Jawaharlal Nehru Engineering College

Laboratory Manual

ANALOG COMMUNICATION ENGINEERING

For

Second Year Students

Lab manual made by

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PROF. S. A.ANNADATE

FOREWORD

It is my great pleasure to present this laboratory manual for second year engineering students for the subject of advance communication engineering keeping in view the vast coverage required for visualization of concepts of communication engineering.

As a student, many of you may be wondering with some of the questions in your mind regarding the subject and exactly what has been tried is to answer through this manual.

Faculty members are also advised that covering these aspects in initial stage itself, will greatly relieve them in future as much of the load will be taken care by the enthusiastic energies of the students once they are conceptually clear.

HOD

LABORATORY MANUAL CONTENTS

This manual is intended for the second year students of ECT branch in the subject of Analog Communication Engineering. This manual typically contains practical/Lab Sessions related to advance communication engineering, covering various aspects, related to the subject to enhance understanding.

Students are advised to thoroughly go through this manual rather than only topics mentioned in the syllabus, as practical aspects are the key to understanding and conceptual visualization of theoretical aspects covered in the books.

Good Luck for your Enjoyable Laboratory Sessions.

Dr. S.N.PAWAR
PROF. S. A.ANNADATE

SUBJECT INDEX

1. DOs and DON'Ts in Laboratory.

2.Lab exercise

    1. To study of RF Signal Generator
    2. To study Amplitude Modulation and Demodulation
    3. To calculate Modulation index by direct method
    4. To calculate Modulation index by indirect method (Trapezoidal method)
    5. To study Frequency Modulation and Demodulation
    6. To study Pre – emphasis and de –emphasis
    7. To study Sampling and its reconstruction.
    8. To study pulse code modulation
    9. To study Time division multiplexing
    10. To study PA System

3. Quiz on the subject.
4. Conduction Viva-Voce Examination.
5. Evaluation and Marking Systems.

DOs and DON'Ts in Laboratory:

1. Do not handle any equipment before reading the instructions/Instruction manuals.

2. Read carefully the power ratings of the equipment before it is switched on whether ratings 230 V/50Hz or 115V/60 Hz. For Indian equipments, the power ratings are normally 230V/50Hz. If you have equipment with 115/60 Hz ratings, do not insert power plug, as our normal supply is 230V/50 Hz, which will damage the equipment.

3. Observe type of sockets of equipment power to avoid mechanical damage.

4. Do not forcefully place connectors to avoid the damage.

5. Strictly observe the instructions given by the teacher/Lab Instructor.

Instruction for Laboratory Teachers:

1. Submission related to whatever lab work has been completed should be done during the next lab session.
2. The promptness of submission should be encouraged by way of marking and evaluation patterns that will benefit the sincere students.

Experiment No.1

AIM: To study RF signal generator.

APPARATUS: oscilloscope, function generator, probes, etc.

THEORY:

RADIO-FREQUENCY (RF) SIGNAL GENERATORS

In addition to the necessary power supply, a typical RF signal generator contains three other main sections: an OSCILLATOR CIRCUIT, a MODULATOR, and an OUTPUT CONTROL CIRCUIT. The modulator modulates the RF signal of the oscillator. In addition, most RF generators are provided with connections through which an external source of modulation of any desired waveform can be applied to the generated signal. Metal shielding surrounds the unit to prevent signals from the oscillator from affecting the circuit under test.

A block diagram of a representative RF signal generator is shown in figure. The function of the oscillator stage is to produce a signal that can be accurately set in frequency at any point within the range of the generator. The type of oscillator circuit used depends on the range of frequencies for which the generator is designed. In lower frequency RF signal generators, the oscillating circuit consists of one of a group of coils combined with a variable capacitor. One of the coils is selected by the position of a range selector switch that connects the coil to a capacitor to provide an inductance-capacitance circuit. The inductive-capacitance circuit then has the correct range of resonant frequencies.

In amplitude modulation, the amplitude of carrier is changed in accordance with Instantaneous value of modulating signal.

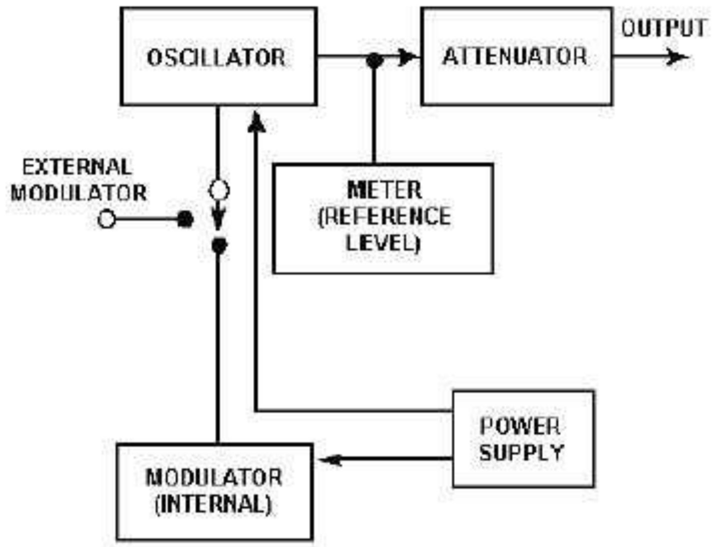


Fig: RF signal generator block diagram

The function of the modulator is to produce an audio (or video) modulating signal that can be superimposed on the RF signal produced by the oscillator. The modulating signal may be provided by an audio oscillator within the generator. This is termed INTERNAL MODULATION. It may also be derived from an external source. This is termed EXTERNAL MODULATION. In some signal generators, either of these two methods of modulation can be employed. In addition, a means of disabling the modulator section is available so that the pure, un-modulated signal from the oscillator can be used when desired.

Procedure:-

1. Connect the probe at output terminal of RF generator n another to the oscilloscope.
2. Check the different waveforms on oscilloscope.

Conclusion:-

Experiment No.2

Aim:- To study Amplitude Modulation and Demodulation.

Apparatus :- Double side Band AM radio Transmitter and Receiver Kits, Multi meter.

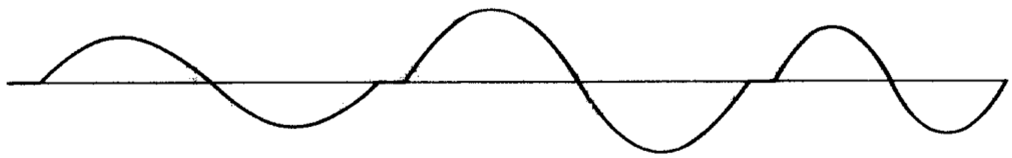
Theory :-

The original broadcast RF signal is picked up by the aerial and selected by a relatively broadband RF filter. This signal is usually amplified before entering the mixer. The mixer then combines the incoming RF signal and the local Oscillator signal. Shift the input signal to a lower frequency. The mixer thus receives two RF signals- one at the true RF, direct from the tuned circuit, and one at RF plus an intermediate frequency (IF), direct from the local oscillator. The result of mixing the two is to extract the difference or beat frequency. This is the IF frequency which now contains the AF components of the original RF signal.

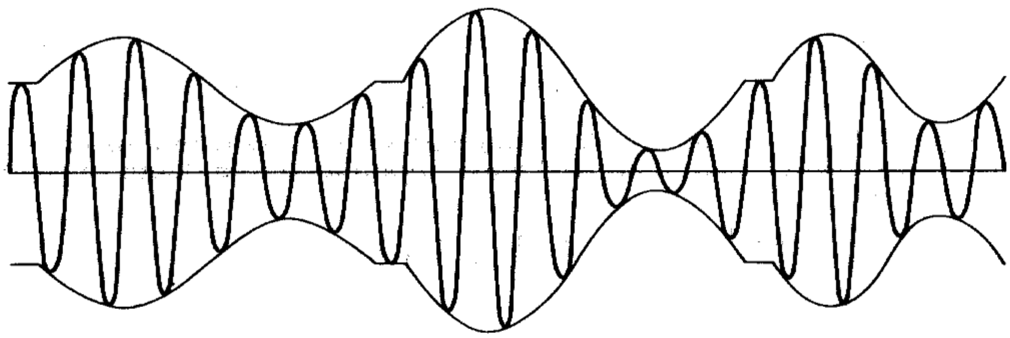
The signal is then passed through a multiple stage fixed filter tuned at 455 kHz (which corresponds to the intermediate frequency). The filter has a bandwidth of 10kHz, which matches the bandwidth of each radio station. The IF signal is then amplified and subsequently passed through an envelope detector before AF amplification.

The block diagram also shows feedback from the detector to the IF amplifier denoted as AGC or automatic gain control. This provides automatic regulation of the gain of the receiver in inverse proportion to the signal strength. This tends to keep the output level constant regardless of the input signal level eliminates much of the need to readjust the volume control when changing radio stations

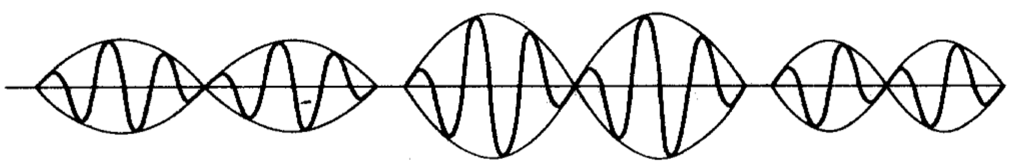
Wave form:-



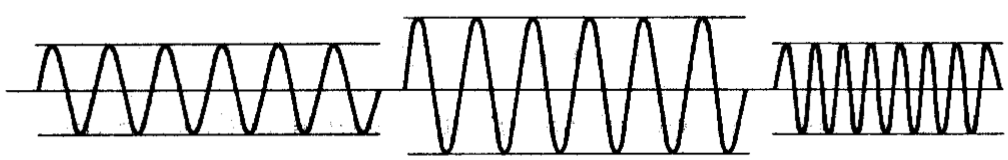
(a) Modulating signal



(b) AM wave



(c) Suppressed-carrier wave



(d) SSB suppressed-carrier wave

Procedure **:-**

(i) Tracing Method to be followed.

(II) Traced circuit diagram.

(iii) Detection of fault.

(iv) Interpretation of fault.

Circuit diagram, Block diagram.

(vi) Proper placement of components.

Conclusion:-

Experiment No.3

AIM: To calculate Modulation index by direct method

EQUIPMENT: AM modulator kit, oscilloscope, AF function generator, probes, wires etc.

THEORY:

In amplitude modulation, the amplitude of carrier is changed in accordance with
Instantaneous value of modulating signal.

$$E= (_{EC\ MAX} + E_M) \sin w_c t$$

Where em = Em max sin wmt is modulating signal.

Ec= EC MAX sinwct is a carrier signal.

E= [Ecmax + Em sin wmt]sinwct .

The modulation index is defined as ,

M= Emmax

=Ecmax

e= Ecmax(1+msinwmt)sinwmt or

e=sinwct + m sinwmt . sinwct

where Ecmax= 1v

e= sinwct + m/2[cos(wc-wm)-cos(wc+wm)

1.First term is a carrier wave of amplitude 1v & frequency w=2 π fc.

2. The second term is a cosine wave of amplitude ½ m &frequency (fc-fm) this is lower side band (LSB).

3. The third term is a cosine wave of amplitude m/2 & frequency (fc+ fm). This is upper side band (USB).

Measurement of carrier wave (RF wave)

1. Give supply to the kit of AM modulator.
2. The carrier wave must have generated carrier wave , open ckt terminal a & b.
3. Connect probe to oscilloscope & other terminal 'a' & measure carrier voltage with respect to ground.
4. Measure the carrier wave & draw the waveform.

Measurement of % modulation.

1. Give supply to the kit of AM modulator.
2. Now short-circuit the terminal a 6 b so that carrier voltages can be applied to the circuit.

3. Using function generator provides a selected frequency to the primary of transformer, which is at emitter of transistor.
4. Now connect probe from o/p ckt to the oscilloscope & measure the amplitude
Modulation Wave.
5. Calculate E_{max} (p-p) & E_{min}(p-p) for 3 set of different readings.

OBSERVATION:

$$\%M = \frac{E_{Max} - E_{Min}}{E_{Max} + E_{Min}} x100$$

SR NO	E_{Max} (p-p)	E_{Min} (p-p)	% M

Conclusion:- Thus we have calculated MI of AM wave.

Experiment No.4

AIM: To calculate Modulation index by indirect method (Trapezoidal method)

APPARATUS: AM modulator kit, oscilloscope, AF function generator, probes, wires etc.

THEORY:

In amplitude modulation, the amplitude of carrier is changed in accordance with
Instantaneous value of modulating signal.

$$E = (E_{C\,MAX} + E_M) \sin w_c t$$

Where em = Em max sin wmt is modulating signal.

Ec= EC MAX sinwct is a carrier signal.

E= [Ecmax + Em sin wmt]sinwct .

The modulation index is defined as ,

M= Emmax

=Ecmax

e= Ecmax(1+msinwmt)sinwmt or

e=sinwct + m sinwmt . sinwct

where Ecmax= 1v

e= sinwct + m/2[cos(wc-wm)-cos(wc+wm)

1.First term is a carrier wave of amplitude 1v & frequency w=2 π fc.

2. The second term is a cosine wave of amplitude ½ m &frequency (fc-fm) this is lower side band (LSB).

3. The third term is a cosine wave of amplitude m/2 & frequency (fc+ fm). This is upper side band (USB).

Measurement of carrier wave (RF wave)

5. Give supply to the kit of AM modulator.
6. The carrier wave must have generated carrier wave , open ckt terminal a & b.
7. Connect probe to oscilloscope & other terminal 'a' & measure carrier voltage with respect to ground.
8. Measure the carrier wave & draw the waveform.

Measurement of % modulation.

4. Give supply to the kit of AM modulator.

5. Now short-circuit the terminal a 6 b so that carrier voltages can be applied to the circuit.
6. Using function generator provides a selected frequency to the primary of transformer, which is at emitter of transistor.
4. Now connect probe from o/p ckt to the oscilloscope & measure the amplitude
   Modulation Wave.
5. Calculate E_{max} (p-p) & E_{min}(p-p) for 3 set of different readings.

OBSERVATION:

%M = (L1- L2) / (L1 + L2) x 100

SR NO	L1	L2	%M

CONCLUSION: Thus we have calculated MI of AM wave.

Experiment No. 5

AIM: To study frequency modulation and Demodulation.

APPARATUS: Matrix trainer kit - Frequency modulator, Function generator, Connecting wires, Oscilloscope

THEORY

A modulator is a circuit which changes frequency of a carrier w.r.t. instantaneous amplitude of modulating signal while keeping other parameters constants.

1. Tuning – allows different channels to be selected.
2. Transmission range - as the carrier is high frequency the transmission range is high.
3. Practical antenna length – as the carrier frequency is high the antenna dimensions are small.
4. Wireless communication – as the carrier is radiated in air, wireless communication is possible.

The F.M. (frequency modulator) is the circuit used for radio / T.V. communication and in other applications.

The advantages of using F.M. are

1. Noiseless reception.
2. Operating range is large.
3. High frequency
4. High efficiency of transmission.

CIRCUIT DESCRIPTION

The practical circuit of the frequency modulator is shown in figure.

The circuit uses function generator I.C. XR 8038 this is a precise waveform generator IC capable of generating sine, square, triangular, sawtooth and pulse waveforms with a minimum number of external components and adjustments. Its operating frequency range is from 0.001Hz to 1 Mhz.

The frequency of the waveform generator is direct function of d.c. voltage at terminal 7. Thus frequency modulation can be achieved by giving modulating signal to pin 7. The pin 2 output sine wave which is then frequency modulation by the modulating signal.

The input modulating signal is applied to pin 7 and to pin 8 for small frequency deviation, large deviation are possible by connecting signal to pin 8 only. C2 and C3 is a timing capacitor.

Selectable through J1 for two frequency range, the preset P5 connected at pin 4 and 5 are for adjusting duty cycle. The preset P3 and P4 at pin 1 and 12 adjusts the sine wave distortion. Pin 2 is sine wave output (modulated). POT P2 for carrier frequency adjustment.

PROCEDURE

1. Keep the power switch on off condition.
2. Connect the power cord to mains supply.
3. Pot p1 works as attenuator provided for controlling the input to modulator (modulating signal).
4. Pot p2 is provided for carrier frequency variation from 70 KHz. to 110 KHz.
5. Put the power switch off the kits on and see the power switch glows.
6. Connect oscilloscope at output.
7. Set output frequency (carrier) above 80 KHz. With pot. P2 and j1 at b side. (j1 at a for low frequency carrier fm observation)
8. Connect the function generator sine wave output to modulator input (any frequency from 20 to 1 KHz).
9. Set the input amplitude with pot P1 such that modulation at the output is seen.

10. Observe and note the input frequency and output frequency on scope.
11. Repeat above experiment for no signal and at different frequencies.
12. Place J2 at A side and connect +v as modulating signal (AF input) and note dc voltages on pin 7 for minimum and maximum FM output frequencies.

Demodulator

In FM signal the amplitude of wave remain constant and modulated signal is used to change the carrier frequency. In such envelope detector can not be used for demodulating of those wave the circuit use to obtain modulating signal from FM wave are discriminator. The most commonly used discriminator are the faster seeley discriminator.

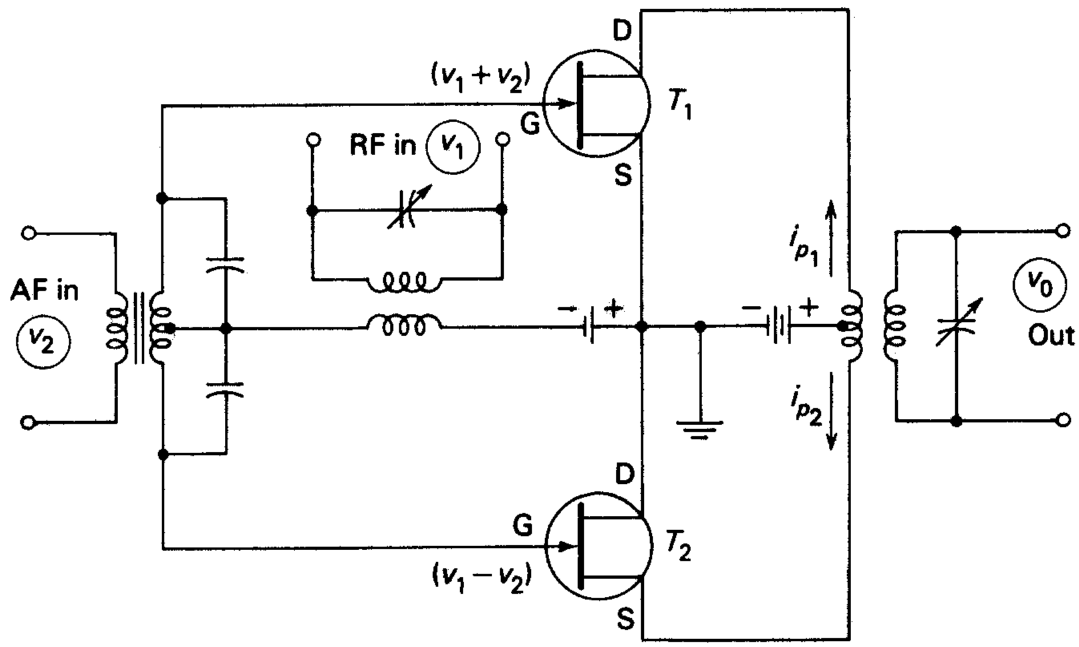
Figure (1) shows the schematic diagram of faster seeley discriminator. It consists of two identical diode detector circuit L,S,D,R,C, AND LS_2, D_2,R_2,C_2 . The FM signal is inductively coupled to see. They also exist on a capacitive coupling between primary an centre tapped sec. the input of diode D_1 = vector sum of the pri voltage.

When incoming frequency is greater than fc then, sec. circuit behave as inductor and lags Vs by an angle less than 90^0. In the result, the ph. Relation of Vs1 and Vs2 with Vp becomes as shown in figure (20).

Signal frequency smaller than the centre tapped fc. The secondary circuit behaves as capacitive and is leads Vs by 90^0 but the reference to centre point. They are 180^0 out of phase as D_1 is smaller than the vector sum of Vs2 and Vp.

It can therefore concluded that effect variation in input signal amp. is to vary the +ve amp of output signal corres and the faster seely discriminator is sensitive output is always proceed by limiter.

Block Diagram:-



PROCEDURE

1. Connect the signal generator at input terminal and keep the signal level at suitable level.
2. Connect oscilloscope one at output and other at input.
3. Connect digital multimeter at output terminal.
4. Vary frequency in step and note down ph. Shift from oscilloscope. Af output on dmm.
5. Also observe the ph shift on oscilloscope when af output is on the side and also on –ve side.

CONCLUSION

Thus, we have studied phase discriminator and graph plotted between frequency AF output is approx linear.

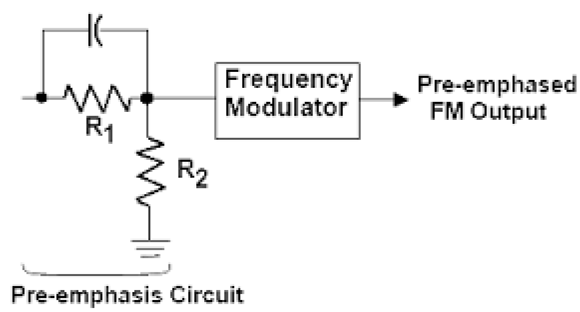
Experiment No.6

AIM: To study Pre emphasis and De emphasis.

APPARATUS: Pre emphasis and De emphasis training kit, Connecting wires, Oscilloscope

THEORY

Pre Emphasis



(a) Pre-emphasis Circuit

In processing electronic audio signals, pre-emphasis refers to a system process designed to increase (within a frequency band) the magnitude of some (usually higher) frequencies with respect to the magnitude of other (usually lower) frequencies in order to improve the overall signal-to-noise ratio by minimizing the adverse effects of such phenomena as attenuation distortion or saturation of recording media in subsequent parts of the system. The mirror operation is called de-emphasis, and the system as a whole is called emphasis.

Pre-emphasis is achieved with a **pre-emphasis network** which is essentially a calibrated filter. The frequency response is decided by special time constants. The cutoff frequency can be calculated from that value.

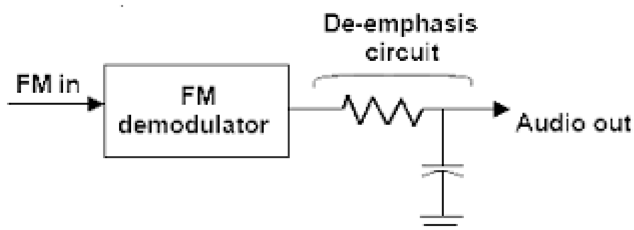
Pre-emphasis is commonly used in telecommunications, digital audio recording, record cutting, in FM broadcasting transmissions, and in displaying the spectrograms of speech signals.

One example of this is the RIAA equalization curve on 33 rpm and 45 rpm vinyl records. Another is the Dolby noise-reduction system as used with magnetic tape.

In high speed digital transmission, pre-emphasis is used to improve signal quality at the output of a data transmission. In transmitting signals at high data rates, the transmission medium may introduce distortions, so pre-emphasis is used to distort the transmitted signal to correct for this distortion. When done properly this produces a received signal which more closely resembles the original or desired signal, allowing the use of higher frequencies or producing fewer bit errors.

Pre-emphasis is employed in frequency modulation or phase modulation transmitters to equalize the modulating signal drive power in terms of deviation ratio. The receiver demodulation process includes a reciprocal network, called a de-emphasis network, to restore the original signal power distribution.

De Emphasis



(c) De-emphasis circuit

In telecommunication, **de-emphasis** is the complement of pre-emphasis, in the anti noise system called emphasis. De-emphasis is a system process designed to decrease, (within a band of frequencies), the magnitude of some (usually higher) frequencies with respect to the magnitude of other (usually lower) frequencies in order to improve the overall signal-to-noise ratio by minimizing the adverse effects of such phenomena as attenuation distortion or saturation of recording media in subsequent parts of the system.

Special time constants dictate the frequency response curve, from which one can calculate the cutoff frequency.

In serial data transmission, de-emphasis has a different meaning, which is to reduce the level of all bits except the first one after a transition. That causes the high frequency content due to the transition to be emphasized compared to the low frequency content which is de-emphasized. This is a form of transmitter equalization; it compensates for losses over the channel which is larger at higher frequencies. Well known serial data standards such as PCI Express, SATA and SAS require transmitted signals to use de-emphasis.

CONCLUSION

Thus, we have studied the Pre emphasis and De emphasis circuit and its use in telecommunications.

Experiment No.7

AIM: To study sampling and its re-construction.

THEORY:

SAMPLING FREQUENCY: The 6.4 MHz Crystal oscillator generates the 6.4 MHz clock. The decade counter divides the frequency by 10 and the ripple counter generate the basic sampling frequencies -320KHz to 20 KHz and the other control frequencies. The basic sampling frequencies is given to a multiplexer For each "Press" on the frequency select
switch, the output of the state counter increases by one and it counts from 000 to 100. As
the state counter counts from 000 to 100, the corresponding input of the multiplexer is
switched to the output. As soon as the count reaches 101, the output of the 3 to 8 decoder
resets the state counter and the whole cycle repeats. Also LED connected to the output of
the decoder is switch ON, which indicates the sampling frequency selected. Refer the
truth table for better understanding.

EQUIPMENTS:
Analog Signal Sampling & Reconstruction Kit, 20 MHz Dual Trace Oscilloscope,Patch Chords

PROCEDURE:

1. Connect power supply in proper polarity to the kit & switch on.
2. Connect the lKHz,5V pp Sinewave signal, generated onboard, to the ANALOG INPUT, by means of the patch-cords provided.
3. Connect the sampling frequency signal in the internal mode, by means of the shorting pin provided.
4. Using switch SW1 select 50% duty cycle as shown in the table.
5. Connect the SAMPLE OUTPUT to the input of the 2nd Order Low Pass Filter.
6. Using frequency selector switch, select desired sampling frequency. The selected sampling frequency is indicated by the glowing LED.

7. Take observation as mentioned below for various sampling frequency: 32 KHz, 16KHz, 8KHz, 4KHz , 2KHz.
OBSERVATIONS:_

CONCLUSION:

From the above observations we conclude that as the Sampling Frequency is increased, the reconstructed output is less distorted and almost original signal is reconstructed. For a sampling frequency of 2KHz, only 2 samples of the 1KHz signal are
taken; whereas that for a sampling frequency of 8KHz, 8 samples of 1KHz signal are taken.
Hence, as the number of samples taken of the signal increases, the distortion of the
reconstructed signal decreases.

As per the Nyquist Criterion atleast two samples are required for the reconstruction of the signal. If the Nyquist Criterion is not satisfied, or if the signal is not
band-limited, then spectral overlap, called "aliasing" occurs, causing higher frequencies to
show up at lower frequencies in the recovered message, and specially in voice transmission
intelligibility is seriously degraded Thus, universally for the voice band (300Hz to 3300Hz), the sampling frequency used is 8KHz, which satisfies the Nyquist Criterion.

Experiment No.8

AIM: To study pulse code modulation

APPARATUS:

PCM kit, oscilloscope, function generator, probes.

THEORY:

Pulse Code Modulation is also known as a digital pulse modulation technique. The pulse modulation

(PCM) is quite complex compared to the analog pulse modulation techniques(i.e. PAM, PWM, PPM) in these sense that the message signal is subjected to a great number of operations.

Fig. 1 shows the basic elements of a PCM system. It consists of three main parts i.e. transmitter, transmission path and receiver. The essential operations in the transmitter of a PCM system are sampling, quantizing and encoding as shown in fig.1.

Sampling is operation in which an analog (i.e. continuous-time) signal is sampled according to the sampling theorem resulting in a discrete-time signal. The quantizing and encoding operations are usually performed in the same circuit which is known as an analog-to-digital converter (ADC).

Also the essential operations in the receiver are regeneration of impaired signals, decoding and demodulation of the train of quantized samples. These operations are usually performed in the same circuit which is known as digital-to-analog converter (DAC).

CONCLUSION:

Hence a digital modulation technique PCM is studied. It is a coding technique since analog signal is converted into binary coded bit stream.

Experiment No. 9

AIM-:-To study time division multiplexing.

APPARATUS-:-ADTRON TRAINER KIT

1. Sine wave signal generators.
2. Dual channel C.R.O.
3. Digital Multimeter etc.

-
- In place of logic data input analog signal may also be connected at the data input terminals with maximum amplitude of $\pm$ 4.5 volts peak to peak.

Theory:-

INTRODUCTION:-

The topic of TDM is an extension of pulse modulation system. It is treated here to permit the two major multiplexing methods to be compared. In time division multiplex, use is made of the fact that narrow pulses with wide spaces between them are generated in any of the pulse modulation systems, so that the spaces can be used by signals from other sources. Moreover, although the spaces are relatively fixed in width, pulses may be made as narrow as desired, thus permitting the generation of high level hierarchies.

The method of achieving TDM is best illustrated by describing the make-up of an actual system, and so a practical basic PMC system used in North America has been selected as the example. In somewhat simplified fashion, this may be described as a 24 channel system, having a sampling rate of 8000 samples per second, 8 bits i.e., 256 sampling levels) per sample, and a pulse width of approximately 0.625 μs. This means that the sampling interval is 1/8000 = 0.000125 s = 125 μs, and the period required for each pulse group is 8 x 0.625 = 5 μs. If there were no multiplexing and by one channel were sent, the transmission would consist of 8000 frames per second, each made up of furious activity during the first 5 μs and nothing at all during the remaining 120 μs. This would clearly be wasteful and would represent an unnecessary complicated method of encoding a single channel, and so this system loads this large spaces between the pulse groups. In fact, each frame is used to provide 24 adjacent channel time slots, with the twenty-fifth shot assigned for synchronization. Each frame consists of 193 bits – 24 x 8 for each channel, plus 1 for sync, and since there are 8000 frames per second, the bit rate is 1.544 Mbit/s.

Slow-speed TDM, as often used in radio telemetry, is produced simply with rotating mechanical switches. A number of channels are fed simultaneously to the switch in the transmitter one channel to each with contact while the output is taken from the moving rotor. This rotates slowly and remains in contact with each channel for a predetermined period, during which time the output of that channel is the only one passed on for transmission. There is a corresponding rotating switch in the receiver, synchronized to the one in the transmitter, which reverses the process to separate the received channels.

The high-speed TDM described here uses electronic switching and delay lines to accomplish the same result. Each sampling circuit, one per channel simultaneously receives a trigger pulse which causes it to sample its signals and each channel output is then fed to an adder. However whereas the output of the first sampler goes straight to the adder, that of the second is delayed by 5 μs, with a delay line or delay circuit. The output of the third sampling circuit is similarly delayed but by 10 μs. In this way each successive interval during the 125 μs frame is occupied by the transmission of a different channel, and the process is repeated 8000 times per second.

In the receiver, the output of the main detector is fed simultaneously to 24 AND gates. An AND gate, or coincidence circuit is a simple device having one output and two or more input terminals. So arranged that an output is obtained only if all (in this case both) input signals are present. In this case each gate has two input terminals, and the second input to each gate is provided from a clock-synchronized gating generator which a monostable multivibrator is providing rectangular pulses of 5 μs duration, 8000 times per second. Delay lines or circuits are used once again. With the gating pulse to the first gate not delayed at all, that to the second gate delayed by 5 μs and so forth. In this fashion each gate is open only during the appropriate time intervals, and the 24 channels are duly separated.

If transmission is by wire, the 1.544-Mbit/s pulse train is the signal sent, but if cable or radio communication is used, the pulse trains, all combined together into a higher TDM hierarchical level.

It should be mentioned that the 1.544 Mbit/s, μ-law system is by no means in world wide use. As a matter of fact is represents yet another of those instances where CCITT has had to produce two sets of parallel recommendations, one for the United States (and Japan in this case) and another for the rest of the world ! This other TDM system

Also used 8 bits per sample and a frame rate of 8000 per second, but has A-law pre emphasis and 32 channels, of which 30 are used for transmission and the remaining two (channels 0 and 16 for signaling and synchronization the pulse rate is 2.048 Mbit/s. In recommendation G.711, CCITT states that;

Digital paths between countries which have adopted different encoding laws should carry signals encoded in accordance with the A-law …. Any necessary conversion should be done by the countries using the μ-law.

PULSE COMMUNICATION TECHNIQUES

Pulse techniques are widely used in radar, satellite communication and telephone where available channel bandwidth has to be traded for noise reduction. In such Multiplex systems, the rate, of sampling pulses, should exceed 2 M. where M. is the modulating frequency. In a typical case of a value of sampling rate of 8000 per a sample of the analogue input signal, is taken every 125 μs, the pulse duration may be a suitable synchronizing pulse is needed to provide unison between the sending the channel distributors. Also sufficient time interval (guard time) is to be provided for transient of a pulse wave to decay and for the associated gates to open and close. An circuit can be used to separate a long synchronized pulse or a Schmitt trigger can loyed if the synchronizing pulse is marked by its amplitude.

A schematic diagram, of a time division multiplex system, is shown above.

CIRCUIT DESCRIPTION:

Block diagrams for generation and decoding of T.D.M. signals are as shown in the diagram.

Clock generator generates the clock pulses which are fed to the counter 1 and 2 synchronously. Clock freq. Can be varied by clock frequency adj. Potentiometer, and its frequency can be observed by blinking of Clk LED.

Counter 1 and counter 2 is two bit BCD counter. Count value A1 – A0 selects one of the input signal Do to D3 and transmits it to the output terminal Y. As clock pulse advances the count value by one, successive dates are transmitted, at the output from D0 to D3 on receipt of next clock pulse.

Similarly count value B1-B0 decodes the data at the X input to one of the output Y0 to Y3 and dates are decoded at the successive output Y0 to Y3 on receipt of next clock pulse.

It should be observed that as the counters are synchronized by the same clock pulses, data's transmitted are decoded and available on the same channel.

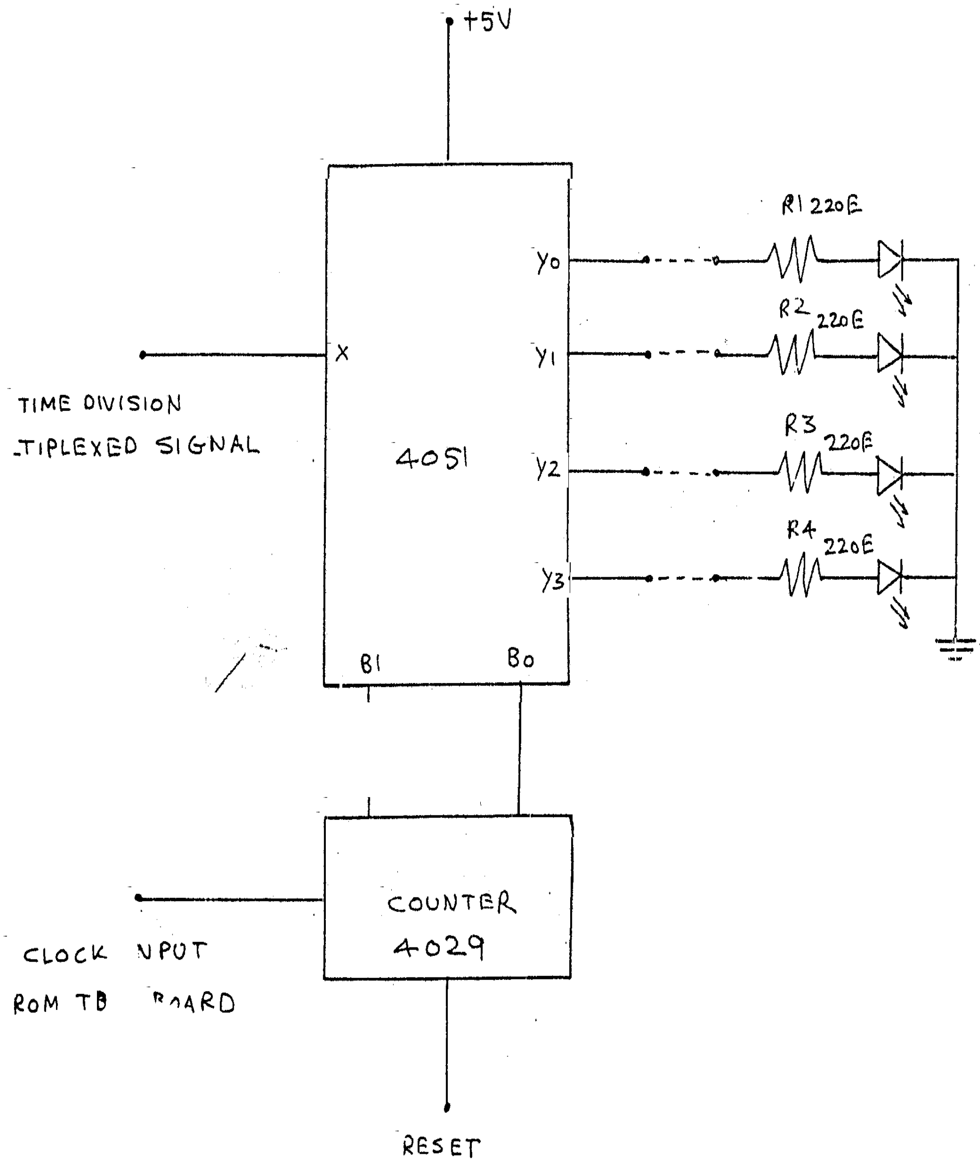
PROCEDURE-:-

- Connect the required supply and switch on the unit see that the supply LED glows.
- With jumper links, select any data from 0000 to 1111 for D0 to D3.
- Connect the output terminal Y to the logic indicator.
- Select low frequency for the clock oscillator by clock frequency adjust potentiometer.
- Clock frequency can be observed by blinking of Clk LED.
- Observe the output.
- You will observe that output selects the data in sequence from D0 to D3 at the rate of clock frequency which demonstrates the principal of T.D.M.

Repeat the above procedures for different clock freq. And so also for different input data.

The training people

TIME DIVISION DEMULTIPLEXING



Experiment No. 10

Aim: - To study PA system.

Apparatus: PA system, CRO and multimeter.

Block Diagram:

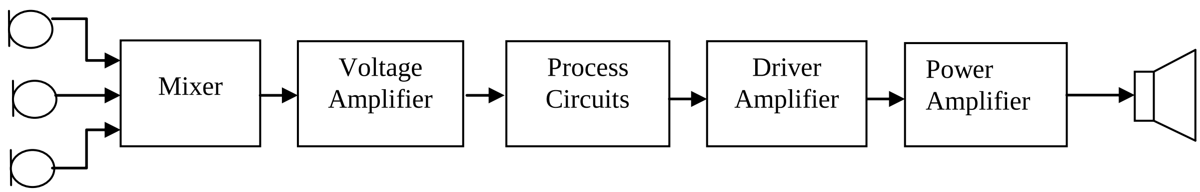


Figure 3.1: Block diagram of a PA system

Theory: -

A **public address system** (**PA system**) is an electronic system comprising microphones, amplifiers, loudspeakers, and related equipment. It increases the apparent volume (loudness) of a human voice, musical instrument, or other acoustic sound source or recorded sound or music. PA systems are used in any public venue that requires that an announcer, performer, etc. be sufficiently audible at a distance or over a large area. Typical applications include sports stadiums, public transportation vehicles and facilities, and live or recorded music venues and events. A PA system may include multiple microphones or other sound sources, a mixing console to combine and modify multiple sources, and multiple amplifiers and loudspeakers for louder volume or wider distribution.

Simple PA systems are often used in small venues such as school auditoriums, churches, and small bars. PA systems with many speakers are widely used to make announcements in public, institutional and commercial buildings and locations—such as schools, stadiums, and passenger vessels and aircraft. Intercom systems, installed in many buildings, have both speakers throughout a building, and microphones in many rooms so occupants can respond to announcements.

Procedure: -

1. Connect the supply and switch on System.
2. Apply the input signals to mixer.
3. Observe/Measure the signal wave parameters at different test points of different sections.
4. Repeat step 3 for different sections.

Conclusion: - The construction, working and Applications of PA system are stidied.

4. Quiz on the subject:

1) What is standard IF value for AM receiver & FM receiver?

2) In a receiver which stage rectifies the IF signal?

3) From which component the three point tracking is achieved?

4) Define double spotting ,image frequency ,sensitivity , selectivity & fiedility .

5) Define carrier signal, modulating signal & modulated signal.

6) How many side bands exists in AM&FM?

7) Define frequency spectrum in AM.

8) What is the use of Bessel's function?

9) What is the disadvantage of FM over AM?

10) What are the different types of detectors in AM & FM?

11) What are the different forms of AM?

12) What is AGC? What are it's types? State use of it ?

13) What is band spreading?

15) What is ganged tuning?

16) Which stage of AM & FM receiver determines the gain?

17) Define bandwidth. State the B.W. of AM & FM signal.

18) Define balanced modulator.

19) What are the o/p components of BM.?

20) What are the different types of BM?

21)) What is SSB ? What are it's types?

22) State the different AM & FM generation methods?

23) What is two & three point tracking?

24) What is IF of audio & video signal in TV?

24) What is aspect ratio?

25) What is PAL?

26) What is channel BW of TV signal?

27) What is interlaced scanning?

28) State the line frequency of TV system in India?

29) How many lines exist in one field in TV?

30) What modulation techniques are used for audio & video signal in TV?

31) What antenna is used for TV reception?

32) What is the use of pattern generator?

33) What is the voltage given to deflection coil?

34) Explain the role of horizontal & vertical blanking pulses?

35) The service range of TV transmitter is controlled by which parameter?

36) What is ghost image?

37) Which TV broadcasting system is used in India?

38) If the width of picture tube is 20 inch, what is it's size?

39) Which type of focusing is used in TV receiver?

40) How many frames are transmitted per second in TV system?

41) What are the different types of picture tube distortions?

42) List out the TV receiver antennas.

43) What are the different types of camera tubes & picture tube?

44) What is persistence of vision?

45) What is ion trap?

46) What is the role of acquadag coating?

47) What are CCIR standards for TV.?

48) What are the three primary colours used in TV?

49) What is the role of colour killer?

50) What is the voltage generated by EHT?

5. Conduction of Viva-Voce Examinations:

Teacher should conduct oral exams of the students with full preparation. Normally, the objective questions with guess are to be avoided. To make it meaningful, the questions should be such that depth of the students in the subject is tested. Oral examinations are to be conducted in co-cordial environment amongst the teachers taking the examination. Teachers taking such examinations should not have ill thoughts about each other and courtesies should be offered to each other in case of difference of opinion, which should be critically suppressed in front of the students.

6. Evaluation and marking system:

Basic honesty in the evaluation and marking system is absolutely essential and in the process, impartial nature of the evaluator is required in the examination system. It is a wrong approach or concept to award the students by way of easy marking to get cheap popularity among the students, which they do not deserve. It is a primary responsibility of the teacher that right students who are really putting up lot of hard work with right kind of intelligence are correctly awarded.

The marking patterns should be justifiable to the students without any ambiguity and teacher should see that students are faced with just circumstances.