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DEPARTMENT OF ELECTRONICS AND TELECOMMUNICATION

ENGINEERING

Dr. BABASAHEB AMBEDKAR TECHNOLOGICAL UNIVERSITY

Lonere-402 103, Tal. Mangaon, Dist. Raigad (MS)

INDIA

Dr. BabasahebAmbedkar Technological University,Lonere
Department of Electronics & Telecommunication Engineering

List of Experiment

Ex. No.	Experiment Name
1.	Basics of Communication System
2	Study of Spectrum Analyzer
3.	Study of Square Law Modulator
4.	Study of AM Generation: To Calculate Modulation Index of DSB Wave 1) By Observation 2) Trapezoidal Pattern.
5.	Study of AM transreceiver 1) Study of AM detection by using Diode Detector.
6.	Study of FM Generation by reactance modulator
7.	Matlab Code for Amplitude Modulation
8.	Matlab code for Frequency Modulation

Experiment 01

Aim : To study communication system

Theory :

Communication

The term communication refers to sending, receiving and processing of informing by electronic means. This information can be of different types such as sound, picture, music, computer data etc.

Electromagnetic wave

Electromagnetic radiation (EM radiation or EMR) is a form of energy emitted and absorbed by charged particles, which exhibits wave-like behaviour as it travels through space. EMR has both electric and magnetic field components, which stand in a fixed ratio of intensity to each other, and which oscillate in phase perpendicular to each other and perpendicular to the direction of energy and wave propagation. In a vacuum, electromagnetic radiation propagates at a characteristic speed, the light. The electromagnetic wave is a combination of electrical signal and magnetic signal. The magnetic signal is horizontally polarized and electric wave signal are vertically polarized.

Electromagnetic Frequency Spectrum

- The EM waves oscillate, they are sinusoidal and their frequency is measured in Hz.
- The frequency of EM signal can be very low or it can be extremely high. This entire range of frequencies is called as Electromagnetic spectrum.
- The electromagnetic spectrum consists of signals such as 50 Hz line frequency and voice signals at the lower end.
- The radio frequencies which are used for the two way communication reside at the centre of the EM spectrum. These frequencies are used for the applications such as radio or TV broadcasting as well.
- The infrared and visible lights are at the upper end of the EM spectrum.

Sr. No.	Name	Frequency	Wavelength
1.	Extremely low frequency(ELF)	30-300Hz	10^7 to 10^6m
2.	Voice Frequency(VF)	300-3000Hz	10^6 to 10^5m
3.	Very Low Frequencies(VLF)	3-30kHz	10^5 to 10^4m
4.	Low frequencies(LF)	30-300kHz	10^4 to 10^3m
5.	Medium frequency(HF)	300kHz-3MHz	10^3 to 10^2m
6.	High frequency(HF)	3-30 MHz	10^2 to 10m
7.	Very high frequency(VHF)	30-300MHz	10 to 1m
8.	Ultra high frequencies(UHF)	300MHz-3GHz	1 to 10^{-1}m
9.	Super high frequencies(SHF)	3-30GHz	10^{-1} to 10^{-2}m
10.	Extremely high frequencies(EHF)	30-300GHz	10^{-2} to 10^{-6}m
11.	Infrared		0.7 to 10µm
12.	Visible light		0.4µm to0.8µm

Basic block diagram of communication system consists of:

1. Information signal
2. Input transducer
3. Transmitter
4. Communication channel
5. Noise
6. Receiver
7. Output transducer

1) Information or input signal
    - The communication system exists to convey useful information from one place to the other.
    - The information can be in the form of a sound signal like speech or music, or it can be in the form of pictures (TV signals) or it can be data information coming from computer.
    - It is a message which is measured in bits or bytes.

2) Input transducer
    - The information is in the form of sound, picture or data signals cannot be transmitted as it is.
    - First it has to be converted into a suitable electrical signal. The input transducer block does this job.
    - The input transducers commonly used in communication systems are microphones, TV, cameras etc.

3) Transmitter
    - The function of the transmitter block is to convert the electrical equivalent of the information to a suitable form.
    - In addition, it increases the power level of the signal. The power level should be increased in order to cover a large range.
    - The transmitter consists of the electronic circuits such as amplifier, mixer, and oscillator and power amplifier.

4) Communication channel or medium

It is the medium used for transmission of electronic signal from one place to other. The communication medium can be conducting wires, cables, optical fibres or free space. Depending on type of communication medium, two types of systems will exist. They are:

a) Wire line communication:
   - Uses mediums like the simple wire or cables or optical fibres.
   - For example, telegraph and telephone systems, cables TV etc.
   - Due physical connections from one point or other, these systems can't be used for communications over long distances.

b) Wireless radio communications
   - Uses free space for communication
   - Doesn't need wires for sending the messages from one place to others
   - For ex: T.V., radio, satellite communication which transmits the signal using transmitting antenna in the free space.
   - The transmitted signal is in the form of electromagnetic waves. A receiving antenna will pick up this signal and feed it to receiver
   - Radio comm. can be used for long distance comm. Such as from one country to other or from one planet to other.

c) Noise
   - Noise is an unwanted electrical signal which gets added to the transmitted signal when it is travelling towards the receiver.
   - Due to noise, the quality of information will degrade, once added, the noise can't be separated out from the information.
   - Hence the noise is a big problem in communication systems.
   - Noise can be either natural or manmade. The radiation from the sun and stars etc.
   - The man made noise are noises from electrical ignition systems of the automobiles, welding machines, electric motors etc.
   - Even though noise can't be completely eliminated, its effect can be reduced by using various techniques.

d) Receiver
   - The reception is exactly the opposite process of transmission. The received signal is amplified, demodulated and converted into a suitable form.
   - The receiver consists of all the electronic circuits like mixer, oscillators and detectors, amplifiers etc.

e) Output transducers
   - The output transducers converts electrical signal at the output of receiver back to the original form i.e. sound and TV, pictures etc.

Conclusion :

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Instructions for student

1. Use one side blank paper where ever required. Draw block/circuit diagram on it.

2. Draw basic block/elements of communication system. Block diagram represent in Principle of Communication by Simon Haykin Book. Refer page number 2 and figure 1.1

3. Answer the following questions and it is your conclusion
    a. What is a communication system?
    b. Communication is transmission of information from one point to another. How communication process takes place? Write it in hierarchical order. (refer page no. 1 of Principle of Communication by Simon Haykin)

4. MATLAB Assignment

    Generate 1V, 1 KHz of sinusoidal signal using Mat Lab. Plot the same, i.e. time & amplitude axis must satisfy the given specifications.

    (This assignment will clarify sampling process of analog signal.)

Experiment No. 2

Aim: Understand the basic controls and measurement techniques of HEMAG spectrum analyzer HM5014-2

Equipment:
- HAMEG 1GHz Spectrum Analyzer HM5014-2
- 3MHz Scientec Function Generator

Theory :

Periodic Signal

All periodic signals are made up of two or more exponential signals (harmonics) of differing amplitude and frequency. One is able to see the shape of these signals with the aid of an oscilloscope, which graphs the signals voltage over an interval of time. We can observe the amplitude and the frequency of the different harmonics that make up the signal with the use of a Spectrum Analyzer. The type of graph that the spectrum analyzer produces is called an Amplitude Spectrum. The amplitude spectrum is comprised of vertical spikes that represent harmonics. These spikes begin at various places along the horizontal (frequency) axis and rise in the positive direction to various heights (amplitudes).

What is a Spectrum Analyzer?

Traditionally oscilloscopes have been used to see how electrical signals vary with time. Some oscilloscopes can perform vector signal analysis, and signal analyzers now have significant amounts of time-domain measurement capability. Oscilloscopes are optimized for time-domain measurements, and signal analyzers are the tool of choice for frequency-domain measurements. However, oscilloscope doesn’t provide the full analysis of signal.

To fully understand the performance of a device/system, a signal (or signals) must also be analyzed in the frequency domain. This is exactly what the spectrum analyzer does. n the frequency domain, complex signals (e.g., comprising more than one frequency) are separated into their frequency components, and the level at each frequency is displayed. Frequency-domain measurements have several distinct advantages. For one, information not discernible on an oscilloscope becomes readily apparent on a spectrum analyzer.

Spectrum analyzers are of several types. We use the heterodyne or scanning analyzer. A scanning spectrum analyzer is essentially a radio receiver. Imagine tuning a conventional FM broadcast receiver from one end to the other, plotting the reading of the tuning meter versus frequency. The graph produce is called a Frequency Domain representation, or Spectrum; it tells you at what frequencies signals occur and how strong they are.

Spectrum Analysis Fundamentals

Measuring signals with a spectrum analyzer also greatly reduces the amount of noise present in the measurement due to the analyzer's ability to narrow the measurement bandwidth. Moreover, many of today's systems are inherently frequency-domain oriented and must be analyzed in the frequency domain to ensure there's no interference from neighboring frequencies.

With a frequency-domain view of the spectrum, it is easy to measure a signal's frequency, power, harmonic content, modulation, spurs, and noise. With these quantities measured, total harmonic distortion, occupied bandwidth, signal stability, output power, intermediation distortion, power bandwidth, carrier-to-noise ratio, and a host of other measurements then can be determined using just a spectrum analyzer.

Frequency-domain measurements (spectrum analysis) are made with either a fast-Fourier transform (FFT) analyzer or a swept-tuned receiver. The FFT analyzer takes a time-domain signal, digitizes it using digital sampling, and then applies the mathematics required to convert it to the frequency domain. The result is displayed as a spectrum. With its real-time signal analysis capability, the analyzer can capture periodic, random, and transient events and can measure phase and magnitude.

The swept-tuned analyzer "sweeps" across the frequency range of interest, displaying all the frequency components present. This enables measurements to be made over a large dynamic range and wide frequency range. The swept-tune analyzer is the most widely accepted, general-purpose tool for frequency-domain measurements.

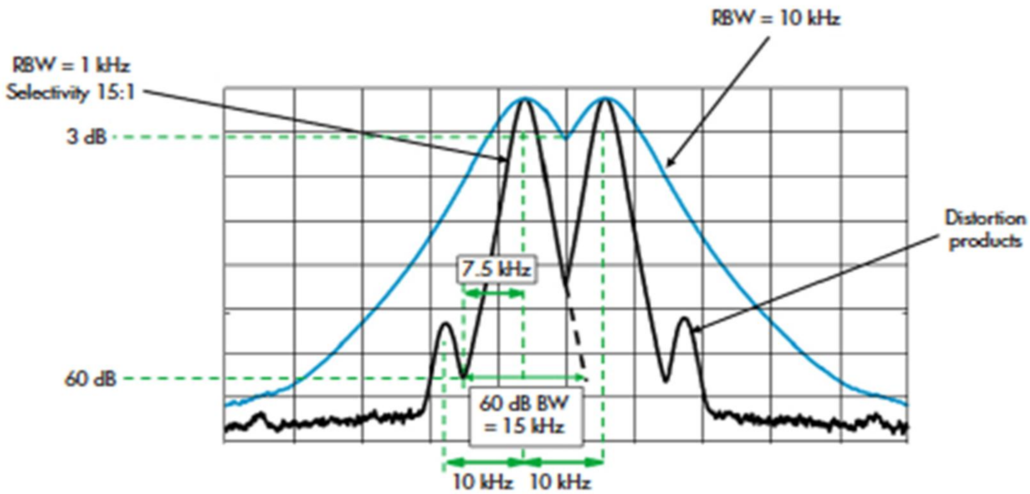
Both the FFT and swept-tuned analyzer can be used for a wide range of measurements (such as frequency, power modulation, distortion, and noise measurements) in applications as varied as spectrum monitoring, spurious emissions, scalar network analysis, and electromagnetic interference.

Measurements are made in the –172-dBm to +30-dBm range, over frequencies ranging from 3 Hz to greater than 325 Hz.

Understanding Analyzer Specifications

It is important to understand the spectrum analyzer's specifications as they communicate the level of performance expected from a particular instrument. Specifications are helpful in predicting how an analyzer will perform in a specific measurement situation, as well as the accuracy of its results. Key spectrum analyzer specifications include:

- <u>Frequency range:</u> Frequency range specifies the range of frequencies over which the analyzer will operate. It is important to find a spectrum analyzer that will cover the fundamental frequencies of your application, but also harmonics or spurious signals on the high end, or baseband, and IF on the low end.
- <u>Frequency accuracy:</u> Frequency accuracy is often labeled as "frequency readout accuracy" and is usually specified as the sum of several sources of errors, including frequency-reference uncertainty, span error, and RBW center-frequency error. Virtually every modern spectrum analyzer uses synthesized local oscillators, which are generally accurate to within a few hundred Hertz.
- <u>Amplitude accuracy:</u> Uncertainty in display fidelity and frequency response has a direct effect on an analyzer's amplitude accuracy. The further a signal is from the reference level, the more the display fidelity will play a factor. To reduce the uncertainty, we must bring the measured signal up to the reference level, making the displayed amplitude of the signal the same as the calibrator. However, this introduces a new error in the measurement, namely IF-gain uncertainty. There are two options for handling this. In one case, we leave the signal in its place on screen, and the display fidelity error is simply accepted. In the other, we move the signal to the reference level, which causes a reference-level switching error. The best option depends on the spectrum analyzer. Some spectrum analyzers have larger display fidelity errors, while others have larger IF-gain uncertainties.
- <u>Resolution:</u> A spectrum analyzer's resolution determines its ability to resolve signals of equal amplitude. A wider RBW may make two signals appear as one. In general, two equal-amplitude signals can be resolved if their separation is greater than or equal to the 3-dB bandwidth of the selected RBW filter. Generally, though, engineers look at signals of unequal amplitudes. Because both signals trace out the filter shape, it is possible to bury a smaller signal under the filter skirt of the larger one. The greater the amplitude difference, the more a lower signal is buried under the skirt of its neighbor's response.
  Examples of a two-tone test, signals are separated by 10 kHz shown in figure given below. With a 10-kHz RBW, resolution of equal-amplitude tones is not a problem, but it may bury distortion products. This is the case for a 3-kHz RBW with a selectivity of 15:1. Here, the required RBW for the measurement is 1 kHz and the two signals unequal in amplitude by 60 dB must be separated by at least one-half the 60-dB bandwidth to resolve the smaller signal.



RBW selectivity (shape factor) and phase noise are important characteristics for determining the resolvability of signals of unequal amplitudes. One way to improve the spectrum analyzer's resolution is to narrow the RBW, but it then takes longer to sweep across the spectrum because the RBW filters require a finite time to fully respond. Spectrum analyzers have auto-coupled sweep time that automatically chooses the fastest allowable sweep time based on selected span, RBW, and VBW. The analyzer type also affects sweep speed.

- Sensitivity: A receiver's sensitivity is an indication of how well it measures small signals. All receivers (including those within spectrum analyzers) add some internally generated noise. Spectrum analyzers usually characterize this by specifying the displayed average noise level (DANL) in dBm, with the smallest RBW setting. DANL is another term for the instrument's noise floor given a particular bandwidth and represents its best-case sensitivity. It is impossible to measure input signals below this noise level. Generally, sensitivity ranges from –135 dBm to –165 dBm. To achieve optimum sensitivity, try using the narrowest RBW possible, sufficient averaging, a minimum RF-input attenuation, and/or a preamplifier, although increasing sensitivity may conflict with other measurement requirements, such as minimizing distortion or maximizing dynamic range.
- Distortion: Although distortion measurements such as third-order intermodulation and harmonic distortion are common measurements for characterizing devices, the spectrum analyzer itself also produces distortion products that may disturb measurements. If this internal distortion is comparable to the external distortion of the device-under-test (DUT) being measured, measurement errors can arise. In the worst-case scenario, it can completely cover up a device's distortion products. The manufacturer may specify the analyzer's distortion performance directly, or it may be lumped into a dynamic-range specification.

Conclusion:

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Instruction to Student

1. Attached photo copy of following pages of HEMAG manual.
2. Draw the spectrum of sinusoidal signal of 1MHz, 1Vp-p, along with its 2^{nd}, 3^{rd}, and 4^{th} harmonic of the input signal.
3. Draw the spectrum of Pulse signal of same specification

NOTE: Specify amplitude/power of the input signal and its corresponding frequency.

Answer the following question as a conclusion of the experiment.

4. What is the significance of two folded spectrum of sinusoidal signal i.e. significance of negative frequency in spectrum analysis.

Experiment No. 3

Aim: Generate amplitude modulated signal using diode as a square law device.

Instrument:

- Spectrum Analyzer
- Diode 1N4007
- MFR resistor, 10KΩ
- CFR resistor, 1KΩ
- Potentiometer, 47KΩ
- Analog voltmeter , 0-1v
- CRO Aplab,3MHZ Bandwidth
- 1 GHz Function Generator

Theory:

Methods of amplitude modulation can be put in the two categories

1. Linear modulation methods and

2. Square law modulation methods.

Linear modulation method utilizes the linear region of the current voltage characteristics of the amplifying device that is transistor or electron tube. Square law modulation circuits make use of non linear current voltage characteristics of diodes or triodes and are in general suited for use at low voltages. Important square law modulation methods are square law diode modulation and balanced modulator.

The given figure depicts the simplest form of AM modulator. Ignoring the higher order terms, the input-output characteristics of the diode-load resistor combination in this figure are given by:

$$v_2(t) = a_0 v_1(t) + a_1 {v_1}^2(t)$$

Where,

$v_1(t)$: Input signal, to the square law device and

$v_2(t)$: Output signal developed across the load resistor of 1KΩ.

Consider resistor, a_0, a_1 are constants.
Where,

$$v_1(t) = V_c \cos(2\pi f_c t) + m(t)$$

Instruction for Student

a) Draw block diagram for AM generation using square law device and draw practical circuit diagram.
b) Derive the equation for $v_2(t)$ before BPF.
c) Determine the spectral content of the output signal $V_2(f)$.
d) To extract the desired AM wave from $v_2(t)$: , we need a band-passfilter. If we were to insert this filter in the figure, determine the cut off frequency and bandwidth of the required filter, assuming that the message signal is limited to the band: $-w_m < f_m < w_m$
e) To avoid spectral distortion by the presence of undesired demodulation products in $v_2(t)$, the condition $w_m < f_m < 3w_m$ must be satisfied validate this condition.

Answer following question as a conclusion

1) Where we select the operating point of diode so that can operate as a square law device?

Experiment No. 4

Aim: Study of Double Sideband AMTransmitter.

Apparatus:AM Generator Kit,CRO, Connecting Probes etc.

Procedure:

This experiment investigates the generation of double sideband amplitude modulated (AM) waveforms, using the ST2201 module. By removing the carrier from such an AM waveforms, the generation of double sideband suppressed carrier (DSBSC) AM is also investigated. To avoid unnecessary loading of monitored signals, X10 oscilloscope probes should be used throughout this experiment.

1. Ensure that the following initial conditions exist on the board.

    a. Audio input select switch should be in INT position:

    b. Mode switch in DSB position.

    c. Output amplifier's gain potentiometer in full clockwise position.

    d. Speakers switch in OFF position.

2. Turn on power to the ST2201 board.

3. Turn the audio oscillator block's amplitude pot to its full clockwise (MAX) position, and examine the block's output (TP14) on an oscilloscope. This is the audio frequency sine wave which will be as our modulating signal

4. Turn the balance pot, in the balanced modulator & band pass filter circuit 1 block to its fully clockwise position. It is this block that we will use to perform double-side band amplitude modulation.

5. Monitor, in turn, the two inputs to the balanced modulator & band pass filter circuits 1 block, at TP1 and TP9. Note that:

    a. The signal at TP1 is the audio-frequency sine wave from the audio oscillator block. This is the modulating input to our double-sideband modulator.

    b. Test Point 9 carries a sine wave of 1MHz frequency and amplitude 120mVpp approx. This is the carrier input to our double-sideband modulator.

6. Next, examine the output of the balanced modulator & band pass filter circuit 1 block (at tp3), together with the modulating signal at TP1 Trigger the oscilloscope on the TP1 signal. Check that the waveforms as shown in Figure 1

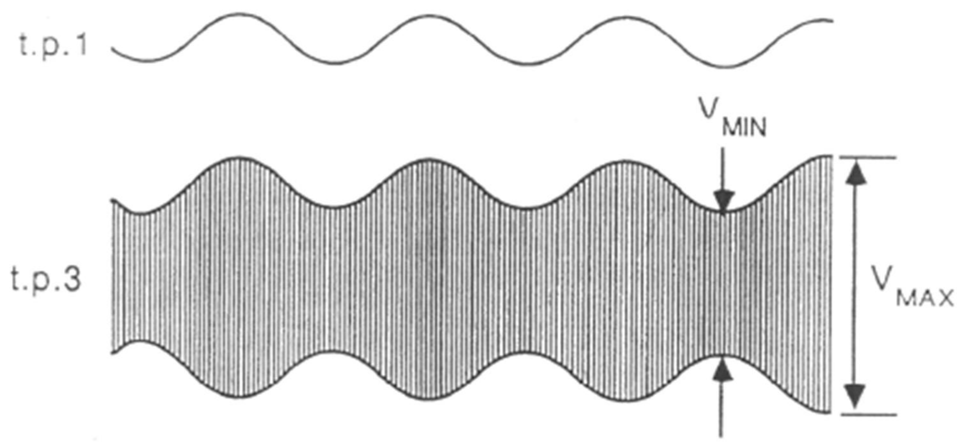


Figure 1

The output from the balanced modulator & band pass filter circuit 1 block (at TP3) is a double-sideband. AM waveform, which has been formed by amplitude-modulating the 1MHz carrier sine wave with the audio-frequency sine wave from the audio oscillator.

The frequency spectrum of this AM waveform is as shown below in Figure 2 whereFm is the frequency of the audio modulating signal.

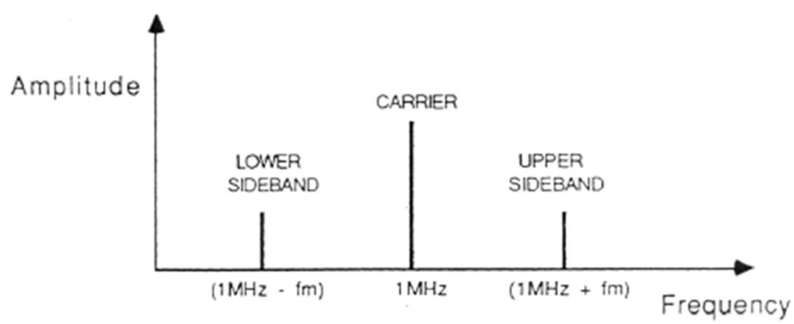


Figure 2

7. To determine the depth of modulation, measure the maximum amplitude (Vmax) and the minimum amplitude (Vmin) of the AM waveform at TP3, and use the following formula:

$$\text{Percentage Modulation} = \frac{\text{Vmax} - \text{Vmin}}{\text{Vmax} + \text{Vmin}}$$

whereVmax and Vmin are the maximum and minimum amplitudes shown in Figure 1.

8. Now vary the amplitude and frequency of the audio-frequency sine wave, by adjusting the amplitude and frequency present in the audio oscillator block.

Note the effect that varying each pot has on the amplitude modulated waveform. The amplitude and frequency amplitudes of the two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero. Do this by turning the amplitude pot to its MIN position, and note that the signal at TP3 becomes an un-modulated sine wave of frequency 1 MHz, indicating that only the carrier component now remains. Return the amplitude pot to its maximumposition. Now turn the balance pot in the balanced modulator & band pass filter circuit 1 block, until the signal at TP3 is as shown in Figure 3

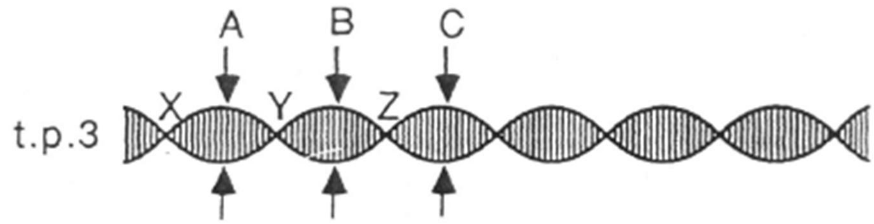


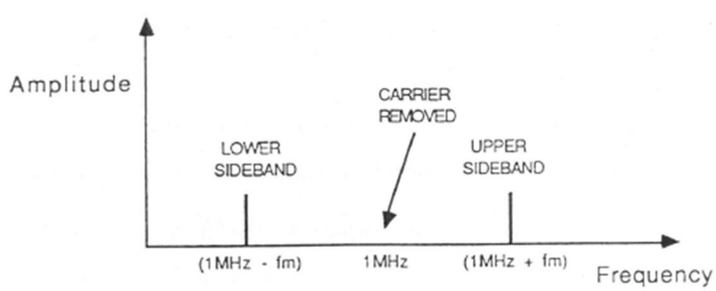
Figure 3

The balance pot varies the amount of the 1 MHz carrier component, which is passed from the modulator's output.

By adjusting the pot until the peaks of the waveform (A, B, C and so on) have the same amplitude, we are removing the carrier component altogether.

We say that the carrier has been 'balanced out' (or 'suppressed') to leave only the two sidebands.

Note that once the carrier has been balanced out, the amplitude of TP3's waveform should be zero at minimum points X, Y; Z etc. If this is not the case, it is because one of the two sidebands is being amplified more than the other. To remove this problem, the band pass filter in the balanced modulator & band pass filter circuit 1 block must be adjusted so that it passes both sidebands equally. This is achieved by carefully trimming transformer T1, until the waveform's amplitude is as close to zero as possible at the minimum points. The waveform at TP3 is known as a double-side suppressed carrier (DSBSC) waveform, and its frequency spectrum is as shown in Figure 4.



Frequency Spectrum of DSBSC Wave Form

9. Change the amplitude and frequency of the modulating audio signal (by adjusting the audio oscillator block's amplitude and frequency pots), and note the effect that these changes on the DSBSC waveform. The amplitudes of the two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero. Do these by turning the amplitude present to its MIN position, and note that the monitored signal becomes a D C level, indicating that there .are now no frequency components present. Return the amplitude pot to its MAX position.

10. Examine the output from the output amplifier block (TP13), together with the audio modulating signal (at TP1), triggering the scope with the audio modulating signal. Note that the DSBSC waveform appears, amplified slightly at TP13, as we will see later, it is the output amplifier's output signal which will be transmitted to the receiver.

11. Perform the experiment number 1 up to step number 6

12. Now apply the modulated waveform to the Y input of the Oscilloscope and the modulating signal to the X input.

13. Press the XY switch, you will observe the waveform similar to the one given

below:

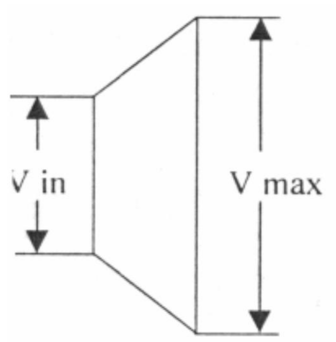


Figure 4

Calculate the modulation index by substituting in the formula

$$\text{Percentage Modulation} = \frac{V\text{max} - V\text{min}}{V\text{max} + V\text{min}}$$

Observation table:

Sr. No	Vmin(V)	Vmax(V)	Vc (V)	Vm(V)	Modulation Index (μ)	Modulation Index(m)

Calculation:

$$m = \frac{Vmax - Vmin}{Vmax + Vmin}$$

$$\mu = \frac{Vc - Vm}{Vc + Vm}$$

EXPERIMENT NO. 5

Aim: Study of Double Sideband AM Receiver.

Apparatus: AM Generator Kit,CRO,Connecting Probes,etc.

Procedure:

This experiment investigates the reception and demodulation of AM waveforms by the ST2201/ ST2202 module. Both AM broadcast signals, and AM transmissions from ST2201, will be examined, and the operation of automatic gain control at the receiver will be investigated.

To avoid unnecessary loading of monitored signals, X10 oscilloscope probes should be used throughout this experiment.

1. Position the ST2201 & ST2202 modules, with the ST2201 board on the left, and a gap of about three inches between them.

2. Ensure that the following initial conditions exist on the ST2201 board.

    a. Audio oscillator's amplitude pot in fully clockwise position.

    b. Audio input select switch in INT position.

    c. Balance pot in balanced modulator & band pass filter circuit 1 block, in full clockwise position;

    d. Mode switch in DSB position.

    e. Output amplifier's gain pot in full counter-clockwise position.

    f. TX output select switch in ANT position:

    g. Audio amplifier’s volume pot in fully counter-clockwise position.

    h. Speaker switch in ON position.

    i. On-board antenna in vertical position, and fully extended.

3. Ensure that the following initial conditions exist on the ST2102 board:

    a. RX input select switch in ANT position.

    b. R.F. amplifier's tuned circuit select switch in INT position.

    c. R.F. amplifier's gain pot in fully clock-wise position;

d. AGC switches in INT position.

e. Detector switches in diode position.

f. Audio amplifier's volume pot in fully counter-clockwise position.

g. Speaker switches in ON position.

h. Beat frequency oscillator switch in OFF position.

i. On-board antenna in vertical position, and fully extended.

4. Turn on power to the modules.

5. On the ST2202 module, slowly turn the audio amplifier's volume pot clockwise, until sounds can be heard from the on-board loudspeaker. Next, turn the vernier tuning dial until a broad cast station can be heard clearly, and adjust the volume control to a comfortable level.

Note: If desired, headphones (supplied with the module) may be used instead of the on-board loudspeaker. To use the headphones, simply plug the headphone jack into the audio amplifier block's headphones socket, and adjust controlled block's volume pot.

6. The first stage or 'front end' of the ST2202 AM receiver is the R.F amplifier stage. This is a wide -bandwidth tuned amplifier stage, which is tuned into the wanted station by means of the tuning dial. Once it has been tuned into the wanted station, the R.F. amplifier, having little selectivity, will not only amplify, but also those frequencies that are close to the wanted frequency. As we will see later, these nearby frequencies will be removed by subsequent stages of the receiver, to leave only the wanted signal.

Examine the envelope of the signal at the R.F. amplifier's output (at TP12), with an arc. -coupled oscilloscope channel.

7. The next stage of the receiver is the mixer stage, which mixes the R.F. amplifier's output with the output of a local oscillator. The Frequency of the local oscillator is also tuned by means of the tuning dial, and is arranged so that its frequency is always 455 KHz above the signal frequency that the R.F. amplifier is tuned to. This fixed frequency difference is always present, irrespective of the position of the tuning dial, and is arranged so that its frequency is always 455 KHz above the signal frequency that the R.F. amplifier is tuned to. This fixed frequency difference is always present, irrespective of the position of the tuning dial, and is known as the intermediate frequency (IF for short). This frequency relationship is shown below, for some arbitrary position of the tuning dial.

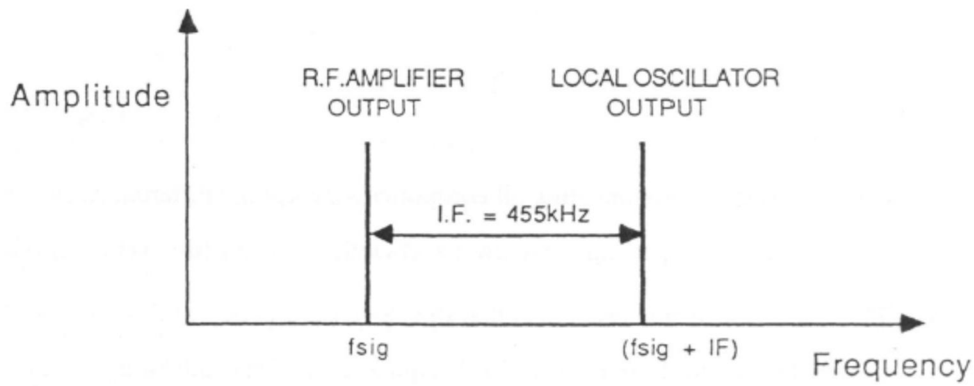


Figure 1

Examine the output of the local oscillator block, and check that its frequency varies as the tuning dial is turned. Re-time the receiver to a radio station.

8. The operation of the mixer stage is basically to shift the wanted signal down to the IF frequency, irrespective of the position of the tuning dial. This is achieved in two stages.

a. By mixing the local oscillator's output sine wave with the output from the R.F. amplifier block. This produces three frequency components:

The local oscillator frequency = (f sig + IF)

The sum of the original two frequencies, f sum = (2 f sig + IF)

The difference between the original two frequencies,

f diff = (f sig + IF -f sig) = IF

These there frequency components are shown in Figure 2.

b. By strongly attenuating all components. Except the difference frequency, IF this is done by putting a narrow-bandwidth band pass filter on the mixer's output.

The end result of this process is that the carrier frequency of the selected AM station is shifted down to 455 KHz (the IF Frequency), and the sidebands of the AM signal are now either side of 455 KHz.

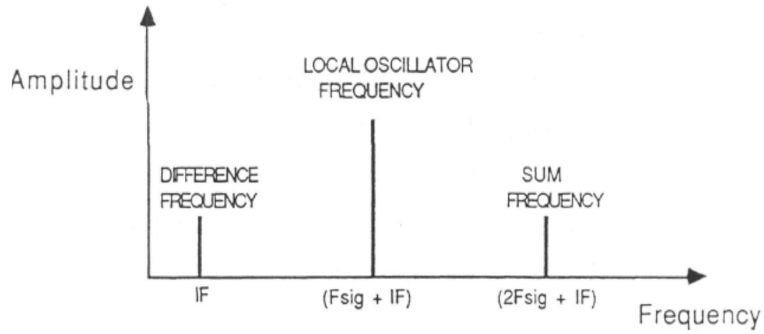


Figure 2

9. Note that, since the mixer's band pass filter is not highly selective, it will not completely remove the local oscillators and sum frequency components from the mixer's output. this is the case particularly with the local oscillator component, which is much larger in amplitude than the sum and difference components. Examine the output of the mixer block (TP20) with an a.c. coupled oscilloscope channel.

10. Tune in to a strong broadcast station again and note that the monitored signal shows little, if any, sign of modulation. This is because the wanted component, which is now at the IF frequency of 455 KHz, is still very small in component, which is now at the IF frequency of 455 KHz, is still very small in comparison to the local oscillator component.

This selective amplification is achieved by using two IF amplifier stages, IF amplifier 1 and IF amplifier 2, which are designed to amplify strongly a narrow band of frequencies around 455 KHz, without amplifying frequencies on either side of this narrow band.

These IF amplifiers are basically tuned amplifiers which have been pre-tuned to the IF frequency-they have a bandwidth just wide enough to amplify the 455 KHz carrier and the AM sidebands either side of it. Any frequencies outside this narrow frequency band will not be amplified. Examine the output of IF amplifier 1 (at. TP24) with an a.c.-coupled oscilloscope channel, and note that :

a. The overall amplitude of the signal is much larger than the signal amplitude at the mixer's output, indicating that voltage amplification has occurred.

b. The dominant component of the signal is now at 455 KHz, irrespective of any particular station you have tuned into. This implies that the wanted signal, at the IF frequency, has been amplified to a level where it dominates over the unwanted components.

c. The envelope of the signal is modulated in amplitude, according to the sound information being transmitted by the station you have tuned into.

11. Examine the output of IF amplifier 2 (TP28) with an a.c.-coupled oscilloscope channel, noting that the amplitude of the signal has been further amplified by this second IF amplitude of the signal has been further amplified by this second IF amplifier stage.

IF amplifier 2 has once again preferentially amplified signals around the IF frequency (455 KHz), so that:

a.The unwanted local oscillator and sum components from the mixer are now so small in comparison, that they can be ignored totally,

b. Frequencies close to the I F frequency, which are due to stations close to the wanted station, are also strongly attenuated. The resulting signal at the output of IF amplifier 2 (TP28) is therefore composed almost entirely of a 455 KHz carrier, and the A.M. sidebands either side of it carrying the wanted audio information.

12. The next step is extract this audio information from the amplitude variations of the signal at the output of IF amplifier 2. This operation is performed by the diode detector block, whose output follows the changes in the amplitude of the signal at its input.

To see how this works, examine the output of the diode detector block (TP31), together with the output from. IF amplifier 2 (at tp28). Note that the signal at the diode detector's output:

• Follows the amplitude variations of the incoming signal as required:

• Contains some ripple at the IF frequency of 455 KHz, and

• The signal has a positive DC offset, equal to half the average peak to peak amplitude of the incoming signal. We will see how we make use of this offset later on, when we look at automatic gain control (AGC).

13. The final stage of the receiver is the audio amplifier block contains a simple low-pass filter which passes only audio frequencies, and removes the high-frequency ripple from the diode detector's output signal. This filtered audio signal is applied to the input of an audio power amplifier, which drives on board loudspeaker (and the headphones, if these are used). The final result is the sound you are listening to!

The audio signal which drives the loudspeaker can be monitored at TP39 (providing that the audio amplifier block's volume pot is not in its minimum volume position). Compare this signal with that at the diode detector's output (TP31), and note how the audio amplifier block's low pass filter has 'cleaned up' the audio signal.

You may notice that the output from the audio amplifier block (tp39) is inverted with respect to the signal at the output of the diode detector (TP31) this inversion is performed by the audio power amplifier IC, and in no way affects the sound produced by the receiver.

14. Now that we have examined the basic principles of operation of the ST2202 receiver for the reception and demodulation of AM broadcast signals, we will try receiving the AM signal from the ST2201 transmitter.

Presently, the gain of ST2201's output amplifier block is zero, so that there is no output from the Transmitter. Now turn the gain pot in ST2201's output amplifier block to its fully clockwise (maximum gain) position, so that the transmitter generates an AM signal.

On the ST2201 module, examine the transmitter's output signal (TP13), together with the audio modulating signal (TP1), triggering the 'scope with the signal'.

Since ST2201 TX output select switch is in the ANT position, the AM signal at tp13 is fed to the transmitter's antenna. Prove this by touching ST2201's antenna, and nothing that the loading caused by your hand reduces the amplitude of the AM waveform. at TP13.

The antenna will propagate this AM signal over a maximum distance of about 1.4 feet. We will now attempt to receive the propagated AM waveform with the ST2201/ ST2202 board, by using the receiver's on board antenna.

15. On the ST2201 module, turn the volume pot (in the audio amplifier block) clockwise, until you can hear the tone of the audio oscillator’s output signal, from the loudspeaker on the board.

16. On the ST2201/ST2202 receiver, adjust the volume pot so that the receiver's output can be clearly heard. Then adjust the receiver's tuning dial until the tone generated at the transmitter is also clearly audible at the receiver (this should be when the tuning dial is set to about 55-65 and adjust the receiver's volume pot until the tone is at a comfortable level.

Check that you are tuned into the transmitter's output signal, by varying ST2201's frequency pot in the audio oscillator block, and nothing that the tone generated by the receiver changes.

The ST2201/2202 receiver is now tuned into AM signal generated by the ST2201 transmitter. Briefly check that the waveforms, at the outputs of the following receiver blocks, are as expected:

R. F. Amplifier (TP12)	Mixer (TP20)	I.F. Amplifier 1 (TP24)
I.F. Amplifier 2 (TP28)	Diode Detector (TP31)	Audio Amplifier (TP39)

Conclusion:

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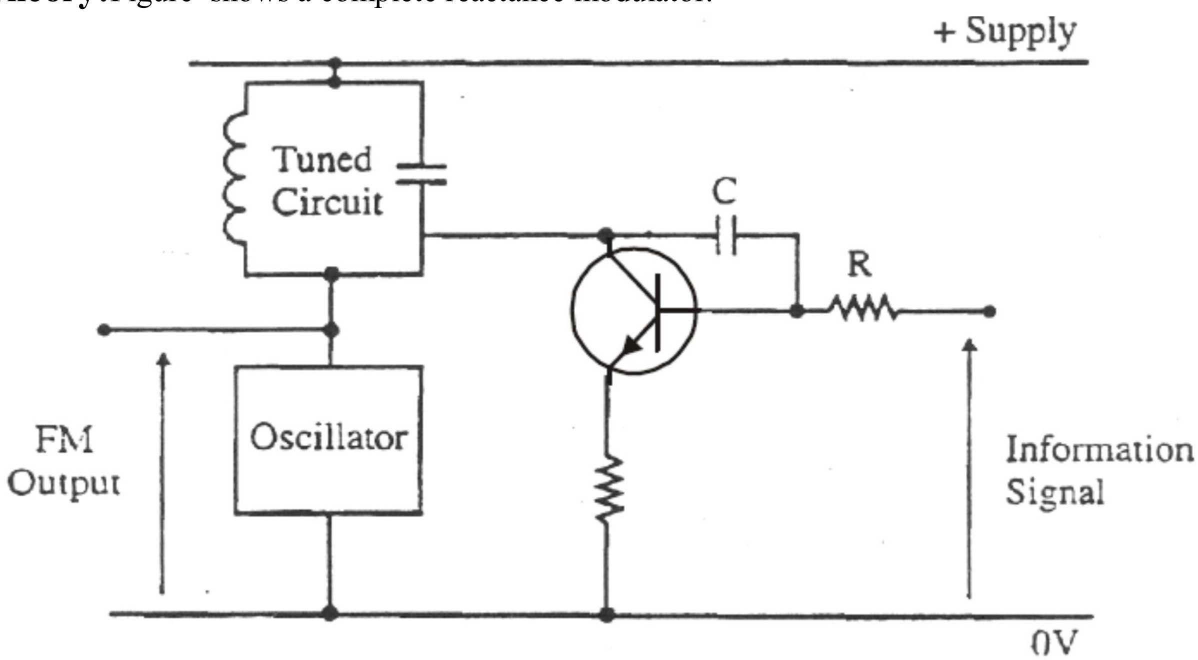
Experiment 6

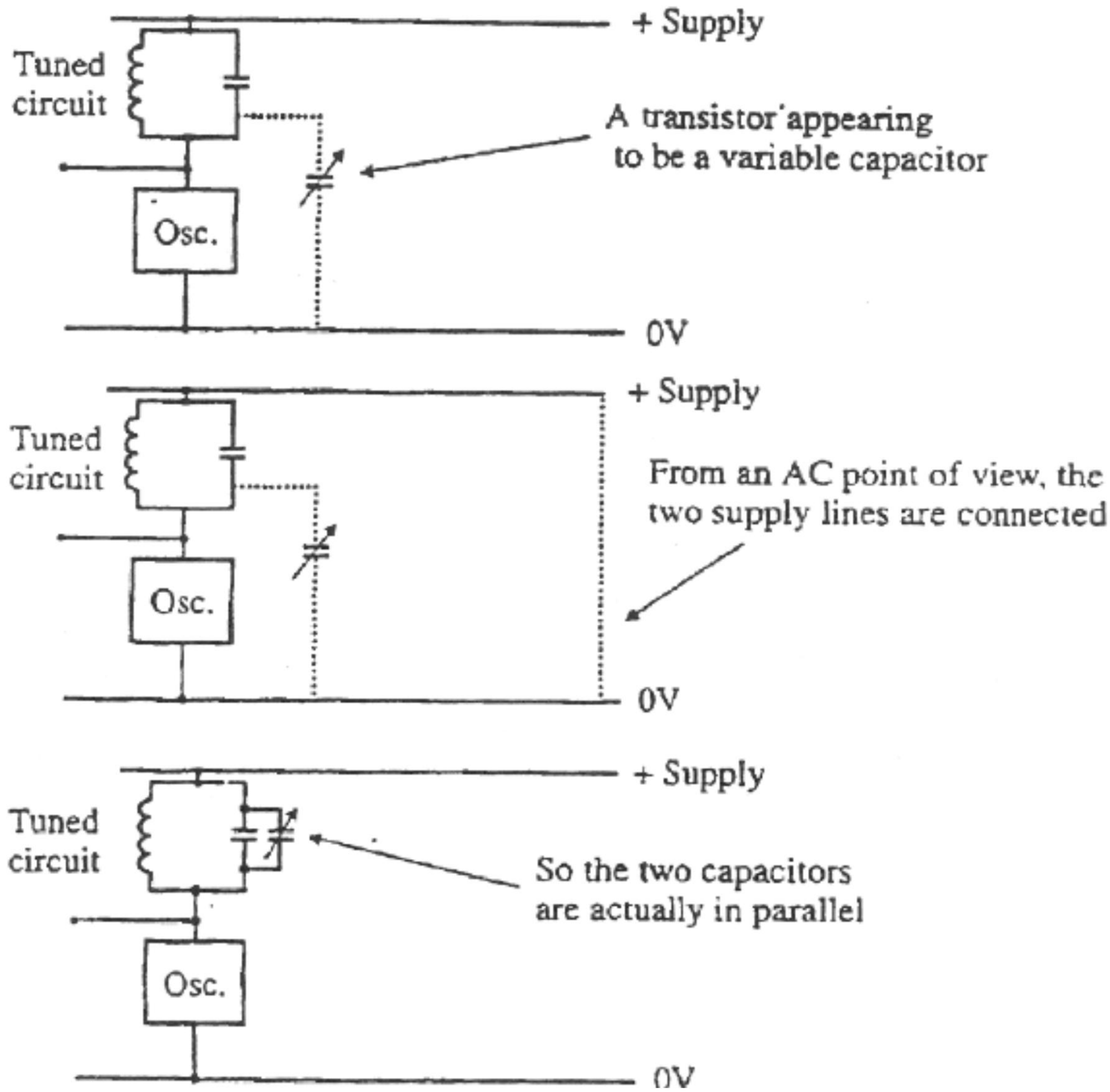
Aim: Study of Frequency Modulation Using Reactance Modulator.

Apparatus: FM Generator Kit, CRO, Connecting Probes etc.

Theory: Figure shows a complete reactance modulator.



In figure 1 the left hand half is the previous varactormodulator simply an oscillator and a tuned circuit, which generates the un-modulated carrier. The capacitor C and the resistor R are the two components used for the phase shifting, and together with the transistor, form the voltage controlled capacitor. This voltage-controlled capacitor is actually in parallel with the tuned circuit. This is not easy to see but figure 18 may be helpful. In the first part of the figure the capacitor and associated components have been replaced by the variable capacitor, shown dotted. In the next part, the two supply lines are connected together. We can justify this by saying that the output of the DC power supply always includes a large smoothing capacitor to keep the DC voltages at a steady value.



This large capacitor will have a very low reactance at the frequencies being used in the circuit less than a milliohm. We can safely ignore this and so the two supply lines can be assumed to be joined together. Remember that this does not affect the DC potentials, which remain at the normal supply voltages. If the two supply voltages are at the same AC potential, the actual points of connection do not matter and so we can redraw the circuit as shown in the third part.

Operation of the Reactance Modulator:

1. The oscillator and tuned circuit provide the un-modulated carrier frequency and this frequency is present on the collector of the transistor.
2. The capacitor and the resistor provide the 90° phase shift between the collector voltage and current. This makes the circuit appear as a capacitor.

3. The changing information signal being applied to the base has the same effect as changing the bias voltage applied to the transistor and, this would have the effect of increasing and decreasing the value of this capacitance.
4. As the capacitance is effectively in parallel with the tuned circuit the variations in value will cause the frequency of resonance to change and hence the carrier frequency will be varied in sympathy with the information signal input.

Procedure:

This experiment investigates how ST2203's reactance modulator circuit performs frequency modulation. This circuit modulates the frequency of a carrier sine wave, according to the audio signal applied to its modulating output. To avoid unnecessary loading of monitored signals, X10 Oscilloscope probes should be used throughout this experiment.

1. Ensure that the following initial conditions exist on the ST2203 Module.

    a. All Switch Faults in 'Off' condition.

    b. Amplitude potentiometer (in the mixer/amplifier block) in fully clockwise.

    c. VCO switch (in phase-locked loop detector block) in 'Off' position.

2. Make the connections as shown in figure 19.

3. Turn on power to the ST2203 module

4. Turn the audio oscillator block's amplitude potentiometer to its fully clockwise (Maximum) positions, and examines the block's output (TP1) on an Oscilloscope.

This is the audio frequency sine wave, which will be used as our modulating signal. Note that the sine wave's frequency can be adjusted from about 300 Hz to approximately 3.4 KHz by adjusting the audio oscillator's frequency potentiometer Note also that the amplitude of this audio modulating signal can be reduced to zero, by turning the audio oscillator's amplitude potentiometer to its fully counter clockwise position.

5. Connect the output socket of the audio oscillator block to the audio input socket of the modulator circuit's block, as shown in figure 19.

6. Put the reactance varactor switch in the reactance position. This switches the output of the reactance modulator through to the input of the mixer/amplifier block~ and also switches off the varactor modulator block to avoid interference between the two modulators.

7. The output signal from the reactance modulator block appears at TP13, before being buffered and amplified by the mixer/amplifier block. Although the output from the reactance modulator

block can be monitored directly at TP13, any capacitive loading affect this point (e.g. due to an Oscilloscope probe) may slightly affect the modulator's output frequency.

In order to avoid this problem we will monitor the buffered FM output signal from the mixer/amplifier block at TP34.

8. Put the reactance modulator's potentiometer in its midway position (arrow pointing towards top of PCB) then examine TP34.

Note : that the monitored signal is a sine wave of approximately 1.2Vpp centered on 0 volts DC This is our FM carrier, and it is presently un-modulated since the reactance modulator's audio input signal has, zero amplitude.

9. The amplitude of the FM carrier (at TP34) is adjustable by means of the mixer/amplifier block's amplitude potentiometer, from zero to its present level.

Try turning this potentiometer slowly anticlockwise, and note that the amplitude of the FM signal can be reduced to zero.

Return the amplitude potentiometer to its fully clockwise position.

10. The frequency of the FM carrier signal (at TP34) should be approximately 455 KHz at the moment This carrier frequency can be varied from 453 KHz to 460 KHz (approximately) by adjusting the carrier frequency potentiometer in the reactance modulator block.

Turn this potentiometer over its range of adjustment and note that the frequency of the monitored signal can be seen to vary slightly. Note also that the carrier frequency is maximum when the potentiometer is in fully clockwise position.

11. Try varying the amplitude & frequency potentiometer in audio oscillators block, and also sees the effect of varying the carrier frequency potentiometer in the mixer/amplifiers block.

12. Monitor the audio input (at TP6) and the FM output (at TP34) triggering the Oscilloscope on the audio input signal. Turn the audio oscillator's amplitude potentiometer throughout its range of adjustment and note that the amplitude of the FM output signal does not change. This is because the audio information is contained entirely in the signal's frequency, and not in its amplitude.

13. The complete circuit diagram for the reactance modulator is given at the end of operating manual. If you wish, follow this circuit diagram and examine the test points in the reactance modulator block, to make sure that you fully understand how the circuit is working.

14. By using the optional audio input module, the human voice can be used as the audio modulating signal, instead of using ST2203’s audio oscillator block.

If you have an audio input module, connect the module's output to the audio input socket in the modulator circuit’s block

The input signal to the audio input module may be taken from an external microphone (supplied with the module), or from a cassette recorder, by choosing the appropriate switch setting on the modules.

Conclusion:____

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Experiment No.7

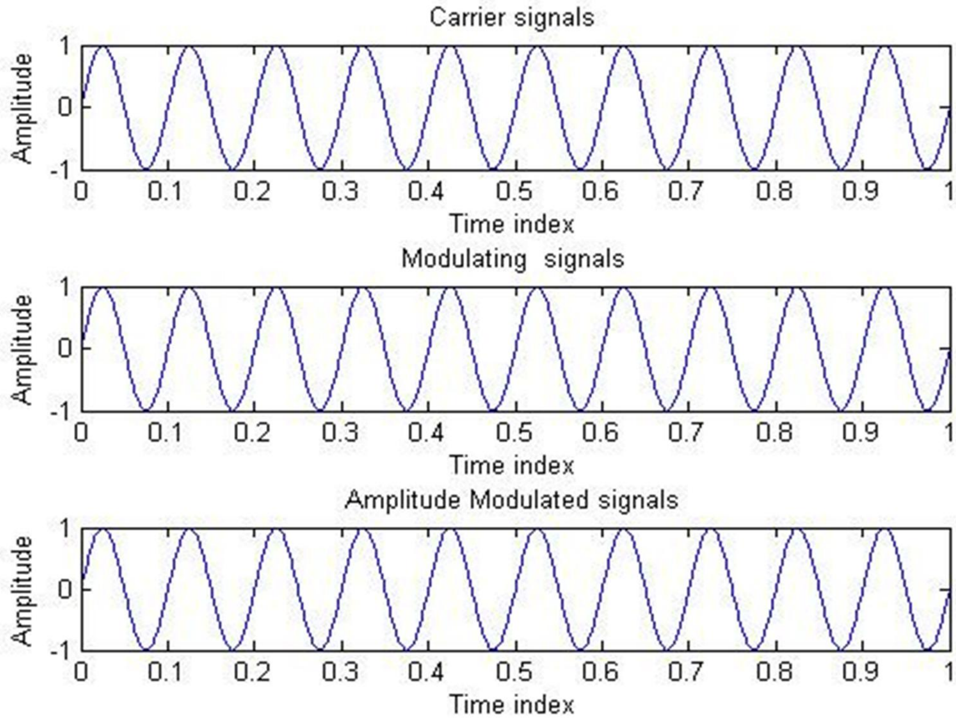
Aim : Matlab code for Amplitude Modulation.

Proceducer:
1. Enter message signal frequency.
2. Enter carrier signal frequency.
3. Enter modulation index.
4. Generate message and carrier signal using sin function.
4. Write equation of amplitude modulation.
5. plot message , carrier and modulated signal.

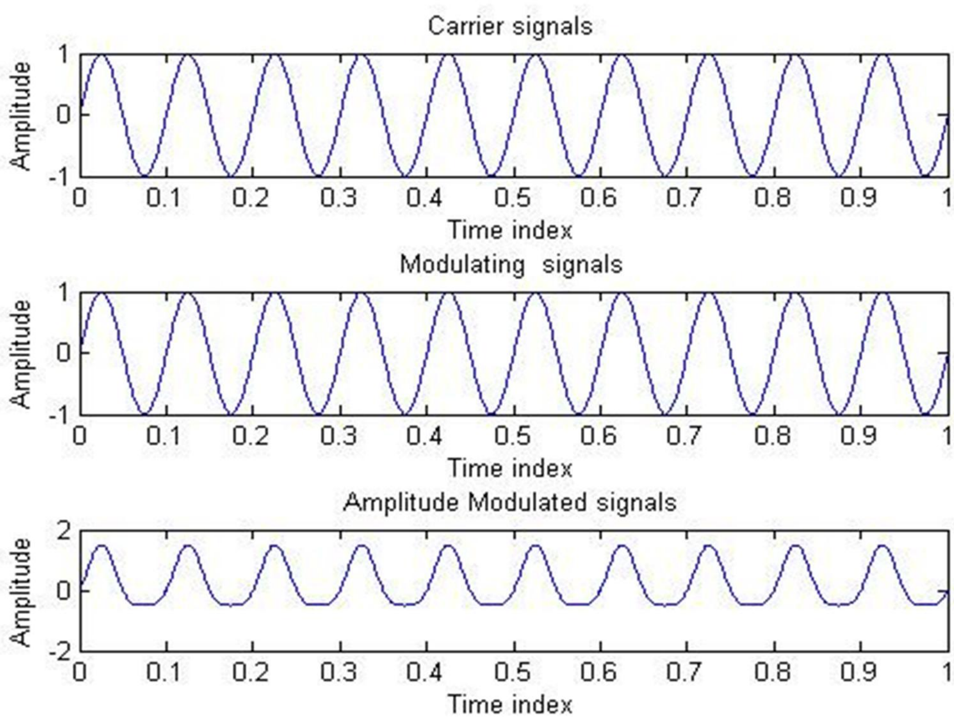
Code:

```
clc;
close all;
clear all;
% AM signal generation
fc=input('Please enter the carrier signal frequency in Hz,fc=');
fm=input('Please enter the modulating signal frequency in
Hz,fm=');
m=input('Modulation index,m=');
n=0:0.001:1;
c=sin(2*pi*fc*n);
M=sin(2*pi*fm*n);
y=(1+m*M).*c;
subplot(3,1,1);
plot (n,c);
ylabel('Amplitude');
xlabel('Time index');
title('Carrier signals');
subplot(3,1,2);
plot (n,M);
ylabel('Amplitude');
xlabel('Time index');
title('Modulating  signals');
subplot(3,1,3);
plot (n,y);
ylabel('Amplitude');
xlabel('Time index');
title('Amplitude Modulated signals');
y1=abs(fft(y));
figure;
plot( y1);
```

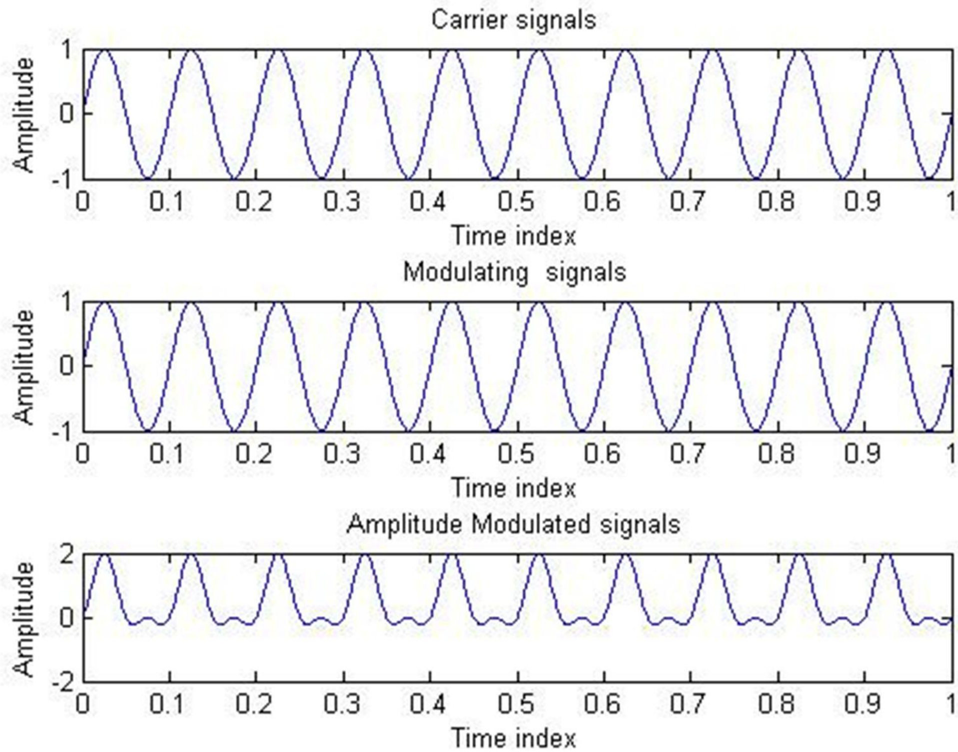
Output: **M=0**



M=0.5



M=1



Carrier signals
Amplitude
1
0
-1
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Time index
Modulating signals
Amplitude
1
0
-1
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Time index
Amplitude Modulated signals
Amplitude
2
0
-2
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Time index

Experiment No.8

Aim : Matlab code for Frequency Modulation.

Procedurer:

1. Enter message signal frequency.
2. Enter carrier signal frequency.
3. Enter modulation index.
4. Generate message and carrier signal using sin function.
4. Write equation of frequency modulation.
5. Plot message , carrier and modulated signal.

Code:

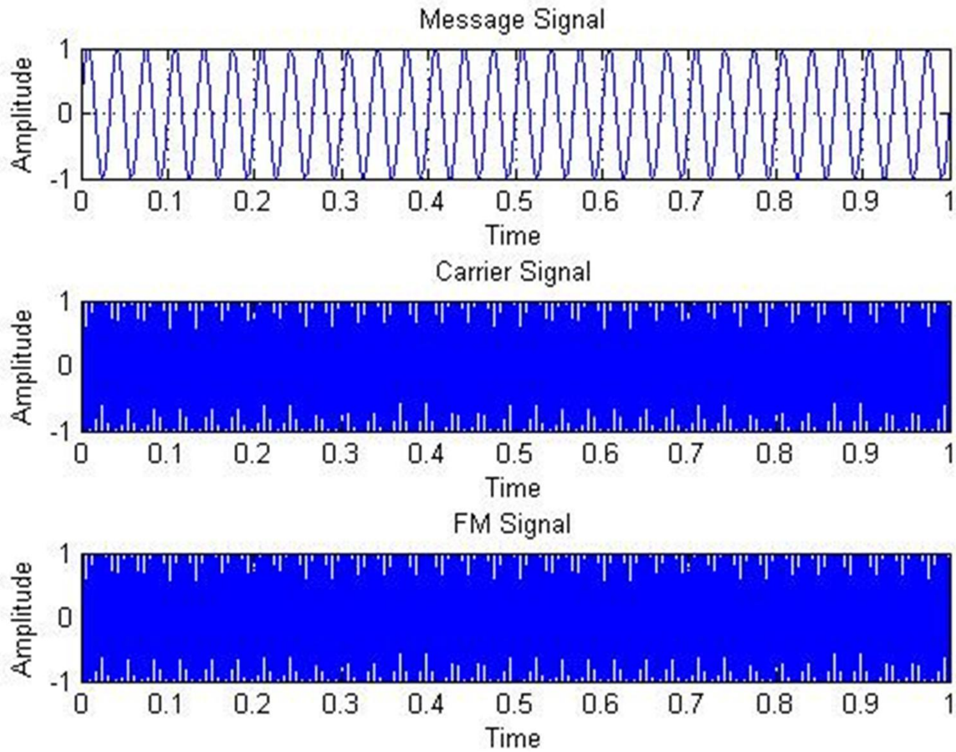
```
clc;
clear all;
close all;
fc1=300;
fm1=30;
mi=0.001;
t=0:0.0001:1;
p=sin(2*pi*fm1*t);
subplot(3,1,1);
plot(t,p);
xlabel('Time');
ylabel('Amplitude');
title('Message Signal');
grid on;

q=sin(2*pi*fc1*t);
subplot(3,1,2);
plot(t,q);
xlabel('Time');
ylabel('Amplitude');
title('Carrier Signal');
grid on;

r=sin(2*pi*fc1*t+(mi.*sin(2*pi*fm1*t))); %Frequency changing
w.r.t Message
subplot(3,1,3);
plot(t,r);
xlabel('Time');
ylabel('Amplitude');
title('FM Signal');
grid on;

figure;
plot(fft(r));
```

Output:



Message Signal
Amplitude
1
0
-1
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Time
Carrier Signal
Amplitude
Time
FM Signal
Amplitude
Time