Dr. Supriya

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INSTRUCTION MANUAL
FOR
PUSH-PULL AMPLIFIER

Push-Pull Amplifier Apparatus has been designed to study the Output Gain, Output wer and Frequency Response of a Push Pull Amplifier.

e instrument comprises of the following built in parts :-

DC Regulated power supply of 12V.

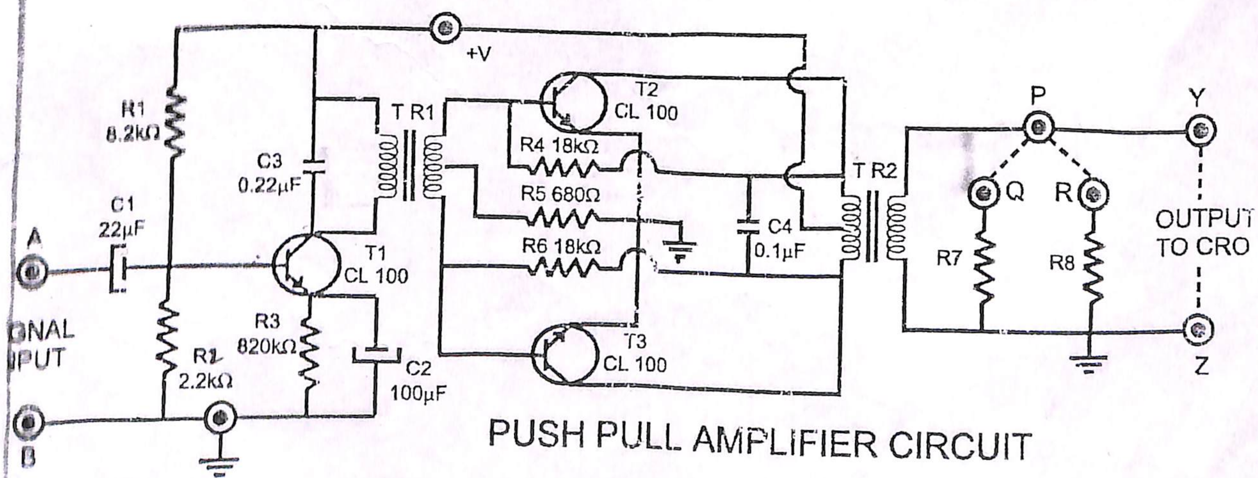
Two L plate (Driver Tranformers) matching transformers.

Three NPN transistors (CL100) are mounted on the front panel & important connections brought out on sockets.

Different types of resistances and capacitors are mounted on the front panel.

THEORY

Push Pull Amplifier is a power amplifier and is frequently employed in the output stages electronic circuits. It is used whenever high output power at high efficiency is required. inted diagram shows the circuit of a push pull amplifier. Two trasistors TR_1 and TR_2 placed ck to back are employed. Both transistors are operated in class B operation i.e collector rrent is nearly zero in the absence of the signal. The centre tapped secondary of driver nsformer T_1 supplies equal and opposite voltage to the base circuit of two transistors. The tput transformer T_2 has the centre-tapped primary winding. The supply voltage Vcc is nnected between the bases and this centre tap. The output load is connected across the condary of this transformer. Input signal appears across the secondary AB of driver



PUSH PULL AMPLIFIER CIRCUIT

ansformer. Suppose during the first half cycle of the signal, end A becomes positive and end negative. This will make the base emitter junction of TR_1 reverse biased and that of Tr_2 rward biased. The circuit will conduct current due to TR_2 only. Therefore, this half cycle of the is amplified by TR_2 only, and appears in the lower half of the primary of the output former. In the next half cycle of the signal, TR_1 is forward biased whearas TR_2 is reverse d. Therefore TR_1 conducts consequently this half cycle of the signal is amplified by TR_1 ppears in the upper half of the output transformer primary. The centre tapped primary of t transformer combines two collector current to form a sine wave output in secondary. It be noted here that push-pull arrangement also permits a maximum transfer of power to ad through impedance matching.

PROCEDURE

Connect Audio Frequency Function Generator across input sockets and set it at sine wave signal of 20mV - 50mV peak to peak amplitude, 1KHz frequency.

Connect CRO across output sockets.

Switch ON the instrument using ON/ OFF toggle switch provided on the front panel.

Observe the amplified output at CRO. Note down the output amplitude.

Calculate the voltage gain of the amplifier, using formula

$$A_V = V_{OUT} / V_{IN}$$

Increase the frequency of the signal towards 100KHz in small steps and note down the voltage gain at differentr frequencies.

Note down the observation in table no. (1) and plot a graph between Voltage Gain vs Frequency

Sr. No.	FREQUENCY	INPUT SIGNAL	OUTPUT SIGNAL	GAIN OUTPUT/ INPUT
1.				
2.				
3.				
4.				
5.				
—				

TABLE No. 1

Connect the load resistance (R7 or R8) across output sockets and calculate the output power by using the formula

$$P = V^2 / R$$ (V is the RMS value of the output signal)

STANDARD ACCESSORIES

One Single point Patchcord.

Instruction Manual.

INSTRUCTION MANUAL
FOR
STUDY OF CLASS 'A', 'B', 'AB' & PUSH PULL AMPLIFIER

Class 'A', 'B', 'AB' & Push Pull Amplifier Circuits has been designed to study the output gain and frequency response of these amplifiers.

The instrument comprises of the following builtin parts:

1. DC Regulated power supply of ±12V & +5V.

2. Four PNP & Four NPN Transistors are mounted on the front panel.

3. Four driver transformers are also mounted on the front panel to perform class B and Push Pull Amplifier experiments.

4. Circuit diagram is printed & components are mounted on the front panel.

THEORY

A practical amplifier always consists of a number of stages that amplify a weak signal until sufficient power is available to operate a loudspeaker or other output device. The first few stages in this multistage amplifier have the function of only voltage amplification. However, the last stage is designed to provide maximum power. This final stage is known as power stage. The term audio means the range of frequencies that we can hear. The range of human hearing extends

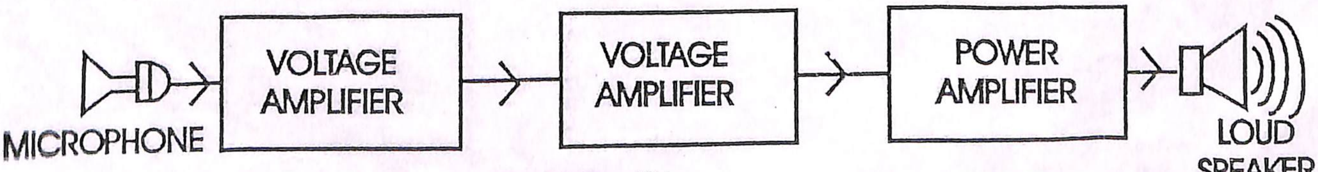


FIG. (1)

from 20Hz to 20kHz. Therefore, audio amplifiers amplify electrical signals that have a frequency range corresponding to the range of human hearing i.e, 20Hz to 20kHz. Figure (1) shows the block diagram of an audio power amplifier. The early stages built up the voltage level of the signal while the last stage built up power to a level sufficient to operate the loudspeaker.

A transistor amplifier which raises the power level of the signals that have audio frequency is known as transistor audio power amplifier. In general, the last stage of a multistage amplifier is the power stage, the amplifier differs from all the previous stages in that here a concentrated effort is made to obtain maximum output power. A transistor that is suitable for power amplification is generally called a power transistor. A power amplifier is required to deliver a large amount of power and such it has to handle large current. In order to achive high power amplification, transformer coupling is used for impedence matching. If the collector current flows at all times during the full cycle of the signal the power amplifier is known as Class 'A' power amplifier. A basic class'A' power amplifier normally consists of a singal transistor, wired in the common emitter mode with the speaker acting as its collector load. The essential feature of this type of amplifier is that its input (Base) is biased so that the collector current takes up a quiescent value roughly halfway between the desired maximum and minimum swings of output current, so that maximum undistorted output signal swings can be obtaind.

The Class 'A' amplifier is simple and produces excellent low distortion audio signal. Its major dis-advantages are that it consumes a high quiescent current and is relatively inefficient. A basic Class 'B' amplifier normally consists of a pair of transistors driven in antiphase but driving a common output load. In this particular design the two transistors are wired in common emitter mode and driven the speaker via Push-Pull transformer. The major advantages of the Class 'B' amplifier are that it consumes near zero quiescent current and has a very high efficiency under all operating conditions. Its major dis-advantages is that it produces high levels of signal distortion. The crossover distortion of the Class 'B' amplifier can be virtually eliminated by applying slight forward bias to the base of each transistor, so that each transistor passes as a modest quiescent current. Such a circuit is known as a Class 'AB' amplifier. Circuits of this type were widely used in early transistor power amplifier systems. The Push-Pull amplifier is a power amplifier and is frequency employed in output stages of electronic circuits. It is used whenever high output power at high effciency is required. Two transistors placed back to back are employed. Both transistor are operated in Class'B' operation i.e, collector current is nearly zero in the absence of the signal. The centre tapped secondary of driver transformer applies equal and opposite volatge to the base circuits of two transistors. The output transformer has the centre tapped primary winding. the supply voltage V_{CC} is connected across the secondary of this centre tap. The loudspeaker is connected across the secondary of this transformer.

NOTE : Circuit diagrams for all the amplifiers are printed on the front panel.

PROCEDURE

FOR CLASS 'A' AMPLIFIER :

1. Connect -12VDC power supply across power supply sockets through patchcords as shown by dotted lines in the circuit diagram of Class 'A' amplifier.

Sr. No.	FREQUENCY	INPUT SIGNAL	OUTPUT SIGNAL	GAIN OUTPUT/ INPUT
1.				
2.				
3.				
4.				
5.				

TABLE No. (1)

2. Connect Audio Frequency Function Generator across input sockets and set it at sine wave signal of 30mV peak to peak amplitude, 100Hz frequency.

3. Connect CRO across output sockets.

4. Switch ON the instrument using ON/ OFF toggle switch provided on the front panel.

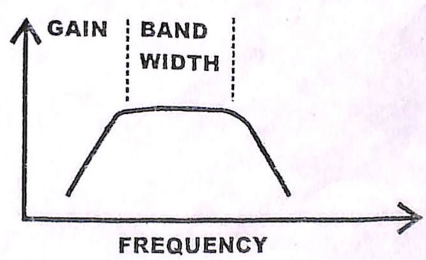
5. Observe the amplified output on CRO. Note down the output amplitude.

6. Calculate the voltage gain of the amplifier, using formula

$$A_V = V_{OUT} / V_{IN}$$

Increase the frequency of the signal towards 100KHz in small steps and note down the voltage gain at different frequencies.

Note down the observation in table no. (1) and plot a graph between Voltage Gain vs Frequency as shown in Fig.



FOR CLASS 'B' AMPLIFIER :

1. Connect -12VDC power supply across power supply sockets through patchcords as shown by dotted lines in the circuit diagram of Class 'B' amplifier.

Sr. No.	FREQUENCY	INPUT SIGNAL	OUTPUT SIGNAL	GAIN OUTPUT/ INPUT
1.				
2.				
3.				
4.				
5.				

TABLE No. (2)

2. Connect Audio Frequency Function Generator across input sockets. Set the Audio Frequency Function generator output to 50mV peak to peak, 10kHz sine wave signal.

3. Connect CRO across output sockets.

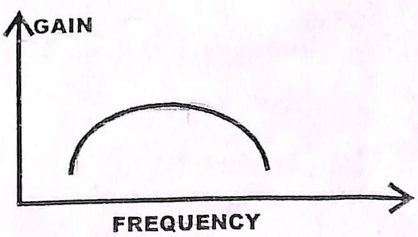
4. Switch ON the instrument using ON/ OFF toggle switch provided on the front panel.

5. Observe the amplified output on CRO. Note down the output amplitude.

6. Calculate the voltage gain of the amplifier, using formula:

$$A_V = V_{OUT} / V_{IN}$$

7. Increase the frequency of the signal towards 100KHz in small steps and note down the voltage gain at different frequencies.



8. Note down the observation in table no. (2) and plot a graph between Voltage Gain vs Frequency as shown in Fig.

FOR CLASS 'AB' AMPLIFIER :

Class 'AB' amplifier is basically a power amplifier, it handies large signal. It amplifies the power level of the signal and does not amplify the volatge level.

Sr. No.	FREQUENCY	INPUT SIGNAL	OUTPUT SIGNAL	GAIN OUTPUT/ INPUT
1.				
2.				
3.				
4.				
5.				

TABLE No. (3)

1. Connect +5VDC power supply across power supply sockets through patchcords as shown by dotted lines in the circuit diagram of Class 'AB' amplifier.

2. Connect Audio Frequency Function Generator across input sockets and set it at sine wave signal of 2V peak to peak amplitude, 10Hz frequency.

3. Connect CRO across output sockets.

4. Switch ON the instrument using ON/ OFF toggle switch provided on the front panel.

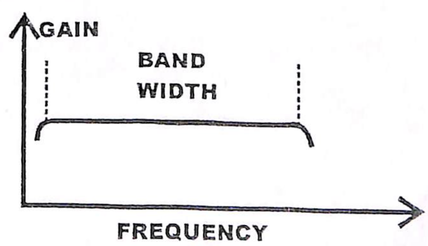
5. Observe the output on CRO. Note down the output amplitude.

6. Calculate the voltage gain of the amplifier, using formula

$$A_V = V_{OUT} / V_{IN}$$

7. Increase the frequency of the signal towards 100KHz in small steps and note down the voltage gain at different frequencies.

8. Note down the observation in table no. (3) and plot a graph between Voltage Gain vs Frequency as shown in Fig.



9. We will observe this amplifier does not amplify the voltage level of the input signal but only the power level of the signal amplifies. If we will connect the loudspeaker load across output socket, the amplifier will drive the loudspeaker.

FOR PUSH PULL AMPLIFIER :

1. Connect +12VDC power supply across power supply sockets through patchcords as shown by dotted lines in the circuit diagram of Push Pull amplifier.

Sr. No.	FREQUENCY	INPUT SIGNAL	OUTPUT SIGNAL	GAIN OUTPUT/ INPUT
1.				
2.				
3.				
4.				
5.				

TABLE No. (4)

2. Connect Audio Frequency Function Generator across input sockets and set it at sine wave signal of 50mV - 100mV peak to peak amplitude, 1.5KHz frequency.

3. Connect CRO across output sockets.

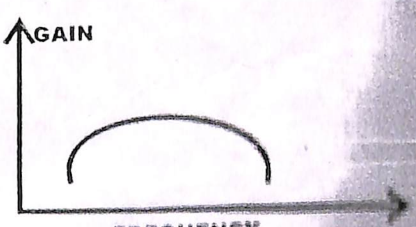
4. Switch ON the instrument using ON/ OFF toggle switch provided on the front panel.

5. Observe the amplified output on CRO. Note down the output amplitude.

6. Calculate the voltage gain of the amplifier, using formula

$$A_V = V_{OUT} / V_{IN}$$

7. Increase the frequency of the signal in small steps and note down the voltage gain at different frequencies.



8. Note down the observation in table no. (4) and plot a graph between Voltage Gain vs Frequency.

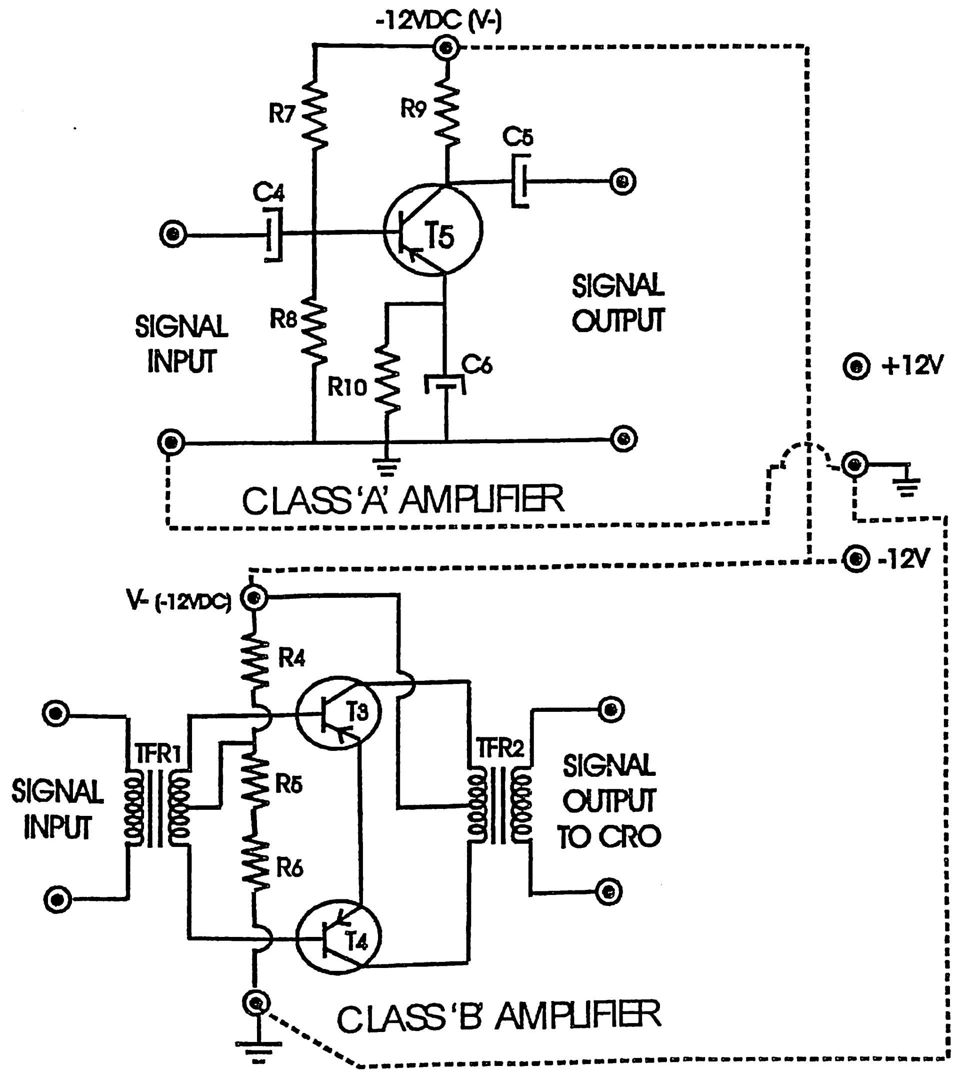
9. Connect the load resistance (R17 or R18) across output sockets and calculate the output power by using the formula:

$$P = V^2 / R$$ (V is the RMS value of the output signal)

STANDARD ACCESSORIES

1. Two Single Point Patchcords for Interconnections.

2. Instruction Manual.



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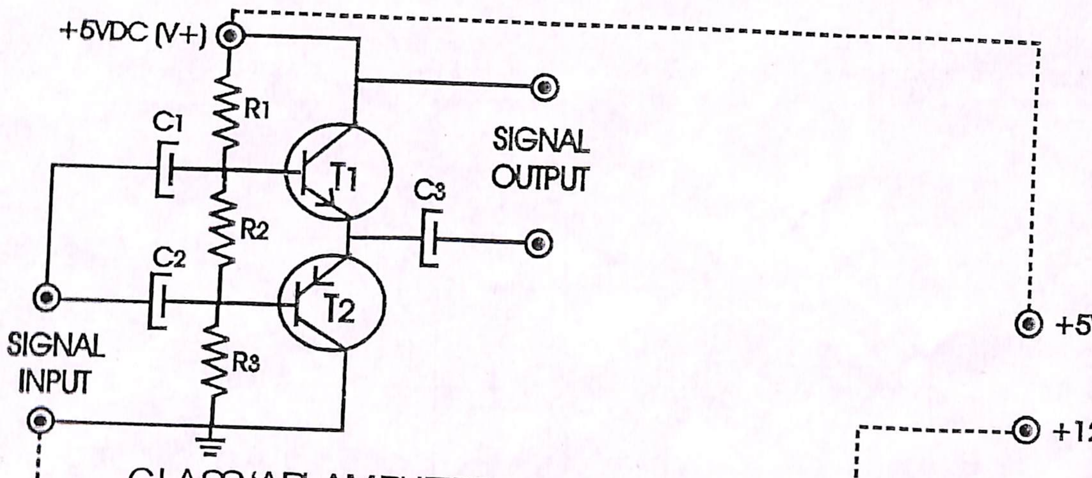
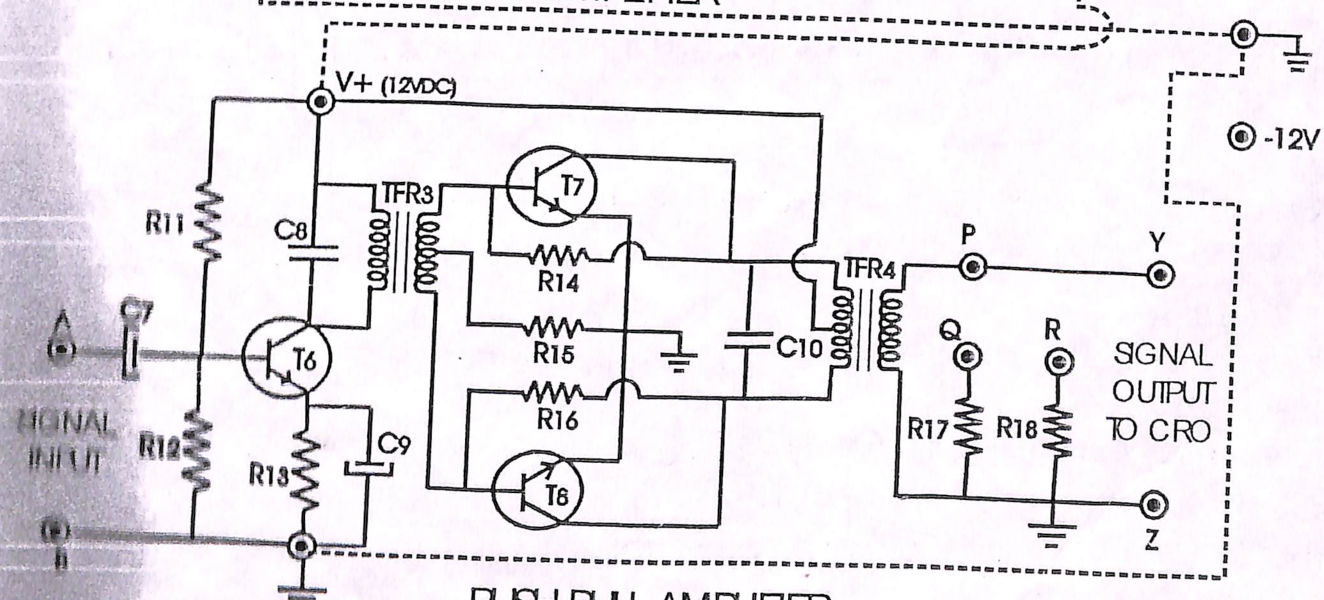


CLASS 'AB' AMPLIFIER



PUSH PULL AMPLIFIER

INSTRUCTION MANUAL
FOR
VOLATGE REGULATOR USING IC 317

Voltage Regulation Circuits using IC 317 has been designed to study the working principle of a voltage regulator.

The instrument comprises of the following built in parts.

1. Step down transformer having 9V-0-9V AC tapes.

2. Rectifier section having two diodes (IN 4007) to convert the AC voltage to DC voltage.

3. Filter section having 2 electrolytic capcitor of 1000μf/ 35V & 1 Inductance.

4. Regulator section consists of IC 317 and combination of resistance & capacitor.

5. Load section having different values of load resistance selectable using Band switch provided on the front panel.

6. Voltmeter and current meter are mounted on front panel to measure DC output voltage, Output current & one AC meter to measure Ripple directly.

THEORY

A voltage regulator maintains the output voltage constant irrespective of a.c. mains fluctuations or load variation. The heart of a voltage regulator is a Zener Diode or Regulator (IC 317). Since Zener Diode or Regulator maintains constant voltage irrespective of their current after breakdown, regulation of voltage can be made available.

In an ordinary power supply, the voltage regulation is poor i,e DC output voltage changes appreciably with load current. Moreover, output voltage also changes due to variations in the input a.c. voltage. A regulated power supply consists of an ordinary power supply and voltage regulating device as shown in circuit diagram. The output of ordinary power supply is fed to the voltage regulator which produces the final output the output voltage (VDC) remains constant whether the load current changes or there are fluctuations in the input A.

PROCEDURE

1 Connect the circuit as shown in Fig 1. Also connects DC voltmeter and current meter in the space provided (shown by dotted line at the front panel).

2. Connect electronic AC voltmeter (1V/ 10Volts) at output to measure the ripple directly.

3. Connect load (R_L) in circuit for measuring DC output current.

4. Switch ON the instrument using ON/ OFF toggle switch provided on the front panel.

5. Note down the observations i.e. DC output voltage, DC current and AC ripples on the meters.

6. Switch ON the toggle switch S1 to connect the capacitor C1 in the circuit again check the DC output voltage, DC current and AC ripples.

7 Switch ON toggle switch S2 so that capacitor C2 also appcars in the circuit. Now the filter circuit is in π type configuration, set the output DC volatge by using potentiometer VR1. Again note down output voltage output current and A.C. ripple.

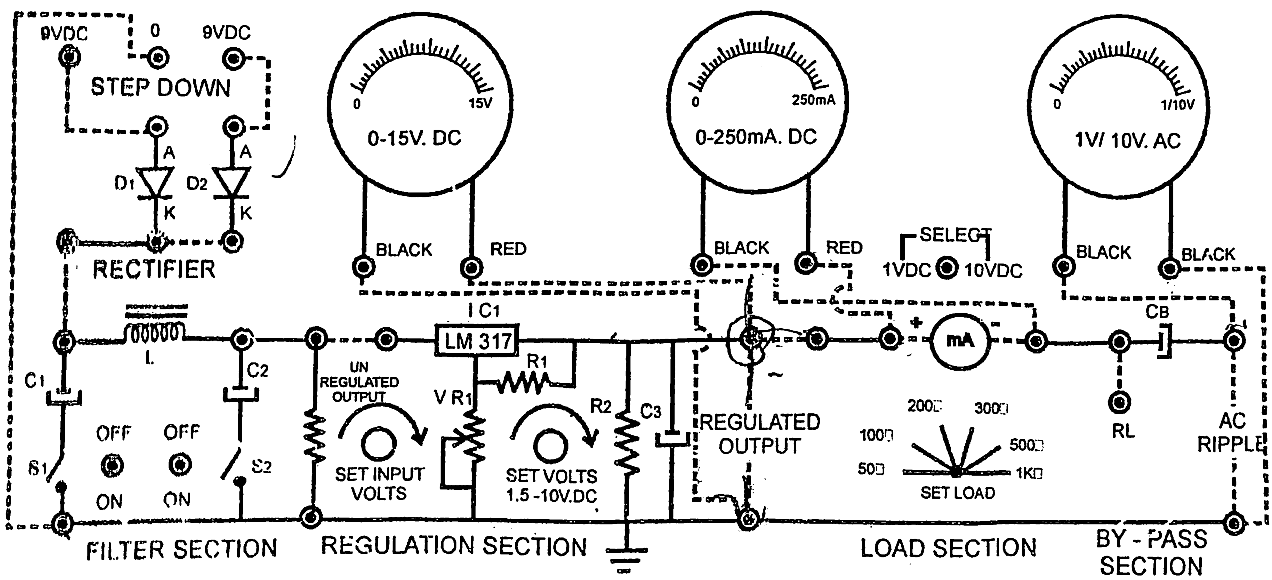
8. Repeat the experiment for different values of load resistances.

STANDERED ACCESSORIES

1. Fourteen single point & one interconnectable patchcords.

2. Instruction Manual.

VOLTAGE REGULATION USING IC 317



INSTRUCTION MANUAL
FOR
DIGITAL COMMUNICATION TRAINER (PAM,PWM,PPM)

PAM,PWM,PPM apparatus has been design to study the following:

1. Pulse Amplitude Modulation & Demodulation
2. Pulse Position Modulation & Demodulation
3. Study the PPM using DC input
4. Study Voice communication using Pulse Amplitude Modulation
5. Study Voice Communication using width Modulation
6. Study Voice communication using Pulse Position Modulation.

INTRODUCTION OF PULSE MODULATION

Pulse Modulation may be used to transmit information such as continuous speech of data. It is a system in which continuous waveforms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times, together with any synchronizing pulses that may be required. At the receiving end, the original wave forms may be reconstituted from the information regarding the samples, if these are taken frequently enough. Despite the fact that information about the signal is not supplied continuously, as in AM and FM, the resulting receiver output can have negligible distortion.

Pulse modulation may be subdivided broadly into two categories. analog and digital. In the former. the indication of sample amplitude is the nearest variable. while in the letter a code. which indicates the sample amplitude to the nearest predetermined level, is sent .Pulse amplitude and pulse time modulation, to be treated next, are both analog

THEORY OF SAMPLING:-

n analog communication system like AM. FM the instantaneous value of the information signal is used change certain parameter of the carrier signal. Pulse modulation system differ from these system in a way that transmit a limited no. of discrete states of a signal at a predetermined time. Sampling can be defined as measuring the value of an information signal at predetermined rate or sampling frequency. It the major parameter. Which decide the quality of the reproduced signal. If the signal is sampled quite frequency (whose limit is specified by Nyquist criterion) then it can be reproduced exactly at the reciver with no distortion.

NYQUIST CRITERION

s shown in the Fig. (1) the lowest sampling frequency that can be used without the side bands overlapping is twice the highest frequency component present in the formation signal. If we reduce this sampling frequency even further. the side bands and information signal will overlap and we can not recover the information signal simply be low pass filtering. This phenomenon is know as fold- over distortion or aliasing.

NYQUIST CRITERION (SAMPLING THEORM)

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

This minimum sampling frequency is calls as a NYQUIST RATE i.e. for faithful reproduction of information signal fs> 2fm.

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 wava has equal pulse and no pulse duration: hence its duty cycle is.5 or 50%

The duty cycle of the sampling of the pulse is an important parameter in pulse Amplitude modulation system they govern the following important aspects.

) The narrower pulses allows us to time division multiplex many such pulse Amplitude modulation panels i.e. we can send many no. of pulse amplitude modulation signals over same channel at a time. Hence lower duty cycle beneficial in this respect.

) The narrower pulses have wider frequency spectrum. Hence the wider bandwidth channel is required

) Narrower pulse 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 majar havoc on the power signal . Hence a pulse of larger duty-cycle is desirous for this sake. In practice an engineering is made between narrower and broader pulse width taking into account the efficiency. requirement and inherent noise of the system.

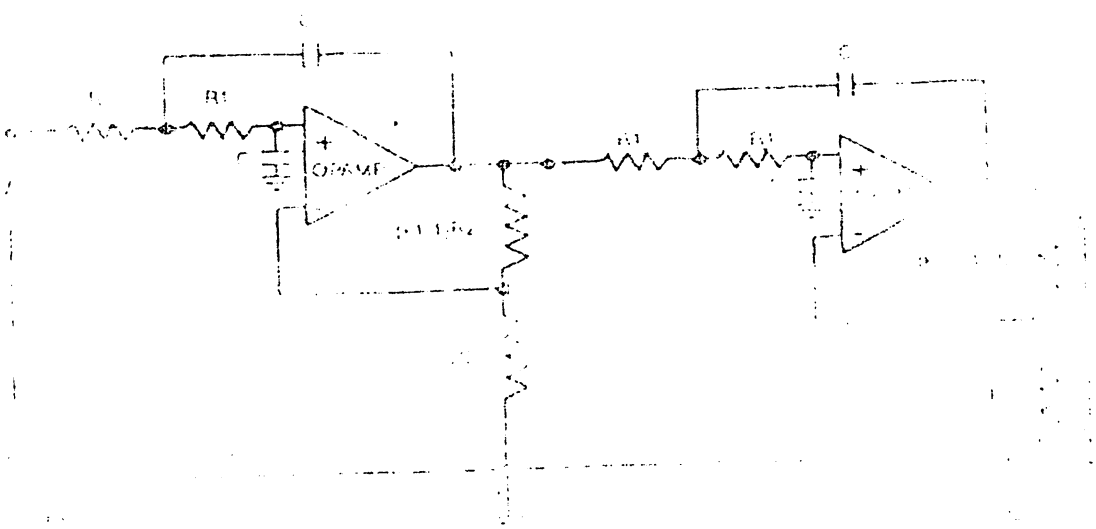
TE:- The frequency spectrum of pulse Amplitude Modulation Signal does not contain those rmonics which when multiplied by duty cycle result in an integer .e.g. the squara wave with duty cle 0.5 (50%) does not contain even harmonics as they result in an integer when multiplied with duty cle. Thus a square wave-sampling signal only contains odd harmonics.

W PASS FILTER

Pulse Amplitude Modulation – Pulse width Modulation – Pulse position modulation system the signal is recovered by a low filter. The type of filter used is very important, as the cut-off frequency would affect the recovered signal if they were not attenuated sufficiently.

OURTH ORDER BUTTRWORTH LOW PASS FILTER

fourth order Butter worth filter can be formed by cascading two- second order butterworth filters. can be seen from fig.(2). The components R&C are identical in both filter stages and



FOURTH ORDER BUTTERWORTH ...

They determine the cut- off frequency. In our circuit the gain of stage has been Set to 1.15 and that of other is set at 2.235

The amplitude/ frequency and phase/ Frequency responses of fourth order Butter Worth low pass filter are shown in fig. (3)

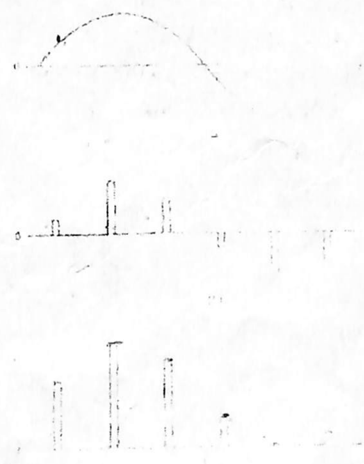
The filter design should be done critically so Any unwanted frequency component existing Close to the desired frequency component Attenuate sufficiently to save the output from Getting corrupted. Through increasing order Filter is desirable. there is a price that we Have to pay for steeper fall-off

a) Additional circuitry increases complexity & cost
b) Increase in order increases phase lag. through it is not so critical in audio circuits.

THEORY

(PAM)

Pulse amplitude modulation the simplest from of pulse modulation, is illustrated in Fig. it form an excellent instruction to pulse modulation in general. Pulse Amplitude modulation is a Pulse modulation System in which the signal is sample at regular intervals. and each sample is made proportional to the amplitude of the signal at the instant of sampling. Either wires then sends the pulses or cable or else is used to modulate a carrier. As shown in fig.(4) Two types are double polarity pulse amplitude modulation.(which is self- explanatory and signal polarity pulse Amplitude modulation) in which a fixed dc level is added to the signal, to ensure that the pulse are always positive. As will be seen shortly. the ability to use constant amplitude pulse is a major advantage of pulse modulation, and(since pulse amplitude modulation does not utilize constant amplitude pulses. it is infrequently used. When it is used.) the pulsed frequency- modulate the carrier, 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. Pulse at the sampling frequencies 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 pulse at the sampling rate equal in amplitude to the signal voltage each instant. The pulse are then passed through a pulse shaping network which gives them flat tops.



AMPLE AND SAMPLE HOLD OUTPUT

the pulse width of the carrier pulse trains used in natural sampling is made very short compared to the the period. the natural pulse amplitude modulation referred to as instantaneous pulse amplitude odulation. As it has been discussed. shorter pulse is desirous for allowing many signals to be include TDM format but the pulse can be highly corrupted by noise due to lesser signal power. One way to aintain reasonable pulse energy is to hold the sample value until the next sample is taken. This chnique is formed as sample and hold techniques. The sample and hold waveform looks as shown nder fig 5(b)

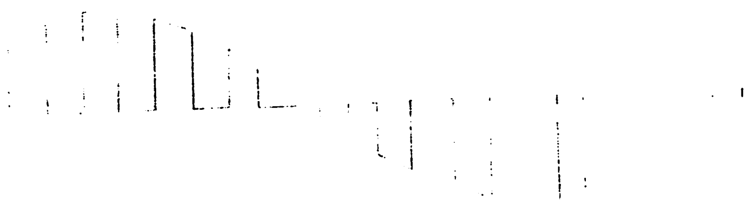
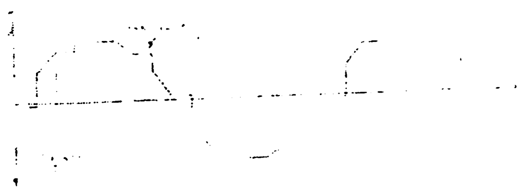


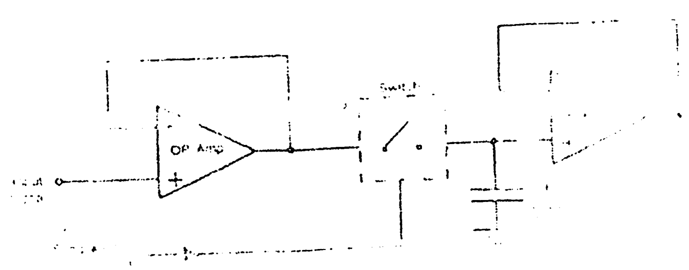
Fig. No (5a)



SAMPLE & HOLD WAVEFORM

Fig. No (5b)

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 signals improved. The hold facility can be provided by a capacitor. When the switch connect the capacitor to pulse amplitude modulation output it change to the instantaneous value. A buffered sample and hold circuit consists of unity gain buffer preceding and succeeding the charging capacitor. The high output impendence of the loading of the message source and also ensure that the capacitor charge by a constant rate irrespective of the source impendence. See fig.(6)



SAMPLE/HOLD CIRCUIT

Fig. No (6)

The high input impendence of the succeeding buffer prevent the change from the capacitor due to loading and hence the capacitor can hold the charger for infinity 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 called as droop rate is impotent parameter is sample and hold circuit design.

PULSE AMPLITUDE MODULATION:-

The modulation signal is applied to pin no. 3 of IC TL 074, this IC buffer signal and is fed to pin no. 3 & 4 of IC DG 211. the pulse input is applied to pin no. 1 & 16 of IC DG 211. the same signal is inverted and applied to pin no. 9. the sample output is available at pin no. 2 the signal is buffered by IC 5b (TL 074) and output is available on socket labeled sample output. Same output as sample output is available at pin no. 15 of IC DG 211. it is applied to sampled & hold circuit comparing of IC TL074. The sample & hold output is available at pin no. 8 of IC TL074 & socket labeled sample & hold output. This sample & hold output is fed to pin no. 11 IC DG 211, and the output is available on the socket labeled flat top output.

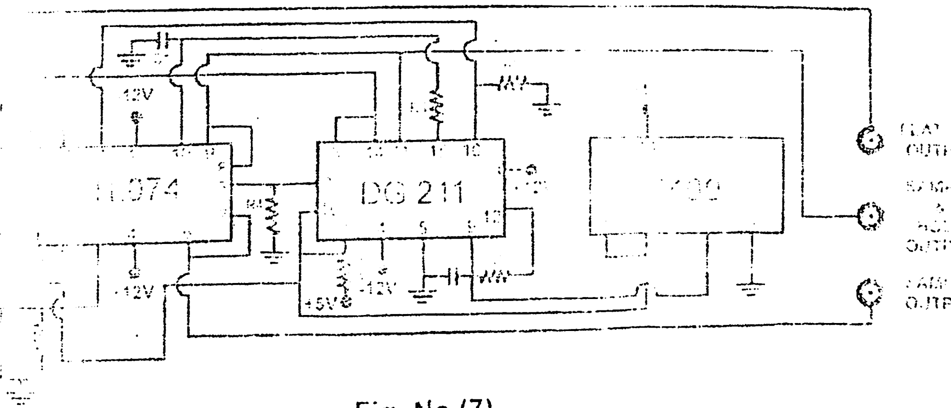


Fig. No (7)

DEMOUALTION:-

The input signal to this block is first attenuated by the resistance 15K & 10K resistor, and is buffered by IC (TL074) quad high speed of amp. IC. This OP amp. Is configured as non-inverting unity gain buffer. The output of this OP-Amp. Is then fed to fourth order butter worth low pass filter. (this filter consists of IC (TL 074) & different values of resistor and capacitors. The output of the filter is fed to the A.C amplifier circuit consists of signal OP-Amp, whose gain can be varied, by varying the potentiometer which is mount on the panel in demodulator section. The final output can be observed at the socket labeled demodulated output. The input to each OP-Amp and the output from each OP-Amp, are a.c coupled with capacitors to remove any d.c off sets.

DEMODULATION :-

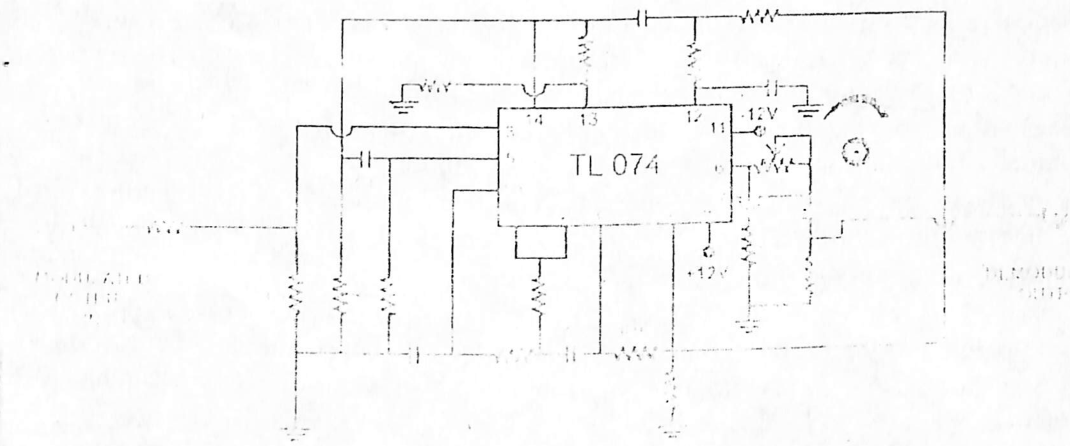


Fig. No (11)

PULSE WIDTH MODULATION:-

The 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. (9), we have a fixed amplitude of the starting time of each pulse, but the width of each pulse is a made proportional to the amplitude of the signal at that instant. Fig. 9 there may be a sequence of signal sample amplitude of (say) 0.9, 0.5, 0 and 0.4V. these can be represented by pulse width of 1.9, 1.5, 1.0 and 0.6 μs, and it has been assumed that signal amplitude ay this point will vary between the limits of + 1V (width = 2μs) and - 1V (Width = 0μs). Zero amplitude is thus the average signal level, and the average pulse width is not possible. It would make the pulse and before if 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 commencement of adjoining pulse is 10 (power 6) / 8000 = 125μs. this adequate not only to accommodate the varying width modulation has the advantage, when compared with pulse position modulation, which will be treated next, that its pulses are of varying width the therefore, of varying power content. This mean that the transmitter must be power full 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 as seen.

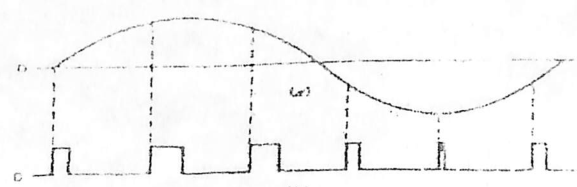


Fig. No (9)

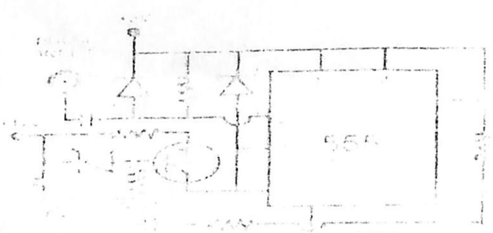


Fig. No (10)

The circuit uses IC 7555 timer IC operation in monostable made. The transistor BC 327 in conjunction with when diode and resistor 1K2 operate as constant. Current source provided a constant current of 4.1 mA at the transistor collector. The pulse input is applied to capacitor In resistor 4K7 & diode at 85 convert wave range into – Ve going pulses, which are applied at pin.2 (trig) input of the 7555 used to trigger the monostable normally, a transistor internal to the 7555 shorter the output of the constant current source to ground via pin no. 7 (discharge) pin when a negative going edge in applied to trigger input of IC 7555. its internal transistor is turn off and the constant current source in allowed to charge capacitor 0.1µs at a constant rate. In addition the 7555's output pin no. 3 is taken high. The voltage across 0.1µs rise linearly. This is monitored by threshold (THR) pin no. 6 input. As soon as the voltage at THR input reaches the level of the CTRL (control) input of 7555's output pin no. 3 is returned to a low state and chip internal transistor turns ON again, shorting out 0.1µ f, via diode bal 85 (pin no. 7)

The cycle is repeated every time a falling edge is applied to the 7555's output pin (pin no.3). if no signal is applied to the modulating input of PWM circuit, the voltage at the IC's CTRL (control pin no. 5) input is a DC level of +3.3V, and the duty cycle of the square wave at its output is approximately 50%. The square wave is the output signal from the pulse width modulator black and can be monitored at the socket labeled PWM, output if analog signal is applied to the modulating input of PWM circuit. This analog input signal is a.c coupled to the 7555's CTRL input pin no. 5 and becomes superimposed on the CTRL input +3.3 d.c level. The voltage on the CTRL input swing on either side +3.3V d.c in sccordance with applied analog signal. This changing voltage can be monitored on pin no.5 suppose, the voltage at the THR pin no. 6 of 7555 IC reaches the voltage level at the CTRL pin no. 5 since capacitor 0.1µf is charged at constant rate, it takes longer for the THR voltage to reach the CTRL voltage, so a longer high level pulse at IC's output pin goes low and capacitor 0.1µf is discharge, resulting in longer high level pulse at IC's output pin than before. Thus causing the duty cycle of the output of pulse width modulator to be more than 50%. Conversely if the voltage at 7555 IC's CTRL input goes below +3.3V, capacitor 0.1µf charged for a period shorter than they were before hence, the duty cycle of pulse width modulation output becomes less than 50%.

DEMODULATION :-

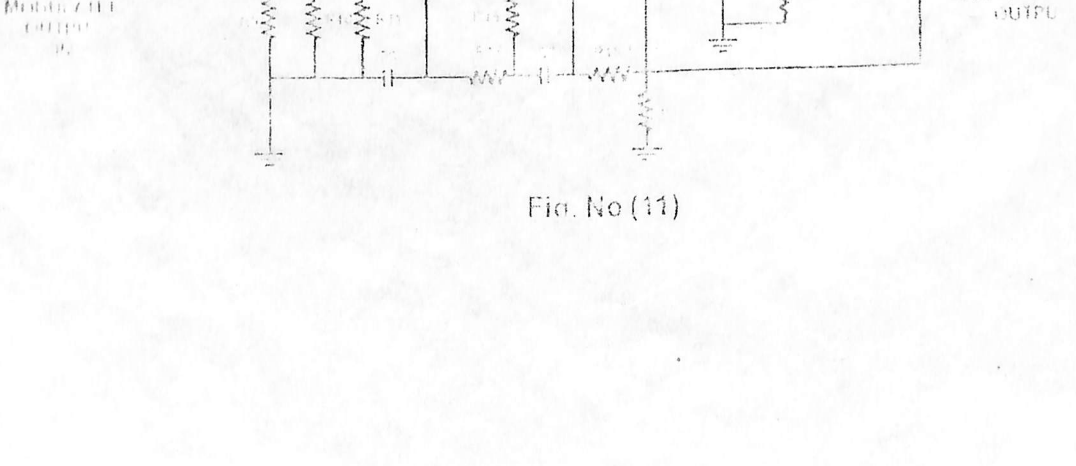


Fig. No (11)

The output signal to this is first attenuated by the resistance 10K resistor and is buffered by (IC TL074) quad high speed op-amp IC. This op-amp is configured as non-inverting unity gain buffer. The output of this op-amp is then fed to fourth order butter worth low pass filter. TWs filter. This filter consists of IC TL 074 & different values of resistor & capacitors. The output of the filter is fed to the A.C amplifier circuit consists of signal op-amp, where gain can be varied, by varying the potentiometer which is mount on the panel is demodulator section. The final output can be observed at the socket labeled demodulated output, the input to each op-amp, and the output from each op-amp. are a.c coupled with capacitor to remove any d.c off sets.

Note:- There is a instant rise in amplitude of demoduiated wave this is due to abrupt change in width of the pulses.

PULSE POSISTION MODULATION:-

The amplitude and width of the pulses is kept constant in this system. while the positions of each pulse, in relation to the position of a recurrent reference pulse is varied by each instantaneous sampled value of the modulation wave.

The analog signal is changed to a PWM signal first and then the PWM signal is changed to a pulse position modulated signal. This double modulation and the transmitter may same redundant, but the improvement in noise immunity is well worth the added effort. The PPM transmitter is far superior to the other will introduce error. The major disadvantage are a more complex circuit and higher costs.

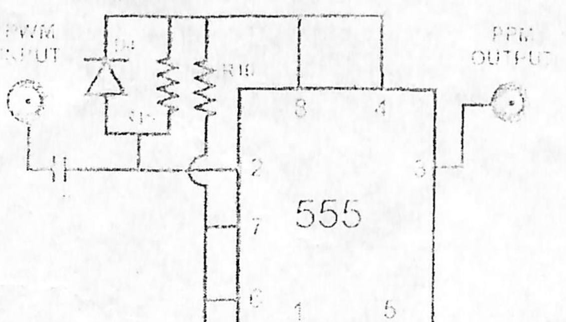


Fig. No (12)

The PPM modulator circuit is shown fig. (12). The signal of the derived frequency is applied to the circuit from which the negative triggered pulse are derived with the help of the diode 4148 and or R-C combination which work as a differentiator. These negative triggered pulse are applied to the input of the 7555 timer -1 which is working in the monostable mode. They decide the stating time of the PWM pulses. The end of the signal at pin no. 5 to which the modulating signal is apply. Therefore the width of the pulses depend upon the value of the modulating signal. This PWM output from pin 3 of 7555 timer-1 is applied in pin no. 2 of 7555 timer 2 through the diode and R1-c1 combination.

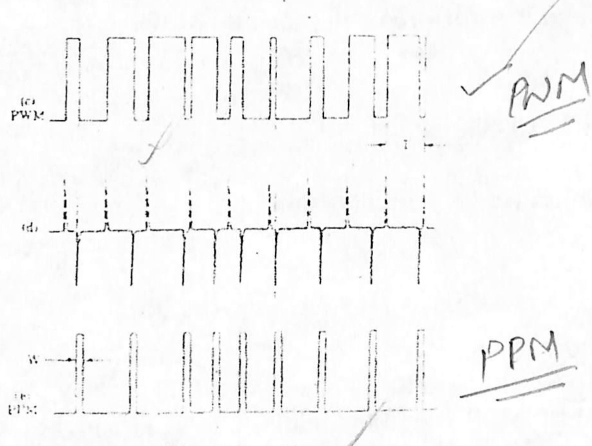


Fig. No (13)

Thus the input to pin no. 2 is the negative trigger pulses which correspond to the tralling edges of the PWM wave form. The 7555 timer 2 is working in a monostable mode and the width of the pulse is constant governed by R2-C2 combination. The negative trigger pulses decide the stating time of the output pulses and thus the output at pin no. 3 is the derived pulse position modulated output.

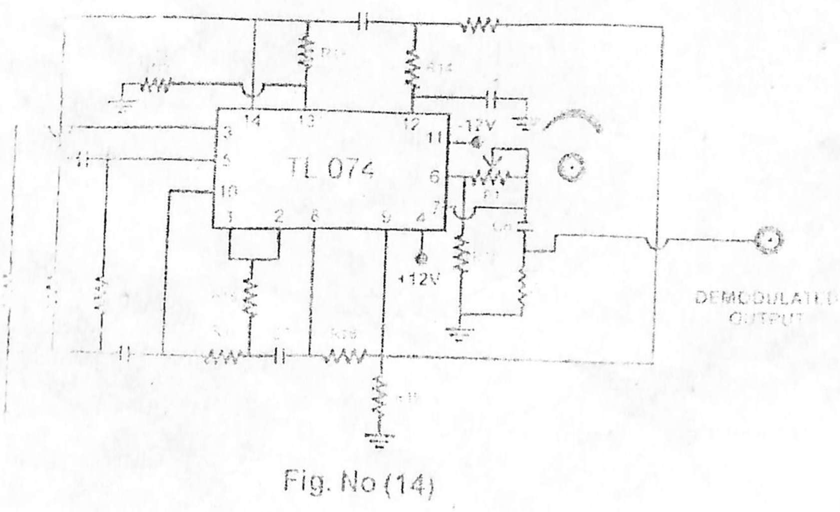


Fig. No (14)

This utility the fact that the gaps between the pulse of PWM signal contains the information regarding the modulating signal. During the gap A-B between the pulses the transmitter is cut off and the capacitor C gets charged through the R-C combination. During the pulse during B-C the capacitor discharge through the monitor and the collector voltage become low. Thus the wave from at the collector is approximately a saw tooth wave from whose cover lope is the modulating signal when this passed through a fourth order low pass filter we get the desired de-modulated output. This filter consists of IC TL074 & different values resistors & capacitors. The output of the filter is fed to the A.C amplifier circuit consists of signal op-amp, whose gain can be varied, by varying the potentiometer which is mount on the panel in demodulator section. The final output can be observed at the socked labeled demodulated output. The input to each op-amp, and the output from each op-amp are a.c coupled with capacitors to remove any d.c off sets.

Note:- There is a instant rise in amplitude of demodulated wave this is to abrupt between the pulses.

AUDIO INPUT BLACK:-

This circuit allows speech and music signals as input the ME 746 trainer. It allows these signals over different communications systems e.g the input block may be connected to PAM, PWM or PPM.

The audio Input block consists of TL082 which is adual op-amp configured as an amplifier. This block gets input from microphone which can be inserted signal. This signal is then applied to the pin no. 2 of IC (TL074).

Through capacitor which is used as a.c coupling. The signal is then amplitude by the op-amp its output further applied to IC (TL074) which again amplifier the signal and output is applied to pin no 9 of IC (TL074) which with is configured as low pass filter with cut off frequency .304KHz. The final output is available at socket labeled o/p. The two zener diodes are to set the reference voltage.

The audio Input block consists of TL082 which is a dual op-amp configured as an amplifier. This block gets input from microphone which can be inserted in the microphone jack provided (labeled as MIC) which converts the sound signal to electrical signal. This signal is then applied to the pin nO.2 of IC (TL074).

Through capacitor which is used as a.c. coupling. The signal is then amplified by the op-amp and its output is further applied to IC (TL074) which again amplifies the signal and output is applied to pin no 9 of IC (TL074) which with is configured as low pass filter with cut off frequency .304KHz . The final output is available at socket labeled *o/p*. The two zener diodes are used to set the reference voltage.

AUDIO OUTPUT BLOCK:-

This circuit converts the audible frequency electric signals to audible output. This circuit incorporates

LM 386 which is a audio amplifier. The output of this IC is available at pin 5 and this output signal after passing through capacitor is applied to load speaker to provide the audible output. To facilitate the use by head phones provided with the module the panel contains the earphone socket. The module incorporates on - board power supply for the amplifier circuitry. Hence the module can be powered from A.C. mains.

FAULTS SWITCHES:-

This block comprises of eight fault switches. Each of them effect different blocks on ME 746.

FAULT SWITCH 1 It disconnects the resistance (1.2K) in PWM block from the ground link.

FAULT SWITC 2: It disconnects the link from transistor T1 collector pin nO.6 & 7 of IC (7555).

FAULT SWITCH 3 It cuts the feedback path. and affects the flat-top output.

FAULT SWITCH 4 It disconnects the modulating liP to the IC (DG 211)

FAULT SWITCH 5: It disconnects the supply voltage of IC 7555 in PPM block.

FAULT SWITCH 6: It disconnects the feed back path of IC (TI074)

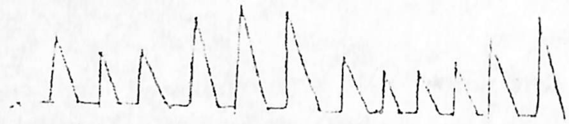
FAULT SWITCH 7: It disconnects the crystal o/p frequency from *liP* pin nO.1 0 of IC(74HC4040)

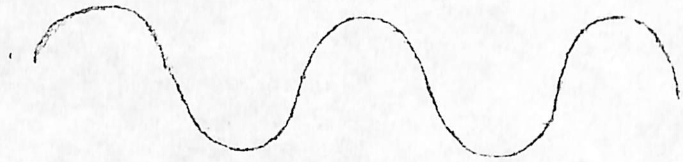
FAULT SWITCH 8: It disconnects the square wave liP to the sine wave converter.

FAULT SWITCH 9: It disconnected the output of not gate to the input of IC074 & its effect of Flat-top output.

FAULT SWITCH 10: Its disconnected of input of demodulation & its effect the demodulated out

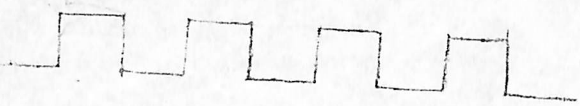
TP 1 (Fro PWM experiment)



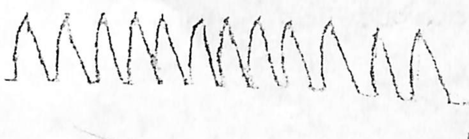


TP 2 (With PAM experiment)

TP 3 (With PAM experiment)- Sinc pulses position changes according to amplitude of modulating signal



TP4 (With PPM experiment)- sine pulses position changes according to amplitude of modulating signal



TP 5 (hold or sinc pulses)

TP 6 (From function generator)-Frequency generated from crystal oscillator.

TP 7 (from function generator)-
To Generated modulating signal (sine wave) of 1KHz and 2KHz frequency.

TP 8 :- Sawtooth wave shape on PAM

TP 9:- Distorted output démodulated section at pin no. 8

Experiment No:-1

PAM

1. Connect the pulse output i.e 8 KHz to pulse input.
2. Connect AF signal output to signal input.
3. Connect CRO channel A with the modulated output socket i.e flat top output.
4. Connect CRO channel B with the AF signal output socket.
5. Keep the amplitude control at minimum position.
6. Keep frequency selector switch to 1KHz frequency.
7. Trigger CRO with the AF sinal.
8. Keep the CRO time/div knob to the 0.2 ms/div.
9. Switch on the instrument using on/off toggles switch.
10. Try varying the amplitude & frequency of sine wave by amplitude pot and frequency change over switch observe the pulse amplitude modulation i.e flat-flop modulation.
11. Also try varying the frequency of pulses, by connecting the pulse input to the 4 frequencies available i.e 8,16,32,64 KHz in pulse output socket & also observed the corresponding effect on the output.
12. Observe the effect on all the output, sample output & sample & hold output.
13. Switch ON fault no 3,4,7,8 one by one & observe their effect on pulse amplitude modulating output and try to locate.
14. Switch off the power supply.

Experiment No.:-2

PAD

1. Connect the modulated output i.e flat top output to the PAM input of demodulator circuit.
2. Keep the frequency selector switch to 1KHz position.
3. Connect the output of low pass filter to the input of AC amplifier.
4. Keep the gain control pot in AC amplifier block in maximum position.
5. Connect CRO channel A with the AC amplifier output socket.
6. Observe the demodulated output.
7. If the output is not pure sine wave adjust the gain control pot of AC amplifier block.
8. Similarly connect the sample & hold & sample outputs to demodulator circuit and see the demodulated wave from at the output of AC amplifier,
9. Switch on the fault switches 1,3,4,6,7,8 one by one and see their effect on demodulated output & try locate them.
10. Switch off the power supply.

Experiment No.:-3

PWM

1. Connect the pulse output i.e 8Khz to the pulse input.
2. Connect the signal output to AF signal input.
3. Connect CRO channel A with the PWM output socket.
4. Connect CRO channel B with the AF signal output socket.
5. Keep amplitude control at minimum position.
6. Keep frequency selector switch to 1KHz frequency.
7. Trigger CRO with the AF signal.

8. Keep CRO time/div knob to the 0.2ms/div.
9. Switch on the instrument using ON/OFF toggle switches.
10. By varying the amplitude and frequency of sine wave by amplitude pot and frequency change over switch observe the pulse width modulated output.
11. Also 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 socket & also observe the corresponding effect on the output.
12. Switch off the power supply.

Experiment No.:-4

PWD

1. Connect the PWM output to the PWM input of demodulator circuit.
2. Keep frequency selector switch to 1Khz position.
3. Connect the output of low pass filter to the input of AC amplifier.
4. Keep gain control pot in AC amplifier block in maximum position.
5. Connect CRO channel A with the AC amplifier output socket.
6. Observe the demodulated output on CRO.
7. If the output is not proper sine wave adjust the gain control pot AC amplifier block.
8. Switch on fault no. 1,2,6,7,8, one by one & observe their effect on demodulated output & try to locate them.
9. Switch off the power supply.

Note: There is an instant rise in amplitude at the demodulated this is due to abrupt change in width of the pulses.

Experiment No.:-5

PPM

1. Connect the pulses output i.e 8Khz to the pulse input of PWM block.
2. Connect AF signal output to the signal input of PWM block.
3. Connect PWM output to PPM input.
4. Connect CRO channel A with the PPM output socket.
5. Connect CRO channels B with the AF signal output socket.
6. Keep frequency selector switch to 1Khz frequency.
7. Trigger CRO with the AF signal.
8. Trigger CRO time/div knob to the o.2ms/div.
9. Switch ON the, instrument using ON/OFF toggle switch.
10. Varying the amplitude and frequency of sine wave by amplitude pot and frequency change over switch. Observe the pulse width modulated output.
11. Also try varying the voltage of pulse, by connecting the pulse input to the 4 frequencies available i.e 8,16,32,64Khz. In pulse output socket & also observe the corresponding effect on the modulated output.
12. Switch on fault no. 1,2,5,7,8 one by one & observe their effect on pulse position modulation output & try to locate them.
13. Switch off the power supply.

Experiment No.:-6

PPD

1. Connect the PPM output to the PPM input of demodulator circuit.
2. Keep frequency selector switch to 1KHz position.
3. Connect the output of low pass filter to the input of AC amplifier.
4. Keep gain control pot in AC amplifier block maximum position.
5. Connect CRO channel A with the AC amplifier output socket.
6. Observe the demodulated output on CRO.
7. If the output is not proper sine wave adjust the gain control pot of AC amplifier block.
8. Switch on the fault no. 1,2,5,7,8 one by one & observe their effect on demodulated output & try to locate them.
9. Switch off the power supply.

Note: There is an instant rise in amplitude at the demodulation this is due to abrupt change in distance between the pulses.

Experiment No.:-7
To study PPM using DC input

1. Connect the circuit as per experiment no-5 expect to connect the AF input of the PWM block.
2. Connect DC output to the AF input of PWM block.
3. Observe the pulse position modulation at PPM output.
4. Vary the DC output while observing the output of pulse position modulation block.
5. Switch on fault no. 1,2,5,7,8 one by one & observe their effect on pulse position modulation output & try to locate them.
6. Switch off the power supply.

Experiment No.:-8

1. Connect a microphone in the MIC socket in audio input block.
2. Connect the output of audio input block to AF signal input of pulse amplitude modulation block.
3. Connect the 8Khz pulse output of pulse amplitude modulation block.
4. Connect sample output of pulse amplitude modulation block to PAM input of demodulator block.
5. Connect output of low pass filter to input of AC amplifier block.
6. Keep gain control pot of AC amplifier in mid position.
7. Connect output of AC amplifier block to input of audio output tblock.
8. Switch on the instrument using on/off toggle switch.
9. Observe the pulse being modulated by radio signal at output of ample output, sample & hold & flat top output.
10. Also, observe its demodulation and hear the same voice in speaker/headphone which was fed in the micro phone in the input.
11. Switch off the power supply.

INSTRUCTION MANUL
FOR
AMPLITUDE MODULATION & DEMODULATION

mplitude modulation & demodulation apparatus has been designed to study he following:

A. Amplitude Modulation & Calculation of Modulation Index Percentage Modulation of Side Band Frequency.
B. Amplitude Demodulator.

The instruments comprises of the following built parts:

1. Fixed output DC regulated power supply of ± 12Volts.
2. Built in Carrier Sine Wave generator of 450 KHz frequency 2.5V peak to peak amplitude.
3. Built in Audio Frequency Function Generator of 1 KHz 1.5V peak to Peak Amplitude.
4. Circuit diagram for modulator & demodulator are printed on the front panel and components are soldered behind the front panel.

THEORY

Modulation is the process in which some property of high frequency wave, also called as carrier wave ωc, is altered in such way by low frequency information signal, called as modulating wave ωm, to transmit from one place to other place through air. In double sideband amplitude modulation the amplitude of carrier wave is altered by modulating wave such to from an envelope upon the carrier on both sides. A non linear element is used to perform the high level modulation and a linear element for low level modulation. Properly biased transistors provide linear/non linear operation with some amplification. The AM wave is represents as shown in fig. (3) and its sidebands as shown in Fig. (4). The top envelope is represents as

Vc+Vm sin ωmt,

Where the bottom envelope is represented as

(Vc+Vm sin ωmt,)

When Vc is the carrier voltage, Vm is the modulating voltage. It is shown that formatting of envelops depends upon the term Vc+Vm, where Vc is kept constant hence the envelope height depends upon Vm only. The ratio between envelope amplitude is called as modulation index or factor mf, which is represented as

$$\frac{V\max - V\min/2}{V\max + V\min/2}$$

Calculating mf with 100 gives percentage modulation m%, which should never > 100. The power radiated in a given load is related in two terms, one the carrier power Pc and the other is modulated carrier power Pm. The difference of two sideband power Ps which is equal to Pusb+Plsb. The Pm is related with Pc as

$$\frac{Pm}{Pc} = 1 + \frac{mf^2}{2}$$

THEORY OF OPERATION

For Amplitude Modulation

Figure (1) that shows the shows the internal circuit of the chip. We see that the carrier signal is applied to pins 8 and 10 in a common mode to a set of cross coupled differential amplifiers (Q1 with Q4 and Q2 with Q3). Transistor Q7 and Q8 serve as the constant current generator for the differential amplifiers, wheras the bias voltage applied to pin 5 determines the amount of current through the

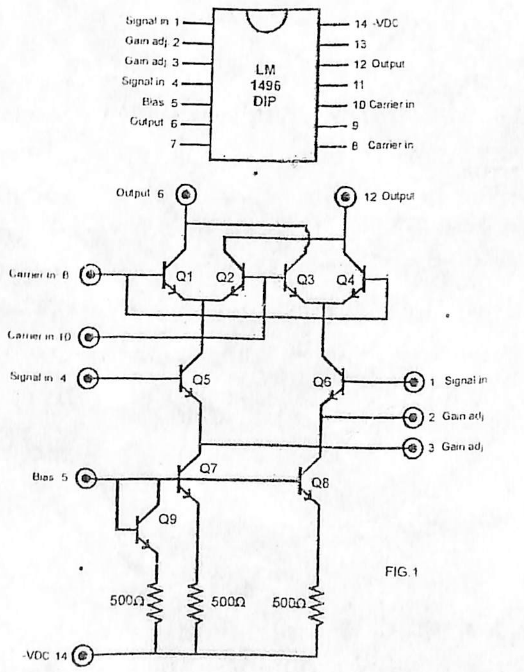


FIG.1

amplifiers. The resistor connected to pins 2 and 3 sets the modulator gain with a smaller resistor resulting in higher gain. The DC voltage difference between pins 1 and 4 will balance the differential amplifiers for complete carrier regection by equalizing the current in each differential amplifier. When the message signal is applied to pins 1 and 4 transistor Q5 and Q6 will alternately increase (or decrease) the current through their associated amplifier to output the sum and difference frequencies in the side band pair. The output is taken pin 6 for the modulator.

DEMODULATION OF AM WAVE

There are may procedure to demodulate the amplitude modulated waves. In present board (envelope detection) linear diode demodulation circuitry is provided. In linear diode detector circuits, diode presents a low ohmic path to input signal in

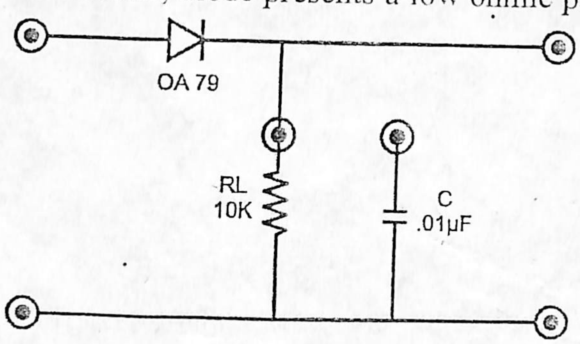


FIG.2

one direction only. For this reason changes in the peak current through the diode will remain confined to straight portion of Volt-ampere characteristics of the diode in other words it can be said that it acts as half wave rectifier carrier wave components. When carrier is presented only the average current Iavg will pass through the diode and has a constant amplitude output. When modulated wave is at the input the current 1 peak varies through forms a low pass filter to remove these HF components (in actual practice pye filters are employed). The maximum time constant is kept max equal RLC= 1/ωcωm.

The detection efficieny of diode detector is calculated from the input modulated signal power, or by mean of modulation depth and detected output as

β = average potential across RL/peak input signal voltage

The maximum detection efficiency lies between 80-90%, and upto 60-70% of modulation index.

PROCEDURE

For Modulation

1. Connect the carrier OSC output to carrier input.
2. Connect AF signal output to AF signal input.
3. Connect CRO channel A with the Amplitude Modulation output sockets.
4. Connect the CRO channel B with the AF signal output sockets.
5. Keep the Amplitude control at minimum position.
6. Switch ON the instrument using ON/OFF toggle switch.

7. Adjust CRO time base for 0.2ms.DV and vert gain at 1V/Div a band will appear upon the screen. Position it at the center of the screen & calculate

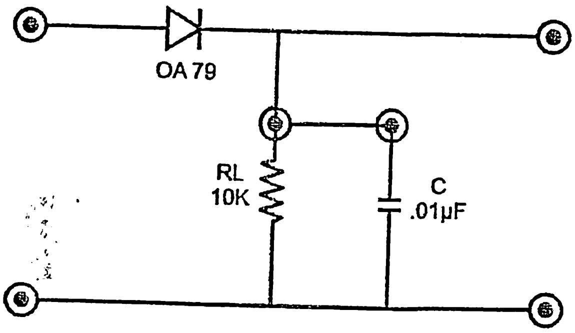


FIG.3

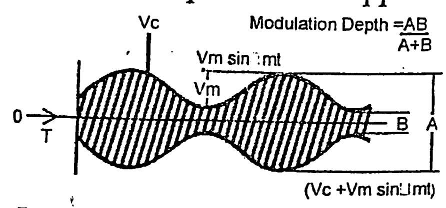
the value of modulation index percentage modulation upper side band frequency and lower side band frequency.

Formula used calculation of modulation index or modulation coefficient and side band frequencies.
ma = 2V max -2V min/2V max +-2V min

Percent modulation = ma x 100%

Upper side band frequency =carrier signal frequency + modulating signal frequency.

8. Connect the resistance box across the modulated output sockets in parallel with CRO leads measure the signal in Vpp.
9. Feed one volt p-p AF signal to the AF input. Trace out the pattern of the modulated wave and measure amplitudes in Vpp as shown in Fig. (4).



Envelop formation of Amplitude Modulated Waveform

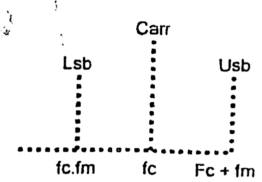


Fig.4

10. Increase AF input to successive levels and note amplitude A and B, for each increment. Calculate modulation factor for each input. Draw a plot between input signal (AF) modulation factor. The curve of the graph shown the modulation process. Increase maximum AF signal to observe the distorted wave form since cut-off and saturation of the transistor Q5.

11. Calculate peak power and Pc = VPP& RL and modulated power 2Pm = VA%RL, where AV is the pp amplitude traces from modulated envelope. Find out the power in side bands as p side bands = Pm. PC.
12. Plot side bands spectrum measuring AF and carrier signal frequencies.
13. Trigger CRO with input signal (For external triggering)

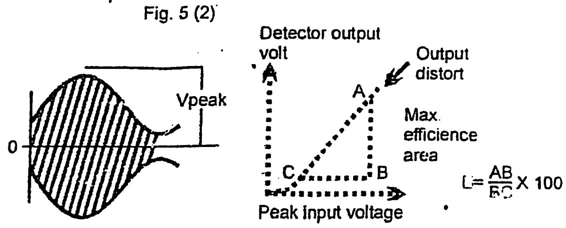
Note: - If resistance box and resistor is not available then modulated output may be terminated in provided RL = 10K in demodulator circuit as shown Fig. (3) & Fig. (4).

For Demodulation

1. Connect the Amplitude modulation circuit to the input of Amplitude demodulation circuit.
2. Connect the CRO channel A with the amplitude demodulation output.

Demodulated signal without 'c'
Fig. 5 (1)


Output wave form with 'c'
Fig. 5 (2)



Different wave form at demodulator and its response curve
Fig. 5 (3)

3. Remain C out Circuit and no RL should be connected across modulated output sockets then the output is as shown in fig. 5 (i).
4. Connect CRO channel B with the AF modulated signal output.
5. Connect C in the Circuit and note its effect upon the RF components as shown in Fig. 5 (ii).
6. Feed AF signal for different modulation factor and note the amplitude of demodulated output voltage in p-p, and input voltage as Vp=Vpp/2 or A/S.
7. Plot a response curve between output voltage and Vp input. Select the linear part of the curve and calculate the efficiency of the detector as

   β=Slope of the curve as shown in Fig. 5 (iii).

INSTRUCTION MANUAL
FOR
B.H CURVE

(2)

The apparatus consists of transformer variable in steps. resistance, capacitor network and a coil wound on former, phase shifting in network is fed to cathode say oscillator to horizontal and vertical plates. The hysteresis produced is noticed on the screen of oscillator, complete circuit diagram is engraved on the panel.

[illegible]ption:

[illegible] ON- To ON the instrument after plugging for each plate and ground of C.R.O

[illegible]minal [illegible] terminal are provided for each plate and ground of C.R.O

[illegible] Pl[illegible] connect horizontal (X) plates of C.R.O

[illegible] Pl[illegible] are to feed the vertical (Y) plates of C.R.O ground terminal are connected to the [illegible] of C.R.O

[illegible] potential of the respective plates.

[illegible] the potential in network the deflection notice on C.R.O can be increase or [illegible] this switch sported by potentiometer (Level).

[illegible] the primary of solenoid =10000
Wire No. S.W.G = 44

[illegible] the secondary of solenoid = 280
Wire No. S.W.G = 24

[illegible] rods are provided to vary the curves in shape after inserting in the solenoid.

[illegible] are to connect ampere meter 1Amp. A.C. A lead is provided to short these [illegible] when ampere meter is not connected No output will be seen if these socket are open.

[illegible] sockets are to connect volt meter 10V A.C.