Kafrelsheikh University - Faculty of Engineering			
Course	Communication systems	Date	//2018
Time	3 Hours	Mark	85
Students	3th year Electronics and Electrical Communications		

Answer all the following questions:

Clarify your answer with the suitable diagrams.

Q1.a Explain The Conventional Amplitude Modulation including its general Equation, the modulated signal in the time domain and its advantages and disadvantages.
(5 Marks)

ANS:

A conventional AM signal consists of a large carrier component in addition to the double-sideband AM modulated signal. The transmitted signal is expressed mathematically as

$$u(t) = A_c[1 + m(t)]\cos(2\pi f_c t + \phi_c) \tag{3.2.7}$$

where the message waveform is constrained to satisfy the condition that $|m(t)| \leq 1$. We observe that $A_c m(t)\cos(2\pi f_c t + \phi_c)$ is a double-sideband AM signal and $A_c\cos(2\pi f_c t + \phi_c)$ is the carrier component. Figure 3.5 illustrates an AM signal in the time domain.

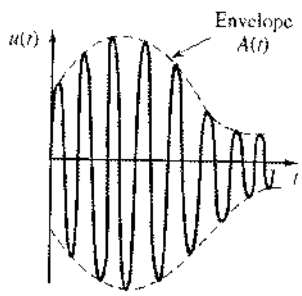

Figure 3.5 A conventional AM signal in the time domain.

The advantage of conventional AM is that the receiver is cheap and simple using an envelope detector, the disadvantage is that the power consumption and bandwidth is not efficient.

Q1.b Explain Why the Full AM is used in Radio communication systems. (5 Marks)

ANS: **The simplicity of the demodulator has made conventional DSB AM a practical choice for AM radio broadcasting. Since there are literally billions of radio receivers, an inexpensive implementation of the demodulator is extremely important. The power inefficiency of conventional AM is justified by the fact that there are few broadcast transmitters relative to the number of receivers. Consequently, it is cost effective to construct powerful transmitters and sacrifice power efficiency in order to simplify the signal demodulation at the receivers.**

Q2.a The modulating signal m(t)=4 $\cos^2(4000\pi t)$ is multiplied by the carrier c(t)=100 $\cos(2\pi f_c t)$ where fc =50 kHz. Determine and sketch spectrum of the DSB signal. (10 Marks)

ANS:

$$u(t) = m(t).c(t)$$
$$u(t) = 4\cos^2(4000\pi t).100\cos(2\pi f_c t)$$
$$u(t) = 2[1+\cos(8000\pi t)].100\cos(100000\pi t)$$
$$u(t) = 200[1+\cos(8000\pi t)]\cos(100000\pi t)$$
$$u(t) = 200[\cos(100000\pi t)+\cos(8000\pi t)\cos(100000\pi t)]$$
$$u(t) = 200\cos(100000\pi t)+100\cos(108000\pi t)+100\cos(92000\pi t)$$
$$U(f) = 100\delta(f-50000)+100\delta(f+50000)+50\delta(f-54000)+50\delta(f+54000) + 50\delta(f-46000)+50\delta(f+46000)$$

Q2.b Explain How to construct Full AM Modulation using Power-Law Modulation. (5 Marks)

Let us consider the use of a nonlinear device such as a P-N diode which has a voltage-current characteristic as shown in figure 3.15. Suppose that the voltage input to such a device is the sum of the message signal m(t) and the carrier $A_c \cos 2\pi f_c t$

As illustrated in figure, The nonlinearity will generate a product of the message m(t) with the carrier, plus additional terms. The desired modulated signal can be filtered out by passing the output of the nonlinear device through a bandpass filter.

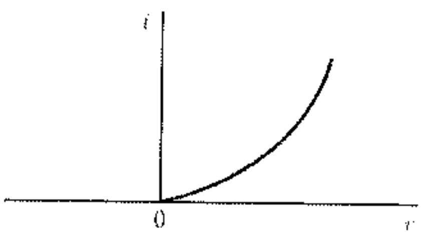

Figure 3.15 Voltage-current characteristic of P-N diode.

To elaborate on this method, suppose that the nonlinear device has an input–output (square-law) characteristic of the form

$$v_0(t) = a_1 v_i(t) + a_2 v_i^2(t) \tag{3.2.29}$$

where $v_i(t)$ is the input signal, $v_0(t)$ is the output signal, and the parameters (a_1, a_2) are constants. Then, if the input to the nonlinear device is

$$v_i(t) = m(t) + A_c \cos 2\pi f_c t, \tag{3.2.30}$$

its output is

$$\begin{aligned} v_o(t) &= a_1[m(t) + A_c \cos 2\pi f_c t] + a_2[m(t) + A_c \cos\cos 2\pi f_c t]^2 \\ &= a_1 m(t) + a_2 m^2(t) + a_2 A_c^2 \cos^2 2\pi f_c t + A_c a_1 \left[1 + \frac{2a_2}{a_1} m(t)\right] \cos 2\pi f_c t \end{aligned} \tag{3.2.31}$$

The output of the bandpass filter with bandwidth $2W$ centered at $f = f_c$ yields

$$u(t) = A_c a_1 \left[1 + \frac{2a_2}{a_1} m(t)\right] \cos 2\pi f_c t \qquad (3.2.32)$$

where $2a_2|m(t)|/a_1 < 1$ by design. Thus, the signal generated by this method is a conventional DSB AM signal.

Q3.a Draw and Explain how the Ring Modulator Works. **(5 Marks)**

ANS:

The switching of the diodes is controlled by a square wave of frequency fc, denoted as c(t), which is applied to the center taps of the two transformers. When c(t) > 0, the top and bottom diodes conduct, while the two diodes in the cross arms are off. In this case, the message signal m(t) is multiplied by+1. When c(t)<0, the diodes in the cross arms of the ring conduct, while the other two are switched off. In this case, the message signal m(t) is multiplied by−1. Consequently, the operation of the ring modulator may be described mathematically as:

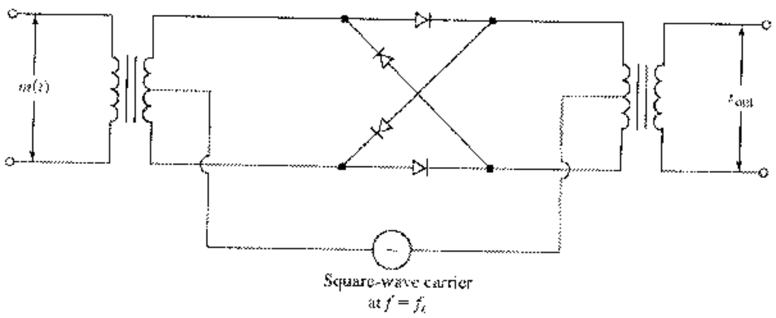

Figure 3.19 Ring modulator for generating DSB-SC AM signal.

a multiplier of $m(t)$ by the square-wave carrier $c(t)$; i.e.,

$$v_0(t) = m(t)c(t) \qquad (3.2.38)$$

as shown in Figure 3.19.

Since $c(t)$ is a periodic function, it is represented by the Fourier series

$$c(t) = \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c(2n-1)t] \qquad (3.2.39)$$

Hence, the desired DSB-SC AM signal u(t) is obtained by passing $v_o(t)$ through a bandpass filter with center frequency fc and bandwidth 2W.

Q3.b Draw the Envelop detector circuit and explain how to use it to demodulate a Conventional Amplitude Modulation signal. **(5 Marks)**

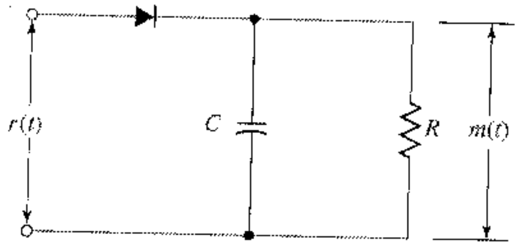

Figure 3.20 An envelope detector.

ANS: As previously indicated, conventional DSB AM signals are easily demodulated by means of an envelope detector. A circuit diagram for an envelope detector is shown in figure 3.20. It consists of a diode and an RC circuit, which is basically a simple lowpass filter. During the positive half-cycle of the input signal, the diode is conducting and the capacitor charges up to the peak value of the input signal.

When the input falls below the voltage on the capacitor, the diode becomes reverse-biased and the input becomes disconnected from the output. During this period, the capacitor discharges slowly through the load resistor R. On the next cycle of the carrier, the diode conducts again when the input signal exceeds the voltage across the capacitor. The capacitor charges up again to the peak value of the input signal and the process is repeated again.

Q4.a An angle-modulated signal has the form u(t)=100 cos[2πfct +4 sin 2πfmt] where fc =10 MHz and fm=1000 Hz. Assuming that this is an FM signal, determine the modulation index and the transmitted signal bandwidth. **(10 Marks)**

Problem 3.34

1) Assuming that $u(t)$ is an FM signal it can be written as

$$\begin{aligned} u(t) &= 100\cos(2\pi f_c t + 2\pi k_f \int_{-\infty}^{\infty} \alpha\cos(2\pi f_m \tau)d\tau) \\ &= 100\cos(2\pi f_c t + \frac{k_f \alpha}{f_m}\sin(2\pi f_m t)) \end{aligned}$$

Thus, the modulation index is $\beta_f = \frac{k_f \alpha}{f_m} = 4$ and the bandwidth of the transmitted signal

$$B_{FM} = 2(\beta_f + 1)f_m = 10 \text{ KHz}$$

3) If the signal $u(t)$ is PM modulated, then

$$\beta_p = \Delta\phi_{max} = \max[4\sin(2\pi f_m t)] = 4$$

The bandwidth of the modulated signal is

$$B_{PM} = 2(\beta_p + 1)f_m = 10 \text{ KHz}$$

Q4.b Draw and explain How to construct an angle modulator using a Voltage controlled oscillator VCO. (5 Marks)

ANS: One approach is to use a varactor diode. A varactor diode is a capacitor whose capacitance changes with the applied voltage. Therefore, if this capacitor is used in the tuned circuit of the oscillator and the message signal is applied to it, the frequency of the tuned circuit, and the oscillator, will change in accordance with the message signal.

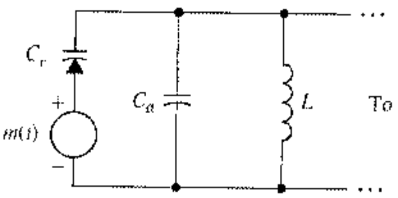

Figure 3.31 Varactor diode implementation of an angle modulator.

Q5.a Explain How to generate a narrow-band angle modulated signal and how to use it to construct an angle modulated signal. (10 Marks)

ANS: Another approach for generating an angle-modulated signal is to first generate a narrowband angle-modulated signal, and then change it to a wideband signal. This method is usually known as the indirect method for generation of FM and PM signals. Due to the similarity of conventional AM signals, generation of narrowband angle-modulated signals is straightforward. In fact any modulator for conventional am generation can be easily modified to generate a narrowband angle-modulated signal. Figure 3.32 shows the block diagram of a narrowband angle modulator.

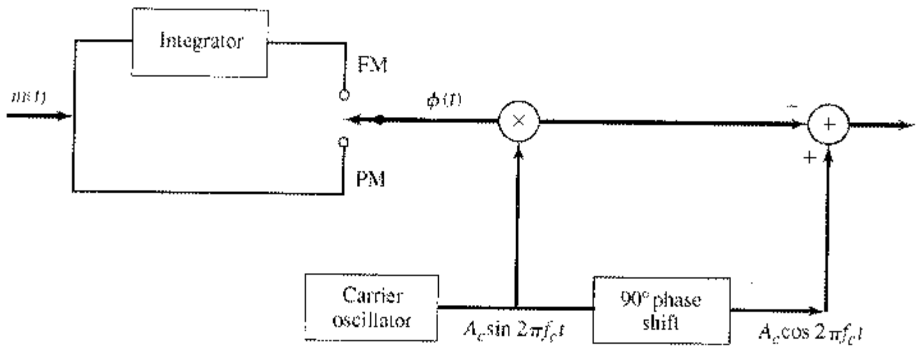

Figure 3.32 Generation of narrowband angle-modulated signal.

The next step is to use the narrowband angle-modulated signal to generate a wideband angle-modulated signal. Figure 3.33 shows the block diagram of a system that generates wideband angle modulated signals from narrowband angle-modulated signals. The first stage of such a system is, of course, a narrowband angle-modulator such as the one shown in figure 3.32. The narrowband angle-modulated signal enters a frequency multiplier that multiplies the instantaneous frequency of the input by some constant n. This is usually done by applying the input signal to a nonlinear element and then passing its output through a bandpass filter tuned to the desired central frequency.

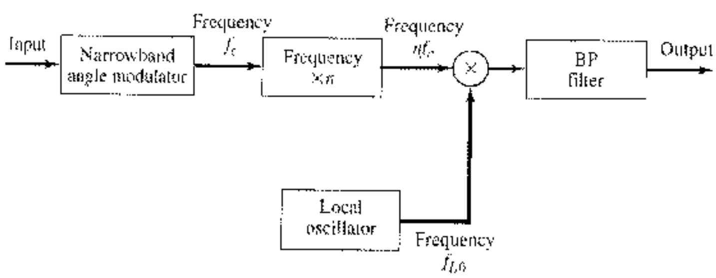

Figure 3.33 Indirect generation of angle-modulated signals.

Q5.b - The output signal from an AM modulator is u(t)=5 cos 1800πt +20 cos 2000πt +5 cos 2200πt.
1-Determine the modulating signal m(t) and the carrier c(t).
2-Determine the modulation index.
3-Determine the ratio of the power in the sidebands to the power in the carrier. (10 Marks)

1)

$$\begin{aligned} u(t) &= 5\cos(1800\pi t) + 20\cos(2000\pi t) + 5\cos(2200\pi t) \\ &= 20(1 + \frac{1}{2}\cos(200\pi t))\cos(2000\pi t) \end{aligned}$$

The modulating signal is $m(t) = \cos(2\pi 100t)$ whereas the carrier signal is $c(t) = 20\cos(2\pi 1000t)$.

2) Since $-1 \leq \cos(2\pi 100t) \leq 1$, we immediately have that the modulation index is $\alpha = \frac{1}{2}$.

3) The power of the carrier component is $P_{\text{carrier}} = \frac{400}{2} = 200$, whereas the power in the sidebands is $P_{\text{sidebands}} = \frac{400\alpha^2}{2} = 50$. Hence,

$$\frac{P_{\text{sidebands}}}{P_{\text{carrier}}} = \frac{50}{200} = \frac{1}{4}$$

Q6.a Draw and Explain the superheterodyne receiver used in AM radio broadcast. (5 Marks)

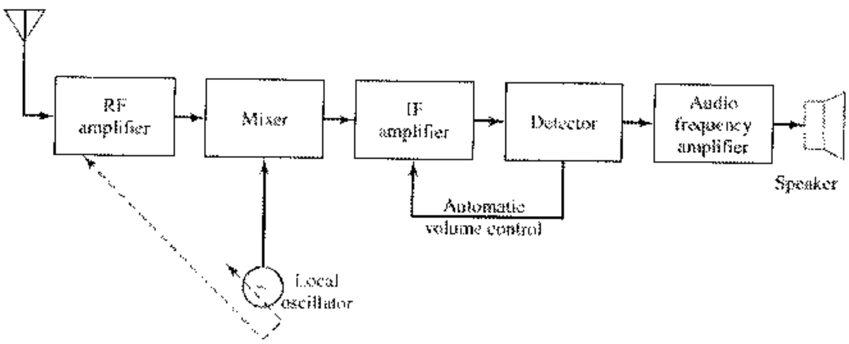

Figure 3.40 Superheterodyne AM receiver.

The receiver most commonly used in AM radio broadcast is the so called superheterodyne receiver shown in figure 3.40. It consists of a radio frequency (RF) tuned amplifier, a mixer, a local oscillator, an intermediate frequency (IF) amplifier, an envelope detector, an audio frequency amplifier, and a loudspeaker. Tuning for the desired radio frequency is provided by a variable capacitor, which simultaneously tunes the RF amplifier and the frequency of the local oscillator.

In the superheterodyne receiver, every AM radio signal is converted to a common IF frequency of fi F = 455 khz. This conversion allows the use of a single tuned IF amplifier for signals from any radio station in the frequency band. The IF amplifier is designed to have a bandwidth of 10 khz, which matches the bandwidth of the transmitted signal.

Q6.b Draw and Explain FM Stereo Broadcasting used in Radio communication. (10 Marks)

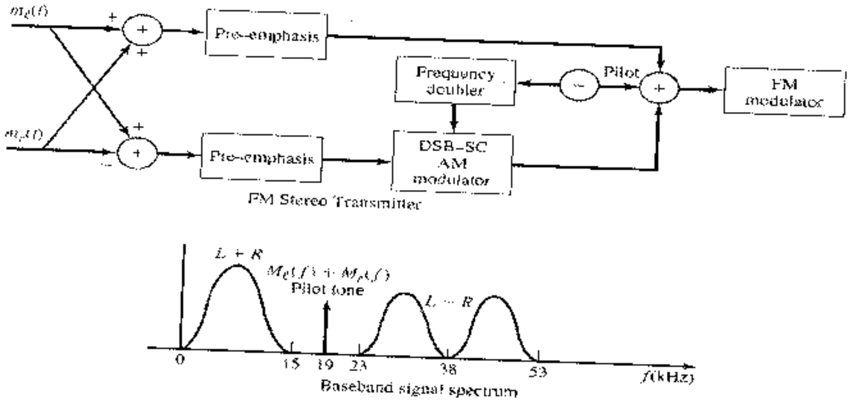

Figure 3.43 FM stereo transmitter and signal spacing.

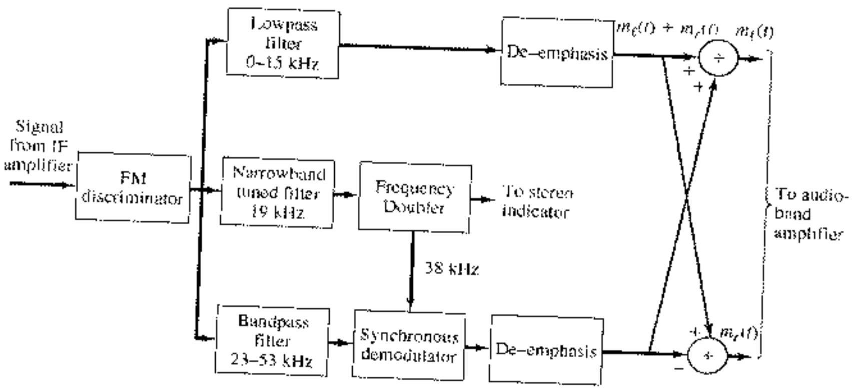

Figure 3.44 FM stereo receiver.

The signals from the left and right microphones, m(t) and mr (t), are added and subtracted as shown. The sum signal m(t)+mr (t) is left as is and occupies the frequency band 0–15 khz. The difference signal m(t) − mr (t) is used to AM modulate (DSB-SC) a 38-khz carrier that is generated from a 19-khz oscillator. A pilot tone at the frequency of 19 khz is added to the signal for the purpose of demodulating the DSB-SC AM signal. The reason for placing the pilot tone at 19 khz instead of 38 khz is that the pilot is more easily separated from the composite signal at the receiver. The combined signal is used to frequency modulate a carrier.

By configuring the baseband signal as an fdm signal, a monophonic fm receiver can recover the sum signal m(t)+mr (t) by use of a conventional fm demodulator. Hence, FM stereo broadcasting is compatible with conventional FM. The second requirement is that the resulting FM signal does not exceed the allocated 200-khz bandwidth.

The FM demodulator for FM stereo is basically the same as a conventional FM demodulator down to the limiter/discriminator. Thus, the received signal is converted to baseband. Following the discriminator, the baseband message signal is separated into the two signals m(t)+mr (t) and m(t)−mr (t) and passed through de-emphasis filters, as shown in figure 3.44. The difference signal is obtained from the DSB-SC signal by means of a synchronous demodulator using the pilot tone.

By taking the sum and difference of the two composite signals, we recover the two signals m(t) and mr (t). These audio signals are amplified by audio-band amplifiers and the two outputs drive dual loudspeakers. As indicated above, an FM receiver that is not configured to receive the FM stereo sees only the baseband signal m(t) + mr (t) in the frequency range 0–15 khz. Thus, it produces a monophonic output signal which consists of the sum of the signals at the two microphones.