

SYLLABUS:

141304 – ANALOG AND DIGITAL COMMUNICATION

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UNIT I FUNDAMENTALS OF ANALOG COMMUNICATION 9

Principles of Amplitude Modulation – AM Envelope – Frequency Spectrum and Bandwidth – Modulation Index and Percent Modulation – AM Voltage Distribution – AM Power Distribution – Angle Modulation – FM and PM Waveforms – Phase Deviation and Modulation Index – Frequency Deviation and Percent Modulation – Frequency Analysis of Angle Modulated Waves – Bandwidth Requirements for Angle Modulated Waves.

UNIT II DIGITAL COMMUNICATION 9

Basics – Shannon Limit for Information Capacity – Digital Amplitude Modulation – Frequency Shift Keying – FSK Bit Rate and Baud – FSK Transmitter – BW Consideration of FSK – FSK Receiver – Phase Shift Keying – Binary Phase Shift Keying – QPSK – Quadrature Amplitude Modulation – Bandwidth Efficiency – Carrier Recovery – Squaring Loop – Costas Loop – DPSK.

UNIT III DIGITAL TRANSMISSION 9

Basics – Pulse Modulation – PCM – PCM Sampling – Sampling Rate – Signal to Quantization Noise Rate – Companding – Analog and Digital – Percentage Error – Delta Modulation – Adaptive Delta Modulation – Differential Pulse Code Modulation – Pulse Transmission – Intersymbol Interference – Eye Patterns.

UNIT IV DATA COMMUNICATIONS 9

Basics – History of Data Communications – Standards Organizations for Data Communication – Data Communication Circuits – Data Communication Codes – Error Control – Error Detection – Error Correction – Data Communication Hardware – Serial and Parallel Interfaces – Data Modems – Asynchronous Modem – Synchronous Modem – Low-Speed Modem – Medium and High Speed Modem – Modem Control.

UNIT V SPREAD SPECTRUM AND MULTIPLE ACCESS TECHNIQUES

9

Basics – Pseudo-Noise Sequence – DS Spread Spectrum with Coherent Binary PSK – Processing Gain – FH Spread Spectrum – Multiple Access Techniques – Wireless Communication – TDMA and CDMA in Wireless Communication Systems – Source Coding of Speech for Wireless Communications.

L: 45 T: 15 Total: 60

TEXT BOOKS

1. Wayne Tomasi, –Advanced Electronic Communication Systems, 6th Edition, Pearson Education, 2007.
2. Simon Haykin, –Communication Systems, 4th Edition, John Wiley and Sons, 2001.

REFERENCES

1. H. Taub, D L Schilling, G Saha , |Principles of Communication|, 3rd Edition, 2007.
2. B. P. Lathi, |Modern Analog and Digital Communication Systems|, 3rd Edition, Oxford University Press, 2007.
3. Blake, –Electronic Communication Systems|, Thomson Delmar Publications, 2002.
4. Martin S. Roden, –Analog and Digital Communication System|, 3rd Edition, PHI, 2002.
5. B. Sklar, |Digital Communication Fundamentals and Applications|, 2nd Edition, Pearson Education, 2007.

LECTURE PLAN

UNIT 1

UNIT I FUNDAMENTALS OF ANALOG COMMUNICATION 9

Principles of Amplitude Modulation – AM Envelope – Frequency Spectrum and Bandwidth – Modulation Index and Percent Modulation – AM Voltage Distribution – AM Power Distribution – Angle Modulation – FM and PM Waveforms – Phase Deviation and Modulation Index – Frequency Deviation and Percent Modulation – Frequency Analysis of Angle Modulated Waves – Bandwidth Requirements for Angle Modulated Waves.

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UNIT1-Introduction

CHAPTER 1: MODULATION SYSTEMS

1. Introduction

a. In the Microbroadcasting services, a reliable radio communication system is of vital importance. The swiftly moving operations of modern communities require a degree of coordination made possible only by radio. Today, the radio is standard equipment in almost all vehicles, and the handie-talkie is a common sight in the populace. Until recently, a-m (amplitude modulation) communication was used universally. This system, however, has one great disadvantage: Random noise and other interference can cripple communication beyond the control of the operator. In the a-m receiver, interference has the same effect on the r-f signal as the intelligence being transmitted because they are of the same nature and inseparable.

b. The engines, generators, and other electrical and mechanical systems of modern vehicles generate noise that can disable the a-m receiver. To avoid this a different type of modulation, such as p-m (phase modulation) or f-m (frequency modulation) is used. When the amplitude of the r-f (radio-frequency) signal is held constant and the intelligence transmitted by varying some other characteristic of the r-f signal, some of the disruptive effects of noise can be eliminated.

c. In the last few years, f-m transmitters and receivers have become standard equipment in America, and their use in mobile equipments exceeds that of a-m transmitters and receivers. The widespread use of frequency modulation means that the technician must be prepared to repair a defective f-m unit, align its tuned circuits, or correct an abnormal condition. To perform these duties, a thorough understanding of frequency modulation is necessary.

2. Carrier Characteristics

The r-f signal used to transmit intelligence from one point to another is called the carrier. It consists of an electromagnetic wave having amplitude, frequency, and phase. If the voltage variations of an r-f signal are graphed in respect to time, the result is a waveform such as that in figure 2. This curve of an unmodulated carrier is the same as those plotted for current or power variations, and it can be used to investigate the general properties of carriers. The unmodulated carrier is a sine wave that repeats itself in definite intervals of time. It swings first in the positive and then in the negative direction about the time axis and represents changes in the amplitude of the wave. This action is similar to that of alternating current in a wire, where these swings represent reversals in the direction of current flow. It must be remembered that the plus and minus signs used in the figure represent direction only. The starting point of the curve in the figure 2 is chosen arbitrarily. It could have been taken at any other point just as well. Once a

starting point is chosen, however, it represents the point from which time is measured. The starting point finds the curve at the top of its positive swing. The curve then swings through 0 to some maximum amplitude in the negative direction, returning through 0 to its original position. The changes in amplitude that take place in the interval of time then are repeated exactly so long as the carrier remains unmodulated. A full set of values occurring in any equal period of time, regardless of the starting point, constitutes one cycle of the carrier. This can be seen in the figure, where two cycles with different starting points are marked off. The number of these cycles that occur in 1 second is called the frequency of the wave.

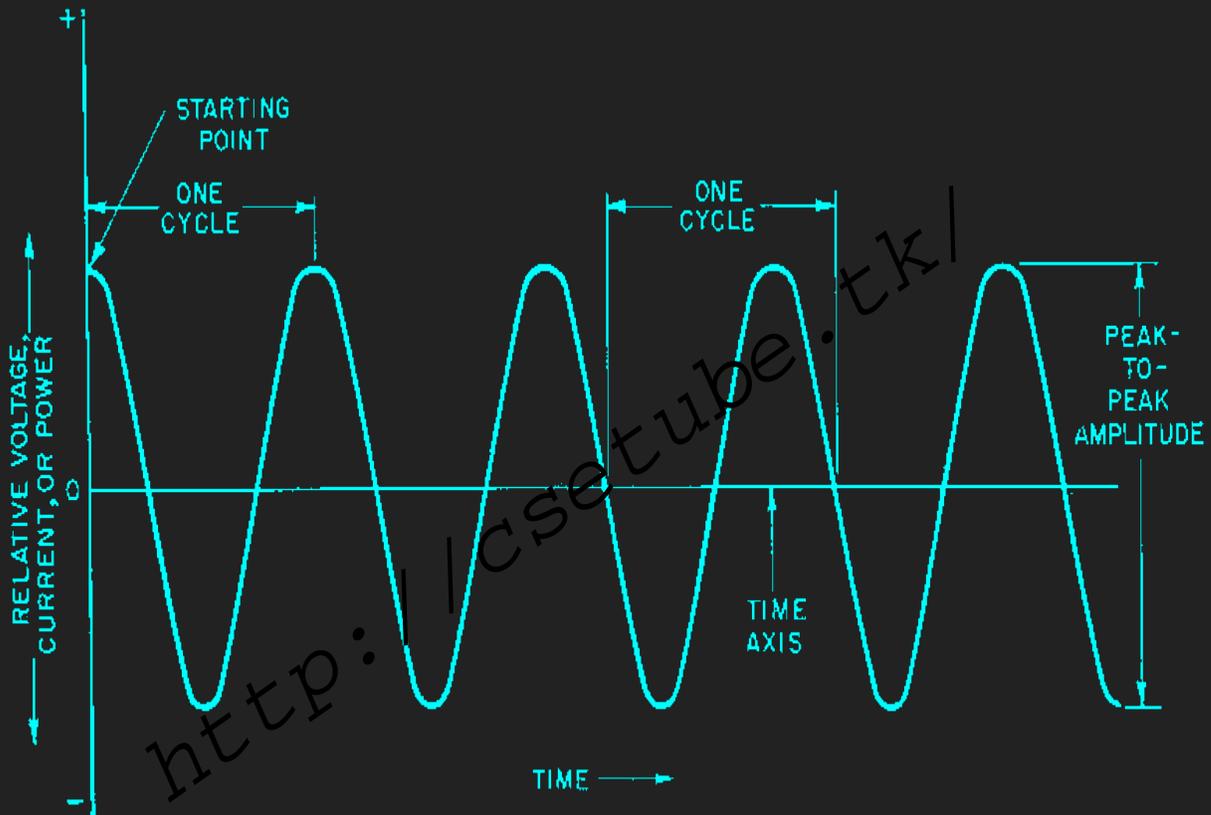


Figure 2. Graph of typical unmodulated carrier.

3. Amplitude Modulation

a. General. The amplitude, phase, or frequency of a carrier can be varied in accordance with the intelligence to be transmitted. The process of varying one of these characteristics is called modulation. The three types of modulation, then are amplitude modulation, phase modulation, and frequency modulation. Other special types, such as pulse modulation, can be considered as subdivisions of these three types. With a sine-wave voltage used to amplitude-modulate the carrier, the instantaneous amplitude of the carrier changes constantly in a sinusoidal manner. The maximum amplitude that the wave reaches in either the positive or the negative direction is termed the peak amplitude. The positive and negative peaks are equal and the full swing of the cycle

from the positive to the negative peak is called the peak-to-peak amplitude. Considering the peak-to-peak amplitude only, it can be said that the amplitude of this wave is constant. This is a general amplitude characteristic of the unmodulated carrier. In amplitude modulation, the peak-to-peak amplitude of the carrier is varied in accordance with the intelligence to be transmitted. For example, the voice picked up by a microphone is converted into an a-f (audio-frequency) electrical signal which controls the peak-to-peak amplitude of the carrier. A single sound at the microphone modulates the carrier, with the result shown in figure 3. The carrier peaks are no longer because they follow the instantaneous changes in the amplitude of the a-f signal. When the a-f signal swings in the positive direction, the carrier peaks are increased accordingly. When the a-f signal swings in the negative direction, the carrier peaks are decreased. Therefore, the instantaneous amplitude of the a-f modulating signal determines the peak-to-peak amplitude of the modulated carrier.

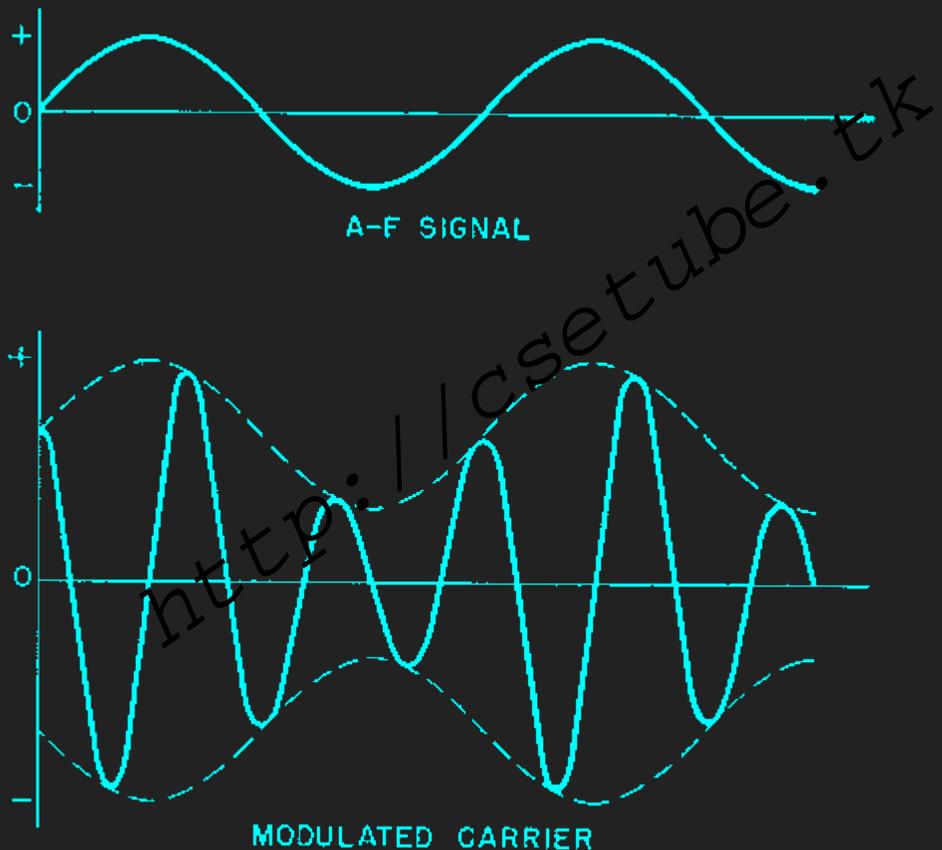


Figure 3. Effect of a-f signal on carrier in amplitude modulation.

b. Percentage of Modulation.

(1) In amplitude modulation, it is common practice to express the degree to which a carrier is modulated as a percentage of modulation. When the peak-to-peak amplitude of the modulating signal is equal to the peak-to-peak amplitude of the unmodulated carrier,

the carrier is said to be 100 percent modulated. In figure 4, the peak-to-peak modulating voltage, E_A , is equal to that of the carrier voltage, E_R , and the peak-to-peak amplitude of the carrier varies from $2E_R$, or $2E_A$, to 0. In other words, the modulating signal swings far enough positive to double the peak-to-peak amplitude of the carrier, and far enough negative to reduce the peak-to-peak amplitude of the carrier to 0.

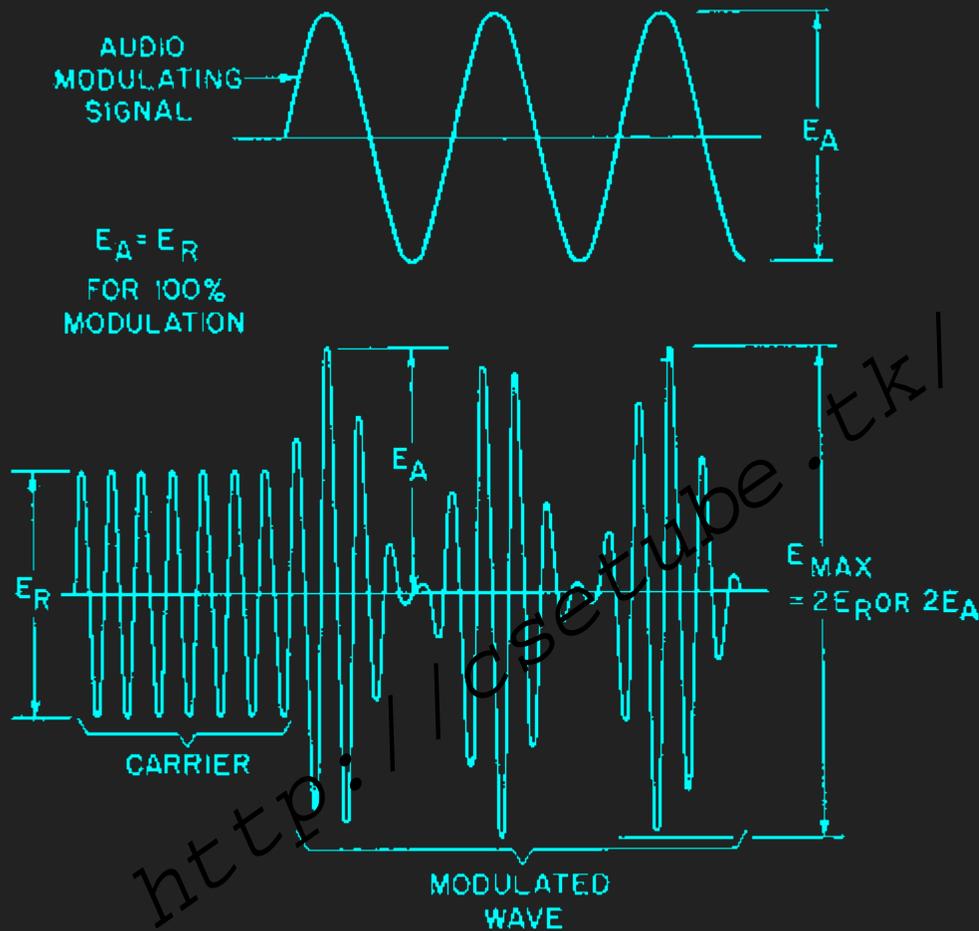


Figure 4. Carrier modulated 100 percent by a modulating signal.

(2) If E_A is less than E_R , percentages of modulation below 100 percent occur. If E_A is one-half E_R , the carrier is modulated only 50 percent (fig. 5). When the modulating signal swings to its maximum value in the positive direction, the carrier amplitude is increased by 50 percent. When the modulating signal reaches its maximum negative peak value, the carrier amplitude is decreased by 50 percent.

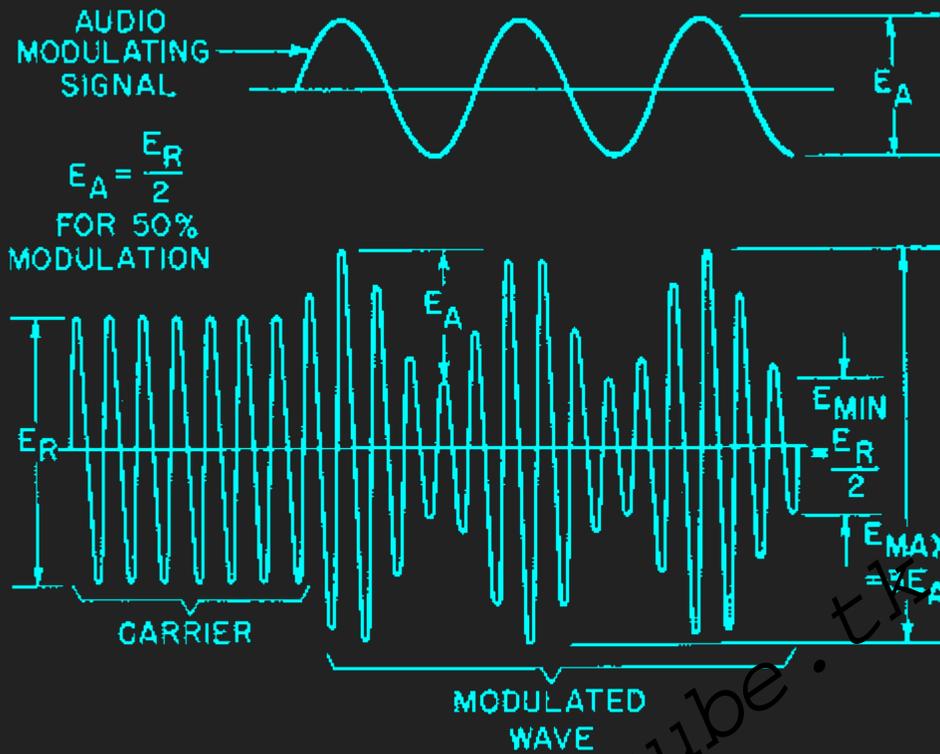


Figure 5. Fifty-percent modulation.

(3) It is possible to increase the percentage of modulation to a value greater than 100 percent by making E_A greater than E_R . In figure 6, the modulated carrier is varied from 0 to some peak-to-peak amplitude greater than $2E_R$. Since the peak-to-peak amplitude of the carrier cannot be less than 0, the carrier is cut off completely for all negative values of E_A greater than E_R . This results in a distorted signal, and the intelligence is received in a distorted form. Therefore, the percentage of modulation in a-m systems of communication is limited to values from 0 to 100 percent.

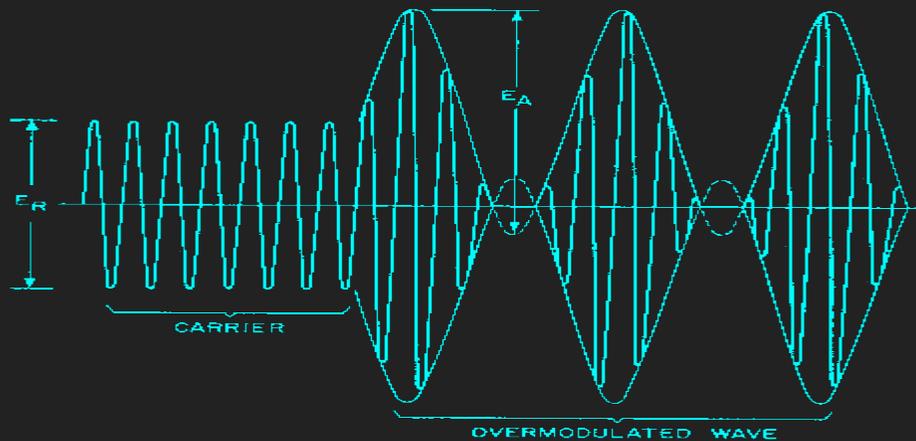


Figure 6. Overmodulation of carrier.

(4) The actual percentage of modulation of a carrier (M) can be calculated by using the following simple formula $M = \text{percentage of modulation} = ((E_{\text{max}} - E_{\text{min}}) / (E_{\text{max}} + E_{\text{min}})) * 100$ where E_{max} is the greatest and E_{min} the smallest peak-to-peak amplitude of the modulated carrier. For example, assume that a modulated carrier varies in its peak-to-peak amplitude from 10 to 30 volts. Substituting in the formula, with E_{max} equal to 30 and E_{min} equal to 10, $M = \text{percentage of modulation} = ((30 - 10) / (30 + 10)) * 100 = (20 / 40) * 100 = 50$ percent. This formula is accurate only for percentages between 0 and 100 percent.

c. Side Bands.

(1) When the outputs of two oscillators beat together, or heterodyne, the two original frequencies plus their sum and difference are produced in the output. This heterodyning effect also takes place between the a-f signal and the r-f signal in the modulation process and the beat frequencies produced are known as side bands. Assume that an a-f signal whose frequency is 1,000 cps (cycles per second) is modulating an r-f carrier of 500 kc (kilocycles). The modulated carrier consists mainly of three frequency components: the original r-f signal at 500 kc, the sum of the a-f and r-f signals at 501 kc, and the difference between the a-f and r-f signals at 499 kc. The component at 501 kc is known as the upper sideband, and the component at 499 kc is known as the lower side band. Since these side bands are always present in amplitude modulation, the a-m wave consists of a center frequency, an upper side-band frequency, and a lower side-band frequency. The amplitude of each of these is constant in value but the resultant wave varies in amplitude in accordance with the audio signal.

(2) The carrier with the two sidebands, with the amplitude of each component plotted against its frequency, is represented in figure 7 for the example given above. The modulating signal, f_A , beats against the carrier, f_C , to produce upper side band f_H and lower side band f_L . The modulated carrier occupies a section of the radio-frequency spectrum extending from f_L to f_H , or 2 kc. To receive this signal, a receiver must have r-f stages whose bandwidth is at least 2 kc. When the receiver is tuned to 500 kc, it also must be able to receive 499 kc and 501 kc with relatively little loss in response.

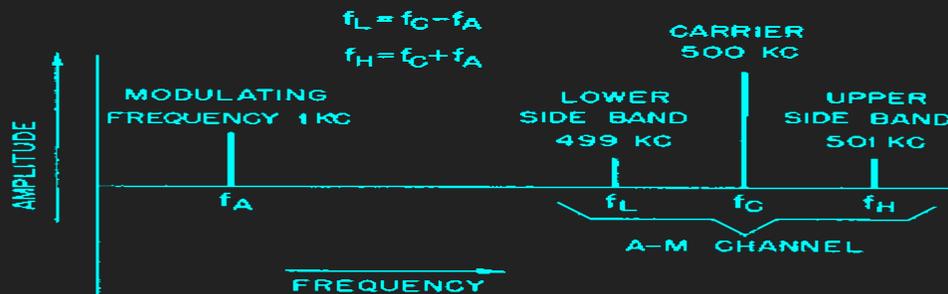


Figure 7. Frequency spectrum of amplitude-modulated wave.

(3) The audio-frequency range extends approximately from 16 to 16,000 cps. To accommodate the highest audio frequency, the a-m frequency channel should extend from 16 kc below to 16 kc above the carrier frequency, with the receiver having a

corresponding bandwidth. Therefore, if the carrier frequency is 500 kc, the a-m channel should extend from 484 to 516 kc. This bandwidth represents an ideal condition; in practice, however, the entire a-m bandwidth for audio reproduction rarely exceeds 16 kc. For any specific set of audio-modulating frequencies, the a-m channel or bandwidth is twice the highest audio frequency present.

(4) The r-f energy radiated from the transmitter antenna in the form of a modulated carrier is divided among the carrier and its two side bands. With a carrier component of 1,000 watts, an audio signal of 500 watts is necessary for 100-percent modulation. Therefore, the modulated carrier should not exceed a total power of 1,500 watts. The 500 watts of audio power is divided equally between the side bands, and no audio power is associated with the carrier.

(5) Since none of the audio power is associated with the carrier component, it contains none of the intelligence. From the standpoint of communication efficiency, the 1,000 watts of carrier-component power is wasted. Furthermore, one side band alone is sufficient to transmit intelligence. It is possible to eliminate the carrier and one side band, but the complexity of the equipment needed cancels the gain in efficiency.

d. Disadvantages of Amplitude Modulation. It was noted previously that random noise and electrical interference can amplitude-modulate the carrier to the extent that communication cannot be carried on. From the military standpoint, however, susceptibility to noise is not the only disadvantage of amplitude modulation. An a-m signal is also susceptible to enemy jamming and to interference from the signals of transmitters operating on the same or adjacent frequencies. Where interference from another station is present, the signal from the desired station must be many times stronger than the interfering signal. For various reasons, the choice of a different type of modulation seems desirable.

4. Phase Modulation.

a. General.

(1) Besides its amplitude, the frequency or phase of the carrier can be varied to produce a signal bearing intelligence. The process of varying the frequency in accordance with the intelligence is frequency modulation, and the process of varying the phase is phase modulation. When frequency modulation is used, the phase of the carrier wave is indirectly affected. Similarly, when phase modulation is used, the carrier frequency is affected. Familiarity with both frequency and phase modulation is necessary for an understanding of either.

(2) In the discussion of carrier characteristics, carrier frequency was defined as the number of cycles occurring in each second. Two such cycles of a carrier are represented by curve A in figure 8. The starting point for measuring time is chosen arbitrarily, and at 0 time, curve A has some negative value. If another curve B, of the same frequency is drawn having 0 amplitude at 0 time, it can be used as a reference in describing curve A.

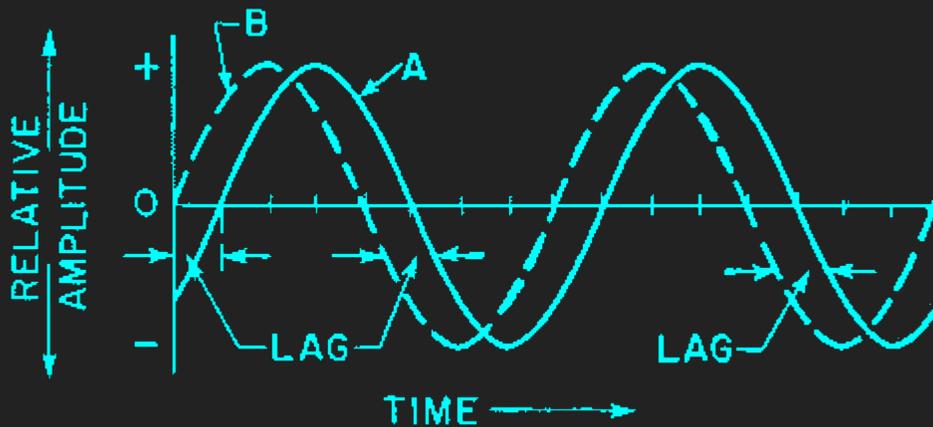


Figure 8. Determining relative phase from a curve of the same frequency.

(3) Curve B starts at 0 and swings in the positive direction. Curve A starts at some negative value and also swings in the positive direction, not reaching 0 until a fraction of a cycle after curve B has passed through 0. This fraction of a cycle is the amount by which A is said to lag B. Because the two curves have the same frequency, A will always lag B by the same amount. If the positions of the two curves are reversed, then A is said to lead B. The amount by which A leads or lags the reference is called its phase. Since the reference given is arbitrary, the phase is relative.

c. Phase Modulation.

(1) In phase modulation, the relative phase of the carrier is made to vary in accordance with the intelligence to be transmitted. The carrier phase angle, therefore, is no longer fixed. The amplitude and the average frequency of the carrier are held constant while the phase at any instant is being varied with the modulating signal (fig. 11). Instead of having the vector rotate at the carrier frequency, the axes of the graph can be rotated in the opposite direction at the same speed. In this way the vector (and the reference) can be examined while they are standing still. In A of figure 11 the vector for the unmodulated carrier is given, and the smaller curved arrows indicate the direction of rotation of the axes at the carrier frequency. The phase angle, ϕ , is constant in respect to the arbitrarily chosen reference. Effects of the modulating signal on the relative phase angle at four different points are illustrated in B, C, D, and E.

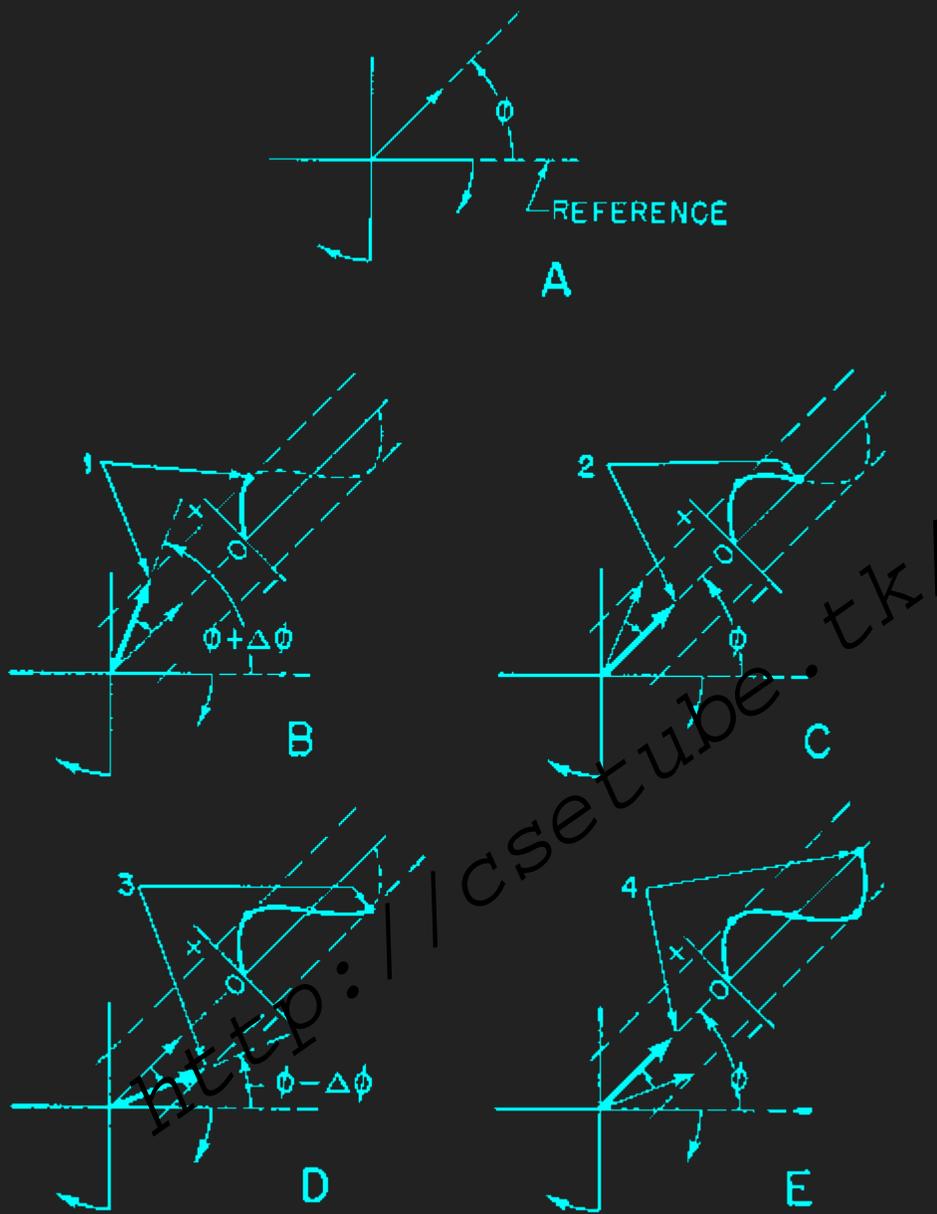


Figure 11. Successive vector representation of a phase-modulated carrier.

(2) The effect of a positive swing of the modulating signal is to speed the rotation of the vector, moving it counterclockwise and increasing the phase angle, ϕ . At point 1, the modulating signal reaches its maximum positive value, and the phase has been changed by the amount $\Delta\phi$. The instantaneous phase condition at 1 is, therefore, $(\phi + \Delta\phi)$. Having reached its maximum value in the positive direction, the modulating signal swings in the opposite direction. The vector speed is reduced and it appears to move in the reverse direction, moving towards its original position.

(3) For each cycle of the modulating signal, the relative phase of the carrier is varied between the values of $(\phi + \Delta\phi)$ and $(\phi - \Delta\phi)$. These two values of instantaneous phase, which occur at the maximum positive and maximum negative values of modulation, are known as the phase-deviation limits. The upper limit is $(\phi + \Delta\phi)$ the lower limit is $(\phi - \Delta\phi)$. The relations between the phase-deviation limits and the carrier vector are given in the figure 12, with the limits of $\pm \Delta\phi$ indicated.

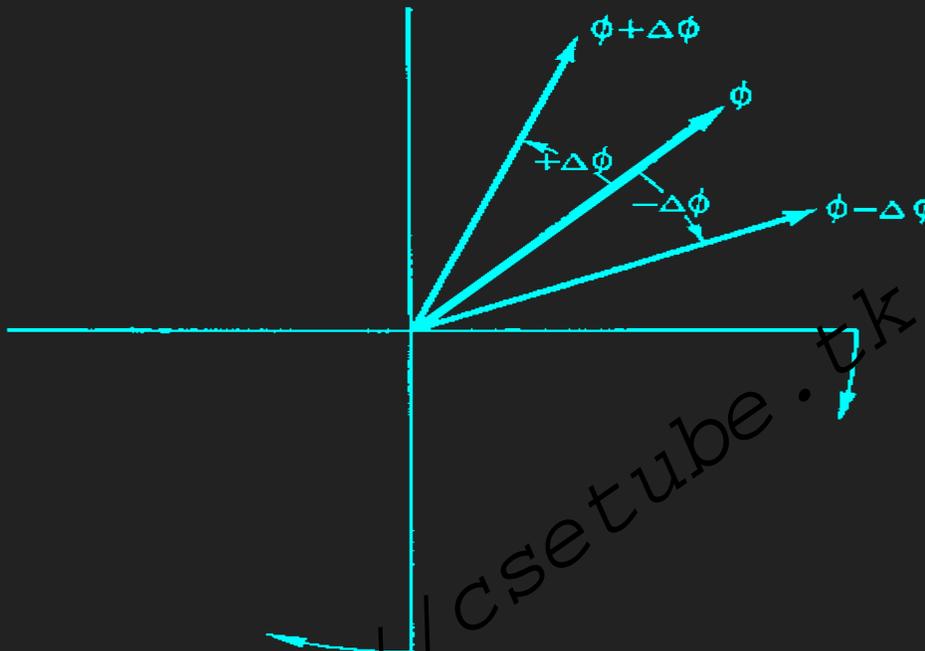


Figure 12. Phase-deviation limits of a modulated carrier.

(4) If the phase-modulated vector is plotted against time, the result is the wave illustrated in the figure 13. The modulating signal is shown in A. The dashed-line waveform, in B, is the curve of the reference vector and the solid-line waveform is the carrier. As the modulating signal swings in the positive direction, the relative phase angle is increased from an original phase lead of 45° to some maximum, as shown at 1 in B. When the signal swings in the negative direction, the phase lead of the carrier over the reference vector is decreased to minimum value, as shown at 2; it then returns to the original 45° phase lead when the modulating signal swings back to 0. This is the basic resultant wave for sinusoidal phase modulation, with the amplitude of the modulating signal controlling the relative phase characteristic of the carrier.

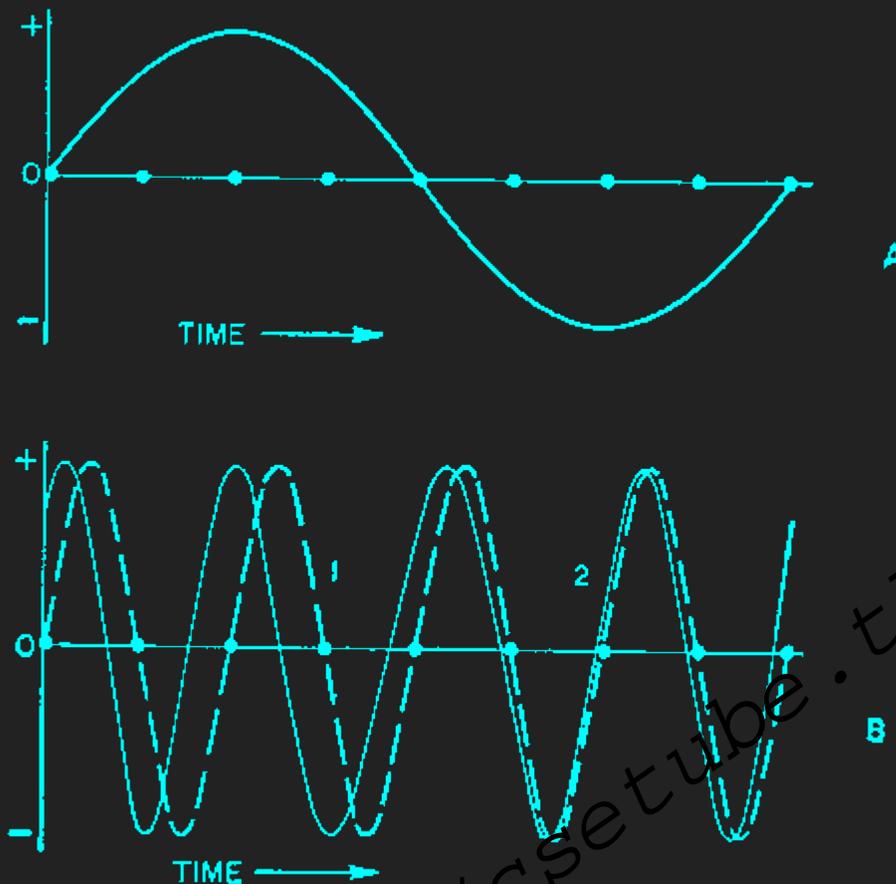


Figure 13. Phase-modulated carrier over 1 cycle of modulating signal.

d. P-M and Carrier Frequency.

(1) In the vector representation of the p-m carrier, the carrier vector is speeded up or slowed down as the relative phase angle is increased or decreased by the modulating signal. Since vector speed is the equivalent of carrier frequency, the carrier frequency must change during phase modulation. A form of frequency modulation, known as equivalent f-m, therefore, takes place. Both the p-m and the equivalent f-m depend on the modulating signal, and an instantaneous equivalent frequency is associated with each instantaneous phase condition.

(2) The phase at any instant is determined by the amplitude of the modulating signal. The instantaneous equivalent frequency is determined by the rate of change in the amplitude of the modulating signal. The rate of change in modulating -signal amplitude depends on two factors -- the modulation amplitude and the modulation frequency. If the amplitude is increased, the phase deviation is increased. The carrier vector must move through a greater angle in the same period of time, increasing its speed, and thereby increasing the carrier frequency shift. If the modulation frequency is increased, the

carrier must move within the phase-deviation limits at a faster rate, increasing its speed and thereby increasing the carrier frequency shift. When the modulating-signal amplitude or frequency is decreased, the carrier frequency shift is decreased also. The faster the amplitude is changing, the greater the resultant shift in carrier frequency; the slower the change in amplitude, the smaller the frequency shift.

(3) The rate of change at any instant can be determined by the slope, or steepness, of the modulation waveform. As shown by curve A in figure 14, the greatest rates of change do not occur at points of maximum amplitude; in fact, when the amplitude is 0 the rate of change is maximum, and when the amplitude is maximum the rate of change is 0. When the waveform passes through 0 in the positive direction, the rate of change has its maximum positive value; when the waveform passes through 0 in the negative direction, the rate of change is a maximum negative value.

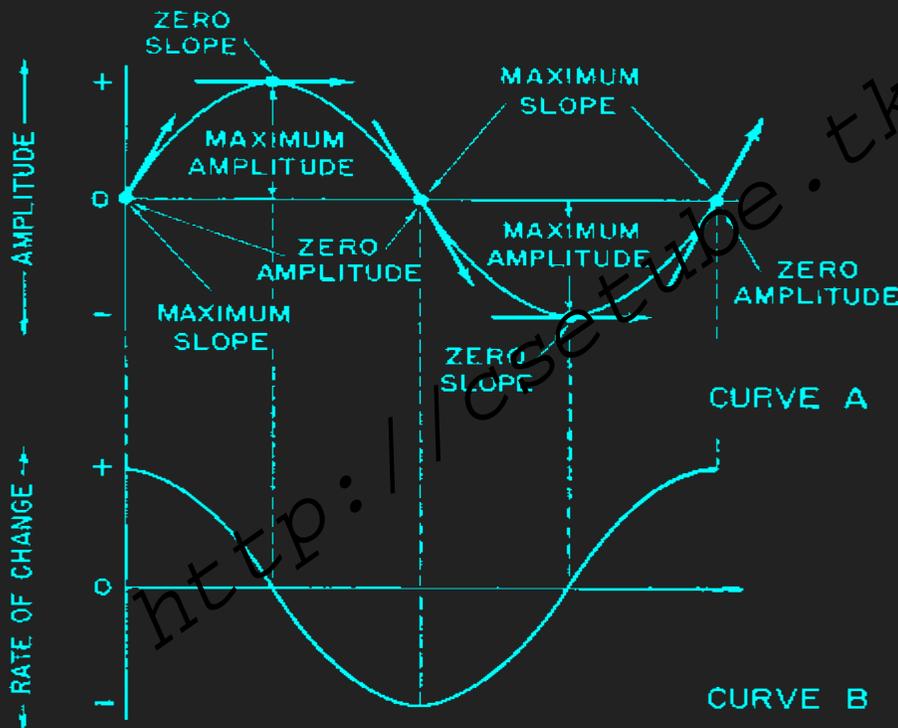


Figure 14. Amplitude compared to rate of change of a signal.

(4) Curve B is a graph of the rate of change of curve A. This waveform is leading A by 90°. This means that the frequency deviation resulting from phase modulation is 90° out of phase with the phase deviation. The relation between phase deviation and frequency shift is shown by the vectors in figure 15. At times of maximum phase deviation, the frequency shift is 0; at times of 0 phase deviation, the frequency shift is maximum. The equivalent-frequency deviation limits of the phase-modulated carrier can be calculated by means of the formula, $\Delta F = \Delta \phi f \cos(2\phi f t)$ where ΔF is the frequency deviation, $\Delta \phi$ is the maximum phase deviation, f is the modulating-signal frequency, $\cos(2\phi f t)$ is the

amplitude variation of the modulating signal at any time, t . When $(2 \pi f t)$ is 0 or 180° , the signal amplitude is 0 and the cosine has maximum values of $+1$ at 360° and -1 at 180° . If the phase deviation limit is 30° , or $\frac{\pi}{6}$ radians, and a $1,000$ -cps signal modulates the carrier, then $F = (1/\frac{\pi}{6}) * 1000 * \pm 1$, $F = \pm 523$ cps, approximately. When the modulating signal is passing through 0 in the positive direction, the carrier frequency is raised by 523 cps. When the modulating signal is passing through 0 in the negative direction, the carrier frequency is lowered by 523 cps.

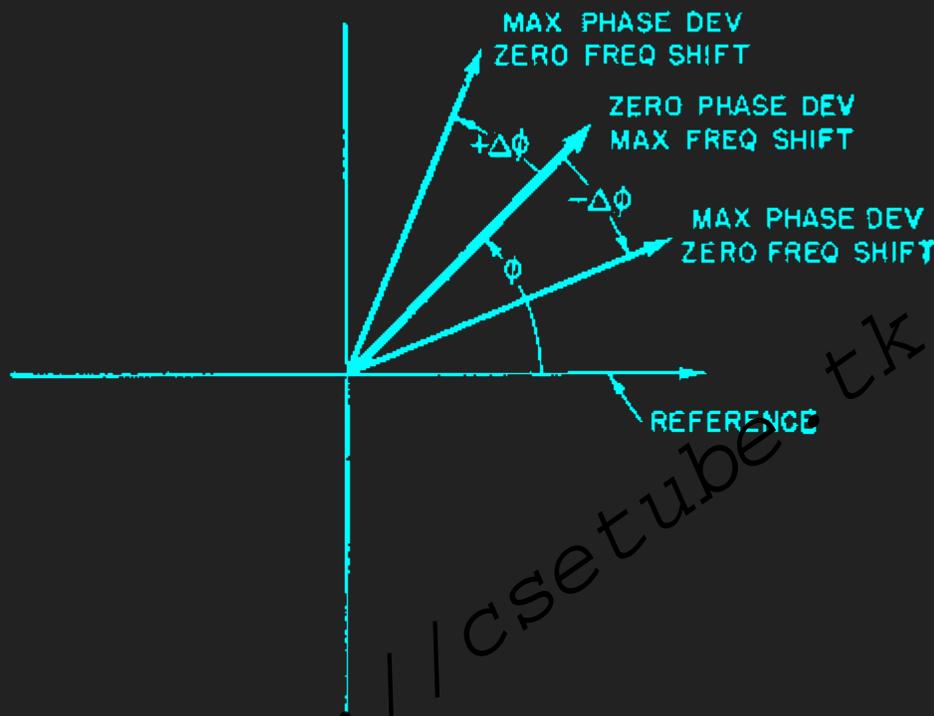


Figure 16. Phase deviation and frequency shift.

5. Frequency Modulation

a. When a carrier is frequency-modulated by a modulating signal, the carrier amplitude is held constant and the carrier frequency varies directly as the amplitude of the modulating signal. There are limits of frequency deviation similar to the phase-deviation limits in phase modulation. There is also an equivalent phase shift of the carrier, similar to the equivalent frequency shift in p-m.

b. A frequency-modulated wave resulting from 2 cycles of modulating signal imposed on a carrier is shown in A of figure 16. When the modulating-signal amplitude is 0 , the carrier frequency does not change. As the signal swings positive, the carrier frequency is increased, reaching its highest frequency at the positive peak of the modulating signal. When the signal swings in the negative direction, the carrier frequency is lowered, reaching a minimum when the signal passes through its peak negative value. The f-m wave can be compared with the p-m wave, in B, for the same 2 cycles of modulation.

signal. If the p-m wave is shifted 90° , the two waves look alike. Practically speaking, there is little difference, and an f-m receiver accepts both without distinguishing between them. Direct phase modulation has limited use, however, and most systems use some form of frequency modulation.

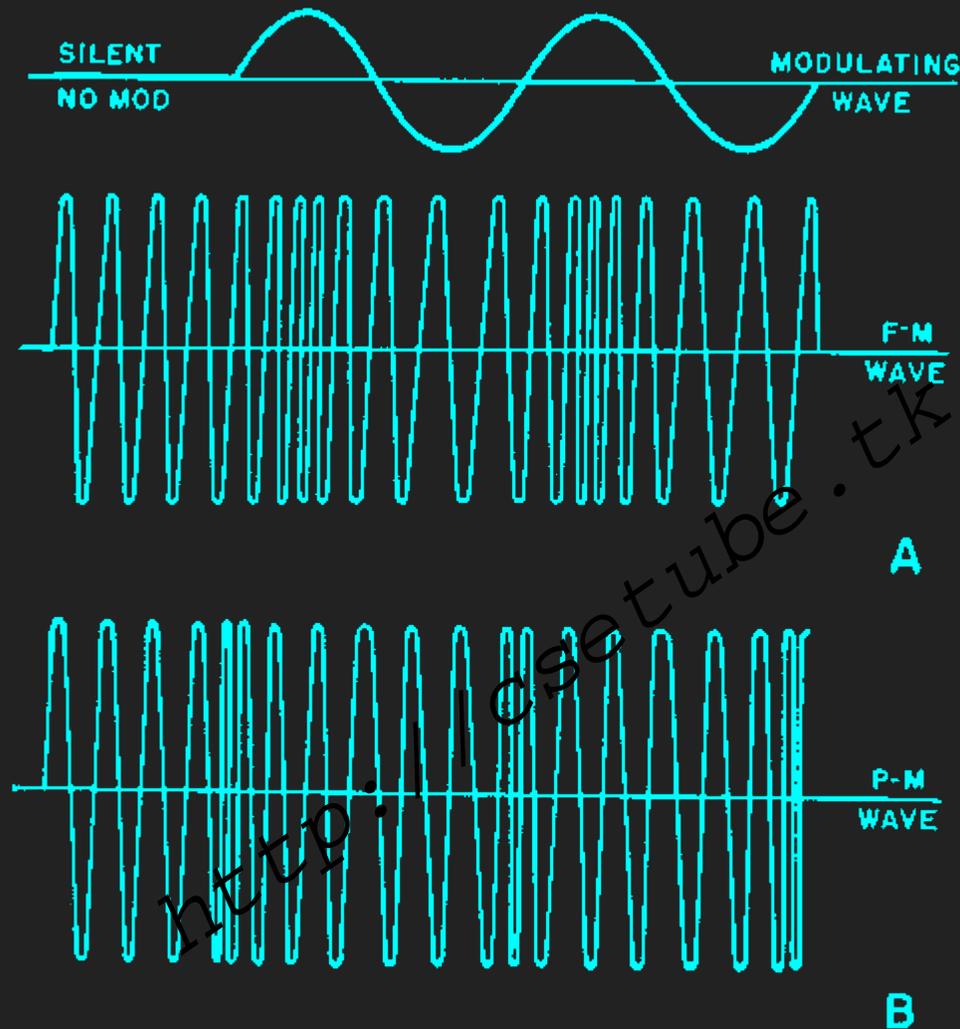


Figure 18. Comparison of f-m and p-m signals.

6. A-M, P-M, and F-M Transmitters

a. General. All f-m transmitters use either direct or indirect methods for producing f-m. The modulating signal in the direct method has a direct effect on the frequency of the carrier; in the indirect method, the modulating signal uses the frequency variations caused by phase-modulation. In either case, the output of the transmitter is a frequency-modulated wave, and the f-m receiver cannot distinguish between them.

b. A-M Transmitter.

(1) In the block diagram of the a-m transmitter (A of fig. 17), the r-f section consists of an oscillator feeding a buffer, which in turn feeds a system of frequency multipliers and/or intermediate power amplifiers. If frequency multiplication is unnecessary, the buffer feeds directly into the intermediate power amplifiers which, in turn, drive the final power amplifier. The input to the antenna is taken from the final power amplifier.

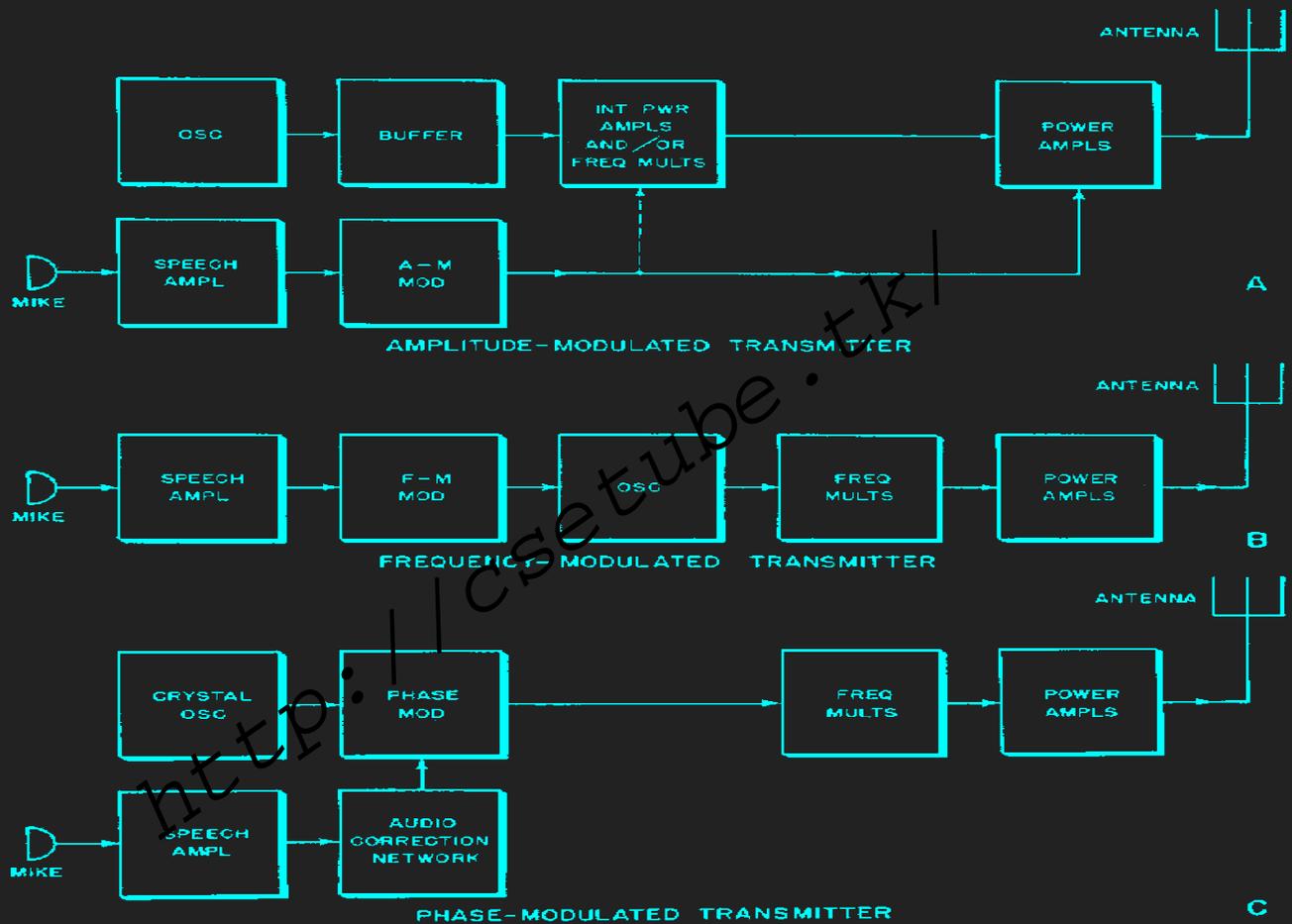


Figure 17. Basic a-m, p-m, and f-m transmitters.

(2) The audio system consists of a microphone which feeds a speech amplifier. The output of this speech amplifier is fed to a modulator. For high-level modulation, the output of the modulator is connected to the final amplifier (solid arrow), where its amplitude modulates the r-f carrier. For low-level modulation, the output of the modulator is fed to the intermediate power amplifier (dashed arrow). The power required in a-m transmission for either high- or low-level modulation is much greater than that required for f-m or p-m.

c. P-M Transmitter. In the p-m, or indirect f-m, transmitter, the modulating signal is passed through some type of correction network before reaching the modulator, as in

C. When comparing the p-m to the f-m wave, it was pointed out that a phase shift of 90° in the p-m wave made it impossible to distinguish it from the f-m wave (fig. 16). This phase shift is accomplished in the correction network. The output of the modulator which is also fed by a crystal oscillator is applied through frequency multipliers and a final power amplifier just as in the direct f-m transmitter. The final output is an f-m wave.

d. F-M Transmitter. In the f-m transmitter, the output of the speech amplifier usually is connected directly to the modulator stage, as in B. The modulator stage supplies an equivalent reactance to the oscillator stage that varies with the modulating signal. This causes the frequency of the oscillator to vary with the modulating signal. The frequency-modulated output of the oscillator then is fed to frequency multipliers which bring the frequency of the signal to the required value for transmission. A power amplifier builds up the signal before it is applied to the antenna.

e. Comparisons.

(1) The primary difference between the three transmitters lies in the method used to vary the carrier. In a-m transmission, the modulating signal controls the amplitude of the carrier. In f-m transmission, the modulating signal controls the frequency of the oscillator. In p-m transmission, the modulating signal controls the frequency of the oscillator output. In p-m, or indirect f-m, transmission, the modulating signal controls the phase of a fixed-frequency oscillator. The r-f sections of these transmitters function in much the same manner, although they may differ appreciably in construction.

(2) The frequency multipliers used in a-m transmitters are used to increase the fundamental frequency of the oscillator. This enables the oscillator to operate at low frequencies, where it has increased stability. In f-m and p-m transmitters, the frequency multipliers not only increase the frequency of transmission, but also increase the frequency deviation caused by the modulating signal.

(3) In all three transmitters, the final power amplifier is used chiefly to increase the power of the modulated signal. In high-level a-m modulation, the final stage is modulated, but this is never done in either f-m or p-m.

7. A-M and F-M Receivers

a. General. The only difference between the a-m superhetrodyne and the two basic types of f-m superhetrodyne receivers (fig. 18) is in the detector circuit used to recover the modulation. In the a-m system, in A, the i-f signal is rectified and filtered, leaving only the original modulation signal. In the f-m system, the frequency variations of the signal must be transformed into amplitude variations before they can be used.

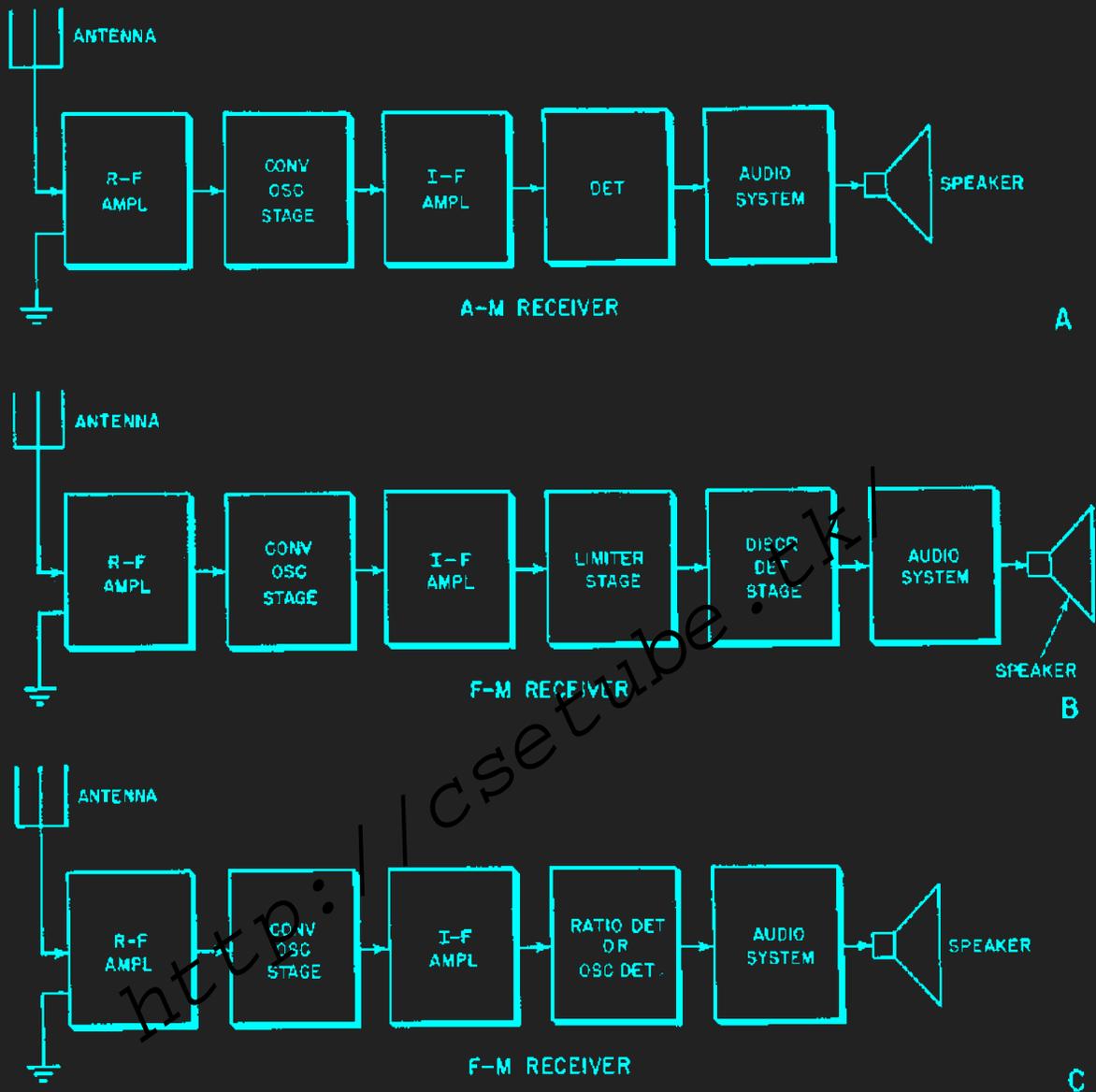


Figure 18. Basic a-m and f-m receivers.

b. F-M Receiver. In the limiter-discriminator detector, in B, the f-m signal is amplitude-limited to remove any variations caused by noise or other disturbances. This signal is then passed through a discriminator which transmorms the frequency variations to corresponding voltage amplitude variation. These voltage variations reproduce the original modulating signal. Two other types of f-m single-stage detectors in general use are the ratio detector and the oscillator detector, shown in C.

AM Reception :

or an L.C. filter. This means if the bottom of your page was 20 KHz wide then the middle half of the top of the page would be 10 KHz wide and this would be considered good!

Back to T.R.F. Receivers - their shape factors were nothing like this. Instead of being shaped like a page they tended to look more like a flat sand hill. The reason for this is it is exceedingly difficult or near impossible to build LC Filters with impressive channel spacing and shape factors at frequencies as high as the broadcast band. And this was in the days when the short wave bands (much higher in frequencies) were almost unheard of. Certain embellishments such as the regenerative detector were developed but they were mostly unsatisfactory.

In the 1930's Major Armstrong developed the superhetrodyne principle.

3. A superhetrodyne receiver works on the principle the receiver has a local oscillator called a variable frequency oscillator or V.F.O.

This is a bit like having a little transmitter located within the receiver. Now if we still have our T.R.F. stages but then mix the received signal with our v.f.o. we get two other signals. (V.F.O. + R.F) and (V.F.O. - R.F).

In a traditional a.m. radio where the received signal is in the range 540 KHz to 1650 KHz the v.f.o. signal is always a constant 455 KHz higher or 995 KHz to 2105 KHz.

Several advantages arise from this and we will use our earlier example of the signal of 540 KHz:

(a) The input signal stages tune to 540 KHz. The adjacent channels do not matter so much now because the only signal to discriminate against is called the i.f. image. At 540 KHz the v.f.o. is at 995 KHz giving the constant difference of 455 KHz which is called the I.F. frequency. However a received frequency of v.f.o. + i.f. will also result in an i.f. frequency, i.e. $995 \text{ KHz} + 455 \text{ KHz} = 1450 \text{ KHz}$, which is called the i.f. image.

Put another way, if a signal exists at 1450 KHz and mixed with the vfo of 995 KHz we still get an i.f. of $1450 - 995 = 455 \text{ KHz}$. Double signal reception. Any reasonable tuned circuit designed for 540 KHz should be able to reject signals at 1450 KHz. And that is now the sole purpose of the r.f. input stage.

(b) At all times we will finish up with an i.f. signal of 455 KHz. It is relatively easy to design stages to give constant amplification, reasonable bandwidth and reasonable shape factor at this one constant frequency. Radio design became somewhat simplified but of course not without its associated problems.

We will now consider these principles in depth by discussing a fairly typical a.m. transistor radio of the very cheap variety.

THE SUPERHETRODYNE TRANSISTOR RADIO

I have chosen to begin radio receiver design with the cheap am radio because:

(a) nearly everyone either has one or can buy one quite cheaply. Don't buy an A.M. / F.M. type because it will only confuse you in trying to identify parts. Similarly don't get one of the newer I.C. types.

Just a plain old type probably with at least 3 transformers. One "red" core and the others likely "yellow" and "black" or "white". Inside will be a battery compartment, a little speaker, a circuit board with weird looking components, a round knob to control volume.

(b) most receivers will almost certainly for the most part follow the schematic diagram I have set out below (there are no limits to my talents - what a clever little possum I am).

(c) if I have included pictures you know I was able to borrow either a digital camera or had access to a scanner.

Important NOTE: If you can obtain discarded "tranny's" (Australian for transistorised am radio receiver) by all means do so because they are a cheap source of valuable parts. So much so that to duplicate the receiver as a kit project for learning purposes costs about \$A70 or \$US45. Incredible. That is why colleges in Australia and elsewhere can not afford to present one as a kit.

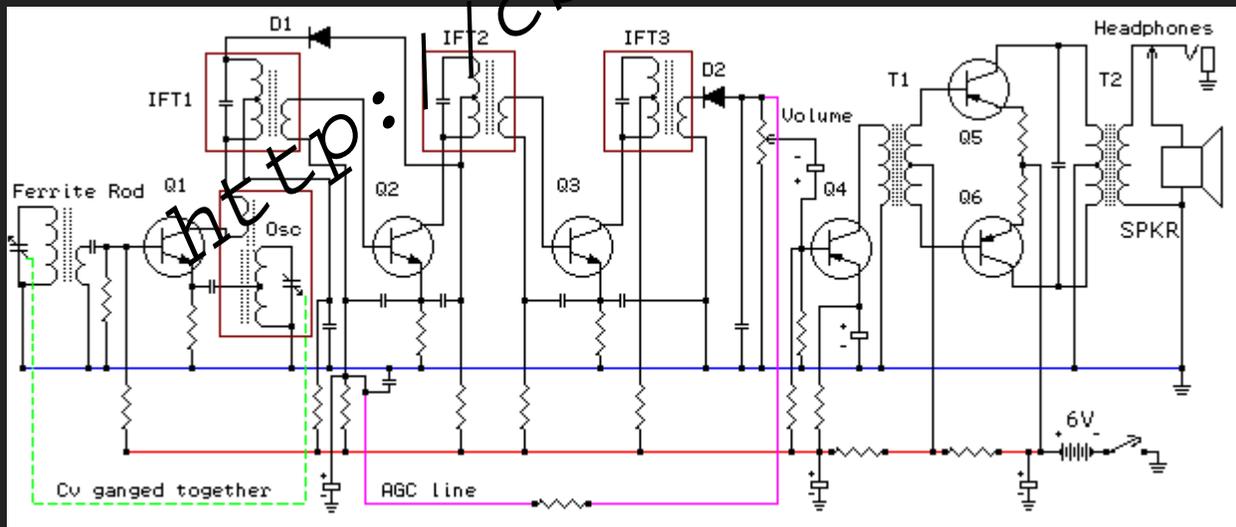
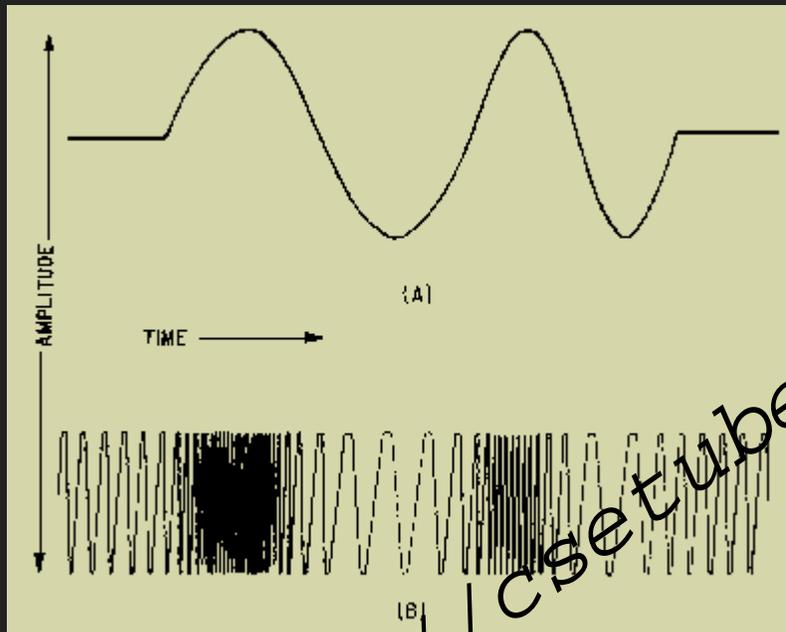


Fig 1 - a.m. bcb radio schematic

Now that's about as simple as it gets. Alright get up off the floor. You will be amazed just how you will be able to understand all this fairly soon.

those illustrated in view (B). The number of times per second that the instantaneous frequency is varied from the average (carrier frequency) is controlled by the frequency of the modulating signal. The amount by which the frequency departs from the average is controlled by the amplitude of the modulating signal. This variation is referred to as the **FREQUENCY DEVIATION** of the frequency-modulated wave. We can now establish two clear-cut rules for frequency deviation rate and amplitude in frequency modulation:

Figure 2-5. - Effect of frequency modulation on an rf carrier.



AMOUNT OF FREQUENCY SHIFT IS PROPORTIONAL TO THE AMPLITUDE OF THE MODULATING SIGNAL

(This rule simply means that if a 10-volt signal causes a frequency shift of 20 kilohertz, then a 20-volt signal will cause a frequency shift of 40 kilohertz.)

RATE OF FREQUENCY SHIFT IS PROPORTIONAL TO THE FREQUENCY OF THE MODULATING SIGNAL

(This second rule means that if the carrier is modulated with a 1-kilohertz tone, then the carrier is changing frequency 1,000 times each second.)

Figure 2-6 illustrates a simple oscillator circuit with the addition of a condenser microphone (M) in shunt with the oscillator tank circuit. Although the condenser microphone capacitance is actually very low, the capacitance of this microphone will be considered near that of the tuning capacitor (C). The frequency of oscillation in this circuit is, of course, determined by the LC product of all elements of the circuit; but, the

product of the inductance (L) and the combined capacitance of C and M are the primary frequency components. When no sound waves strike M, the frequency is the rf carrier frequency. Any excitation of M will alter its capacitance and, therefore, the frequency of the oscillator circuit. Figure 2-7 illustrates what happens to the capacitance of the microphone during excitation. In view (A), the audio-frequency wave has three levels of intensity, shown as X, a whisper; Y, a normal voice; and Z, a loud voice. In view (B), the same conditions of intensity are repeated, but this time at a frequency twice that of view (A). Note in each case that the capacitance changes both positively and negatively; thus the frequency of oscillation alternates both above and below the resting frequency. The amount of change is determined by the change in capacitance of the microphone. The change is caused by the amplitude of the sound wave exciting the microphone. The rate at which the change in frequency occurs is determined by the rate at which the capacitance of the microphone changes. This rate of change is caused by the frequency of the sound wave. For example, suppose a 1,000-hertz tone of a certain loudness strikes the microphone. The frequency of the carrier will then shift by a certain amount, say plus and minus 40 kilohertz. The carrier will be shifted 1,000 times per second. Now assume that with its loudness unchanged, the frequency of the tone is changed to 4,000 hertz. The carrier frequency will still shift plus and minus 40 kilohertz; but now it will shift at a rate of 4,000 times per second. Likewise, assume that at the same loudness, the tone is reduced to 200 hertz. The carrier will continue to shift plus and minus 40 kilohertz, but now at a rate of 200 times per second. If the loudness of any of these modulating tones is reduced by one-half, the frequency of the carrier will be shifted plus and minus 20 kilohertz. The carrier will then shift at the same rate as before. This fulfills all requirements for frequency modulation. Both the frequency and the amplitude of the modulating signal are translated into variations in the frequency of the rf carrier.

Figure 2-6. - Oscillator circuit illustrating frequency modulation.

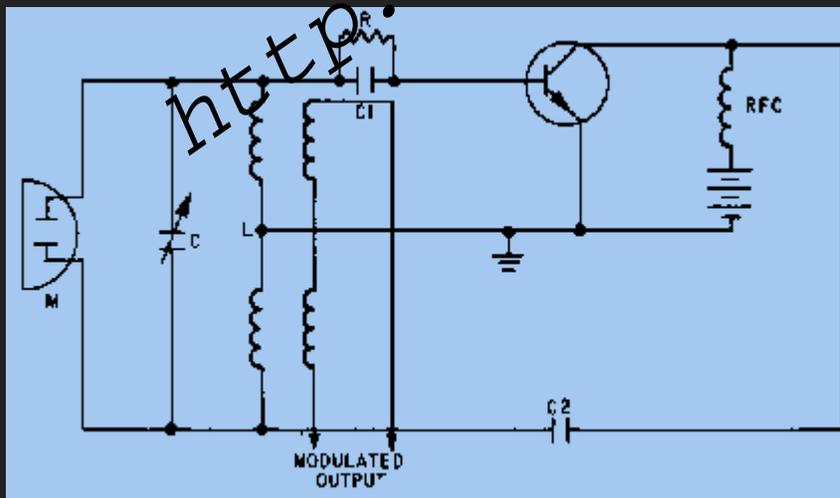


Figure 2-7A. - Capacitance change in an oscillator circuit during modulation. CHANGE IN INTENSITY OF SOUND WAVES CHANGES CAPACITY

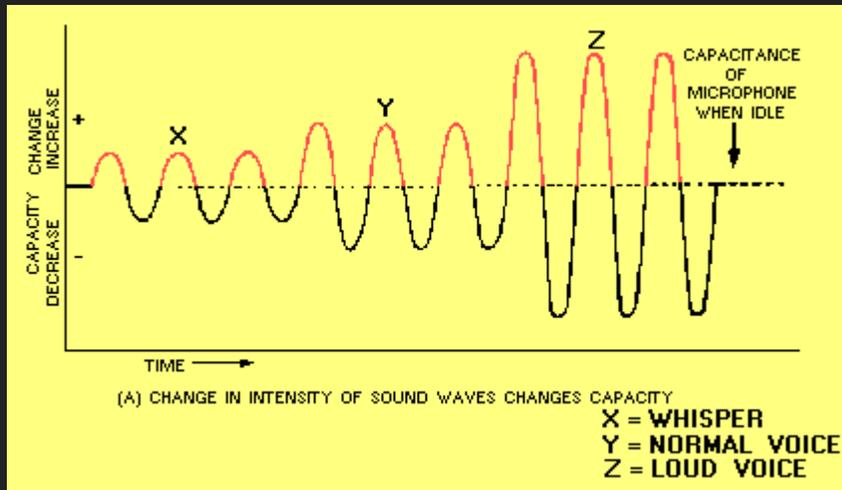


Figure 2-7B. - Capacitance change in an oscillator circuit during modulation. AT A FREQUENCY TWICE THAT OF (A), THE CAPACITY CHANGES THE SAME AMOUNT, BUT TWICE AS OFTEN

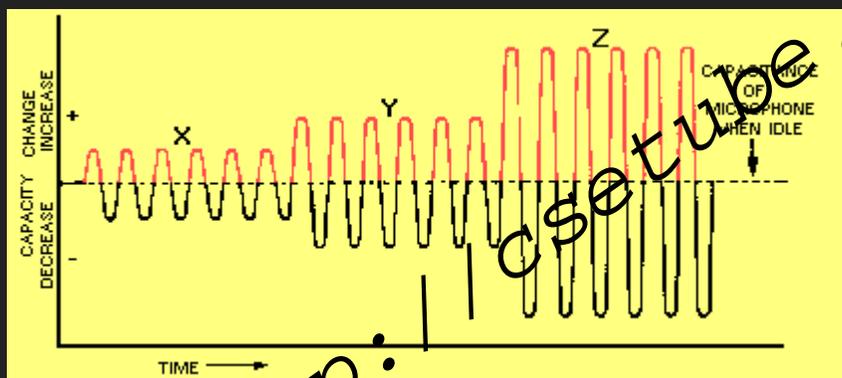


Figure 2-8 shows how the frequency shift of an fm signal goes through the same variations as does the modulating signal. In this figure the dimension of the constant amplitude is omitted. (As these remaining waveforms are presented, be sure you take plenty of time to study and digest what the figures tell you. Look each one over carefully, noting everything you can about them. Doing this will help you understand this material.) If the maximum frequency deviation is set at 75 kilohertz above and below the carrier, the audio amplitude of the modulating wave must be so adjusted that its peaks drive the frequency only between these limits. This can then be referred to as 100-PERCENT MODULATION, although the term is only remotely applicable to fm. Projections along the vertical axis represent deviations in frequency from the resting frequency (carrier) in terms of audio amplitude. Projections along the horizontal axis represent time. The distance between A and B represents 0.001 second. This means that carrier deviations from the resting frequency to plus 75 kilohertz, then to minus 75 kilohertz, and finally back to rest would occur 1,000 times per second. This would equate to an audio frequency of 1,000 hertz. Since the carrier deviation for this period (A to B) extends to the full allowable limits of plus and minus 75 kilohertz, the wave is fully modulated. The distance from C to D is the same as that from A to B, so the time interval and frequency

are the same as before. Notice, however, that the amplitude of the modulating wave has been decreased so that the carrier is driven to only plus and minus 37.5 kilohertz, one-half the allowable deviation. This would correspond to only 50-percent modulation if the system were AM instead of fm. Between E and F, the interval is reduced to 0.0005 second. This indicates an increase in frequency of the modulating signal to 2,000 hertz. The amplitude has returned to its maximum allowable value, as indicated by the deviation of the carrier to plus and minus 75 kilohertz. Interval G to H represents the same frequency at a lower modulation amplitude (66 percent). Notice the GUARD BANDS between plus and minus 75 kilohertz and plus and minus 100 kilohertz. These bands isolate the modulation extremes of this particular channel from that of adjacent channels.

PERCENT OF MODULATION. - Before we explain 100-percent modulation in an fm system, let's review the conditions for 100-percent modulation of an AM wave. Recall that 100-percent 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-percent modulation 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 MODULATION INDEX is more useful in fm modulation discussions. Modulation index is frequency deviation divided by the frequency of the modulating signal.

MODULATION INDEX. - This ratio of frequency deviation to frequency of the modulating signal is useful because it also describes the ratio of amplitude to tone for the audio signal. These factors determine the number and spacing of the side frequencies of the transmitted signal. The modulation index formula is shown below:

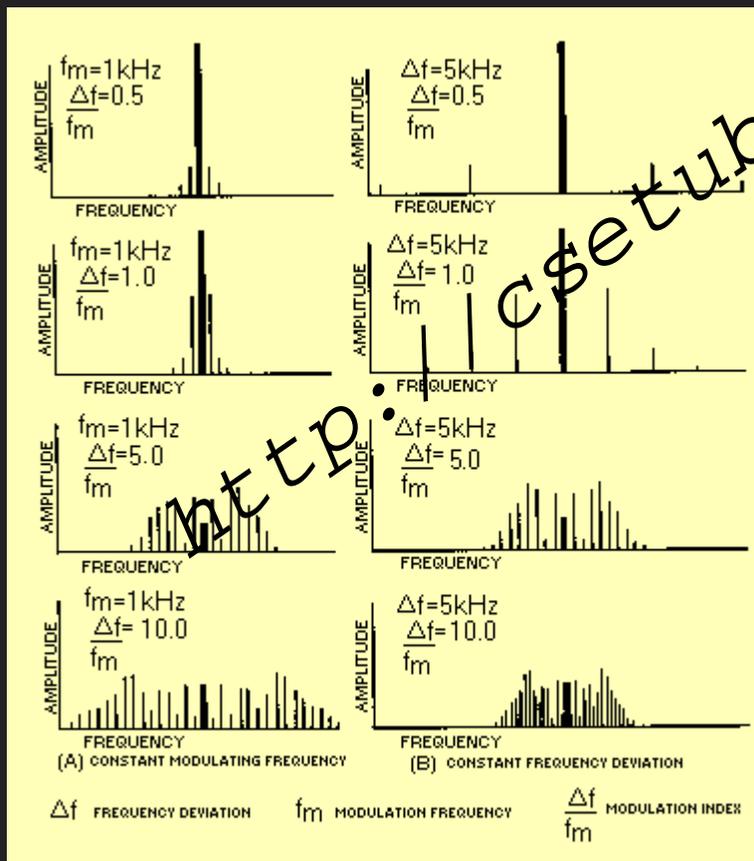
$$\text{modulation index} = \frac{\Delta f}{f_m}$$

Where:

Δf = frequency deviation
 f_m = modulating frequency

Views (A) and (B) of figure 2-9 show the frequency spectrum for various fm signals. In the four examples of view (A), the modulating frequency is constant; the deviation frequency is changed to show the effects of modulation indexes of 0.5, 1.0, 5.0, and 10.0. In view (B) the deviation frequency is held constant and the modulating frequency is varied to give the same modulation indexes.

Figure 2 - 9. - Frequency spectra of fm waves under various conditions.



You can determine several facts about fm signals by studying the frequency spectrum. For example, table 2-1 was developed from the information in figure 2-9. Notice in the top spectrums of both views (A) and (B) that the modulation index is 0.5. Also notice as you look at the next lower spectrums that the modulation index is 1.0. Next down is 5.0, and finally, the bottom spectrums have modulation indexes of 10.0. This information was

used to develop table 2-1 by listing the modulation indexes in the left column and the number of significant sidebands in the right. SIGNIFICANT SIDEBANDS (those with significantly large amplitudes) are shown in both views of figure 2-9 as vertical lines on each side of the carrier frequency. Actually, an infinite number of sidebands are produced, but only a small portion of them are of sufficient amplitude to be important. For example, for a modulation index of 0.5 [top spectrums of both views (A) and (B)], the number of significant sidebands counted is 4. For the next spectrums down, the modulation index is 1.0 and the number of sidebands is 6, and so forth. This holds true for any combination of deviating and modulating frequencies that yield identical modulating indexes.

Table 2-1. - Modulation index table

MODULATION INDEX	SIGNIFICANT SIDEBANDS
.01	2
.4	2
.5	4
1.0	6
2.0	8
3.0	12
4.0	14
5.0	16
6.0	18
7.0	22
8.0	24
9.0	26
10.0	28
11.0	32
12.0	32
13.0	36
14.0	38
15.0	38

You should be able to see by studying figure 2-9, views (A) and (B), that the modulating frequency determines the spacing of the sideband frequencies. By using a significant sidebands table (such as table 2-1), you can determine the bandwidth of a given fm signal. Figure 2-10 illustrates the use of this table. The carrier frequency shown is 500 kilohertz. The modulating frequency is 15 kilohertz and the deviation frequency is 75 kilohertz.

$$\Delta f = 75\text{kHz}$$

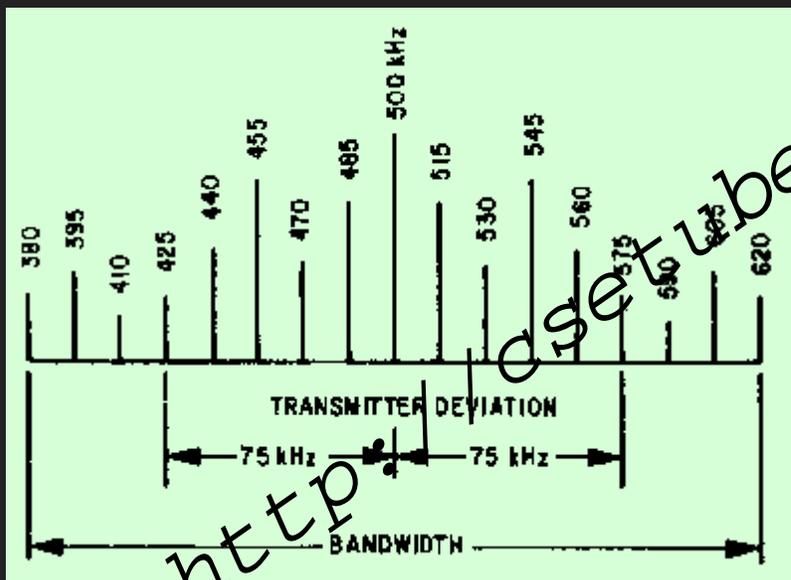
$$f_m = 15\text{kHz}$$

$$MI = \frac{\Delta f}{f_m}$$

$$MI = \frac{75\text{kHz}}{15\text{kHz}}$$

$$MI = 5$$

Figure 2-10. - Frequency deviation versus bandwidth.



From table 2-1 we see that there are 16 significant sidebands for a modulation index of 5. To determine total bandwidth for this case, we use:

$$bw = \text{modulating frequency} \times \text{no. of significant sidebands}$$

$$bw = 15\text{kHz} \times 16$$

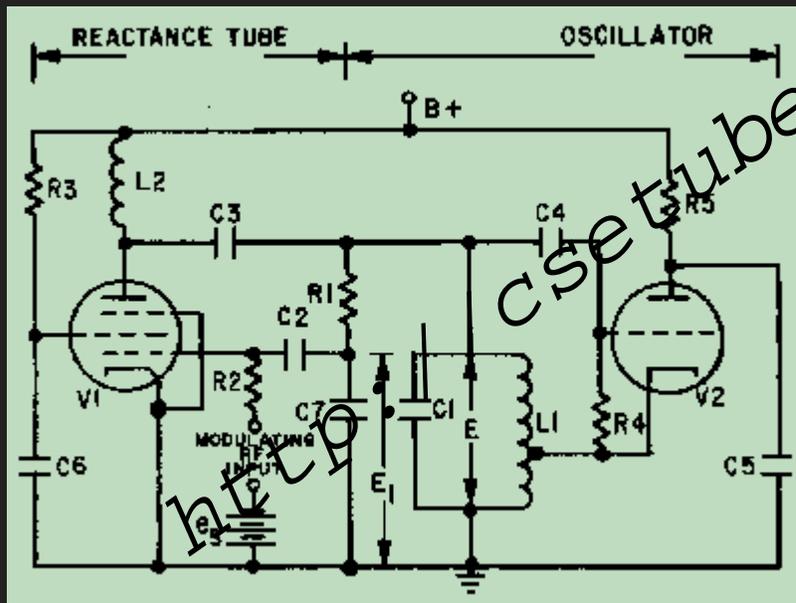
$$bw = 240\text{kHz}$$

The use of this math is to illustrate that the actual bandwidth of an fm transmitter (240 kHz) is greater than that suggested by its maximum deviation bandwidth (□75 kHz, or 150 kHz). This is important to know when choosing operating frequencies or designing equipment.

METHODS OF FREQUENCY MODULATION. - The circuit shown earlier in figure 2-6 and the discussion in previous paragraphs were for illustrative purposes only. In reality, such a circuit would not be practical. However, the basic principle involved (the change in reactance of an oscillator circuit in accordance with the modulating voltage) constitutes one of the methods of developing a frequency-modulated wave.

Reactance-Tube Modulation. - In direct modulation, an oscillator is frequency modulated by a REACTANCE TUBE that is in parallel (SHUNT) with the oscillator tank circuit. (The terms "shunt" or "shunting" will be used in this module to mean the same as "parallel" or "to place in parallel with" components.) This is illustrated in figure 2-11. The oscillator is a conventional Hartley circuit with the reactance-tube circuit in parallel with the tank circuit of the oscillator tube. The reactance tube is an ordinary pentode. It is made to act either capacitively or inductively; that is, its grid is excited with a voltage which either leads or lags the oscillator voltage by 90 degrees.

Figure 2-11. - Reactance-tube fm modulator.



When the reactance tube is connected across the tank circuit with no modulating voltage applied, it will affect the frequency of the oscillator. The voltage across the oscillator tank circuit (L1 and C1) is also in parallel with the series network of R1 and C7. This voltage causes a current flow through R1 and C7. If R1 is at least five times larger than the capacitive reactance of C7, this branch of the circuit will be essentially resistive. Voltage E1, which is across C7, will lag current by 90 degrees. E1 is applied to the control grid of reactance tube V1. This changes plate current (I_p), which essentially flows only through the LC tank circuit. This is because the value of R1 is high compared to the impedance of the tank circuit. Since current is inversely proportional to impedance, most of the plate current coupled through C3 flows through the tank circuit.

At resonance, the voltage and current in the tank circuit are in phase. Because E_1 lags E by 90 degrees and I_p is in phase with grid voltage E_1 , the superimposed current through the tank circuit lags the original tank current by 90 degrees. Both the resultant current (caused by I_p) and the tank current lag tank voltage and current by some angle depending on the relative amplitudes of the two currents. Because this resultant current is a lagging current, the impedance across the tank circuit cannot be at its maximum unless something happens within the tank to bring current and voltage into phase. Therefore, this situation continues until the frequency of oscillations in the tank circuit changes sufficiently so that the voltages across the tank and the current flowing into it are again in phase. This action is the same as would be produced by adding a reactance in parallel with the LC tank. Because the superimposed current lags voltage E by 90 degrees, the introduced reactance is inductive. In NEETS, Module 2, Introduction to Alternating Current and Transformers, you learned that total inductance decreases as additional inductors are added in parallel. Because this introduced reactance effectively reduces inductance, the frequency of the oscillator increases to a new fixed value.

Now let's see what happens when a modulating signal is applied. The magnitude of the introduced reactance is determined by the magnitude of the superimposed current through the tank. The magnitude of I_p for a given E_1 is determined by the transconductance of V_1 . (Transconductance was covered in NEETS, Module 6, Introduction to Electronic Emission, Tubes, and Power Supplies.)

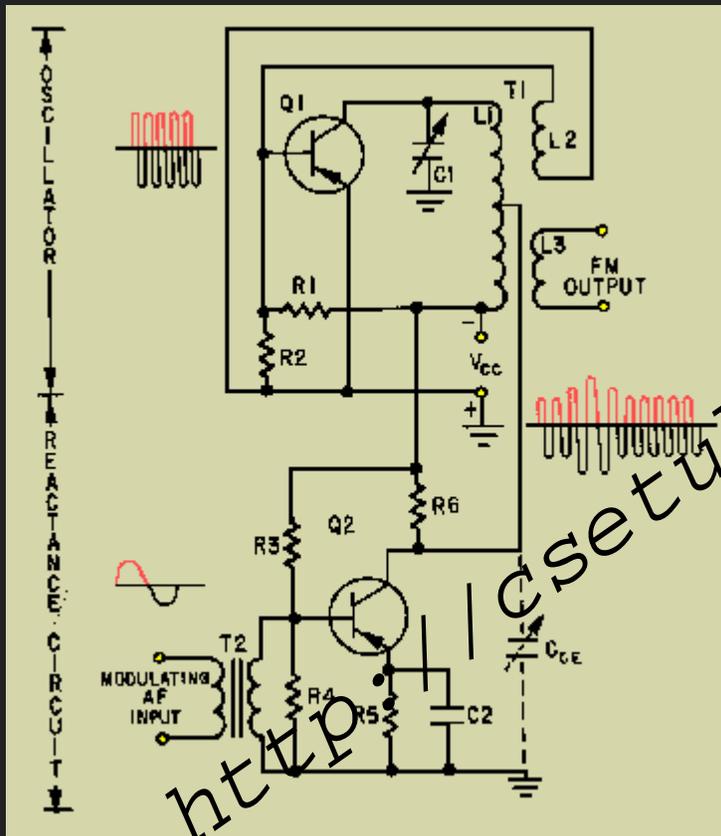
Therefore, the value of reactance introduced into the tuned circuit varies directly with the transconductance of the reactance tube. When a modulating signal is applied to the grid of V_1 , both E_1 and I_p change, causing transconductance to vary with the modulating signal. This causes a variable reactance to be introduced into the tuned circuit. This variable reactance either adds to or subtracts from the fixed value of reactance that is introduced in the absence of the modulating signal. This action varies the reactance across the oscillator which, in turn, varies the instantaneous frequency of the oscillator. These variations in the oscillator frequency are proportional to the instantaneous amplitude of the modulating voltage. Reactance-tube modulators are usually operated at low power levels. The required output power is developed in power amplifier stages that follow the modulators.

The output of a reactance-tube modulated oscillator also contains some unwanted amplitude modulation. This unwanted modulation is caused by stray capacitance and the resistive component of the RC phase splitter. The resistance is much less significant than the desired XC, but the resistance does allow some plate current to flow which is not of the proper phase relationship for good tube operation. The small amplitude modulation that this produces is easily removed by passing the oscillator output through a limiter-amplifier circuit.

Semiconductor Reactance Modulator. - The SEMICONDUCTOR-REACTANCE MODULATOR is used to frequency modulate low-power semiconductor transmitters.

Figure 2-12 shows a typical frequency-modulated oscillator stage operated as a reactance modulator. Q1, along with its associated circuitry, is the oscillator. Q2 is the modulator and is connected to the circuit so that its collector-to-emitter capacitance (CCE) is in parallel with a portion of the rf oscillator coil, L1. As the modulator operates, the output capacitance of Q2 is varied. Thus, the frequency of the oscillator is shifted in accordance with the modulation the same as if C1 were varied.

Figure 2-12. - Reactance-semiconductor fm modulator.



When the modulating signal is applied to the base of Q2, the emitter-to-base bias varies at the modulation rate. This causes the collector voltage of Q2 to vary at the same modulating rate. When the collector voltage increases, output capacitance CCE decreases; when the collector voltage decreases, CCE increases. An increase in collector voltage has the effect of spreading the plates of CCE farther apart by increasing the width of the barrier. A decrease of collector voltage reduces the width of the pn junction and has the same effect as pushing the capacitor plates together to provide more capacitance.

When the output capacitance decreases, the instantaneous frequency of the oscillator tank circuit increases (acts the same as if C1 were decreased). When the output capacitance increases, the instantaneous frequency of the oscillator tank circuit decreases. This decrease in frequency produces a lower frequency in the output because of the shunting effect of CCE. Thus, the frequency of the oscillator tank circuit increases and decreases at

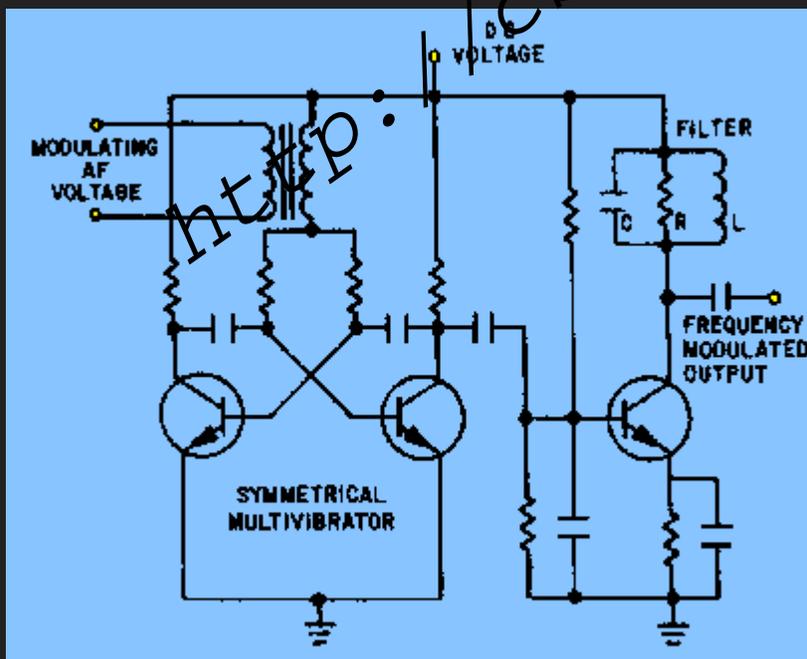
an audio frequency (af) modulating rate. The output of the oscillator, therefore, is a frequency modulated rf signal.

Since the audio modulation causes the collector voltage to increase and decrease, an AM component is induced into the output. This produces both an fm and AM output. The amplitude variations are then removed by placing a limiter stage after the reactance modulator and only the frequency modulation remains.

Frequency multipliers or mixers (discussed in chapter 1) are used to increase the oscillator frequency to the desired output frequency. For high-power applications, linear rf amplifiers are used to increase the steady-amplitude signal to a higher power output. With the initial modulation occurring at low levels, fm represents a savings of power when compared to conventional AM. This is because fm noise-reducing properties provide a better signal-to-noise ratio than is possible with AM.

Multivibrator Modulator. - Another type of frequency modulator is the astable multivibrator illustrated in figure 2-13. Inserting the modulating af voltage in series with the base-return of the multivibrator transistors causes the gate length, and thus the fundamental frequency of the multivibrator, to vary. The amount of variation will be in accordance with the amplitude of the modulating voltage. One requirement of this method is that the fundamental frequency of the multivibrator be high in relation to the highest modulating frequencies. A factor of at least 100 provides the best results.

Figure 2-13. - Astable multivibrator and filter circuit for generating an fm carrier.



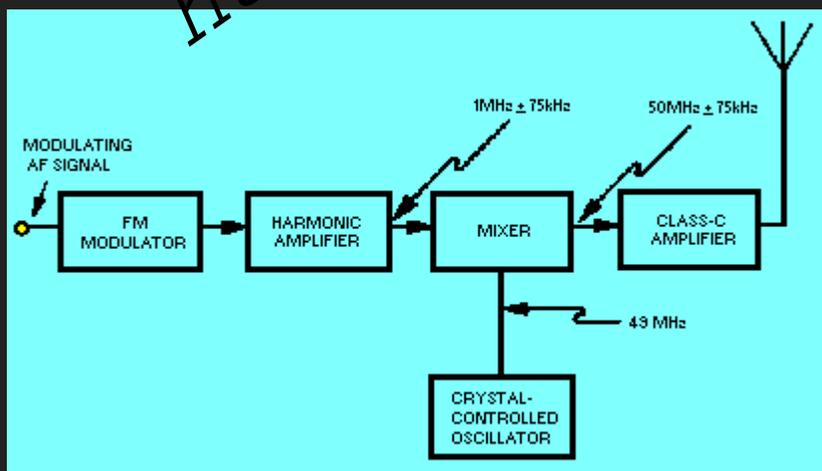
Recall that a multivibrator output consists of the fundamental frequency and all of its harmonics. Unwanted even harmonics are eliminated by using a SYMMETRICAL

MULTIVIBRATOR circuit, as shown in figure 2-13. The desired fundamental frequency, or desired odd harmonics, can be amplified after all other odd harmonics are eliminated in the LCR filter section of figure 2-13. A single frequency-modulated carrier is then made available for further amplification and transmission.

Proper design of the multivibrator will cause the frequency deviation of the carrier to faithfully follow (referred to as a "linear" function) the modulating voltage. This is true up to frequency deviations which are considerable fractions of the fundamental frequency of the multivibrator. The principal design consideration is that the RC coupling from one multivibrator transistor base to the collector of the other has a time constant which is greater than the actual gate length by a factor of 10 or more. Under these conditions, a rise in base voltage in each transistor is essentially linear from cutoff to the bias at which the transistor is switched on. Since this rise in base voltage is a linear function of time, the gate length will change as an inverse function of the modulating voltage. This action will cause the frequency to change as a linear function of the modulating voltage.

The multivibrator frequency modulator has the advantage over the reactance-type modulator of a greater linear frequency deviation from a given carrier frequency. However, multivibrators are limited to frequencies below about 1 megahertz. Both systems are subject to drift of the carrier frequency and must, therefore, be stabilized. Stabilization may be accomplished by modulating at a relatively low frequency and translating by heterodyne action to the desired output frequency, as shown in figure 2-14. A 1-megahertz signal is heterodyned with 49 megahertz from the crystal-controlled oscillator to provide a stable 50-megahertz output from the mixer. If a suitably stable heterodyning oscillator is used, the frequency stability can be greatly improved. For instance, at the frequencies shown in figure 2-14, the stability of the unmodulated 50-megahertz carrier would be 50 times better than that which harmonic multiplication could provide.

Figure 2-14. - Method for improving frequency stability of fm system.



Varactor FM Modulator. - Another fm modulator which is widely used in transistorized circuitry uses a voltage-variable capacitor (VARACTOR). The varactor is simply a diode,

or pn junction, that is designed to have a certain amount of capacitance between junctions. View (A) of figure 2-15 shows the varactor schematic symbol. A diagram of a varactor in a simple oscillator circuit is shown in view (B). This is not a working circuit, but merely a simplified illustration. The capacitance of a varactor, as with regular capacitors, is determined by the area of the capacitor plates and the distance between the plates. The depletion region in the varactor is the dielectric and is located between the p and n elements, which serve as the plates. Capacitance is varied in the varactor by varying the reverse bias which controls the thickness of the depletion region. The varactor is so designed that the change in capacitance is linear with the change in the applied voltage. This is a special design characteristic of the varactor diode. The varactor must not be forward biased because it cannot tolerate much current flow. Proper circuit design prevents the application of forward bias.

Figure 2-15A. - Varactor symbol and schematic. SCHEMATIC SYMBOL

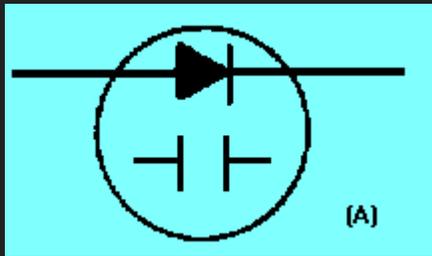
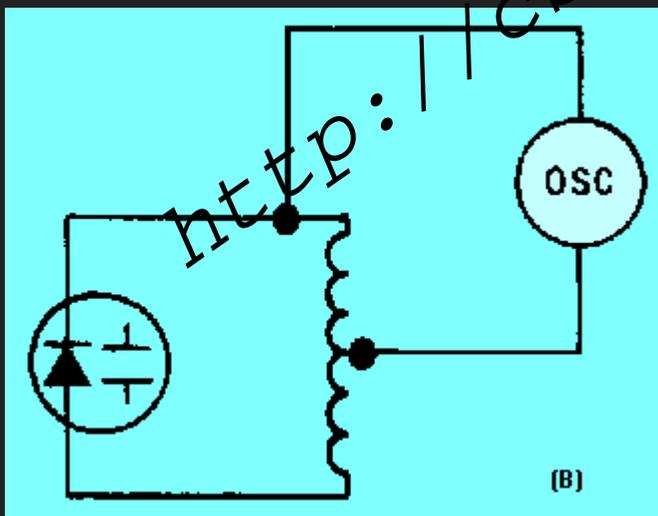
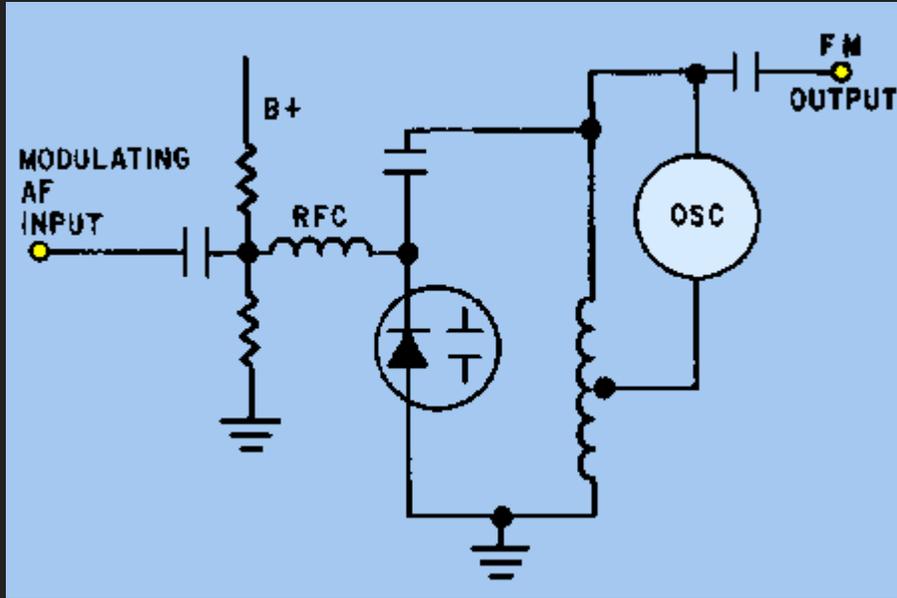


Figure 2-15B. - Varactor symbol and schematic. SIMPLIFIED CIRCUIT



Notice the simplicity of operation of the circuit in figure 2-16. An af signal that is applied to the input results in the following actions: (1) On the positive alternation, reverse bias increases and the dielectric (depletion region) width increases. This decreases capacitance which increases the frequency of the oscillator. (2) On the negative alternation, the reverse bias decreases, which results in a decrease in oscillator frequency.

Figure 2-16. - Varactor fm modulator.

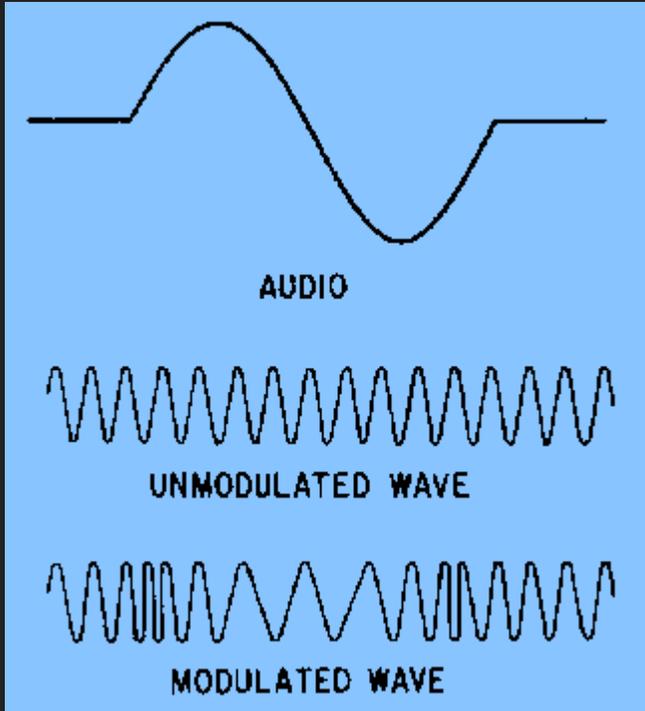


Many different fm modulators are available, but they all use the basic principles you have just studied. The main point to remember is that an oscillator must be used to establish the reference (carrier) frequency. Secondly, some method is needed to cause the oscillator to change frequency in accordance with an af signal.

PHASE MODULATION

Frequency modulation requires the oscillator frequency to deviate both above and below the carrier frequency. During the process of frequency modulation, the peaks of each successive cycle in the modulated waveform occur at times other than they would if the carrier were unmodulated. This is actually an incidental phase shift that takes place along with the frequency shift in fm. Just the opposite action takes place in phase modulation. The af signal is applied to a PHASE MODULATOR in pm. The resultant wave from the phase modulator shifts in phase, as illustrated in figure 2-17. Notice that the time period of each successive cycle varies in the modulated wave according to the audio-wave variation. Since frequency is a function of time period per cycle, we can see that such a phase shift in the carrier will cause its frequency to change. The frequency change in fm is vital, but in pm it is merely incidental. The amount of frequency change has nothing to do with the resultant modulated wave shape in pm. At this point the comparison of fm to pm may seem a little hazy, but it will clear up as we progress.

Figure 2-17. - Phase modulation.



Let's review some voltage phase relationships. Look at figure 2-18 and compare the three voltages (A, B, and C). Since voltage A begins its cycle and reaches its peak before voltage B, it is said to lead voltage B. Voltage C, on the other hand, lags voltage B by 30 degrees. In phase modulation the phase of the carrier is caused to shift at the rate of the af modulating signal. In figure 2-19, note that the unmodulated carrier has constant phase, amplitude, and frequency. The dotted wave shape represents the modulated carrier. Notice that the phase on the second peak leads the phase of the unmodulated carrier. On the third peak the shift is even greater; however, on the fourth peak, the peaks begin to realign phase with each other. These relationships represent the effect of 1/2 cycle of an af modulating signal. On the negative alternation of the af intelligence, the phase of the carrier would lag and the peaks would occur at times later than they would in the unmodulated carrier.

Figure 2-18. - Phase relationships.

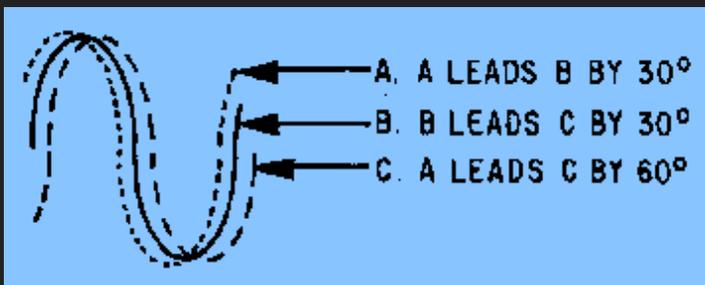
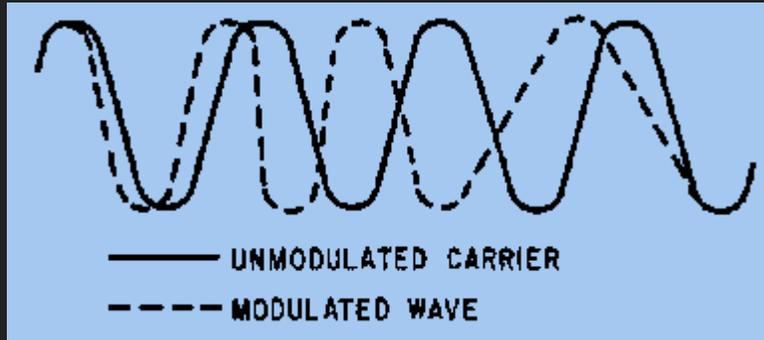


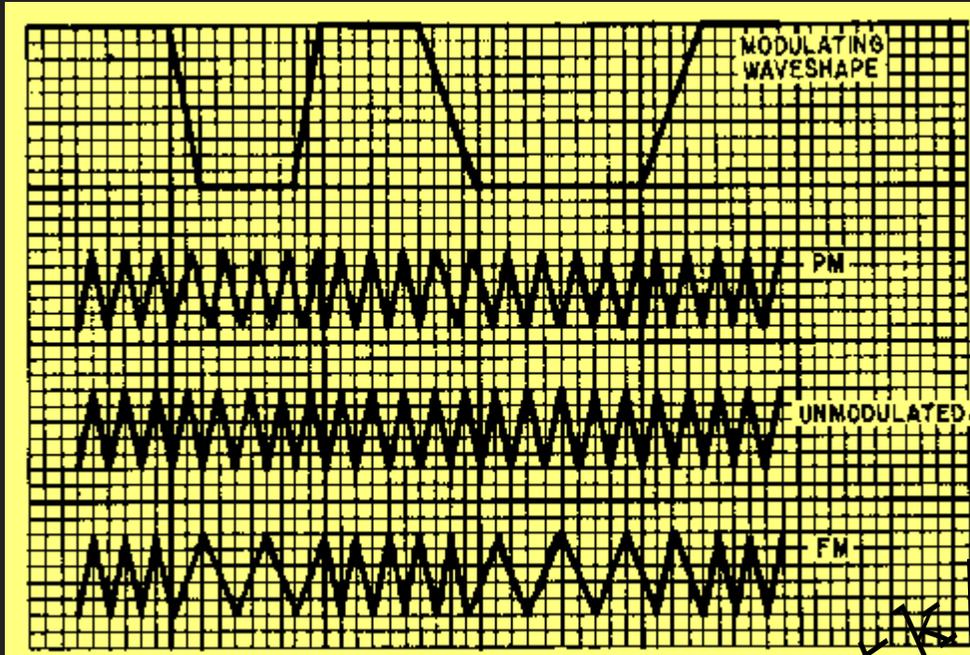
Figure 2-19. - Carrier with and without modulation.



The presentation of these two waves together does not mean that we transmit a modulated wave together with an unmodulated carrier. The two waveforms were drawn together only to show how a modulated wave looks when compared to an unmodulated wave.

Now that you have seen the phase and frequency shifts in both fm and pm, let's find out exactly how they differ. First, only the phase shift is important in pm. It is proportional to the af modulating signal. To visualize this relationship, refer to the wave shapes shown in figure 2-20. Study the composition of the fm and pm waves carefully as they are modulated with the modulating wave shape. Notice that in fm, the carrier frequency deviates when the modulating wave changes polarity. With each alternation of the modulating wave, the carrier advances or retards in frequency and remains at the new frequency for the duration of that cycle. In pm you can see that between one alternation and the next, the carrier phase must change, and the frequency shift that occurs does so only during the transition time; the frequency then returns to its normal rate. Note in the pm wave that the frequency shift occurs only when the modulating wave is changing polarity. The frequency during the constant amplitude portion of each alternation is the REST FREQUENCY.

Figure 2-20. - Pm versus fm.



The relationship, in pm, of the modulating af to the change in the phase shift is easy to see once you understand AM and fm principles. Again, we can establish two clear-cut rules of phase modulation:

AMOUNT OF PHASE SHIFT IS PROPORTIONAL TO THE AMPLITUDE OF THE MODULATING SIGNAL.

(If a 10-volt signal causes a phase shift of 20 degrees, then a 20-volt signal causes a phase shift of 40 degrees.)

RATE OF PHASE SHIFT IS PROPORTIONAL TO THE FREQUENCY OF THE MODULATING SIGNAL.

(If the carrier were modulated with a 1-kilohertz tone, the carrier would advance and retard in phase 1,000 times each second.)

Phase modulation is also similar to frequency modulation in the number of sidebands that exist within the modulated wave and the spacing between sidebands. Phase modulation will also produce an infinite number of sideband frequencies. The spacing between these sidebands will be equal to the frequency of the modulating signal. However, one factor is very different in phase modulation; that is, the distribution of power in pm sidebands is not similar to that in fm sidebands, as will be explained in the next section.

Modulation Index

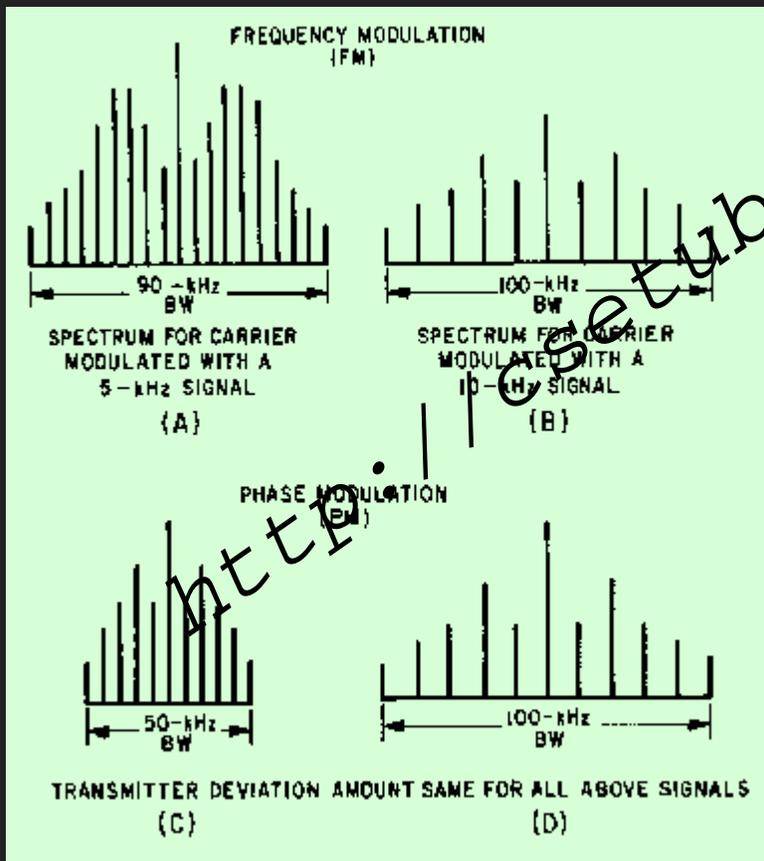
Recall from frequency modulation that modulation index is used to calculate the number of significant sidebands existing in the waveform. The higher the modulation index, the

greater the number of sideband pairs. The modulation index is the ratio between the amount of oscillator deviation and the frequency of the modulating signal:

$$MI = \frac{\text{transmitter deviation}}{\text{modulating frequency}}$$

In frequency modulation, we saw that as the frequency of the modulating signal increased (assuming the deviation remained constant) the number of significant sideband pairs decreased. This is shown in views (A) and (B) of figure 2-21. Notice that although the total number of significant sidebands decreases with a higher frequency-modulating signal, the sidebands spread out relative to each other; the total bandwidth increases.

Figure 2-21. - Fm versus pm spectrum distribution.



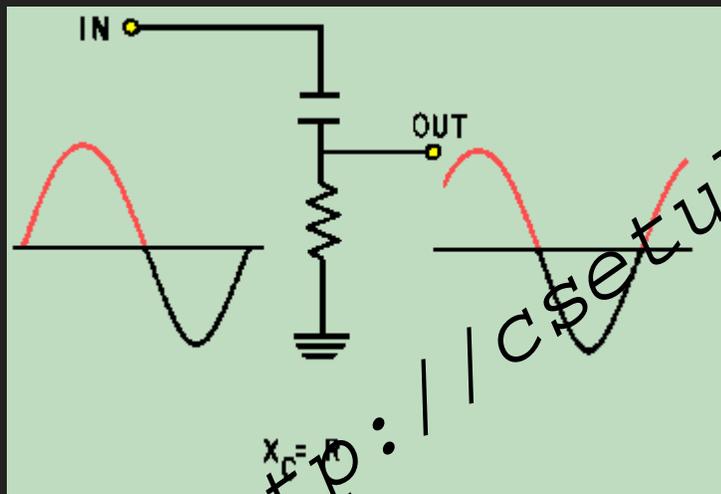
In phase modulation the oscillator does not deviate, and the power in the sidebands is a function of the amplitude of the modulating signal. Therefore, two signals, one at 5 kilohertz and the other at 10 kilohertz, used to modulate a carrier would have the same sideband power distribution. However, the 10-kilohertz sidebands would be farther apart, as shown in views (C) and (D) of figure 2-21. When compared to fm, the bandwidth of the pm transmitted signal is greatly increased as the frequency of the modulating signal is increased.

As we pointed out earlier, phase modulation cannot occur without an incidental change in frequency, nor can frequency modulation occur without an incidental change in phase. The term fm is loosely used when referring to any type of angle modulation, and phase modulation is sometimes incorrectly referred to as "indirect fm." This is a definition that you should disregard to avoid confusion. Phase modulation is just what the words imply - phase modulation of a carrier by an af modulating signal. You will develop a better understanding of these points as you advance in your study of modulation.

Basic Modulator

In phase modulation you learned that varying the phase of a carrier at an intelligence rate caused that carrier to contain variations which could be converted back into intelligence. One circuit that can cause this phase variation is shown in figure 2-22.

Figure 2-22. - Phase shifting a sine wave.



The capacitor in series with the resistor forms a phase-shift circuit. With a constant frequency rf carrier applied at the input, the output across the resistor would be 45 degrees out of phase with the input if $X_C = R$.

Now, let's vary the resistance and observe how the output is affected in figure 2-23. As the resistance reaches a value greater than 10 times X_C , the phase difference between input and output is nearly 0 degrees. For all practical purposes, the circuit is resistive. As the resistance is decreased to 1/10 the value of X_C , the phase difference approaches 90 degrees. The circuit is now almost completely capacitive. By replacing the resistor with a vacuum tube, as shown in view (A) of figure 2-24, we can vary the resistance (vacuum-tube impedance) by varying the voltage applied to the grid of the tube. The frequency applied to the circuit (from a crystal-controlled master oscillator) will be shifted in phase by 45 degrees with no audio input [view (B)]. With the application of an audio signal, the phase will shift as the impedance of the tube is varied.

Figure 2-23. - Control over the amount of phase shift.

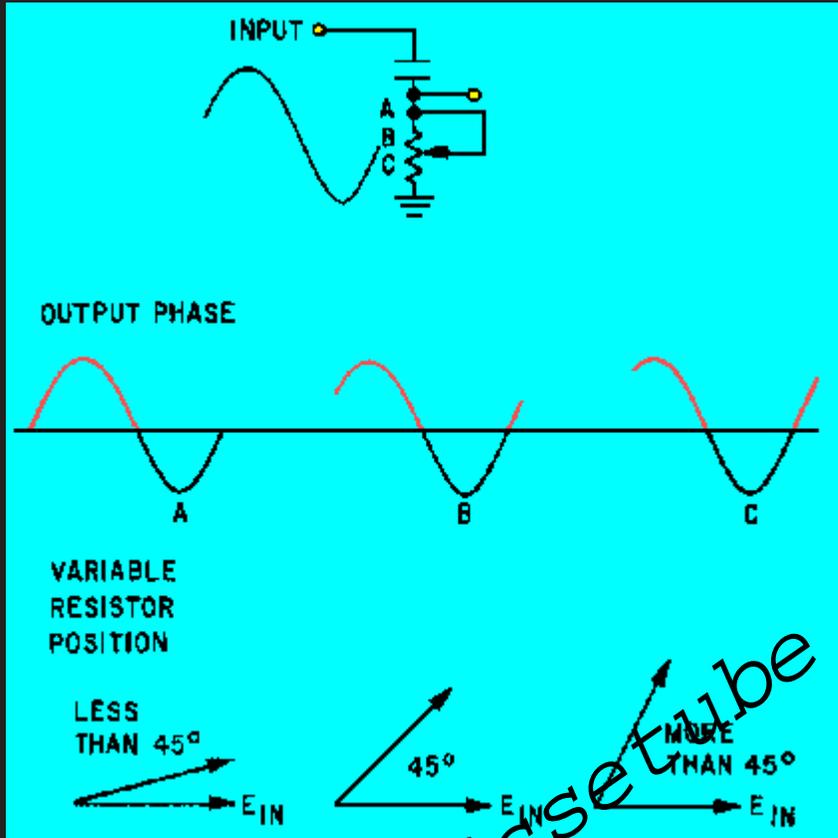


Figure 2-24A. - Phase modulator.

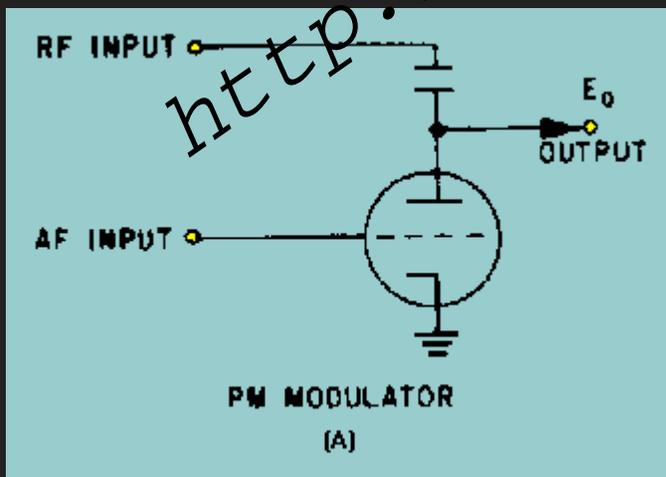
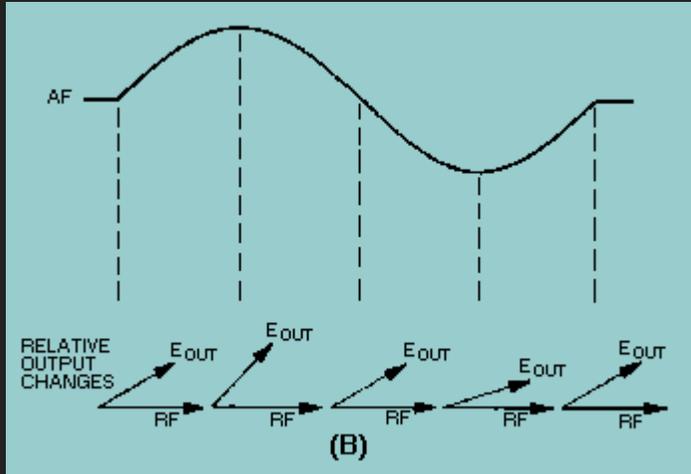


Figure 2-24B. - Phase modulator.



In practice, a circuit like this could not provide enough phase shift to produce the desired results in the output. Several of these circuits are arranged in cascade to provide the desired amount of phase shift. Also, since the output of this circuit will vary in amplitude, the signal is fed to a limiter to remove amplitude variations.

The major advantage of this type modulation circuit over frequency modulation is that this circuit uses a crystal-controlled oscillator to maintain a stable carrier frequency. In fm the oscillator cannot be crystal controlled because it is actually required to vary in frequency. That means that an fm oscillator will require a complex automatic frequency control (afc) system. An afc system ensures that the oscillator stays on the same carrier frequency and achieves a high degree of stability.

FM Transmitters

MODULATORS

There are two types of FM modulators - direct and indirect. Direct FM involves varying the frequency of the carrier directly by the modulating input. Indirect FM involves directly altering the phase of the carrier based on the input (this is actually a form of direct phase modulation).

Direct modulation is usually accomplished by varying a capacitance in an LC oscillator or by changing the charging current applied to a capacitor.

The first method can be accomplished by the use of a reverse biased diode, since the capacitance of such a diode varies with applied voltage. A varactor diode is specifically

designed for this purpose. Figure 1 shows a direct frequency modulator which uses a varactor diode.

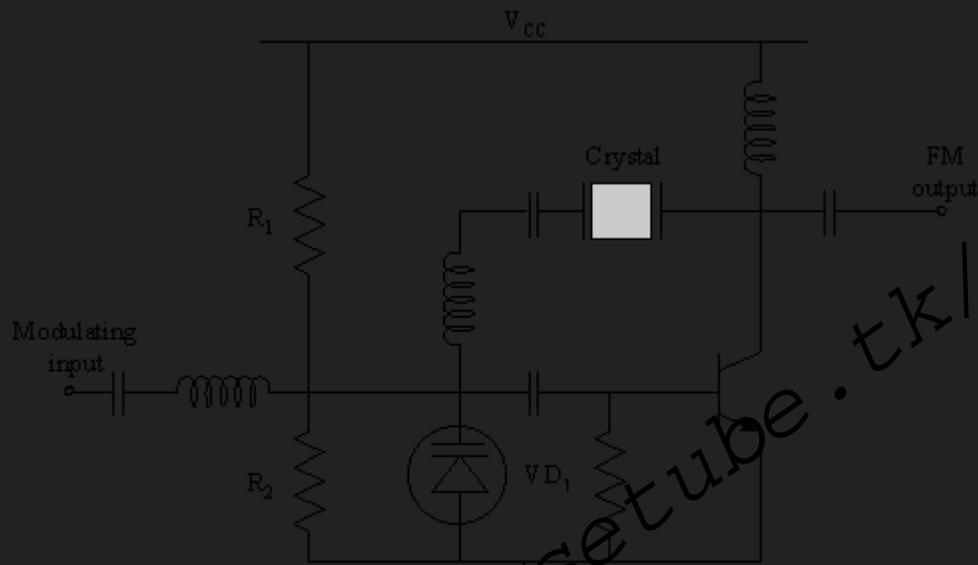


Figure 1 Varactor Diode Frequency Modulator

This circuit deviates the frequency of the crystal oscillator using the diode. R_1 and R_2 develop a DC voltage across the diode which reverse biases it. The voltage across the diode determines the frequency of the oscillations. Positive inputs increase the reverse bias, decrease the diode capacitance and thus increase the oscillation frequency. Similarly, negative inputs decrease the oscillation frequency.

The use of a crystal oscillator means that the output waveform is very stable, but this is only the case if the frequency deviations are kept very small. Thus, the varactor diode modulator can only be used in limited applications.

The second method of direct FM involves the use of a voltage controlled oscillator, which is depicted in figure 2.

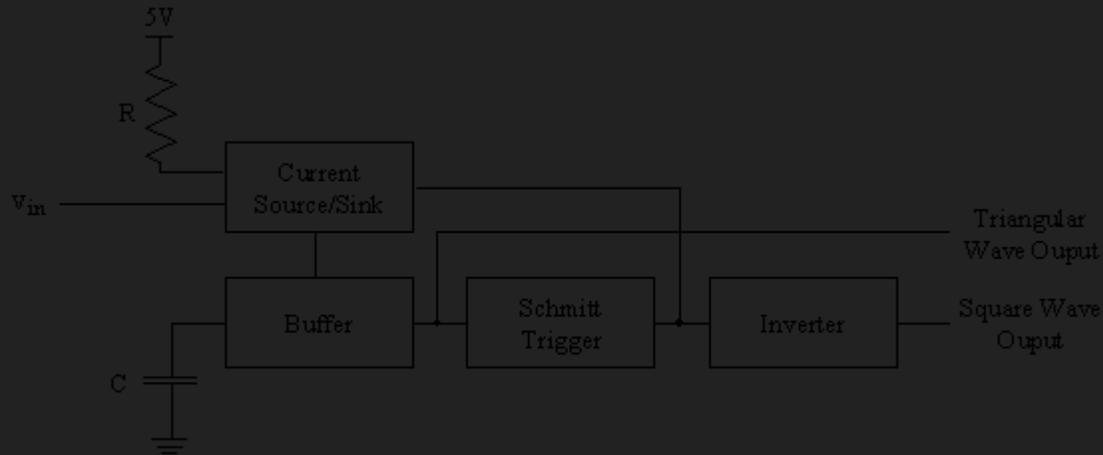


Figure 2 Block diagram of 566 VCO

The capacitor repeatedly charges and discharges under the control of the current source/sink. The amount of current supplied by this module is determined by V_{IN} and by the resistor R . Since the amount of current determines the rate of capacitor charging, the resistor effectively controls the period of the output. The capacitance C also controls the rate of charging. The capacitor voltage is the input to the Schmitt trigger which changes the mode of the current source/sink when a certain threshold is reached. The capacitor voltage then heads in the opposite direction, generating a triangular wave. The output of the Schmitt trigger provides the square wave output. These signals can then be low-pass filtered to provide a sinusoidal FM signal.

The major limitation of the voltage controlled oscillator is that it can only work for a small range of frequencies. For instance, the 566 IC VCO only works a frequencies up to 1MHz.

A varactor diode circuit for indirect FM is shown in figure 3.

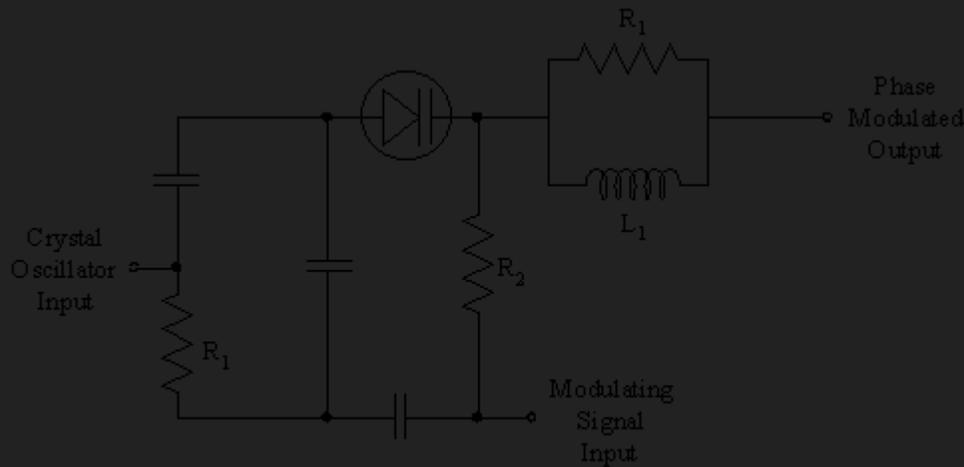


Figure 3 Varactor Diode Circuit for Phase Modulation

The modulating signal varies the capacitance of the diode, which then changes the phase shift incurred by the carrier input and thus changes the phase of the output signal. Because the phase of the carrier is shifted, the resulting signal has a frequency which is more stable than in the direct FM case

TRANSMITTERS

As previously stated, if a crystal oscillator is used to provide the carrier signal, the frequency cannot be varied too much (this is a characteristic of crystal oscillators). Thus, crystal oscillators cannot be used in broadcast FM, but other oscillators can suffer from frequency drift. An automatic frequency control (AFC) circuit is used in conjunction with a non-crystal oscillator to ensure that the frequency drift is minimal.

Figure 4 shows a Crosby direct FM transmitter which contains an AFC loop. The frequency modulator shown can be a VCO since the oscillator frequency is much lower than the actual transmission frequency. In this example, the oscillator centre frequency is 5.1MHz which is multiplied by 18 before transmission to give $f_t = 91.8\text{MHz}$.

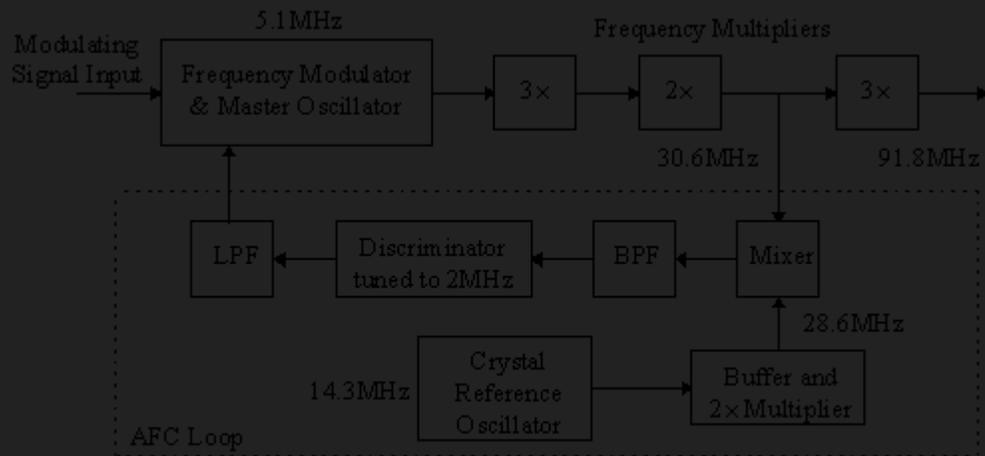


Figure 4 Crosby Direct FM transmitter

When the frequency is multiplied, so are the frequency and phase deviations. However, the modulating input frequency is obviously unchanged, so the modulation index is multiplied by 18. The maximum frequency deviation at the output is 75kHz, so the maximum allowed deviation at the modulator output is

$$\Delta f = \frac{75\text{kHz}}{18} = 4166.7\text{Hz}$$

Since the maximum input frequency is $f_m = 15\text{kHz}$ for broadcast FM, the modulation index must be

$$\beta = \frac{\Delta f}{f_m} = 0.2778$$

The modulation index at the antenna then is $\beta = 0.2778 \times 18 = 5$.

The AFC loop aims to increase the stability of the output without using a crystal oscillator in the modulator.

The modulated carrier signal is mixed with a crystal reference signal in a non-linear device. The band-pass filter provides the difference in frequency between the master oscillator and the crystal oscillator and this signal is fed into the frequency discriminator. The frequency discriminator produces a voltage proportional to the difference between the input frequency and its resonant frequency. Its resonant frequency is 2MHz, which will allow it to detect low frequency variations in the carrier.

The output voltage of the frequency discriminator is added to the modulating input to correct for frequency deviations at the output. The low-pass filter ensures that the frequency discriminator does not correspond to the frequency deviation in the FM signal (thereby preventing the modulating input from being completely cancelled).

Indirect transmitters have no need for an AFC circuit because the frequency of the crystal is not directly varied. This means that indirect transmitters provide a very stable output, since the crystal frequency does not vary with operating conditions.

Figure 5 shows the block diagram for an Armstrong indirect FM transmitter. This works by using a suppressed carrier amplitude modulator and adding a phase shifted carrier to this signal. The effect of this is shown in figure 6, where the pink signal is the output and the blue signal the AM input. The output experiences both phase and amplitude modulation. The amplitude modulation can be reduced by using a carrier much larger than the peak signal amplitude, as shown in figure 7. However, this reduces the amount of phase variation.

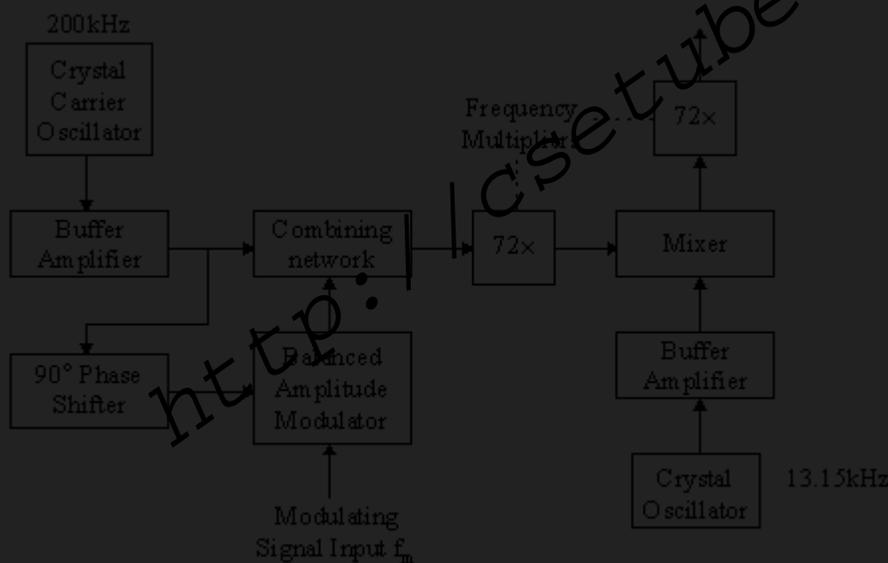


Figure 5 Armstrong Indirect FM transmitter

Figure 6 Phase modulation using amplitude

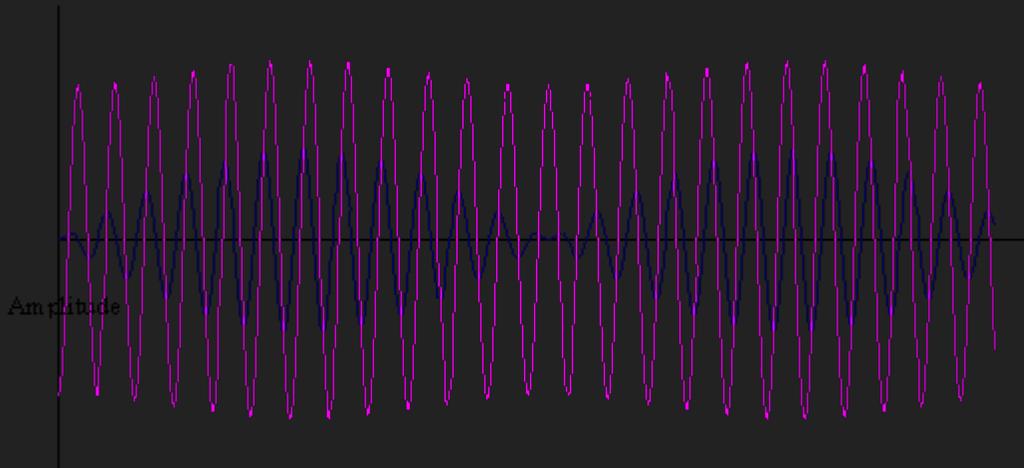


Figure 7 Better phase modulation with less amplitude



The disadvantage of this method is the limited phase shift it can provide. The rest of figure 5 shows the frequency shifting to the FM broadcast band by means of frequency multiplication (by a factor of 72), frequency shifting and frequency multiplication again. This also multiplies the amount of phase shift at the antenna, allowing the required phase shift to be produced by a small phase variation at the modulator output.

FM Receiver

An FM waveform carries its information in the form of frequency, so the amplitude is constant. Thus the information is held in the zero crossings. The FM waveform can be clipped at a low level without the loss of information. Additive noise has less of an effect on zero crossings than the amplitude. Receivers therefore often clip, or limit the amplitude of the received waveform prior to frequency detection.

This produces a constant waveform as an input to the discriminator. This clipping has the effect of introducing higher harmonic terms which are rejected by a post-detection low-pass filter. A simplified FM receiver is shown in figure 1a, a more sophisticated system is shown in figure 1b.



Figure 1a Simplified FM Receiver

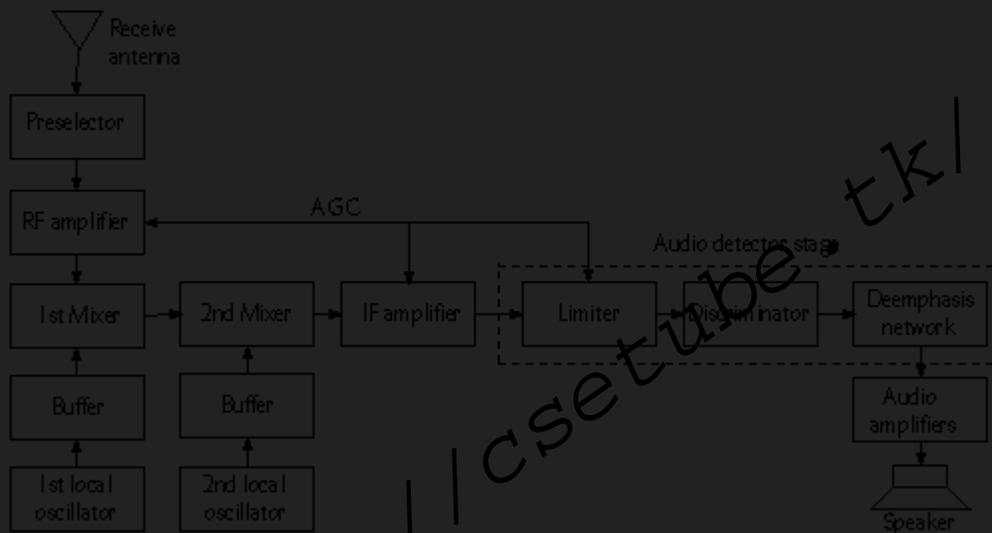


Figure 1b Double-conversion FM Receiver

DEMODULATORS

FM demodulators are frequency-dependant circuits that produce an output voltage that is directly proportional to the instantaneous frequency at its input. The signal received is $f_m(t)$ and is known to the receiver in the form,

$$\lambda_{fm}(t) = A \cos 2\pi \left(f_c t + k_f \int_0^t s(\tau) d\tau \right)$$

Several circuits are used for demodulating FM signals slope detector, Foster-Seeley discriminator, ratio detector, PLL demodulator and quadrature detector. The first three are tuned circuit frequency discriminators, they work by converting the FM signal to AM then demodulate using conventional peak detectors.

Discriminators

A block diagram of a discriminator is shown in figure 2.



Figure 2

The differentiator effectively converts the FM signal into an AM signal. The differentiated FM signal is,

$$\frac{d\lambda}{dt} = -2\pi A [f_c + k f_s(t)] \sin 2\pi \left(f_c t + k f \int_0^t s(\tau) d\tau \right)$$

The envelope detector removes the sine term, this is possible because the slight changes in frequency are not detected by the envelope detector. The envelope is given by

$$2\pi |A [f_c + k f_s(t)]|$$

from which the signal $s(t)$ can be found.

When a differentiator is used like this it is called a slope detector or discriminator. A requirement for a discriminator is that the transfer function be linear throughout the range of frequencies of the FM wave. This is the simplest type of discriminator. Two discriminators can be used by subtracting the characteristic of one from a shifted version of itself, see figure 3.

This method is called a balanced slope detector. It has several disadvantages like poor linearity and difficulty in tuning. Another way is to approximate the derivative by using the difference between two adjacent sample values of the waveform, see figure 4, the Foster-Seeley discriminator or also known as a phase shift demodulator.

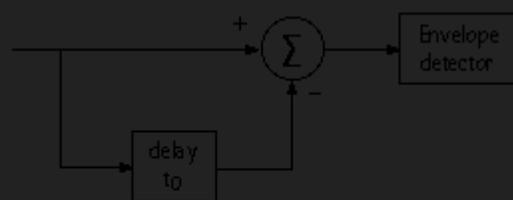


Figure 4

The Foster-Seeley circuit is easier to tune but must be preceded by a separate limiter circuit to clip the amplitude before demodulating. The ratio detector has the property that it is immune to amplitude variations in its input signal, so a preceding limiter is not required.

Radio receiver frequency modulation (FM) demodulation

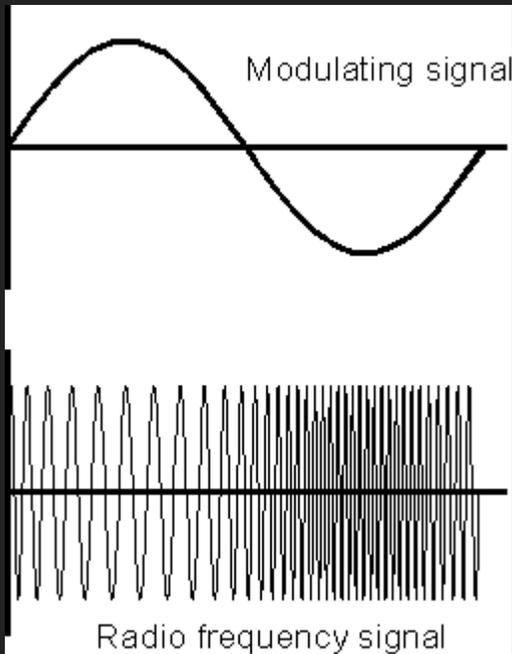
- overview or tutorial of the basics of frequency modulation or FM demodulation using Foster-Seeley, ratio, phase locked loop (pll) and quadrature detectors or demodulators.

Frequency modulation is widely used in radio communications and broadcasting, particularly on frequencies above 30 MHz. It offers many advantages, particularly in mobile radio applications where its resistance to fading and interference is a great advantage. It is also widely used for broadcasting on VHF frequencies where it is able to provide a medium for high quality audio transmissions.

In view of its widespread use receivers need to be able to demodulate these transmissions. There is a wide variety of different techniques and circuits that can be used including the Foster-Seeley, and ratio detectors using discrete components, and where integrated circuits are used the phase locked loop and quadrature detectors are more widely used.

What is FM?

As the name suggests frequency modulation uses changes in frequency to carry the sound or other information that is required to be placed onto the carrier. As shown in Figure 1 it can be seen that as the modulating or base band signal voltage varies, so the frequency of the signal changes in line with it. This type of modulation brings several advantages with it. The first is associated with interference reduction. Much interference appears in the form of amplitude variations and it is quite easy to make FM receivers insensitive to amplitude variations and accordingly this brings about a reduction in the levels of interference. In a similar way fading and other strength variations in the signal have little effect. This can be particularly useful for mobile applications where changes in location as the vehicle moves can bring about significant signal strength changes. A further advantage of FM is that the RF amplifiers in transmitters do not need to be linear. When using amplitude modulation or its derivatives, any amplifier after the modulator must be linear otherwise distortion is introduced. For FM more efficient class C amplifiers may be used as the level of the signal remains constant and only the frequency varies



Frequency modulating a signal

Wide band and Narrow band

When a signal is frequency modulated, the carrier shifts in frequency in line with the modulation. This is called the deviation. In the same way that the modulation level can be varied for an amplitude modulated signal, the same is true for a frequency modulated one, although there is not a maximum or 100% modulation level as in the case of AM.

The level of modulation is governed by a number of factors. The bandwidth that is available is one. It is also found that signals with a large deviation are able to support higher quality transmissions although they naturally occupy a greater bandwidth. As a result of these conflicting requirements different levels of deviation are used according to the application that is used.

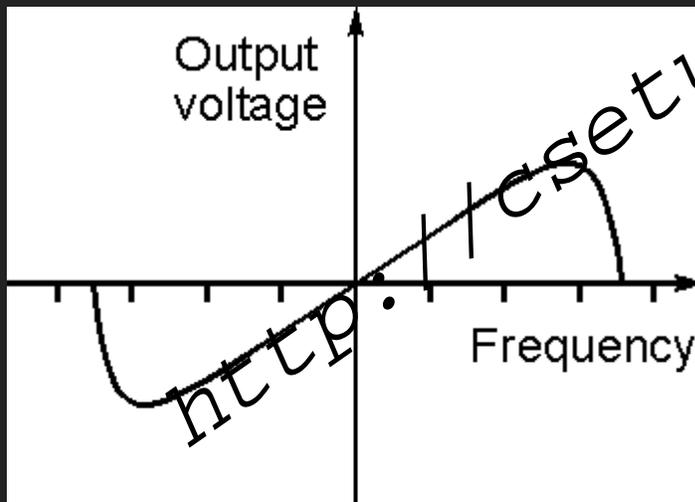
Those with low levels of deviation are called narrow band frequency modulation (NBFM) and typically levels of ± 3 kHz or more are used dependent upon the bandwidth available. Generally NBFM is used for point to point communications. Much higher levels of deviation are used for broadcasting. This is called wide band FM (WBFM) and for broadcasting deviation of ± 75 kHz is used.

Receiving FM

In order to be able to receive FM a receiver must be sensitive to the frequency variations of the incoming signals. As already mentioned these may be wide or narrow band. However the set is made insensitive to the amplitude variations. This is achieved by having a high gain IF amplifier. Here the signals are amplified to such a degree that the amplifier runs into limiting. In this way any amplitude variations are removed.

In order to be able to convert the frequency variations into voltage variations, the demodulator must be frequency dependent. The ideal response is a perfectly linear voltage to frequency characteristic. Here it can be seen that the centre frequency is in the middle of the response curve and this is where the un-modulated carrier would be located when the receiver is correctly tuned into the signal. In other words there would be no offset DC voltage present.

The ideal response is not achievable because all systems have a finite bandwidth and as a result a response curve known as an "S" curve is obtained. Outside the bandwidth of the system, the response falls, as would be expected. It can be seen that the frequency variations of the signal are converted into voltage variations which can be amplified by an audio amplifier before being passed into headphones, a loudspeaker, or passed into other electronic circuitry for the appropriate processing.



Characteristic "S" curve of an FM demodulator

To enable the best detection to take place the signal should be centred about the middle of the curve. If it moves off too far then the characteristic becomes less linear and higher levels of distortion result. Often the linear region is designed to extend well beyond the bandwidth of a signal so that this does not occur. In this way the optimum linearity is achieved. Typically the bandwidth of a circuit for receiving VHF FM broadcasts may be about 1 MHz whereas the signal is only 200 kHz wide.

FM demodulators

There are a number of circuits that can be used to demodulate FM. Each type has its own advantages and disadvantages, some being used when receivers used discrete components, and others now that ICs are widely used.

Slope detection

The very simplest form of FM demodulation is known as slope detection or demodulation. It simply uses a tuned circuit that is tuned to a frequency slightly offset from the carrier of the signal. As the frequency of the signal varies up and down in frequency according to its modulation, so the signal moves up and down the slope of the tuned circuit. This causes the amplitude of the signal to vary in line with the frequency variations. In fact at this point the signal has both frequency and amplitude variations. The final stage in the process is to demodulate the amplitude modulation and this can be achieved using a simple diode circuit. One of the most obvious disadvantages of this simple approach is the fact that both amplitude and frequency variations in the incoming signal appear at the output. However the amplitude variations can be removed by placing a limiter before the detector. Additionally the circuit is not particularly efficient as it operates down the slope of the tuned circuit. It is also unlikely to be particularly linear, especially if it is operated close to the resonant point to minimise the signal loss.

Ratio and Foster-Seeley FM detectors

When circuits employing discrete components were more widely used, the Ratio and Foster-Seeley detectors were widely used. Of these the ratio detector was the most popular as it offers a better level of amplitude modulation rejection of amplitude modulation. This enables it to provide a greater level of noise immunity as most noise is amplitude noise, and it also enables the circuit to operate satisfactorily with lower levels of limiting in the preceding IF stages of the receiver.

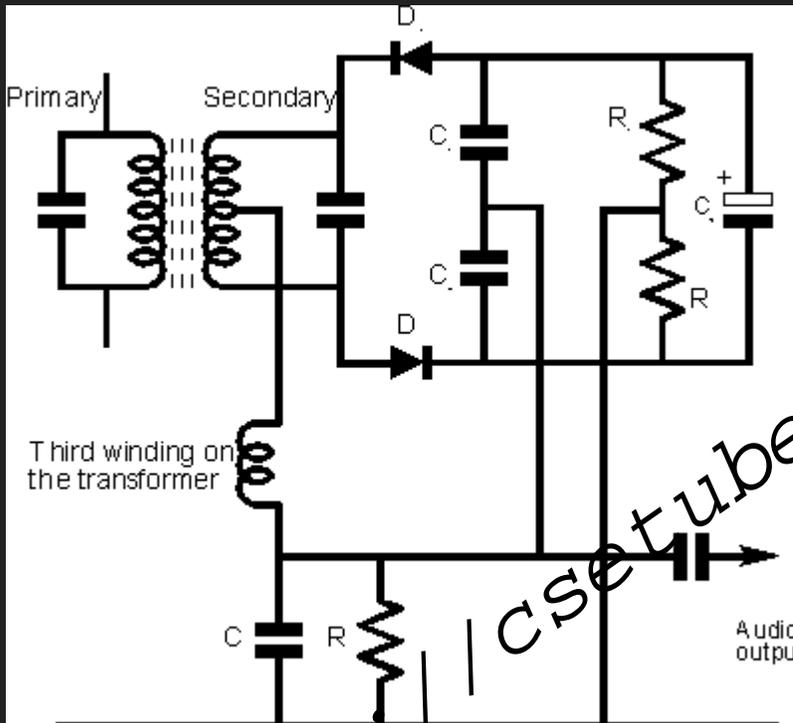
The operation of the ratio detector centres around a frequency sensitive phase shift network with a transformer and the diodes that are effectively in series with one another. When a steady carrier is applied to the circuit the diodes act to produce a steady voltage across the resistors R1 and R2, and the capacitor C3 charges up as a result.

The transformer enables the circuit to detect changes in the frequency of the incoming signal. It has three windings. The primary and secondary act in the normal way to produce a signal at the output. The third winding is un-tuned and the coupling between the primary and the third winding is very tight, and this means that the phasing between signals in these two windings is the same.

The primary and secondary windings are tuned and lightly coupled. This means that there is a phase difference of 90 degrees between the signals in these windings at the centre frequency. If the signal moves away from the centre frequency the phase difference will change. In turn the phase difference between the secondary and third windings also varies.

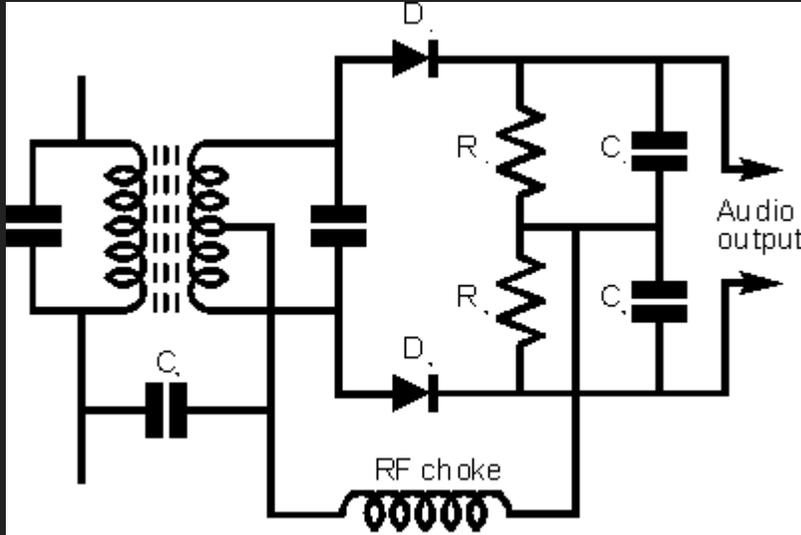
When this occurs the voltage will subtract from one side of the secondary and add to the other causing an imbalance across the resistors R1 and R2. As a result this causes a current to flow in the third winding and the modulation to appear at the output.

The capacitors C1 and C2 filter any remaining RF signal which may appear across the resistors. The capacitor C4 and R3 also act as filters ensuring no RF reaches the audio section of the receiver.



The ratio detector

The Foster Seeley detector has many similarities to the ratio detector. The circuit topology looks very similar, having a transformer and a pair of diodes, but there is no third winding and instead a choke is used.



The Foster-Seeley detector

Like the ratio detector, the Foster-Seeley circuit operates using a phase difference between signals. To obtain the different phased signals a connection is made to the primary side of the transformer using a capacitor, and this is taken to the centre tap of the transformer. This gives a signal that is 90 degrees out of phase.

When an un-modulated carrier is applied at the centre frequency, both diodes conduct, to produce equal and opposite voltages across their respective load resistors. These voltages cancel each other out at the output so that no voltage is present. As the carrier moves off to one side of the centre frequency the balance condition is destroyed, and one diode conducts more than the other. This results in the voltage across one of the resistors being larger than the other, and a resulting voltage at the output corresponding to the modulation on the incoming signal.

The choke is required in the circuit to ensure that no RF signals appear at the output. The capacitors C1 and C2 provide a similar filtering function.

Both the ratio and Foster-Seeley detectors are expensive to manufacture. Wound components like coils are not easy to produce to the required specification and therefore they are comparatively costly. Accordingly these circuits are rarely used in modern equipment.

Quadrature FM detector

Another form of FM detector or demodulator that can be these days is called the quadrature detector. It lends itself to use with integrated circuits and as a result it is in widespread use. It has the advantage over the ratio and Foster-Seeley detectors that it only requires a simple tuned circuit.

For the quadrature detector, the signal is split into two components. One passes through a network that provides a basic 90 degree phase shift, plus an element of phase shift dependent upon the deviation and into one port of a mixer. The other is passed straight into another port of the mixer. The output from the mixer is proportional to the phase difference between the two signals, i.e. it acts as a phase detector and produces a voltage output that is proportional to the phase difference and hence to the level of deviation on the signal.

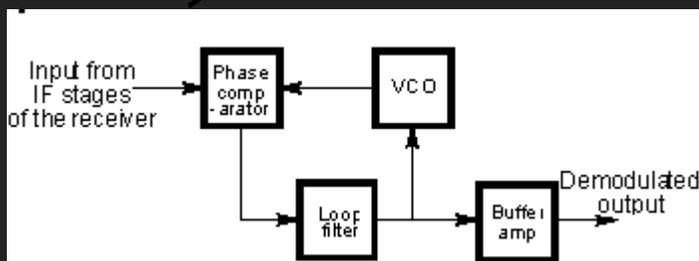
The detector is able to operate with relatively low input levels, typically down to levels of around 100 microvolts and it is very easy to set up requiring only the phase shift network to be tuned to the centre frequency of the expected signal. It also provides good linearity enabling very low levels of distortion to be achieved.

Often the analogue multiplier is replaced by a logic AND gate. The input signal is hard limited to produce a variable frequency pulse waveform. The operation of the circuit is fundamentally the same, but it is known as a coincidence detector. Also the output of the AND gate has an integrator to "average" the output waveform to provide the required audio output, otherwise it would consist of a series of square wave pulses.

Phase locked loop (PLL)

Another popular form of FM demodulator comes in the form of a phase locked loop. Like the quadrature detector, phase locked loops do not need to use a coil, and therefore they make a very cost effective form of demodulator.

The way in which they operate is very simple. The loop consists of a phase detector into which the incoming signal is passed, along with the output from the voltage controlled oscillator (VCO) contained within the phase locked loop. The output from the phase detector is passed into a loop filter and then used as the control voltage for the VCO.



Phase locked loop (PLL) FM demodulator

With no modulation applied and the carrier in the centre position of the pass-band the voltage on the tune line to the VCO is set to the mid position. However if the carrier deviates in frequency, the loop will try to keep the loop in lock. For this to happen the VCO frequency must follow the incoming signal, and for this to occur the tune line

voltage must vary. Monitoring the tune line shows that the variations in voltage correspond to the modulation applied to the signal. By amplifying the variations in voltage on the tune line it is possible to generate the demodulated signal.

It is found that the linearity of this type of detector is governed by the voltage to frequency characteristic of the VCO. As it normally only swings over a small portion of its bandwidth, and the characteristic can be made relatively linear, the distortion levels from phase locked loop demodulators are normally very low.

Frequency Vs Phase Modulation:

The difference between FM & PM in a digital oscillator is that FM is added to the frequency before the phase integration, while PM is added to the phase after the phase integration. Phase integration is when the old phase for the oscillator is added to the current frequency (in radians per sample) to get the new phase for the oscillator. The equivalent PM modulator to obtain the same waveform as FM is the integral of the FM modulator. Since the integral of sine waves are inverted cosine waves this is no problem. In modulators with multiple partials, the equivalent PM modulator will have different relative partial amplitudes. For example, the integral of a square wave is a triangle wave; they have the same harmonic content, but the relative partial amplitudes are different. These differences make no difference since we are not trying to exactly recreate FM, but real (or nonreal) instruments.

The reason PM is better is because in PM and FM there can be non-zero energy produced at 0 Hz, which in FM will produce a shift in pitch if the FM wave is used again as a modulator, however in PM the DC component will only produce a phase shift. Another reason PM is better is that the modulation index (which determines the number of sidebands produced and which in normal FM is calculated as the modulator amplitude divided by frequency of modulator) is not dependant on the frequency of the modulator, it is always equal to the amplitude of the modulator in radians. The benefit of solving the DC frequency shift problem, is that cascaded carrier-modulator pairs and feedback modulation are possible. The simpler calculation of modulation index makes it easier to have voices keep the same harmonic structure throughout all pitches.

Important Questions: UNIT I

PART A

1. Define modulation.
2. What is the need for modulation?
3. Draw the block diagram of communication system.
4. Give the types of modulation.
5. What is amplitude modulation
6. Define sensitivity
7. For an AM commercial broadcast band receiver (535KHz-1605KHz) with an input filter Q factor of 54, determine the bandwidth at low and high ends of the RF spectrum.
8. Define AM envelope
9. Draw the frequency spectrum of an AMDSBFC wave.
10. Define modulation index and percent modulation in AM.
11. What are the types of conventional AM modulators. State the difference.
12. What is meant by repetition rate of an AM envelope.
13. When there is 100% modulation, what is the relationship between voltage amplitudes of the side frequencies and the carrier.
14. What is the predominant disadvantage of AMDSBFC.
15. State disadvantage of low level modulator (AM).
16. What is the maximum modulating frequency that can be used with an AMDSBFC system with a 20KHz bandwidth.
17. For an unmodulated carrier amplitude of 16V, and a modulation coefficient $m=0.4$, determine the amplitudes of the modulated carrier and side frequencies.
18. For $m=0.4$, $P_c=2000W$, Determine i) Total sideband power ii) Total transmitted power
iii) Power of the modulated and unmodulated carrier
19. What is image frequency. Give the expression of IFRR.
20. For an AM receiver using high side injection with an RF carrier of 27MHz & an IF center frequency of 455KHz, determine Local oscillator frequency, Image frequency, IFRR for a preselector Q of 100
21. What is heterodyning.

PART-B

1. Explain with neat sketches the different types of amplitude modulator.
2. Explain the voltage distribution of AM wave.
3. For an AM modulator with $f_c=200kHz$ and $f_m(\max) = 10KHz$ Determine
 - a. limits for upper and lower sidebands,
 - b. bandwidth,
 - c. upper and lower side frequency when $f_m=3kHz$ tone
 - d. Draw output frequency spectrum.
4. Explain AM Receiver parameters
5. Draw and explain the block diagram of low level AM transmitter
6. Draw and explain the block diagram of high level AM transmitter
7. Explain the working of a super heterodyne receiver with suitable block diagram.
8. Explain the working of a tuned radiofrequency receiver with suitable block diagram.
9. Explain the working of a double conversion AM receiver with suitable block diagram.

UNIT II DIGITAL COMMUNICATION

Basics- Shannon Limit for Information Capacity – Digital Amplitude Modulation –

Frequency Shift Keying – FSK Bit Rate and Baud – FSK Transmitter – BW

Consideration of FSK – FSK Receiver – Phase Shift Keying – Binary Phase Shift

Keying – QPSK – Quadrature Amplitude Modulation – Bandwidth Efficiency –

Carrier Recovery – Squaring Loop – Costas Loop – DPSK.

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UNIT II DIGITAL COMMUNICATION

The techniques used to modulate digital information so that it can be transmitted via microwave, satellite or down a cable pair are different to that of analogue transmission. The data transmitted via satellite or microwave is transmitted as an analogue signal. The techniques used to transmit analogue signals are used to transmit digital signals. The problem is to convert the digital signals to a form that can be treated as an analogue signal that is then in the appropriate form to either be transmitted down a twisted cable pair or applied to the RF stage where is modulated to a frequency that can be transmitted via microwave or satellite.

The equipment that is used to convert digital signals into analogue format is a modem. The word modem is made up of the words –modulator and –demodulator.

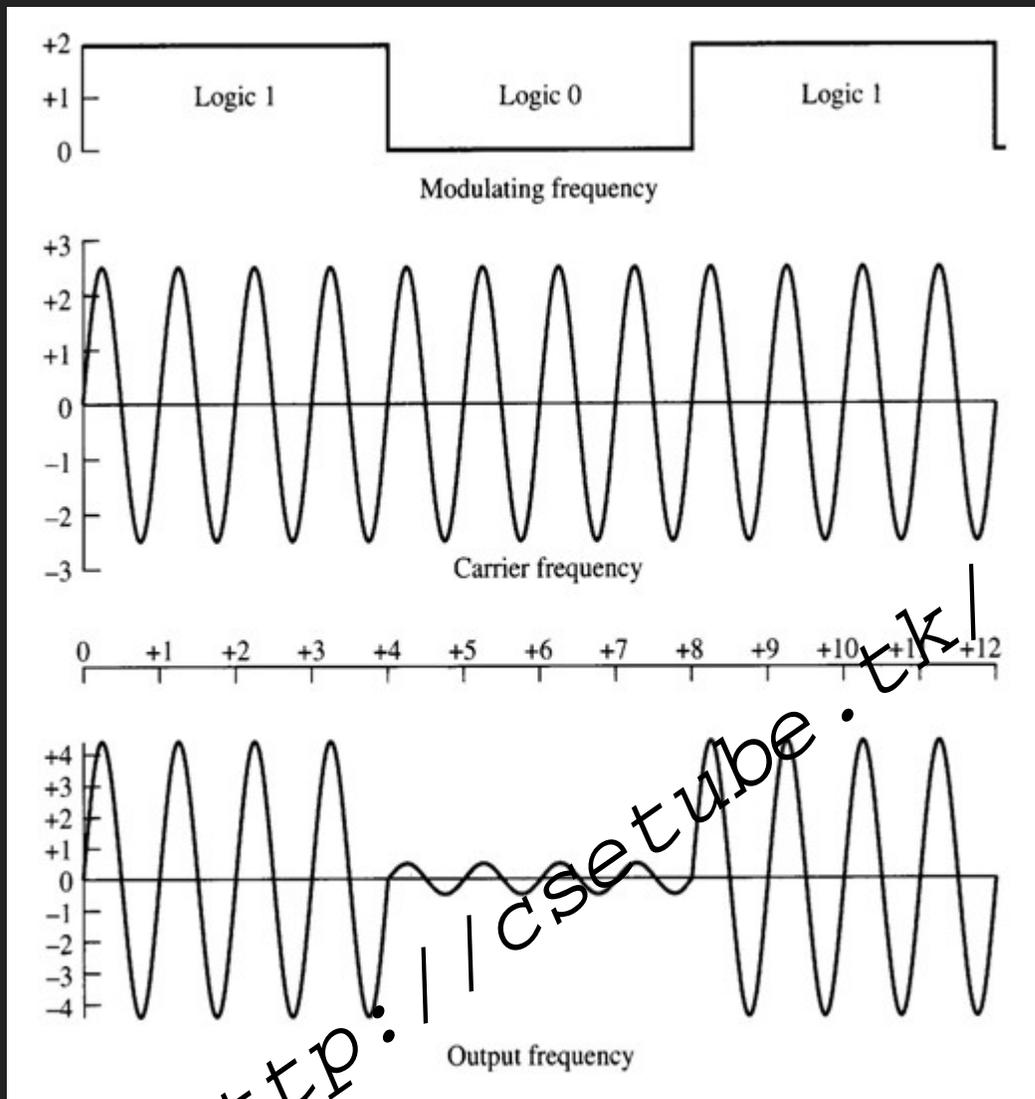
A modem accepts a serial data stream and converts it into an analogue format that matches the transmission medium.

There are many different modulation techniques that can be utilised in a modem. These techniques are:

- Amplitude shift key modulation (ASK)
- Frequency shift key modulation (FSK)
- Binary-phase shift key modulation (BPSK)
- Quadrature-phase shift key modulation (QPSK)
- Quadrature amplitude modulation (QAM)

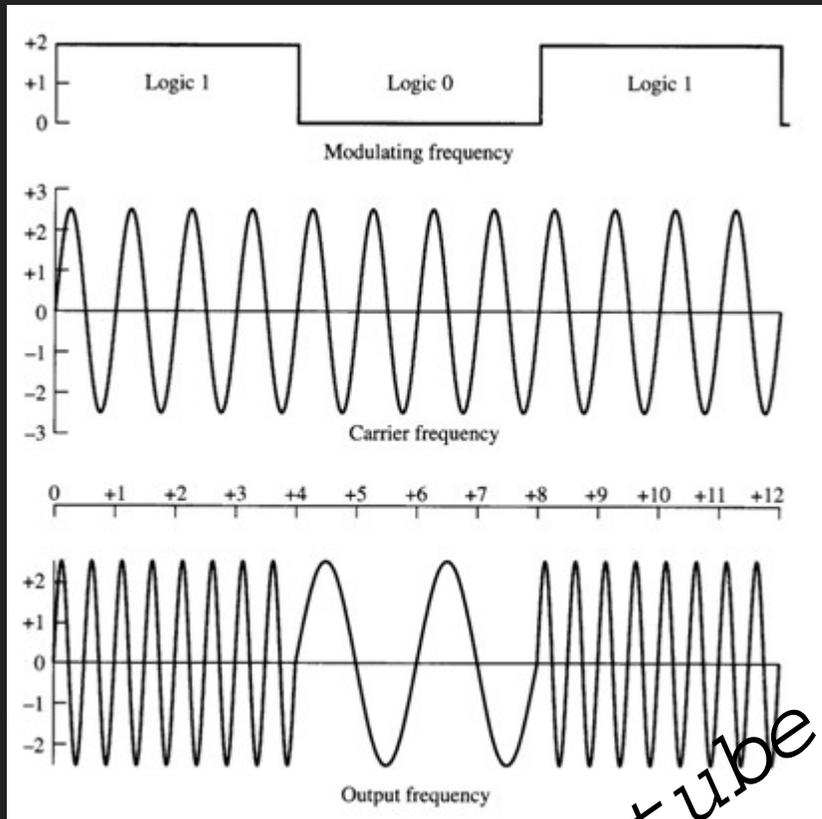
Amplitude Shift Key Modulation

In this method the amplitude of the carrier assumes one of the two amplitudes dependent on the logic states of the input bit stream. A typical output waveform of an ASK modulator is shown in the figure below. The frequency components are the USB and LSB with a residual carrier frequency. The low amplitude carrier is allowed to be transmitted to ensure that at the receiver the logic 1 and logic 0 conditions can be recognised uniquely.



Frequency Shift Key Modulation

In this method the frequency of the carrier is changed to two different frequencies depending on the logic state of the input bit stream. The typical output waveform of an FSK is shown below. Notice that a logic high causes the centre frequency to increase to a maximum and a logic low causes the centre frequency to decrease to a minimum.



Phase Shift Key Modulation

With this method the phase of the carrier changes between different phases determined by the logic states of the input bit stream.

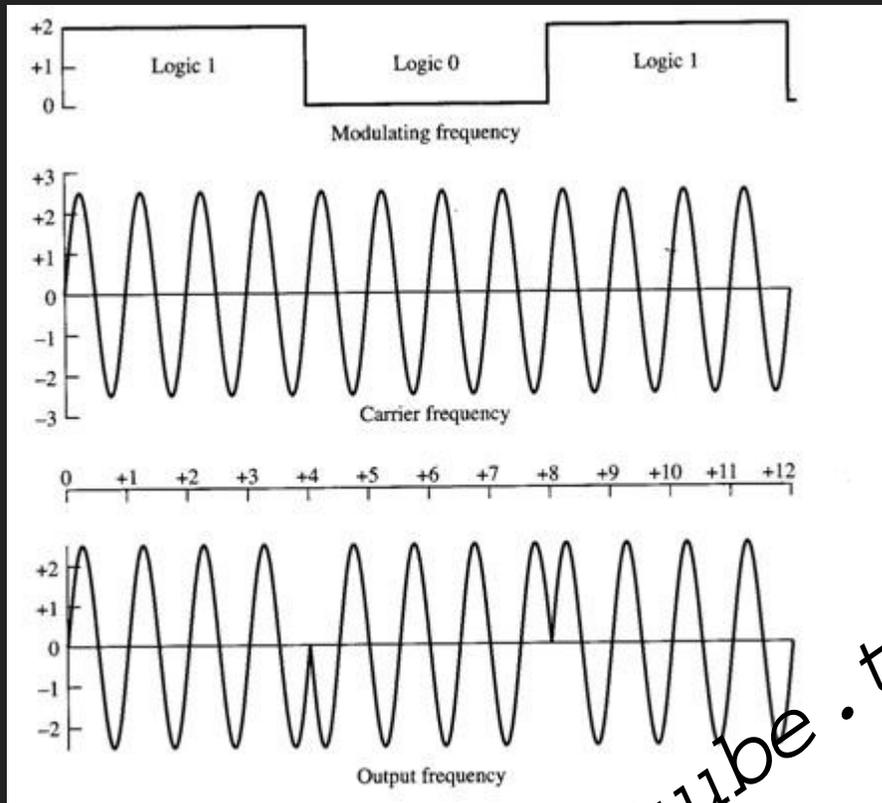
There are several different types of phase shift key (PSK) modulators.

- Two-phase (2 PSK)
- Four-phase (4 PSK)
- Eight-phase (8 PSK)
- Sixteen-phase (16 PSK)
- Sixteen-quadrature amplitude (16 QAM)

The 16 QAM is a composite modulator consisting of amplitude modulation and phase modulation. The 2 PSK, 4 PSK, 8 PSK and 16 PSK modulators are generally referred to as binary phase shift key (BPSK) modulators and the QAM modulators are referred to as quadrature phase shift key (QPSK) modulators.

Two-Phase Shift Key Modulation

In this modulator the carrier assumes one of two phases. A logic 1 produces no phase change and a logic 0 produces a 180° phase change. The output waveform for this modulator is shown below.



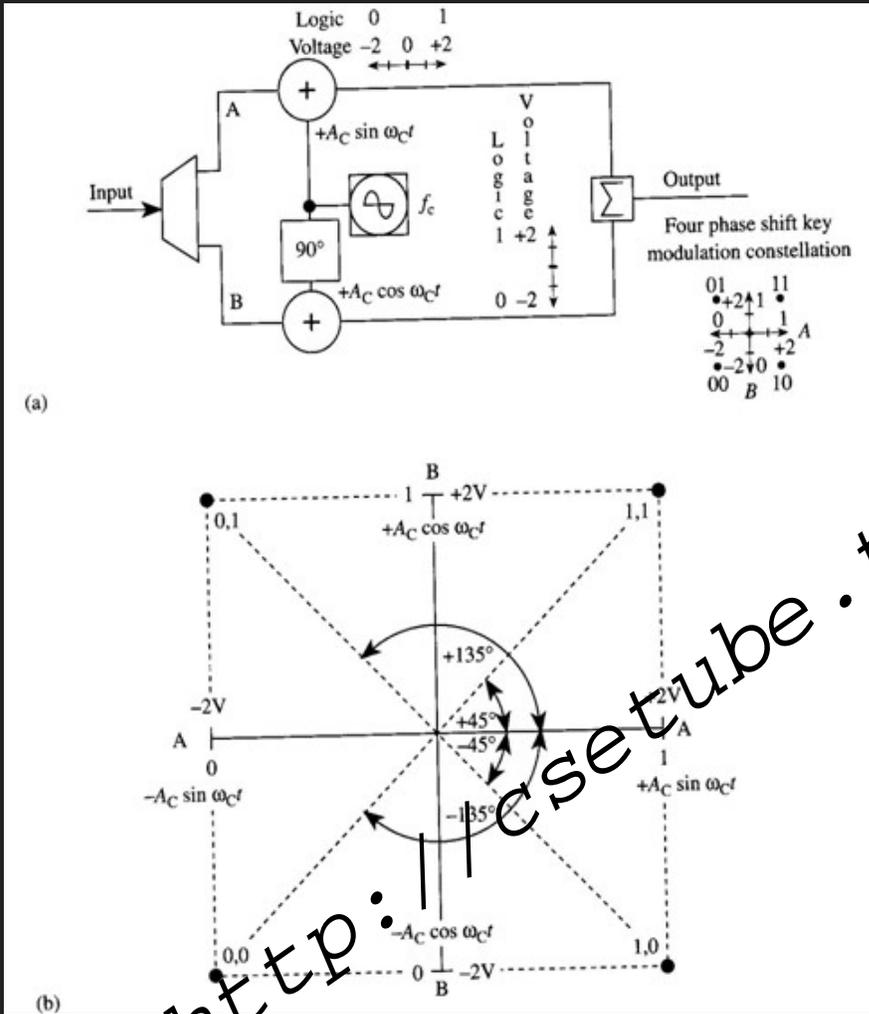
Four-Phase Shift Key Modulation

With 4 PSK, 2 bits are processed to produce a single phase change. In this case each symbol consists of 2 bits, which are referred to as a dibit. The actual phases that are produced by a 4 PSK modulator are shown in the table below.

Dibit	Phase Change
00	+225°/-135°
01	+135°/-225°
10	+315°/-45°
11	+45°/-315°

Figure 8: PSK Table

Because the output bit rate is less than the input bit rate, this results in a smaller bandwidth. A typical 4 PSK circuit and the constellation is shown below.



Eight-Phase Shift Key Modulation

With this modulator 3 bits are processed to produce a single phase change. This means that each symbol consists of 3 bits.

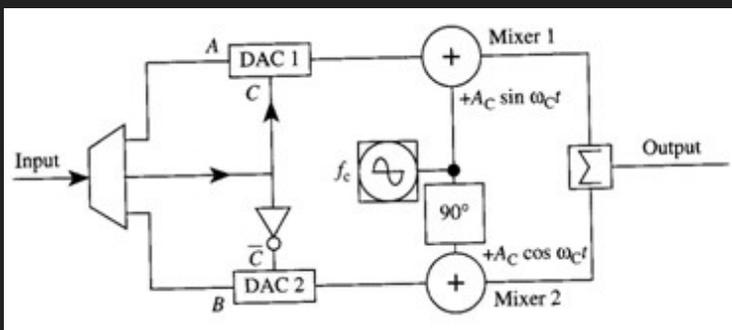


Figure 10: 8 PSK Modulator

Figure 10 above shows a typical circuit for the 8 PSK modulator. With this modulator bit A controls the output polarity of the first digital-to-analogue converter (DAC1). Bit B is used to control the output polarity of the second DAC 2 and bit C is used to control the output amplitude of both DACs.

A	Polarity	B	Polarity	C	Amplitude	\bar{C}	Amplitude
0	-	0	-	0	0.5	1	1.21
1	+	1	+	1	1.21	0	0.5

Figure 11: Digital to Analogue Conversion Condition for 8 PSK modulator

The conditions shown in the table above (Figure 11) produce the positions shown in table below (Figure 12) for all the different permutations.

\bar{C}	C	B	A	C	A	Polarity	Amplitude	\bar{C}	B	Polarity	Amplitude	Quad
1	0	0	0	0	0	-	0.5	1	0	-	1.21	3
1	0	0	1	0	1	+	0.5	1	0	-	1.21	4
1	0	1	0	0	0	-	0.5	1	1	+	1.21	2
1	0	1	1	0	1	+	0.5	1	1	+	1.21	1
0	1	0	0	1	0	-	1.21	0	0	-	0.5	3
0	1	0	1	1	1	+	1.21	0	0	-	0.5	4
0	1	1	0	1	0	-	1.21	0	1	+	0.5	2
0	1	1	1	1	1	+	1.21	0	1	+	0.5	1

Figure 12: Input permutations and Positions

The constellation diagram can be drawn according to the above table and is shown below.

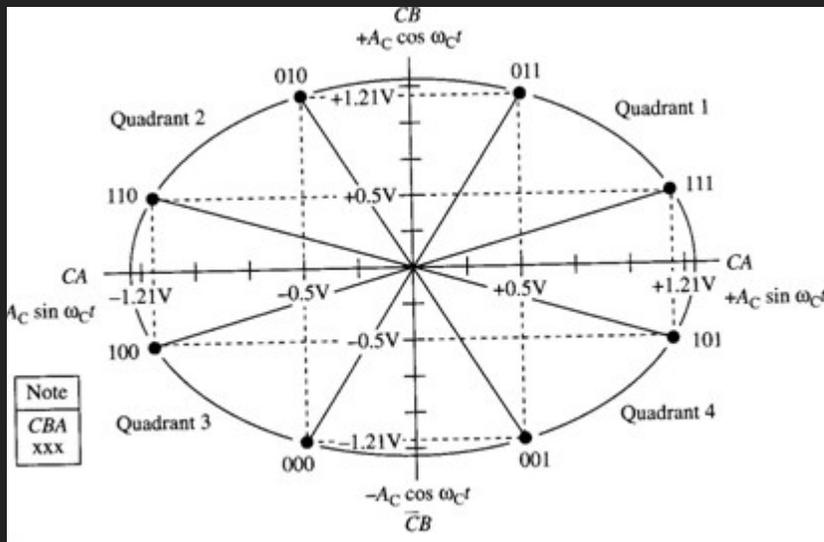


Figure 13: 16 PSK Constellation

Sixteen-Phase Shift Key Modulation

With this modulator 4 bits are processed to produce a single phase change. This means that each symbol consists of 4 bits. The constellation for this modulator scheme is shown below.

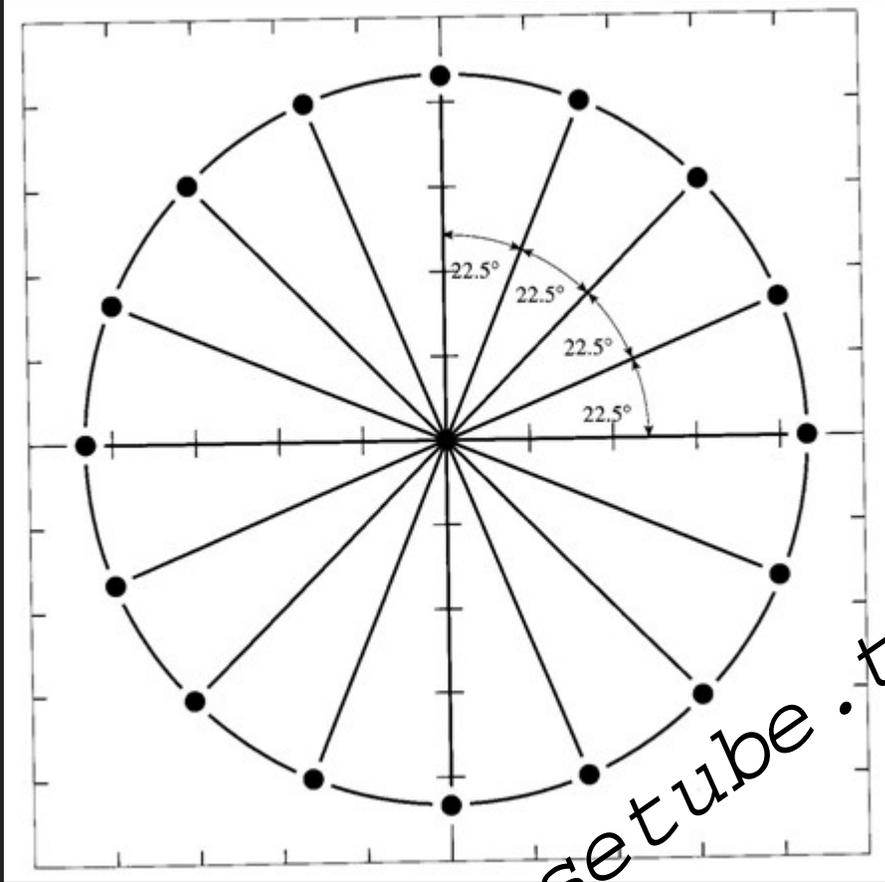


Figure 14: 16 PSK Modulation Constellation

Sixteen-Quadrature Amplitude Modulation

With this modulator, 4 bits are processed to produce a single vector. The resultant constellation consists of three different amplitudes distributed in 12 different phases as shown below.

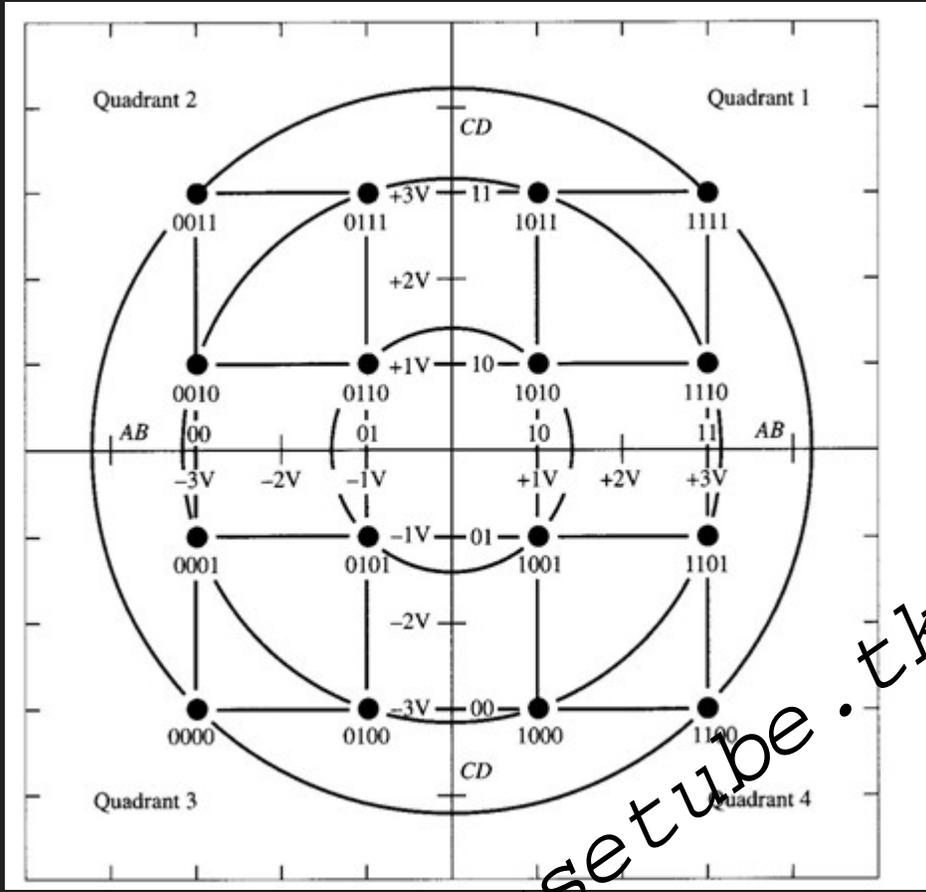


Figure 15: 16 QAM Constellation

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Important Questions:UNIT II

PART A

1. Define Angle modulation
2. What is frequency modulation?
3. Define frequency deviation.give an expression for modulation index.
4. What is the bandwidth required for an FM signal in which the modulating frequency is 2KHz and maximum deviation is 10KHz.
5. What is deviation ratio in FM.
6. If frequency deviation is 5Khz for a 10v modulating signal, determine deviation sensitivity.
7. What are the types of FM demodulators?
8. What is the bandwidth required for an FM signal in which the modulating frequency is 2KHz and maximum deviation is 10KHz.
9. What is phase modulation?.
- 10 What are the types of FM modulators?
11. What are the types of PM modulators & demodulators??
12. Define Adjacent channel interference
13. Define Carson's rule.
14. calculate the bandwidth using carson's rule for maximum frequency deviation and modulating signal
15. Give the modulation index for FM and PM

PART B

1. Explain the working of FET Reactance modulator with circuit diagram.
2. Explain the working of Varactor diode FM modulator.
3. Explain the working of direct PM modulators
4. Explain the working of foster seeley discriminator with diagram.
5. Sketch and explain the working of direct FM transmitter.
6. Sketch and explain the PLL FM demodulator.
7. Explain about Direct FM transmitter.
8. Explain about Indirect FM transmitter
9. Explain the commercial Broadcast band FM.
- 10.Explain about FM Noise Suppression

UNIT III DIGITAL TRANSMISSION 9

Basics – Pulse Modulation – PCM – PCM Sampling – Sampling Rate – Signal to

Quantization Noise Rate – Companding – Analog and Digital – Percentage Error –

Delta Modulation – Adaptive Delta Modulation – Differential Pulse Code Modulation

– Pulse Transmission – Intersymbol Interference – Eye Patterns.

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UNIT III DIGITAL TRANSMISSION :

PULSE MODULATION :

Pulse-width modulation (PWM) is a commonly used technique for controlling power to inertial electrical devices, made practical by modern electronic power switches.

The average value of voltage (and current) fed to the load is controlled by turning the switch between supply and load on and off at a fast pace. The longer the switch is on compared to the off periods, the higher the power supplied to the load is.

The PWM switching frequency has to be much faster than what would affect the load, which is to say the device that uses the power. Typically switchings have to be done several times a minute in an electric stove, 120 Hz in a lamp dimmer, from few kilohertz (kHz) to tens of kHz for a motor drive and well into the tens or hundreds of kHz in audio amplifiers and computer power supplies.

The term duty cycle describes the proportion of 'on' time to the regular interval or 'period' of time; a low duty cycle corresponds to low power, because the power is off for most of the time. Duty cycle is expressed in percent, 100% being fully on.

The main advantage of PWM is that power loss in the switching devices is very low. When a switch is off there is practically no current, and when it is on, there is almost no voltage drop across the switch. Power loss, being the product of voltage and current, is thus in both cases close to zero. PWM works also well with digital controls, which, because of their on/off nature, can easily set the needed duty cycle.

PWM has also been used in certain communication systems where its duty cycle has been used to convey information over a communications channel.

Principle

Fig. 1: a pulse wave, showing the definitions of y_{min} , y_{max} and D .

Pulse-width modulation uses a rectangular pulse wave whose pulse width is modulated resulting in the variation of the average value of the waveform. If we consider a pulse waveform $f(t)$ with a low value y_{min} , a high value y_{max} and a duty cycle D (see figure 1), the average value of the waveform is given by:

As $f(t)$ is a pulse wave, its value is y_{max} for and y_{min} for . The above expression then becomes:

This latter expression can be fairly simplified in many cases where $y_{min} = 0$ as . From this, it is obvious that the average value of the signal () is directly dependent on the duty cycle D .

Fig. 2: A simple method to generate the PWM pulse train corresponding to a given signal is the intersective PWM: the signal (here the green sinewave) is compared with a sawtooth waveform (blue). When the latter is less than the former, the PWM signal (magenta) is in high state (1). Otherwise it is in the low state (0).

The simplest way to generate a PWM signal is the intersective method, which requires only a sawtooth or a triangle waveform (easily generated using a simple oscillator) and a comparator. When the value of the reference signal (the green sine wave in figure 2) is more than the modulation waveform (blue), the PWM signal (magenta) is in the high state, otherwise it is in the low state.

Delta

In the use of delta modulation for PWM control, the output signal is integrated, and the result is compared with limits, which correspond to a reference signal offset by a constant. Every time the integral of the output signal reaches one of the limits, the PWM signal changes state.

Fig. 3 : Principle of the delta PWM. The output signal (blue) is compared with the limits (green). These limits correspond to the reference signal (red), offset by a given value. Every time the output signal reaches one of the limits, the PWM signal changes state.

Delta-sigma

In delta-sigma modulation as a PWM control method, the output signal is subtracted from a reference signal to form an error signal. This error is integrated, and when the integral of the error exceeds the limits, the output changes state.

Fig. 4 : Principle of the sigma-delta PWM. The top green waveform is the reference signal, on which the output signal (PWM, in the middle plot) is subtracted to form the error signal (blue, in top plot). This error is integrated (bottom plot), and when the integral of the error exceeds the limits (red lines), the output changes state.

Space vector modulation

Space vector modulation is a PWM control algorithm for multi-phase AC generation, in which the reference signal is sampled regularly; after each sample, non-zero active switching vectors adjacent to the reference vector and one or more of the zero switching vectors are selected for the appropriate fraction of the sampling period in order to synthesize the reference signal as the average of the used vectors.

Direct torque control (DTC)

Direct torque control is a method used to control AC motors. It is closely related with the delta modulation (see above). Motor torque and magnetic flux are estimated and these are controlled to stay within their hysteresis bands by turning on new combination of the

device's semiconductor switches each time either of the signal tries to deviate out of the band.

Time proportioning

Many digital circuits can generate PWM signals (e.g. many microcontrollers have PWM outputs). They normally use a counter that increments periodically (it is connected directly or indirectly to the clock of the circuit) and is reset at the end of every period of the PWM. When the counter value is more than the reference value, the PWM output changes state from high to low (or low to high).[2] This technique is referred to as time proportioning, particularly as time-proportioning control[3] – which proportion of a fixed cycle time is spent in the high state.

The incremented and periodically reset counter is the discrete version of the intersecting method's sawtooth. The analog comparator of the intersecting method becomes a simple integer comparison between the current counter value and the digital (possibly digitized) reference value. The duty cycle can only be varied in discrete steps, as a function of the counter resolution. However, a high-resolution counter can provide quite satisfactory performance.

Types

Fig. 5 : Three types of PWM signals (blue): leading edge modulation (top), trailing edge modulation (middle) and centered pulses (both edges are modulated, bottom). The green lines are the sawtooth waveform (first and second cases) and a triangle waveform (third case) used to generate the PWM waveforms using the intersective method.

Four types of pulse-width modulation (PWM) are possible:

1. The pulse center may be fixed in the center of the time window and both edges of the pulse moved to compress or expand the width.
2. The lead edge can be held at the lead edge of the window and the tail edge modulated.
3. The tail edge can be fixed and the lead edge modulated.
4. The pulse repetition frequency can be varied by the signal, and the pulse width can be constant. However, this method has a more-restricted range of average output than the other three.

Spectrum

The resulting spectra (of the three cases) are similar, and each contains a dc component, a base sideband containing the modulating signal and phase modulated carriers at each harmonic of the frequency of the pulse. The amplitudes of the harmonic groups are restricted by a sinc / x envelope (sinc function) and extend to infinity.

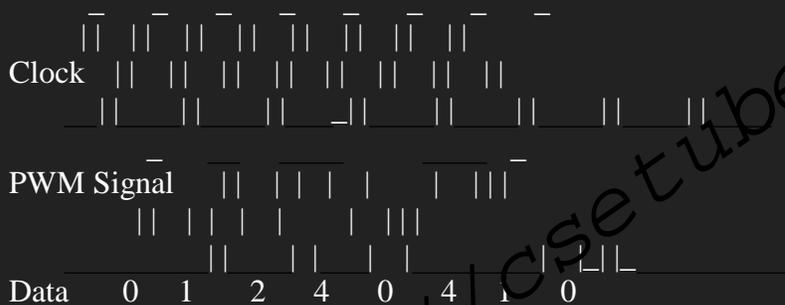
On the contrary, the delta modulation is a random process that produces continuous spectrum without distinct harmonics.

Applications

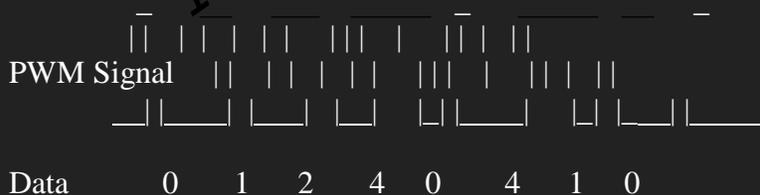
Telecommunications

In telecommunications, the widths of the pulses correspond to specific data values encoded at one end and decoded at the other.

Pulses of various lengths (the information itself) will be sent at regular intervals (the carrier frequency of the modulation).



The inclusion of a clock signal is not necessary, as the leading edge of the data signal can be used as the clock if a small offset is added to the data value in order to avoid a data value with a zero length pulse.



Power delivery

PWM can be used to adjust the total amount of power delivered to a load without losses normally incurred when a power transfer is limited by resistive means. The drawback are the pulsations defined by the duty cycle, switching frequency and properties of the load. With a sufficiently high switching frequency and, when necessary, using additional passive electronic filters the pulse train can be smoothed and average analog waveform recovered.

High frequency PWM power control systems are easily realisable with semiconductor switches. As has been already stated above almost no power is dissipated by the switch in either on or off state. However, during the transitions between on and off states both voltage and current are non-zero and thus considerable power is dissipated in the switches. Luckily, the change of state between fully on and fully off is quite rapid (typically less than 100 nanoseconds) relative to typical on or off times, and so the average power dissipation is quite low compared to the power being delivered even when high switching frequencies are used.

Modern semiconductor switches such as MOSFETs or Insulated-gate bipolar transistors (IGBTs) are quite ideal components. Thus high efficiency controllers can be built. Typically frequency converters used to control AC motors have efficiency that is better than 98 %. Switching power supplies have lower efficiency due to low output voltage levels (often even less than 2 V for microprocessors are needed) but still more than 70-80 % efficiency can be achieved.

Variable-speed fan controllers for computers usually use PWM, as it is far more efficient when compared to a potentiometer or rheostat. (Neither of the latter is practical to operate electronically; they would require a small drive motor.)

Light dimmers for home use employ a specific type of PWM control. Home-use light dimmers typically include electronic circuitry which suppresses current flow during defined portions of each cycle of the AC line voltage. Adjusting the brightness of light emitted by a light source is then merely a matter of setting at what voltage (or phase) in the AC halfcycle the dimmer begins to provide electrical current to the light source (e.g. by using an electronic switch such as a triac). In this case the PWM duty cycle is the ratio of the conduction time to the duration of the half AC cycle defined by the frequency of the AC line voltage (50 Hz or 60 Hz depending on the country).

These rather simple types of dimmers can be effectively used with inert (or relatively slow reacting) light sources such as incandescent lamps, for example, for which the additional modulation in supplied electrical energy which is caused by the dimmer causes only negligible additional fluctuations in the emitted light. Some other types of light sources such as light-emitting diodes (LEDs), however, turn on and off extremely rapidly and would perceivably flicker if supplied with low frequency drive voltages. Perceivable flicker effects from such rapid response light sources can be reduced by increasing the PWM frequency. If the light fluctuations are sufficiently rapid, the human visual system can no longer resolve them and the eye perceives the time average intensity without flicker (see flicker fusion threshold).

In electric cookers, continuously-variable power is applied to the heating elements such as the hob or the grill using a device known as a Simmerstat. This consists of a thermal oscillator running at approximately two cycles per minute and the mechanism varies the duty cycle according to the knob setting. The thermal time constant of the heating elements is several minutes, so that the temperature fluctuations are too small to matter in practice.

Voltage regulation

PWM is also used in efficient voltage regulators. By switching voltage to the load with the appropriate duty cycle, the output will approximate a voltage at the desired level. The switching noise is usually filtered with an inductor and a capacitor.

One method measures the output voltage. When it is lower than the desired voltage, it turns on the switch. When the output voltage is above the desired voltage, it turns off the switch.

Audio effects and amplification

PWM is sometimes used in sound (music) synthesis, in particular subtractive synthesis, as it gives a sound effect similar to chorus or slightly detuned oscillators played together. (In fact, PWM is equivalent to the difference of two sawtooth waves. [1]) The ratio between the high and low level is typically modulated with a low frequency oscillator, or LFO. In addition, varying the duty cycle of a pulse waveform in a subtractive-synthesis instrument creates useful timbral variations. Some synthesizers have a duty-cycle trimmer for their square-wave outputs, and that trimmer can be set by ear; the 50% point was distinctive, because even-numbered harmonics essentially disappear at 50%.

A new class of audio amplifiers based on the PWM principle is becoming popular. Called "Class-D amplifiers", these amplifiers produce a PWM equivalent of the analog input signal which is fed to the loudspeaker via a suitable filter network to block the carrier and recover the original audio. These amplifiers are characterized by very good efficiency figures ($\geq 90\%$) and compact size/light weight for large power outputs. For a few decades, industrial and military PWM amplifiers have been in common use, often for driving servo motors. They offer very good efficiency, commonly well above 90%. Field-gradient coils in MRI machines are driven by relatively-high-power PWM amplifiers.

Historically, a crude form of PWM has been used to play back PCM digital sound on the PC speaker, which is driven by only two voltage levels, typically 0 V and 5 V. By carefully timing the duration of the pulses, and by relying on the speaker's physical filtering properties (limited frequency response, self-inductance, etc.) it was possible to obtain an approximate playback of mono PCM samples, although at a very low quality, and with greatly varying results between implementations.

In more recent times, the Direct Stream Digital sound encoding method was introduced, which uses a generalized form of pulse-width modulation called pulse density modulation, at a high enough sampling rate (typically in the order of MHz) to cover the whole acoustic frequencies range with sufficient fidelity. This method is used in the SACD format, and reproduction of the encoded audio signal is essentially similar to the method used in class-D amplifiers.

Companing

In telecommunication, signal processing, and thermodynamics, companing (occasionally called compansion) is a method of mitigating the detrimental effects of a channel with limited dynamic range. The name is a portmanteau of compressing and expanding.

While the compression used in audio recording and the like depends on a variable-gain amplifier, and so is a locally linear process (linear for short regions, but not globally), companing is non-linear and takes place in the same way at all points in time. The dynamic range of a signal is compressed before transmission and is expanded to the original value at the receiver.

The electronic circuit that does this is called a compondor and works by compressing or expanding the dynamic range of an analog electronic signal such as sound. One variety is a triplet of amplifiers: a logarithmic amplifier, followed by a variable-gain linear amplifier and an exponential amplifier. Such a triplet has the property that its output voltage is proportional to the input voltage raised to an adjustable power. Compondors are used in concert audio systems and in some noise reduction schemes such as dbx and Dolby NR (all versions).

Companing can also refer to the use of compression, where gain is decreased when levels rise above a certain threshold, and its complement, expansion, where gain is increased when levels drop below a certain threshold.

The use of companing allows signals with a large dynamic range to be transmitted over facilities that have a smaller dynamic range capability. For example, it is employed in professional wireless microphones since the dynamic range of the microphone audio signal itself is larger than the dynamic range provided by radio transmission. Companing also reduces the noise and crosstalk levels at the receiver.

Companing is used in digital and telephony systems, compressing before input to an analog-to-digital converter, and then expanding after a digital-to-analog converter. This is equivalent to using a non-linear ADC as in a T-carrier telephone system that implements A-law or μ -law companing. This method is also used in digital file formats for better signal-to-noise ratio (SNR) at lower bit rates. For example, a linearly encoded 16-bit PCM signal can be converted to an 8-bit WAV or AU file while maintaining a decent SNR by compressing before the transition to 8-bit and expanding after a conversion back to 16-bit. This is effectively a form of lossy audio data compression.

Many of the music equipment manufacturers (Roland, Yamaha, Korg) used companing for data compression in their digital synthesizers. This dates back to the late 80's when memory chips would often come as one the most costly parts in the instrument.

Manufacturers usually express the amount of memory as it is in the compressed form. i.e. 24MB waveform ROM in Korg Trinity is actually 48MB of data. Still the fact remains that the unit has a 24MB physical ROM. In the example of Roland SR-JV expansion boards, they usually advertised them as 8MB boards which contain '16MB-equivalent content'. Careless copying of the info and omitting the part that stated "equivalent" can often lead to confusion.

Pulse-code modulation

Pulse-code modulation (PCM) is a method used to digitally represent sampled analog signals, which was invented by Alec Reeves in 1937. It is the standard form for digital audio in computers and various Blu-ray, Compact Disc and DVD formats, as well as other uses such as digital telephone systems. A PCM stream is a digital representation of an analog signal, in which the magnitude of the analogue signal is sampled regularly at uniform intervals, with each sample being quantized to the nearest value within a range of digital steps.

PCM streams have two basic properties that determine their fidelity to the original analog signal: the sampling rate, which is the number of times per second that samples are taken; and the bit depth, which determines the number of possible digital values that each sample can take.

Modulation

Sampling and quantization of a signal (red) for 4-bit PCM

In the diagram, a sine wave (red curve) is sampled and quantized for pulse code modulation. The sine wave is sampled at regular intervals, shown as ticks on the x-axis. For each sample, one of the available values (ticks on the y-axis) is chosen by some algorithm. This produces a fully discrete representation of the input signal (shaded area) that can be easily encoded as digital data for storage or manipulation. For the sine wave example at right, we can verify that the quantized values at the sampling moments are 7, 9, 11, 12, 13, 14, 14, 15, 15, 15, 14, etc. Encoding these values as binary numbers would result in the following set of nibbles: 0111 ($2^3*0+2^2*1+2^2*1+2^0*1=0+4+2+1=7$), 1001, 1011, 1100, 1101, 1110, 1110, 1111, 1111, 1111, 1110, etc. These digital values could then be further processed or analyzed by a purpose-specific digital signal processor or general purpose DSP. Several Pulse Code Modulation streams could also be multiplexed into a larger aggregate data stream, generally for transmission of multiple streams over a single physical link. One technique is called time-division multiplexing, or TDM, and is widely used, notably in the modern public telephone system. Another technique is called Frequency-division multiplexing, where the signal is assigned a

frequency in a spectrum, and transmitted along with other signals inside that spectrum. Currently, TDM is much more widely used than FDM because of its natural compatibility with digital communication, and generally lower bandwidth requirements.

There are many ways to implement a real device that performs this task. In real systems, such a device is commonly implemented on a single integrated circuit that lacks only the clock necessary for sampling, and is generally referred to as an ADC (Analog-to-Digital converter). These devices will produce on their output a binary representation of the input whenever they are triggered by a clock signal, which would then be read by a processor of some sort.

Demodulation

To produce output from the sampled data, the procedure of modulation is applied in reverse. After each sampling period has passed, the next value is read and a signal is shifted to the new value. As a result of these transitions, the signal will have a significant amount of high-frequency energy. To smooth out the signal and remove these undesirable aliasing frequencies, the signal would be passed through analog filters that suppress energy outside the expected frequency range (that is, greater than the Nyquist frequency $f_s / 2$). Some systems use digital filtering to remove some of the aliasing, converting the signal from digital to analog at a higher sample rate such that the analog filter required for anti-aliasing is much simpler. In some systems, no explicit filtering is done at all; as it's impossible for any system to reproduce a signal with infinite bandwidth, inherent losses in the system compensate for the artifacts — or the system simply does not require much precision. The sampling theorem suggests that practical PCM devices, provided a sampling frequency that is sufficiently greater than that of the input signal, can operate without introducing significant distortions within their designed frequency bands.

The electronics involved in producing an accurate analog signal from the discrete data are similar to those used for generating the digital signal. These devices are DACs (digital-to-analog converters), and operate similarly to ADCs. They produce on their output a voltage or current (depending on type) that represents the value presented on their inputs. This output would then generally be filtered and amplified for use.

Limitations

There are two sources of impairment implicit in any PCM system:

Choosing a discrete value near the analog signal for each sample leads to quantization error, which swings between $-q/2$ and $q/2$. In the ideal case (with a fully linear ADC) it is uniformly distributed over this interval, with zero mean and variance of $q^2/12$.

Between samples no measurement of the signal is made; the sampling theorem guarantees non-ambiguous representation and recovery of the signal only if it has no energy at frequency $f_s/2$ or higher (one half the sampling frequency, known as

the Nyquist frequency); higher frequencies will generally not be correctly represented or recovered.

As samples are dependent on time, an accurate clock is required for accurate reproduction. If either the encoding or decoding clock is not stable, its frequency drift will directly affect the output quality of the device. A slight difference between the encoding and decoding clock frequencies is not generally a major concern; a small constant error is not noticeable. Clock error does become a major issue if the clock is not stable, however. A drifting clock, even with a relatively small error, will cause very obvious distortions in audio and video signals, for example.

Extra information: PCM data from a master with a clock frequency that can not be influenced requires an exact clock at the decoding side to ensure that all the data is used in a continuous stream without buffer underrun or buffer overflow. Any frequency difference will be audible at the output since the number of samples per time interval can not be correct. The data speed in a compact disk can be steered by means of a servo that controls the rotation speed of the disk; here the output clock is the master clock. For all "external master" systems like DAB the output stream must be decoded with a regenerated and exact synchronous clock. When the wanted output sample rate differs from the incoming data stream clock then a sample rate converter must be inserted in the chain to convert the samples to the new clock domain.

Digitization as part of the PCM process

In conventional PCM, the analog signal may be processed (e.g., by amplitude compression) before being digitized. Once the signal is digitized, the PCM signal is usually subjected to further processing (e.g., digital data compression).

PCM with linear quantization is known as Linear PCM (LPCM).[1]

Some forms of PCM combine signal processing with coding. Older versions of these systems applied the processing in the analog domain as part of the A/D process; newer implementations do so in the digital domain. These simple techniques have been largely rendered obsolete by modern transform-based audio compression techniques.

DPCM encodes the PCM values as differences between the current and the predicted value. An algorithm predicts the next sample based on the previous samples, and the encoder stores only the difference between this prediction and the actual value. If the prediction is reasonable, fewer bits can be used to represent the same information. For audio, this type of encoding reduces the number of bits required per sample by about 25% compared to PCM.

Adaptive DPCM (ADPCM) is a variant of DPCM that varies the size of the quantization step, to allow further reduction of the required bandwidth for a given signal-to-noise ratio.

Delta modulation is a form of DPCM which uses one bit per sample.

In telephony, a standard audio signal for a single phone call is encoded as 8,000 analog samples per second, of 8 bits each, giving a 64 kbit/s digital signal known as DS0. The default signal compression encoding on a DS0 is either μ -law (mu-law) PCM (North America and Japan) or A-law PCM (Europe and most of the rest of the world). These are logarithmic compression systems where a 12 or 13-bit linear PCM sample number is mapped into an 8-bit value. This system is described by international standard G.711. An alternative proposal for a floating point representation, with 5-bit mantissa and 3-bit radix, was abandoned.

Where circuit costs are high and loss of voice quality is acceptable, it sometimes makes sense to compress the voice signal even further. An ADPCM algorithm is used to map a series of 8-bit μ -law or A-law PCM samples into a series of 4-bit ADPCM samples. In this way, the capacity of the line is doubled. The technique is detailed in the G.726 standard.

Later it was found that even further compression was possible and additional standards were published. Some of these international standards describe systems and ideas which are covered by privately owned patents and thus use of these standards requires payments to the patent holders.

Some ADPCM techniques are used in Voice over IP communications.

Encoding for transmission

Pulse-code modulation can be either return-to-zero (RZ) or non-return-to-zero (NRZ). For a NRZ system to be synchronized using in-band information, there must not be long sequences of identical symbols, such as ones or zeroes. For binary PCM systems, the density of 1-symbols is called ones-density.[2]

Ones-density is often controlled using precoding techniques such as Run Length Limited encoding, where the PCM code is expanded into a slightly longer code with a guaranteed bound on ones-density before modulation into the channel. In other cases, extra framing bits are added into the stream which guarantee at least occasional symbol transitions.

Another technique used to control ones-density is the use of a scrambler polynomial on the raw data which will tend to turn the raw data stream into a stream that looks pseudo-random, but where the raw stream can be recovered exactly by reversing the effect of the polynomial. In this case, long runs of zeroes or ones are still possible on the output, but are considered unlikely enough to be within normal engineering tolerance.

In other cases, the long term DC value of the modulated signal is important, as building up a DC offset will tend to bias detector circuits out of their operating range. In this case special measures are taken to keep a count of the cumulative DC offset, and to modify the codes if necessary to make the DC offset always tend back to zero.

Many of these codes are bipolar codes, where the pulses can be positive, negative or absent. In the typical alternate mark inversion code, non-zero pulses alternate between being positive and negative. These rules may be violated to generate special symbols used for framing or other special purposes

Modulation

In electronics, modulation is the process of varying one or more properties of a high frequency periodic waveform, called the carrier signal, with respect to a modulating signal. This is done in a similar fashion as a musician may modulate a tone (a periodic waveform) from a musical instrument by varying its volume, timing and pitch. The three key parameters of a periodic waveform are its amplitude ("volume"), its phase ("timing") and its frequency ("pitch"), all of which can be modified in accordance with a low frequency signal to obtain the modulated signal. Typically a high-frequency sinusoid waveform is used as carrier signal, but a square wave pulse train may also occur.

In telecommunications, modulation is the process of conveying a message signal, for example a digital bit stream or an analog audio signal, inside another signal that can be physically transmitted. Modulation of a sine waveform is used to transform a baseband message signal into a passband signal, for example low-frequency audio signal into a radio-frequency signal (RF signal). In radio communications, cable TV systems or the public switched telephone network for instance, electrical signals can only be transferred over a limited passband frequency spectrum, with specific (non-zero) lower and upper cutoff frequencies. Modulating a sine wave carrier makes it possible to keep the frequency content of the transferred signal as close as possible to the centre frequency (typically the carrier frequency) of the passband.

In music synthesizers, modulation may be used to synthesise waveforms with a desired overtone spectrum. In this case the carrier frequency is typically in the same order or much lower than the modulating waveform. See for example frequency modulation synthesis or ring modulation.

A device that performs modulation is known as a modulator and a device that performs the inverse operation of modulation is known as a demodulator (sometimes detector or demod). A device that can do both operations is a modem (short for "Modulator-Demodulator").

Aim

The aim of digital modulation is to transfer a digital bit stream over an analog bandpass channel, for example over the public switched telephone network (where a bandpass filter limits the frequency range to between 300 and 3400 Hz), or over a limited radio frequency band.

The aim of analog modulation is to transfer an analog baseband (or lowpass) signal, for example an audio signal or TV signal, over an analog bandpass channel, for example a limited radio frequency band or a cable TV network channel.

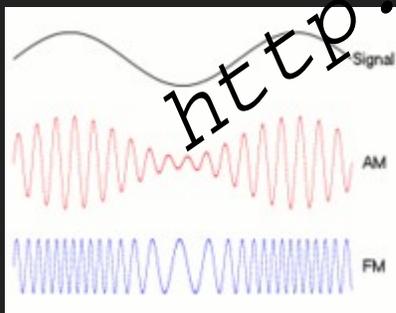
Analog and digital modulation facilitate frequency division multiplexing (FDM), where several low pass information signals are transferred simultaneously over the same shared physical medium, using separate passband channels.

The aim of digital baseband modulation methods, also known as line coding, is to transfer a digital bit stream over a baseband channel, typically a non-filtered copper wire such as a serial bus or a wired local area network.

The aim of pulse modulation methods is to transfer a narrowband analog signal, for example a phone call over a wideband baseband channel or, in some of the schemes, as a bit stream over another digital transmission system.

Analog modulation methods

In analog modulation, the modulation is applied continuously in response to the analog information signal.



A low-frequency message signal (top) may be carried by an AM or FM radio wave.

Common analog modulation techniques are:

Amplitude modulation (AM) (here the amplitude of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal)

- Double-sideband modulation (DSB)
 - Double-sideband modulation with carrier (DSB-WC) (used on the AM radio broadcasting band)
 - Double-sideband suppressed-carrier transmission (DSB-SC)

- Double-sideband reduced carrier transmission (DSB-RC)
- Single-sideband modulation (SSB, or SSB-AM),
 - SSB with carrier (SSB-WC)
 - SSB suppressed carrier modulation (SSB-SC)
- Vestigial sideband modulation (VSB, or VSB-AM)
- Quadrature amplitude modulation (QAM)

Angle modulation

- Frequency modulation (FM) (here the frequency of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal)
- Phase modulation (PM) (here the phase shift of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal)

The accompanying figure shows the results of (amplitude-)modulating a signal onto a carrier (both of which are sine waves). At any point along the y-axis, the amplitude of the modulated signal is equal to the sum of the carrier signal and the modulating signal amplitudes.

Simple example of amplitude modulation.

Digital modulation methods

In digital modulation, an analog carrier signal is modulated by a digital bit stream. Digital modulation methods can be considered as digital-to-analog conversion, and the corresponding demodulation or detection as analog-to-digital conversion. The changes in the carrier signal are chosen from a finite number of M alternative symbols (the modulation alphabet).

Schematic of 4 baud (8 bits) data link.

A simple example: A telephone line is designed for transferring audible sounds, for example tones, and not digital bits (zeros and ones). Computers may however communicate over a telephone line by means of modems, which are representing the digital bits by tones, called symbols. If there are four alternative symbols (corresponding to a musical instrument that can generate four different tones, one at a time), the first symbol may represent the bit sequence 00, the second 01, the third 10 and the fourth 11. If the modem plays a melody consisting of 1000 tones per second, the symbol rate is 1000 symbols/second, or baud. Since each tone (i.e., symbol) represents a message consisting of two digital bits in this example, the bit rate is twice the symbol rate, i.e. 2000 bits per second. This is similar to the technique used by dialup modems as opposed to DSL modems.

According to one definition of digital signal, the modulated signal is a digital signal, and according to another definition, the modulation is a form of digital-to-analog conversion. Most textbooks would consider digital modulation schemes as a form of digital transmission, synonymous to data transmission; very few would consider it as analog transmission.

Fundamental digital modulation methods

The most fundamental digital modulation techniques are based on keying:

In the case of PSK (phase-shift keying), a finite number of phases are used.

In the case of FSK (frequency-shift keying), a finite number of frequencies are used.

In the case of ASK (amplitude-shift keying), a finite number of amplitudes are used.

In the case of QAM (quadrature amplitude modulation), a finite number of at least two phases, and at least two amplitudes are used.

In QAM, an inphase signal (the I signal, for example a cosine waveform) and a quadrature phase signal (the Q signal, for example a sine wave) are amplitude modulated with a finite number of amplitudes, and summed. It can be seen as a two-channel system, each channel using ASK. The resulting signal is equivalent to a combination of PSK and ASK.

In all of the above methods, each of these phases, frequencies or amplitudes are assigned a unique pattern of binary bits. Usually, each phase, frequency or amplitude encodes an equal number of bits. This number of bits comprises the symbol that is represented by the particular phase, frequency or amplitude.

If the alphabet consists of $M = 2^N$ alternative symbols, each symbol represents a message consisting of N bits. If the symbol rate (also known as the baud rate) is f_s symbols/second (or baud), the data rate is Nf_s bit/second.

For example, with an alphabet consisting of 16 alternative symbols, each symbol represents 4 bits. Thus, the data rate is four times the baud rate.

In the case of PSK, ASK or QAM, where the carrier frequency of the modulated signal is constant, the modulation alphabet is often conveniently represented on a constellation diagram, showing the amplitude of the I signal at the x-axis, and the amplitude of the Q signal at the y-axis, for each symbol.

Modulator and detector principles of operation

PSK and ASK, and sometimes also FSK, are often generated and detected using the principle of QAM. The I and Q signals can be combined into a complex-valued signal

$I+jQ$ (where j is the imaginary unit). The resulting so called equivalent lowpass signal or equivalent baseband signal is a complex-valued representation of the real-valued modulated physical signal (the so called passband signal or RF signal).

These are the general steps used by the modulator to transmit data:

1. Group the incoming data bits into codewords, one for each symbol that will be transmitted.
2. Map the codewords to attributes, for example amplitudes of the I and Q signals (the equivalent low pass signal), or frequency or phase values.
3. Adapt pulse shaping or some other filtering to limit the bandwidth and form the spectrum of the equivalent low pass signal, typically using digital signal processing.
4. Perform digital-to-analog conversion (DAC) of the I and Q signals (since today all of the above is normally achieved using digital signal processing, DSP).
5. Generate a high-frequency sine wave carrier waveform, and perhaps also a cosine quadrature component. Carry out the modulation, for example by multiplying the sine and cosine wave form with the I and Q signals, resulting in that the equivalent low pass signal is frequency shifted into a modulated passband signal or RF signal. Sometimes this is achieved using DSP technology, for example direct digital synthesis using a waveform table, instead of analog signal processing. In that case the above DAC step should be done after this step.
6. Amplification and analog bandpass filtering to avoid harmonic distortion and periodic spectrum

At the receiver side, the demodulator typically performs:

1. Bandpass filtering.
2. Automatic gain control, AGC (to compensate for attenuation, for example fading).
3. Frequency shifting of the RF signal to the equivalent baseband I and Q signals, or to an intermediate frequency (IF) signal, by multiplying the RF signal with a local oscillator sinewave and cosine wave frequency (see the superheterodyne receiver principle).
4. Sampling and analog-to-digital conversion (ADC) (Sometimes before or instead of the above point, for example by means of undersampling).
5. Equalization filtering, for example a matched filter, compensation for multipath propagation, time spreading, phase distortion and frequency selective fading, to avoid intersymbol interference and symbol distortion.
6. Detection of the amplitudes of the I and Q signals, or the frequency or phase of the IF signal.
7. Quantization of the amplitudes, frequencies or phases to the nearest allowed symbol values.
8. Mapping of the quantized amplitudes, frequencies or phases to codewords (bit groups).
9. Parallel-to-serial conversion of the codewords into a bit stream.

10. Pass the resultant bit stream on for further processing such as removal of any error-correcting codes.

As is common to all digital communication systems, the design of both the modulator and demodulator must be done simultaneously. Digital modulation schemes are possible because the transmitter-receiver pair have prior knowledge of how data is encoded and represented in the communications system. In all digital communication systems, both the modulator at the transmitter and the demodulator at the receiver are structured so that they perform inverse operations.

Non-coherent modulation methods do not require a receiver reference clock signal that is phase synchronized with the sender carrier wave. In this case, modulation symbols (rather than bits, characters, or data packets) are asynchronously transferred. The opposite is coherent modulation.

List of common digital modulation techniques

The most common digital modulation techniques are:

Phase-shift keying (PSK):

- Binary PSK (BPSK), using $M=2$ symbols
- Quadrature PSK (QPSK), using $M=4$ symbols
- 8PSK, using $M=8$ symbols
- 16PSK, using $M=16$ symbols
- Differential PSK (DPSK)
- Differential QPSK (DQPSK)
- Offset QPSK (OQPSK)
- $\pi/4$ -QPSK

Frequency-shift keying (FSK):

- Audio frequency-shift keying (AFSK)
- Multi-frequency shift keying (M-ary FSK or MFSK)
- Dual-tone multi-frequency (DTMF)
- Continuous-phase frequency-shift keying (CPFSK)

Amplitude-shift keying (ASK)

On-off keying (OOK), the most common ASK form

- M-ary vestigial sideband modulation, for example 8VSB

Quadrature amplitude modulation (QAM) - a combination of PSK and ASK:

- Polar modulation like QAM a combination of PSK and ASK.[citation needed]

Continuous phase modulation (CPM) methods:

- Minimum-shift keying (MSK)
- Gaussian minimum-shift keying (GMSK)

Orthogonal frequency-division multiplexing (OFDM) modulation:

- discrete multitone (DMT) - including adaptive modulation and bit-loading.

Wavelet modulation

Trellis coded modulation (TCM), also known as trellis modulation

Spread-spectrum techniques:

- Direct-sequence spread spectrum (DSSS)
- Chirp spread spectrum (CSS) according to IEEE 802.15.4a CSS uses pseudo-stochastic coding
- Frequency-hopping spread spectrum (FHSS) applies a special scheme for channel release

MSK and GMSK are particular cases of continuous phase modulation. Indeed, MSK is a particular case of the sub-family of CPM known as continuous-phase frequency-shift keying (CPFSK) which is defined by a rectangular frequency pulse (i.e. a linearly increasing phase pulse) of one symbol-time duration (total response signaling).

OFDM is based on the idea of frequency-division multiplexing (FDM), but is utilized as a digital modulation scheme. The bit stream is split into several parallel data streams, each transferred over its own sub-carrier using some conventional digital modulation scheme. The modulated sub-carriers are summed to form an OFDM signal. OFDM is considered as a modulation technique rather than a multiplex technique, since it transfers one bit stream over one communication channel using one sequence of so-called OFDM symbols. OFDM can be extended to multi-user channel access method in the orthogonal frequency-division multiple access (OFDMA) and multi-carrier code division multiple access (MC-CDMA) schemes, allowing several users to share the same physical medium by giving different sub-carriers or spreading codes to different users.

Of the two kinds of RF power amplifier, switching amplifiers (Class C amplifiers) cost less and use less battery power than linear amplifiers of the same output power. However, they only work with relatively constant-amplitude-modulation signals such as angle modulation (FSK or PSK) and CDMA, but not with QAM and OFDM. Nevertheless, even though switching amplifiers are completely unsuitable for normal QAM constellations, often the QAM modulation principle are used to drive switching amplifiers with these FM and other waveforms, and sometimes QAM demodulators are used to receive the signals put out by these switching amplifiers.

Digital baseband modulation or line coding

The term digital baseband modulation (or digital baseband transmission) is synonymous to line codes. These are methods to transfer a digital bit stream over an analog baseband channel (a.k.a. lowpass channel) using a pulse train, i.e. a discrete number of signal levels, by directly modulating the voltage or current on a cable. Common examples are unipolar, non-return-to-zero (NRZ), Manchester and alternate mark inversion (AMI) codings.

Pulse modulation methods

Pulse modulation schemes aim at transferring a narrowband analog signal over an analog baseband channel as a two-level signal by modulating a pulse wave. Some pulse

modulation schemes also allow the narrowband analog signal to be transferred as a digital signal (i.e. as a quantized discrete-time signal) with a fixed bit rate, which can be transferred over an underlying digital transmission system, for example some line code. These are not modulation schemes in the conventional sense since they are not channel coding schemes, but should be considered as source coding schemes, and in some cases analog-to-digital conversion techniques.

Analog-over-analog methods:

Pulse-amplitude modulation (PAM)

Pulse-width modulation (PWM)

Pulse-position modulation (PPM)

Analog-over-digital methods:

Pulse-code modulation (PCM)

○ Differential PCM (DPCM)

○ Adaptive DPCM (ADPCM)

Delta modulation (DM or Δ -modulation)

Sigma-delta modulation ($\Sigma\Delta$)

Continuously variable slope delta modulation (CVSDM), also called Adaptive-delta modulation (ADM)

Pulse-density modulation (PDM)

Differential Pulse Code Modulation (DPCM)

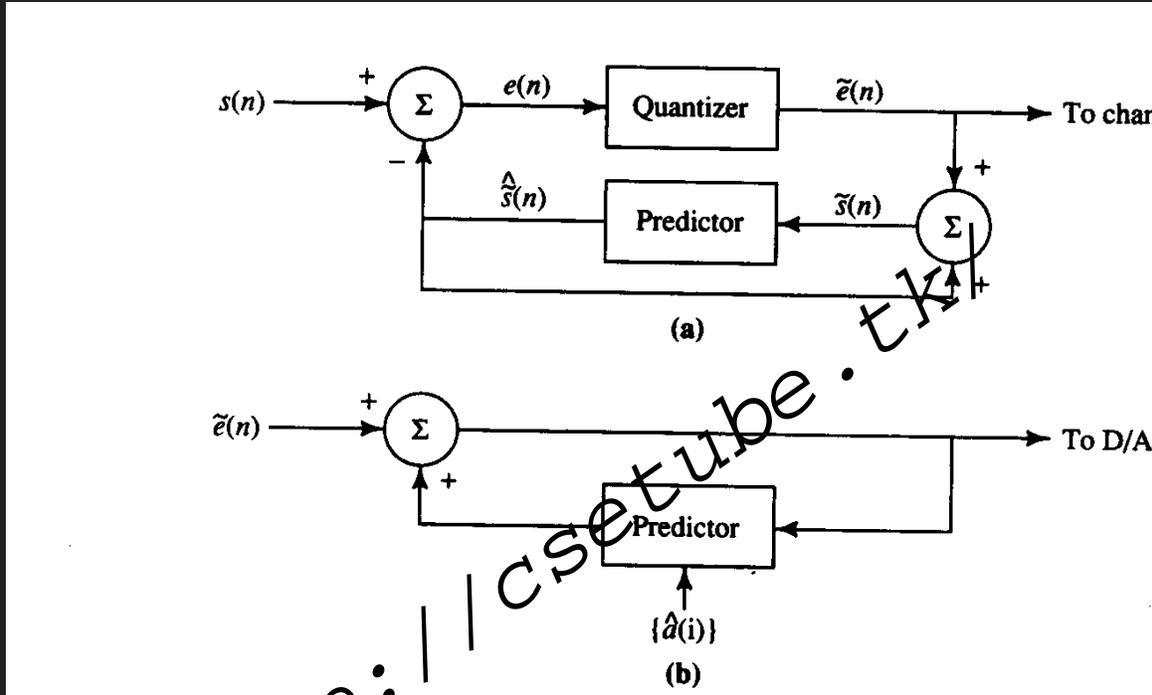
In PCM, each sample of the waveform is encoded independently of all the other samples. However, most source signals including speech sampled at the Nyquist rate or faster exhibit significant correlation between successive samples. In other words, the average change in amplitude between successive samples is relatively small. Consequently an encoding scheme that exploits the redundancy in the samples will result in a lower bit rate for the source output.

A relatively simple solution is to encode the differences between successive samples rather than the samples themselves. The resulting technique is called differential pulse code modulation (DPCM). Since differences between samples are expected to be smaller than the actual sampled amplitudes, fewer bits are required to represent the differences. In this case we quantize and transmit the differenced signal sequence

$$e(n) = s(n) - \hat{s}(n-1),$$

where $s(n)$ is the sampled sequence of $s(t)$.

A natural refinement of this general approach is to predict the current sample based on the previous M samples utilizing linear prediction (LP), where LP parameters are dynamically estimated. Block diagram of a DPCM encoder and decoder is shown below. Part (a) shows DPCM encoder and part (b) shows DPCM decoder at the receiver.



The "dpcm_demo" shows the use of DPCM to approximate a input sine wave signal and a speech signal that were sampled at 2 KHz and 44 KHz, respectively. The source code file of the MATLAB code and the output can be viewed using MATLAB.

Intersymbol interference

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. ISI is usually caused by multipath propagation or the inherent non-linear frequency response of a 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. Ways to fight intersymbol interference include adaptive equalization and error correcting codes.

Causes

Multipath propagation

One of the causes of intersymbol interference is what is known as multipath propagation in which a wireless signal from a transmitter reaches the receiver via many different 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 all of these paths are different lengths - plus some of these effects will also slow the signal down - this results in the different versions of the signal arriving at different times. This delay means 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, the amplitude of the frequency components 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.

As opposed to multipath propagation, 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 MHz - 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 , $1/T$ 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).

Sampling (signal processing)

In signal processing, sampling is the reduction of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous-time signal) to a sequence of samples (a discrete-time signal).

A sample refers to a value or set of values at a point in time and/or space.

A sampler is a subsystem or operator that extracts samples from continuous signal. A theoretical ideal sampler multiplies a continuous signal with a Dirac comb. This multiplication "picks out" values but the result is still continuous-valued. If this signal is then discretized (i.e., converted into a sequence) and quantized along all dimensions it becomes a discrete signal.

Reconstruction Of message from samples:

Any real signal will be transmitted along some form of channel which will have a finite bandwidth. As a result the received signal's spectrum cannot contain any frequencies above some maximum value, f_m . However, the spectrum obtained using the Fourier method described in the previous section will be characteristic of a signal which repeats after the interval, T . This means it can be described by a spectrum which only contain the frequencies, 0 (d.c.), $f_0, 2f_0, 3f_0, \dots, Nf_0$, where N is the largest integer which satisfies the inequality $Nf_0 \leq f_m$. As a consequence we can specify everything we know about the signal spectrum in terms of a d.c. level plus the amplitudes and phases of just N frequencies — i.e. all the information we have about the spectrum can be specified by just $2N + 1$ numbers. Given that no information was lost when we calculated the spectrum it immediately follows that everything we know about the shape of the time domain signal pattern could also be specified by just $2N + 1$ values.

For a signal whose duration is T this means that we can represent all of the signal information by measuring the signal level at $2N + 1$ points equally spaced along the signal waveform. If we put the first point at the start of the message and the final one at its end

this means that each sampled point will be at a distance $\frac{T}{2N+1}$ from its neighbours. This result is generally expressed in terms of the Sampling Theorem which can be stated as:

'If a continuous function contains no frequencies higher than f_m Hz it is completely determined by its value at a series of points less than $\frac{1}{2f_m}$ apart.'

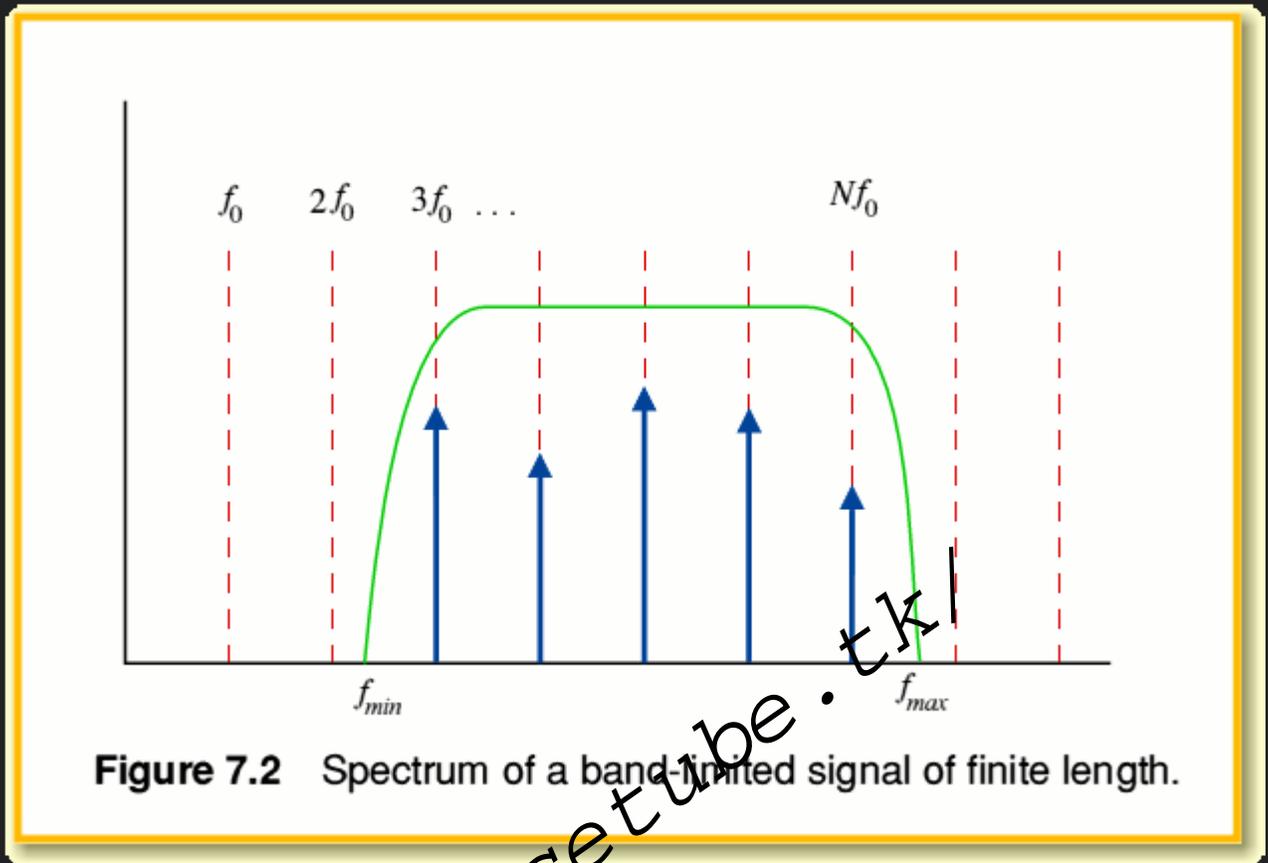
Consider a signal, $A(t)$, which is observed over the time interval, $0 \leq t \leq T$, and which we know cannot contain any frequencies above f_m . We can sample this signal

to obtain a series of values, x_i , which represent the signal level at the instants, $t_i = \frac{iT}{N}$, where i is an integer in the range 0 to K . (This means there are $K + 1$ samples.) Provided that $\frac{K}{N} \gg 2B$, where N is defined as above, we have satisfied the requirements of the Sampling Theorem. The samples will then contain all of the information present in the original signal and make up what is called a Complete Record of the original.

In fact, the above statement is a fairly 'weak' form of the sampling theorem. We can go on to a stricter form:

If a continuous function only contains frequencies within a bandwidth, B Hertz, it is completely determined by its value at a series of points spaced less than $\frac{1}{(2B)}$ seconds apart.

This form of the sampling theorem can be seen to be true by considering a signal which doesn't contain any frequencies below some lower cut-off value, f_L . This means the values of A_n and B_n for low n (i.e. low values of f_n) will all be zero. This limits the number of spectral components present in the signal just as the upper limit, f_H , means that there are no components above f_H . This situation is illustrated in figure 7.2.



From the above argument a signal of finite length, T , can be described by a spectrum which only contains frequencies $f_0, 2f_0, \dots, Nf_0$. If the signal is restricted to a given bandwidth, $B = f_{max} - f_{min}$, only those components inside the band have non-zero values. Hence we only need to specify the A_n and B_n values for those components to completely define the signal. The minimum required sampling rate therefore depends upon the bandwidth, not the maximum frequency. (Although in cases where the signal has components down to d.c. the two are essentially the same.)

The sampling theorem is of vital importance when processing information as it means that we can take a series of samples of a continuously varying signal and use those values to represent the entire signal without any loss of the available information. These samples can later be used to reconstruct all of the details of the original signal — even recovering details of the actual signal pattern in between the sampled moments. To demonstrate this we can show how the original waveform can be reconstructed from a complete set of samples.

The approach used in the previous section to calculate a signal's spectrum depends upon being able to integrate a continuous analytical function. Now, however, we need to deal with a set of sampled values instead of a continuous function. The integrals must be replaced by equivalent summations. These expressions allow us to calculate a frequency

spectrum (i.e. the appropriate set of A_n and B_n values) from the samples which contain all of the signal information. The most obvious technique is to proceed in two steps. Firstly, to take the sample values, x_i , and calculate the signal's spectrum. Given a series of samples we must use the series expressions

$$A_n = \frac{2}{K} \sum_{i=0}^{K-1} x_i \cos\{2\pi n f_s t_i\} \quad B_n = \frac{2}{K} \sum_{i=0}^{K-1} x_i \sin\{2\pi n f_s t_i\}$$

to calculate the relevant spectrum values. These are essentially the equivalent of the integrals, 7.10 and 7.11, which we would use to compute the spectrum of a continuous function. The second step of this approach is to use the resulting A_n and B_n values in the expression

$$x\{t\} = \sum_{n=0}^N A_n \cos\{2\pi n f_s t\} + B_n \sin\{2\pi n f_s t\}$$

to compute the signal level at any time, t , during the observed period. In effect, this second step is simply a restatement of the result shown in expression 7.9. Although this method works, it is computationally intensive and indirect. This is because it requires us to perform a whole series of numerical summations to determine the spectrum, followed

by another summation for each $x\{t\}$ we wish to determine. A more straightforward method can be employed, based upon combining these operations. Expressions 7.12 and 7.13 can be combined to produce

$$x\{t\} = \sum_{n=0}^N \frac{2}{K} \sum_{i=0}^{K-1} x_i [\cos\{2\pi n f_s t_i\} \cos\{2\pi n f_s t\} + \sin\{2\pi n f_s t_i\} \sin\{2\pi n f_s t\}]$$

which, by a fairly involved process of algebraic manipulation, may be simplified into the form

$$x\{t\} = \sum_{i=0}^{K-1} x_i \operatorname{Sinc}\left\{\frac{\pi(t - t_i)}{\Delta t}\right\}$$

where the Sinc function can be defined as

$$\operatorname{Sinc}\{z\} = \frac{\sin\{z\}}{z}$$

and $\Delta t = T/K$ is the time interval between successive samples.

Given a set of samples, x_{kT} , taken at the instants, kT , we can now use expression 7.15 to calculate what the signal level would have been at any time, t , during the sampled signal interval.

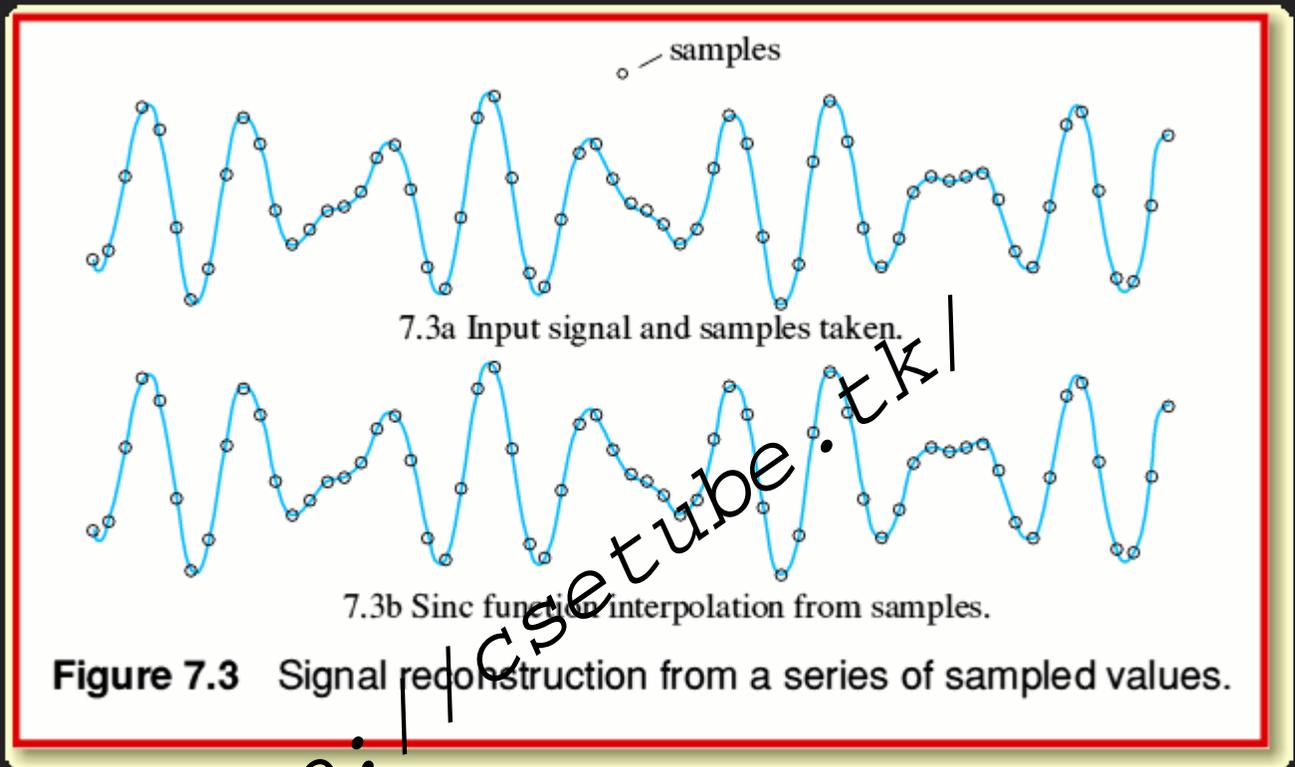


Figure 7.3 Signal reconstruction from a series of sampled values.

Clearly, by using this approach we can calculate the signal value at any instant by performing a single summation over the sampled values. This method is therefore rather easier (and less prone to computational errors!) than the obvious technique. Figure 7.2 was produced by a BBC Basic program to demonstrate how easily this method can be used.

Although the explanation given here for the derivation of expression 7.15 is based upon the use of a Fourier technique, the result is a completely general one. Expression 7.15 can be used to interpolate any given set of sampled values. The only requirement is that the samples have been obtained in accordance with the Sampling Theorem and that they do, indeed, form a complete record. It is important to realise that, under these circumstances, the recovered waveform is not a guess but a reliable reconstruction of what we would have observed if the original signal had been measured at these other moments.

Summary

You should now be aware that the information carried by a signal can be defined either in terms of its Time Domain pattern or its Frequency Domain spectrum. You should also know that the amount of information in a continuous analog signal can be specified by a finite number of values. This result is summarised by the Sampling Theorem which states that we can collect all the information in a signal by sampling at a rate $\geq 2B$, where B is the signal bandwidth. Given this information we can, therefore, reconstruct the actual shape of the original continuous signal at any instant 'in between' the sampled instants. It should also be clear that this reconstruction is not a guess but a true reconstruction.

Discrete PAM signals:

Pulse-amplitude modulation

From Wikipedia, the free encyclopedia

(Redirected from Pulse amplitude modulation)

Jump to: navigation, search

Principle of PAM; (1) original Signal, (2) PAM-Signal, (a) Amplitude of Signal, (b) Time

Pulse-amplitude modulation, acronym PAM, is a form of signal modulation where the message information is encoded in the amplitude of a series of signal pulses.

Example: A two bit modulator (PAM-4) will take two bits at a time and will map the signal amplitude to one of four possible levels, for example -3 volts, -1 volt, 1 volt, and 3 volts.

Demodulation is performed by detecting the amplitude level of the carrier at every symbol period.

Pulse-amplitude modulation is widely used in baseband transmission of digital data, with non-baseband applications having been largely superseded by pulse-code modulation, and, more recently, by pulse-position modulation.

In particular, all telephone modems faster than 300 bit/s use quadrature amplitude modulation (QAM). (QAM uses a two-dimensional constellation).

It should be noted, however, that some versions of the widely popular Ethernet communication standard are a good example of PAM usage. In particular, the Fast Ethernet 100BASE-T2 medium, running at 100Mb/s, utilizes 5 level PAM modulation (PAM-5) running at 25 megapulses/sec over two wire pairs. A special technique is used to reduce inter-symbol interference between the unshielded pairs. Later, the gigabit Ethernet 1000BASE-T medium raised the bar to use 4 pairs of wire running each at 125 megapulses/sec to achieve 1000Mb/s data rates, still utilizing PAM-5 for each pair.

The IEEE 802.3an standard defines the wire-level modulation for 10GBASE-T as a Tomlinson-Harashima Precoded (THP) version of pulse-amplitude modulation with 16 discrete levels (PAM-16), encoded in a two-dimensional checkerboard pattern known as DSQ128. Several proposals were considered for wire-level modulation, including PAM with 12 discrete levels (PAM-12), 10 levels (PAM-10), or 8 levels (PAM-8), both with and without Tomlinson-Harashima Precoding (THP).

To achieve full-duplex operation, parties must ensure that their transmitted pulses do not coincide in time. This makes use of bus topology (featured by older Ethernet implementations) practically impossible with these modern Ethernet mediums. This technique is called Carrier Sense Multiple Access and is used in some home networking protocols such as HomePlug.

Eye pattern

eye diagram of a 4 level signal

In telecommunication, an eye pattern, also known as an eye diagram is an oscilloscope display in which a digital data signal from a receiver is repetitively sampled and applied to the vertical input, while the data rate is used to trigger the horizontal sweep. It is so called because, for several types of coding, the pattern looks like a series of eyes between a pair of rails.

Several system performance measures can be derived by analyzing the display. If the signals are too long, too short, poorly synchronized with the system clock, too high, too low, too noisy, too slow to change, or have too much undershoot or overshoot, this can be observed from the eye diagram. An open eye pattern corresponds to minimal signal distortion. Distortion of the signal waveform due to intersymbol interference and noise appears as closure of the eye pattern.

Important Questions:UNIT III

PART A

- 1.Determine the 12 bit linear code,8 bit compressed code and recovered 12 bit code for resolution of 0.01V and analog sample voltage of 0.05V.
- 2.What are the types Pulse Modulation.
- 3.Define Sampling rate.
4. Define Nyquist rate
- 5.What is alising?
6. what is Companding

7. Define SNR.
8. what is Redudancy.
- 9.What is the use of Eyepattern.
- 10.Define Modem.
- 11.Give data communication codes.
- 12.Draw the block diagram of DPCM.
13. what are the errors occur in Delta Modulation.

PART-B

- 1 .Explain the working of Delta modulation
- 2 Explain the working of PCM transmitter and receiver
3. Explain the working of Adaptive Delta modulation
- 4.What is ISI. How can it be determine.
5. Explain the operation of RS-232 serial interface with timing diagram.
- 6 .Explain low and high speed modems.
7. Explain delta modulation and DPCM
- 8.Explain about data communication circuit.
- 9.Explain about data communication codes.
- 10.Explain about MODEM Control .

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UNIT IV DATA COMMUNICATIONS 9

Basics – History of Data Communications – Standards Organizations for Data

Communication – Data Communication Circuits – Data Communication Codes –

Error Control – Error Detection – Error Correction – Data Communication Hardware

– Serial and Parallel Interfaces – Data Modems – Asynchronous Modem –

Synchronous Modem – Low-Speed Modem – Medium and High Speed Modem –

Modem Control.

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UNIT – IV

DATA COMMUNICATIONS

Error detection and correction

In information theory and coding theory with applications in computer science and telecommunication, error detection and correction or error control are techniques that enable reliable delivery of digital data over unreliable communication channels. Many communication channels are subject to channel noise, and thus errors may be introduced during transmission from the source to a receiver. Error detection techniques allow detecting such errors, while error correction enables reconstruction of the original data.

The general definitions of the terms are as follows:

Error detection is the detection of errors caused by noise or other impairments during transmission from the transmitter to the receiver.

Error correction is the detection of errors and reconstruction of the original, error-free data.

Error correction may generally be realized in two different ways:

Automatic repeat request (ARQ) (sometimes also referred to as backward error correction): This is an error control technique whereby an error detection scheme is combined with requests for retransmission of erroneous data. Every block of data received is checked using the error detection code used, and if the check fails, retransmission of the data is requested – this may be done repeatedly, until the data can be verified.

Forward error correction (FEC): The sender encodes the data using an error-correcting code (ECC) prior to transmission. The additional information (redundancy) added by the code is used by the receiver to recover the original data. In general, the reconstructed data is what is deemed the "most likely" original data.

ARQ and FEC may be combined, such that minor errors are corrected without retransmission, and major errors are corrected via a request for retransmission: this is called hybrid automatic repeat-request (HARQ).

Introduction

The general idea for achieving error detection and correction is to add some redundancy (i.e., some extra data) to a message, which receivers can use to check consistency of the delivered message, and to recover data determined to be erroneous. Error-detection and correction schemes can be either systematic or non-systematic: In a systematic scheme, the transmitter sends the original data, and attaches a fixed number of check bits (or

parity data), which are derived from the data bits by some deterministic algorithm. If only error detection is required, a receiver can simply apply the same algorithm to the received data bits and compare its output with the received check bits; if the values do not match, an error has occurred at some point during the transmission. In a system that uses a non-systematic code, the original message is transformed into an encoded message that has at least as many bits as the original message.

Good error control performance requires the scheme to be selected based on the characteristics of the communication channel. Common channel models include memory-less models where errors occur randomly and with a certain probability, and dynamic models where errors occur primarily in bursts. Consequently, error-detecting and correcting codes can be generally distinguished between random-error-detecting/correcting and burst-error-detecting/correcting. Some codes can also be suitable for a mixture of random errors and burst errors.

If the channel capacity cannot be determined, or is highly varying, an error-detection scheme may be combined with a system for retransmissions of erroneous data. This is known as automatic repeat request (ARQ), and is most notably used in the Internet. An alternate approach for error control is hybrid automatic repeat request (HARQ), which is a combination of ARQ and error-correction coding.

Error detection schemes

Error detection is most commonly realized using a suitable hash function (or checksum algorithm). A hash function adds a fixed-length tag to a message, which enables receivers to verify the delivered message by recomputing the tag and comparing it with the one provided.

There exists a vast variety of different hash function designs. However, some are of particularly widespread use because of either their simplicity or their suitability for detecting certain kinds of errors (e.g., the cyclic redundancy check's performance in detecting burst errors).

Random-error-correcting codes based on minimum distance coding can provide a suitable alternative to hash functions when a strict guarantee on the minimum number of errors to be detected is desired. Repetition codes, described below, are special cases of error-correcting codes: although rather inefficient, they find applications for both error correction and detection due to their simplicity.

Repetition codes

A repetition code is a coding scheme that repeats the bits across a channel to achieve error-free communication. Given a stream of data to be transmitted, the data is divided into blocks of bits. Each block is transmitted some predetermined number of times. For example, to send the bit pattern "1011", the four-bit block can be repeated three times, thus producing "1011 1011 1011". However, if this twelve-bit pattern was received as

"1010 1011 1011" – where the first block is unlike the other two – it can be determined that an error has occurred.

Repetition codes are not very efficient, and can be susceptible to problems if the error occurs in exactly the same place for each group (e.g., "1010 1010 1010" in the previous example would be detected as correct). The advantage of repetition codes is that they are extremely simple, and are in fact used in some transmissions of numbers stations.[citation needed]

Parity bits

A parity bit is a bit that is added to a group of source bits to ensure that the number of set bits (i.e., bits with value 1) in the outcome is even or odd. It is a very simple scheme that can be used to detect single or any other odd number (i.e., three, five, etc.) of errors in the output. An even number of flipped bits will make the parity bit appear correct even though the data is erroneous.

Extensions and variations on the parity bit mechanism are horizontal redundancy checks, vertical redundancy checks, and "double," "dual," or "diagonal" parity (used in RAID-DP).

Checksums

A checksum of a message is a modular arithmetic sum of message code words of a fixed word length (e.g., byte values). The sum may be negated by means of a one's-complement prior to transmission to detect errors resulting in all-zero messages.

Checksum schemes include parity bits, check digits, and longitudinal redundancy checks. Some checksum schemes, such as the Luhn algorithm and the Verhoeff algorithm, are specifically designed to detect errors commonly introduced by humans in writing down or remembering identification numbers.

Cyclic redundancy checks (CRCs)

A cyclic redundancy check (CRC) is a single-burst-error-detecting cyclic code and non-secure hash function designed to detect accidental changes to digital data in computer networks. It is characterized by specification of a so-called generator polynomial, which is used as the divisor in a polynomial long division over a finite field, taking the input data as the dividend, and where the remainder becomes the result.

Cyclic codes have favorable properties in that they are well suited for detecting burst errors. CRCs are particularly easy to implement in hardware, and are therefore commonly used in digital networks and storage devices such as hard disk drives.

Even parity is a special case of a cyclic redundancy check, where the single-bit CRC is generated by the divisor $x+1$.

Cryptographic hash functions

A cryptographic hash function can provide strong assurances about data integrity, provided that changes of the data are only accidental (i.e., due to transmission errors). Any modification to the data will likely be detected through a mismatching hash value. Furthermore, given some hash value, it is infeasible to find some input data (other than the one given) that will yield the same hash value. Message authentication codes, also called keyed cryptographic hash functions, provide additional protection against intentional modification by an attacker.

Error-correcting codes

Any error-correcting code can be used for error detection. A code with minimum Hamming distance, d , can detect up to $d-1$ errors in a code word. Using minimum-distance-based error-correcting codes for error detection can be suitable if a strict limit on the minimum number of errors to be detected is desired.

Codes with minimum Hamming distance $d=2$ are degenerate cases of error-correcting codes, and can be used to detect single errors. The parity bit is an example of a single-error-detecting code.

The Berger code is an early example of a unidirectional error(-correcting) code that can detect any number of errors on an asymmetric channel, provided that only transitions of cleared bits to set bits or set bits to cleared bits can occur.

Error correction

Automatic repeat request

Automatic Repeat reQuest (ARQ) is an error control method for data transmission that makes use of error-detection codes, acknowledgment and/or negative acknowledgment messages, and timeouts to achieve reliable data transmission. An acknowledgment is a message sent by the receiver to indicate that it has correctly received a data frame.

Usually, when the transmitter does not receive the acknowledgment before the timeout occurs (i.e., within a reasonable amount of time after sending the data frame), it

retransmits the frame until it is either correctly received or the error persists beyond a predetermined number of retransmissions.

Three types of ARQ protocols are Stop-and-wait ARQ, Go-Back-N ARQ, and Selective Repeat ARQ.

ARQ is appropriate if the communication channel has varying or unknown capacity, such as is the case on the Internet. However, ARQ requires the availability of a back channel, results in possibly increased latency due to retransmissions, and requires the maintenance of buffers and timers for retransmissions, which in the case of network congestion can put a strain on the server and overall network capacity.

Error-correcting code

An error-correcting code (ECC) or forward error correction (FEC) code is a system of adding redundant data, or parity data, to a message, such that it can be recovered by a receiver even when a number of errors (up to the capability of the code being used) were introduced, either during the process of transmission, or on storage. Since the receiver does not have to ask the sender for retransmission of the data, a back channel is not required in forward error correction, and it is therefore suitable for simplex communication such as broadcasting. Error-correcting codes are frequently used in lower-layer communication, as well as for reliable storage in media such as CDs, DVDs, hard disks, and RAM.

Error-correcting codes are usually distinguished between convolutional codes and block codes:

Convolutional codes are processed on a bit-by-bit basis. They are particularly suitable for implementation in hardware, and the Viterbi decoder allows optimal decoding.

Block codes are processed on a block-by-block basis. Early examples of block codes are repetition codes, Hamming codes and multidimensional parity-check codes. They were followed by a number of efficient codes, Reed-Solomon codes being the most notable due to their current widespread use. Turbo codes and low-density parity-check codes (LDPC) are relatively new constructions that can provide almost optimal efficiency.

Shannon's theorem is an important theorem in forward error correction, and describes the maximum information rate at which reliable communication is possible over a channel that has a certain error probability or signal-to-noise ratio (SNR). This strict upper limit is expressed in terms of the channel capacity. More specifically, the theorem says that there exist codes such that with increasing encoding length the probability of error on a discrete memoryless channel can be made arbitrarily small, provided that the code rate is smaller than the channel capacity. The code rate is defined as the fraction k/n of k source symbols and n encoded symbols.

The actual maximum code rate allowed depends on the error-correcting code used, and may be lower. This is because Shannon's proof was only of existential nature, and did not show how to construct codes which are both optimal and have efficient encoding and decoding algorithms.

Hybrid schemes

Hybrid ARQ is a combination of ARQ and forward error correction. There are two basic approaches:

Messages are always transmitted with FEC parity data (and error-detection redundancy). A receiver decodes a message using the parity information, and requests retransmission using ARQ only if the parity data was not sufficient for successful decoding (identified through a failed integrity check).

Messages are transmitted without parity data (only with error-detection information). If a receiver detects an error, it requests FEC information from the transmitter using ARQ, and uses it to reconstruct the original message.

The latter approach is particularly attractive on an erasure channel when using a rateless erasure code.

Applications

Applications that require low latency (such as telephone conversations) cannot use Automatic Repeat reQuest (ARQ); they must use Forward Error Correction (FEC). By the time an ARQ system discovers an error and re-transmits it, the re-sent data will arrive too late to be any good.

Applications where the transmitter immediately forgets the information as soon as it is sent (such as most television cameras) cannot use ARQ; they must use FEC because when an error occurs, the original data is no longer available. (This is also why FEC is used in data storage systems such as RAID and distributed data store).

Applications that use ARQ must have a return channel. Applications that have no return channel cannot use ARQ.

Applications that require extremely low error rates (such as digital money transfers) must use ARQ.

The Internet

In a typical TCP/IP stack, error control is performed at multiple levels:

Each Ethernet frame carries a CRC-32 checksum. Frames received with incorrect checksums are discarded by the receiver hardware.

The IPv4 header contains a checksum protecting the contents of the header. Packets with mismatching checksums are dropped within the network or at the receiver.

The checksum was omitted from the IPv6 header in order to minimize processing costs in network routing and because current link layer technology is assumed to provide sufficient error detection (see also RFC 3819).

UDP has an optional checksum covering the payload and addressing information from the UDP and IP headers. Packets with incorrect checksums are discarded by the operating system network stack. The checksum is optional under IPv4, only, because the IP layer checksum may already provide the desired level of error protection.

TCP provides a checksum for protecting the payload and addressing information from the TCP and IP headers. Packets with incorrect checksums are discarded within the network stack, and eventually get retransmitted using ARQ, either explicitly (such as through triple-ack) or implicitly due to a timeout.

Deep-space telecommunications

Development of error-correction codes was tightly coupled with the history of deep-space missions due to the extreme dilution of signal power over interplanetary distances, and the limited power availability aboard space probes. Whereas early missions sent their data uncoded, starting from 1968 digital error correction was implemented in the form of (sub-optimally decoded) convolutional codes and Reed-Muller codes.[3] The Reed-Muller code was well suited to the noise the spacecraft was subject to (approximately matching a bell curve), and was implemented at the Mariner spacecraft for missions between 1969 and 1977.

The Voyager 1 and Voyager 2 missions, which started in 1977, were designed to deliver color imaging amongst scientific information of Jupiter and Saturn.[4] This resulted in increased coding requirements, and thus the spacecrafts were supported by (optimally Viterbi-decoded) convolutional codes that could be concatenated with an outer Golay (24,12,8) code. The Voyager 2 probe additionally supported an implementation of a Reed-Solomon code: the concatenated Reed-Solomon-Viterbi (RSV) code allowed for very powerful error correction, and enabled the spacecraft's extended journey to Uranus and Neptune.

The CCSDS currently recommends usage of error correction codes with performance similar to the Voyager 2 RSV code as a minimum. Concatenated codes are increasingly falling out of favor with space missions, and are replaced by more powerful codes such as Turbo codes or LDPC codes.

The different kinds of deep space and orbital missions that are conducted suggest that trying to find a "one size fits all" error correction system will be an ongoing problem for some time to come. For missions close to earth the nature of the channel noise is different

from that of a spacecraft on an interplanetary mission experiences. Additionally, as a spacecraft increases its distance from earth, the problem of correcting for noise gets larger.

Satellite broadcasting (DVB)

The demand for satellite transponder bandwidth continues to grow, fueled by the desire to deliver television (including new channels and High Definition TV) and IP data. Transponder availability and bandwidth constraints have limited this growth, because transponder capacity is determined by the selected modulation scheme and Forward error correction (FEC) rate.

Overview

QPSK coupled with traditional Reed Solomon and Viterbi codes have been used for nearly 20 years for the delivery of digital satellite TV.

Higher order modulation schemes such as 8PSK, 16QAM and 32QAM have enabled the satellite industry to increase transponder efficiency by several orders of magnitude.

This increase in the information rate in a transponder comes at the expense of an increase in the carrier power to meet the threshold requirement for existing antennas.

Tests conducted using the latest chipsets demonstrate that the performance achieved by using Turbo Codes may be even lower than the 0.8 dB figure assumed in early designs.

Data storage

Error detection and correction codes are often used to improve the reliability of data storage media.

A "parity track" was present on the first magnetic tape data storage in 1951. The "Optimal Rectangular Code" used in group code recording tapes not only detects but also corrects single-bit errors.

Some file formats, particularly archive formats, include a checksum (most often CRC32) to detect corruption and truncation and can employ redundancy and/or parity files to recover portions of corrupted data.

Reed Solomon codes are used in compact discs to correct errors caused by scratches.

Modern hard drives use CRC codes to detect and Reed-Solomon codes to correct minor errors in sector reads, and to recover data from sectors that have "gone bad" and store that data in the spare sectors.

RAID systems use a variety of error correction techniques, to correct errors when a hard drive completely fails.

Error-correcting memory

DRAM memory may provide increased protection against soft errors by relying on error correcting codes. Such error-correcting memory, known as ECC or EDAC-protected memory, is particularly desirable for high fault-tolerant applications, such as servers, as well as deep-space applications due to increased radiation.

Error-correcting memory controllers traditionally use Hamming codes, although some use triple modular redundancy.

Interleaving allows distributing the effect of a single cosmic ray potentially upsetting multiple physically neighboring bits across multiple words by associating neighboring bits to different words. As long as a single event upset (SEU) does not exceed the error threshold (e.g., a single error) in any particular word between accesses, it can be corrected (e.g., by a single-bit error correcting code), and the illusion of an error-free memory system may be maintained.

Command and Data modes (modem)

Command and Data modes refer to the two modes in which a computer modem may operate. These modes are defined in the Hayes command set, which is the de-facto standard for all modems. These modes exist because there is only one channel of communication between the modem and the computer, which must carry both the computer's commands to the modem, as well as the data that the modem is enlisted to transmit to the remote party over the telephone line.

When a modem is in command mode, any characters sent to it are interpreted as commands for the modem to execute, per the Hayes command set. A command is preceded by the letters 'AT', which stand for 'Attention'. For example, if a modem receives 'ATDT5551212' while in the command mode, it interprets that as an instruction to dial the numbers 5551212 on the telephone, using touch-tone dialing. While in command mode, the modem may send responses back to the computer indicating the outcome of the command. For example, the modem may respond with the word "BUSY" in response to the ATDT command, if it hears a busy signal after dialing and is configured to listen for busy signals.

Any communication in command mode (in both directions) is terminated by a carriage return.

When a modem is in data mode, any characters sent to the modem are intended to be transmitted to the remote party. The modem enters data mode immediately after it makes

a connection. For example, if ATDT5551212 resulted in a phone call that was answered by another computer modem, the modem would report the word "CONNECT" and then switch to data mode. Any further characters received over the serial link are deemed to be from the remote party, and any characters sent are transmitted to the remote party.

When a voice-capable modem is in "voice data" mode, any data sent to the modem is interpreted as audio data to be played over the phone line, rather than character bytes to be transmitted digitally to the other party.

Switching between modes

Modems always start out in command mode when powered up. Here are the ways a modem can switch to data mode:

After a successful dial-out connection in response to an "ATD" dial command in which the modem reaches another modem.

After answering the phone with the "ATA" answer command, if another modem is on the other end.

After answering the phone automatically for some pre-configured reason (such as auto-answer), and connecting to another modem. (Almost all modems support auto-answering when given the command "ATS0=1".)

After being given the "ATO" (that's three letters A-T-Oh, not A-T-zero) command after being put back in the command mode with an escape sequence (see below).

In response to similar dialing or connecting commands for fax or voice communications.

Modems switch back into command mode from data mode for the following reasons:

The connection got broken (for example, the other party hung up).

The computer issued an escape command, which is usually a 1-second pause, then the three characters "+++", then another 1-second pause. The connection remains, but the modem can accept commands, such as "ATH" for hangup. The computer can issue the "ATO" command to return to data mode.

The computer instructed the modem to terminate the call by setting the Data Terminal Ready (DTR) pin to an "off" state. (This is usually how computers invoke the termination of a modem call nowadays - the +++ escape sequence is rarely used, and usually disabled to avoid malfunction in case these characters are legitimately a part of the data stream).

Practical contemporary use

Today, most modems are configured with the characters "&C1&D2" in the initialization string, or otherwise behave this way by default. This causes the following behavior:

The modem uses the Data Carrier Detect (DCD) pin to signal whether it's connected to a host. The computer can generally just read the DCD pin at any time and always know if the modem is in command or data mode. DCD high means data mode, and low means command mode. DCD is a signal sent from the modem to the computer. (Note that DCD also remains high if in command mode after a +++ escape sequence, but in practice, software uses this so rarely, if at all, so it's not really an issue.)

The modem interprets the Data Terminal Ready (DTR) pin as a signal from the computer to know when it wants to terminate a call. DTR is a signal from the computer to the modem. The computer keeps DTR high at all times until it wants to terminate a call, at which time the computer lowers DTR for a second or two. The computer also keeps DTR low when no programs are running that want to use the modem - this keeps the modem from answering calls due to auto-answer or otherwise doing something unexpected or undesired. The computer may safely assume that after DTR has been brought low for a couple seconds, that the modem will be in command mode.

Data Link Escape (DLE) messages

For normal dial-up data communications, modems enter data mode only once - starting when the session connects, and ending when the session disconnects. However, when modems are used for fax and voice (audio) communication, they rapidly switch between command and data modes several times during a call. This is because the role of the modem changes more frequently - rather than simply handing bytes from point A to point B, it is either negotiating parameters and pages with a fax machine, or switching between recording, pausing, and playback audio modes.

In fax and voice data modes, sometimes events occur that the modem wishes to signal to the computer regardless of whether it's in command or data mode. An example of such an event is a caller pressing a touch-tone key in voice mode. Other common DLE messages are notifications from the modem that data was lost because the computer is sending data either too slow or too fast, or that the modem hears an unexpected dial-tone on the line (meaning the caller probably hung up), or that the extension handset was picked up or hung up.

This type of event is signalled with a DLE message, which is a single-character message preceded by a Data Link Escape code. DLE is a character sent from the modem to the computer whose definition is unrelated to either command or data mode, and which uses a specific ASCII code (0x10) which never occurs in any AT commands or responses, so that it can be isolated from the command/response stream on that merit alone. In these modes, whenever a DLE (0x10) is sent, the character following it has a special non-command and non-data meaning. For example, a DLE followed by the number 2 means the caller pressed the number 2 on his telephone keypad. This could happen in either command or data mode so long as the phone line is in use by the modem and someone is on the other end of the line.

Of course, the DLE character could legitimately appear in fax or audio data. For this, a special exception is made: two DLE's in a row means "interpret one literal byte 0x10".

DLE is also used lightly in communication from the computer to the modem. One specific DLE event signals the end of a fax page, or the end of an audio file. That event returns the modem back into command mode. Unlike in standard dial-up data mode, dropping DTR isn't an appropriate way to resume command mode since a hangup is not desired, and an escape code with mandatory pauses isn't suitable either. Because of this, literal 0x10 bytes in data are doubled from the computer to the modem as well.

DLE is never used in standard modem-to-modem data modes such as the one used for dial-up Internet access, at least not by the modem itself. In these modes, DLE is passed over the line just like any other character. DLE (and DLE-escaped messages) are only used by a modem for fax and voice applications, when it is specifically placed in a fax or voice mode.

Important Questions:UNIT IV & V

PART A

1. State Shannon's Fundamental theorem of information theory.
2. Give the applications of spread spectrum modulation
3. Define Processing gain. Give an expression for processing gain.
4. What is spread spectrum modulation. State its advantages
5. Differentiate between FDM and TDM
6. Mention the Processing gain of DS and FH Spread spectrum techniques.
7. What is Time division multiple access?
8. Draw the Signal Constellation diagram of 8-PSK Modulation.
9. State the correlation property of Maximal Length Sequence.
10. Define Pulse amplitude modulation.
11. What is fast and slow frequency hopping?
12. What are the three Properties of PN Sequences?
13. Define MSK.
14. Define LPC
15. Give the applications of wireless communications.
16. What is Near-far problem?
17. what is Frequency reuse?
18. What is Code division multiple access?
19. Draw the waveform of ASK,FSK for a sequence.
20. Define DPSK

PART B

1. What is PN sequence. Explain the operation of direct spread spectrum with coherent BPSK.
2. Sketch and explain working of RAKE receiver.
3. What are the different types of multiple access techniques.
4. Explain working of costas loop.

5. Explain the working of Multi pulse excited LPC and Code excited LPC by Suitable diagrams.
6. Explain in detail about QPSK modulation scheme.
7. Write about the performance of M-ary PSK.
8. Explain the function of DBPSK transmitter and receiver.
9. With neat block diagram explain the operation of QAM transmitter. Draw its output signal constellation diagram.
10. Discuss about the power spectrum and bandwidth efficiency of M-ary modulation schemes.
11. Explain BPSK transmitter and receiver with help of block diagram ?
12. The bit stream 1011100011 is to be transmitted using DPSK. Determine the encoded sequence and transmitted phase sequence.
13. Explain about Wireless Communications.
14. Derive the Processing gain of SS Modulation.
15. Explain about Carrier and Clock recovery.

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UNIT V SPREAD SPECTRUM AND MULTIPLE ACCESS TECHNIQUES

Basics – Pseudo-Noise Sequence – DS Spread Spectrum with Coherent Binary PSK
– Processing Gain – FH Spread Spectrum – Multiple Access Techniques – Wireless
Communication – TDMA and CDMA in Wireless Communication Systems – Source
Coding of Speech for Wireless Communications.

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UNIT 5

An Introduction to Direct-Sequence Spread-Spectrum Communications

Introduction

As spread spectrum techniques become increasingly popular, electrical engineers outside the field are eager for understandable explanations of the technology. There are many books and web sites on the subject but many are hard to understand or describe some aspects while ignoring others (the DSSS type, for instance, with extensive focus on PN-code generation).

The following discussion covers the full spectrum (pun intended).

A Short History

Spread-spectrum (SS) communications technology was first described on paper by an actress and a musician! In 1941, Hollywood actress Hedy Lamarr and pianist George Antheil described a secure radio link to control torpedos and received U.S. patent #2.292.387. It was not taken seriously at that time by the U.S. Army and was forgotten until the 1980s, when it came alive, and has become increasingly popular for applications that involve radio links in hostile environments.

Typical applications for the resulting short-range data transceivers include satellite-positioning systems (GPS), 3G mobile telecommunications, W-LAN (IEEE802.11a, IEEE802.11b, IEEE802.11g), and Bluetooth. SS techniques also aid in the endless race between communication needs and radio-frequency availability.

Theoretical Justification for SS

SS is apparent in the Shannon and Hartley channel-capacity theorem:

$$C = B \times \log_2(1 + S/N)$$

In this equation, C is the channel capacity in bits per second (bps), which is the maximum data rate for a theoretical bit-error rate (BER). B is the required channel bandwidth in Hz, and S/N is the signal-to-noise power ratio. To be more explicit, one assumes that C , which represents the amount of information allowed by the communication channel, also represents the desired performance. Bandwidth (B) is the price to be paid, because frequency is a limited resource. S/N ratio expresses the environmental conditions or the physical characteristics (obstacles, presence of jammers, interferences, etc.).

An elegant interpretation of this equation, applicable for difficult environments (low S/N ratio caused by noise and interference), says that one can maintain or even increase communication performance (high C) by allowing or injecting more bandwidth (high B), even when signal power is below the noise floor. (The equation does not forbid that condition!)

Modify the above equation by changing the log base from 2 to e (the Napierian number), and by noting that $\ln = \log_e$:

$$C/B = (1/\ln 2) \times \ln(1+S/N) = 1.443 \times \ln(1+S/N)$$

Applying the MacLaurin series development for $\ln(1+x) = x - x^2/2 + x^3/3 - x^4/4 + \dots + (-1)^{k+1}x^k/k + \dots$:

$$C/B = 1.443 \times (S/N - 1/2 \times (S/N)^2 + 1/3 \times (S/N)^3 - \dots)$$

S/N is usually low for spread-spectrum applications. (As just mentioned, the signal power density can be even below the noise level.) Assuming a noise level such that $S/N \ll 1$, Shannon's expression becomes simply:

$$C/B \approx 1.433 \times S/N$$

Very roughly,

$$C/B \approx S/N$$

Or : $N/S \approx B/C$

To send error-free information for a given noise-to-signal ratio in the channel, therefore, we need only perform the fundamental SS signal-spreading operation: increase the transmitted bandwidth. That principle seems simple and evident, but its implementation is complex mainly because spreading the baseband (by a factor that can be several orders of magnitude) forces the electronics to act and react accordingly, making necessary the spreading and despreading operations.

Definitions

Different SS techniques are available, but all have one idea in common: the key (also called code or sequence) attached to the communication channel. The manner of inserting this code defines precisely the SS technique in question. The term "spread spectrum" refers to the expansion of signal bandwidth, by several orders of magnitude in some cases, which occurs when a key is attached to the communication channel.

The formal definition of SS is more precise: Spread spectrum is an RF communications system in which the baseband signal bandwidth is intentionally spread over a larger bandwidth by injecting a higher-frequency signal. As a direct consequence, energy used in transmitting the signal is spread over a wider bandwidth, and appears as noise. The ratio (in dB) between the spread baseband and the original signal is called processing gain. Typical SS processing gains run from 10dB to 60dB.

To apply an SS technique, simply inject the corresponding SS code somewhere in the transmitting chain before the antenna. (That injection is called the spreading operation.) The effect is to diffuse the information in a larger bandwidth. Conversely, you can remove the SS code (despreading operation) at a point in the receive chain before data retrieval. The effect of a despreading operation is to reconstitute the information in its original bandwidth. Obviously, the same code must be known in advance at both ends of the transmission channel. (In some circumstances, it should be known only by those two parties.)

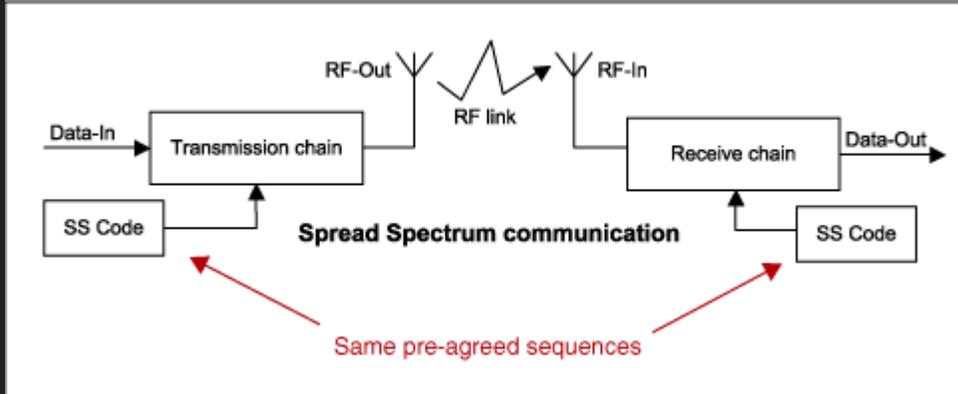
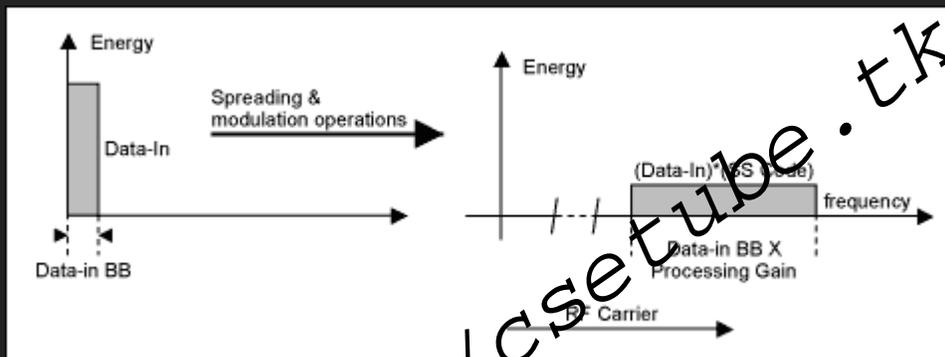


Figure 1.

Bandwidth Effects of the Spreading Operation

The simple drawings below illustrate the evaluation of signal bandwidth in a communication link.



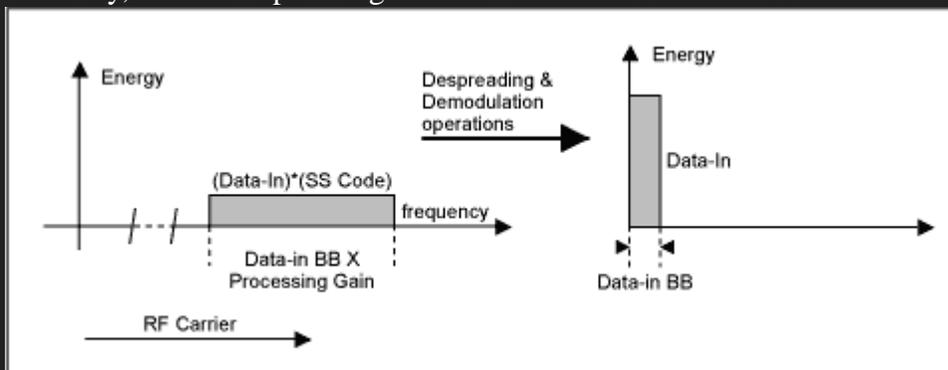
Figure

2.

SS modulation is applied on top of a conventional modulation such as BPSK or direct conversion. One can demonstrate that all other signals not receiving the SS code will stay as they are, unspread.

Bandwidth Effects of the Despreading Operation

Similarly, despreading can be seen as follows:



Figure

3.

An SS demodulation has been made on top of the normal demodulation operations above.

One can also demonstrate that signals added during the transmission (such as an interferer or jammer) will be spread during the despreading operation!

Waste of Bandwidth Due to Spreading is Offset by Multiple Users

Spreading results directly in the use of a wider frequency band (by a factor corresponding exactly to the "processing gain" mentioned earlier), so it doesn't spare the limited frequency resource. That overuse is well compensated, however, by the possibility that many users will share the enlarged frequency band.

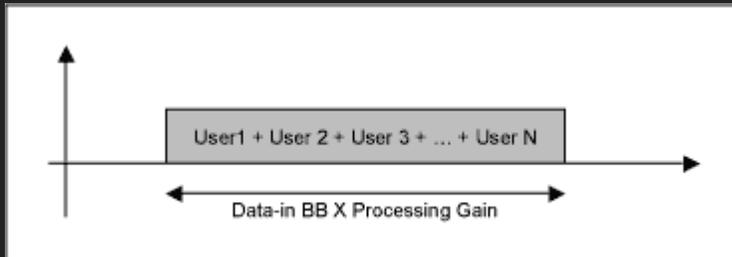


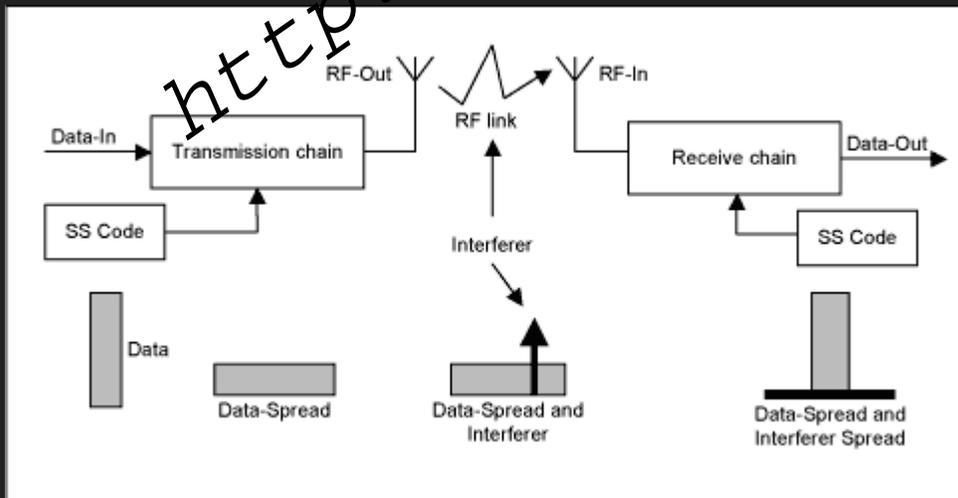
Figure 4.

SS is Wideband Technology

As opposed to regular narrowband technology, the SS process of spreading is a wideband technology. W-CDMA and UMTS, for example, are wideband technologies that require a relatively large frequency bandwidth (compared to that of narrowband radio).

Resistance to Interference, and Anti-jamming Effects

This characteristic is the real beauty of SS. Intentional or un-intentional interference and jamming signals are rejected because they do not contain the SS key. Only the desired signal, which has the key, will be seen at the receiver when the despreading operation is exercised.



Figure

5.

You can practically ignore the interference (narrowband or wideband) if it does not include the key used in the despreading operation. That rejection also applies to other SS

signals not having the right key, which allows different SS communications to be active simultaneously in the same band (such as CDMA). Note that SS is a wideband technology, but the reverse is not true. Wideband techniques need not involve SS technology.

Resistance to Interception

Resistance to interception is the second advantage provided by SS techniques. Because non-authorized listeners do not have the key used to spread the original signal, they cannot decode it. Without the right key, the SS signal appears as noise or as an interferer (scanning methods can break the code, however, if the key is short.) Even better, signal levels can be below the noise floor, because the spreading operation reduces the spectral density (total energy is the same, but it is widely spread in frequency). The message is thus made invisible, an effect that is particularly strong with the DSSS technique. Other receivers cannot "see" the transmission; they only register a slight increase in the overall noise level!

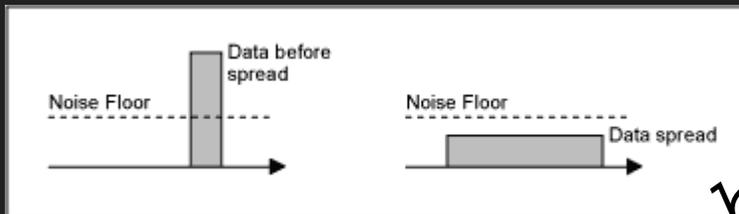
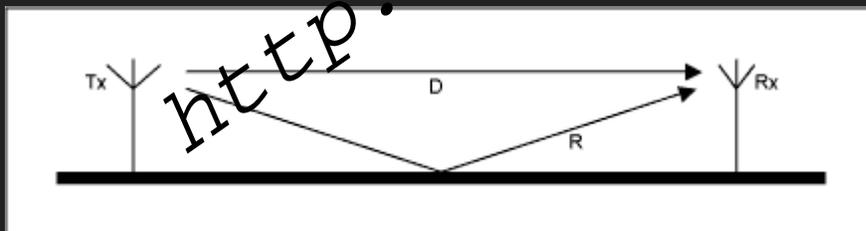


Figure 6.

Resistance to Fading (Multipath Effects)

Wireless channels often include multiple-path propagation, in which the signal has more than one path from the transmitter to the receiver. Such multipaths can be caused by atmospheric reflection or refraction, and by reflection from the ground or from objects such as buildings.



Figure

7.

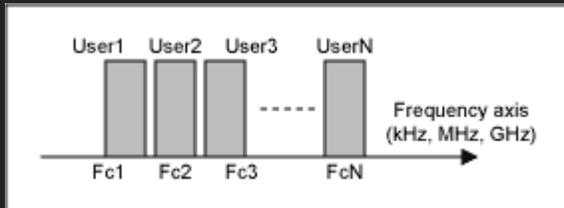
The reflected path (R) can interfere with the direct path (D) in a phenomenon called fading. Because the despreading process synchronizes to signal D, signal R is rejected even though it contains the same key. Methods are available to use the reflected-path signals by despreading them and adding the extracted results to the main one.

SS Allows CDMA

Note that SS is not a modulation scheme, and should not be confused with other types of modulation. One can, for example, use SS techniques to transmit a signal modulated via FSK or BPSK. Thanks to the coding basis, SS can also be used as another method for implementing multiple access (the real or apparent coexistence of multiple and

simultaneous communication links on the same physical media). So far, three main methods are available:

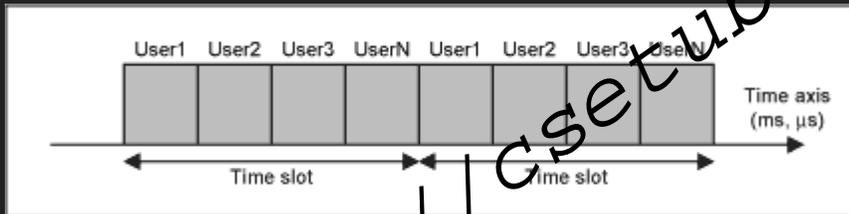
FDMA: Frequency Division Multiple Access. FDMA allocates a specific carrier frequency to a communication channel, and the number of different users is limited to the number of slices in the frequency spectrum. FDMA is the least efficient in term of frequency-band usage. Methods of FDMA access include radio broadcasting, TV, AMPS, and TETRAPOLE.



Figure

8.

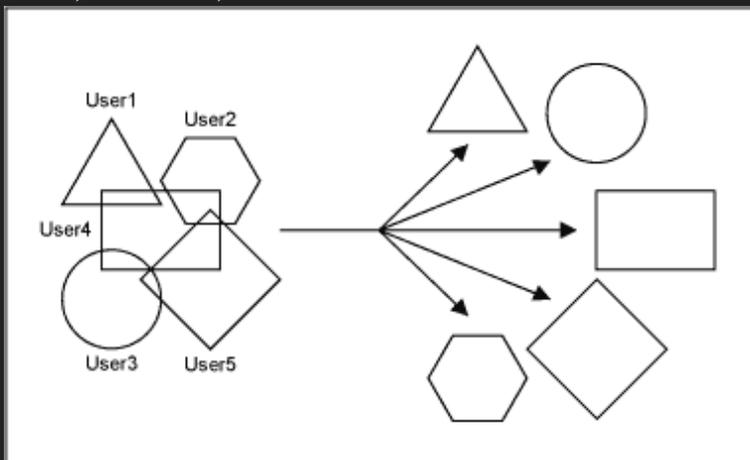
TDMA: Time Division Multiple Access. Here, the different users speak and listen to each other according to a defined allocation of time slots. Different communication channels can then be established for a unique carrier frequency. Examples of TDMA are GSM, DECT, TETRA, and IS-136.



Figure

9.

CDMA: Code Division Multiple Access. CDMA access to the air is determined by a key or code. In that sense, spread spectrum is a CDMA access. The key must be defined and known in advance at the transmitter and receiver ends. Growing examples are IS-95 (DS), IS-98, Bluetooth, and WLAN.



Figure

10.

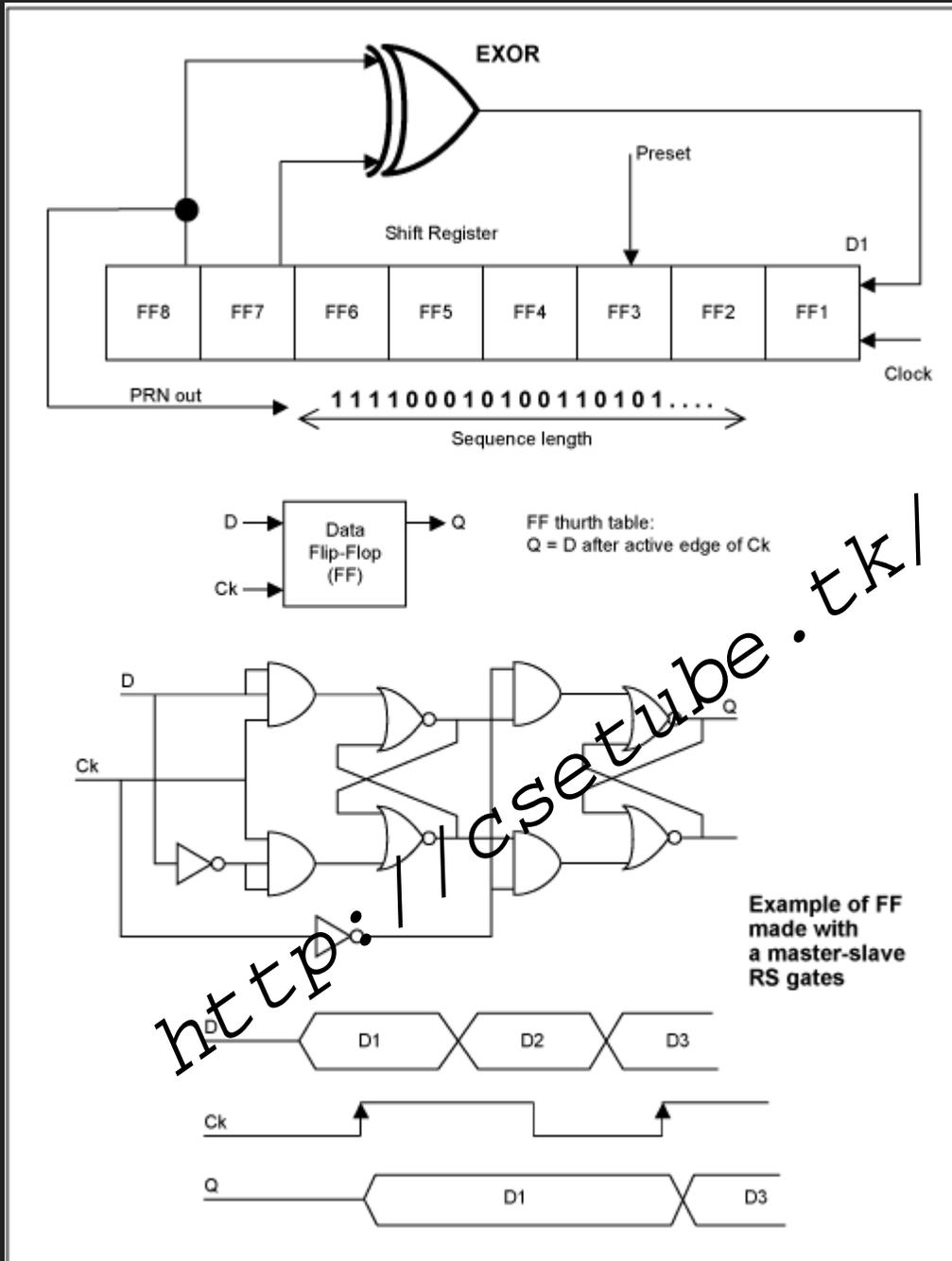
One can, of course, combine the above access methods. GSM, for instance, combines TDMA and FDMA. It defines the topological areas (cells) with different carrier frequencies, and sets time slots within each cell.

Spread Spectrum and (De)coding "Keys"

At this point, we know that the main SS characteristic is the presence of a code or key, which must be known in advance by the transmitter and receiver(s). In modern communications, the codes are digital sequences that must be as long and as random as possible to appear as "noise-like" as possible. But in any case, they must remain reproducible. Otherwise, the receiver will be unable to extract the message that has been sent. Thus, the sequence is "nearly random." Such a code is called a pseudo-random number (PRN) or sequence. The method most frequently used to generate pseudo-random

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codes is based on a feedback shift register:



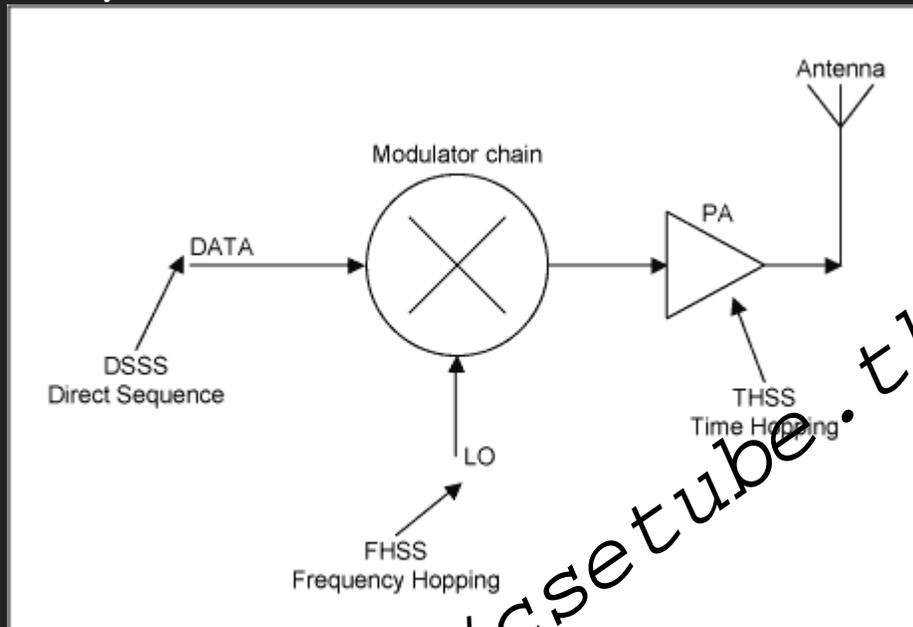
Figure

Many books are available on the generation of PRNs and their characteristics, but that development is outside the scope of this basic tutorial. We simply note that the construction or selection of proper sequences (or sets of sequences) is not trivial. To guarantee efficient SS communications, the PRN sequences must respect certain rules, such as length, auto-correlation, cross-correlation, orthogonality, and bits balancing. The more popular PRN sequences have names: Barker, M-Sequence, Gold, Hadamard-Walsh, etc. Keep in mind that a more complex sequence set provides a more robust SS link. But,

the price to pay is a more complex electronics (both in speed and behavior), mainly for the SS despreading operations. Purely digital SS despreading chips can contain more than several million equivalent 2-input NAND gates, switching at several tens of megahertz.

Different Modulation Spreading Techniques for Spread Spectrum

Different SS techniques are distinguished according to the point in the system at which a pseudo-random code (PRN) is inserted in the communication channel. This is very basically illustrated in the here below RF front end schematic :



Figure

12.

If the PRN is inserted at the data level, we have the direct sequence form of spread spectrum (DSSS). (In practice, the pseudo-random sequence is mixed or multiplied with the information signal giving an impression that the original data flow was "hashed" by the PRN.) If the PRN acts at the carrier-frequency level, we have the frequency hopping form of spread spectrum (FHSS). Applied at the LO stage, FHSS PRN codes force the carrier to change or hop according to the pseudo-random sequence. If the PRN acts as an on/off gate to the transmitted signal, we have a time hopping spread spectrum technique (THSS). There is also the chirp technique, which linearly sweeps the carrier frequency in time. One can mix all the above techniques to form a hybrid SS technique, such as DSSS + FHSS. DSSS and FHSS are the two techniques most in use today.

Direct Sequence Spread Spectrum (DSSS)

In this technique, the PRN is applied directly to data entering the carrier modulator. The modulator therefore sees a much larger bit rate, which corresponds to the chip rate of the PRN sequence. The result of modulating an RF carrier with such a code sequence is to produce a direct-sequence-modulated spread spectrum with $((\sin x)/x)^2$ frequency spectrum, centered at the carrier frequency.

The main lobe of this spectrum (null to null) has a bandwidth twice the clock rate of the

modulating code, and the sidelobes have null-to-null bandwidths equal to the code's clock rate. Illustrated below is the most common type of direct-sequence-modulated spread spectrum signal. Direct-sequence spectra vary somewhat in spectral shape, depending on the actual carrier and data modulation used. Below is a binary phase shift keyed (BPSK) signal, which is the most common modulation type used in direct sequence systems.

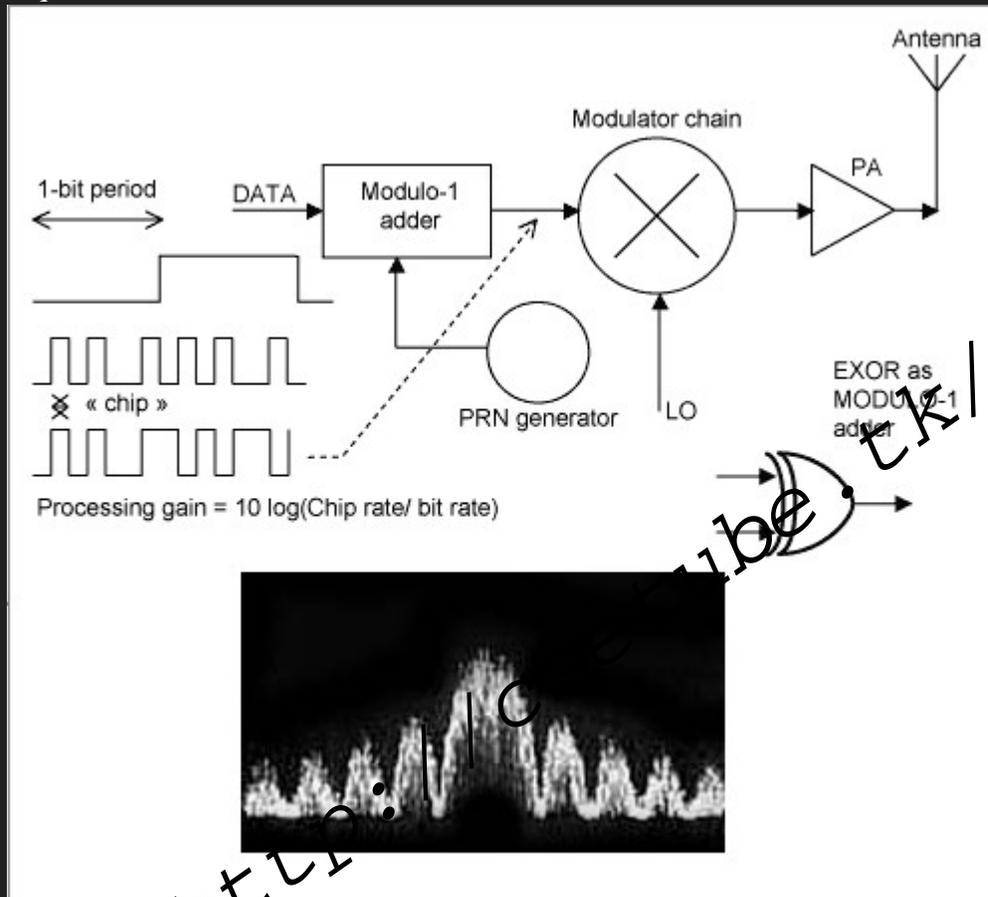


Figure 13. Spectrum-analyzer photo of a direct-sequence (DS) spread-spectrum signal. Note the original signal (non-spread) would only occupy half of the central lobe.

Frequency Hopping Spread Spectrum (FHSS)

This method does exactly what its name implies, it causes the carrier to hop from frequency to frequency over a wide band according to a sequence defined by the PRN. The speed at which the hops are executed depends on the data rate of the original information, but one can distinguish between Fast Frequency Hopping (FFHSS) and Low Frequency Hopping (LFHSS). The latter method (the most common) allows several consecutive data bits to modulate the same frequency. FFHSS, on the other hand, is characterized by several hops within each data bit.

The transmitted spectrum of a frequency hopping signal is quite different from that of a direct sequence system. Instead of a $((\sin x)/x)^2$ -shaped envelope, the frequency hopper's output is flat over the band of frequencies used (see below). The bandwidth of a frequency-hopping signal is simply N times the number of frequency slots available,

where N is the bandwidth of each hop channel.

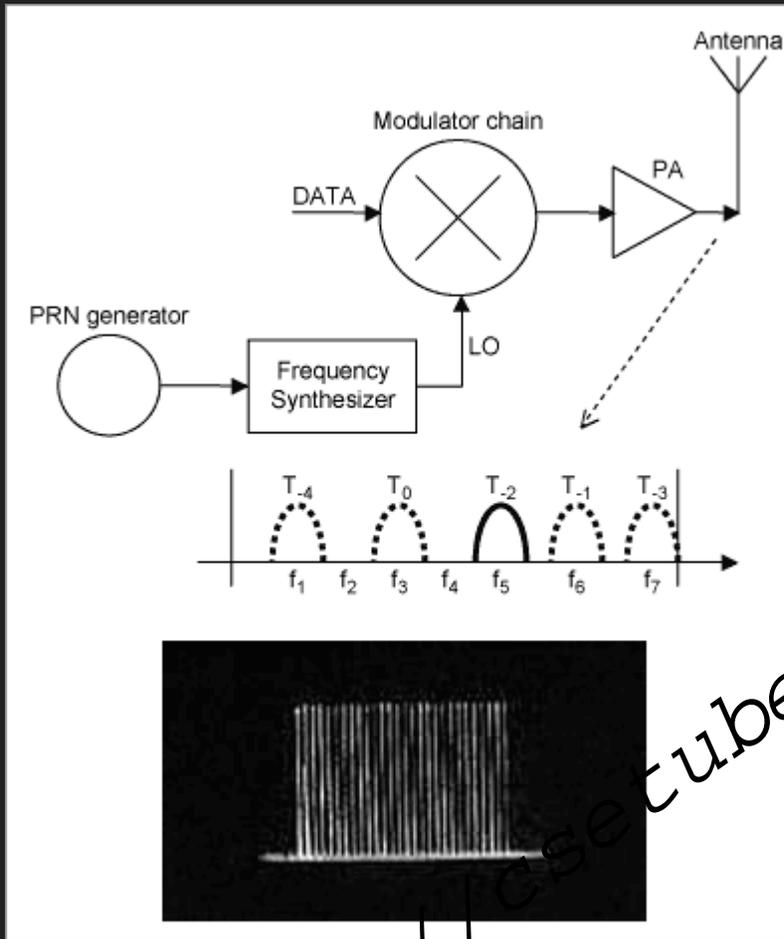
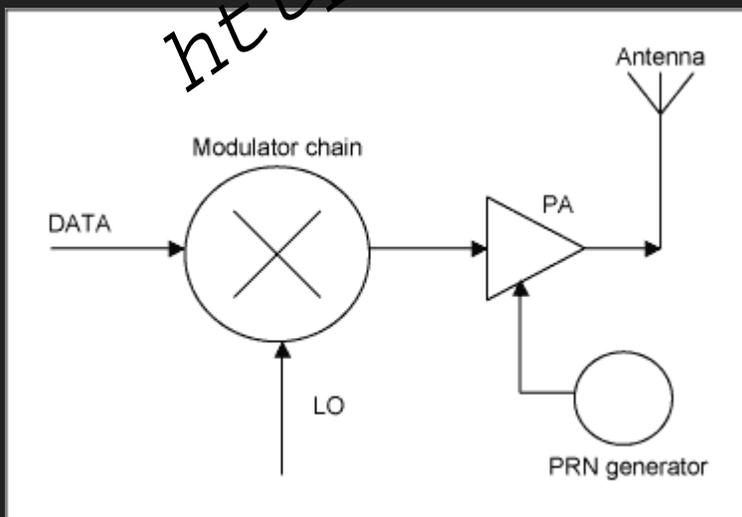


Figure 14. Spectrum-analyzer photo of a frequency-hop (FH) spread-spectrum signal.

Time Hopping Spread Spectrum (THSS)



Figure

Here, in a method not well developed today, the on and off sequences applied to the PA are dictated according to the PRN sequence.

Implementations and Conclusions

A complete SS communication link requires various advanced and up-to-date technologies and disciplines: RF antenna, powerful and efficient PA, low-noise, highly linear LNA, compact transceivers, high-resolution ADCs and DACs, rapid low-power digital signal processing (DSP), etc. Though designers and manufacturers compete, they are also joining in their effort to implement SS systems.

The most difficult area is the receiver path, especially at the despreading level for DSSS, because the receiver must be able to recognize the message and synchronize with it in real time. The operation of code recognition is also called correlation. Because correlation is performed at the digital-format level, the tasks are mainly complex arithmetic calculations including fast, highly parallel binary additions and multiplications. The most difficult aspect of today's receiver design is synchronization. More time, effort, research, and money has gone toward developing and improving synchronization techniques than toward any other aspect of SS communications.

Several methods can solve the synchronization problem, and many of them require a large number of discrete components to implement. Perhaps the biggest breakthroughs have occurred in DSP and in application specific integrated circuits (ASICs). DSP provides high-speed mathematical functions that analyze, synchronize, and decorrelate an SS signal after slicing it in many small parts. ASIC chips drive down costs via VLSI technology, and by the creation of generic building blocks suitable for any type of application.

Code division multiple access

Code division multiple access (CDMA) is a channel access method utilized by various radio communication technologies. It should not be confused with the mobile phone standards called cdmaOne and CDMA2000 (which are often referred to as simply "CDMA"), that use CDMA as their underlying channel access methods.

One of the basic concepts in data communication is the idea of allowing several transmitters to send information simultaneously over a single communication channel. This allows several users to share a bandwidth of frequencies. This concept is called multiplexing. CDMA employs spread-spectrum technology and a special coding scheme (where each transmitter is assigned a code) to allow multiple users to be multiplexed over the same physical channel. By contrast, time division multiple access (TDMA) divides access by time, while frequency-division multiple access (FDMA) divides it by frequency. CDMA is a form of "spread-spectrum" signaling, since the modulated coded signal has a much higher data bandwidth than the data being communicated.

An analogy to the problem of multiple access is a room (channel) in which people wish to communicate with each other. To avoid confusion, people could take turns speaking (time division), speak at different pitches (frequency division), or speak in different directions (spatial division). In CDMA, they would speak different languages. People speaking the same language can understand each other, but not other people. Similarly, in radio CDMA, each group of users is given a shared code. Many codes occupy the same channel, but only users associated with a particular code can understand each other.

CDMA has been used in many communications and navigation systems, including the Global Positioning System and the OmniTRACS satellite system for transportation logistics.

Uses

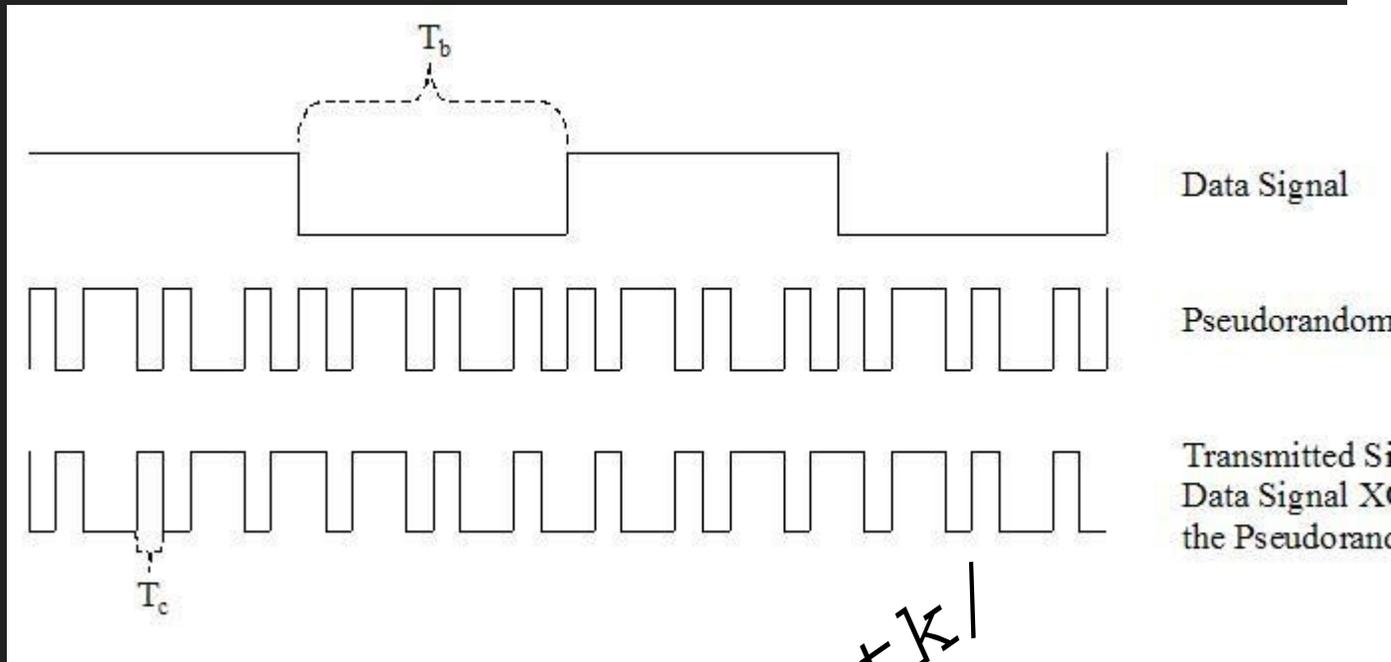
One of the early applications for code division multiplexing—predating, and distinct from cdmaOne—is in GPS.

The Qualcomm standard IS-95, marketed as cdmaOne.

The Qualcomm standard IS-2000, known as CDMA2000. This standard is used by several mobile phone companies, including the Globalstar satellite phone network.

Technical details

CDMA is a spread spectrum multiple access technique. In CDMA a locally generated code runs at a much higher rate than the data to be transmitted. Data for transmission is simply logically XOR (exclusive OR) added with the faster code. The figure shows how spread spectrum signal is generated. The data signal with pulse duration of T_b is XOR added with the code signal with pulse duration of T_c . (Note: bandwidth is proportional to $1/T$ where $T =$ bit time) Therefore, the bandwidth of the data signal is $1/T_b$ and the bandwidth of the spread spectrum signal is $1/T_c$. Since T_c is much smaller than T_b , the bandwidth of the spread spectrum signal is much larger than the bandwidth of the original signal. [1]



Each user in a CDMA system uses a different code to modulate their signal. Choosing the codes used to modulate the signal is very important in the performance of CDMA systems. The best performance will occur when there is good separation between the signal of a desired user and the signals of other users. The separation of the signals is made by correlating the received signal with the locally generated code of the desired user. If the signal matches the desired user's code then the correlation function will be high and the system can extract that signal. If the desired user's code has nothing in common with the signal the correlation should be as close to zero as possible (thus eliminating the signal); this is referred to as cross correlation. If the code is correlated with the signal at any time offset other than zero, the correlation should be as close to zero as possible. This is referred to as auto-correlation and is used to reject multi-path interference. [2]

In general, CDMA belongs to two basic categories: synchronous (orthogonal codes) and asynchronous (pseudorandom codes).

Code Division Multiplexing (Synchronous CDMA)

Synchronous CDMA exploits mathematical properties of orthogonality between vectors representing the data strings. For example, binary string "1011" is represented by the vector (1, 0, 1, 1). Vectors can be multiplied by taking their dot product, by summing the products of their respective components. If the dot product is zero, the two vectors are said to be orthogonal to each other. (Note: If $u=(a,b)$ and $v=(c,d)$, the dot product $u \cdot v = a \cdot c + b \cdot d$) Some properties of the dot product help to understand how WCDMA works. If vectors a and b are orthogonal, then

Each user in synchronous CDMA uses an orthogonal codes to modulate their signal. An example of four mutually orthogonal digital signals is shown in the figure. Orthogonal

codes have a cross-correlation equal to zero; in other words, they do not interfere with each other. In the case of IS-95 64 bit Walsh codes are used to encode the signal to separate different users. Since each of the 64 Walsh codes are orthogonal to one another, the signals are channelized into 64 orthogonal signals. The following example demonstrates how each users signal can be encoded and decoded.

Example

Start with a set of vectors that are mutually orthogonal. (Although mutual orthogonality is the only condition, these vectors are usually constructed for ease of decoding, for example columns or rows from Walsh matrices.) An example of orthogonal functions is shown in the picture on the left. These vectors will be assigned to individual users and are called the "code", "chipping code" or "chip code". In the interest of brevity, the rest of this example uses codes (v) with only 2 digits.

An example of four mutually orthogonal digital signals.

Each user is associated with a different code, say v . If the data to be transmitted is a digital zero, then the actual bits transmitted will be $-v$, and if the data to be transmitted is a digital one, then the actual bits transmitted will be v . For example, if $v=(1,-1)$, and the data that the user wishes to transmit is (1, 0, 1, 1) this would correspond to $(v, -v, v, v)$ which is then constructed in binary as $((1,-1),(-1,1),(1,-1),(1,-1))$. For the purposes of this article, we call this constructed vector the transmitted vector.

Each sender has a different, unique vector v chosen from that set, but the construction method of the transmitted vector is identical.

Now, due to physical properties of interference, if two signals at a point are in phase, they add to give twice the amplitude of each signal, but if they are out of phase, they "subtract" and give a signal that is the difference of the amplitudes. Digitally, this behaviour can be modelled by the addition of the transmission vectors, component by component.

If sender0 has code (1,-1) and data (1,0,1,1), and sender1 has code (1,1) and data (0,0,1,1), and both senders transmit simultaneously, then this table describes the coding steps:

Step	Encode sender0	Encode sender1
0	vector0=(1,-1), data0=(1,0,1,1)=(1,-1,1,1)	vector1=(1,1), data1=(0,0,1,1)=(-1,-1,1,1)
1	encode0=vector0.data0	encode1=vector1.data1
2	encode0=(1,-1).(1,-1,1,1)	encode1=(1,1).(-1,-1,1,1)
3	encode0=((1,-1),(-1,1),(1,-1),(1,-1))	encode1=((-1,-1),(-1,-1),(1,1),(1,1))

signal0=(1,-1,1,1,-1,1,-1) signal1=(1,-1,-1,1,1,-1,1)

Because signal0 and signal1 are transmitted at the same time into the air, they add to produce the raw signal:
 $(1,-1,-1,1,1,-1,1,-1) + (-1,-1,-1,-1,1,1,1,1) = (0,-2,-2,0,2,0,2,0)$

This raw signal is called an interference pattern. The receiver then extracts an intelligible signal for any known sender by combining the sender's code with the interference pattern, the receiver combines it with the codes of the senders. The following table explains how this works and shows that the signals do not interfere with one another:

Step	Decode sender0	Decode sender1
0	vector0=(1,-1), 2,0,2,0,2,0)	vector1=(1,1), 2,0,2,0,2,0)
1	decode0=pattern.vector0	decode1=pattern.vector1
2	decode0=((0,-2),(-2,0),(2,0),(2,0)).(1,-1)	decode1=((0,-2),(-2,0),(2,0),(2,0)).(1,1)
3	decode0=((0+2),(-2+0),(2+0),(2+0))	decode1=((0-2),(-2+0),(2+0),(2+0))
4	data0=(2,-2,2,2)=(1,0,1,1)	data1=(-2,-2,2,2)=(0,0,1,1)

Further, after decoding, all values greater than 0 are interpreted as 1 while all values less than zero are interpreted as 0. For example, after decoding, data0 is (2,-2,2,2), but the receiver interprets this as (1,0,1,1).

We can also consider what would happen if a receiver tries to decode a signal when the user has not sent any information. Assume signal0=(1,-1,-1,1,1,-1,1,-1) is transmitted alone. The following table shows the decode at the receiver:

Step	Decode sender0	Decode sender1
0	vector0=(1,-1), 1)	vector1=(1,1), 1)
1	decode0=pattern.vector0	decode1=pattern.vector1
2	decode0=((1,-1),(-1,1),(1,-1),(1,-1)).(1,-1)	decode1=((1,-1),(-1,1),(1,-1),(1,-1)).(1,1)
3	decode0=((1+1),(-1-1),(1+1),(1+1))	decode1=((1-1),(-1+1),(1-1),(1-1))
4	data0=(2,-2,2,2)=(1,0,1,1)	data1=(0,0,0,0)

When the receiver attempts to decode the signal using sender1's code, the data is all zeros, therefore the cross correlation is equal to zero and it is clear that sender1 did not transmit any data.

Asynchronous CDMA

The previous example of orthogonal Walsh sequences describes how 2 users can be multiplexed together in a synchronous system, a technique that is commonly referred to as Code Division Multiplexing (CDM). The set of 4 Walsh sequences shown in the figure will afford up to 4 users, and in general, an $N \times N$ Walsh matrix can be used to multiplex N users. Multiplexing requires all of the users to be coordinated so that each transmits their assigned sequence v (or the complement, $-v$) starting at exactly the same time. Thus, this technique finds use in base-to-mobile links, where all of the transmissions originate from the same transmitter and can be perfectly coordinated.

On the other hand, the mobile-to-base links cannot be precisely coordinated, particularly due to the mobility of the handsets, and require a somewhat different approach. Since it is not mathematically possible to create signature sequences that are orthogonal for arbitrarily random starting points, unique "pseudo-random" or "pseudo-noise" (PN) sequences are used in Asynchronous CDMA systems. A PN code is a binary sequence that appears random but can be reproduced in a deterministic manner by intended receivers. These PN codes are used to encode and decode a users signal in Asynchronous CDMA in the same manner as the orthogonal codes in synchronous CDMA (shown in the example above). These PN sequences are statistically uncorrelated, and the sum of a large number of PN sequences results in Multiple Access Interference (MAI) that is approximated by a Gaussian noise process (following the "central limit theorem" in statistics). If all of the users are received with the same power level, then the variance (e.g., the noise power) of the MAI increases in direct proportion to the number of users. In other words, unlike synchronous CDMA, the signals of other users will appear as noise to the signal of interest and interfere slightly with the desired signal in proportion to number of users.

All forms of CDMA use spread spectrum process gain to allow receivers to partially discriminate against unwanted signals. Signals encoded with the specified PN sequence (code) are received, while signals with different codes (or the same code but a different timing offset) appear as wideband noise reduced by the process gain.

Since each user generates MAI, controlling the signal strength is an important issue with CDMA transmitters. A CDM (Synchronous CDMA), TDMA or FDMA receiver can in theory completely reject arbitrarily strong signals using different codes, time slots or frequency channels due to the orthogonality of these systems. This is not true for Asynchronous CDMA; rejection of unwanted signals is only partial. If any or all of the unwanted signals are much stronger than the desired signal, they will overwhelm it. This leads to a general requirement in any Asynchronous CDMA system to approximately match the various signal power levels as seen at the receiver. In CDMA cellular, the base station uses a fast closed-loop power control scheme to tightly control each mobile's transmit power. See Near-far problem for further information on this problem.

Advantages of Asynchronous CDMA over other techniques

Asynchronous CDMA's main advantage over CDM (Synchronous CDMA), TDMA and FDMA is that it can use the spectrum more efficiently in mobile telephony applications.

(In theory, CDMA, TDMA and FDMA have exactly the same spectral efficiency but practically, each has its own challenges - power control in the case of CDMA, timing in the case of TDMA, and frequency generation/filtering in the case of FDMA.) TDMA systems must carefully synchronize the transmission times of all the users to ensure that they are received in the correct timeslot and do not cause interference. Since this cannot be perfectly controlled in a mobile environment, each timeslot must have a guard-time, which reduces the probability that users will interfere, but decreases the spectral efficiency. Similarly, FDMA systems must use a guard-band between adjacent channels, due to the random doppler shift of the signal spectrum which occurs due to the user's mobility. The guard-bands will reduce the probability that adjacent channels will interfere, but decrease the utilization of the spectrum.

Most importantly, Asynchronous CDMA offers a key advantage in the flexible allocation of resources. There are a fixed number of orthogonal codes, timeslots or frequency bands that can be allocated for CDM, TDMA and FDMA systems, which remain underutilized due to the bursty nature of telephony and packetized data transmissions. There is no strict limit to the number of users that can be supported in an Asynchronous CDMA system, only a practical limit governed by the desired bit error probability, since the SIR (Signal to Interference Ratio) varies inversely with the number of users. In a bursty traffic environment like mobile telephony, the advantage afforded by Asynchronous CDMA is that the performance (bit error rate) is allowed to fluctuate randomly, with an average value determined by the number of users times the percentage of utilization. Suppose there are $2N$ users that only talk half of the time, then $2N$ users can be accommodated with the same average bit error probability as N users that talk all of the time. The key difference here is that the bit error probability for N users talking all of the time is constant, whereas it is a random quantity (with the same mean) for $2N$ users talking half of the time.

In other words, Asynchronous CDMA is ideally suited to a mobile network where large numbers of transmitters each generate a relatively small amount of traffic at irregular intervals. CDM (Synchronous CDMA), TDMA and FDMA systems cannot recover the underutilized resources inherent to bursty traffic due to the fixed number of orthogonal codes, time slots or frequency channels that can be assigned to individual transmitters. For instance, if there are N time slots in a TDMA system and $2N$ users that talk half of the time, then half of the time there will be more than N users needing to use more than N timeslots. Furthermore, it would require significant overhead to continually allocate and deallocate the orthogonal code, time-slot or frequency channel resources. By comparison, Asynchronous CDMA transmitters simply send when they have something to say, and go off the air when they don't, keeping the same PN signature sequence as long as they are connected to the system.

Spread Spectrum Characteristics of CDMA

Most modulation schemes try to minimize the bandwidth of this signal since bandwidth is a limited resource. However, spread spectrum techniques use a transmission bandwidth that is several orders of magnitude greater than the minimum required signal bandwidth.

One of the initial reasons for doing this was military applications including guidance and communication systems. These systems were designed using spread spectrum because of its security and resistance to jamming. Asynchronous CDMA has some level of privacy built in because the signal is spread using a pseudorandom code; this code makes the spread spectrum signals appear random or have noise-like properties. A receiver cannot demodulate this transmission without knowledge of the pseudorandom sequence used to encode the data. CDMA is also resistant to jamming. A jamming signal only has a finite amount of power available to jam the signal. The jammer can either spread its energy over the entire bandwidth of the signal or jam only part of the entire signal. [3]

CDMA can also effectively reject narrowband interference. Since narrowband interference affects only a small portion of the spread spectrum signal, it can easily be removed through notch filtering without much loss of information. Convolution encoding and interleaving can be used to assist in recovering this lost data. CDMA signals are also resistant to multipath fading. Since the spread spectrum signal occupies a large bandwidth only a small portion of this will undergo fading due to multipath at any given time. Like the narrowband interference this will result in only a small loss of data and can be overcome.

Another reason CDMA is resistant to multipath interference is because the delayed versions of the transmitted pseudorandom codes will have poor correlation with the original pseudorandom code, and will thus appear as another user, which is ignored at the receiver. In other words, as long as the multipath channel induces at least one chip of delay, the multipath signals will arrive at the receiver such that they are shifted in time by at least one chip from the intended signal. The correlation properties of the pseudorandom codes are such that this slight delay causes the multipath to appear uncorrelated with their intended signal, and it is thus ignored. However, spread spectrum signals can also exploit the multipath delay components to improve the performance of the system by using a Rake receiver which anticipates multipath propagation delays of the transmitted spread spectrum signal and combines the information obtained from several resolvable multipath components to produce a stronger version of the signal. [4]

Frequency reuse is the ability to reuse the same radio channel frequency at other cell sites within a cellular system. In the FDMA and TDMA systems frequency planning is an important consideration. The frequencies used in different cells need to be planned carefully in order to ensure that the signals from different cells do not interfere with each other. In a CDMA system the same frequency can be used in every cell because channelization is done using the pseudorandom codes. Reusing the same frequency in every cell eliminates the need for frequency planning in a CDMA system; however, planning of the different pseudorandom sequences must be done to ensure that the received signal from one cell does not correlate with the signal from a nearby cell. [5]

Since adjacent cells use the same frequencies, CDMA systems have the ability to perform soft handoffs. Soft handoffs allow the mobile telephone to communicate simultaneously with two or more cells. The best signal quality is selected until the handoff is complete.

SOURCE CODING OF SPEECH FOR WIRELESS COMMUNICATION

SOURCE CODING OF SPEECH FOR WIRELESS COMMUNICATION

LPC

I) CELP (CODE EXCITED LPC)

II) MELP (MULTIPULSE EXCITED LPC)

UNIVERSITY QUESTION PAPERS
B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2010
Third Semester
Computer Science and Engineering
EC1207 - ANALOG AND DIGITAL COMMUNICATION
(Regulation 2008)

Time: Three hours Maximum: 100 Marks

Answer ALL Questions

PART A- (10 x 2 = 20 Marks)

1. In an Amplitude modulation system, the carrier frequency is $f_c = 100\text{ kHz}$. The maximum frequency of the signal is 5 kHz . Determine the lower and upper side bands and the bandwidth of AM signal.
2. The maximum frequency deviation in an FM is 10 kHz and signal frequency is 10 kHz . Find out the bandwidth using Carson's rule and the modulation index.
3. Draw the ASK and FSK signals for the binary signal $s(t) = 1011001$.
4. What are the advantages of QPSK?
5. Define Nyquist sampling theorem.
6. For the signal $m(t) = 3 \cos 500t + 4 \sin 1000t$, Determine the Nyquist sampling rate.
7. What is meant by ASCII code?
8. Which error detection technique is simple and which one is more reliable?
9. What are the applications of spread spectrum modulation?
10. Design processing gain in spread spectrum modulation.

PART B- (5 x 16 = 80 Marks)

<http://csetube.weebly.com/>

11. (a) (i) Distinguish between FM and PM by giving its

mathematical analysis. (8 Marks)

(ii) Derive the relationship between the voltage amplitudes of the side band frequencies and the carrier and draw the frequency spectrum. (8 Marks)

(Or)

(b) (i) Discuss about the sets of side bands produced when a carrier is frequency modulated by a single frequency sinusoid. (8 Marks)

(ii) In an AM modulator, 500 KHz carrier of amplitude 20 V is modulated by 10 KHz modulating signal which causes a change in the output wave of ± 7.5 V. Determine:

(1) Upper and lower side band frequencies

(2) Modulation Index

<http://csetube.tk/>

- (3) Peak amplitude of upper and lower side frequency
- (4) Maximum and minimum amplitudes of envelope. (8 Marks)

12. (a) What is known as Binary phase shift keying? Discuss in detail the BPSK transmitter and Receiver and also obtain the minimum double sided Nyquist bandwidth. (16 Marks)

(Or)

- (b) (i) Illustrate the concept of 8 QAM transmitter with the truth table. (8 Marks)
- (ii) What is the need for carrier Recovery? Explain the Costas loop method of carrier recovery. (8 Marks)

13. (a) (i) What is called companding? Briefly discuss the Analog companding. (8 Marks)

(ii) Discuss about the causes of ISI. (8 Marks)

(Or)

- (b) (i) Explain in detail the Delta modulation transmitter and Receiver. (10 Marks)
- (ii) Discuss the draw backs of delta modulation and explain the significance of adaptive delta modulator. (5 Marks)

14. (a) (i) Describe the most common error detection techniques. (12 Marks)

(ii) Discuss the function of a data modem. (4 Marks)

(Or)

- (b) (i) Explain in detail the characteristics of IEEE 488 Bus. (10 Marks)
- (ii) Briefly explain the three methods of error connection. (6 Marks)

15. (a) (i) What is a Pseudo noise sequence? What are the properties of Pseudo noise sequence? (8 Marks)

(ii) Describe the application of CDMA in wireless communication system. (8 Marks)

(Or)

(b) (i) With a block diagram explain, DS spread spectrum with coherent binary PSK. (10 Marks)

(ii) Explain the near- far problem in spread spectrum modulation? (6 Marks)

B.E./B.Tech EXAMINATIONS, NOVEMBER/DECEMBER 2009
THIRD SEMESTER

COMPUTER SCIENCE AND ENGINEERING EC1207-
ANALOG AND DIGITAL COMMUNICATIONS
(REGULATIONS 2008)

Time: 3hrs Maximum: 100 marks

Answer all questions

PART A-(10X2=20 marks)

1. A carrier wave is represented by equation $s(t) = 12\sin\omega t$. Draw the wave form of an AM wave for depth of modulation of 0.5.
2. Compare FM with AM
3. What is coherent detection?
4. Why is binary ASK called on-off keying?
5. What are the errors in DM?
6. Define companding
7. What are the two methods of error detection and correction?
8. What do you mean by signaling rate?
9. Define processing gain
10. What is CDMA?

PART A-(5X16=80 marks)

11. (a) (i) The output voltage of a transmitter is given by $500(1 + 0.4\sin 3140t)\cos 6.28 \times 10^7 t$

This voltage is fed to a load of 600 ohms. Determine

1. Carrier frequency
2. Modulating frequency
3. Carrier power (9)

(ii) Explain in detail about superheterodyne receiver (7)

Or

(b) (i) Carrier frequency modulated with a sinusoidal signal of 2kHz resulting in a maximum frequency deviation of 5kHz. Find

1. Modulation index
2. Bandwidth of modulated signal (4)

(ii) Explain the method of generating FM signal using indirect method (12)

12. (a) (i) Explain the coherent binary FSK system with a neat diagram of transmitter and receiver (12)

(ii) Enumerate the advantages and disadvantages of FSK over PSK system (4)

Or

(b)(i) Explain the generation and detection of coherent QPSK system in detail(12)

(ii) What is DPSK? Explain its bandwidth requirements.

13.(a)(i) Explain in detail about DPCM with suitable diagram(10)

(ii) 1kHz signal is sampled by 8kHz sampling signal and the samples are encoded with 12

bit PCM system. Find

1. Required bandwidth for PCM system

2. Total number of bits in the digital output signal in 10 cycles(6)

Or

(b)(i) Write short notes on ISI (6)

(ii) What is eye pattern? Explain how is the performance of a baseband pulse transmission system measured with this?(10)

14(a)(i) Write short notes on error correcting codes

(ii) Find the generator polynomial of $(7,4)$ cyclic code and find the codeword for the message

1001(12)

Or

(b) Explain in detail about the serial interface with its control signals and timing information (16)

15(a)(i) What are pseudo noise sequences? How are they generated?(6)

(ii) Explain direct sequence spread spectrum system in detail(10)

Or

(b)(i) Explain 2 types of FH spread spectrum systems with suitable diagrams(16)

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