Dronacharya Group of Institutions

COMMUNICATION LAB - II MANUAL

LIST OF EXPERIMENTS

(Communication Lab – II)

1. To construct a triangular wave with the help of Fundamental Frequency and its Harmonic component.
2. To construct a Square wave with the help of Fundamental Frequency and its Harmonic component.
3. Study of Pulse code modulation (PCM) and its demodulation using Bread Board.
4. Study of delta modulation and demodulation and observe effect of slope overload.
5. Study of pulse data coding techniques for NRZ formats.
6. Study of Data decoding techniques for NRZ formats.
7. Study of Manchester coding and decoding.
8. Study of Amplitude shift keying modulator and demodulator.
9. Study of Frequency shift keying modulator and demodulator.
10. Study of Phase shift keying modulator and demodulator.
11. Study of single bit error detection and correction using Hamming code.
12. Measuring the input impedance and Attenuation of a given Transmission Line.

EXPERIMENT No. : 01

Aim: To construct a triangular wave with the help of fundamental frequency and its harmonic component.

Apparatus Required:

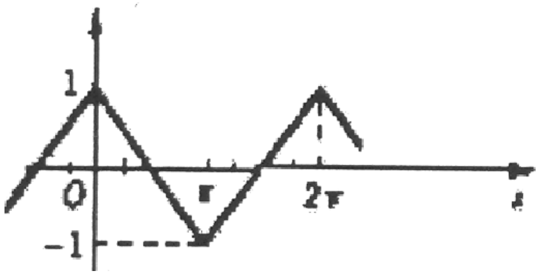
S. No.	Component	Quantity
1.	Trainer kit **ST2603**	1
2.	CRO	1
3.	Connecting leads	2

Theory: Fourier synthesis is a method of electronically constructing a signal with a specific, desired periodic waveform. It works by combining a sine wave signal and sine wave or cosine-wave harmonics (signals at multiples of the lowest, or fundamental, frequency) in certain proportions. The scheme gets its name from a French mathematician and physicist named *Jean Baptiste Joseph Baron de Fourier*, who
lived during the 18th and 19th centuries.
A mathematical theorem stating that a Periodic function $f(x)$ which is reasonably continuous may be expressed as the sum of a series of sine or cosine terms (called the *Fourier series*), each of which has specific Amplitude and Phase coefficients known as *Fourier coefficients.*Many waveforms represent signal energy at a fundamental frequency and also at harmonic frequencies (whole-number multiples of the fundamental). The relative proportions of energy concentrated at the fundamental and harmonic frequencies determine the shape of the wave. The wave function (usually amplitude, frequency, or phase versus time) can be expressed as of a sum of sine and cosine functions called a Fourier series, uniquely defined by constants known as Fourier coefficients. If these coefficients are represented by $a0, a1, a2, a3, ..., an, ...$ and $b1, b2, b3, ..., bn ...$, then the Fourier series $F(x)$, where x is an independent variable (usually time), has the following form :

$$F(x) = a0/2 + a1 \cos x + b1 \sin x + a2 \cos 2x + b2 \sin 2x + ... + an \cos nx + bn \sin nx + ...$$

In Fourier synthesis, it is necessary to know, or to determine, the coefficients $a0$, $a1$ $a2$, $a3$, ..., an ... and $b1$ $b2$, $b3$, ..., bn ... that will produce the waveform desired when "plugged into" the generalized formula for the Fourier series. Then, sine and cosine waves with the proper amplitudes (as defined by the coefficients)
must be electronically generated and combined, up to the highest possible value of n. The larger the value of n for which sine-wave and cosine-wave signals are generated, the more nearly the synthesized waveform matches the desired waveform.

Following is the fourier series equation and wave form for the triangular wave.

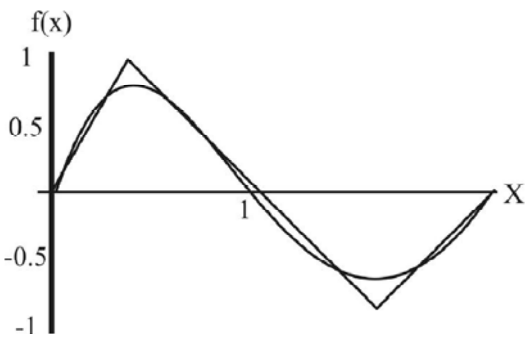


Triangular wave :

$$\frac{8}{\pi^2}\sum_{n=0}^{\infty}\frac{1}{(2n+1)^2}\cos(2n+1)x$$

i.e equation for triangular wave is
A*(cos x + 1/9 cos 3x + 1/25 cos 5x + ………)
where A = 8/(□)2
Example:- (0V DC + 3Cos(x) + 0.44 Cos (3x) + 0.304 Cos(5x) + 0.152 Cos(7x) + 0.092 Cos(9x))

Procedure :
1. Switch ‘On’ the Power Switch and LCD.
2. Minimize all the amplitudes of frequencies by potentiometer.
3. Set DC potentiometer to 0V DC by using Output BNC & Oscilloscope DC mode.
4. Switch each frequency for cosine and positive value.
5. Use “Harmonic Output” BNC to set and observe individual harmonic and use “Output” BNC to observe the resultant waveform.
6. Use Harmonic select push button switch and Set the fundamental cosine Frequency (1 KHz) to 3V, observe it on oscilloscope.
7. Use Harmonic select push button switch and Set the 3rd harmonics i.e. cos 3x to 0.44V.
8. Use Harmonic select push button switch and Set the 5th harmonics i.e. cos 5x to 0.30V, 7th harmonic i.e. cos7x to 0.16V & 9th harmonic i.e. cos9x to 0.08V



Observation:

Result: Traced the waveform from CRO & observed the effect of harmonics on the waveform.

Precautions:

- Make sure that kit is powered off when connections are made.
- Handle the trainer kit properly.

Pre Expeiment Questions:

Q:1 What is Fourier Synthesis?

Q:2 What is the harmonics of the Fourier Series?

Q:3 The effect of various harmonics on the waveform?

Post Experiment Questions:

EXPERIMENT No. : 02

Aim: To construct a square wave with the help of fundamental Frequency and its harmonic component.

Apparatus Required:

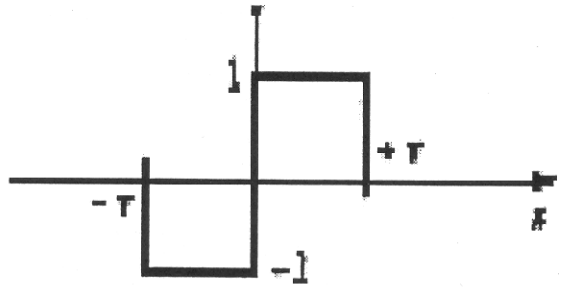
S. No.	Component	Quantity
1.	Trainer kit **ST2603**	1
2.	CRO	1
3.	Connecting leads	2

Theory: Fourier synthesis is a method of electronically constructing a signal with a specific, desired periodic waveform. It works by combining a sine wave signal and sine wave or cosine-wave harmonics (signals at multiples of the lowest, or fundamental, frequency) in certain proportions. The scheme gets its name from a French mathematician and physicist named *Jean Baptiste Joseph Baron de Fourier*, who
lived during the 18th and 19th centuries.
A mathematical theorem stating that a Periodic function *f(x)* which is reasonably continuous may be expressed as the sum of a series of sine or cosine terms (called the *Fourier series*), each of which has specific Amplitude and Phase coefficients known as *Fourier coefficients.*Many waveforms represent signal energy at a fundamental frequency and also at harmonic frequencies (whole-number multiples of the fundamental). The relative proportions of energy concentrated at the fundamental and harmonic frequencies determine the shape of the wave. The wave function (usually amplitude, frequency, or phase versus time) can be expressed as of a sum of sine and cosine functions called a Fourier series, uniquely defined by constants known as Fourier coefficients. If these coefficients are represented by $a0, a1, a2, a3, ..., an, ...$ and $b1, b2, b3, ..., bn ...$, then the Fourier series F(x), where x is an independent variable (usually time), has the following form :

$$F(x) = a0/2 + a1 \cos x + b1 \sin x + a2 \cos 2x + b2 \sin 2x + ... + an \cos nx + bn \sin nx + ...$$

In Fourier synthesis, it is necessary to know, or to determine, the coefficients $a0$, $a1$ $a2$, $a3$, ..., an ... and $b1$ $b2$, $b3$, ..., bn ... that will produce the waveform desired when "plugged into" the generalized formula for the Fourier series. Then, sine and cosine waves with the proper amplitudes (as defined by the coefficients)
must be electronically generated and combined, up to the highest possible value of n. The larger the value of n for which sine-wave and cosine-wave signals are generated, the more nearly the synthesized waveform matches the desired waveform.

Following is the fourier series equation and wave form for the square wave.



Square wave :

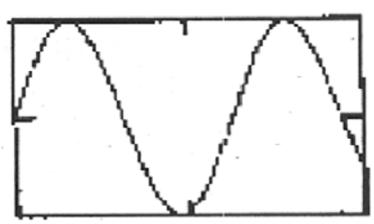
$$\frac{4}{\pi}\sum_{n=0}^{\infty}\frac{1}{2n+1}\sin(2n+1)x$$

i.e. equation for rectangular sawtooth wave is
A *(sin x + 1/3 sin3x + 1/5sin5x)
where A = 4/□
Example:- 0V DC + 5 Sin(x) + 1.72 Sin(3x) + 1.04 Sin(5x) + 0.74 Sin(7x) + 0.56 Sin(9x)

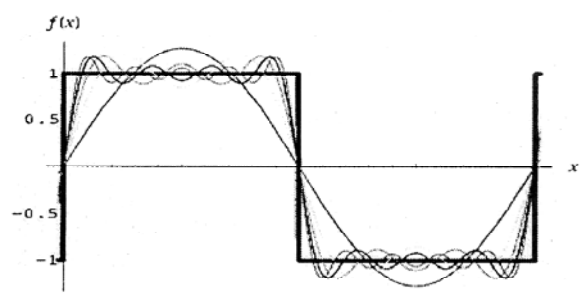
Procedure :

1. Switch 'On' the Power and LCD Switch.
2. Minimize all the amplitudes of frequencies by potentiometer.
3. Set DC potentiometer to 0V DC by using Output BNC & Oscilloscope DC mode.
4. Switch each odd frequency for sine and positive value.
5. Use "Harmonic Output" BNC to set and observe individual harmonic and use "Output" BNC to observe the resultant waveform.
6. Use Harmonic select push button switch and set the fundamental sine Frequency H1 (1 KHz) to 5V observe it on oscilloscope.

Sin (x) 0



7. Use Harmonic select push button switch and set the 3rd harmonics H3 i.e. sin3x to 1.72V output should be similar to following waveform.
8. Use Harmonic select push button switch and set the harmonic H5 (sin5x) to 1.04V and observe the output.
9. Use Harmonic select push button switch and set the harmonic H7 (sin7x) to 0.74V and observe the output.
10. Use Harmonic select push button switch and set the harmonic H9 (sin9x) to 0.56V and observe the output.



Observation:

Result: Traced the waveform from CRO & observed the effect of harmonics on the waveform.

Precautions:

- Make sure that kit is powered off when connections are made.
- Handle the trainer kit properly.

Pre Expeiment Questions:

Q:1 What is Fourier Synthesis?

Q:2 What is the harmonics of the Fourier Series?

Q:3 The effect of various harmonics on the waveform?

Post Experiment Questions:

EXPERIMENT No. : 03

Aim: Study of Pulse code modulation (PCM) and its demodulation.

Apparatus Required:

S. No.	Component	Quantity
1.	PCM modulation / demodulation ST2103 trainer.	1
2.	CRO	1
3.	Connecting leads	2

Theory: Pulse Code Modulation technique involves following steps:

(a) Sampling:

The analog signal is sampled according to the nyquist criteria. The nyquist criteria states that for faithful reproduction of a band limited signal, the sampling rate must be at least twice the highest frequency component present in the signal.

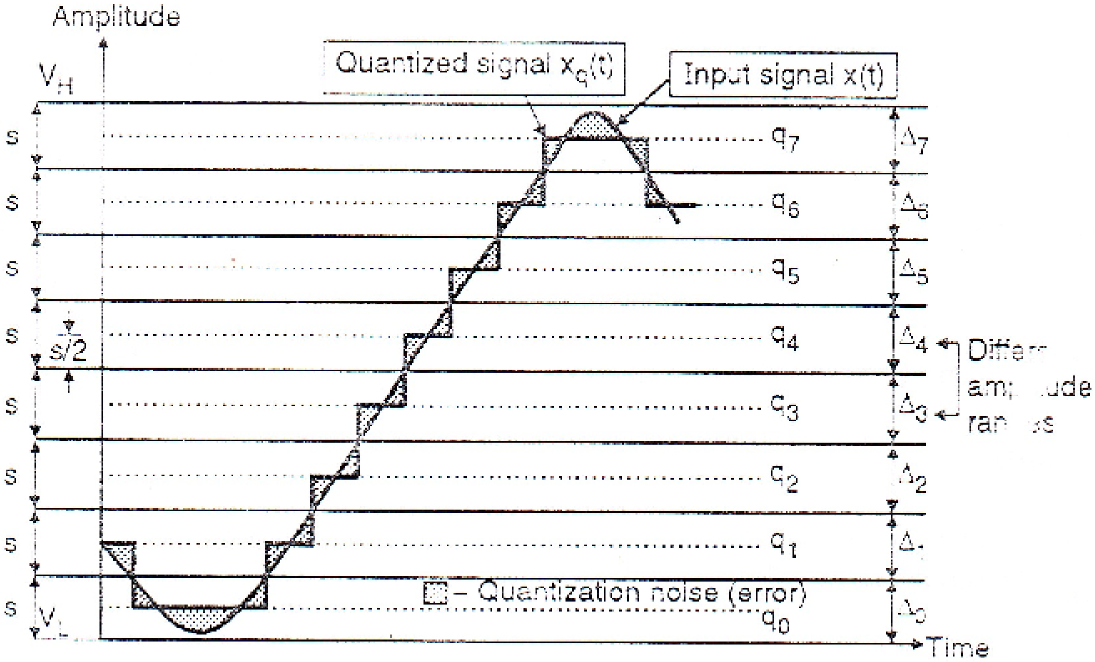
So sampling frequency ≥ 2 fm, where fm is maximum frequency component present in the signal

Practically the sampling frequency is kept slightly more than the required rate.

(b) Allocation of binary codes:

Each binary word defines a particular narrow range of amplitude level. The sampled value is then approximated to the nearest amplitude level. The sample is then assigned a code corresponding to the amplitude level, which is then transmitted.

This process is called quantization and it is generally carried out by the A/D Converter as shown below in fig1



Process of quantization

Fig.1

Procedure:

1. Ensure that the MODE switch should be in FAST mode.
2. Connect CH 0 & CH 1 to DC1 AND DC2.
3. Ensure that the DC1 and DC2 controls in Function Generator Block should be in fully clockwise direction and ~1KHz and 2 KHz signal controls set art 10Vpp.
4. Now turn ON the kit and see that the LED glows.
5. With the help of Digital Voltmeter, adjust the DC1 amplitude control until the DC1 output measures 0V.
6. Observe the output on the A/D Converter Block LED's (D0 to D6). The LED's represent the state of the binary PCM word allocated to the PAM sample being processed.
7. Adjust the D.C input from +5V to –5V in steps of 1V.
8. Observe the output of +5V is as follows:

D6	D5	D4	D3	D2	D1	D0
1	1	1	1	1	1	1

Where for the negative values it is less than 1000000.For –5V the output is as follows:

D6	D5	D4	D3	D2	D1	D0
0	0	0	0	0	0	0

This is obtained at the approximately full anti clockwise position of the DC Control.

9. Turn the DC1 control fully anticlockwise and repeat the above procedure by varying the DC2 control.
10. Trigger the dual trace oscilloscope externally by the CH.1 signal available at t.p.12 and observe the signal at CH.0 and CH.1 at t.p.5 with reference to the signal at t.p.7.
11. Now connect the oscilloscope channel 1 to CH1 sample at t.p.6 and sketch the three waveforms.

Observations:

Result: The PCM Modulation / Demodulation studied.

Precautions:

1. Connections should be checked before switching ON the kit.
2. Observations should be taken properly.

Pre Expeiment Questions:

Q:1 What does PCM stands for?

Q:2 What is Quantization noise?

Q:3 Whether PCM signal is a digital signal or analog signal?

Post Experiment Questions:

EXPERIMENT No. : 04

Aim: To study delta modulation & demodulation and observe the effect of slope overload.

Apparatus Required:

S. No.	Component	Quantity
1.	Delta modulation / demodulation trainer ST2105	1
2.	CRO	1
3.	Connecting leads	2

Theory : Delta Modulation is a system of Digital Modulation scheme in which the difference between the sample value at sampling time K and sample value at the previous sampling time (k – 1) is encoded into just a single bit. One way in which delta modulator and demodulator is assembled is as shown in fig.1 and fig.2.

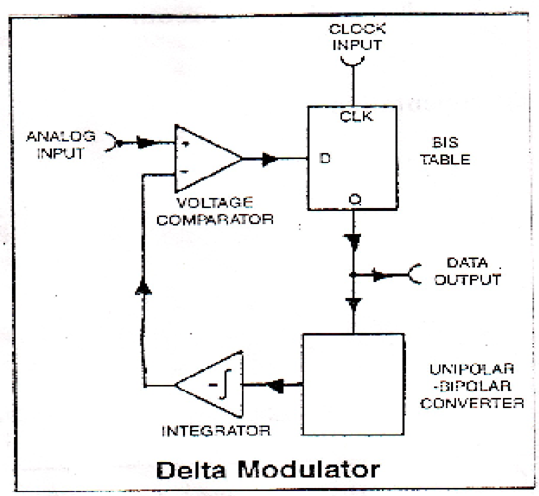


Fig.1

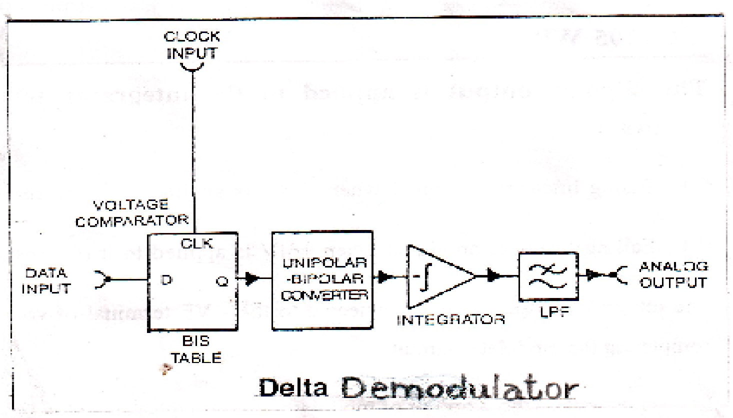


Fig.2

The baseband signal m(t) and its quantized approximation m'(t) are applied as inputs to a comparator. A comparator simply makes a comparison between inputs. If signal amplitude has increased, then modulators output is at logic level 1. If the signal amplitude has decreased, the modulator output is at logic level 0. Thus the output from the modulator is a series of 0's and 1's to indicate rise and fall of the waveform since the previous value. The comparator output is then latched into a D flip-flop which is clocked by the transmitter clock. Thus the output of the flip-flop is a latched 1 or 0 synchronous with the transmitter clock edge. The binary sequence is transmitted to receive and is also fed to the unipolar to bipolar converter. This block converts logic 0 to voltage level of +4V and 1 to voltage level of – 4V. The bipolar output ia applied to the integrator whose output is :

a) Rising linear ramp signal when – 4V is applied to it

b) Falling linear ramp signal when + 4V is applied to it.

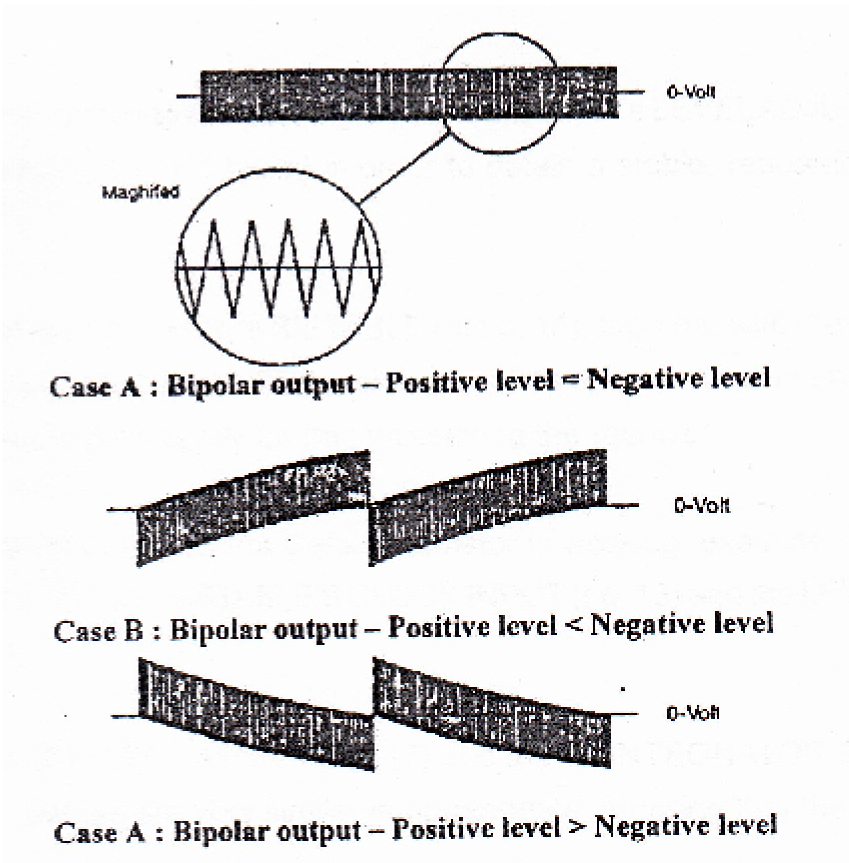
The integrator output is then connected to the – ve terminal of voltage comparator.

Procedure:

1. Make connections as per diagram.
2. Ensure that the clock frequency selector block switches A & B are in A=0 and B=0 position.
3. Now turn On the kit and see that LED glows.
4. In order to ensure for the correct operation of the system, we first connect 0 volts to the +ve input of the comparator. Now observe the output of the integrator 1 (i.e. tp 17) and the output of transmitter's level changer (i.e. tp 15). When the positive and negative output levels of the level changer will be equal the output will be a triangular waveform as shown in fig3 (Case A). When the negative level is greater than positive level, the integrator's output level will be as shown in fig3 (Case B). And when the positive output level is greater, then the integrator's output will be as shown in fig3 (Case C). The levels can be adjusted by turning the potentiometer from one extreme to another.
5. Adjust the transmitter's level changer preset until the output of integrator is a triangular wave centered at 0 volts. The peak to peak amplitude of the wave should be 0.5 volts (approx.), this amplitude is known as the integrator Step Size.
6. Now observe the output of the transmitter's bistable circuit (i.e. tp 14). It is now a stream of alternate '1' and '0'. This is the output of a delta modulator and the Delta modulator is now said to be balanced for correct operation.
7. Now examine the output of integrator at the receiver (i.e. tp 47). It should be a triangular wave with step size equal to that of integrator in transmitter and ideally centered around 0 volts.
8. Now observe the output of low pass filter. It will be a DC level centered around 0 volts. This is the output of Delta demodulator and it is balanced for correct operation.
9. Now disconnect the 0 volts from the +ve input of the comparator and reconnect it to 250 Hz signal of the function generator block. Now observe the output of voltage comparator (tp 9), integrator (tp 17). Also observe the delta modulated output at the output of bistable circuit. It has been encoded into stream of '0' and '1'.
10. Also observe the output of low pass filter in the receiver (tp 51), which is the output of demodulator.

11. Now disconnect 250 Hz from the +ve input of comparator and reconnect it to 500 Hz , 1 Khz, and 2 Khz outputs in turn. Now note the frequency of the analog signal increases, so the low pass filter's output becomes more distorted and reduced in amplitude. This effect is known as 'Slope Overloading'.

Observations:



Case A : Bipolar output – Positive level = Negative level

Case B : Bipolar output – Positive level < Negative level

Case A : Bipolar output – Positive level > Negative level

Result: The Delta modulation / demodulation and Slope Overloading effect has been studied.

Precautions:

1. The connections should be made properly and tightly.
2. Check all the connections before switching ON the kit.

<u>**EXPERIMENT No. : 05**</u>

Aim : Study of Pulse data coding techniques for NRZ formats.

Apparatus Required:

1. Data Encoding kit (Trainer 2106)
2. Data bit generator
3. Patch cords
4. CRO
5. CRO Probes

Block Diagram:

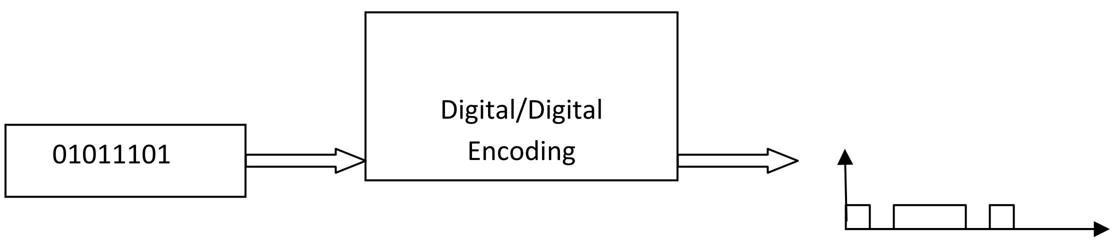
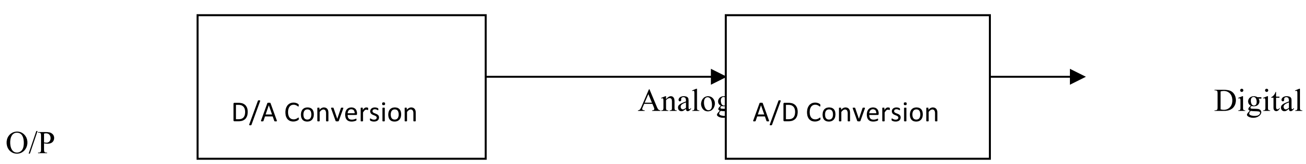


Fig 1: Digital-to-Digital Encoding



Theory: Digital to Digital conversion is the representation of digital information by a digital signal. In this conversion, the binary 1's and 0's generated by a computer are translated into a sequence of voltage pulses that can be propagated over a wire. Fig1 shows the relationship between the digital information, the digital-to-digital encoding hardware and the resultant digital signal. There are many mechanisms for digital-to-digital conversion; these are unipolar, polar and bipolar encoding/conversion. In our present experiment we are using polar conversion method.

Polar Encoding: It uses two-voltage levels- one positive and one negative. Of many existing variations of polar conversion we will examine only the three most popular: nonreturn to zero (NRZ), return to zero (RZ), and biphase. NRZ encoding includes two methods: nonreturn to zero, level (NRZ-

L), and nonreturn to zero, invert (NRZ-I). Biphase also refers to two methods. The first, Manchester, is the method used by Ethernet LANs. The second, Differential Manchester, is the method used by Token Ring LANs.

Nonreturn to Zero(NRZ): In NRZ encoding, the level of the signal is always either positive or negative. The two most popular method of NRZ transmission are:

NRZ-L: In this encoding method, the level of the signal depends on the type of bit it represents. A positive voltage usually means the bit is a 0, and a negative voltage means the bit is a 1(or vice-versa); thus, the level of the signal is dependent upon the state of the bit.

NRZ-I: In this method, an inversion of the voltage level represents a 1 bit. It is the transition between a positive and negative volatage, not the voltages themselves, that represents a 1 bit. A 0 bit is represented by no change.

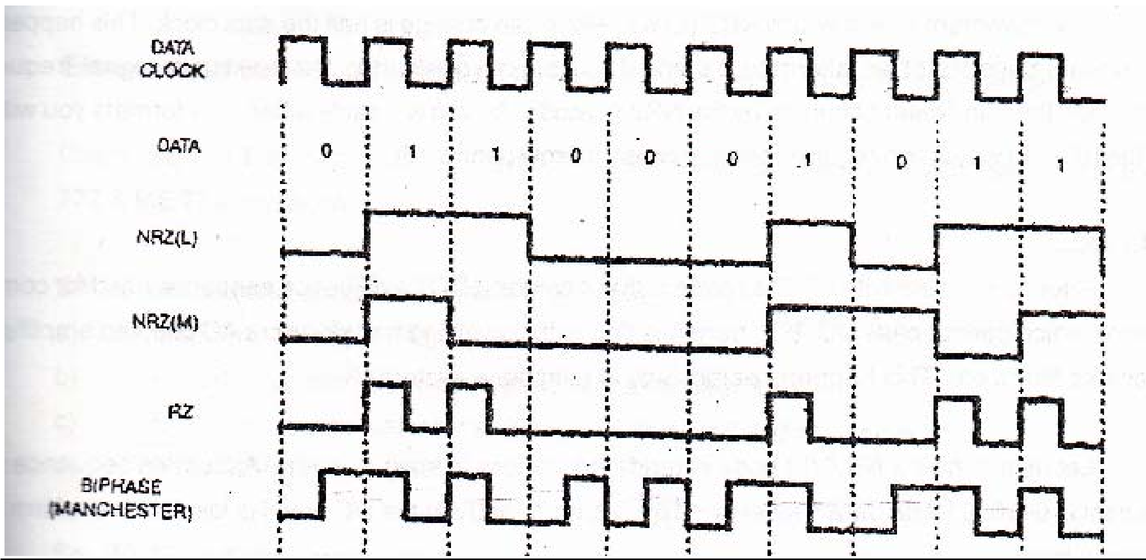
Return to Zero: This method uses three values: positive, negative and zero. The signal changes not between bits but during each bit. A positive voltage means 1 and negative voltage means 0.

Biphase: In this the signal changes at the middle of the bit interval but does not return to zero. Instead it continues to the opposite pole.

Procedure :

1. Data is generated with the help of a data bit generator.
2. Connect the data O/P of the data generator to the Tx data I/P of the trainer 2106.
3. Now connect the clock of the generator to the Tx clock of the kit and ground with the ground terminal of the kit.
4. Select the data on the data generator and load it in the trainer 2106 by pressing load button.
5. Now observe the O/P of the NRZ-L, NRZ-M and Biphase.

bservations :



Result :

Different pulse data coding techniques has been studied.

Precautions:

1. The connections should be made properly and tightly.
2. Check all the connections before switching ON the kit.

EXPERIMENT No. : 06

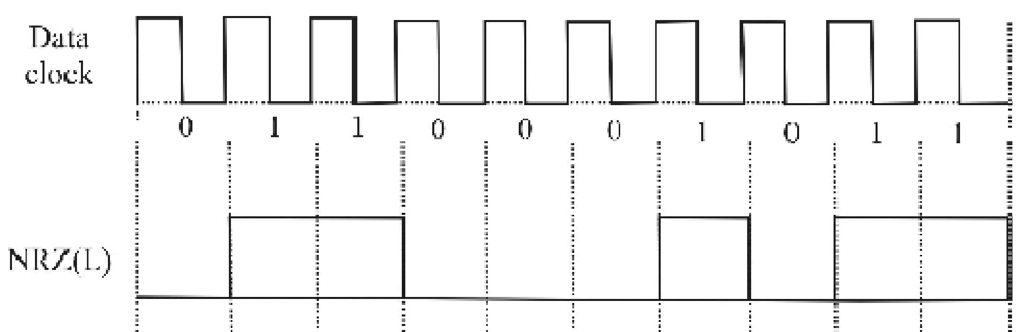
Aim : Study of data decoding techniques for NRZ formats.

Apparatus Required:

1. Data Encoding kit (Trainer 2106)
2. Data bit generator
3. Patch cords
4. CRO
1. CRO Probes

Theory: The symbols ‘0’ and ‘1’ in digital systems can be represented in various formats with different levels & waveforms. The selection of particular format for communication depends on the system bandwidth, system’s ability to pass DC level information, error checking facility, ease of clock regeneration & synchronizations at receiver, system complexity & cost etc.

Non - Return To Zero (Level) NRZ (L) :
It is the simplest form of data representation. The NRZ (L) waveform simply goes low for one bit time to represent a data '0' & high for one bit time to represent a data '1'. Thus the signal alternates only when there is a data change. See figure 1



NRZ (L) Encoding (Figure 1)

Clock Regeneration :
Since the level transition takes place at a predetermined moment (e.g. at rising/falling edge of the data clock), it is possible to extract clock information at the receiver. However the synchronization & clock information is sparse & sometimes even lost when a long stream of zero or ones are encountered. The clock regeneration is very difficult in such cases. This makes the clock regeneration design more complex.

Bandwidth :
The maximum rate at which NRZ (L) waveform can change is half the data clock. This happens when the data stream consists of an alternate 0's and 1's. As it is known, it is the maximum signal frequency which determines the bandwidth occupied by the NRZ (L) code. As you will study other data formats you will appreciate that the NRZ (L) waveform requires comparatively narrow bandwidth.

DC Levels :

Another problem with NRZ (L) code is that it contains DC Level hence cannot be used for communication systems which cannot pass DC. e.g. transmission paths involving transformers AC coupled amplifiers or series capacitors filters etc. This happens particularly in telephone systems.

Let us see now an NRZ (L) code is rendered useless in such systems. Assume a sequence of repetitive data sent is 0110001 with data 1 level at + 5V & data '0' at 0V. If the DC Level is lost, the waveform balances at the mean level.

Mean level = total value of samples ÷ no of samples.

= (0+5+5+0+0+0+5) ÷ 7 = 15 ÷ 7 = 2.14V

Thus if the DC Level information is lost, the whole signal balances about 2.14V.

Thus the peak value of + 5V will shift to 5 - 2.14 = 2.86V

It may slip down to a level where the receiver cannot recognize as level '1' & thus the data could be misread. In extreme case where the input is constant series of logic 0's then the NRZ (L) output would be a constant level. Now if the input changed to a stream of logic 0's, the output would still be a constant level. The only difference is the DC Level. Therefore if the DC Level information is lost, we have no way of knowing whether the original input will have all 0's or all 1's.

Non - Return - To Zero (Mark) : [NRZ (M)] :

The NRZ (M) code is very much similar to the NRZ (L) code. Here if logic 1 is to be transmitted. The new level is inverse of the previous level i.e. change in level occurs. It a data '0' is to be transmitted the level remains unchanged. Thus in the case of NRZ (M) waveform the present level is related to the previous levels. See figure 2.

Thus, no longer the absolute value of signal is necessary instead it is the change in the level for which we look now.

Remember, ----- A change means a logic '1'

----- No change means logic '0'

NZR (M) Format

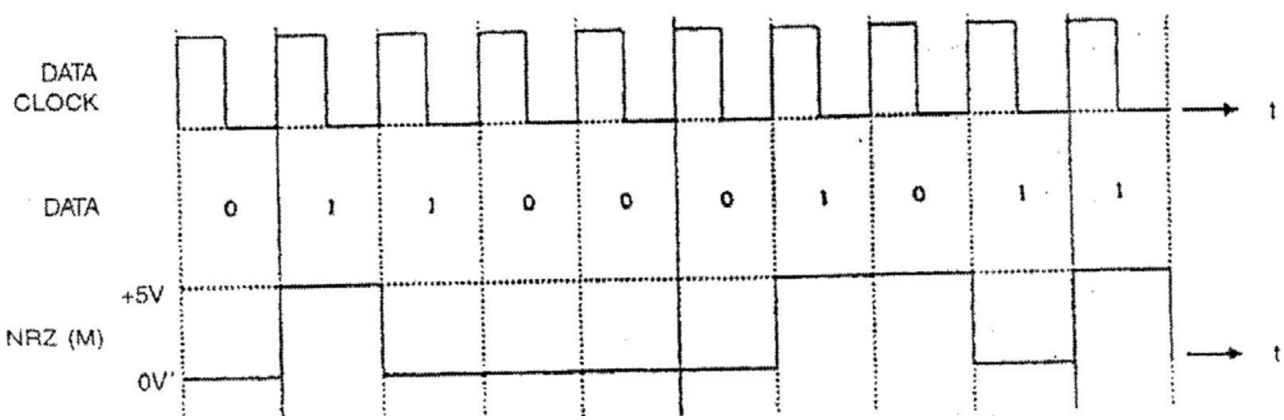


Figure 2

Clock Recovery :
The receiver can extract the timing information from the NRZ (M) waveform unless there are long periods of no level changes corresponding to long streams of ‘0's. Here long stream of 1's are not a problem as in NRZ (L) because now it causes a level change continuously & the receiver can easily extract the clock information. This is a slight improvement over NRZ (L) waveform.
Band Width A DC Level :
The NRZ (M) is simil ar to NRZ (L) waveform in respect of the bandwidth utilized & the passing of DC Levels. A considerable advantage of NRZ (M) is that it is independent of the absolute level of the incoming data. The receiver simply has to know the level changes. This is an advantage in phase shift keying as will be discusses later on, where the receiver looks for a change in phase of the incoming signal.
Decoding :
The NRZ (M) can be converted to NRZ (L) code by a bit decoder. The bit decoder samples the incoming data bit, holds it for a moment takes a new sample & compare the two, to see whether the changes has occurred. If it has occurred it gives output logic '1' & if not it gives outputs logic '0'. This is the required NRZ (L) code.

Procedure for NRZ (M) Coding :
Steps (1) to (13) : Follow the set up procedure for steps 1 to 13 as given in experiment1. Also Set pulse generator delay adjust potentiometer fully clock wise in step 6.
1. Monitor the NRZ (M) output in data formatting circuit block (TP6). The waveform will be identical to what's shown in figure 1 before or it will be the logical inverse of the other. The reason is that the present level does not depend on the current data bit, but also of the previous level.
2. Switch off the instruments. Make the following additional connections as shown in figure 4.
a. Between **ST2106** and **ST2107**

ST2106	**ST2107**
NRZ (M) output (TP6) to	Bit decoder input (TP39)

b. Between **ST2107** and **ST2104**

ST2107	**ST2104**
Bit decoder (TP39) input Clock regeneration circuits input (TP3)	
Bit decoder output (TP40) to	PCM data input (TP1)
ST2104	**ST2107**
Clock regeneration circuit to	Clock input (TP41) Output (TP8)

On **ST2104** trainer connect clock regeneration circuit's output (TP8) to RX clock input (TP46)
3. Switch on the power.
4. The bit decoder samples the NRZ (M) waveform at the centre of each data bit. If the sampled level bit is same as the previous level, the decoder output a low level or logic '0'. If however, the sampled data level differs from the previous one, the decoder's output is logic '1'. Thus the decoders output is a NRZ (L)
signal which can be monitored at decoder's output (TP40). Observe that the output is identical to **ST2106** NRZ (L) output signal (TP5) but delayed one bit time. Also observe the bit decoder output (TP40) with respect to clock input (TP4)
5. Switch on the **ST2103** trainer’s pseudo - random sync code generator.
6. Observe that now the receiver is frame synchronized to the transmitter. This is indicated by the fact that **ST2103** trainer's A/D converter LEDs and **ST2104** trainer's D/A converter LEDs carry the same data.

7. Also observe that now the NRZ (M) waveform is non repeatable. This is because the sync data bit is different for different frames.
8. Switch off the trainers. On **ST2103** disconnect the two inputs to channel 0 & channel 1. Instead connect CH 0 to 1 KHz signal & CH.1 to 2 KHz signal.
9. Switch on the power. Observe the two channel outputs on **ST2104** trainer (TP33 & TP36) simultaneously. Use dual trace oscilloscope. Also observe that there is no interference between the two waveforms and the output changes in amplitude by varying the function generator potentiometer.
Result : Observed & traced the waveform of NRZ(L) and NRZ(M).

Precautions:

1. The connections should be made properly and tightly.
2. Check all the connections before switching ON the kit.
3. Set the correct D.C. level of NRZ(L) coding.

Pre Experiment Questions:
1. What is NRZ(L) and NRZ(M) coding techniques.
2. The effect of D.C. level of NRZ(L) coding techniques.
3. Circuit of coding waveform generation.

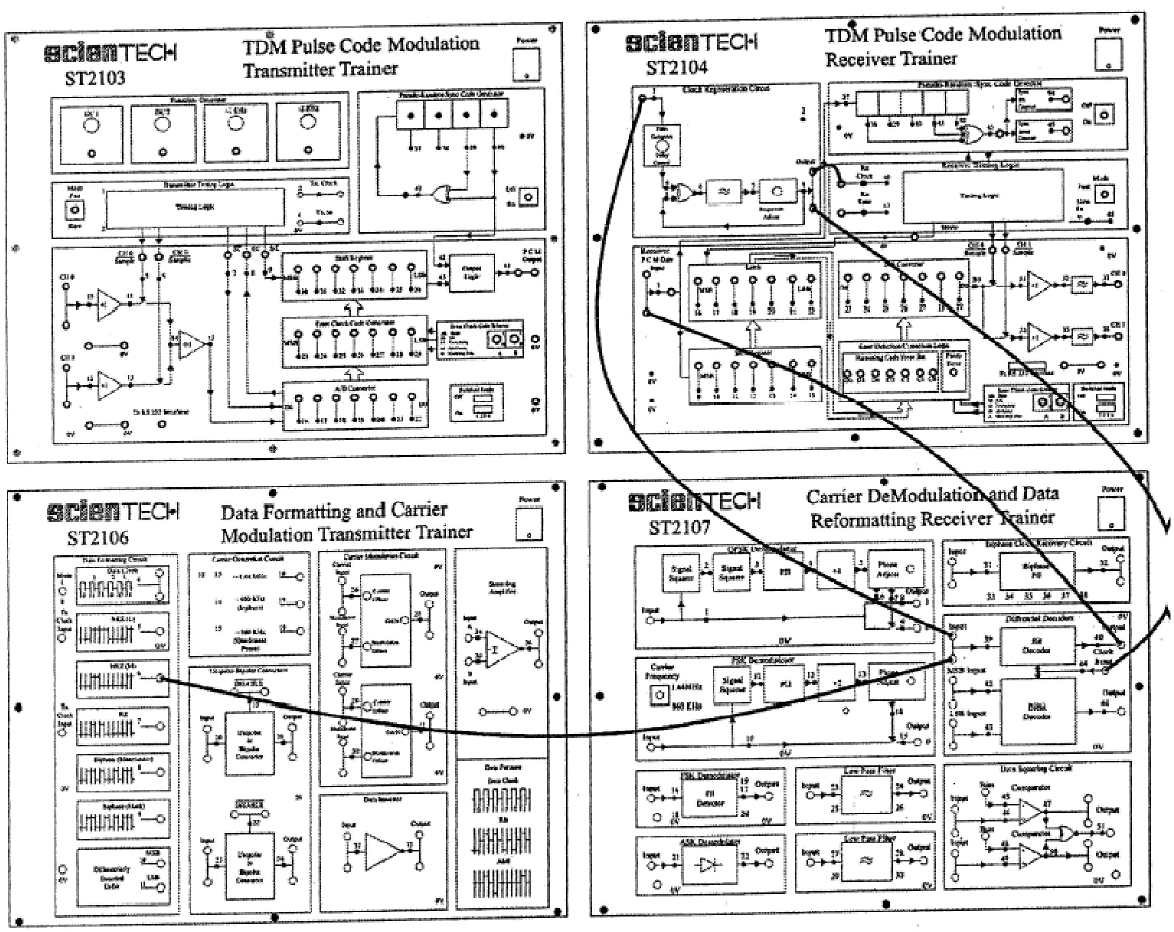


Figure 4

EXPERIMENT No. : 07

Aim: Study of Manchester coding and decoding.

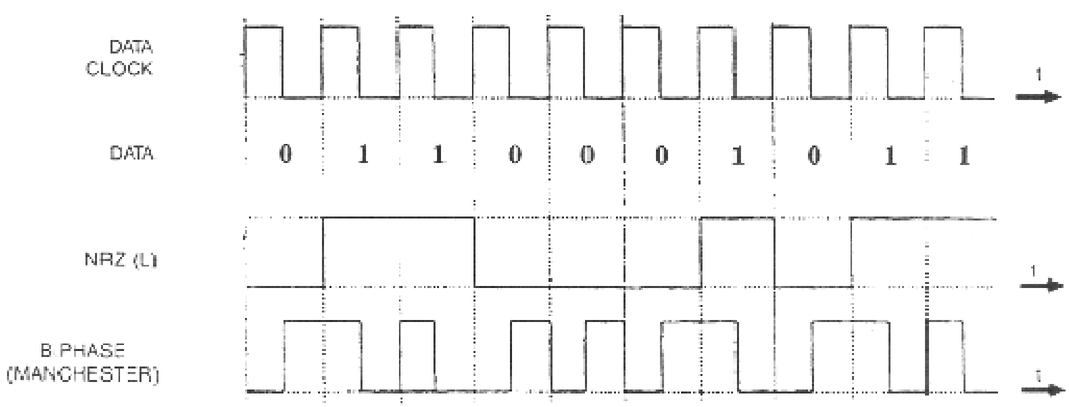
Apparatus Reqired :

1. Data Encoding kit (Trainer 2106)
2. Data bit generator
3. Patch cords
4. CRO
5. CRO Probes

Theory:

The main disadvantage with all the previous formats is their inability to provide reliable clock synchronized information to the receiver clock. biphase codes overcome this problem by providing the transition in both 0's and '1's. The two most common biphase codes in practice are biphase (Manchester) & biphase (Mark) codes. Also these codes are independent of the DC Levels i.e. they have zero DC component.

Biphase (Manchester) Coding :
The encoding rules for biphase (Manchester) code are as follows. A data '0' is encoded as a low level during first half of the bit time and a high level during the second half. A data '1' is encoded as a high level during first half of the bit time and a low level during the second half. Thus string of l's or 0's as well as any mixture of them will not pass any synchronization problem in receiver. Figure 1 shows the biphase
(Manchester) waveform for a given data stream.



Biphase (Manchester) Format (Figure 1)

Bandwidth :
The Biphase (Manchester) code always contains at least one transition per bit time, irrespective of the data being transmitted. Hence the maximum frequency of the Biphase (Manchester) code is equal to

the data clock rate when a stream of consecutive data '1' & '0' is transmitted. Therefore the required bandwidth is same as that of the RZ code & double as that of the NRZ (L) code.

DC Level :

Since the biphase (Manchester) code has a high level for half of each data bit time & low level for second half irrespective of the data. The effective DC level of the biphase coded waveform is zero. This allows it to be used in AC coupled communication systems.

Problem In Decoding :

This form of coding certainly provides plenty of rising edges for clock synchronization but they do not all occur at same time e.g. we have a rising transition at the start of code for data 'I' where as for data '0' we have it at the midway of the data bit time. This causes confusion in the clock regeneration circuit. To overcome this, we employ a special biphase clock recovery circuit which can be synchronized by the rising edge occurring at either time.Rest of the decoding is same as for the RZ code. Since the valid data is carried for in first half of each clock period, we ensure that the regenerated receiver clock's rising edge occurs at this time.

Procedure for Biphase (Manchester) Code and its Detection

Steps (1) to (13) : Follow the set up procedure for steps 1 to 13 as given in experiment . Also set pulse generator delay adjusts fully clockwise in step 6.

1. Use the other trace of your oscilloscope to observe the biphase (Manchester) waveform at (TP8) of **ST2106**. This is identical to the waveform shown in figure1.

2. To view the complete waveform adjust the time base & X- position controls until you have exactly two clock pulse within each of the oscilloscope's vertical graticule lines. Monitor biphase (Manchester) waveform again.

3. Readjust the time base & X-Position controls to original set-up again. Switch off the power. As it has been stated earlier in the biphase (Manchester) theory section, we need a biphase clock recovery circuit, on **ST2107** trainer. The function of biphase clock recovery circuit is that it accepts rising edges at both

the centre of a bit interval as well as at the start. Its output is a square wave whose period is equal to one bit interval & where as rising edge occupies one quarter (1/4) of the way through the bit interval.

4. Make the following additional connections. (see figure 2) Biphase (Manchester) output (TP8) on **ST2106** to biphase clock recovery circuit input (TP31) on **ST2107**.

5. Make the following connections between **ST2107** and **ST2104** trainers.

6. Biphase clock recovery circuit input (TP31) ‘On’ **ST2107** to PCM data input (TP1) on **ST2104**.

7. Biphase clock recovery circuit output (TP32) on **ST2107** to 'RX clock input (TP46) on **ST2104** see figure 2.

8. Switch on the trainers. On close observation of the regenerated data output.We see that the frequency of the signal is same but is quarter cycle delayed as compared to the data clock.

9. Switch off the trainers. Disconnect the CH0 & CH 1 inputs & connect 1 KHz output to CH0 & 2 KHz output to CH 1.

10. Turn ‘On’ the power. Observe the two channel outputs on **ST2104** trainer (TP33 & 36). Also observe they are independent is found, it can be removed by adjusting the pulse generator delay adjust potentiometer control slightly.

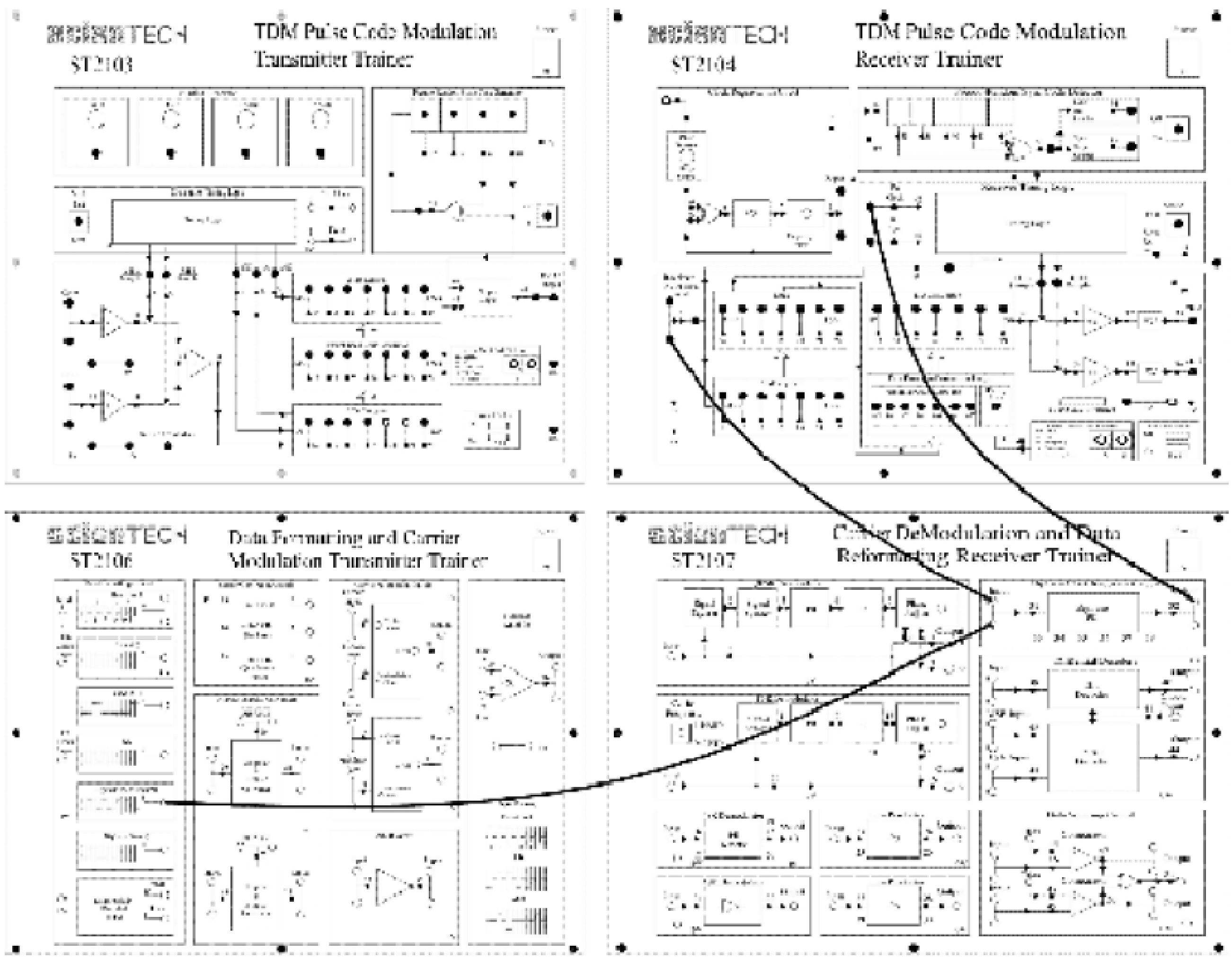


Figure 2

Result : Observed & traced the waveform of Manchester code.

Precautions:

1. The connections should be made properly and tightly.
2. Check all the connections before switching ON the kit.
3. Set the correct D.C. level of Manchester coding.

Pre Experiment Questions:

1. What is Manchester coding techniques.
2. The effect of D.C. level of Manchester coding techniques.
3. Circuit of coding waveform generation.

EXPERIMENT No. : 08

Aim: Study of Amplitude Shift Keying Modulator and Demodulator.

Apparatus Required: ASK Modulator/Demodulator Trainer kit (ST 2106, ST 2107), Binary Data Generator, CRO, CRO probes.

Theory:

Carrier Modulation Introduction: To transmit the digital data from one place to another, we have to choose the transmission media. The simplest possible method to connect the transmitter to the receiver with a piece of wire. This works satisfactorily for short distances in some cases. But for long distance communication & in situations like communication with the aircraft, ship, and vehicle this is not feasible. Here we have to opt for the radio transmission. It is not possible to send the digital data directly over the antenna because the antennae of practical size works on very high frequencies, much higher than our data
transmission rate. To be able to transmit the data over antenna, we have to 'modulate' the signal's phase,
frequency or amplitude etc. is varied in accordance with the digital data. At receiver we separate the signal from digital information by the process of demodulation. After this process we are left with high frequency signal (called as carrier signal) which we discard & the digital information, which we utilize.
Modulation also allows different data streams to be transmitted over the same channel (transmission medium). This process is called as 'multiplexing' & results in a considerable saving in no. of channels to be used. Also it increases the channel efficiency. The variations of particular parameter variation of the carrier wave give rise to various modulation techniques. Some of the basic modulation techniques are
described as under.

a. Amplitude Shift Keying (ASK) :
In this technique modulation involves the variation of the amplitude of the carrier wave in accordance with the data stream. The simplest method of modulating a carrier with a data stream is to change the amplitude of the carrier wave every time the data changes. This modulation technique is known *amplitude shift keying*.
The simplest way of achieving amplitude shift keying is by switching 'On' the carrier whenever the data bit is '1' & switching off. Whenever the data bit is '0' i.e. the transmitter outputs the carrier for a' 1 ' & totally suppresses the carrier for a '0'. This technique is known as '*On-Off' keying* figure 20 illustrates the amplitude shift keying for the given data stream.
Thus, Data = 1 carrier transmitted & Data = 0 carrier suppressed.

The ASK waveform is generated by a balanced modulator circuit, also known as a *linear multiplier*. As the name suggests, the device multiplies the instantaneous signal at its two inputs. The output voltage being product of the two input voltages at any instance of time. One of the input is AC coupled 'carrier' wave of high frequency. Generally, the carrier wave is a sine wave since any other waveform would increase the bandwidth, without providing any advantages. The other input which is the information signal to be transmitted, is DC coupled. It is known as modulating signal.

ASK modulation

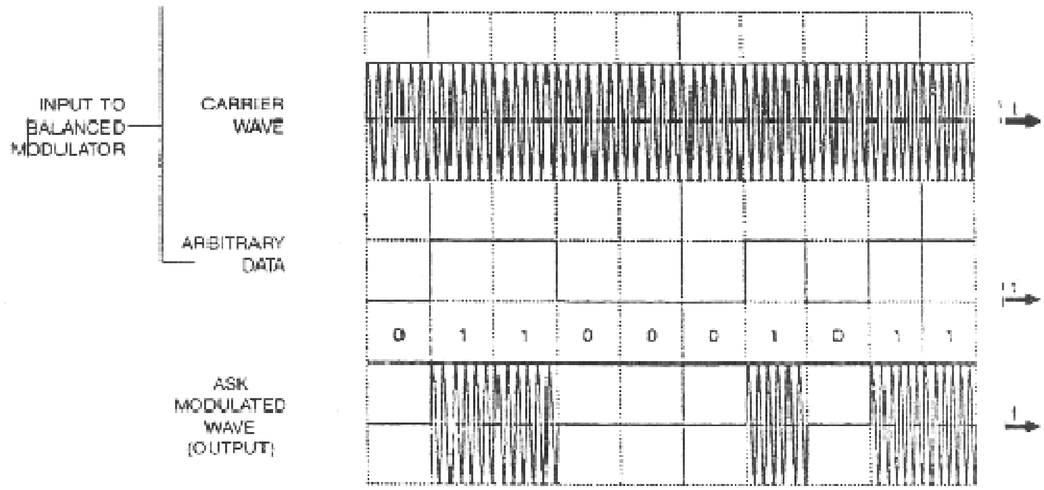


Figure 1: ASK Waveform

In order to generate ASK waveform it is necessary to apply a sine wave at carrier input & the digital data stream at modulation input. The *double - balanced modulator* is shown in figure 2.

Amplitude Shift Keying:

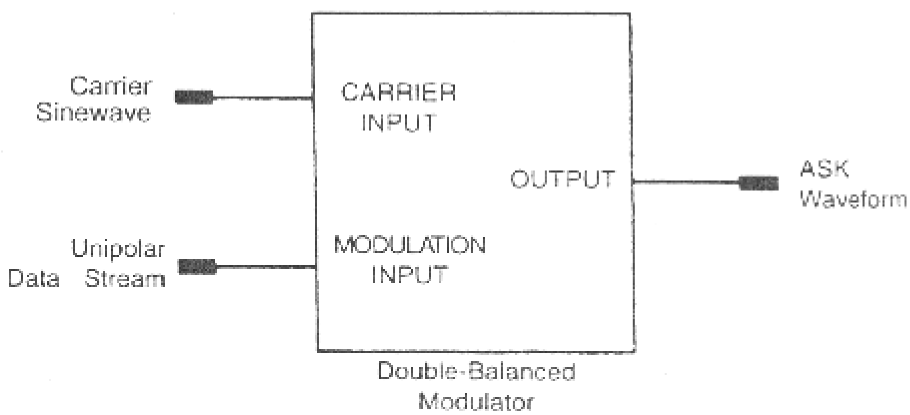


Figure 2: ASK Modulator block diagram

The data stream applied is unipolar i.e. 0 volts at logic '0' & + 5 Volts at logic '1'. The output of balanced modulator is a sine wave, unchanged in phase when a data bit ‘l' is applied to it. In this case the carrier is multiplied with a positive constant voltage when the data bit '0' is applied, the carrier is multiplied by 0 volts, giving rise to 0 volt signal at modulator's output.

The ASK modulation result in a great simplicity at the receiver. The method to demodulate the ASK modulation results in a great simplicity at the receiver. The method to demodulate the ASK waveform is to rectify it, pass it through the filter & 'Square Up' the resulting waveform. The output is the original data stream. Figure 3 shows the functional blocks required in order to demodulate the ASK waveform at receiver.

ASK Demodulator:

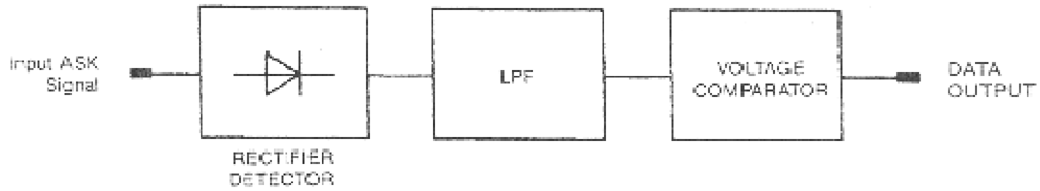


Figure 3: ASK Demodulator block diagram

The various steps involved are summed below :

Step A : The ASK waveform is *rectified by a diode* rectifier, giving a positive going signal. This signal is too rounded to be used as digital data. Also the carrier component is still present & it is of unreliable amplitude due to the attenuation & noise in transmission path. In fact it is a great drawback associated with ASK modulation. The data level may be misinterpreted by the receiver if the amplitude change is too much.

Step B : After rectification, the signal is *passed through the low pass filter* to remove the carrier component. This result in slightly rounded pulses of unreliable amplitude.

Step C : These rounded pulsed are then 'Squared Up' (i.e. shaped in a square wave fashion) by passing it through voltage comparator set at a threshold level. If the input voltage exceeds the threshold level, the comparator output is a +5V signal and in other case it is O\Z Thus at the end we have the true copy of the original input data see figure 4.

ASK Modulation:

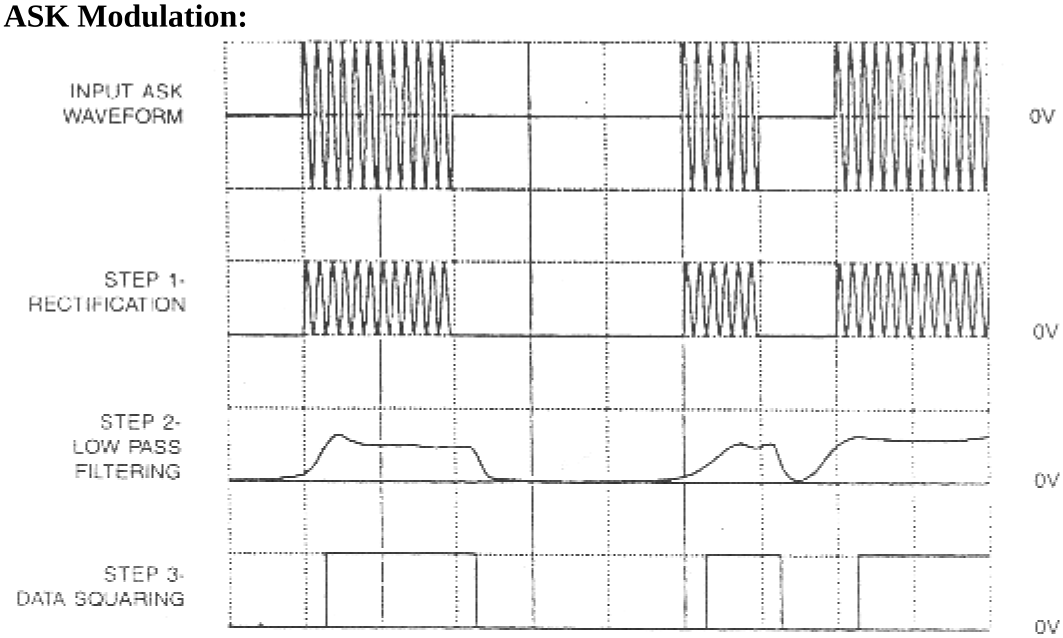


Figure 4

Procedure :

ASK Modulation and Demodulation:

1. The experiment makes use of four trainers namely **ST2103, ST2104, ST2106** & **ST2107. ST2103** TDM pulse code modulation transmitter trainer serves as a data source while **ST2104** TDM pulse code modulation receiver trainer serves as analog signal recover. **ST2106** serves as data formatting (conditioning) device while **ST2107** reformats (recondition) the data. **ST2103** & **ST2106** Trainers serves as transmitter for our system & **ST2107** & **ST2104 t**rainer serves as receiver.
2. Ensure that all trainers are switched off, until the complete connections are made.
3. Check **ST2104** Trainer's clock regeneration circuit Set up for correct operation as given at the end of **ST2103 / 4** work book
4. Set up the following conditions on **ST2103** trainer
a. Mode switch set in FAST position.
b. Pseudo - random sync code generator switched on.
c. Error check code selector switches A & B in A=0 & B=0 positions.
d. All switched faults 'Off'
5. Set **ST2106** trainer's mode switch in position 1
6. Set up following conditions on **ST2104** trainers:
a. Mode switch set in FAST position
b. Pseudo - random sync code detector in 'On' position.
c. Error check code selector switch A & B in A = 0 & B = 0 position.
d. All switched faults to be kept 'Off'
7. Make the following connections between **ST2103, ST2106** trainers.
ST2103 trainer ST2106 trainer
a. TX clock output (TP3) to TX clock input
b. PCM output (TP44) to TX data input
8. Connect the TX to output (TP4) on **ST2103** trainer to external trigger input of the oscilloscope. Set to negative edge triggered mode in oscilloscope. It may be necessary to adjust the trigger level manually to obtain a stable waveform.
9. n **ST2103** trainer make following connections
a. DC 1 to CH 0 input
b. CH 0 input to CH 1 input
This is done to supply the same voltage level to each of the two time division multiplexed channels. Thus we are able to get the same data stream for any time frame.

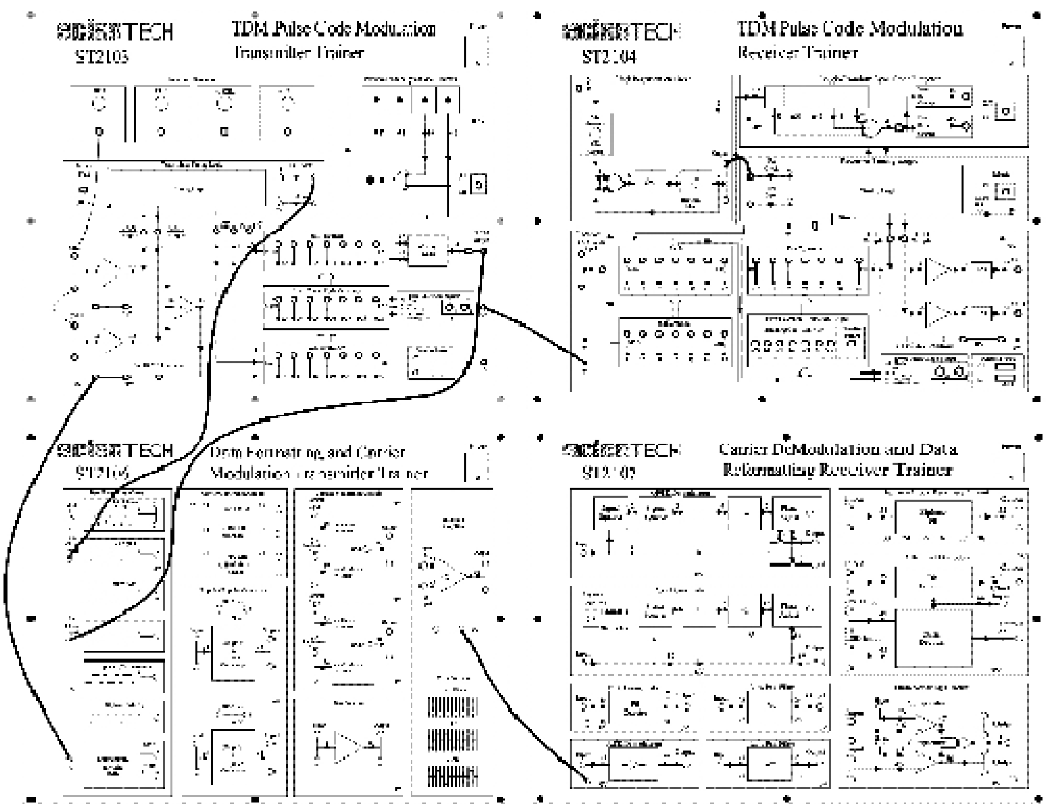


Figure 5

10. Make the rest of the connections as shown in configuration figure 5.

11. Switch ‘on’ the power

12. On **ST2103** trainer adjust the DC1 potentiometer until the 7 bit code displayed at A/D converter LEDs is D6 D5 D4 D3 D2 D1 D0

0 1 0 0 0 1 1

13. Observe the data clock output at TP4 on **ST2106** trainer's data format block with Oscilloscope. Adjust the oscilloscopes time base & position control until each rising edge of data clock coincides with one of scope's vertical graticule line as shown in figure 6. Each main division on scope's horizontal axis now represents one data bit time. Adjust the trigger level (manually, if necessary, to obtain a stable trace.) This sets convenient reference against which to observe the other wave forms.

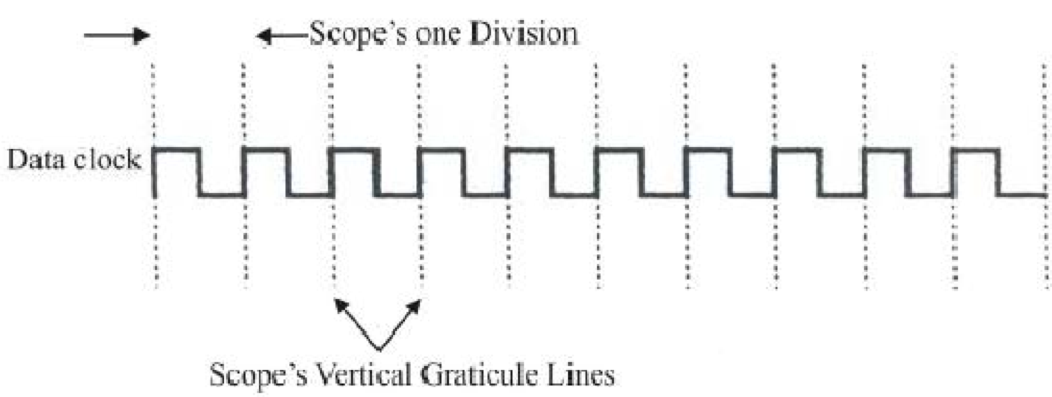
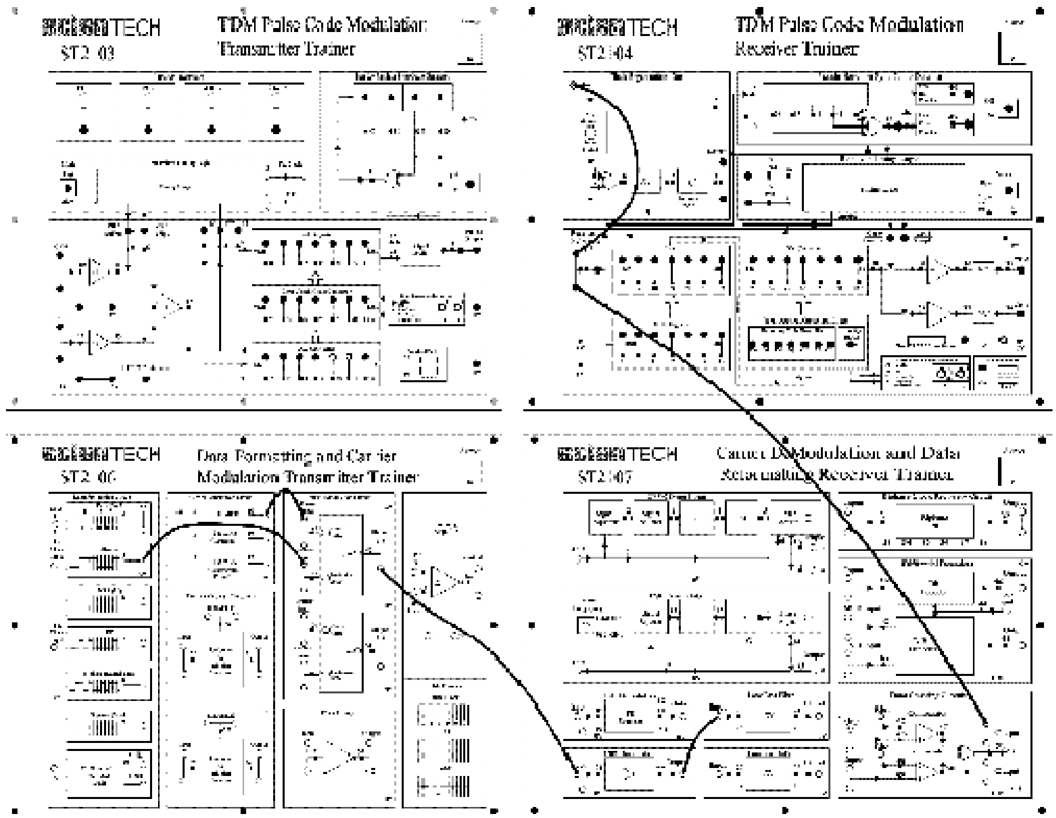


Figure 6 (Data clock output)

14. Switch off the power. Connect additional board as shown in figure 7 as follows :
a. On ST2106 trainer : i) NRZ (L) output (TP5) to carrier modulation circuit's modulation input (TP27).
ii) 1.44 MHz carrier output (TP16) to carrier input socket of modulator circuit (TP26).
b. Between ST2106 and ST2107 :
i.) Modulator 1 output (TP28) to ASK demodulator input (TP21).
c. On ST2106 and ST2107 :
i.) ASK demodulator output (TP22) to Low Pass Filter 1 input (TP23).
ii.) Low Pass Filter 1 output (TP24) to comparator 1 input (TP46)
d. On ST2104 trainer :
i.) PCM. data input (TP1) to clock regent circuit input (TP3)
ii.) Clock regent circuit output TP8 to RX clock input (TP46)
e. Connect :
i.) Comparator 1 output (TP47) on **ST2107** to PCM data input (TP1) on **ST2104.**
15. Turn 'On' the trainers : Monitor NRZ (L) output (TP5) from **ST2106** trainer on one channel of the oscilloscope Use the other channel to monitor the output of modulator 1 (TP28) in **ST2106** trainer.
16. Three variables have been provided in the modulators block. Their use may be necessary to obtain a required ASK waveform. These variables are
a. Gain : This pot adjusts the amplification of the modulator's output. Adjust this pot till the output is not a 2Vpp signal in 'On' state.
b. Modulation Offset : This control is used to adjust the amplitude of the 'Off' signal. Adjust this control till the amplitude of the 'Off' signal is an close to zero as possible.
c. Carrier Offset : This control adjusts the 'Off' bias level of the ASK waveform. Adjust this control till the 'Off' level occurs midway between the 'On' signal peaks.
17. To see the demodulation process, observe the output at the ASK demodulator (TP22) & low pass filter (TP24) on **ST2107** trainer.
18. The last stage of demodulation is 'squaring up' of filter output. In order to achieve this it is necessary to adjust the bias level for comparator 1 so that the output has the correct pulse width.
19. Adjust it till the output signal pulse width is not similar to the NRZ (L) data pulse width. You can observe these two simultaneously on the dual trace oscilloscope. The two will be identical once the bias level is adjusted except for short delay between them.
20. Turn 'On' the pseudo-random sync code generator on **ST2103** trainer. This pulls the transmitter & receiver in 'Frame - Synchronization'. Observe the A/D Converter LEDs on **ST2103** trainer & D/A Converter LEDs on **ST2104** trainer. Now they will be carrying the same data. Change the position of the DC l potentiometer Observe that the same change is reflected at the receiver.
21. Turn 'Off' the power : Disconnect CH0 & CH1. Instead connect them to 1 KHz & 2KHz signal respectively. Turn 'On' the trainers. & check the constructed analog output on **ST2104** CH 0 & CH 1 (TP33 & 36).
22. You can use any data format available on **ST2106** trainer to modulate the carrier. Remember to demodulate the carrier as described above & convert the date format back to NRZ (L) format by means of reformatting techniques described in the earlier sections.
23. The same experiment can be performed by using 960 KHz (1) signal instead of 1.44 MHz signal.



TDM Pulse Code Modulation Transmitter Trainer
ST2103
TDM Pulse Code Modulation Receiver Trainer
ST2104
Data Formatting and Carrier Modulation Transmitter Trainer
ST2106
Carrier Demodulation and Data Reformatting Receiver Trainer
ST2107

Figure 7

EXPERIMENT No. : 09

Aim: Study of Frequency Shift Keying Modulator and Demodulator.

Apparatus Required: FSK Modulator/Demodulator Trainer kit (ST 2106, ST 2107), Binary Data Generator, CRO, CRO probes.

Block Diagram:

Block Diagram:

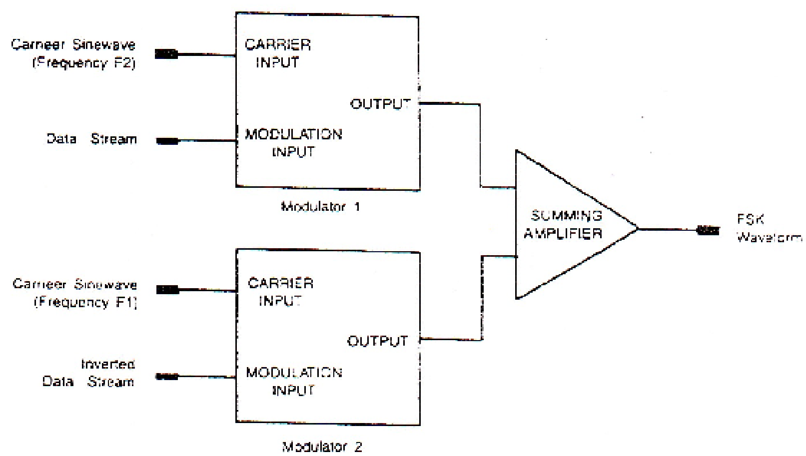


Fig1 : FSK MODULATOR

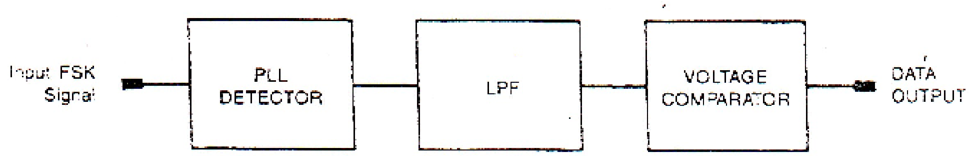


Fig2 : FSK DEMODULATOR

Theory: In Frequency shift keying, the carrier frequency is shifted (i.e. from one frequency to another) corresponding to the digital modulating signal. If the higher frequency is used to represent a data ‘1’ & lower frequency a data ‘0’, the resulting FSK waveform appears. Thus

Data =1 High Frequency

Data =0 Low Frequency

It is also represented as a sum of two ASK signals. The two carriers have different frequencies & the digital data is inverted. The demodulation of FSK can be carried out by a PLL. As known, the PLL tries to 'lock' the input frequency. It achieves this by generating corresponding O/P voltage to be fed to the VCO, if any frequency deviation at its I/P is encountered. Thus the PLL detector follows the frequency changes and generates proportional O/P voltage. The O/P voltage from PLL contains the carrier components. Therefore to remove this, the signal is passed through Low Pass Filter. The resulting wave is too rounded to be used for digital data processing. Also, the amplitude level may be very low due to channel attenuation.

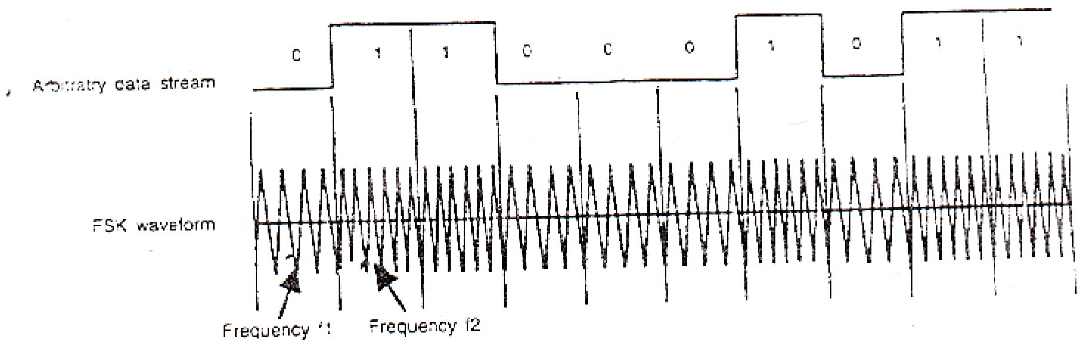
Procedure:

1. Turn ON the trainer kits. Monitor NRZ(L) O/P (t.p.5) from ST2106 trainer on one channel of the CRO. Use the other channel to monitor the O/P of modulator 1 (t.p. 28) in ST2106 trainer.
2. Observe the O/P of the summing amplifier on the ST 2106 trainer at t.p. 36. Note that it is the FSK waveform for the given data. Adjust the GAIN control of modulator 2, if necessary to make the amplitude of two frequency components equal.
3. Display the FSK waveform simultaneously with NRZ(L) O/P. Observe that, for data bit '0' the FSK signal is at lower frequency (960 Khz) & for bit '1', the FSK signal is at higher frequency (1.44 Mhz).
4. Now, to study about demodulator, examine the input (t.p 16) & the O/P (t.p 17) of ST 2107 FSK demodulator. The PLL detector has been used as the FSK demodulator.
5. The unwanted frequency component is removed by passing it through the LPF. On a dual trace oscilloscope examine the I/P (t.p 23) & O/P (t.p 24) of ST 2107 LPF1 simultaneously. Observe that the O/P contains no carrier frequency components.
6. The rounded O/P of the LPF is removed by passing it through the Data Squaring Circuit but prior to it, the BIAS level of the comparator1 is to be adjusted to a value until the O/P pulse width (t.p 47) is same as the NRZ(L) input t.p 5 on ST2106.

Observations:

DATA = 1 HIGH FREQUENCY

DATA = 0 LOW FREQUENCY



FSK WAVEFORM

Result:

The FSK modulator and demodulator circuit has been studied.

Precautions:

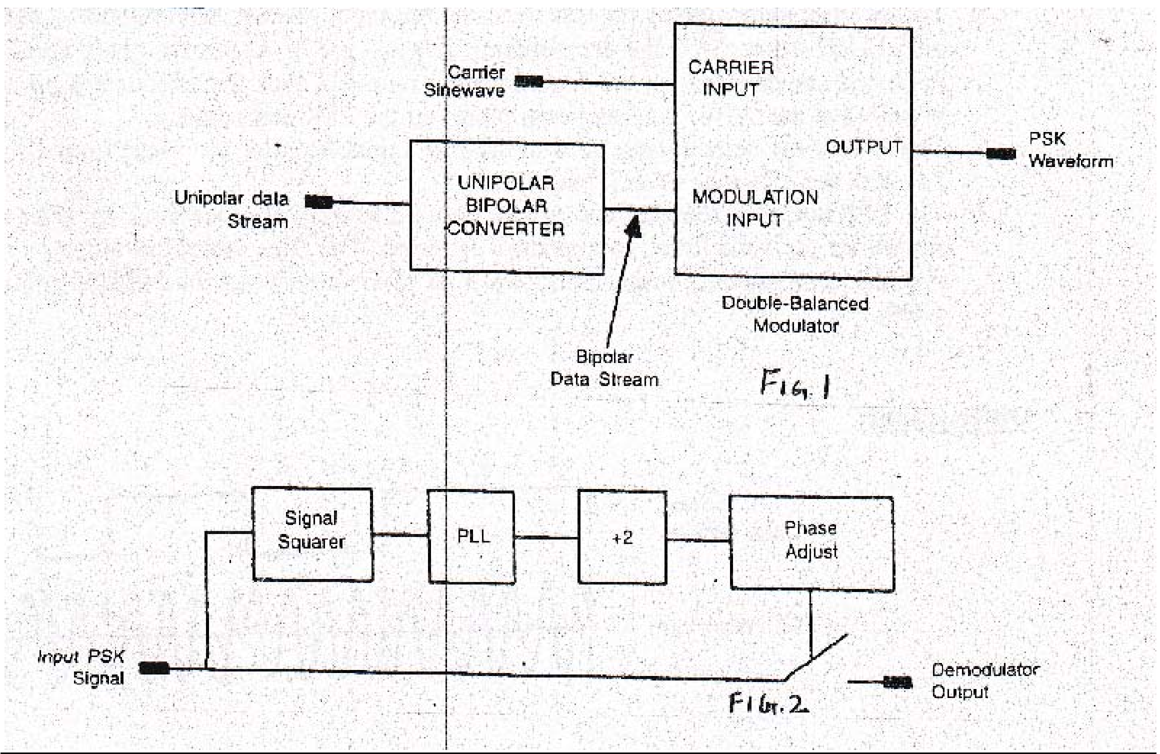
1. Check the connections before switching ON the kit.
2. Observations should be taken properly.

EXPERIMENT No. : 10

Aim: Study of Phase Shift Keying Modulator and Demodulator.

Apparatus Required: PSK Modulator/Demodulator Trainer kit, Binary Data Generator, CRO, CRO probes.

Block Diagram:

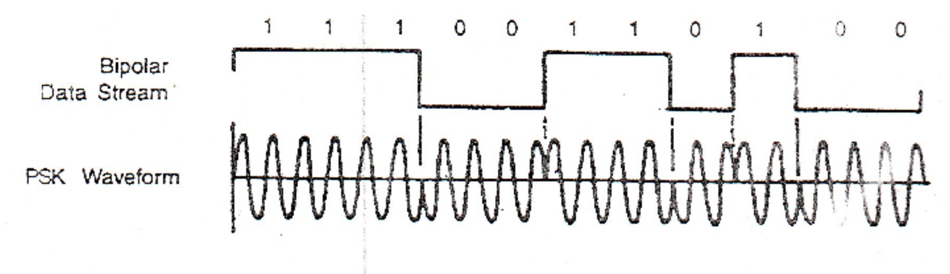


Theory: Phase shift keying involves the phase change of the carrier sine wave between 0 and 180 in accordance with the data stream to be transmitted. PSK is also known as Phase reversal keying. PSK modulator is shown in figure 1. Functionally, the PSK modulator is very similar to the ASK modulator. Both uses balanced modulator to multiply the carrier with the modulating signal. But in contrast to ASK techniques, the digital signal applied to the modulator input for PSK generation is bipolar i.e. have equal +ve and –ve voltage levels. The unipolar – bipolar converter converts the unipolar data stream to bipolar data. At receiver, the square loop detector circuit is used to demodulate the transmitted PSK signal. The demodulator is shown in figure 2. The incoming PSK signal with 0 & 180 phase changes is first fed to the signal square, which multiplies the input signal by itself. The phase adjust circuit allows the phase of the digital signal to be adjusted w.r.t the input PSK signal. Also its O/P controls the closing of an analog switch. When the output is high the switch closes and the original PSK signal is switched through the detector.

Procedure:

1. Connections to be done on ST2106 trainer:
2. Carrier input of modulator 1 to 960 Khz carrier
3. NRZ(M) output t.p 6 to unipolar-bipolar converter input
4. Unipolar-bipolar converter output tp modulator1 input.
5. Connections between ST2106 & ST2107 trainers: Modulator 1 output (t.p. 28) to PSK demodulator input (t.p 10).
6. Connections on ST2107 trainer:
7. PSK demodulator output (t.p 15) to LPF input (t.p.13)
8. LPF output (t.p24) to comparator input (t.p 46)
9. Comparator output (t.p 47) to bit decoder input (t.p 39)
10. Switch ON the trainer kits and monitor the modulator output (t.p 28) in ST2106 trainer with reference to its input (t.p27) by using a dual trace CRO.
11. To see PSK demodulator process examine the input of PSK demodulator (t.p10) on ST2107 trainer with the demodulator output (t.p 15). Adjust the phase control knob and see its effect on the demodulator's output. Check the various test points provided at the O/P of the functional blocks of the PSK demodulator.
12. The O/P of the demodulator goes to the LPF input. Monitor the filters output (t.p 24) with the reference to its input (t.p 28)
13. The LPF output is rounded and cannot be used for digital processing. In order to square up the waveform, comparators are used. The Bias control is adjusted so that the comparator's output pulse width at t.p47 is same as the NRZ(M) pulse width.

Observations:



PSK WAVEFORM

Result:

The PSK modulator and demodulator circuit has been studied.

Precautions:

1. Check the connections before switching ON the kit.
2. Observations should be taken properly.

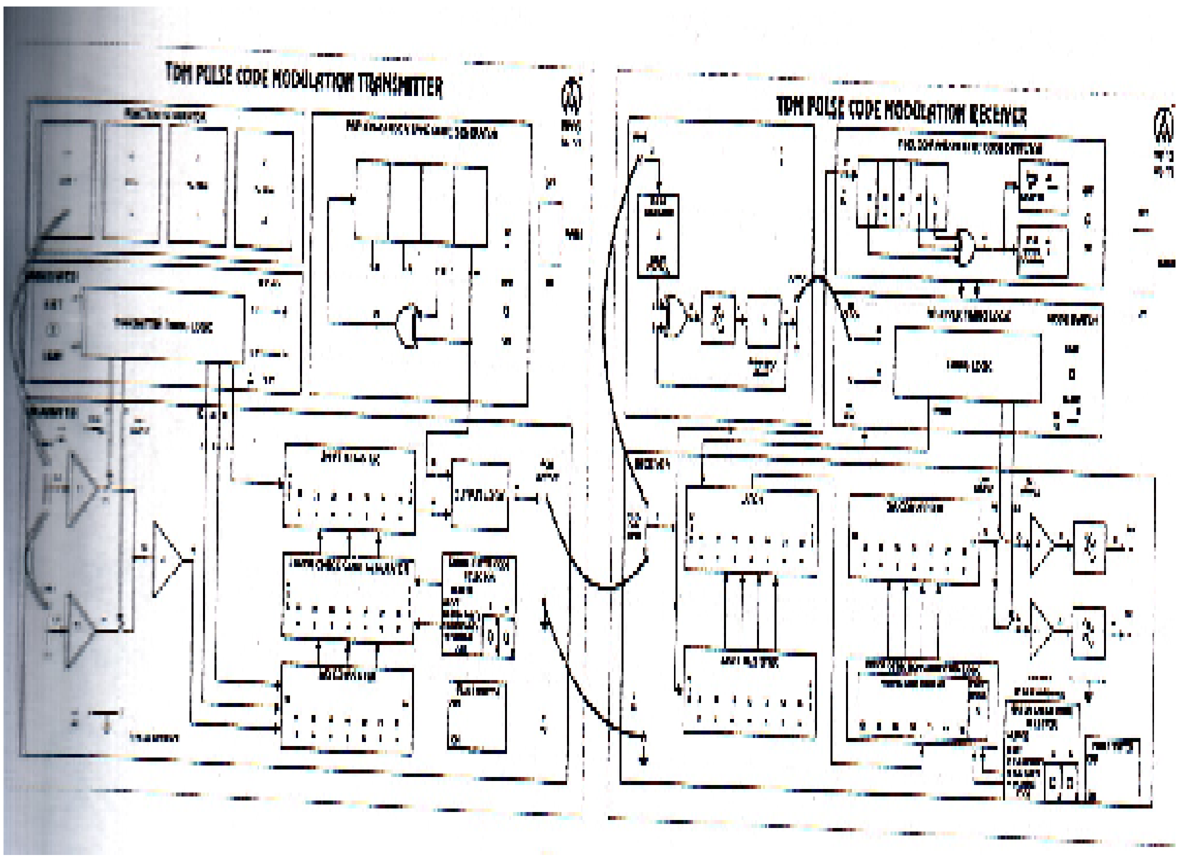
EXPERIMENT No. : 11

Aim: Detection and Correction of Errors using Hamming Code.

Apparatus Required:

1. TDM PCM modulation / demodulation trainer kit.
2. Connecting wires.
3. CRO

Connection Diagram:



Theory: Hamming Code decodes each word at transmitter into a new code by stuffing the word with extra redundant bit. As the name suggests, the redundant bits do not convey information but also provides a method of allowing the receiver to decide when an error has occurred & which bits is in error since the system is binary, the bit in error is easily corrected. Three bit hamming code provides single bit error detection and correction. The kit involves the use of a 7-bit word. Therefore only four bits are used for transmitting data if hamming code is selected. The format becomes: D6 D5 D4 D3 C2 C1 C0

Where, C2, C1 and C0 are hamming code bits.

Procedure:

1. Set up the following initial conditions on TDM-PCM transmitter kit:
    a) Mode switch in FAST position.
    b) DC1 & DC2 amplitude controls in function generator block in fully clockwise position.
    c) Set 1 Khz and 2 Khz signal levels in function generator block to 10 Vpp.
    d) Pseudo-Random syn code generator switched ON.
    e) Error check code selector switches A & B in A=0 & B=0 position.
    f) All switched faults off.
2. Set up the following initial conditions on TDM-PCM receiver kit:
    a) Mode switch in FAST position.
    b) Pseudo-Random syn code generator switched ON.
    c) Error check code selector switches A & B in A=0 & B=0 position.
    d) All switched faults off.
    e) Pulse generator delay adjusts control in fully clockwise position.
3. On TDN-PCM Tx kit connect DC output to CH 0 input (tp 10) and CH 0 (tp 10) input to CH 1 (tp 12) input to ensure that the two channels contain the same information.
4. On TDM-PCM Rx kit connect PCM data input (tp 1) to clock regeneration circuit input (tp 3) and output of clock regeneration circuit (tp 8) to Rx clock input (tp 46).
5. Connect PCM output (tp 44) of Tx kit to PCM data input (tp1) of Rx kit. Also connect the grounds of both the kits.
6. Turn On the power. Ensure that the frequency of the VCO in the receiver clock regeneration circuit has been correctly adjusted.
7. Connect channel 1 of CRO to tp 10 on Tx kit and channel 2 of CRO to tp 33 on Rx kit.
8. Vary DC 1 and note that the data is transferred correctly between the two trainers. This can be verified if the data in the A/D converter blocks of both the kits is same.
9. Select even parity with error check code selector switches A & B at A=0 & B=1 position on both the kits. Set up various codes from A/D converter's output LED's some containing even no. of 1's & some odd. Check the output of error check code generator output on Tx kit, data latch output (tp 16 & tp 22) & D/A converter input (tp 23 & tp 29) on Rx kit.
10. Compare the output of tha data latch LED (tp 16 & tp 22) with input to the D/A converter LED in each case. Once the error detection logic has decided whether an error has occurred, it must pass the received code to D/A converter. But since D0 bit was used as parity bit, it is always forced to a '0'.
11. Set up the error check selector switches to A=1 & B=0 position on both trainers to select the odd parity mode. Repeat steps 9 & 10, but with odd parity as selection.
12. Carry out the same experiment with 1 Khz sine wave applied at CH 0 & CH 1 input of Tx kit. Adjust the 1 Khz amplitude fully clockwise.
13. Switch ON the hamming code error check mode on the kit. Disconnect the sine wave and connect the DC output from the function generator block to CH 0 and CH 1.Adjust the DC control such that the A/D converter's output LED's show 110100 on D6-D0 bits. Note the binary code on the error check code generator..
14. Vary the DC control such that output of A/D converter goes from 1101000 to 1101111. Notice the changes in the binary code output of the error check code. Observe that the error check code generator is only concerned with checking bits D6, D5, D4, D3 only. D2, D1, D0 outputs from A/D converter are ignored by the error check code generator in hamming code as parity check bits C2, C1 and C0 are output in their place depending on the value of D6, D5, D4, D3

bits. It is for this reason that C2, C1 and C0 bits do not change although the data at D2, D1, and D0 have changed.

Observations:

S No.	Data Received	Case 1 D6 D5 D4 C2	Case 2 D6 D5 D3 C1	Case 3 D6 D4 D3 C0	Bit in Error	Corrected Output
1.	0101011					
2.	0110001					
3.	1101101					
4.	1101001					

Result:

Error detection and correction using Hamming Code has been verified.

Precautions:

1. Check the connections before switching ON the kit.
2. Observations should be taken properly.

EXPERIMENT No. : 12

Aim: Measuring the input impedance and Attenuation of a given Transmission Line.

Apparatus Required:

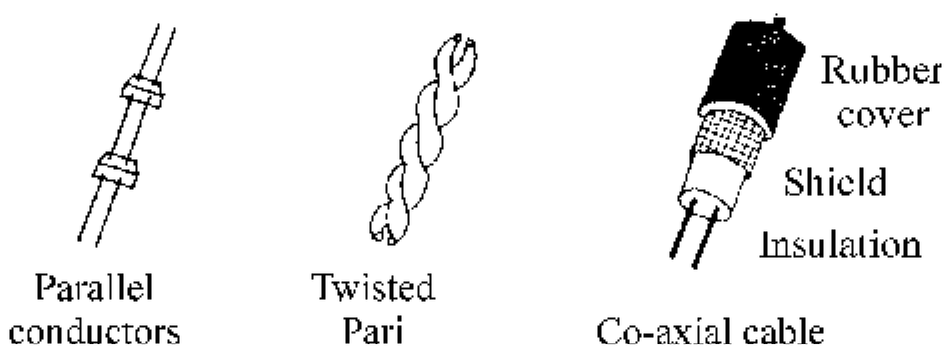
SNo.	Component	Quantity
1.	Transmission line kit	1
2.	Connecting leads	1
3.	Dual trace CRO	1
4.	Digital multimeter	1

Theory: Basic Principles of Transmission Line:

Transmission lines are a means of conveying signals or power from one point to another. From such a broad definition, any system of wires can be considered as forming one or more transmission lines. However, if the properties of these lines must be taken into account, the lines might as well be arranged in some simple, constant pattern. This will make the properties much easier to calculate, and it will also make them constant for any type of transmission line.

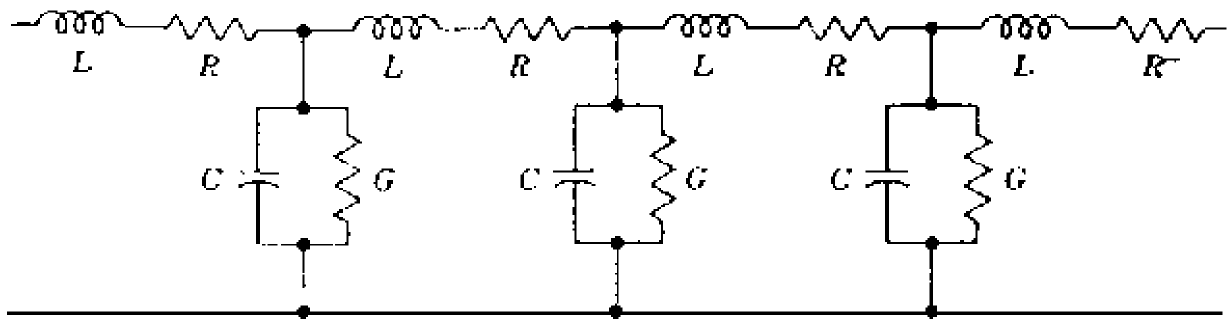
Types of Transmission Line:

One of the simplest forms of a transmission line is the open-wire line or the twisted pair. See figure 1. Coaxial lines are the more popular of the two in RF communication. A coaxial line consists of a central conductor and an outer conductor with the outer conductor referred to as shield normally grounded. Due to the outer conductor normally grounded the two conductors do not have similar relationship with respect to ground and that is why a coaxial line is an unbalanced line. However, due to shielding, coaxial lines have extremely low radiation loss.



Equivalent Circuit Representation of A Transmission Line:

Since each conductor has a certain length and diameter, it must have resistance and inductance, since there are two wires close to each other, there must be capacitance between them. Finally, the wires are separated by a medium called the dielectric, which cannot be perfect in its insulation; the current leakage through it can be represented by a shunt conductance. The resulting equivalent circuit is as shown in figure.



Losses in Transmission Line:

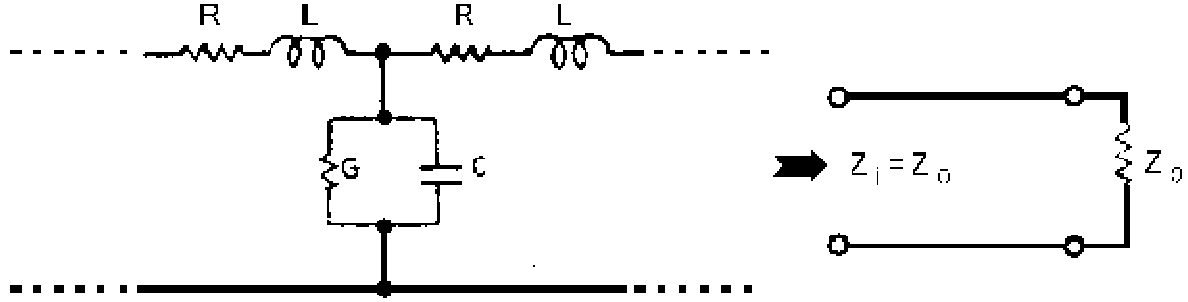
The three major sources of losses in RF transmission lines are :
1. Copper losses
2. Dielectric losses
3. Radiation losses

Characteristic Impedance of a Transmission Line:
Characteristic impedance of a transmission line is its input impedance if it was infinitely long. Refer to the transmission line equivalent circuit of figure. It can be proved with simple mathematics that the characteristic impedance is given by :

$Z = \sqrt{(R + j\omega L) / (G + j\omega C)}$

Where, R = Distributed resistance per unit length
L = Distributed inductance per unit length
G = Distributed Shunt conductance per unit length
C = Distributed Shunt capacitance per unit length.
In a loss less transmission line R = 0, G = 0
Therefore, Characteristic Impedance (Z_0) $= \sqrt{L / C}$.

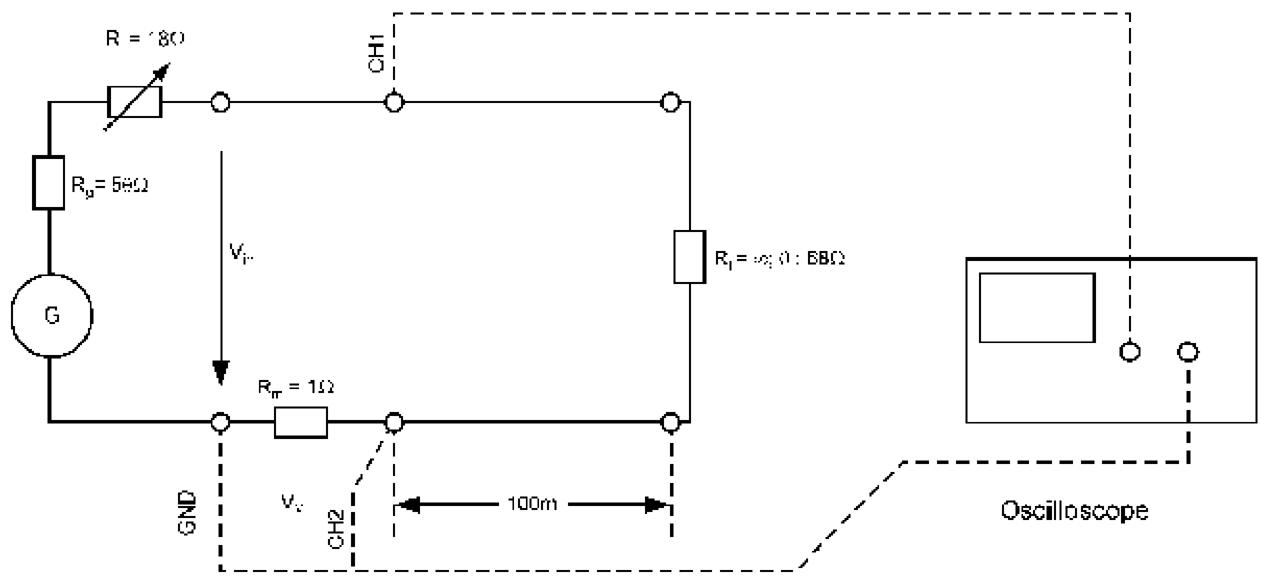
The input impedance of a finite line terminated in its characteristic impedance is equal to its characteristic impedance only. See figure.



Basic Properties of the Coaxial Cable (Used in the Trainer)
Type : RG 174
Length : 100 meters
Series Inductance : 28µH (Frequency 1 KHz, 100 m) approximately
Parallel capacitance : 11µF (Frequency 1 KHz, 100 m) approximately
Conductance : 0.4 µmhos
Impedance : 50 Ω approximately
.
Important points to note :

1. The coaxial line used in the trainer is placed in the coiled form of 25 meters each. This is done for space saving. In practice the lines are s straight. The coil form has caused some deterioration. For the convenience of the students the basic properties given above are of coiled form.

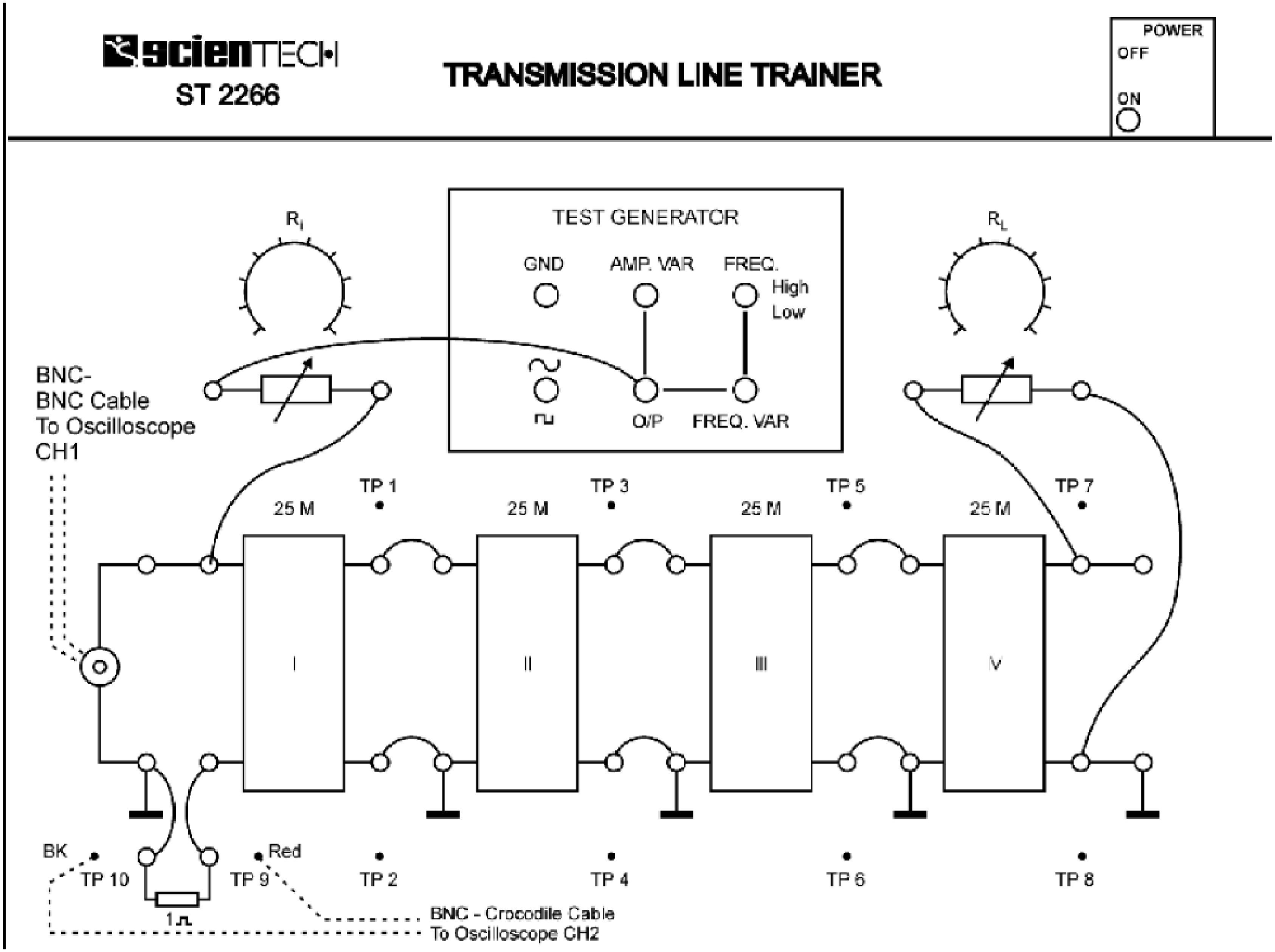
2. The attenuation of RG 174 cable is 40dB / 100 m at 200 MHz. But due to coiled form it will show 3dB/ 100 m at 3.6 MHz.

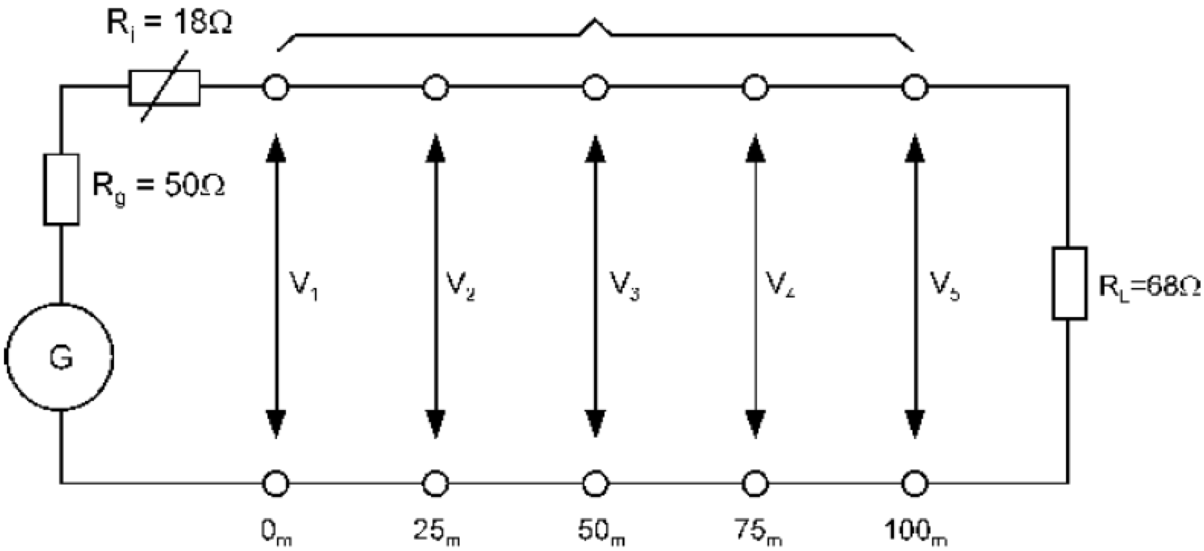


Measuring the attenuation of line:
The ohmic resistance R & the conductance G are responsible for energy dissipation in the form of heat. These losses, which determine the attenuation characteristics, are expressed in terms of "attenuation" "a" and can be calculated by :

$$a = 20 \log (V2 / V1)$$

Where, V1 = amplitude of signal at input
V2 = amplitude of signal at output
a = attenuation for given length
In this experiment we will measure the attenuation for the different trunks of transmission line available on the trainer. See figure


scienTECH
ST 2266
TRANSMISSION LINE TRAINER
POWER
OFF
ON
TEST GENERATOR
GND
AMP. VAR
FREQ.
High
Low
O/P
FREQ. VAR
R_i
R_L
BNC-
BNC Cable
To Oscilloscope
CH1
TP 1
TP 2
TP 3
TP 4
TP 5
TP 6
TP 7
TP 8
TP 9
TP 10
25 M
25 M
25 M
25 M
I
II
III
V
BK
Red
1 Ω
BNC - Crocodile Cable
To Oscilloscope CH2



Procedure for input impedance measurement :

1. Adjust Ri and RL for 18Ωand 68Ωrespectivelywith the help of DMM.
2. Make the connections as shown in figure.
3. A 1Ωresistance in series between the generator and the transmission line as shown, in figure 12 allows to measure the value of input current.
4. Set the input at 0.4p-p and freq 100 KHz of sine-wave (both measurement on (CRO).
5. Take readings of Vin and Vm (across 1Ω) on oscilloscope.
6. Calculate the input impedance according to the following formula :

$$Zin = Vin / I = Vin / Vm / 1\Omega$$

7. Change the frequency to 1MHz and note the values of Vin and Vm at this frequency.
8. Note down these results. The input impedance at 100 KHz is around 80 Ωand at 1 MHz is around 50Ω.

Procedure for attenuation measurement :
1. Adjust Ri and RL for 18 Ωand 68 Ωrespectively with the help of DMM.
2. Make connections as shown in figure 9.
3. Set the sine-wave frequency to approximately 100 KHz and level to 0.4 V.
4. Oscilloscope CH 1 shows applied input CH 2 shows outputs.
5. Measure signal level at Input, and at 25, 50, 75, and 100 m lengths.
6. Tabulate as under :

Length(m)	V_1(input)	V_2(output)
25		
50		
75		
100		

7. Now, calculate the attenuations in dB at various lengths by the formula given below : a = 20 Log V2 / V1
8. The attenuation is approximately -2 dB at 100 m.
9. Try the same with open-ended line and short-ended line.

Result: The 7 input impedance for 100KHz signal is approx 80 ohm and the attenuation measured for transmission line for various length.

Precaution:

1. Do not make interconnections on the board with power switched ON.

2. The frequency from generator should be set to high for attenuation measurement.

Pre Experiment Question:

1. What is attenuation?

2. What is the effect of frequency on attenuation?

Post Experiment Question

1. What is the input impedance of the transmission line?

2. Calculate the percentage attenuation of transmission line?