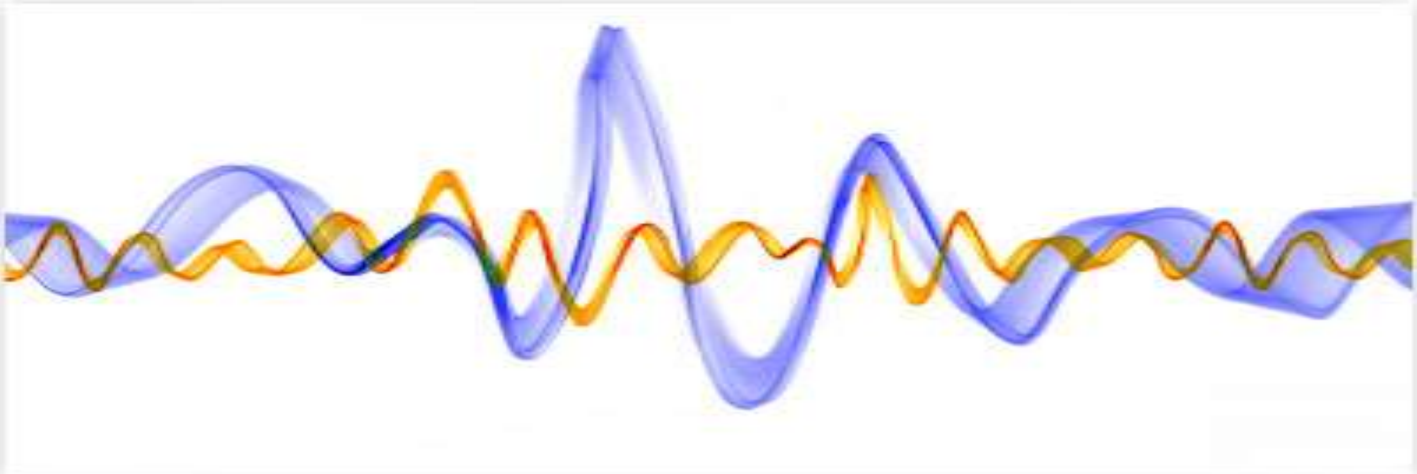

SWEDISH COLLEGE OF ENGINEERING & TECHNOLOGY

Communication Systems

Lab Manual

Submitted by:…………………………………………

Roll No.:………………………………………………

Board Roll No.:………………………………………

Submitted to: Ahmad Bilal

Ahmad Bilal
Communication Engineering

COMMUNICATION SYSTEMS

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UNDERSTANDING SAMPLING

Trainer Specifications

TECHNICAL SPECIFICATIONS

Audio Generator	Sine Wave output with Frequency & Amplitude Adjustable
Sampling Frequency	16KHz, 8KHz, 4KHz, 1KHz with Channel selector
Clock	4.096MHz
Duty Cycle Control	Adjustable Duty Cycle using Decimal Rotary Switch
Sampling Generator	Sampling / Sample & Hold Output
Low Pass Filter	Dual Low Pass Filter (5^{th} Order Butterworth)
Audio Amplifier	Audio Amplifier for Regenerated Audio Signal

THEORY

The aim of any communication system is to transmit information from one location to another. In case of voice communication, this information will be speech.

The signal which contains the information to be transmitted is known as information signal and in the case of voice communication this will be a continuously changing signal containing speech information. The aim is to reproduce this information signal as accurately as possible, at the distant, receiving end of the communication system.

In the exercises to follow, you will simulate audio signal by a 1 KHz test signal provided On-board. The repetitive, non-changing (in amplitude, frequency or phase), waveform does not contain information, but that does not mean we cannot use it. Provided the frequency of the test-signal lies within the frequency range which an information signal will occupy, a test signal of this type can be extremely helpful in system analysis and testing.

The voice signals are limited to the range 300 Hz to 3.4 KHz, A 1 KHz frequency fits conveniently in this range and can be used to demonstrate and test many techniques used in communications.

Theory of Sampling:

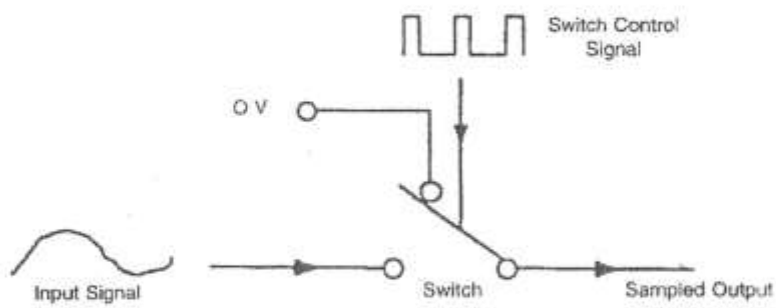
In analog communication systems like AM, FM, the instantaneous value of the information signal is used to hang certain parameter of the carrier signal.

Pulse modulation systems differ from these systems in a way that they transmit a limited number of discrete states of a signal at a predetermined time; sampling can be defined as measuring the value of an information signal at predetermined time intervals. The rate at which the signal is sampled is known as the sampling rate or sampling frequency.

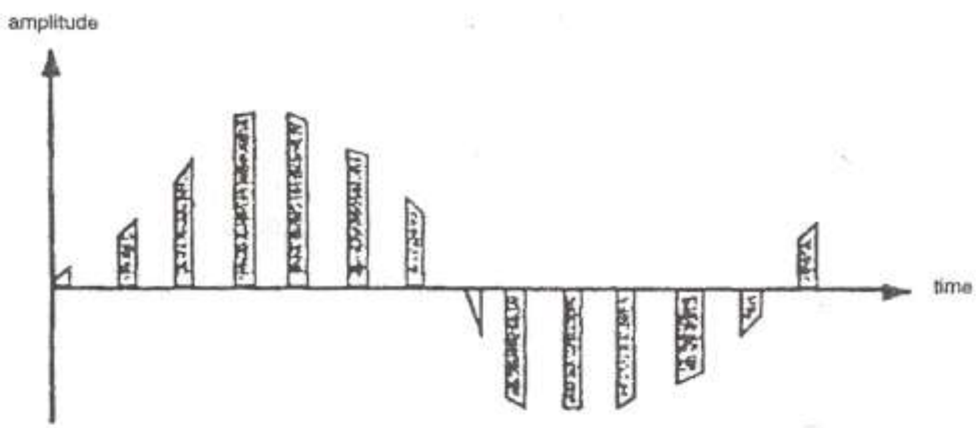
It is the major parameter which decides the quality of the reproduced signal. If the signal is sampled quite frequently (whose limit is specified by Nyquist Criterion), then it can be reproduced exactly at the receiver with no distortion.

Pulse Amplitude Modulation

The pulse amplitude modulation (PAM) system is analog system where the train of Pulses corresponding to the samples of each signal are modulated in amplitude in accordance with the signal itself i.e. the height of the transmitted pulses vary with the amplitude of the message. (Fig.1 & 2)



Basic Sampling Process
Fig.1



Sampled Output
Fig. 2

PAM is an analog system because the amplitude of pulse can vary infinitely i.e. the levels is not discrete.

An information signal sent through an ideal switch which is operated by a control signal, isolated from the information signal, produces a PAM signal. When the switch is open, the voltage is zero; when switch is closed the output voltage is equal to the instantaneous signal voltage. The sample width depends upon how long a switch remains closed.

In practice, electronic switching is used. Sampling by this method is same as multiplying the information signal by a rectangular pulse train. Let us see how the signal can be recovered from PAM signal.

Let the information signal m (t) of highest frequency component F_m (known as the base band signal) is applied to a multiplier along with a train of pulses of unit amplitude, width dt and period Ts. The output of the multiplier is S (t) m (t).

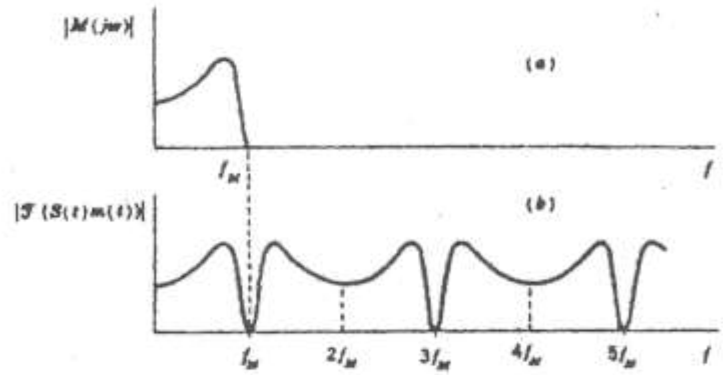
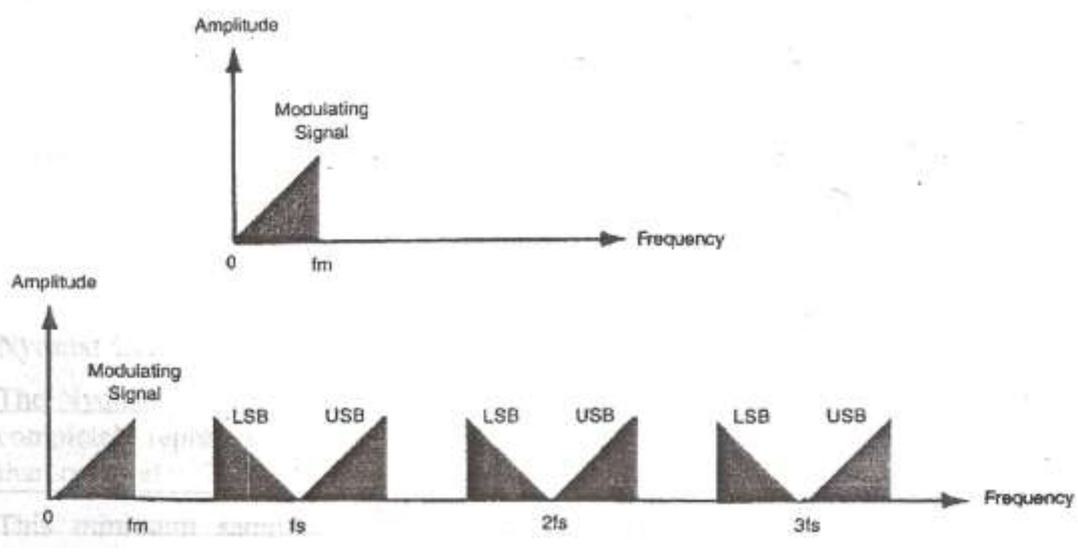


Fig.3

As it can be seen from above Fig.3 the multiplier output has some value as m (t) when the pulse occurs, otherwise it is zero. Rectangular waveforms can represented as summation of Sine/Cosine waveforms of fundamental frequency plus infinite number of harmonics.

Since PAM is amplitude modulation of pulse, we expect the sidebands to be formed around fundamental frequency and each harmonics.

If the sampling frequency is Fs, the frequency spectrum of the PAM signal is as shown below Fig.4.



As it can be seen from the Fig. the spectrum of PAM signal is very much similar to that of AM signal, except the following :

1. The PAM signal contains the spectrum of the base band signal unlike that in and where it is absent it is due to this fact that we can recover the original signal.

2. In AM, a fixed amplitude carrier component is also present at the un-modulated frequency fc. In PAM no such component exists in PAM spectrum.

The information signal can be recovered from the PAM signal by using a low pass filter of cut-off frequency F_M.

Nyquist Criterion

As shown-in the Fig.5 the lowest sampling frequency that can be used without the sidebands overlapping is twice the highest frequency component present in the information signal. If we reduce this sampling frequency even further, the sidebands and the information signal will overlap and we cannot recover the information signal simply by low pass filtering. This phenomenon is known as fold-over distortion or aliasing.

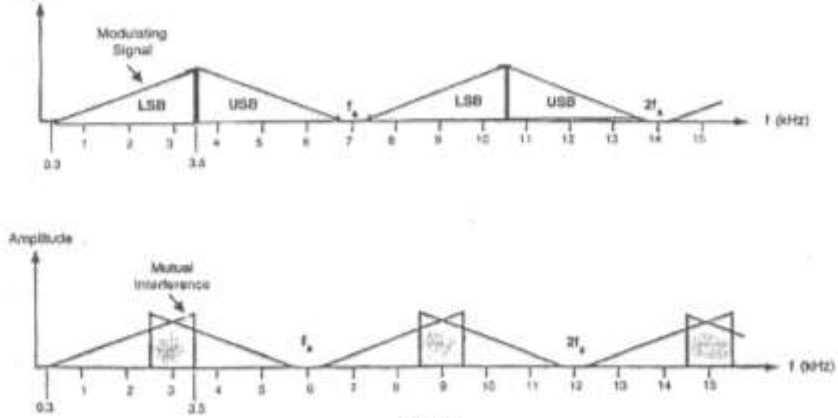


Fig.5

Nyquist Criterion (Sampling Theorem)

The Nyquist criterion states that a continuous signal band limited to F_m Hz can be completely represented by and reconstructed from the samples taken at a rate greater than or equal to $2F_m$ samples/second.

This minimum sampling frequency is called as Nyquist Rate i.e. for faithful reproduction of information signal $fs > 2\ fm$.

Effect of Duty Cycle on Information Recovery

The duty cycle of a, signal is defined as the ratio of pulse duration to the pulse repetition period. This ratio can also be expressed as percentage, e.g. the square wave has equal pulse and no pulse duration, and hence its duty cycle is 0.5 or 50%.

The duty cycle of the sampling pulses is an important parameter in PAM system. They govern the following important aspects:

1. The narrower pulses allow us to time division multiplex many such PAM signals i.e. we can send many no. of PAM signals over same channel at a time. Hence lower duty cycle beneficial in this respect.

2. The narrower pulses has wider frequency spectrum. Hence the wider bandwidth channel is required.

3. Narrower pulses have less power as the power content of a pulse depends on its amplitude and width. During transmission and demodulation the inherent noise can play a major havoc on the low power signal. Hence a pulse of larger duty-cycle is desirous for this sake.

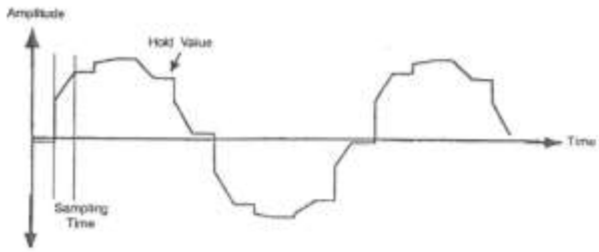
In practice an engineering compromise is made between narrower and broader pulse width taking into account the efficiency, requirement and inherent noise of the system.

Note: The frequency spectrum of PAM signal does not contain those harmonics which when multiplied by duty cycle results in an integer. e.g. the square wave with duty cycle 0.5 (50%), does not contain even harmonics as they result in An integer when multiplied with duty cycle. Thus a square wave sampling signal only contains odd harmonics.

Effect of Sample and Sample / Hold Outputs

If the pulse width of the carrier pulse train used in natural sampling is made very short compared to the pulse period, the natural PAM is referred to as instantaneous PAM As it has been discussed, shorter pulse is desirous for allowing many signals to be included in TDM format but the pulse can be highly corrupted by noise due to lesser signal power.

One way to maintain reasonable pulse energy is to hold the sample value until the next sample is taken. This technique is termed as sample-and-hold techniques. The sample-and-hold waveform looks as shown in Fig.6.

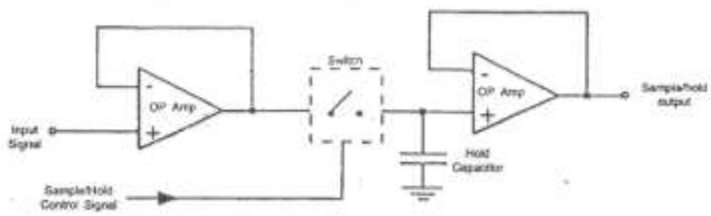


Sample & Hold Waveform
Fig.6

Now, the area under the curve (which is equivalent to the signal power) is greater and so the filter output amplitude and quality of reproduced signal is improved.

The 'hold' facility can be provided by a capacitor when the switch connects the capacitor to PAM output it charges to the instantaneous value.

A buffered sample and hold circuit consists of unit gain buffers preceding and succeeding the charging capacitor. The high input impedance of the preceding buffer prevents the loading of the message source and also ensures that the capacitor charges by a constant rate irrespective of the source impedance see Fig.7.



Sample Hold Circuit

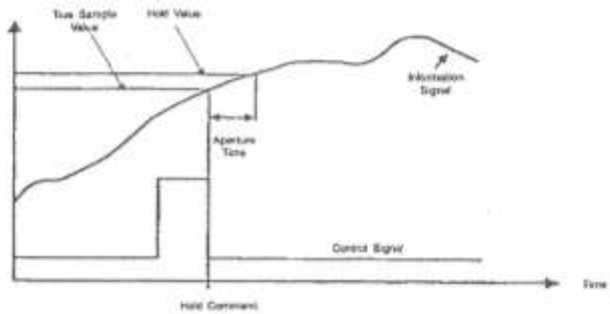
Fig.7

The high input impedance of the succeeding buffers prevents the charge from the capacitor due to loading and hence the capacitor can hold the charge for infinite time, at least theoretically. However, small leakage current through the capacitor dielectric into '+'ve input of second buffer is always present which causes gradual charge loss. The rate of change of voltage with respect to time dv / dt is called as droop rate and is important parameter in sample and Hold circuit design.

Important Parameters of Sample & Hold Circuit

1. Aperture time :

The aperture time is defined as the delay time between the beginnings of the hold command to the time the capacitor voltage ceases to follow the information signal. Hence the hold value is different from the true sample value. The aperture time cannot be reduce to zero because on application of finite time taken by a switch to closet open on application of the hold signal. Therefore a small value of aperture time is sought after.



Timing Diagram for Sample And Hold Circuit

Fig.8

2. Acquisition Time :

In sample mode, it takes finite time for the capacitor to charge to the information signal value depending on the RC time constant. This is called as the acquisition time.The acquisition time is dependent on the current flowing from the input buffer through switch and hence on RC time constant. The maximum acquisition time occurs when the capacitor voltage has to change by the full amplitude of the information signal.

3. Drop Rate :

As it has been discussed earlier, the presence of leakage current through capacitor dielectric to +ve input of succeeding buffer causes charge loss of capacitor. Hence the voltage level at the output falls with time. This rate of change of voltage with respect to time dv/dt is known as droop rate. Over value of droop rate is desirable as the circuit should be able to maintain the sample at a relatively constant level until the next sample.

4. **Feed Through :**

At high frequencies, the stray capacitance within the switch causes some of the input signal to appear at the output during the hold state (switch open). The fraction of input signal appearing at the output of sample and hold circuit is called feed through.

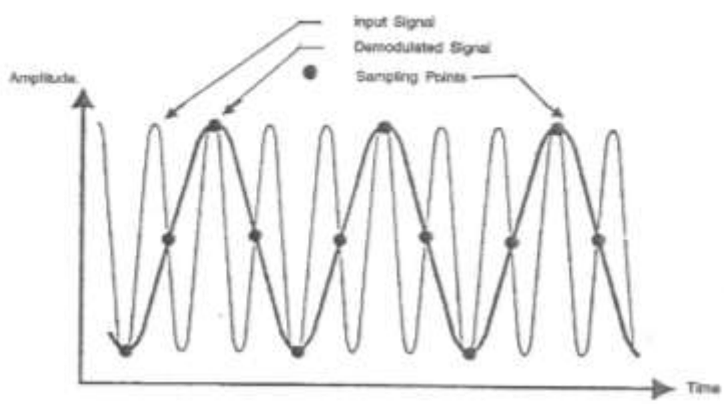
The sample and hold feature provides both problem and benefit will be seen afterwards.

Aliasing :

If the signal is sampled at a rate lower than stated by Nyquist criterion, then there is an overlap between the information signal and the sidebands of the harmonics. Thus the higher and the lower frequency components get mixed and cause unwanted signals to appear at the demodulator output. This phenomenon is turned as aliasing or fold over distortion. The various reasons for aliasing and its prevention are as described.

1. **Aliasing due to Under-Sampling :**

If the signal is sampled at rate lower than $2F_m$ then it causes aliasing. Let us assume a sinusoidal waveform of frequency F_{IN} which is being sampled at rate $F_s < 2F_m$. In the Fig.9 dots represents the sample points.



Aliasing due to Under - Sampling

Fig.9

The low-pass filter at demodulator effectively 'joins' the sample causing an unwanted frequency component to appear at the output. This unwanted component has frequency equal to (F_S-F_M).

2. Aliasing due to wide Band Signal :

The system is designed to take samples at frequency slightly greater than that stated by Nyquist rate. If higher frequencies are ever present in the information signal or it is affected by high frequency noise then the aliasing will occur.

This does not generally happen in properly designed telephone network where speech channels are band-limited by filters before sampling.

In control engineering and telemetry, however, out of band high frequencies either from source or due to noise pick-up can be present. In this case band-limiting filters, generally known as anti-aliasing filters are usually installed prior to sampling to prevent aliasing.

As a principle, the system is designed to sample at rate higher than the rate to take into account the equipment tolerances, ageing and filter response.

3. Aliasing Due to Filter Roll-off :

Roll-off is a term applied to the cut-off gradient of a filter. No filter is ideal and therefore frequencies above the nominal cut-off frequency may still have significant amplitudes at a filter's output. If proper sampling rate and appropriate filter response is not chosen, aliasing will occur.

4. Aliasing due to Noise :

If very small duty cycle is used in sample-and-hold circuit aliasing may occur if the signal has been affected by noise. High frequency noises generally 'mix' with the high frequency component of the signal and hence causes undesirable frequency components to be present at the output.

EXPERIMENT NO 01

UNDERSTANDING OF ST2101

Objective:
Familiarization with ST2101 Module

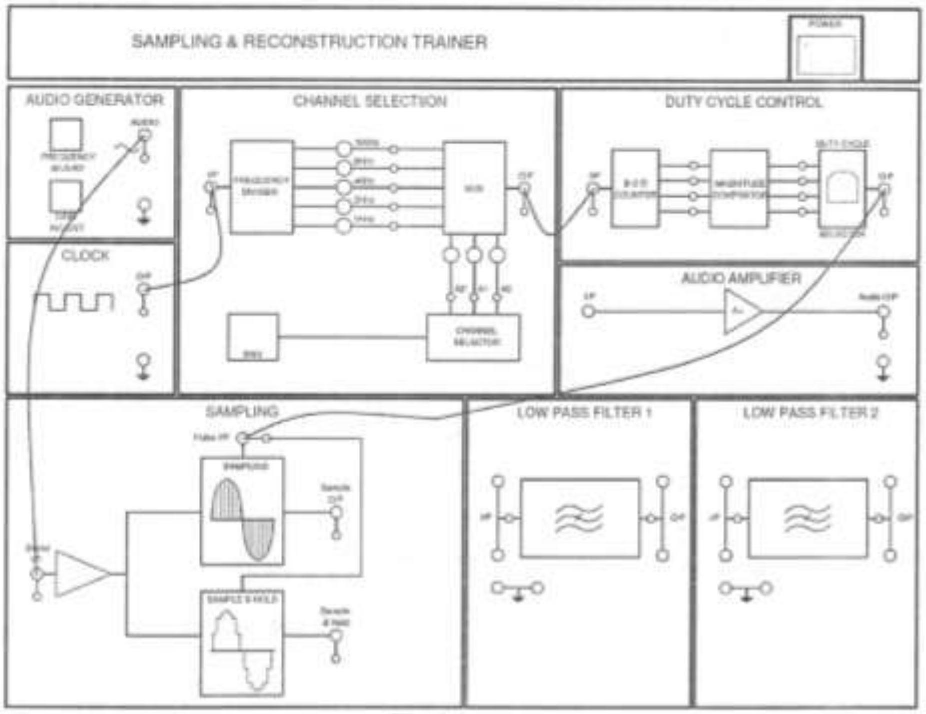


Fig. 1.1

Procedure:

Upon completion of this experiment, students will be familiarized with ST2101 module

Ensure that the following initial conditions exist on the board.

a. Frequency Adj. & Gain Adj. knobs of Audio Generator in center position

b. Amplitude knob of Audio Amplifier is in center position

1. Turn on the power of ST2101 module
2. Connect Oscilloscope probe to Audio O/P test point and examine the waveform. Frequency of this will change by turning the FREQUENCY knob. Amplitude will change by turning the AMPLITUDE knob. Note the minimum and maximum frequency generated by Audio Generator by turning the Frequency Adj. fully CCW and the fully CW respectively.
3. Connect Oscilloscope probe to Clock O/P and verify a clock of 4.096MHz
4. Make connections, using 2mm patch cord, as shown in Fig. 1.1
5. Connect CH1 to 16KHz test point and verify the frequency. Similarly verify other frequencies.
6. Switch SW2 is used to select the Frequency to be output through MUX. Press SW2 and wait for 3 seconds, Channel Selection LED's will indicate the channel selected for Output and the corresponding Frequency LED will also be ON. Verify the Frequency at the MUX O/P with respect to the selected channel.
7. Decimal Rotary switch is used to convert the clock into Pulse. Observe the O/P of Duty Cycle section while rotating the rotary switch.
8. Connect CH1 to Sampling O/P test point. Observe the change in sampling with respect to Duty Cycle selection. Also observe the change in sampling with respect to change in Sampling Frequency Channel.
9. Connect CH1 to Sample & Hold O/P test point. Observe the change in O/P with respect to Duty Cycle selection. Also observe the change in O/P with respect to change in Sampling Frequency Channel.
10. Remove all the patch cord from the trainer and switch off power supply

EXPERIMENT 1: UNDERSTANDING OF ST2101

The first Lab objective is to familiarize you with, apparatus which we will be using throughout this semester

PRE LAB - QUESTIONS

Answer the following Questions. Define and Explain following with formulas

I. **Frequency**

II. **Clock**

III. **Duty Cycle**

IV. **Gain**

V. **Effect of increasing Gain on a signal**

VI. **What is filter?**

VII. **What is low pass filter?**

VIII. **What is high pass filter?**

IX. **What is by pass filter?**

X. **What is the function of duty cycle controller?**

LAB OBSERVATION SHEET

Trainer ST2101

What is the maximum and minimum frequency that can be generated from the Audio Frequency?

What function does Frequency Adjustment and gain knob do?

What does channel selector do?

Write down the status of LED A2 A1 and A0 when following Frequency is selected

A2	A1	A0	Frequency

What is the frequency of the clock generator?

BONUS Question

Can you design a circuit which can perform the same function as Channel Selector?

EXPERIMENT NO 02

STUDY OF SIGNAL SAMPLING

Objective:
Study of Signal Sampling and Reconstruction techniques

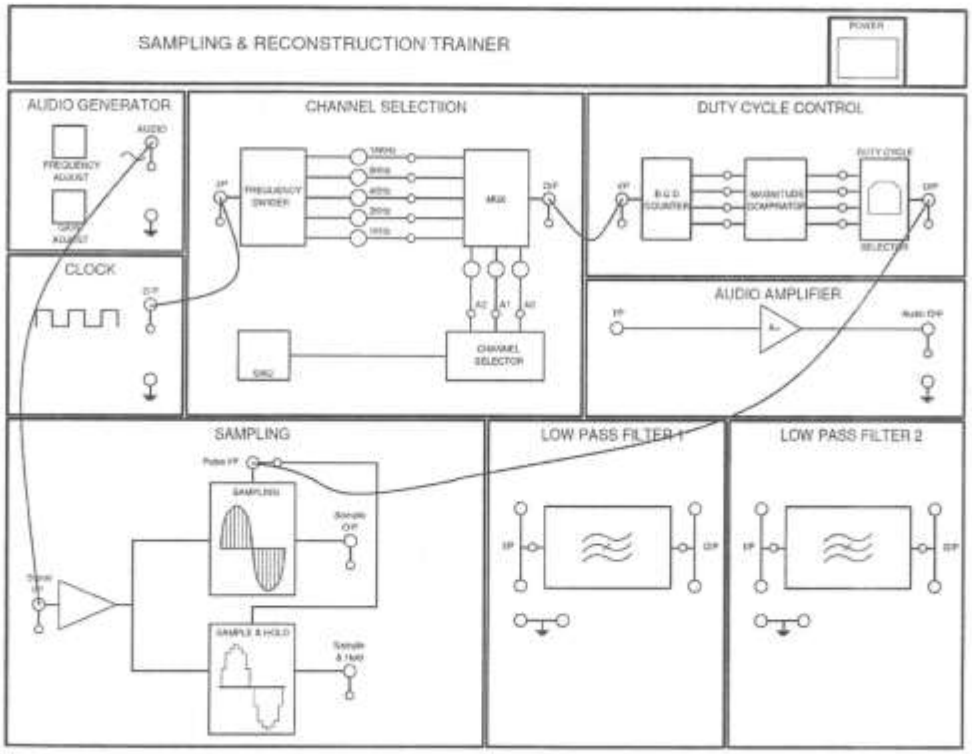


Fig. 2.1

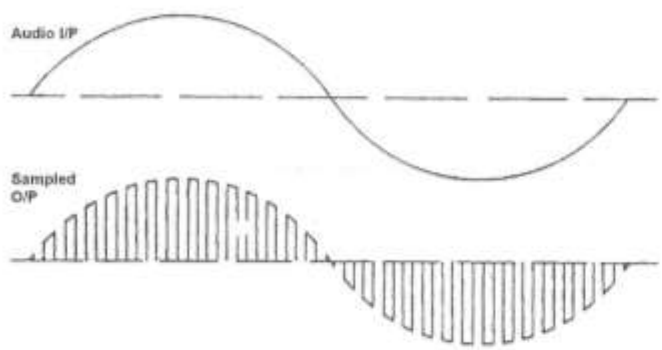
Procedure:

Ensure that the following initial conditions exist on the board.

c. Frequency Adj. & Gain Adj. knobs of Audio Generator in center position

d. Amplitude knob of Audio Amplifier is in center position

1. Turn on the power of ST2101 module
2. Connect Oscilloscope probe to Audio O/P and adjust the frequency to 100Hz
3. Make connections using 2mm patch cords as shown in Fig. 2.1
4. Select the 16KHz channel.
5. Select minimum Duty Cycle
6. Connect CH1 to Sample O/P test point. The display shows 1KHz. Sine wave being sampled at 16 KHz.



Sampled Signal

Fig. 2.2

7. By successive presses of Frequency Selector switch, change the sampling frequency to 2KHz, 4KHz, 8KHz, and back to 16KHz. Observe how SAMPLE output changes in each case.
8. Connect Sampled O/P to LPF 1 I/P and connect CH2 to LPF O/P. Change the sampling frequency by pressing Frequency Selector switch and observe how at lower sampling frequencies introduce distortion into the filter output waveform.
9. Now connect LPF 1 O/P to LPF 2 I/P. Connect CH1 to Signal I/P test point and connect CH2 to LPF 2 O/P and observe a clear Audio signal.
10. Remove all the patch cord from the trainer and switch off power supply

EXPERIMENT 2: STUDY OF SIGNAL AND SAMPLING

PRE LAB - QUESTIONS

Answer the following Questions. Define and Explain following with formulas

I. **Define Nyquist Theorem**

II. **Define aliasing**

III. Sketch a signal in matlab and sample it a frequency as following

   a. Sampling frequency < Signal frequency
   b. Sampling frequency = Signal frequency
   c. Sampling frequency > twice of signal frequency

Reconstruct the signal and discuss the results: Cut and Paste your results below : Every diagram should contain the name of the student in diagram label

---

LAB *OBSERVATION SHEET*

Draw and Label a schematic diagram of circuit with which you have performed the experiment

Fill out the following table

Signal	Value
Audio Generator MAX Freq	
Audio Generator Min Freq	
Audio Generator Minimum Amplitude	
Frequency of Clock	
Amplitude of clock	
Result of OP signal when Sampled with 2 Khz Frequency Amplitude	
Result of OP signal when Sampled with 4 Khz Frequency Amplitude	
Result of OP signal when Sampled with 8 Khz Frequency Amplitude	
Result of OP signal when Sampled with 16 Khz Frequency Amplitude	

Does the received signal have double polarity or single polarity

What output is given if OP signal is feeded to Low pass filter 1

What is the function of duty cycle module here .

EXPERIMENT NO 03

STUDY OF SAMPLE & HOLD SIGNAL

Objective:

1. Study the effect of Sample & Hold circuitry on reconstructed waveform
2. Effect of sampling pulse duty cycle on the reconstructed waveform in sample and sample hold output

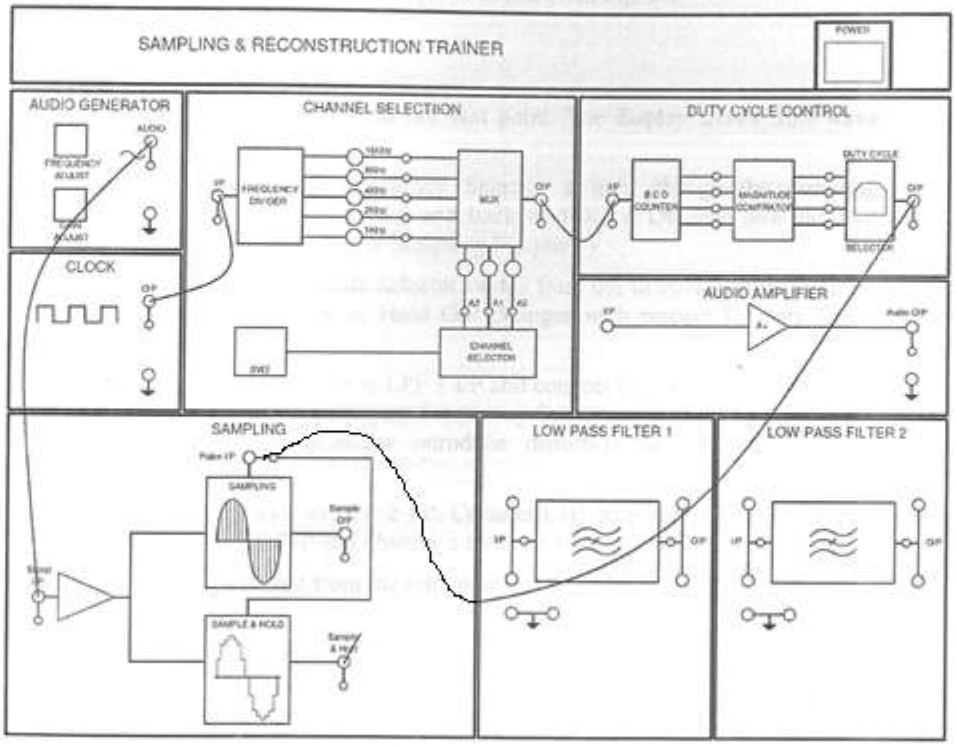


Fig. 3.1

Procedure:

Ensure that the following initial conditions exist on the board.

a. Frequency Adj. & Gain Adj. knobs of Audio Generator in center position

b. Amplitude knob of Audio Amplifier is in center position

1. Turn on the power of ST2101 module
2. Connect Oscilloscope probe to Audio O/P and adjust the frequency to 100Hz
3. Make connections using 2mm patch cords as shown in Fig. 3.1
4. Select the 16KHz channel
5. Select minimum Duty Cycle
6. Connect CH1 to Sample & Hold O/P test point. The display shows Sine wave being Sample & Hold at 16 KHz.

PULSE MODULATION

Trainer Specifications

Modulation Types:
Pulse Amplitude Modulation
Pulse Position Modulation
Pulse Width Modulation

Sampling Frequency:
8 KHz, 16 KHz, 32 KHz, 64 KHz

Waveform Generator
Sine wave

Low Pass Filter
5^{th} order Butterworth Filter

Audio Communication
Dynamic Mic & Speaker

AC Amplifier
DC Variable Level

Interconnections
2mm Gold plated pins

Power Supply
230 V +/- 10%, 50 Hz

Trainer Specifications

Modulation Types:
Pulse Amplitude Modulation
Pulse Position Modulation
Pulse Width Modulation

Sampling Frequency:
8 KHz, 16 KHz, 32 KHz, 64 KHz

Waveform Generator
Sine wave

Low Pass Filter
5^{th} order Butterworth Filter

Audio Communication
Dynamic Mic & Speaker

AC Amplifier
DC Variable Level

Interconnections
2mm Gold plated pins

Power Supply
230 V +/- 10%, 50 Hz

Pulse Modulation Methods

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

Analog-over-analog methods:

- Pulse-amplitude modulation (PAM)
- Pulse-width modulation (PWM)
- Pulse-position modulation (PPM)

Analog-over-digital methods:

- Pulse-code modulation (PCM)
  - Differential PCM (DPCM)
  - Adaptive DPCM (ADPCM)
- Delta modulation (DM or Δ-modulation)
- Sigma-delta modulation (∑Δ)
- Continuously variable slope delta modulation (CVSDM), also called Adaptive-delta modulation (ADM)
- Pulse-density modulation (PDM)

Pulse Amplitude Modulation (PAM)

In pulse-amplitude modulation (PAM) the amplitude of a train of constant-width pulses is varied in proportion to the sample values of the modulating (message) signal. The pulses are usually spaced at equal time interval. Pulse amplitude modulation, the simplest form of pulse modulation, is illustrated in Fig 2.

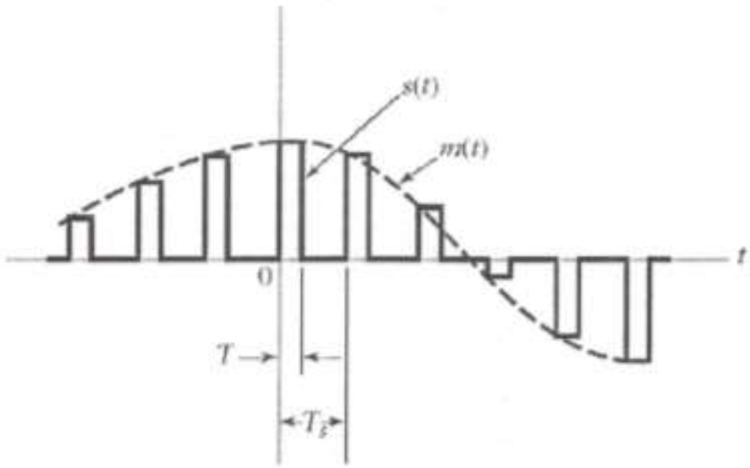


Fig. 2

It is very easy to generate and demodulate pulse amplitude modulation. In a generator, the signal to be converted to Pulse Amplitude Modulation is fed to one input of an AND gate. Pulses at the sampling frequency are applied to the other input of the AND gate to open it during the wanted time intervals. The output of the gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant. The pulses are then passed through a pulse shaping network, which gives them flat tops.

Sample & Hold Output:

The purpose of a sample and hold macro is to quickly store the amplitude of a sampled input waveform, and then to maintain that amplitude until the next sampling pulse. These parts are used in such cases as sampled-data filters or in taking an analog signal that is to be processed by digital circuitry. For example, the sample and hold circuit would hold the value of an input waveform until it takes the next sample. During the hold time, the held voltage would then be converted into an equivalent digital signal for processing by the digital circuitry.

The Sample-and-Hold waveform looks as shown under Fig 3.

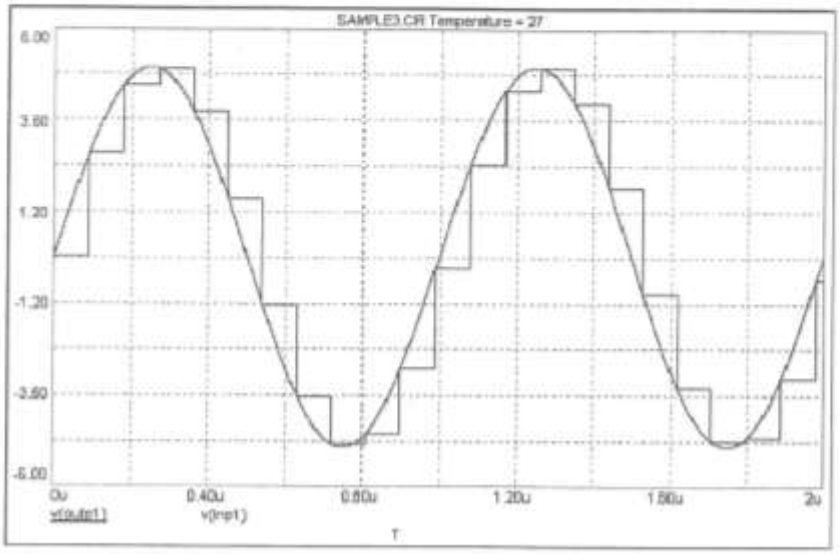


Fig. 3 Sample & Hold Waveform

Now the area under the curve (which is equivalent to the signal power) is greater and so the filter output amplitude and quality of reproduced signal is improved. The hold facility can be provided by a capacitor. When the switch connects the capacitor to pulse amplitude modulation output it changes to the instantaneous value. A buffered Sample and Hold circuit consists of unity gain buffers preceding and succeeding the charging capacitor. The high input impedance of the proceeding buffer prevents the loading of the message source and also ensures that the capacitor charges by a constant rate irrespective of the source impedance. See Fig 4.

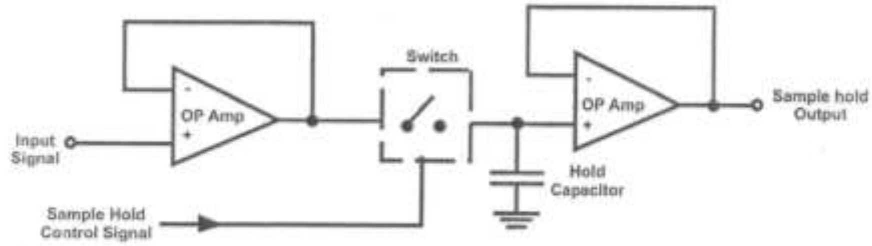


Fig. 4 Sample/Hold Circuit

The high input impedance of the succeeding buffers prevents the charge from the capacitor due to loading and hence the capacitor can hold the charge for infinite time, at least theoretically. However, small leakage current through the capacitor dielectric into positive input of second buffer is always present which causes gradual charge loss. The rate of change of voltage with respect to time dv/dt is called as droop rate is important parameter is sample and Hold circuit design.

Important Parameters of Sample & Hold Circuit:

1. Aperture time: The aperture time is defined as the delay time between the beginnings of the hold command to the time the capacitor voltage ceases to follow the information signal. Hence the hold value is different from the true sample value. The aperture time cannot be reducing to zero because on application of finite time taken by a switch to close/open on application of the hold signal. Therefore a small value of aperture time is sought after. See Fig 5.

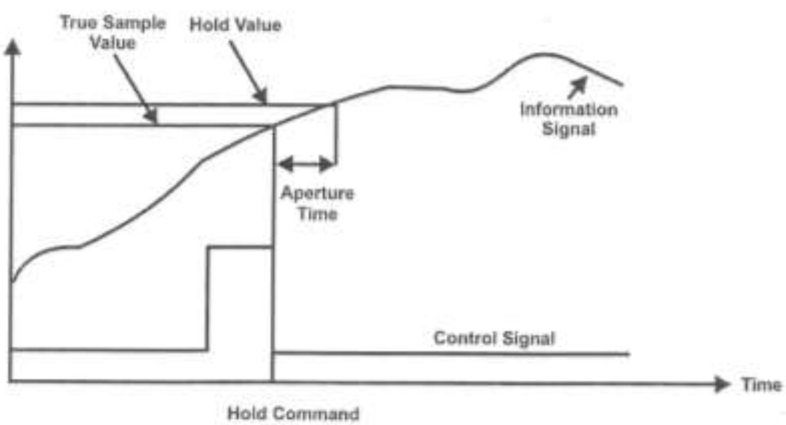
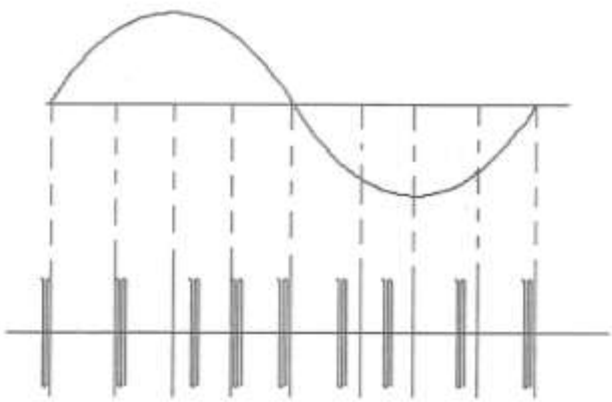


Fig. 5

Pulse Position Modulation

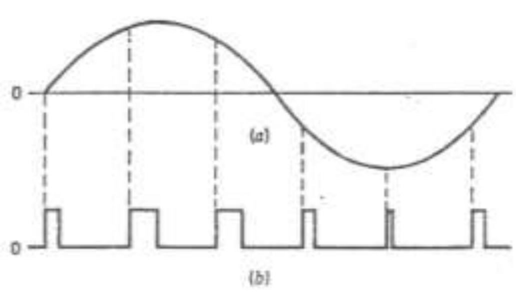
The Amplitude and width of the pulses is kept constant in this system, while the position of each pulse, in relation to the position of a recurrent reference

Pulse is varied by each instantaneous sampled value of the modulating wave. As mentioned in connection with pulse width modulation, pulse-position modulations has the advantage of requiring constant transmitter power output, but the disadvantages of depending on transmitter receiver is synchronization.



Pulse Width Modulation

In pulse width modulation of pulse amplitude modulation is also often called PDM (pulse duration modulation) and less often, PLM (pulse length modulation). In this system, as shown in Fig 6, we have fixed amplitude and starting time of each pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant.



Pulse width modulation. (a) Signal (b) PWM (width variations exaggerated)
Fig. 6

In Fig 6, there may be a sequence of signal sample amplitudes of (say) 0.9, 0.5, 0 and -0.4V. These can be represented by pulse widths of 1.9, 1.5, 1.0 and 0.6μs respectively. The width corresponding to zero amplitude was chosen in this system to be 1.0μs, and it has been assumed that signal amplitude at this point will vary between the limits of + 1 V (width = 2μs) and -1 V (width = 0μs). Zero amplitude is thus the average signal level, and the average pulse width of 1μs has been made to correspond to it. In this context, a negative pulse width is not possible. It would make the pulse end before it began, as it were, and thus throw out the timing in the receiver. If the pulses in a practical system have a recurrence rate of 8000 pulses per second, the time between the commencements of adjoining pulses is 10^6 /8000 = 125μs. This is adequate not only to accommodate the varying widths but also to permit time-division multiplexing. Pulse width modulation has the disadvantage, when compared with pulse position modulation, which will be treated next, that its pulses are of varying width and therefore, of varying power content. This means that the transmitter must be powerful enough to handle the maximum-width pulses, although the average power transmitted is perhaps only half of the peak power. On the other hand, pulse width modulation still works if synchronization between transmitter and receiver fails, whereas pulse-position modulation does not, as will be seen.

EXPERIMENT NO 04

Objective:
To Study Pulse Amplitude Modulation using Natural & Flat top Sampling

Procedure:

1. Connect the circuit as shown in Fig 16.
   a. Output of sine wave to modulation signal IN in PAM block keeping the sine wave nearly 1 KHz.
   b. 8 KHz pulse output to pulse IN.
2. Switch ON the power supply.
3. Monitor the outputs at given test points these are natural flat top outputs - respectively.
4. Observe the difference between the two outputs and try giving reasons behind them.
5. Try Varying the amplitude & frequency of sine wave by amplitude pot and frequency change over switch. Observe the effect on all the two outputs.
6. Also, try varying the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 KHz in Pulse output block.
7. Switch Off the power supply.

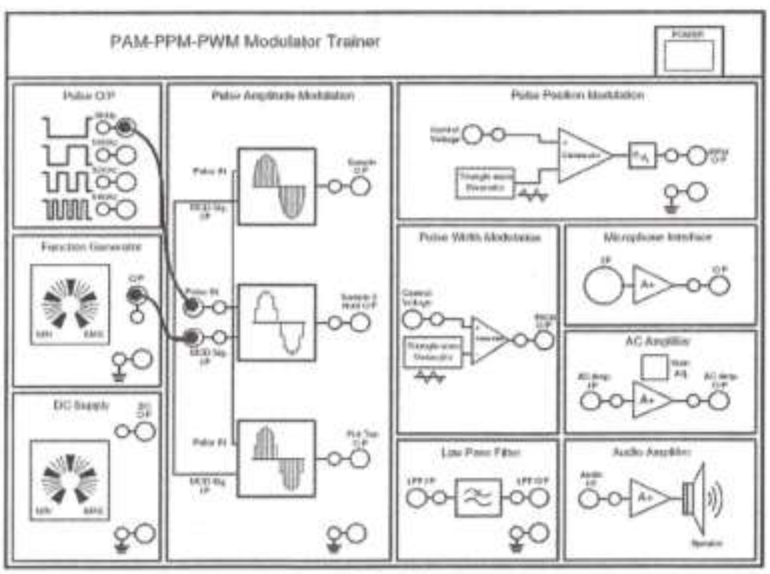


Fig. 16

EXPERIMENT 4: PAM USING NATURAL AND FLAT TOP SAMPLING

PRE LAB - QUESTIONS

Answer the following Questions. Define and Explain following with formulas

I. **Define PAM**

II. **Discuss advantages and disadvantages of PAM using natural and flat top sampling**

LAB OBSERVATION SHEET

Draw and Label a schematic diagram of circuit with which you have performed the experiment

What are the rate of different Pulse signals available on trainer

What is the maximum Frequency that can be generated via frequency generator on Trainer

What is the maximum output that is given by DC Supply

What are the three type of Pulse amplitude modulation techniques available on trainer

Using the same experiment, sketch the output received via flat and natural PAM options on trainer. Don't forget to mention input parameters and output parameters of signal like frequency amplitude of input and output signal

Which Signal has highest frequencies? Pulse signals or Signal produced via frequency generator?

Looking at the frequencies of both signal which should be message signal and which of the signal can be defined as carrier signal

EXPERIMENT NO 05

Objective:
To Study PAM using Sample & Hold sampling

Procedure:

1. Connect the circuit as shown in Fig 17.
   a. Output of sine wave to modulation signal IN of PAM block.
   b. 8 KHz pulse output to pulse IN of PAM block.
   c. Keep the frequency to 1 KHz .
2. Switch ON the power supply.
3. Monitor the output of sample & hold circuit at given test point.
4. Vary the amplitude of input sine wave & its frequency to nearly 2KHz.
5. Also vary the input pulse's frequency by connecting the pulse input to different Square wave frequencies available on-board viz. 16, 32, & 64 KHz.
6. Switch OFF the power supply.

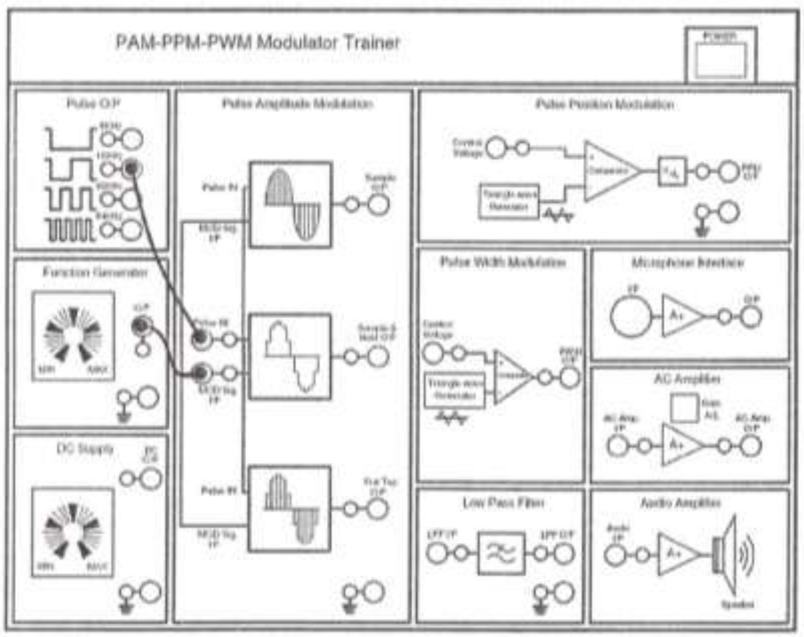


Fig. 17

Communication Lab

EXPERIMENT 5: STUDY PAM USING HOLD AND SAMPLING

PRE LAB - QUESTIONS

Answer the following Questions. Define and Explain following with formulas

I. **Differentiate between sampling and hold sampling**

II. **What is the effect on energy parameter of signal if Hold sampling method is used**

III. **What is the advantage of using Hold sampling**

LAB OBSERVATION SHEET

Draw and Label a schematic diagram of circuit with which you have performed the experiment

If pulse signal shows the carrier signal and the message signal is displayed by the sinusoidal signal , than what is the result of following modulation

Carrier Signal		Message Signal		Modulated Signal	
Frequency	Amplitude	Frequency	Amplitude	Frequency	Amplitude

Using the above table write your observations regarding message and Carrier and modulated signal

EXPRERIMENT NO 06

Objective:
To study Pulse Amplitude Modulation & Demodulation with Sample, Sample & Hold & Flat Top

Procedure:

1. Connect the circuit as shown in Fig 18.
   a. Output of sine wave to modulation signal IN in PAM block keeping the frequency 1 KHz.
   b. 8 KHz pulse output to pulse IN.
   c. Connect the sample output low pass filter input.
   d. Output of low pass filter to input of AC amplifier. Keep the gain pot in AC amplifier block in max position.
2. Follow the steps of experiment 1.

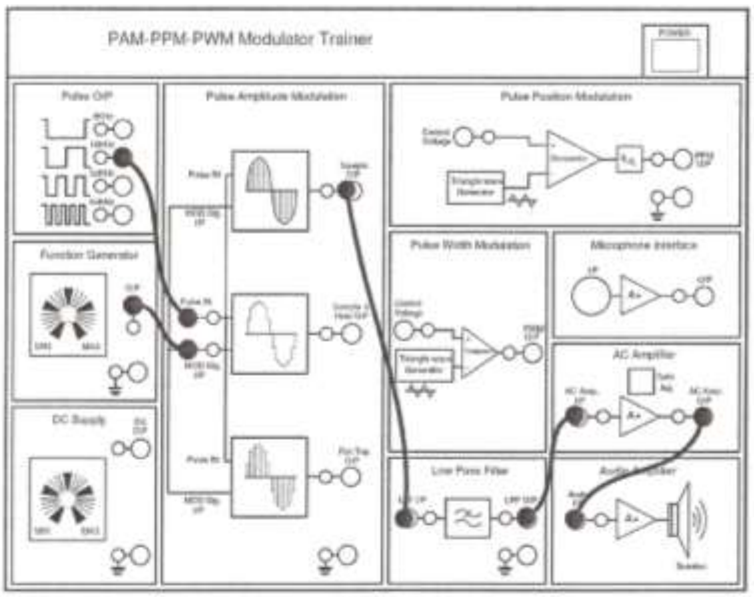


Fig. 18

3. Monitor the output of AC amplifier. It should be a pure sine wave similar to input.
4. Try varying the amplitude of input, the amplitude of output will vary.
5. Similarly connect the sample & hold & flat top outputs to low pass filter and see the demodulated waveform at the output of AC amplifier.
6. Switch OFF the power supply.

EXPERIMENT 6: PAM MODULATION AND DEMODULATION

PRE LAB - QUESTIONS

Answer the following Questions. Define and Explain following with formulas

I. **Draw a circuit which can detect the envelop of AM signal**

II. **What is the function of audio amplifier ?**

III. **What type of low pass filter is being used in trainer? (Answer is available in manual)**

LAB *OBSERVATION SHEET*

If pulse signal shows the carrier signal and the message signal is displayed by the sinusoidal signal , than what is the result of following modulation .

Carrier Signal		**Message Signal**		**Modulated Signal**		**Demodulated Signal**	
Frequency	**Amplitude**	**Frequency**	**Amplitude**	**Frequency**	**Amplitude**	**Frequency**	**Amplitude**

What is the effect of increasing decreasing the amplitude of message signal on Modulated wave and output signal

Draw the block diagram of this experiment and draw signals at every node. Done for get to mention the parameters of signal (Kindly use pencil)

EXPERIMENT NO 07

Objective :
To study PPM using DC Input.

Procedure :

1. Connect the circuit as shown in Fig 19.
    a. Connect the DC output to input of PPM block.
2. Switch ON the power supply.
3. Observe the output of PPM block at given test point.
4. Vary the DC output while observing the output of PPM block.
5. Switch OFF the power supply.

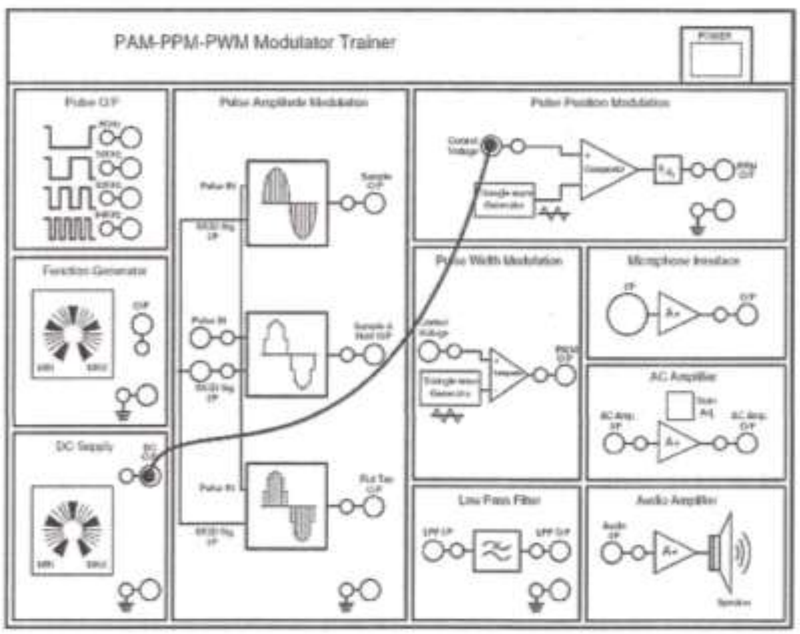


Fig. 19

EXPERIMENT NO 08

Objective:
To study PPM using Sine wave Input

Procedure:

1. Connect the circuit as shown in Fig 20.
   a. Input of pulse position modulation blocks to sine wave output of FG block.
2. Switch ON the power supply.
3. Keep the oscilloscope at 0.5mS / div, time base speed and in X-5 mode, and observe the pulse position modulated waveform at the pulse position modulation block output.
4. Vary the amplitude of sine wave and observe the pulse position modulation, keep the amplitude preset in center. Here you can best observe the pulse modulation.
5. Switch OFF the power supply.

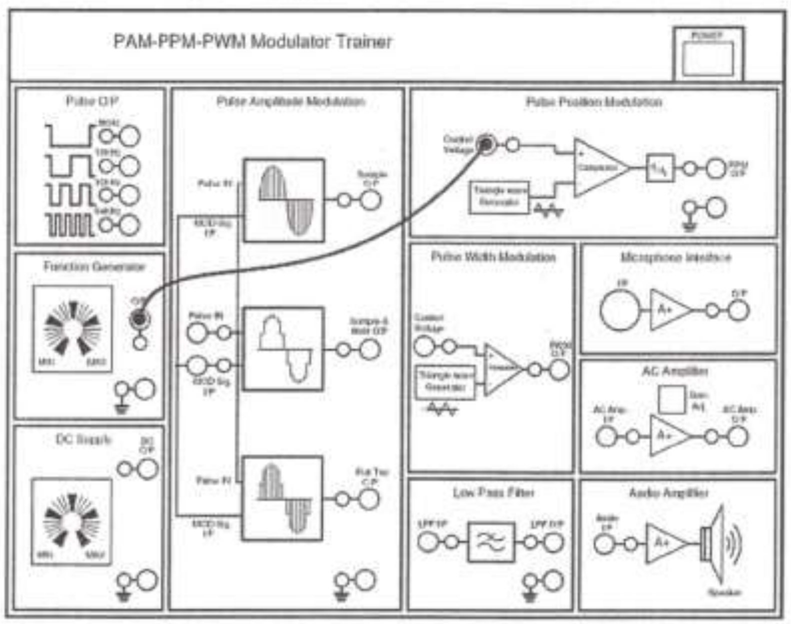


Fig. 20

EXPERIMENT 8 : STUDY PPM USING SINE WAVE INPUT

A very good interactive tutorial is also available on website

PRE LAB - QUESTIONS

Answer the following Questions. Define and Explain following with formulas

I. **Define PPM**

II. **Write some advantages and disadvantages of PPM**

LAB *OBSERVATION SHEET*

If pulse signal shows the carrier signal and the message signal is displayed by the sinusoidal signal , than what is the result of following modulation .

Message Signal		**Modulated Signal**	
Frequency	**Amplitude**	**Frequency**	**Amplitude**

Draw the block diagram of experiment and sketch the wave form of any of the above Experiment.

Which Trainer component/block should be used if we need to demodulate the signal?

EXPERIMENT NO 09

Objective:
To study PPM Demodulation

Procedure:

1. Connect the circuit as shown in Fig 21.
    a. Sine wave of 1 KHz to input of PPM block.
    b. Output PPM block to input of low pass filter.
    c. Output of low pass filter to input of AC amplifier.
    d. Keep the gain pot. In amplifier block at middle position.
2. Switch ON the power supply.
3. Perform experiment 5.
4. Observe the waveform at the given test point output of low pass filter block.
5. Then observe the demodulated output at the given test point output of AC amplifier.
6. Switch OFF the power supply.

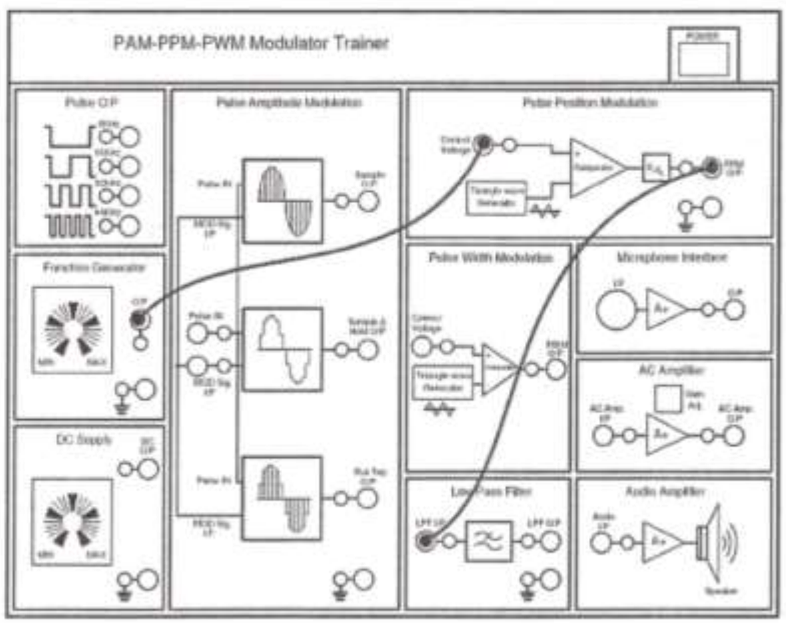


Fig. 21

EXPERIMENT NO 10

Objective:
To study PWM using DC voltage

Procedure:

1. Connect the circuit as shown in Fig 22.
2. Switch ON the power supply.
3. Observe the output of PWM block.
4. Vary the amplitude of DC and see its effect on pulse output.
5. Switch OFF the power supply.

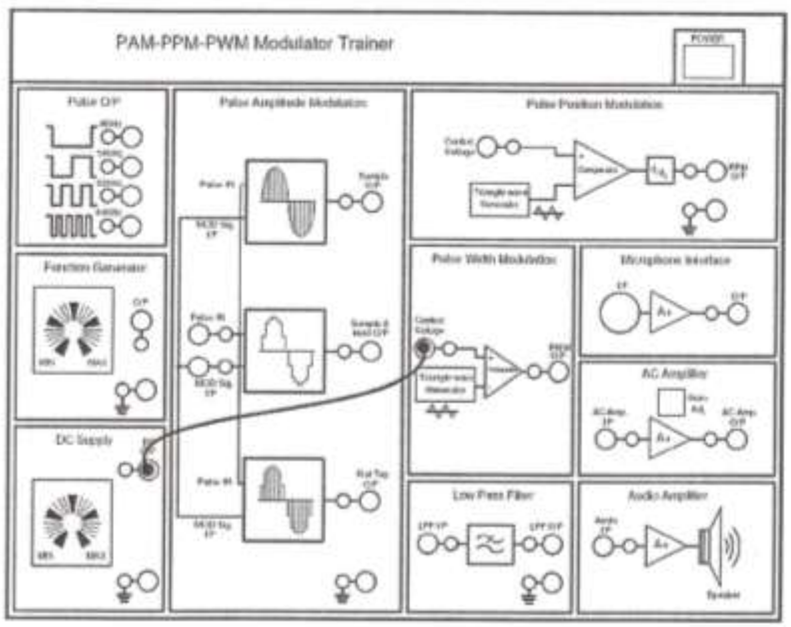


Fig. 22

EXPERIMENT NO 11

Objective:
To study Voice Link Using Pulse Amplitude Modulation

Procedure:

1. Connect the circuit as shown in Fig 23.
    a. Connect a microphone in the MIC socket in audio input block.
    b. Connect the output of audio input block to MOD input of PAM block.
    c. Connect the 8 KHz pulse output to pulse IN of PAM block.
    d. The sample output of PAM block to input of low pass filter.
    e. Output of low pass filter to AC amplifier.
    f. Gain pot of AC amplifier in mid position.
    g. Output of AC amplifier to input of audio output block.
2. Switch ON the power supply.
3. You can observe the pulse being modulated by audio signal at output of sample output, sample & hold & flat top outputs.
4. Also, you can observe its demodulation and hear the same voice in speaker /headphone which was fed in the microphone in the input.
5. Switch OFF the power supply.

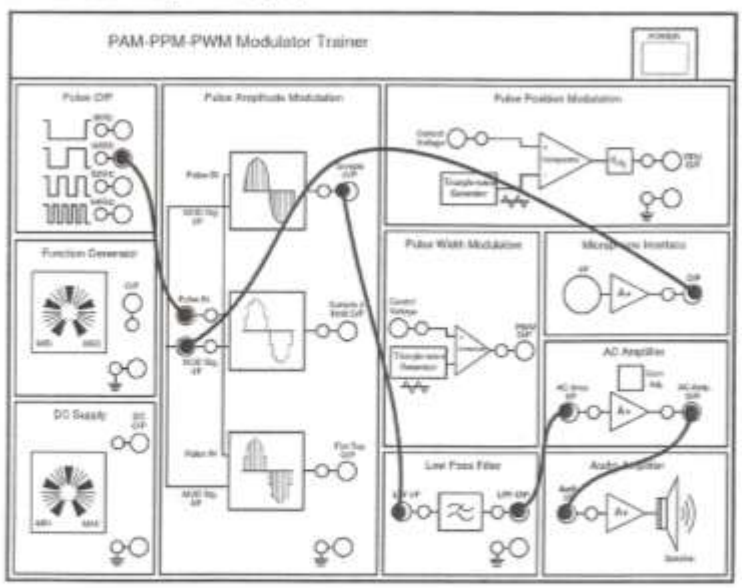


Fig. 23

EXPERIMENT 11: THE FUN PART – THE PAM WAY

1. Have you ever seen your voice on oscilloscope?
2. Have you ever heard your modulated voice?
3. Have you heard your voice after demodulation?

<u>Connect a audio mic to Trainer and see if you can hear your voice via Speaker</u>

<u>Can you see your voice wave form on oscilloscope?</u>

Can you sketch it

Modulate your voice signal and listen to it (Use any flat top or natural PAM) . Also fill the following table Demodulate the signal and listen to your voice again. Can you hear your original voice?

EXPERIMENT NO 12

Objective:
To study Voice Link using Pulse Position Modulation

Procedure:

1. Connect the circuit as shown in Fig 24.
   a. Connect a microphone in MIC socket of audio input block.
   b. Output of audio input block to input of PPM block.
   c. Output of PPM block to input of low pass filter block.
   d. Output of low pass filter block to input of AC amplifier block.
   e. Keep the frequency 1 KHz range.
   f. Keep the gain preset of AC amplifier in mid position.
   g. Connect the output of AC amplifier block to input of audio output block.
2. Switch ON the power supply.
3. You can study the PPM using voice, by observing the waveforms at different stages.
4. The input is heard by means of speaker or headphone.
5. Switch OFF the power supply.

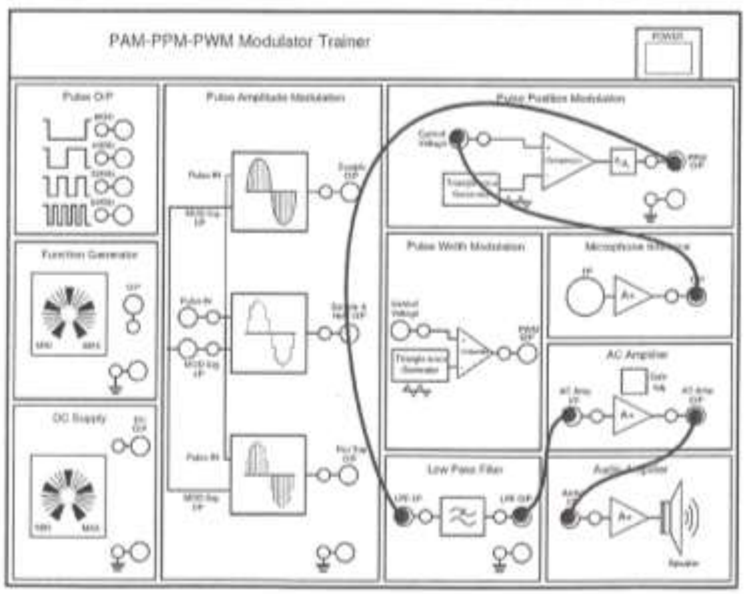


Fig. 24

EXPERIMENT 12: THE FUN PART – THE PPM WAY

1. Have you ever seen your voice on oscilloscope?
2. Have you ever heard your modulated voice?
3. Have you heard your voice after demodulation?

Connect a audio mic to Trainer and see if you can hear your voice via Speaker

Can you see your voice wave form on oscilloscope?

Can you sketch it

Demodulate the signal and listen to your voice again. Can you hear your original voice?

MATLAB EXCERCISES

EXPERIMENT 13 # NYQUIST THEOREM

I. **Sketch a signal in matlab and sample it a frequency as following**
   a. **Sampling frequency < Signal frequency**
   b. **Sampling frequency = Signal frequency**
   c. **Sampling frequency > twice of signal frequency**

Reconstruct the signal and discuss the results: Cut and Paste your results below : Every diagram should contain the name of the student in diagram label

EXPERIMENT 14 # Modulation Index and Amplitude Modulation

AM modulation index basics

Modulation indices are described for various forms of modulation. The amplitude modulation, AM, modulation index can be defined as the measure of extent of amplitude variation about an un-modulated carrier.

As with other modulation indices, the modulation index for amplitude modulation, AM, indicates the amount by which the modulated carrier varies around its static un-modulated level.

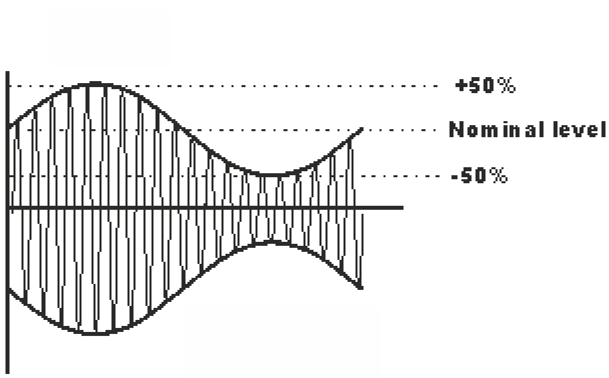
When expressed as a percentage it is the same as the depth of modulation. In other words it can be expressed as:

$$M = \text{(RMS value of modulating signal) / (RMS value of unmodulated signal)}$$

From this it can be seen that for an AM modulation index of 0.5, the modulation causes the signal to increase by a factor of 0.5 and decrease to 0.5 of its original level.

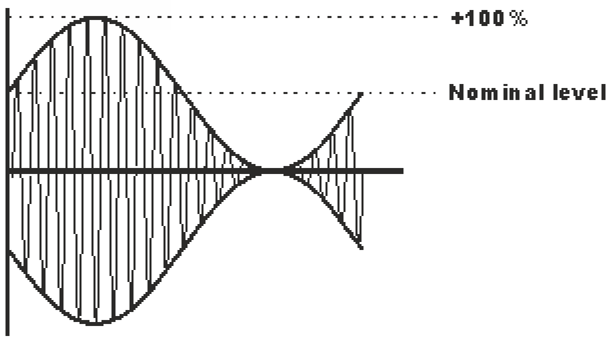
Modulation index / modulation depth examples

Typically the modulation index of a signal will vary as the modulating signal intensity varies. However some static values enable the various levels to visualised more easily.



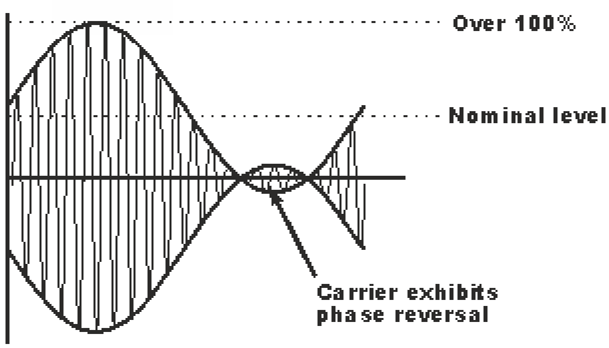
Amplitude modulated index of 0.5

When the modulation index reaches 1.0, i.e. a modulation depth of 100%, the carrier level falls to zero and rise to twice its non-modulated level.



<u>Amplitude modulated index of 1.0</u>

Any increase of the modulation index above 1.0, i.e. 100% modulation depth causes over-modulation. The carrier experiences 180° phase reversals where the carrier level would try to go below the zero point. These phase reversals give rise to additional sidebands resulting from the phase reversals (phase modulation) that extend out, in theory to infinity. This can cause serious interference to other users if not filtered.



<u>Amplitude modulated index of more than 1.0 i.e. over-modulated</u>

Broadcast stations in particular take measures to ensure that the carries of their transmissions never become over modulated. The transmitters incorporate limiters to prevent more than 100% modulation. Hover they also normally incorporate automatic audio gain controls to keep the audio levels such that near 100% modulation levels are achieved for most of the time

PRE LAB - QUESTIONS

What is Amplitude Modulation?

What are the Characteristics of Message signal and Carrier Signal

Represent Amplitude Modulation via Mathematical Expressions

What is modulation index, and what is its effect on Modulation for different values

LAB - QUESTIONS

For the following Amplitude Modulation Code, draw the signal wave forms

```
clc;
clear all;
close all;
t=0:0.001:1;
set(0,'defaultlinelinewidth',2);
A=5;%Amplitude of signal
fm=input('Message frequency=');%Accepting input value
fc=input('Carrier frequency=');%Accepting input value (f2>f1)
mi=input('Modulation Index=');%Modulation Index

Sm=A*sin(2*pi*fm*t);%Message Signal
subplot(3,1,1);%Plotting frame divided in to 3 rows and this fig
appear at 1st
plot(t,Sm);
xlabel('Time');
ylabel('Amplitude');
title('Message Signal');
grid on;

Sc=A*sin(2*pi*fc*t);%Carrier Signal
subplot(3,1,2);
plot(t,Sc);
xlabel('Time');
ylabel('Amplitude');
```

```
title('Carrier Signal');
grid on;

Sfm=(A+mi*Sm).*sin(2*pi*fc*t);%AM Signal, Amplitude of Carrier
changes to (A+Message)
subplot(3,1,3);
plot(t,Sfm);
xlabel('Time');
ylabel('Amplitude');
title('AM Signal');
grid on;
```

Paste the result of AM signal , while keeping Frequency same for all cases, and taking different values of modulation index

On the basis of result, answer the following question

a)The most efficient modulation is carried out when m=?

b)Comment about modulation when m =0, 50, -50