PRACTICAL WORK BOOK
For The Course
TC-497 Multimedia Communication

For

Final Year

(Telecommunication Engineering)

Name of Student: ____

Class: ________________________________Batch :____________________

Discipline: ____

Class Roll No.: ________________Examination Seat No.________________

Complied by: Engr. Shoaib Siddiqui (Assistant Professor)

TELECOMMUNICATION LAB

DEPARTMENT OF ELECTRICAL ENGINEERING
NED University of Engineering & Technology, Karachi-75270, Pakistan

INDEX

MULTIMEDIA COMMUNICATION (TC)

S. NO.	DATE	OBJECT	SIGNATURE
1		To observe and analyze the operation of ISDN. (Integrated Services Digital Network)	
2		To examine the operation of the system PCM/EV.	
3		To follow the phases for the connection and disconnection between two telephones	
4		Selection of the PCM channels. To select and observe the PCM transmission (Time slots) and reception channels, which compose the voice signal circuits.	
5		To understand and observe the linear delta modulation.	
6		To understand and observe the adaptive delta modulation.	
7		To understand and observe the linear delta demodulator.	
8		To understand and observe the adaptive delta (CVSD) demodulator.	
9		Providing a functional description of the types of frequency synthesis carried out in the module T10L/EV.	
10		To analyze frequency synthesis with conversion using the module T10L/EV.	

EXPERIMENT NO: 1

Object: To observe and analyze the operation of ISDN (Integrated Services Digital Network)

Theory: ISDN is used to integrate digitized voice, video, and data, using the same simple twisted pair wire as that used for telephone loops. It is integrated because all types of information- voice, data, packet switched messages and directly connected users- come to the user node on one line from one central source.

Two classes of service called the basic rate and the primary rate are available with ISDN. The basic rate is known as 2B+D and consists of two 64 –kbit/s B channels that can carry either voice or data ,plus one 16-kbit/s D channel used for network signaling and control as well as user packet data. There is an additional 48-kbit/s channel used for other purposes, such as echoing the D channel back to the source, synchronizing the terminal units and conveying other internal network information. The overall basic rate is (64+64+16+48) =192 kbits/s.

In North America and Japan, the primary rate is called 23B+D and consists of 23B channels at 64 kbits/s each and one D channel at 16 kbits/s.The primary rate is compatible with 1.544-Mbit/s transmission system since it is the same as the Bell system T-1 trunk rate and is used to interconnect the computer centers and to connect central offices to the network rather than to serve local users directly.

Equipment and accessories required:

1 ISDN panel
1 PBX System
3 ISDN Telephone
1 ISDN Tester.

Accessories:

1 Digital storage oscilloscope
3 BNC cable
1 PC

Procedure:

Refer to the diagram as shown in figure 1 and configure the system. Check the connections of diagram with that of ISDN panel.

When there is voltage on the bus, the red LED on the test adopter will come on. Use the ISDN tester to measure the resistance on a correctly wired S0 bus.

Observations:

Automatic line test:

A customer calls to report a fault. The customer is aware that they possess an ISDN line but is unable to give any technical details about it. For this reason you conduct an initial test using the ISDN tester to check:

- Whether the B channels are present.
- The interface on which the test is being performed.
- The line configuration
- And the protocol the connection is using.

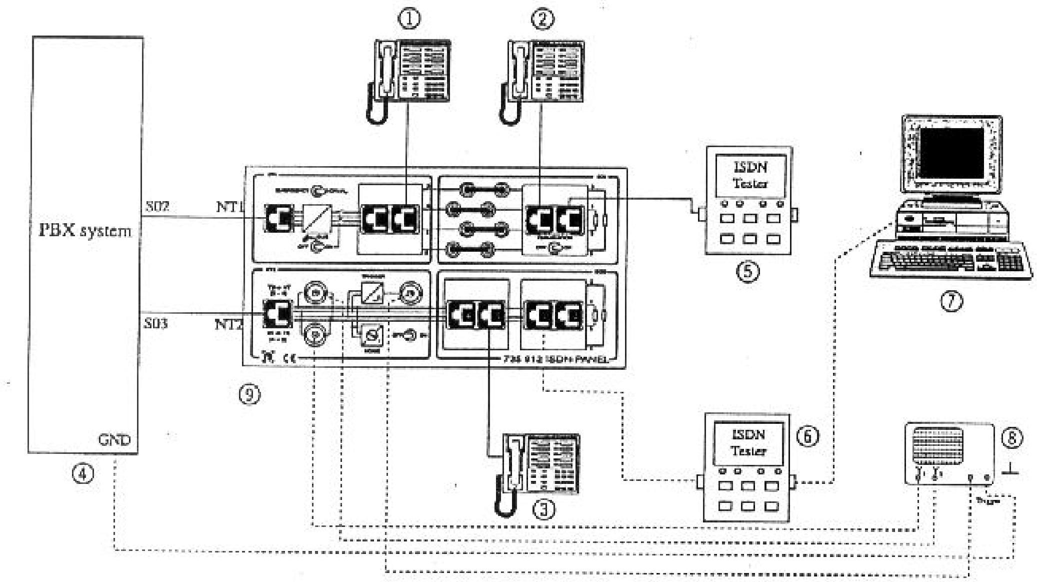
Observations:

1. How can you conduct a simple automatic line test using the ISDN tester?
2. What results are shown on the display after completion of the test?
3. How do you interpret these results?
4. How do the test results appear when a B channel is used during the course of the test? How can a B channel be occupied using the existing terminal devices for the duration of the line test?
5. How do the test results appear when two B channels on the S0 bus are used? How can you occupy both B channels on the internal S0 bus, even though only three ISDN telephones in total are available to you? Can you achieve the exercise with only two telephones?
6. How does the testing device react when it is not connected to the S0 interface during the line test?

1 Installation techniques

1.1 Configuration of the training system

1.1.1 Graphic representation of the basic configuration



①②③	ISDN Telephone	(735 915)	⑦	PC	
④	PBX system	(735 913)	⑧	Oscilloscope	(575 293)
⑤	ISDN tester	(735 916)	⑨	ISDN panel	(735 912)
⑥	Option				

The experiment set-up shown here represents a configuration with the maximum amount of equipment. In the following experiments, the number of terminal devices may be less than shown for example. All the measuring equipment may also be unnecessary for some experiments.

Figure. 1.

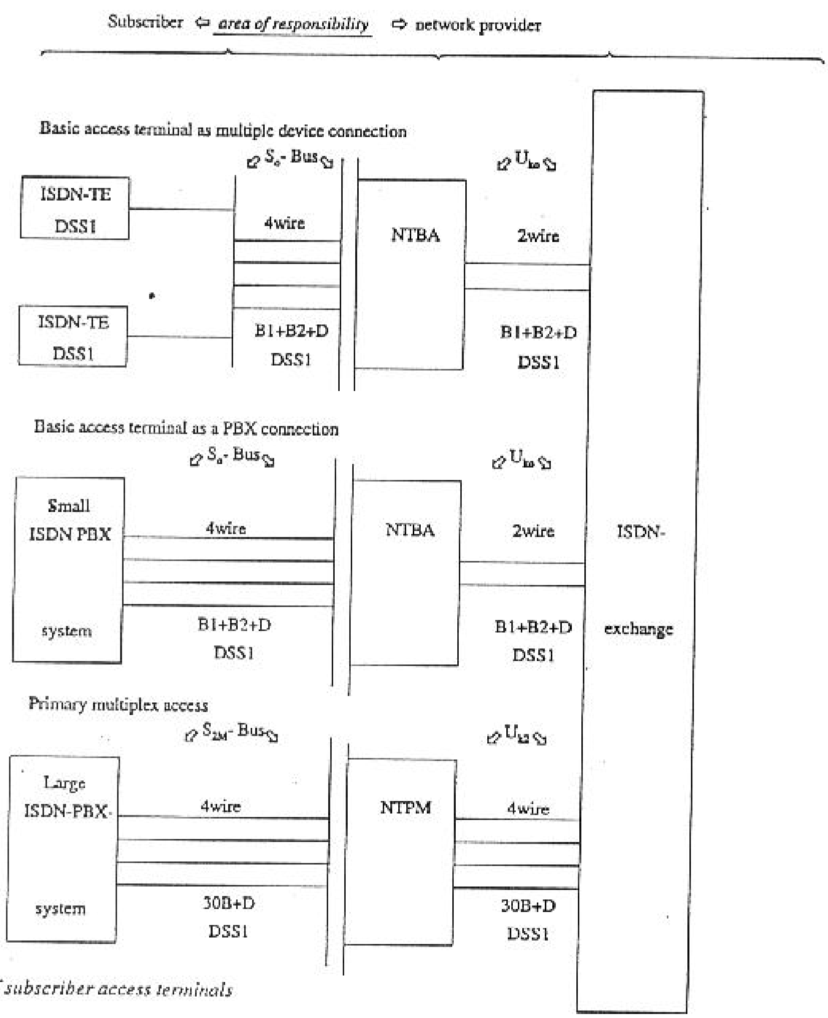


Fig. 1.4-1: Types of subscriber access terminals

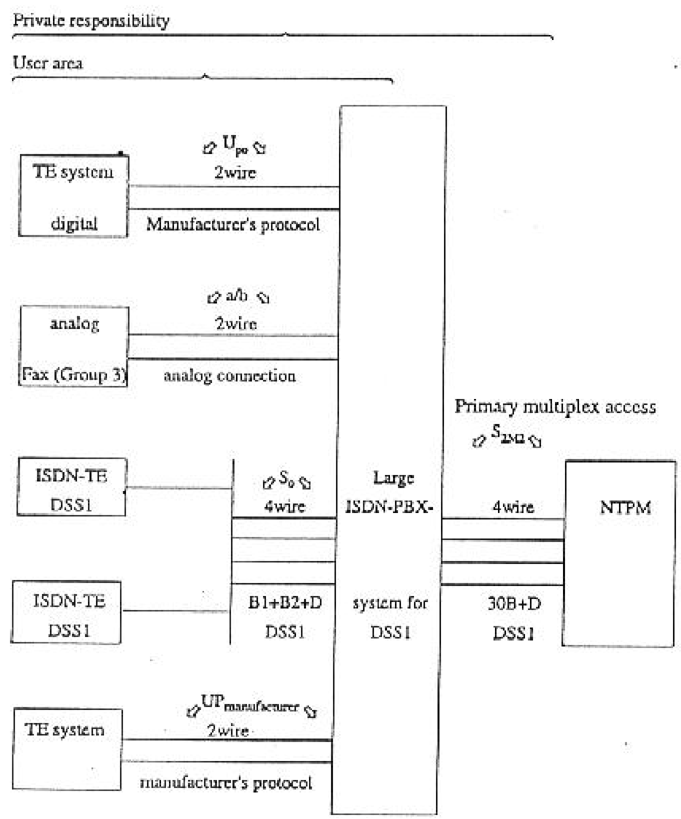


Fig. 1.5-1: Internal interfaces of a private digital exchange

Automatic testing of the services:

You are installing an ISDN line for a customer. After installing the NTBA and laying the S0 bus, it is required to prove the functionality of the line. In doing this also check whether various terminal devices (e.g. telephones, group-3 faxes, group-4 faxes etc) can be operated on the bus.i.e whether they can receive and transmit the call. Since you are not in possession of all these types of device where you are, use the ISDN tester to carry out this check.

Observations:

1. What sort of test can you use to check the various services most easily?
2. Which test menu do you call up on the ISDN tester for this?
3. How do you carry out this test with the ISDN tester?
4. Carry out the services test by means of self calling. What are the objectives of the test when calling the test's own number? Interpret the results of the test.
5.Conduct a test of services by calling a external number(establishing connection to the S02 bus).What are the objectives of the test when calling a different number to that of the test device? Interpret the results of the test.
6. Why do the LEDs L1, L2 and L3 turn on or flash during the test?

Testing of an individual service:

Your customer has bought an ISDN PC card and is planning to use it as a group -3 fax receiver. The card has already been installed by customer. As it turns out, faxes can neither be transmitted nor received, it is required to solve this problem. To find out if the terminal device service for fax group three is functioning at all on this line, make use of the ISDN tester to determine whether connections using this service can be established.

Note: As an alternative to a PC card the customer could be connecting a group 3 fax machine via a terminal adapter (TA a/b) or a small PBX system (point to multipoint configuration).

Observations:

1. What sort of test can you use to check a specific service most easily?
2. Which test menu on the do you call up on the ISDN tester for this?
3. How do you carry out this test with ISDN tester? Describe the testing process step by step?
4. What information are you given by the device's display during establishment of the connection (=dialing)?
5. What is the result of the test when a connection is successfully established? How do you interpret this result?
6. What is the result of the test when there is no terminal device to receive the call at the dialed number? How do you interpret this result?

Assignment:

Students are required to perform the practical as described above and write the observations.

EXPERIMENT NO:2

Object: The object of this experiment is to examine the operation of the system PCM/EV.

Required material: Unit PCM/EV
1 pulse telephone set
1 multifrequeny telephone set.

Procedure for use:

- Connect the power supply cable to the socket on the back of the container.
- Connect the telephones to the proper line plugs of the panel and check that the telephones of the lines 1, 2, 3 are pulse ones and that the telephone # 4 is multifrequency.
- Turn on the system with the switch on the panel. Check the status of leads. In normal conditions it should be the following:

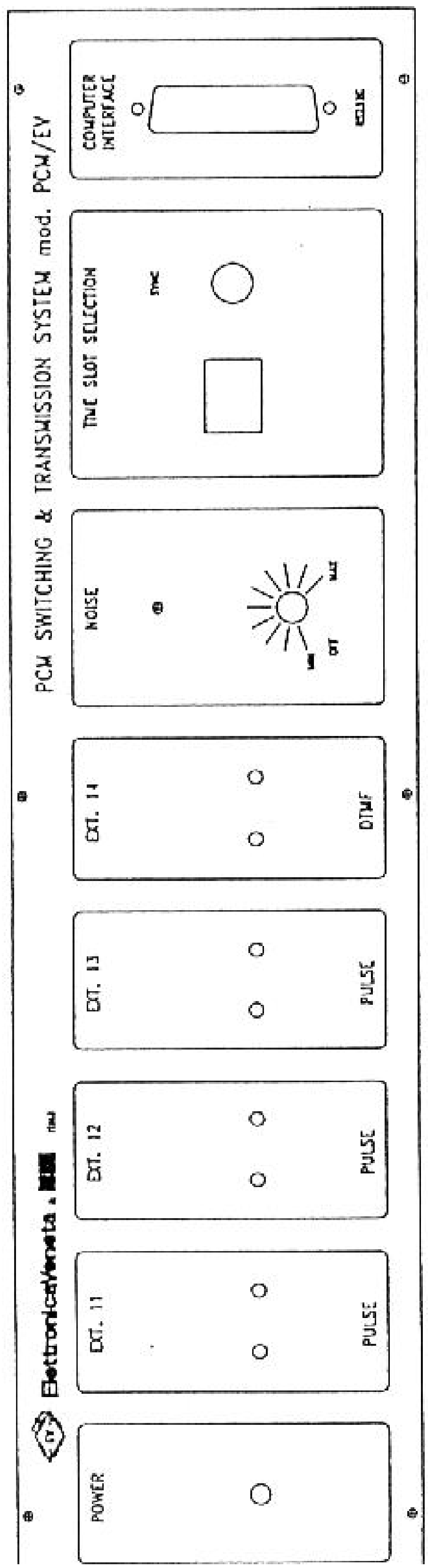
  Normally on

  - Power supply led on the front panel.
  - Voice led on the section RX MPX.
  - 4 sleds on the decoder DTMF
  - Voice led on the section RX MPX
  - Led 1 (switch mode) on section µP
  - Led TRAP INT on section µP

  Normally off

  All the other legs.

  The BER Led can be on or off in a fixed way, not flashing. In case of flashing of the BER Led or lighting on of the Frame Sync and Multiframe Sync leads, check that the noise generator is off.


PCM SWITCHING & TRANSMISSION SYSTEM mod. PCM/EY
COMPUTER INTERFACE
TIME SLOT SELECTION
SYNC
NOISE
OFF
MAX
EXT. 14
DTMF
EXT. 13
PULSE
EXT. 12
PULSE
EXT. 11
PULSE
ElettronicaVeneta
POWER

Lift the telephone # 1 checking the starting on of the engage LED (SW HOOK DETECTOR) on the silk screen panel, wait for the invitation tone to dialing (“Dial”)

Compose the dial number of one of the other telephones noting the tone disappears at the dialing of the first digit. The dial numbers of the telephones for inner connections (which do not concern the external line) are the following:

11 for dialing telephone # 1
12 for dialing telephone # 2
13 for dialing telephone # 3
14 for dialing telephone # 4

For the external connection (passing from the external line) the dial numbers are:

21 for dialing telephone # 1
22 for dialing telephone # 2
23 for dialing telephone # 3
24 for dialing telephone # 4

Observe the engage LED pulsing during the decade dialing; wait for the call signal to the dialed telephone noting the presence of the free tone on telephone # 1 (calling)

Lift the called telephone; check the voice signal connection between the two terminals.

Hang down the telephones; repeat the same operations with other telephones calling, in particular, from # 4. In this case, you can feel the presence in line of the dialing tones (DTMF) and on the LEDs of the silk screen panel connected to the outputs of the DTMF decoder, note the appearance of the binary code corresponding to the dialed number.

Observations: Students are required to perform the practical as described above and write the observations.

EXPERIMENT NO:3

Objective

The purpose of this experiment is to follow the phases for the connection and disconnection between two telephones. Table reports these phases and their signaling.

PHASE	TYPE OF SIGNAL	DIRECTION of the SIGNAL	
		CALLING USER	CALLED USER
User engage (unhook)	Line off on the user terminal	User ->Central	
Invite to dialing	“DIAL” Tone (invite to dialing)	User ->Central	
Dialing	Decade or “DTMF”	User ->Central	
Busy	Busy tone	User <-Central	
Free	Free tone	User <-Central	
Call	Call current		User <-Central
Response engage (unhook)	Line off on the user terminal		User ->Central
SETTING UP the CONNECTION			
Disengage (hang up)	Opening the user loop	User ->Central	
CONNECTION and DISCONNECTION			

In particular, the purpose of the experiment is to:

* analyze the activity promoted by the central exchange after the engagement of a line:

- The closing of a user's loop determines the transition to the active state of the engage signal (SWITCH HOOK DETECTOR) of the user's interface
- The central exchange answers with the invitation tone to dialing

* detect the sequence of pulses for opening the user's line due to the decade dialing:

- At each opening pulse corresponds a still transition of the engage signal supplied by the user's interface
- The control unit counts these transitions to identify the transmitted digits, considers the duration of the interdigit pauses (in off state) and acquires the dial number with which he can set up the required connection

*observe the behavior of the multi-frequency signaling with which the dialing digits are coded with pairs of acoustic tones which are decoded by the user's interface:

- The decoding circuit supplies the value of the transmitted digits as 4-bit binary codes, plus a line of "valid datum"; the binary codes are displayed by LED (ON= state "1")
- check the connection and disconnection.

Required material

- Unit PCM/EV with telephone equipment
- Oscilloscope
- Multimeter

Operation

- Turn on the system keeping the microtelephones down
- Connect the multimeter (50 Vfs) between TP1 (Test Point 1) and TP2: the voltage line #1 presents a value of about 40 Vdc

- hang up the telephone #1 and check the changing of the line voltage, which will go to 10 V as a consequence of the closing of the user's loop on the d.c. impedance of the circuit

- connect the oscilloscope (in a.c.) between TP1 and ground: adjusting the scales in a proper way, you point out the invitation tone to dialing; the presence of this tone is also signaled with LED
- the same tone can be observed between TP11 and ground, as voice signal input to the user's interface (RECEIVE INPUT)
- with the input of the oscilloscope in d.c. (1V/Div), connect the probe to TP8, which corresponds to the engage signal (SW HOOK DETECTOR) of the user's inter face: the active logic state is "0", which is signaled with LED

- select one digit on telephone #1 and note that the invitation tone disappears. On the oscilloscope you will note the state transitions due to the decade dialing (pulses for line opening). The dial pulses are also signaled by the LED

- select other digits adjusting the time scale of the oscilloscope so to evaluate the pulse and the pause duration.

- According to the dialed digits, the telephone can receive the busy or the free tone, with the call signal of one of the remaining telephones

- Unhook the microtelephone and observe the return of the engage line (SW HOOK DETECTOR, TP8) to still state (high level)

- Connect the oscilloscope (a.c., 0.5 V/Div, probe 10:1) to TP7 and unhook the telephone #4 (multi-frequency) observing the turning on of the led connected to the engage line (SW HOOK DETECTOR) of the user's interface

- Select a digit after the invitation tone: note the tone disappearance and the presence of a signal on the oscilloscope (the duration of this signal is short)

- Select other digits and adjust the time scale and the trigger on the oscilloscope so to optimize the observation of the DTMF tones which come to the decoding circuit. The presence of these tones can also be heard at the receiver

The signal present across TP7 is the sum of the two tones, high and low, which code the dialed digit.

- on the LEDs connected to the decoder lines, observe the correspondence between the dialed digit and the codes supplied by the circuit:

Q1	Q2	Q3	Q4	DIGIT:
ON	OFF	OFF	OFF	1
OFF	ON	OFF	OFF	2
ON	ON	OFF	OFF	3
OFF	OFF	ON	OFF	4
ON	OFF	ON	OFF	5
OFF	ON	ON	OFF	6
ON	ON	ON	OFF	7
OFF	OFF	OFF	ON	8
ON	OFF	OFF	ON	9
OFF	ON	OFF	ON	0

According to the dialed digits the telephone can receive the busy tone or the free tone, with the call signal in one of the remaining telephones.

- Unhook the telephone observing the turn off the engage LED.
- connect one probe of the oscilloscope (2V/Div) to TP9 (RING COMMAND) and the other probe (50V/Div) to TP2 (RING)
- from a telephone different from #1 dial the number of the line #1 (11 or 21) after the invitation tone. When the call signal reaches telephone #1, TP9 shows the command of the call current (active low) supplied by the control unit, and TP2 the signal of the call current present in the line
- note the correspondence between the call line state, the activation of the relay with the starting up of the call LED, the presence of the line current and the free tone sent to the calling telephone: all these signaling have the same timing (a second of activity every any five seconds)
- when the telephone #1 is unhooked, the engage LED turns on and instantaneously the call current is interrupted, and this is caused by the user's interface. Right after the control unit removes the call command and the free tone on the calling telephone (these events are visible only if the unhook occurs during the active phase)

At this point there is the connection between the two telephones.

- check the voice signal connection with the caller. Hang up and then unhook the telephone #1 noting the voice signal remains. Hang up the calling telephone noting the connection is freed. Hang up the telephones
- connect the oscilloscope (alternated; 1ms/Div; 0.5V/Div; probe 10:1) to TP11 (RECEIVE INPUT). Unhook the telephone #1 and observe the turning on of the LED which indicates the presence of the invitation tone to dial. Examine the characteristics of the tone properly adjusting the scales of the oscilloscope
- unhook one of the remaining telephones and dial its number (12, 13, 14 or 22, 23, 24). Note the LED lights on which indicates the presence of the busy tone on telephone #1. Examine the characteristics of the tone by properly adjusting the scales of the oscilloscope
- hang up both telephones and then redial from telephone #1. Wait for the call signal from the dialed telephone and the lighting on of the LED in presence of the free tone on telephone #1. Examine the characteristics of the tone by properly adjusting the scales of the oscilloscope
- unhook the called telephone. Check the presence of the voice signal connection. Hang up the telephones.

Observations: Students are required to perform the practical as describe above and write the observations.

EXPERIMENT NO:4

Object: Selection of the PCM channels. The purpose of this experiment is to select and observe the PCM transmission (Time slots) and reception channels, which compose the voice signal circuits. On the PCM transmission (DX0) line 4 channels on 32 are busy:

- channel 9 for telephone # 1
- channel 11 for telephone # 2
- channel 13 for telephone # 3
- channel 15 for telephone # 4
- the remaining channels are all at level “1”.

On the PCM reception (DR0) line all channels have ,in still state, the PCM value “0” (succession of “1” and “0” ,according to ADI coding: Alternate Digit Inversion).In presence of one or more voice signal circuits, the reception data appear in the following time slots:

- channel 1 for telephone # 1 (default)
- channel 2 for telephone # 2 (default)
- channel 3 for telephone # 3 (default)
- channel 4 for telephone # 4 (default)

Required material:

Unit PCM/EV with telephone sets.
Oscilloscope.

Operation:

- Adjust the oscilloscope to observe the PCM signaling at 2048 Kb/s. Synchronize the external time base with the signal “SYNC” taken from the front panel with a coaxial cable.
- Connect the probes 10:1 to TP16(DX0, frame PCM in transmission) and TP19 (DR0,PCM frame is reception)
- With the decade selector of the front panel (TIME SLOT SELECTION) detect the transmission channels of the line DX0,which can be distinguished from the other channels (not

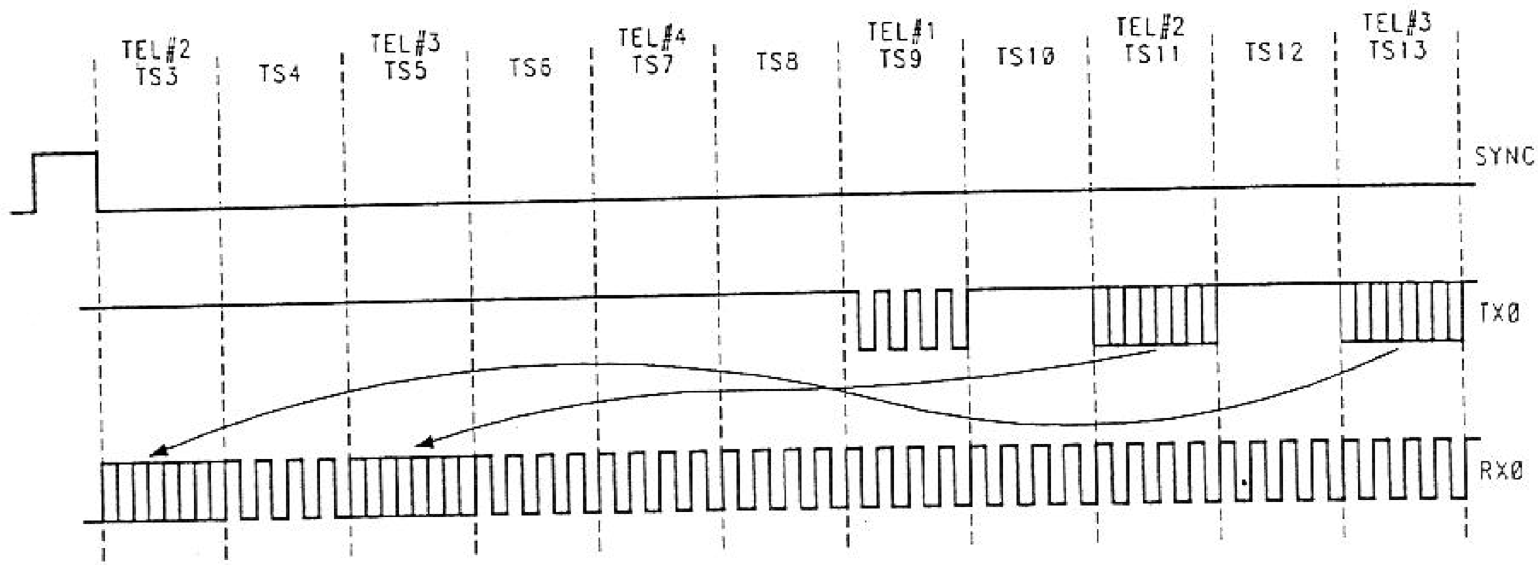

TEL#2 TS3
TS4
TEL#3 TS5
TS6
TEL#4 TS7
TS8
TEL#1 TS9
TS10
TEL#2 TS11
TS12
TEL#3 TS13
SYNC
TX0
RX0

figure 3.1

used) for the presence of PCM codes near zero(succession of one and zero)

- The reception channels (line DR0) cannot be distinguished between them , as in still state they all have the PCM value "0"
- Connect two telephones for example # 2 and # 4 and synchronize the oscilloscope first to the Time Slot 10 (select 10 on the front panel) and then on the time slot 2 (select 2).observe that:
- In presence of voice signals the transmission channels(11 and 15) take variable codes
- the voice signal data appearing on the reception channels (3 and 7) are variable codes in respect to the fixed configurations of the PCM zeroes which are present on the other channels.
- Connecting two telephones which are near, for example #2 and #3, the respective reception and transmission "Time slots" can be set near quite enough to observe them simultaneously on the oscilloscope as shown in figure 3.1
- In this case ,talking in one of the two telephones(e.g. # 2) you will see that the variations of the related transmission channels(Time slot # 11) will affect the reception channel of telephone # 3(Time Slot # 5),and vice versa.
- Repeat the test with different connections, also simultaneous.

Observations: Students are required to perform the practical as described above and write the observations.

EXPERIMENT NO:5

Object: To understand and observe the linear delta modulation.

Theory: Delta modulation is technique used to convert an anolog signal into bits.

This process is carried out through two typical operations: the sampling and the coding of the signal to be transmitted. So, Delta Modulation may be considered as a PCM system, even though the term PCM applies to well-defined coding technique.

The advantage of Delta Modulation is that the modulator and the demodulator circuits are much simpler than those used in traditional PCM's. Some limitations of Delta (Also called one Linear) Modulation are overcome in the Delta-Sigma and Adaptive Delta Systems.

Description of the Circuit:

Audio Signal and clock generator:

Refer to the functional diagram in fig 2.1 and the electric diagram shown in fig 2.2

Notice that the modulator and demodulator are included (IC1 and IC2 respectively), as well as a clock generator, a frequency divider by 32 (IC3) and two low pass filters (IC4 and IC5).

The clock pulse necessary for modulator and demodulator is generated by IC3 (CD4060), C8, R19 and RV1.An almost square wave with a frequency ranging from 16 kHz to approx 32 kHz can be measured at the set point TP1.Inside IC3 this signal is then divided by 32: a square wave with a frequency totaling 1/32 of the clock frequency is obtained in pin 13.

This square wave is filtered by the low-pass filter made up by IC4 (TL082).A sine wave is available at the filter output(Audio OUT).This can be used as an output signal for the modulator IC1.

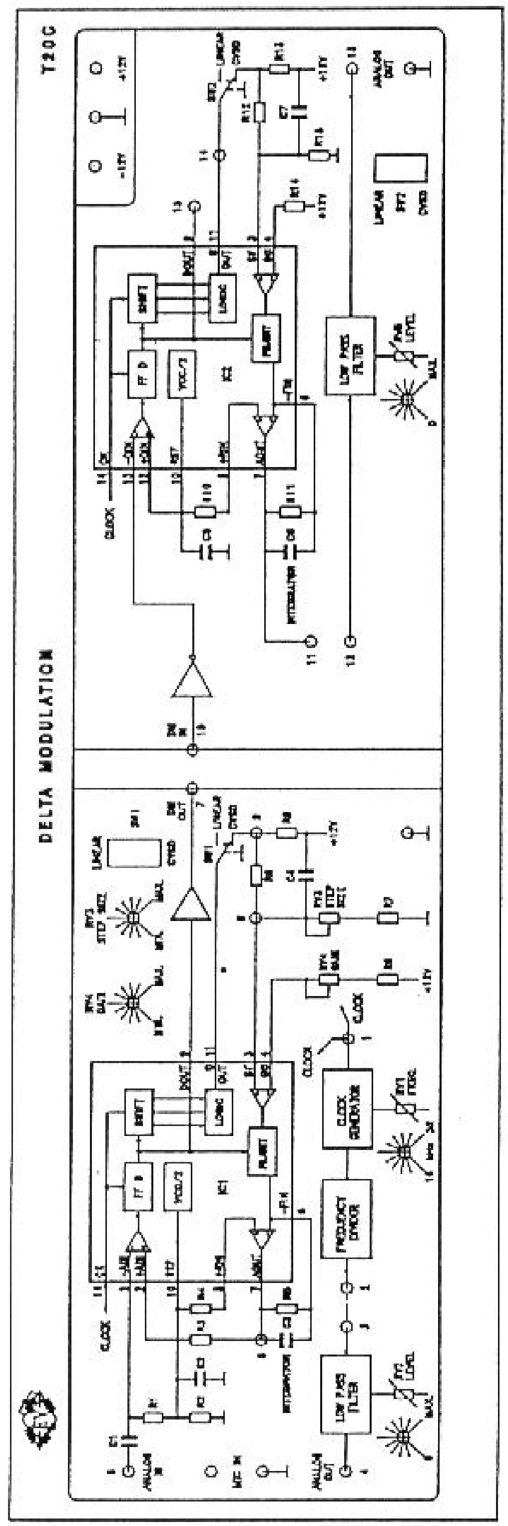


fig. 2.1

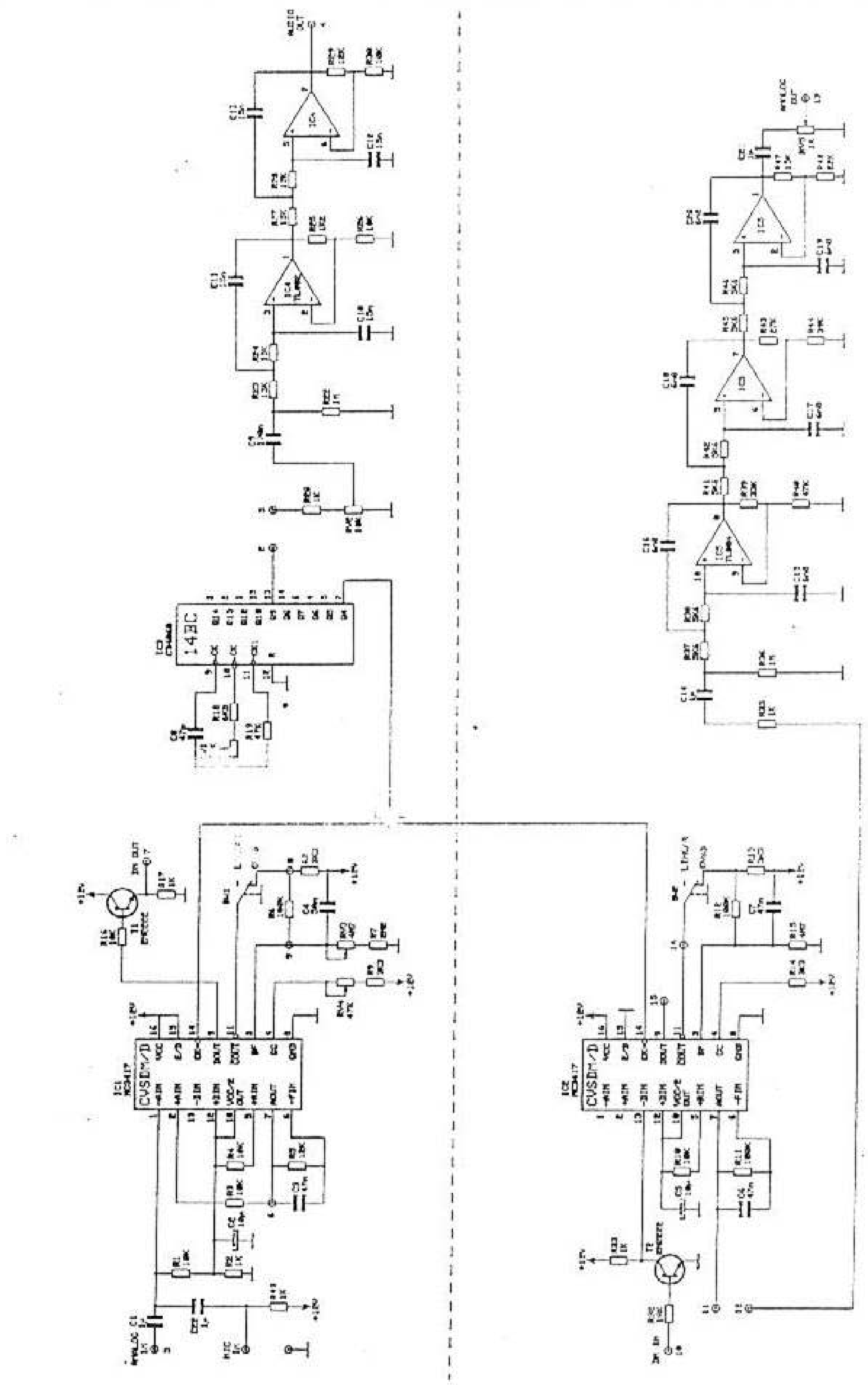


fig.2.2

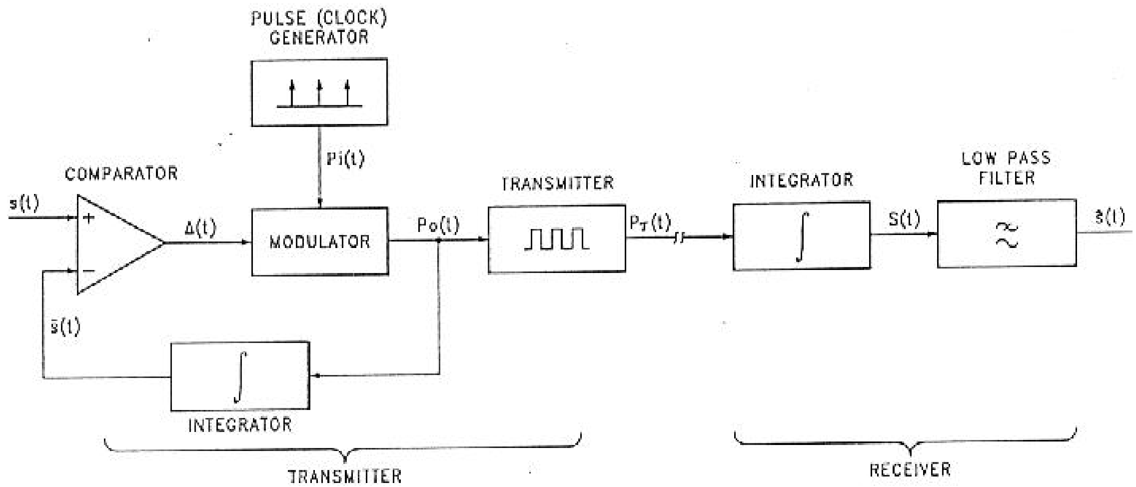


Fig.1.1 Communication system using Linear Delta Modulation

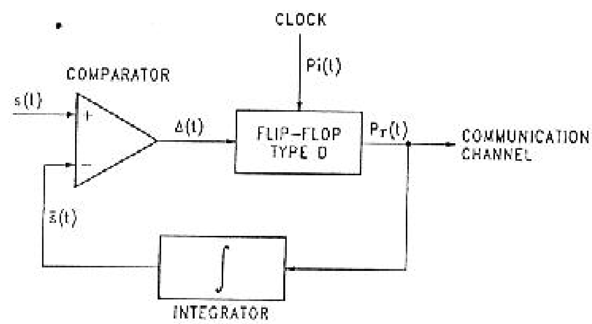


fig.1.3 Basic Diagram of Delta Modulator

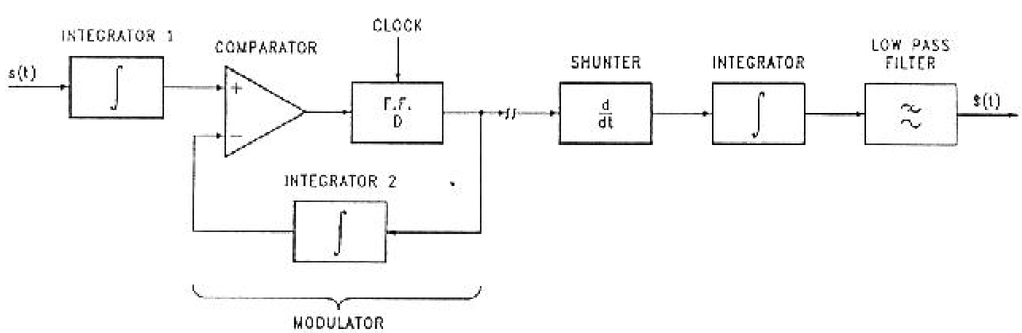


fig. 1.5 Block Diagram of a DELTA-SIGMA Communication System

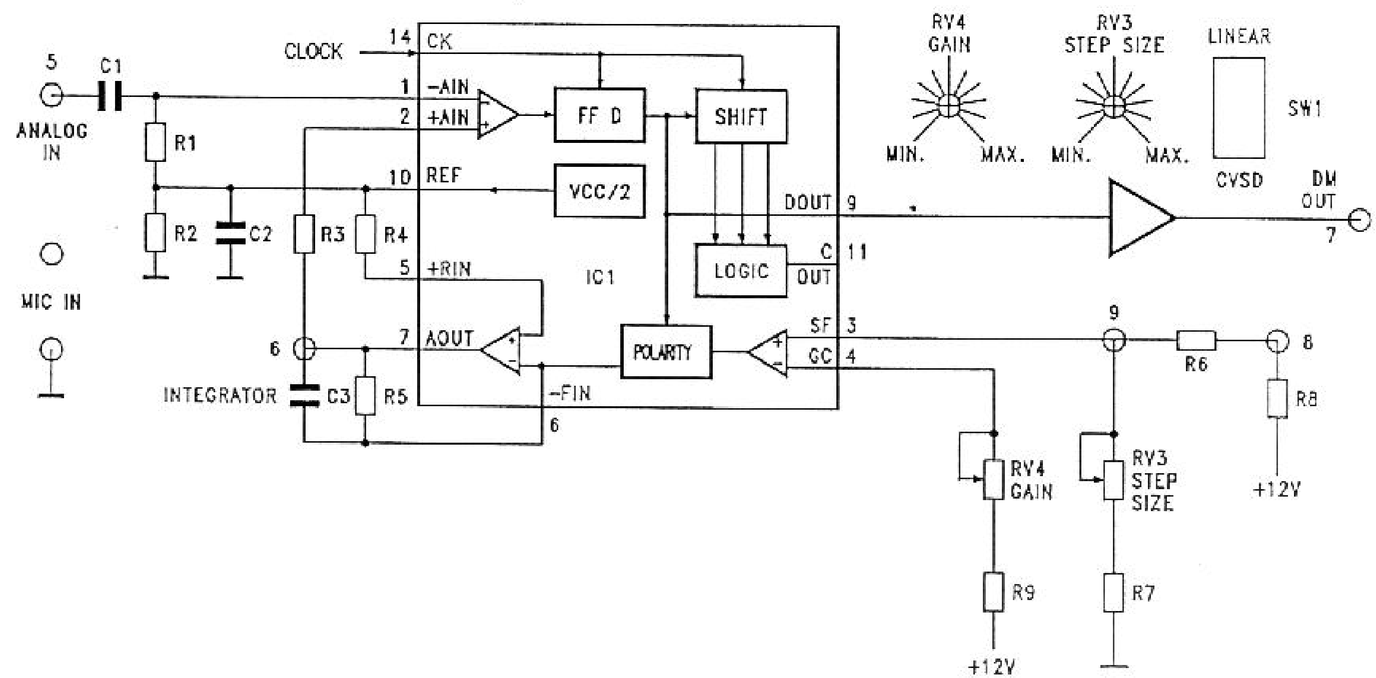


fig.2.3 Linear Delta and CVSD Modulator

Required Instruments:

- Dual trace oscilloscope
- Frequency meter
- Low frequency generator
- Power supply 12V dc.
- Delta modulator circuit.

Procedure:

- Connect TP4(ANALOG OUT) to TP5(ANALOG IN)
- Turn the two way switch SW1 to "LINEAR"
- Turn RV3 (STEP SIZE) completely to the right and RV4 (GAIN) almost to the maximum level (last but one notch).
- Turn Rv1 to 32KHz
- Power supply the circuit with 12V dc
- Connect the oscilloscope to TP5:adjust RV2 in order to obtain a signal of approx. 150mVpp
- Connect the other probe of the oscilloscope to TP6(integrator output of the Delta Modulator)
- Fig 3.2 shows the generated waveforms: if the oscilloscope synchronization is difficult, adjust RV2 slightly until the two displayed waveforms are synchronized.
- The signal approximating the input analog signal is available at the integrator output (TP6): this signal is obtained by integrating the rectangular waveform resulting from delta modulation.
- Connect the oscilloscope to TP6 and TP7 (DM OUT): Fig 3.3 shows the rectangular waveforms generated at the delta modulator output (TP7) and the waveform obtained through its integration.
- With an oscilloscope displaying three traces, it is possible to simultaneously see the input analog signal of the modulator, the digital output of the modulator and the signal obtained by the integration from the modulator digital output Fig 3.4 shows these three signals: the input analog signal and the saw-tooth signal(approximating the analog signal) are superimposed. Notice that, when the saw-tooth signal is higher than the analog signal, the

fig.3.2

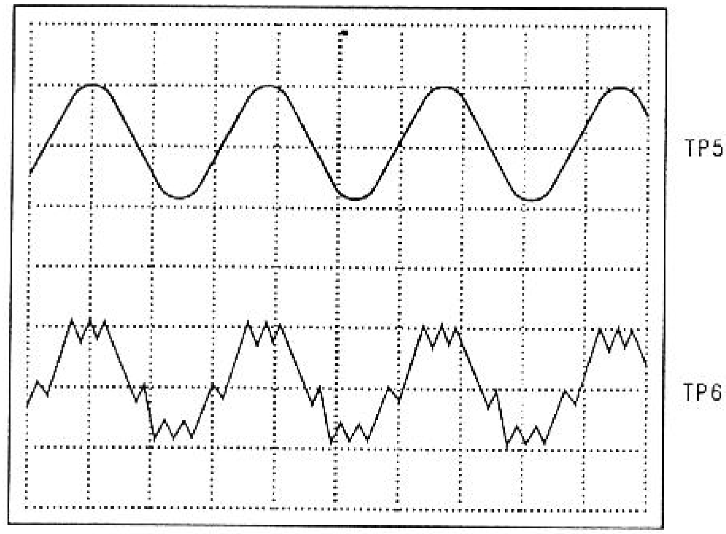
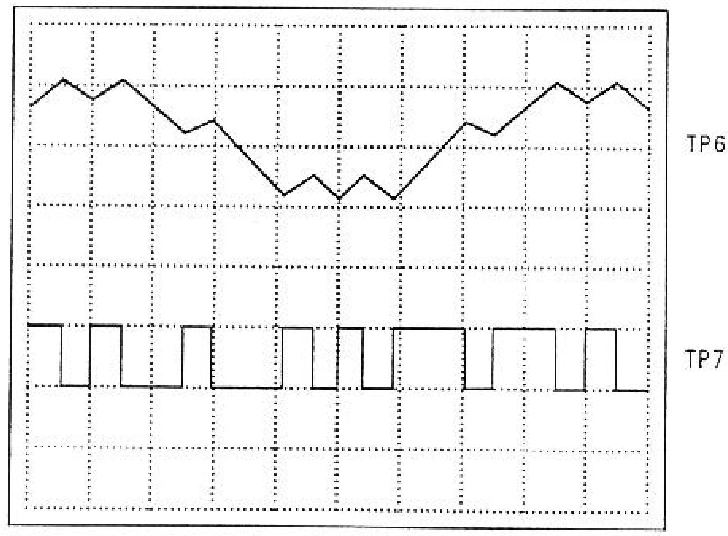


fig.3.3



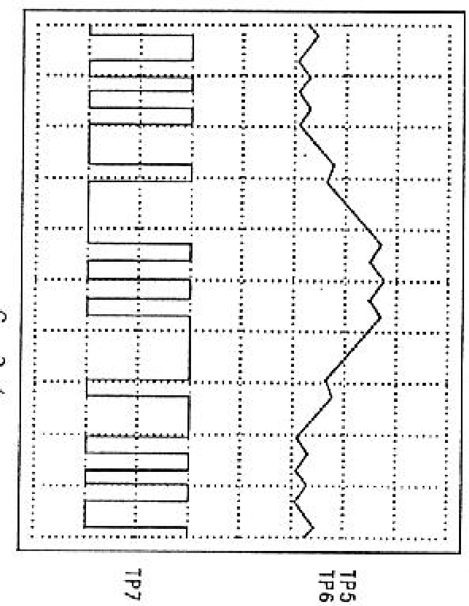


fig.3.4

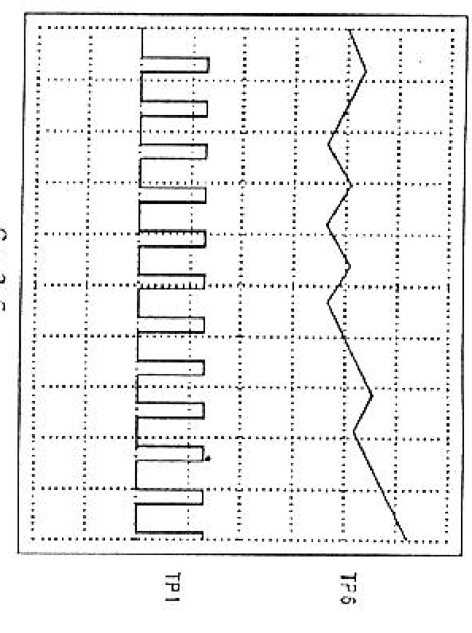


fig.3.5

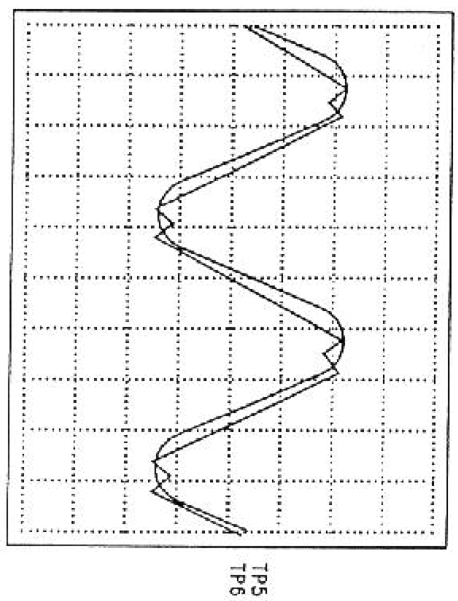


fig.3.6

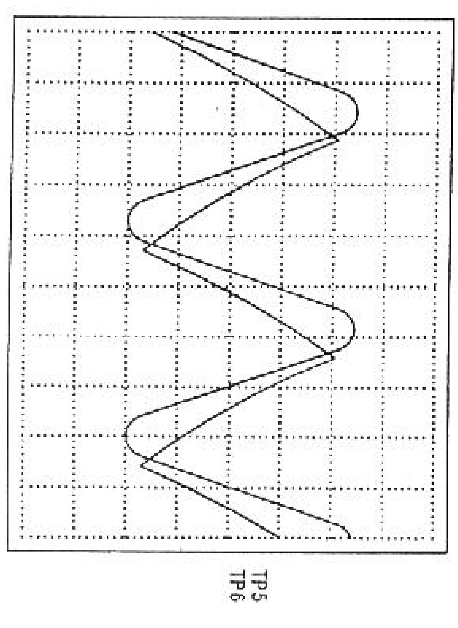


fig.3.7

digital output is high, whereas it is low when the saw-tooth signal is lower than the analog signal.

- Fig 3.5 shows that the analog signal and the saw-tooth signal are compared as the negative edge of the clock pulse occurs.
- Increase the amplitude of the input analog signal to approx 500mVpp: in TP5 and TP6 two waveforms are found, similar to the ones shown in fig 3.6
- Increase the amplitude still further (bringing it to almost 600mVpp) and make sure that the integrator cannot follow the analog input signal any mare with a saw-tooth signal Fig 3.7 highlights this fact.
- Bring the integrator GAIN (RV4) to the maximum value and make sure that now the integrator can approximate the input analog signal in a mare accurately.
- Turn RV2 to zero, in order wave to have no analog input signal: a square wave is found at the TP7 output. When this wave is integrated it generates a triangular wave in point 6: change RV3 and RV4 and observe the corresponding amplitude variation of the triangular wave.

Observations: Students are required to perform the practical as described above and write the observations.

EXPERIMENT NO:6

Object: To understand and observe the adaptive delta modulation.
(CVDM: Continuously Variable Delta Modulation)

THEORY: Granular noise and slope overload are peculiar drawbacks of Linear Delta Modulation: usually as one decreases the other increase and vice versa. Using Delta-Sigma modulation the slope overload can be reduced without worsening the granular noise, but this improvement is still limited.

The above-mentioned disadvantages can be effectively minimized using the companding (compressing/expanding) process. With this system, the modulator varies its gain according to the amplitude of the input analog signal.So, when the signal is low, the ramp amplitude is low and when the signal increases, the amplitude is higher too.

Therefore the ramp amplitude adapts itself to the signal amplitude, so this modulation is called adaptive or "continuously variable". The CVSD (Continuously Variable Slope Deltamod) is one of the most frequent and effective methods used for Adaptive Delta Modulation. It is applied to several integrated circuits on the market.

Required Instruments:

- Dual trace oscilloscope
- Frequency meter
- Low frequency generator
- Power supply 12V dc.
- Delta modulator circuit.

Procedure:

- Connect TP4 (ANALOG OUT) to TP5(ANALOG IN)
- Turn the two way switch SW1 to "CVSD"
- Turn RV3 (STEP SIZE) and RV4(Gain) to the maximum
- Turn RV1 to 32KHz
- Power supply the circuit with 12V dc.

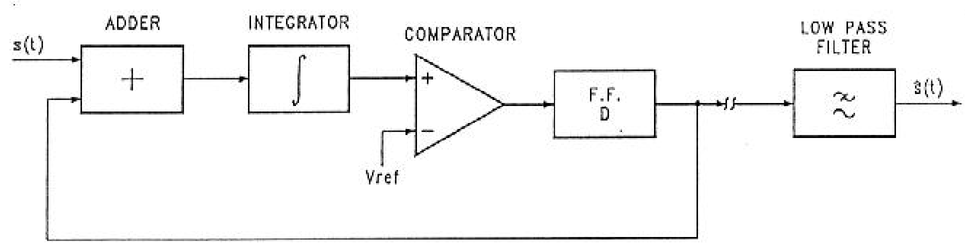


fig.1.7 DELTA-SIGMA Modulator and Demodulator

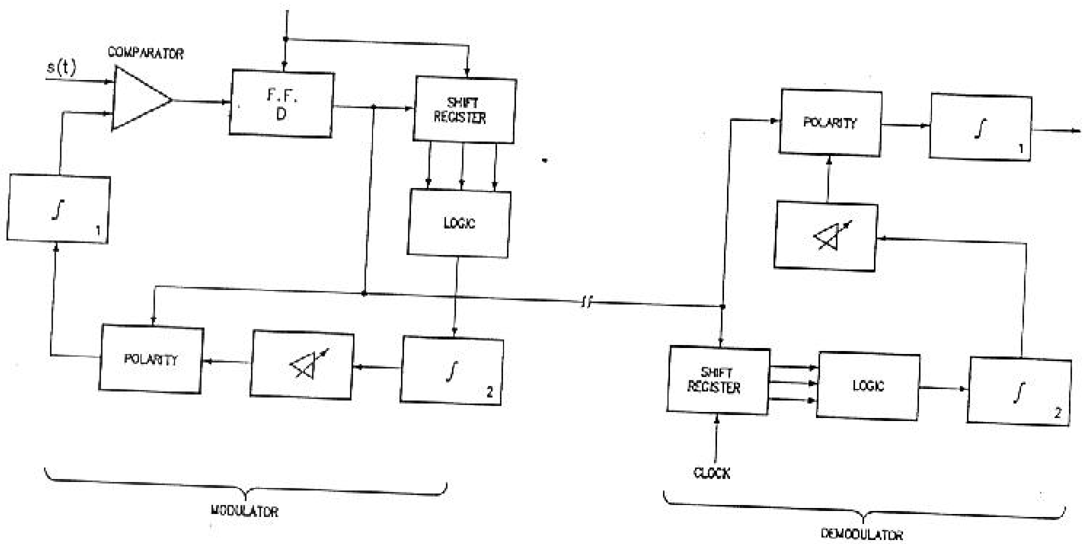
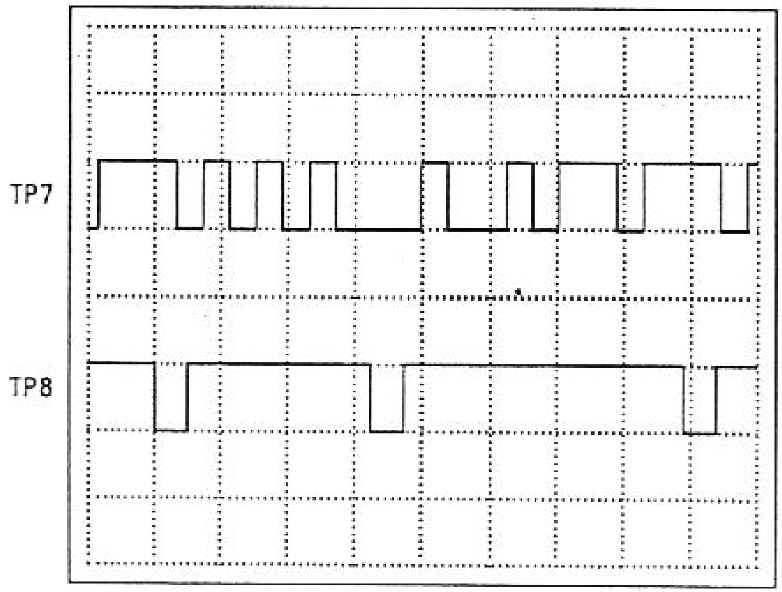


fig.1.8 Block diagram of Adaptive (CVSD) Modulator



flg.3.8

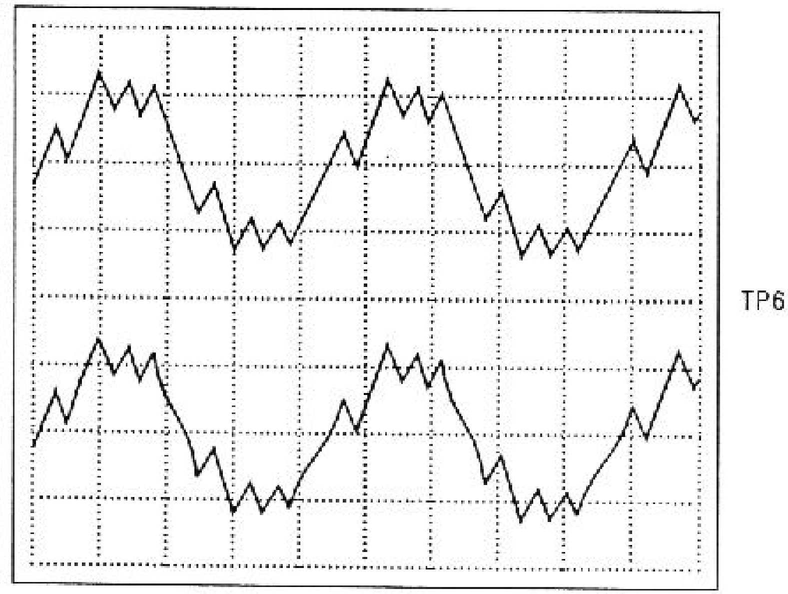


fig.3.9

Connect the oscilloscope to TP5 and adjust RV2 in order to obtain a signal of about 500mVpp

- Connect the other probe of the oscilloscope toTP6 and check that the saw-tooth signal follows the analog signal.
- Turn the switch SW1 to "LINEAR" and check that the sawtooth signal is reduced to a triangular signal so that it does not approximate the analog signal any more.
- Turn the switch back to CVSD and notice the step amplitude(approx 100mV)
- Increase the analog signal to approx 1Vpp and check the corresponding increase of the step amplitude (approx 150mV): this is due to the increase of the integrator gain brought about by the syllabic filter.
- Connect the oscilloscope to TP 7 and TP 8 .The digital signal generated by the delta modulation is displayed in TP7, whereas a negative pulse is delivered in TP8 (fig 3.8). Whenever the modulator output is kept high or low for at least three bits. These pulses are integrated by the syllabic filter and the resulting d.c voltage varies the integrator gain, as well as the ramp slope
- Measure the d.c voltage in TP 9.Oberve that it decreases as the amplitude of the input analog signal increases: measured values are included between 6V and 10V approximately.
- Check the influence of the syllabic filter on the integrator gain (and hence on the ramp slope) shunting a 6K8-resistor to R6 (between TP8 and TP9). This makes the time constant of the integrator decrease. Consequently the d.c voltage which controls the gain varies the ramp slope faster. Fig 3.9 shows the integrator output (TP6) under two conditions: 6K8 disconnected (high time constant) and connected (low time constant).

Observations: Students are required to perform the practical as described above and write the observations.

EXPERIMENT NO:7

Object: To understand and observe the linear delta demodulator

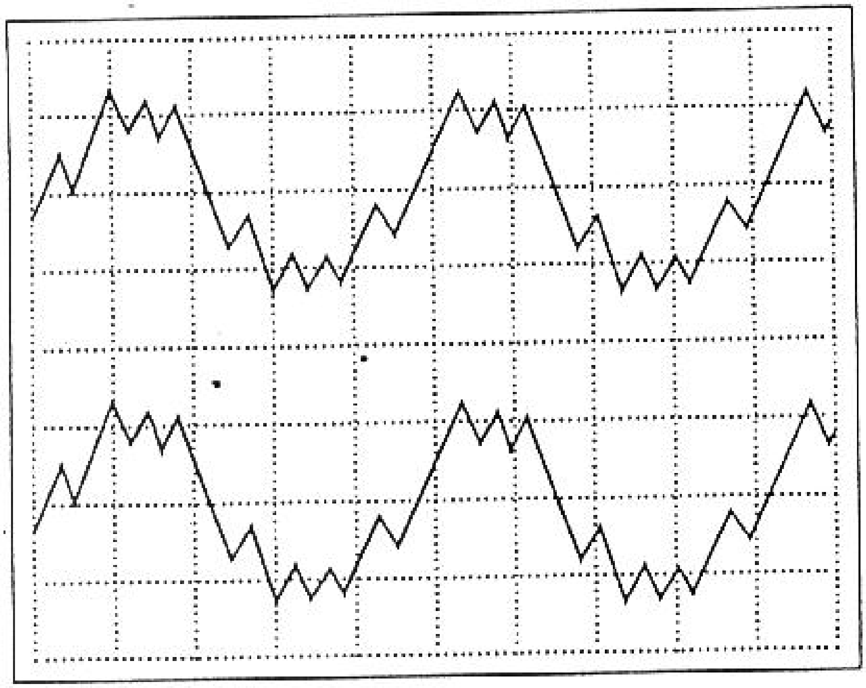
Required Instruments:

- Dual trace oscilloscope
- Frequency meter
- Low frequency generator
- Power supply 12V dc.
- Delta modulator /demodulator circuit.

Procedure:

- Prearrange the modulator and the demodulator for the linear operation.
- Connect TP7 to TP10 (DM OUT to DM IN)
- Power supply the circuit with 12V dc
- Turn RV1 to 32KHz
- Apply a sinusoidal signal with amplitude of approx. 40mVpp to the modulator analog input (TP5). This signal can be delivered from TP4.
- Turn RV3 (STEP SIZE) and RV4 (GAIN) to the minimum.
- Connect the oscilloscope to TP6 and TP11.Slightly adjust the GAIN (RV4) in order to obtain the waveforms related to the reproduced analog signals inTP6 e TP11.The first signal is used in the modulation process, the second one is the received signal proper (fig 3.10). The two signals are identical; the only difference lies in the delay of 1 clock time of the reception signal as compared with the transmission signal.

Observations: Students are required to perform the practical as described above and write the observations.



TP6

TP11

fig.3.10

EXPERIMENT NO:8

Object: To understand and observe the adaptive delta (CVSD) demodulator

Required Instruments:

- Dual trace oscilloscope
- Frequency meter
- Low frequency generator
- Power supply 12V dc.
- Delta modulator/demodulator circuit.

Procedure:

- Prearrange the modulator and the demodulator for the CVSD (Continuously Variable Slope Deltamod) operation.
- Carry out the same tests as in the previous experiment but increase the amplitude of the analog signal entering the demodulator. Check that the right modulation and demodulation can be carried out in "CVSD" operation.
- Connect the oscilloscope to TP8 and TP14; check that the pulses are the same (the demodulator pulse is 1 clock time delayed compared with the demodulator pulse).
- Connect the oscilloscope to TP14 and TP10.Check that in TP4 there is a pulse for each sequence of 3 bits either at high or low level in the delta signal entering the demodulator.

Observations: Students are required to perform the practical as described above and write the observations.

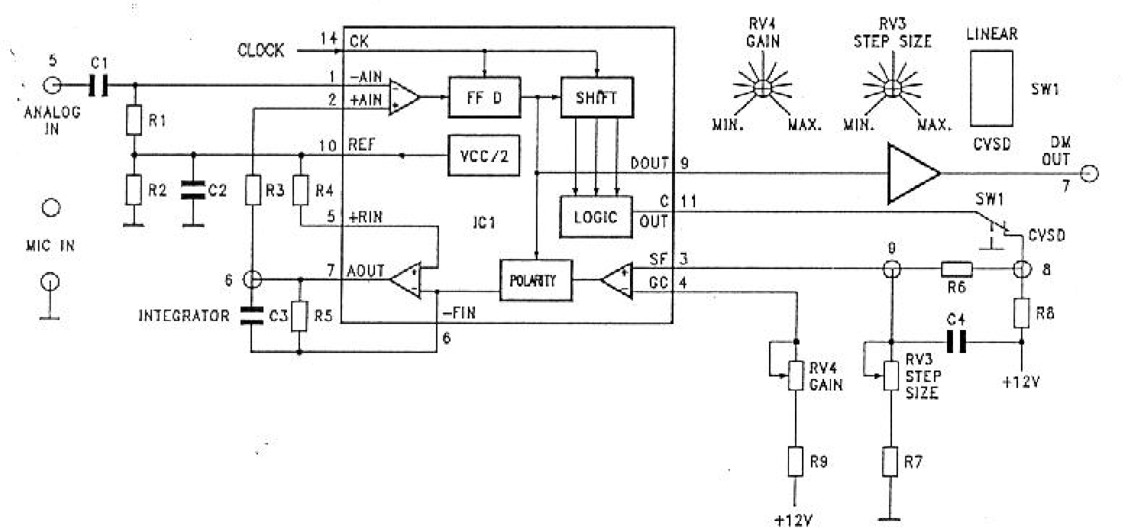


fig.2.4 Delta CVSD Modulator

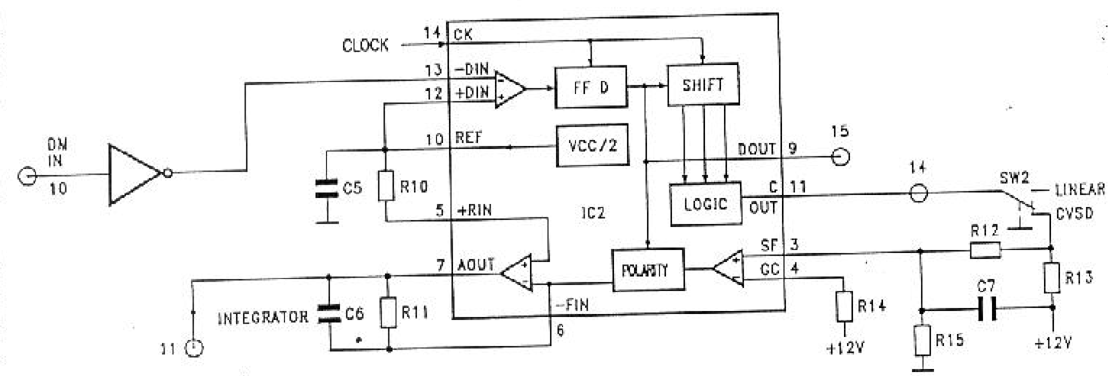


fig.2.5 Linear Delta and CVSD Demodulator

EXPERIMENT NO:9

Object: 1. Providing a functional description of the types of frequency synthesis carried out in the module T10L/EV.

2. Describing the configurations of the desired functions.

3. Checking the various operational modes.

Theory: Frequency synthesis is a process that enables to generate a frequency value being N times as high of a reference frequency, fr; this reference frequency is generally a fixed value obtained from a quartz oscillator by division.

N often is a decimal number of n digits, cn-1,cn-2,…co that can be varied individually from 0 to 9.

The minimum variation step of the output frequency, corresponding to one unit variation of N, is the reference frequency called the synthesis resolution.

When a frequency synthesis occurs as described above, it is defined as direct synthesis. The aim of frequency synthesizer consists in generating a sequence of discrete frequency values within a certain interval distributed around N/2.

When the desired frequency values have to be distributed within an interval included in a range of very high frequencies, it is better that the output frequencies of the synthesizer are down-converted within a range of lower values. This process is called synthesis with frequency conversion.

In this case the conversion frequencies will maintain the relation with the reference frequency, fixed by the factor N. The basic device normally used for the frequency synthesis is the PLL.

Necessary Equipment:

- Power supply PSU or PS1 with module holder.
- Testing module T10L/EV.
- Oscilloscope.
- Frequency meter.

Configuring the module:

Refer to the block diagram shown in the figure:1.1

Prearrange the module with J3 to B, J4 to A, J1 and J2 connected. and the selector S2 in the position of 1 kHz.

Prearrange the selection of the PROGRAMMABLE DIVIDER at 450.

Power the module with 12 Vdc.

Direct Synthesis:

After being prearranged as indicated above, the module is configured for the direct synthesis with resolution of 1 kHz.

This type of synthesis will undergo the following checks:

1. connecting the probes of the frequency meter and oscilloscope with TP12 where a sine wave of 2 Vpp is available, check that a frequency value of 450 kHz is available at the output.
2. measuring the voltage of the signal in TP11 with the other probe of the oscilloscope check that the mean control voltage of VCO is of approx 0V: within +/- 0.5 V.
3. switching the units of the selector PROGRAMMABLE DIVIDER upwards and downwards,check (frequency meter) the variation of the output frequency available in TP12 by corresponding steps of 1 kHz.
4. switching the tens of the PROGRAMMABLE DIVIDER upwards or downwards will lead to variation of the output frequency by steps of 10 kHz :in this case also the variations of the control voltage of VCO can be detected from TP11.

Far from the center frequency of the VCO the system is affected by the deviations of PLL parameters from the values of the optimum operation , so the PLL will lose the lock in state to

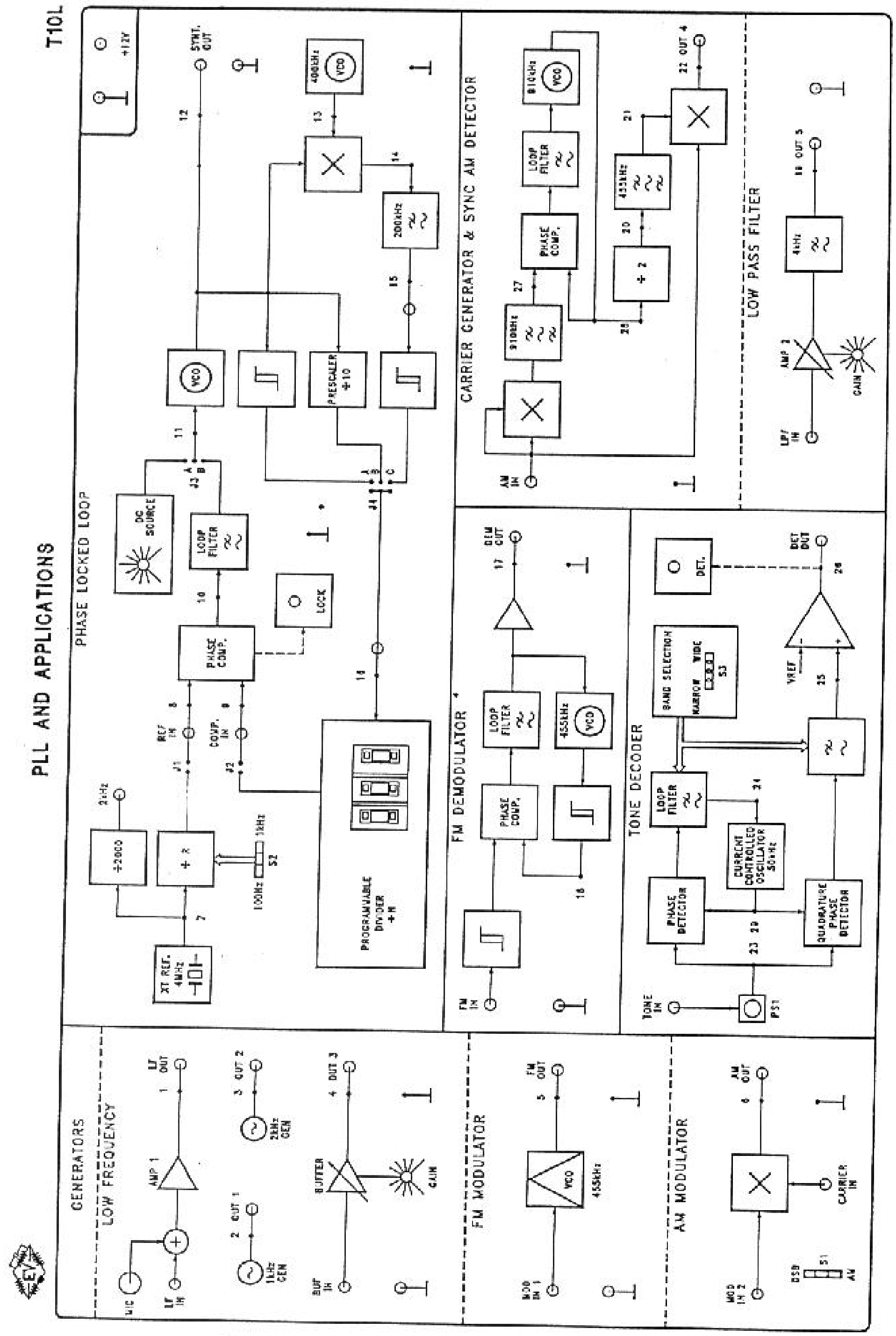


Fig. 1.1
Block diagram of the module T10L/EV

the reference frequency beyond a certain distance from the center alignment(450 kHz).

This cutoff frequency can be assessed as follows:

5. turn the PROGRAMMABLE DIVIDER to 600 and then increase the value of the tens up to the point where the measure of the output frequency becomes unstable and is not coherent any longer with the set synthesis value and the oscilloscope displays some jitters in the signal of VCO(TP12);now even the LED LOCK starts blinking: the upper limit of the synthesis corresponds to the frequency value set immediately before the last step(for this measurement with resolution of 10 kHz);

6. repeat the operations described in the point 5.above starting from an initial frequency of 300 kHz and reducing it by steps of 10 kHz, down to the lower limit of the synthesis that will always be the frequency value set immediately before the last step.

Assignment: Students are required to perform the practical as mentioned above and submit the observations.

EXPERIMENT NO:10

Object: To analyze frequency synthesis with conversion using the module T10L/EV.

Necessary Equipment:

- Power supply PSU or PS1 with module holder.
- Testing module T10L/EV.
- Oscilloscope.
- Frequency meter.

Theory: The down conversion applied to the signal of the VCO translates its frequencies to the range (centered at 50 kHz) that forms the input of the programmable divider.

In fact, the local conversion oscillator is of 400 kHz; consequently the difference with the center value of 450 kHz of the VCO is of 50 kHz that is the center value of the divider input.

Operation: When operating in this configuration with the reference of 1 kHz, and consequently with resolution of 1 kHz, set the hundreds of the division selector to zero.

This mode of synthesis can be assessed as follows:

1. position S2 at 1 kHz and J4 in C, setting the value 50 on the division selector; keep the probes of the frequency meter and of the oscilloscope connected with TP12 and shift the other probe(of the oscilloscope) from TP11 to TP16 (converter output);
2. check that the value of the synthesized frequency (TP12) is of 450 kHz and that of the frequency generated by the converter is of 50 kHz: if generally the assessment of frequency value carried out with the oscilloscope is sufficient for the converter output (TP 16),it is necessary that the trigger is enabled by this signal(in fact the sine waves of TP12 and TP16 cannot be observed simultaneously, being the trigger enabled by one of the two signals, because the ratio of the two frequency values will not be an integer hardly ever);

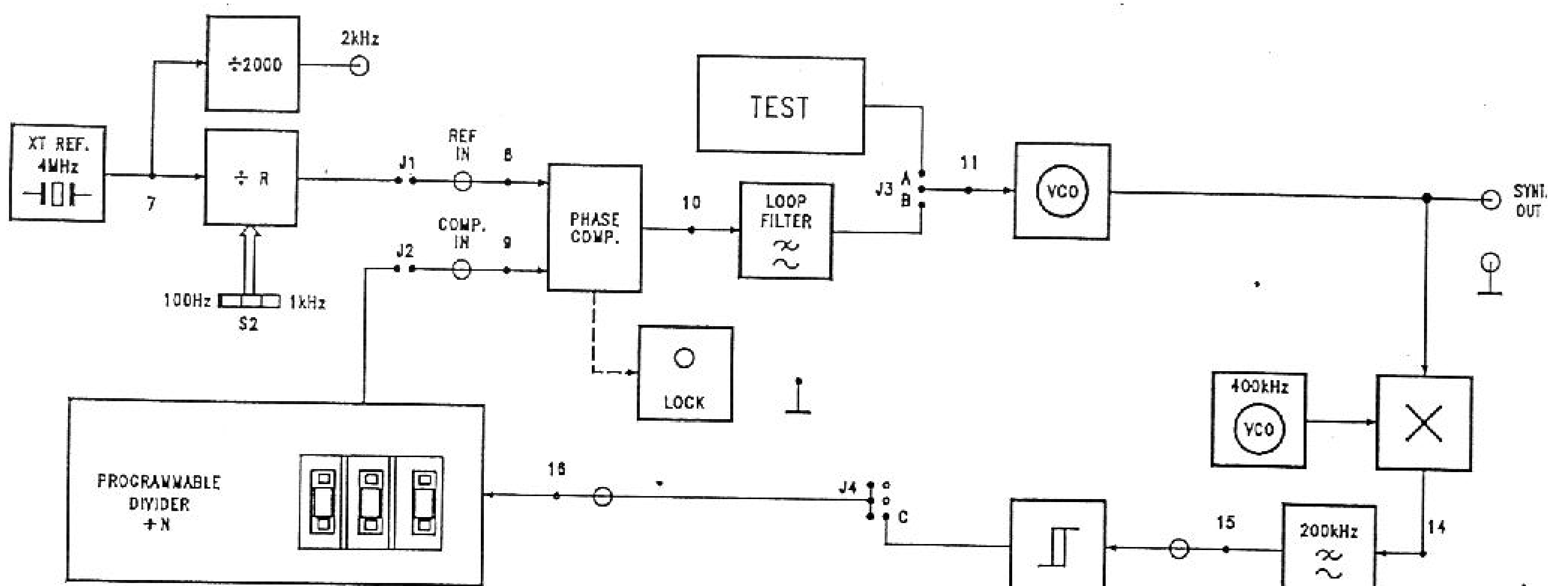


Fig. 4.2

Diagram of synthesis using a PLL and frequency conversion

3. varying the units of the selector (PROGRAMMABLE DIVIDER) will vary both the frequencies ,extracted from TP12 and TP16,by steps of 1 kHz;
4. switching the selector of tens of the PROGRAMMABLE DIVIDER will detect simultaneous frequency variations of 10 kHz in the signals of TP12 and TP16,with the method described above at the point 3.

Operational limits of synthesis:

As regards the synthesis under examination, it is important to remark that in this case too the mean operating of the PLL is shifted from the basic set-up.

The center frequency of VCO is always of 450 kHz, but the division factor is 10 times as low of the nominal value.

Assignment: Students are required to perform the practical as mentioned above and submit the observations.