IInd MIT TERM

GOVT. WOMEN ENGG. COLLEGE, AJMER

Sub - Analog Communication

V-sem Batch (ECE)

Time: 1hr **MM: 20**

All questions carry equal marks

Q.1 Explain the principle of FM wave transmission using direct method [5]

Q.2 what is the difference between narrow-band FM and wide-band FM. [5]

Q.3 Discuss PWM modulation using suitable figure. [5]

Q.4 Write a note on sampling [5]

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Solutions

1. Armstrong method for FM generation:

- The direct methods cannot be used for the broadcast applications. Thus the alternative method i.e. indirect method called as the Armstrong method of FM generation is used.
- In this method the FM is obtained through phase modulation. A crystal oscillator can be used hence the frequency stability is very high.

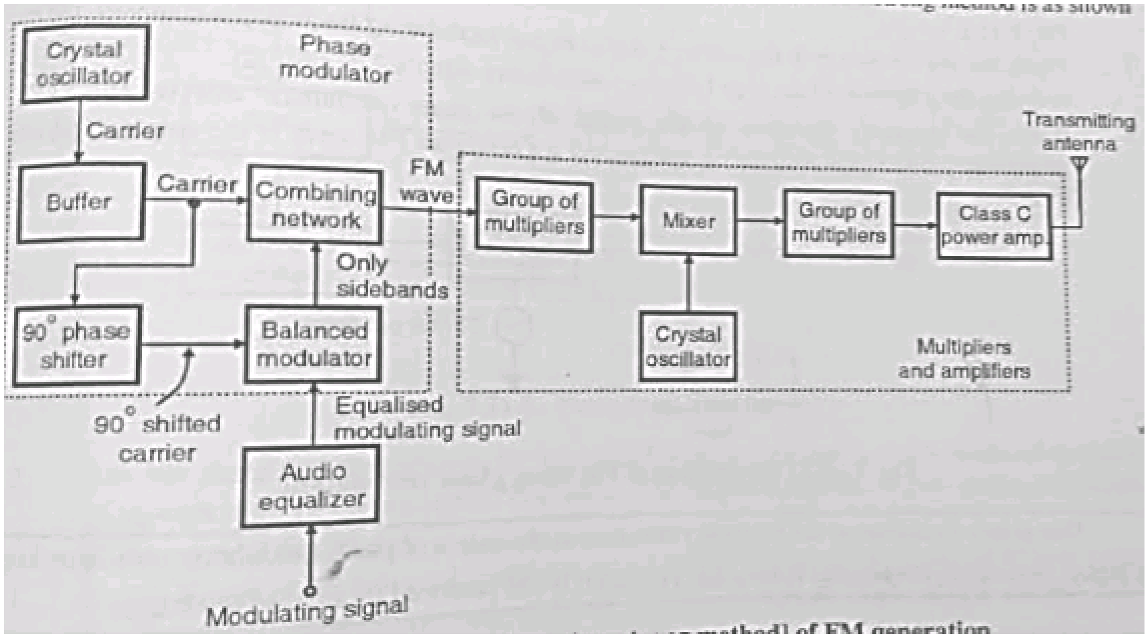


Fig: Indirect method [Armstrong method] of FM generation

Operation:

- The crystal oscillator generates the carrier at low frequency typically at 1MHz. This is applied to the combining network and a 90° phase shifter.
- The modulating signal is passed through an audio equalizer to boost the low modulating frequencies .The modulating signal is then applied to a balanced modulator.
- The balanced modulator produced two side bands such that their resultant is 90° phase shifted with respect to the unmodulated carrier.
- The unmodulated carrier and 90° phase shifted sidebands are added in the combining network.
- At the output of the combining network we get F_mFm wave. This wave has a low carrier frequency $f_c$fc and low value of the modulation index $m_f$mf .
- The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a mixer and then the $f_c$fc and $m_f$mf both are raised to required high values using the second group of multipliers.
- The FM signal with high $f_c$fc and high $m_f$mf is then passed through a class C power amplifier to raise the power level of the FM signal.
- The Armstrong method uses the phase modulation to generate frequency modulation. This method can be understood by dividing it into four parts as follows:

1 Generation of FM from phase modulator:

- The modulating signal is passed through a low pass RC filter.

- The filter output is then applied to a phase modulator along with carrier.
- Hence the extra deviation in the carrier fcfc due to higher modulating frequency is compensated by reducing the amplitude of the high frequency modulating signals.
- Hence the frequency deviation at the output of the phase modulator will be effectively proportional only to the modulating voltage and we obtain an FM wave at the output of phase modulator.

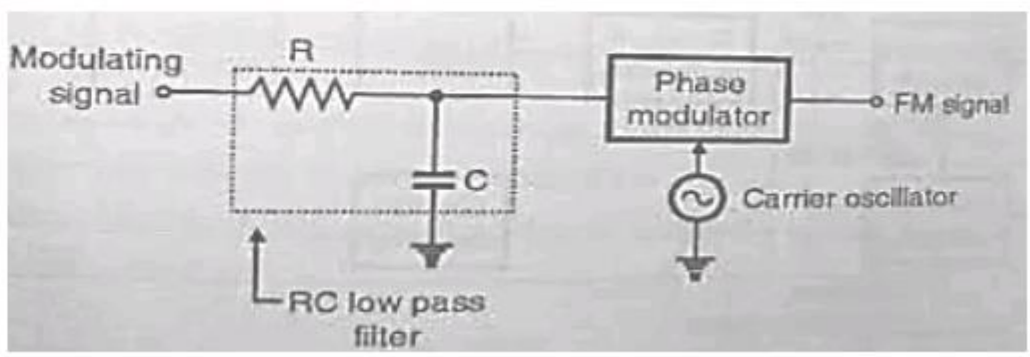


Fig: Generation of FM using phase modulation

2 Implementation of phase modulator:

- The crystal oscillator produces a stable unmodulated carrier which is applied to the “90° phase shifter” as well as the “combining network” through a buffer.

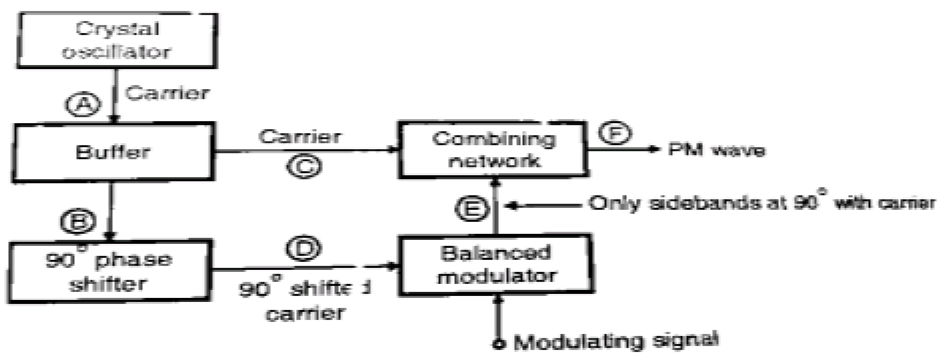


Fig: Phase modulator circuit

- The 90° phase shifter produces a 90° phase shifted carrier. It is then applied to the balanced modulator along with the modulation signal.
- At the output of the balanced modulator we get DSBSC signal i.e. AM signal without carrier. This signal consists of only two sidebands with their resultant in phase with their resultant in phase with the 90° phase shifted carrier.

Combining parts 1 and 2 to obtain The FM:

- Combining the parts 1 and 2 we get the block diagram of the Armstrong method of FM generation

Use of frequency multipliers and amplifiers:

- The FM signal produced at the output of phase modulator has a low carrier frequency and low modulation index. They are increased to an adequately high value with the help of frequency multipliers and mixer. The power level is raised to the desired level by the amplifier.

2. **Narrow Band FM**

In NBFM $\beta << 1$, therefor s(t) reduces as follows: $s(t) = Ac \cos(2\pi fct + \beta \sin(2\pi fmt)) = Ac \cos(2\pi fct) \cos(\beta \sin(2\pi fmt)) - Ac \sin(2\pi fct)\sin(\beta \sin(2\pi fmt))$ Since, β is very small, the above equation reduces to $s(t) = Ac \cos(2\pi fct) - Ac\beta \sin(2\pi fmt)\sin(2\pi fct)$. The above equation is similar to AM. Hence, for NBFM the bandwidth is same as that of AM i.e., 2 × message bandwidth(2 × B). A NBFM signal is generated. DSB−SC oscillator + m(t) A cos(t) ω −Asin(t) ω NBFM signal Phase shifter π/2 c c Figure 2: Generation of NBFM signal

Wide-Band FM (WBFM)

A WBFM signal has theoritically infinite bandwidth. Spectrum calculation of WBFM signal is a tedious process. For, practical applications however the Bandwidth of a WBFM signal is calculated as follows: Let m(t) be bandlimited to BHz and sampled adequately at 2BHz. If time period $T = 1/2B$ is too small, the signal can be approximated by sequence of pulses

3. **PWM:** Pulse-width modulation (PWM), as it applies to motor control, is a way of delivering energy through a succession of pulses rather than a continuously varying (analog) signal. By increasing or decreasing pulse width, the controller regulates energy flow to the motor shaft. The motor's own inductance acts like a filter, storing energy during the "on" cycle while releasing it at a rate corresponding to the input or reference signal. In other words, energy flows into the load not so much the switching frequency, but at the reference frequency. PWM is somewhat like pushing a playground-style merry-go-round. The energy of each push is stored in the inertia of the heavy platform, which accelerates gradually with harder, more frequent, or longer-lasting pushes. The riders receive the kinetic energy in a very different manner than how it's applied.

Modulation:

Modulation is a process through which audio, video, image or text information is added to an electrical or optical carrier signal to be transmitted over a telecommunication or electronic medium. Modulation enables the transfer of information on an electrical signal to a receiving device that demodulates the signal to extract the blended information.

Modulation is primarily used in telecommunication technologies that require the transmission of data via electrical signals. It is considered the backbone of data communication because it enables the use of electrical and optical signals as information carriers. Modulation is achieved by altering the periodic waveform or the carrier. This includes carrying its amplitude, frequency and phase. Modulation has three different types:

1. Amplitude Modulation (AM): Amplitude of the carrier is modulated.
2. Frequency Modulation (FM): Frequency of the carrier is modulated.
3. Phase Modulation (PM): Phase of the carrier is modulated.

A modem is a common example/implementation of a modulation technique in which the data is modulated with electrical signals and transmitted over telephone lines. It is later demodulated to receive the data.

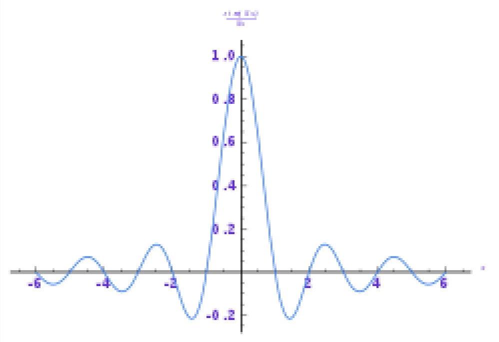
4. Sampling:

Sampling is a process of converting a signal (for example, a function of continuous time and/or space) into a numeric sequence (a function of discrete time and/or space). Shannon'sversion of the theorem states:[2]

If a function x(t) contains no frequencies higher than *B* hertz, it is completely determined by giving its ordinates at a series of points spaced 1/(2*B*) seconds apart.

A sufficient sample-rate is therefore 2*B* samples/second, or anything larger. Equivalently, for a given sample rate f_s, perfect reconstruction is guaranteed possible for a bandlimit $B < f_s/2$.

When the bandlimit is too high (or there is no bandlimit), the reconstruction exhibits imperfections known as aliasing. Modern statements of the theorem are sometimes careful to explicitly state that $x(t)$ must contain no sinusoidal component at exactly frequency *B*, or that *B* must be strictly less than ½ the sample rate. The two thresholds, 2*B* and $f_s/2$ are respectively called the **Nyquist rate** and **Nyquist frequency**. And respectively, they are attributes of $x(t)$ and of the sampling equipment. The condition described by these inequalities is called the **Nyquist criterion**, or sometimes the *Raabe condition*. The theorem is also applicable to functions of other domains, such as *space*, in the case of a digitized image. The only change, in the case of other domains, is the units of measure applied to *t*, f_s, and *B*.



The normalized sinc function: sin(π*x*) / (π*x*) ... showing the central peak at *x*= 0, and zero-crossings at the other integer values of *x*.

The symbol $T = 1/f_s$ is customarily used to represent the interval between samples and is called the **sample period** or **sampling interval**. And the samples of function $x(t)$ are commonly denoted by $x[n] = x(nT)$ (alternatively "x_n" in older signal processing literature), for all integer values of *n*. A mathematically ideal way to interpolate the sequence involves the use of sinc functions. Each sample in the sequence is replaced by a sinc function, centered on the time axis at the original location of the sample, *nT*, with the amplitude of the sinc function scaled to the sample value, $x[n]$. Subsequently, the sinc functions are summed into a continuous function. A mathematically equivalent method is to convolve one sinc function with a series of Dirac delta pulses, weighted by the sample values. Neither method is numerically practical. Instead, some type of approximation of the sinc functions, finite in length, is used. The imperfections attributable to the approximation are known as *interpolation error*.

Practical digital-to-analog converters produce neither scaled and delayed sinc functions, nor ideal Dirac pulses. Instead they produce a piecewise-constant sequence of scaled and delayed rectangular pulses (the zero-order hold), usually followed by an "anti-imaging filter" to clean up spurious high-frequency content.