Telecom Equipment - 2

TEM - 4

Centre for Electronics Design & Technology of India
A Scientific Society under Department of Electronics,
Govt. of India, New Delhi

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FOREWORD

The information technology and telecom sectors have suddenly opened up avenues, which require a very large specially trained manpower. These sectors are highly dynamic and need training and re-training of manpower at a rapid rate. The growing gap of requirement of the industry and its fulfillment has created a challenging situation before manpower training institutes of the country. To meet this challenge most effectively, Centre for Electronics Design and Technology of India (CEDTI) has launched its nation-wide franchising scheme.

Centre for Electronics Design and Technology of India (CEDTI) is an Autonomous Scientific Society under the Govt. of India, Department of Electronics with its Headquarters at New Delhi. It operates seven centres located at Aurangabad, Calicut, Gorakhpur, Imphal, Mohali, Jammu and Tezpur. The scheme will be implemented and coordinated by these centres.

The scheme endeavours to promote high quality computer and information technology education in the country at an affordable cost while ensuring uniform standards in order to build a national resource of trained manpower. Low course fees will make this education available to people in relatively small, semi urban and rural areas. State-of-the-art training will be provided keeping in view the existing and emerging needs of the industrial and Govt. sectors. The examinations will be conducted by CEDTI and certificates will also be awarded by CEDTI. The scheme will be operated through all the seven centres of CEDTI.

The CEDTI functions under the overall control and guidance of the Governing Council with Secretary, Department of Electronics as its Chairman. The members of the council are drawn from scientific, government and industrial sectors. The Centres have separate executive committees headed by Director General, CEDTI. The members of these committees are from academic/professional institutes, state governments, industry and department of electronics.

CEDTI is a quality conscious organisation and has taken steps to formally get recognition of the quality and standards in various activities. CEDTI, Mohali was granted the prestigious ISO 9002 certificate in 1997. The other centres have taken steps to obtain the certification as early as possible. This quality consciousness will assist CEDTI in globalizing some of its activities. In keeping with its philosophy of ‘Quality in every Activity’, CEDTI will endeavour to impart state of the art – computer and IT training through its franchising scheme.

The thrust of the Software Courses is to train the students at various levels to carry out the Management Information System functions of a medium sized establishment, manufacture Software for domestic and export use, make multimedia presentations for management and effectively produce various manufacturing and architectural designs.

The thrust of the Hardware Courses at Technician and Telecommunication Equipment Maintenance Course levels is to train the students to diagnose the faults and carry out repairs at card level in computers, instruments, EPABX, Fax etc. and other office equipment. At Engineer and Network Engineer levels the thrust is to train them as System Engineers to install and supervise the Window NT, Netware and Unix Networking Systems and repair Microcontrollers / Microprocessor based electronic applications.

An Advisory Committee comprising eminent and expert personalities from the Information Technology field have been constituted to advise CEDTI on introduction of new courses and revising the syllabus of existing courses to meet the changing IT needs of the trade, industry and service sectors. The ultimate objective is to provide industry-specific quality education in modular form to supplement the formal education.

The study material has been prepared by the CEDTI, document centre. It is based on the vast and rich instructional experience of all the CEDTI centres. Any suggestions on the improvement of the study material will be most welcome.

(R. S. Khandpur)
Director General (CEDTI)

TABLE OF CONTENTS

UNIT	CHAPTER NAME	PAGE NO.
1	Facsimile Machines	11
2	Modems	27
3	Electronic Private Automatic Branch Exchange	47
4	Cellular Telephony	67
5	Paging	75

PREFACE

Telephone is the most basic instrument in the field of telecommunication. The notes have been designed with the aim to provide the hardware knowledge of various telephone instruments, specifically covering the Push Button Telephones (Decadic & DTMF) and the Cordless Phones.

TEM-III has two chapters in all in which Chapter-1 covers the Push Button Phones and Chapter-2 covers the Cordless Phones. Each chapter provides the basic concepts, block diagrams, specifications, functional description alongwith detailed block diagrams and ICs details, features, followed by assimilation exercises and troubleshooting charts. The troubleshooting charts discuss the most common problems occuring in a particular instrument. The information provided in this document is according to the equipments currently available in the market.

Care has been taken to support the theoretical concept by illustrative figures. The information acquired by the document aims to be practical and comprehensive.

FACSIMILE MACHINES

COMPETENCY OBJECTIVES

The objective of this Chapter is to introduce the students to:-

- Introduction
- Facsimile Process
- CCITT Facsimile Standards
- Troubleshooting
- Features

Chapter 1

FACSIMILE MACHINES

INTRODUCTION

In this fast paced business world, obtaining the right information at the right time and in the right form offers a strategic advantage. Unlike telephones, which usually transmit only sound, fax transmits any graphic image, including signature, text, illustrations and photographs. With its unique ability to transmit existing graphics as well as text instantaneously, fax provides a communication capability that no other alternative technology offers.

Although a facsimile systems have existed for more than 25 years, only recently have we seen explosive growth in the use of facsimile with the corresponding increase in facsimile standard activity. Current standard work concentrates on improving the quality and speed of image transmission an increasing functionality. The goal is to benefit from the developing technology, higher-resolution scanners and printers and improved processing, storage and communication technology. Twenty five years ago, a fax machines were expensive, slow and use only by international stock brokers. However, recent technological advances have made these faster, quicker, and less expensive, allowing businesses to take advantage of their unique ability.

Facsimile is one of the original electrical engineering arts, invented by “Alexander Bain” in 1842. Many characteristics of signalling appeared first in facsimile. In its long development, facsimile has served as a mother art, spawing a variety of devices and methods, including the photocell, linear phase filters, adaptive equalizers, compression encoding, the daughter art of television, and the application to transform theory to signals and images.

DIGITAL MODULATION TECHNIQUES

In order to transmit digital information over bandpass channels, we have to transfer the information to a carrier wave of appropriate frequency. Digital information can be impressed upon a carrier wave in many different ways. In this chapter, we will study some of the most commonly used digital modulation techniques wherein the digital information modifies the amplitude, the phase, or the frequency of the carrier in discrete steps.

Figure shows four different modulation waveforms for transmitting binary information over bandpass channels. The waveform shown in Figure (a) corresponds to discrete amplitude modulation or an amplitude-shift keying (ASK) scheme where the amplitude of the carrier is switched between two values (on and off). The resultant waveform consists of “on” (mark) pulses representing binary 1 and “off” (space) pulses representing binary 0. The waveform shown in Fig.(b) is generated by switching the frequency of the carrier between two volumes corresponding to the binary information to be transmitted. This method, where the frequency of the carrier is changed, is called frequency-shift keying (FSK). In the third method of digital modulation shown in Fig.(c), the carrier phase is shifted between two values and hence this method is called phase-shift keying (PSK). It should be noted here that in PSK and FSK methods, the amplitude of the carrier remains constant. Further, in all cases the modulated waveforms are continuous at all times. Finally Fig.(d) shows a modulation waveform generated by the discrete PAM scheme described in the previous chapter.

The modulation scheme using baseband pulse shaping followed by analog modulation (DSB or VSB) requires the minimum transmission bandwidth. However, the equipment required to generate, transmit, and demodulate the waveform shown in fig.(d) is quite complex. In contrast, the digital modulation schemes are extremely simple to implement. The price paid for this simplicity is excessive bandwidth and possible increase in transmitter power requirements. When bandwidth is not the major consideration, then digital modulation schemes provide very good performance with minimum equipment complexity and with a good degree of immunity to certain channel impairments.

In a binary modulation scheme, it is assumed that the digital data source is binary giving out one of the two symbols at a time denoted by 0 and 1 and a waveform is transmitted by the modulator corresponding to each symbol 0 and 1. A modulator may wait for the source to give out more than one symbol, say k, and corresponding to each pattern of k length sequence a waveform may transmit a single waveform. This modulation scheme is called M-ary modulation scheme with $M = 2_k$.

We will now describe the binary modulation schemes generally used.

(a) Binary ASK

In this modulation scheme, the amplitude of a carrier is switched between two levels depending on whether 0 or 1 is to be transmitted. If one of the levels is zero volts, i.e., the carriers switched off, and the other symbol is represented by transmitting the carrier of amplitude,

$$\sqrt{\frac{(2E_b)}{T_b}}$$

Where T_b = duration of transmission for one binary symbol, E_b = Energy of the waveform representing one binary symbol.

(b) Binary PSK (BPSK):

In binary PSK modulation scheme the two signals representing the binary symbols are given by,

$$S_0(t) = \sqrt{2\frac{E_b}{T_b}}\cos(2\pi f_c t) \qquad 0 \le t \le T_b$$

$$S_1(t) = \sqrt{2\frac{E_b}{T_b}}\cos(2\pi f_c t + \pi) = \sqrt{2\frac{E_b}{T_b}}\cos(2\pi f_c t) \qquad 0 \le t \le T_b$$

This is also known as ON-OFF keying. Fig.(a) shows the waveform of an of-off keying modulation for a binary sequence 0 1 1 0 1 0 0 1.

Where f_c is the frequency of the carrier chosen to be equal to a large multiple of $1/T_b$.

Note that the signals corresponding to 0 and 1 differ only in their phase by 180 degrees all other characteristics of the carrier being same. Fig(b) shows the waveform of a PSK modulated binary sequence 0 1 1 0 1 0 0 1. Binary PSK is also called antipodal signalling.

(c) Binary FSK (BFSK)

In this modulation scheme, 0 and 1 are distinguished by transmitting one of the two sinusoidal waveforms that differ only in their frequency by a fixed amount. They are expressed as

$$S_0(t) = \sqrt{2\frac{E_b}{T_b}}\cos(2\pi f_0 t) \qquad 0 \le t \le T_b$$

$$S_1(t) = \sqrt{2\frac{E_b}{T_b}}\cos(2\pi f_1 t) \qquad 0 \le t \le T_b$$

where generally, $f_0 - f_1 = 1/T_b$. Fig(c) shows the corresponding FSK waveform for the same sequence considered for ASK and PSK.

QUADRATURE MODULATION

In a quadrature modulation scheme both the carrier $\cos(2\pi f_c t)$ and its 90 degrees shifted version $\sin(2\pi f_c t)$ are simultaneously modulated and transmitted as a single waveform expressed by

$$S(t) = S_1(t) = \cos(2\pi f_c t) - S_Q(t)\sin(2\pi f_c t) \qquad \ldots\ldots(1)$$

where $S_1(t)$ and $S_0(t)$ are called respectively the inphase and the quadrature components of the modulated wave. Both $S_1(t)$ and $S_Q(t)$ are related to the input binary data stream in a way that is characteristics of the type of modulation used. Note that the inphase component and the quadrature component can represent one binary symbol each and hence a quadrature modulated wave represented by two binary symbols. We elaborate this aspect with Quadriphase Shift Keying as an example.

Quadriphase Shift Keying (QPSK)

In this quadrature modulation scheme the phase of the carrier takes one of four equally spaced values $\pi/4$, $3(\pi/4)$, $5(\pi/4)$ and $7(\pi/4)$ radians as expressed by

$$S_i(t) = \sqrt{2\frac{E}{T}}\cos\left\{2\pi f_c t + (2i-1)\frac{\pi}{4}\right\} \quad 0 \le t \le T \quad i = 1, 2, 3, 4 \quad ......(2)$$

Where E denotes the energy of the signal and T denotes the symbol duration. Note that energy E is used to transmit two bits at a time and hence the energy per bit denoted by E_b is equal to E/2.

M-ARY MODULATION TECHNIQUES

While discussing Quadrature modulation techniques we saw that two binary symbols can be communicated at time by a single waveform corresponding to all possible pairs of binary symbols 00, 01, 10 and 11. Extending this technique to more than 2 say k symbols at a time and using M waveforms, where M = 2_k, is called a M-ary modulation scheme. We give two examples of M-ary communication systems by way of illustration.

(i) M-ary FSK (MFSK)

The signal used to transmit in this modulation scheme are given by

The signals used in this M-ary scheme can be expressed as

$$S_i(t) = \sqrt{2\frac{E}{T}}\cos\left\{2\pi f_i t + 2\frac{\pi}{M}i\right\} \quad 0 \le t \le T \quad i = 0, 1, 2, ..., M-1$$

Note that in this case M different phases are used to designate all possible M binary sequences of length k whereas in the previous case M different frequencies are used justifying their names.

ROLE OF FAX

The fax machine is somewhat like an office copier, or rather, two officer copiers electrically connected by a telephone line. Compare calling on a telephone with sending a printed page to a distant point. The speaker’s voice is changed by a microphone within the telephone into electrical signals sent over the telephone line to the listener’s telephone where the signals are changed back to sound by its earphone. For fax, the imaging portion of the “office copier” produces tone signals representing the page being sent. These tone signals pass over the telephone line connection the same way as the voice signals do for speech. At the receiving end, these signals are changed back into an image and printed by the other half of the “office copier.”

PRINCIPLE OF OPERATION

For sending by fax, the reading is done electronically, also starting at the top left corner of a page. Fax may read two or more pages a minute, much faster than most people do. Fax does not recognize the printed letters but reads the small black dots that form each character. Imagine the page being sent as printed on very fine graph paper with 200 squares per inch. Fax starts reading a line of type by reading only the rwo of squares across the tops of the printed characters. Each square is either a black or white dot. Starting at the left end of this row, the black dots plus all of the white dots are read in sequence (1728 dots in all). The process then repeats for the next row just below. It takes 10 to 20 additional rows (scanning lines) down the page to fully read one line of text. A whole page takes 2200 scanning lines (at fine resolution).

At the fax receiver, the imaginary squares corresponding to those squares covered by black markings at the fax transmitter are filled in with black dots. After all of the scanning lines have been read and sent to the fax receiver, all of the printed black dots are in the squares that match those at the transmitter. The black dots printed on the recording page form the characters and lines of the page sent. Thus a fax transmission is converted into a fax copy. The distance between the fax machine sending the page and the fax machine printing the page can be as far as can be reached by telephone.

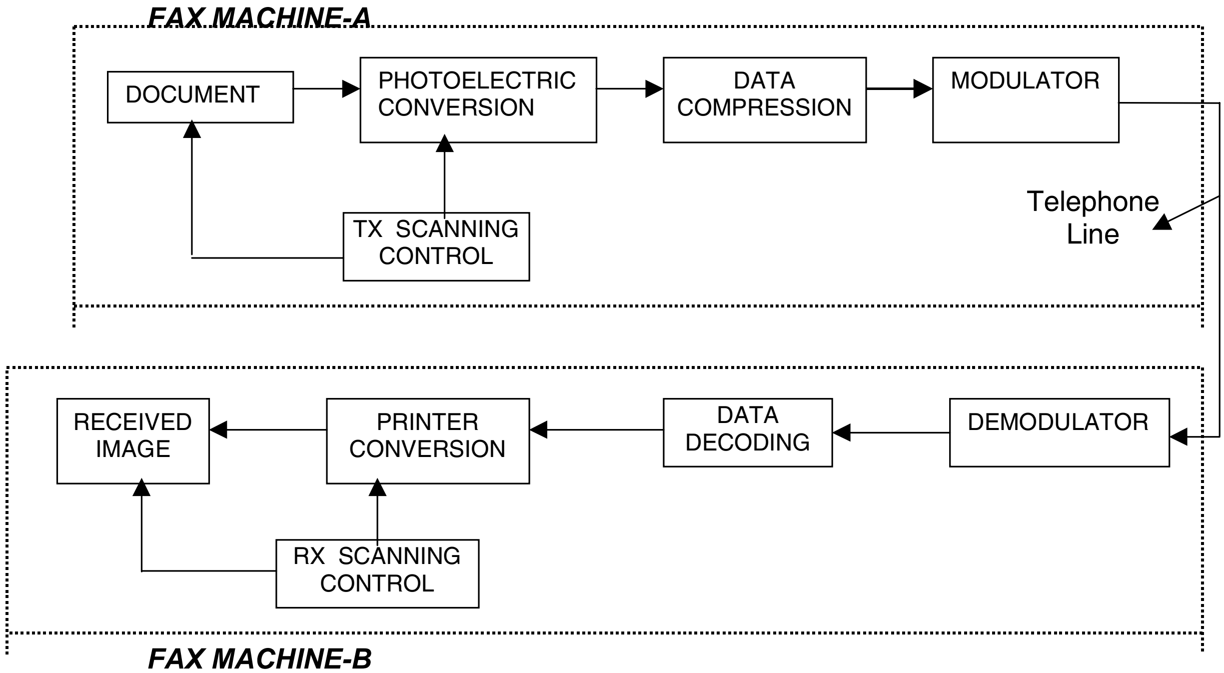
A light sensitive electronic device [e.g. a charge-coupled device (CCD) silicon chip] coverts the page image into electrical signals, with a strong pulse for each white dot and a weak pulse for each black dot. This electronic image signal is processed into digital format, compressed for faster transmission, and then coded by a modem to send even faster. The modem has a tone-type signal that will go through the regular dial-up telephone system in a manner similar to voice. The receiving portion of the called fax machine answers the call, decodes the received signal, and prints a copy of the original page.

The analog tone signals that Group 3 fax sends over a telephone line are generated by a very efficient, high speed synchronous digital modem. The receiving modem continually synchronizes on the symbols of the received signal. This results in decoding and storing in the fax receiver line memory, the position of black dots for a line across the page scanned at the transmitter. Each square across the page has its own writing element coupled to the corresponding memory element so that each black dot is in the correct position. The time that each dot is printed is sometime later than the time it was scanned.

FACSIMILE PROCESS

The facsimile process involves the following:

1. Optical Scanning
2. Recording
3. Encoding
4. Decoding
5. Signal Processing
6. Modulation
7. Demodulation
8. Transmission
9. Printing
10. Photocopying



HOW FACSIMILE MACHINE WORKS

A document, for, photograph, chart, graph, records, reports, data, text signatures, passport visas or any type of written matter can be transmitted by facsimile machine on ordinary telephone lines.

The document to be sent is first scanned and converted into an electronic format that can be sent over ordinary telephone lines. At the other end, the facsimile machine receives the messages, converts the electronic information and prints a copy which is an exact replica of the original document sent. The transmission times varies from a few seconds to minutes depending upon the speed of transmission, type of the machine, line conditions and other related factors. Facsimile is 40 less expensive than Telex, and has the additional advantage of sending the document in its original form with the company logo, format, information, signature etc.

ELECTRONIC SCANNERS

Photosensitive Arrays

Most current facsimile scanners use photosensitive arrays to scan flood-illuminated copy. An objective lens forms a reduced image of the subject copy exactly on the straight-line array of silicon sensors. For digital facsimile using CCITT Group 3, the array has 1,728 sensors in a 1.02 inch row to view an 8.5 inch wide subject copy, necessitating an image size reduction of approximately 8.3 times by the objective lens.

For lesser resolution requirement, the facsimile using CCITT Group 2, use three separate sensor units of 512 photo-sensors each providing 1.536 pixels per sweep. Although alignment of the mating edges requires high precision, the shorter optical paths save space. Two types of silicon photo-sensor arrays are in general use. The photodiode array is a monolithic, self-scanning linear array of silicon photodiodes. In each of the 0.00059-inch wide individual is a photodiode 0.000275-inch wide and 0.00063-inch high. Each photodiode is connected to a storage capacitor and an MOS switch. As light shines one the photodiode, it induces a reverse current through the photodiode, allowing capacitor charge to leak off. Every photodiode is sampled by switching in turn to an output. The size of the recharging current pulse to the capacitor measures the integrated light incident on the photodiode for the period since the last sampling. The charge-coupled-device (CCD) linear image sensor is functionally similar to the photodiode array. A CCD is semiconductor device in which isolated charge packets are moved by sequential clocking of any array of gates from one charge retention cell to the next. Since the amplitude of the charge packets can take on analogue values, a CCD can be used as an analogue shift register.

The CCD as a linear image sensor uses silicon sensors, 1,728 of them are used for Group 3 facsimiles. While light is incident on the silicon, charge is accumulated proportionally to the incident light flux. Transfer gates below the sensors transfer the accumulated charge to a CCD analogue shift register for the odd-number sensors. Clocking signals cause these charges to move down their respective analogue shift registers, where they emerge alternately into a preamplifier, the output of which is a discrete time series of analogue pulses corresponding to the diffuse reflectance of a row of elemental areas on the subject copy, reported left to right in turn

.The expected alignment of pixels reproduce a narrow vertical line. Horizontal position errors are called jitter, and vertical position errors are called clogging. Jitter or clogging errors in excess of 0.001 inches are usually disturbing. Mechanical sweep caused by errors in timing, and the clogging caused by errors in traverse or feed. Observed jitter and clogging on received copy is the sum of jitter and clogging due to the scanner and recorder.

Recording Media

Facsimile systems mark record copy by systematically applying (a) Electricity (b) Heat (c) Light (d) Ink jet or Pressure to the recording medium. Elemental areas of the record medium are marked individually and sequentially by rectilinear array or multispot scan. The tool used to apply the mark to the elemental areas is called the marking transducer of the facsimile system.

Marking transducers may apply the finished mark directly on the record medium in a one-step process, or the marking transducer may create a latent image, which is then rendered visible in a process requiring two , three, or more steps. Except for ink-jet, the simplest facsimile machines uses a one-step process requiring specially coated recording media. The faster and most complex facsimile recorders are designed to use plane paper, relaying on the paper cost difference to justify the multistep marking process.

CCITT FACSIMILE INTERFACE STANDARDS

The International Telegraph and Consultative Committee (CCITT), a part of the International Telecommunications Union (ITU), an arm of the United Nations, sets international telecommunications definitions and standards, including definitions of four categories of facsimile equipment and standards for the first three. The categories are called groups, as follows:

GROUP 1 : Facsimile apparatus which enables an ISO A4 page 21 0mm by 297 mm or 8.3” by 11.7”, to be transmitted over a telephone type circuit in approximately six minutes. Group 1 uses analog frequency-shift modulation signaling which will support a gray scale.

GROUP 2 : Facsimile machine runs at double the speed of Group 1 by using duo binary bandwidth reduction to have its Fourier spectrum group 2 delivers a page in three minutes over telephone type circuits.

GROUP 3 : Facsimile apparatus enables a typical A4 page to be transmitted by a digital modem over a telephone-type circuit in one minute or less by employing digital - data - compression techniques. The facsimiles used in our country generally fall into the CCITT recommended Group 3. We will be dealing with this group in the various facsimiles used in our country.

GROUP 4 : Facsimile apparatus uses digital-data-compression techniques and is interfaced directly to a digital data network in which procedures are incorporated to ensure error-free reception. Under these circumstances no modem is necessary. The Group 4 facsimile applications imply a store and forward character in which the minimum transmission time per total scan line can be zero. With zero minimum time and infinite K (constant) the time of transmission for high-resolution pages using Modified Read code is cut almost to half compared to Group 3 for the same bit rates.

Resolution: Resolution refers to the detail with which a document or image can be reproduced by a fax machine, similar to the photography. Basically, when it receives a document, a fax machine reconstructs the image of that document dot-by-dot. The more dots per inch, the finer is the resolution and clearer is the image. The Group 3 standard which is mostly used in facsimile defines two levels of resolution:

Standard and Fine: The type of resolution one uses depends largely upon the amount of visual detail in the document. Standard resolution is best suited for text-based information and simple line drawings. Fine resolution is most appropriate for material with very small print and illustrations that contain fine details.

Standard Resolution constructs images consisting of 203 horizontal lines per inch and 98 vertical lines per inch. Each inch of document is divided into 203 horizontal lines, each horizontal line consists of 98 dots. In comparison, a modern computer laser printer has a resolution of 300 dots per inch. A good quality 12-inch diagonal computer display has a resolution of about 100 dots.

Fine Resolution contains the same number of horizontal lines but each inch contains 196 instead of 98 dots. Because each fine resolution dot is half the size of a dot in regular resolution, an image consists of twice as twice as much information, and thus has significantly more detail.

Although facsimile deliver more detail than most computer and telecom screen, these are still unable to produce the fine details of a modern laser printer. As laser printers and facsimiles become integrated and advanced Group 4 modes become available, fax will gain even higher resolution.

CCITT GROUP 3 EQUIPMENT

(1) EQUIPMENT DIMENSIONS: The following dimensions are used for Equipment using A4 size.
- (a) A standard resolution and an optional higher resolution of 3.85 line/mm ± 1% and 7.7 line/mm ± 1% respectively, in a vertical direction.
- (b) 1728 black and white picture elements along the standard scan line length of 215mm±1%.
- (c) Optionally, 2048 black and white picture elements along a scan line length of 255mm±1%
- (d) Optionally, 2432 black and white picture elements along a scan line length of 303mm±1%

Dimensions for Equipment using A5 and A6 facilities:

- (a) Optionally, 864 black and white picture elements along a scan line length of 107mm ± 1%.
- (b) Optionally, 1216 black and white picture elements along a scan line length of 151mm ± 1%.
- (c) Optionally, 1728 black and white picture elements along a scan line length of 107mm ± 1%.
- (d) Optionally, 1728 black and white picture elements along a scan line length of 151mm ± 1%.

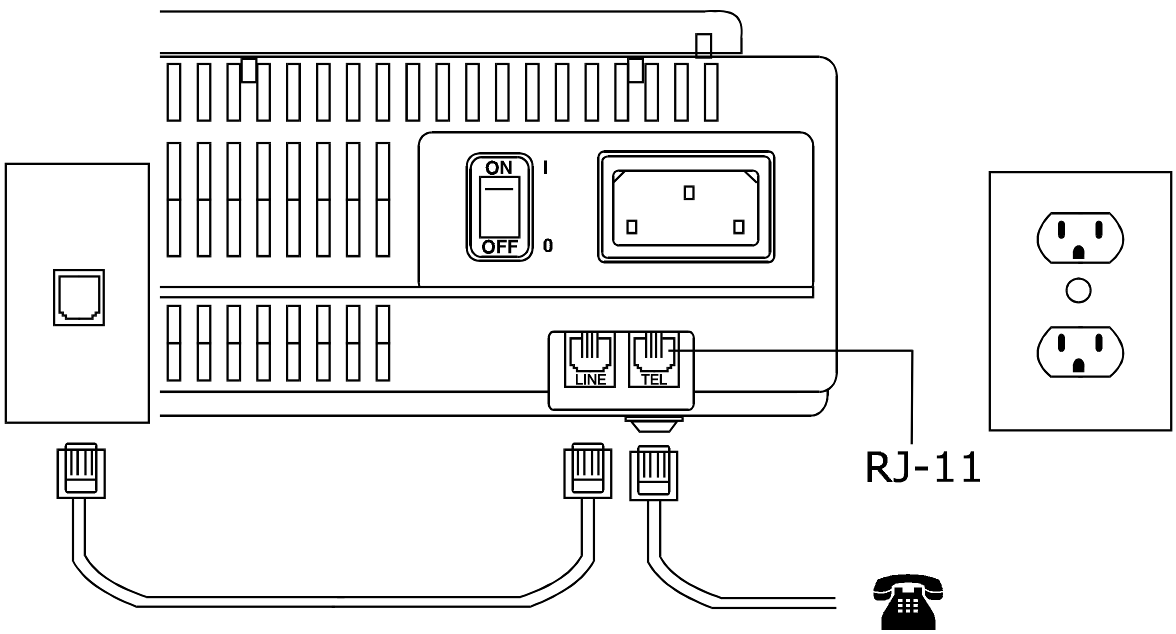
The normal method of inter-working when transmitting from an A5 or A6 machine to an A4 machine not signaling such capabilities is that the A5 or A6 content will be enlarged to fill the A4 page. This means that if the document is then retransmitted, or if it has been stored for later retransmission, it will be received without additional reduction.

FAX MACHINE INSTALLATION

SETUP

1. **Power Cord**
   Plug the power cord into a standard grounded, 3 wire outlet.

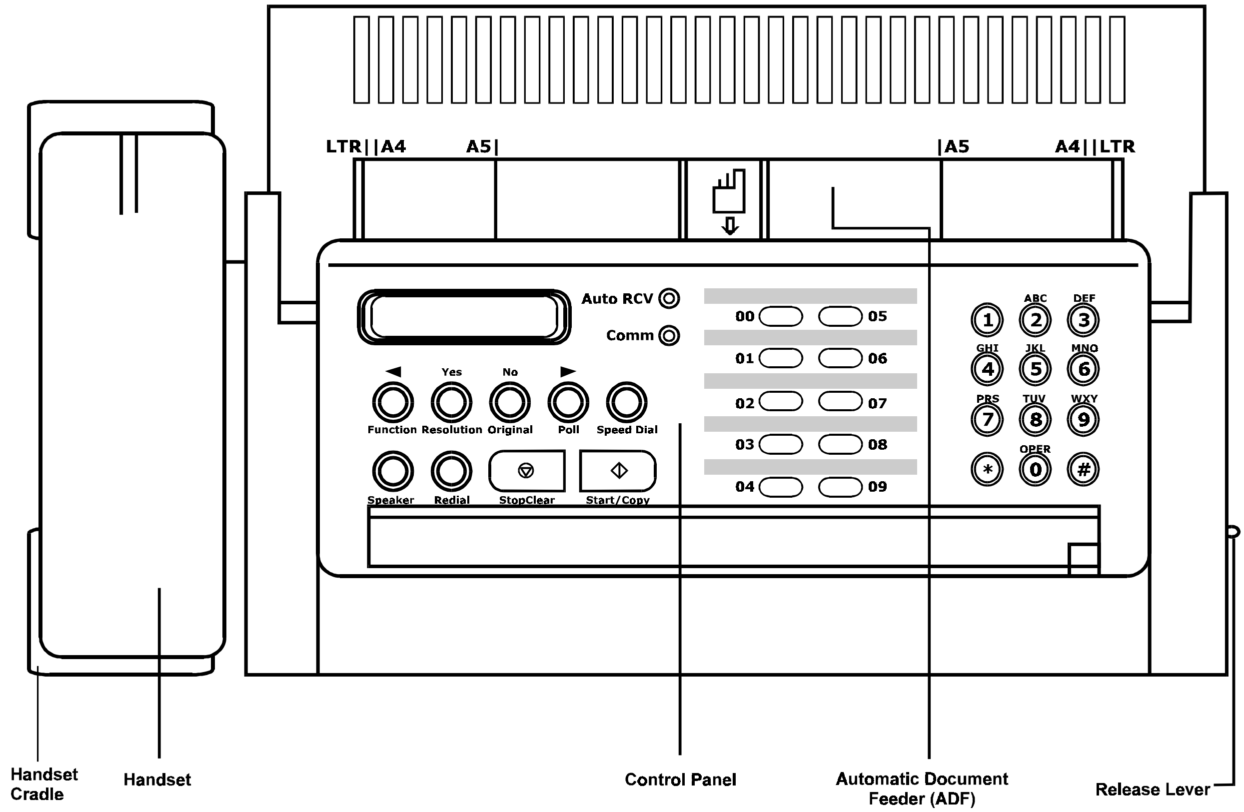
2. **Telephone Cable**
   Connect the one end of telephone cable to the jack of machine labeled “LINE” at the rear side of machine, and connect the other end of cable to the RJ11 wall jack.



3. **Handset cable (curl cord)**
   Connect the one end of handset cable to the jack of machine labeled “handset” at the left side of machine, under the cradle, and connect the other end of cable to the jack of Handset.

4. **Paper Load**
   a. Open the operator panel assembly.
   b. Load paper roll following the instructions of the paper ex label attached on the frame.
   c. Close the operator panel assembly.

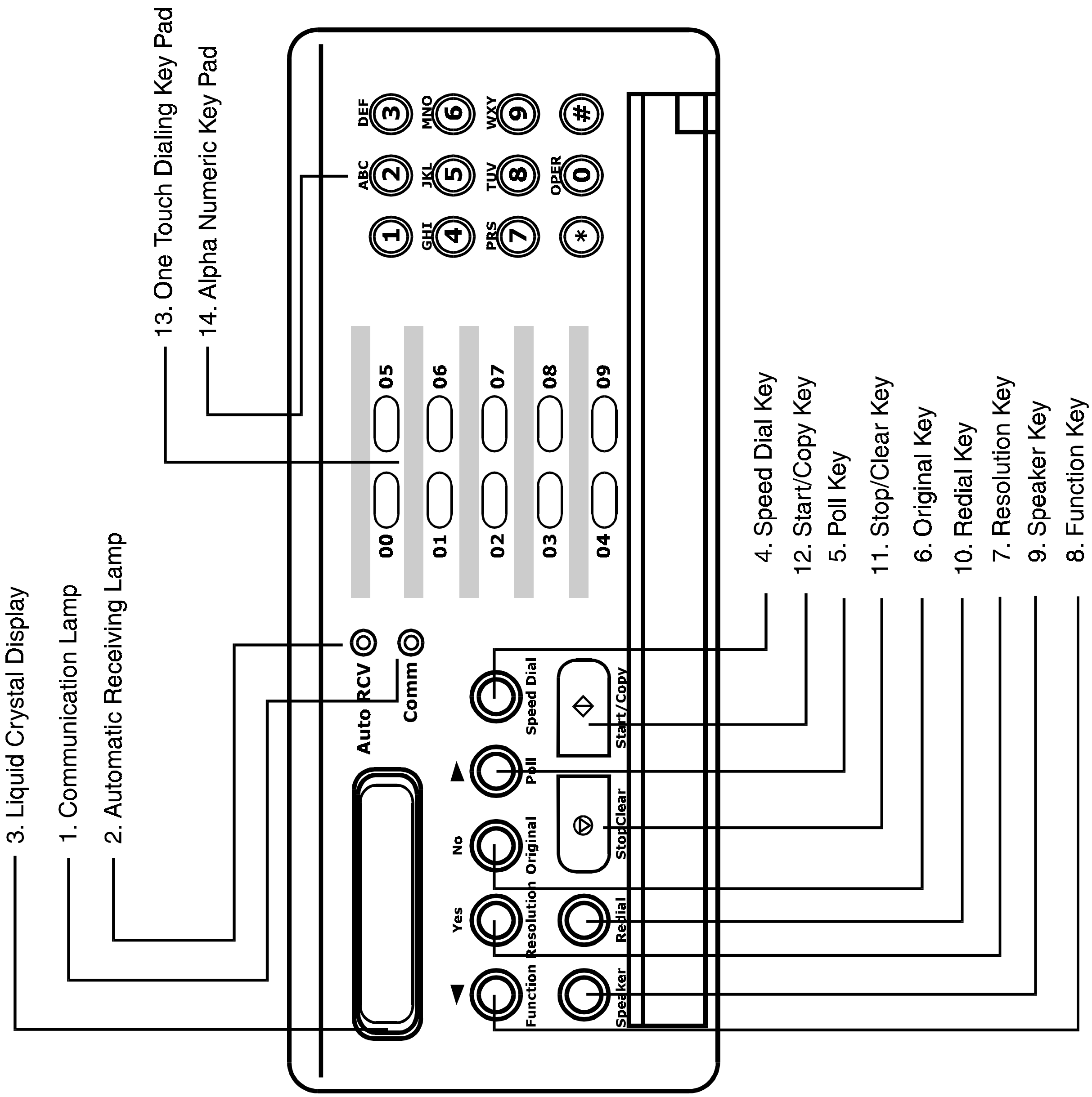
5. **Install the stacker**
   Insert the end of stacker to the hole on the top Cover.



SPECIFICATIONS

Type
Scanning System
Compatibility
Transmission Speed
Transmission Time
Compression System
Resolution
Document Size
Scanning Width
Recording System
Recording Paper Size
Display
Speed Dial
Document Feeder
Image Control

OPERATION PANEL



1. Communication Lamp

When line is used for sending/receiving documents or for telephone call, it lights. And when the speaker key is pressed for voice request during sending or receving documents, it also lights.

2. Automatic Receiving Lamp

When the fax is set to receive documents automatically, it lights. When the fax is performing settings, registrations or other functions, it blinks.

3. Liquid Crystal Display (LCD)

This 32-character LCD shows various messages regarding the operations, programming and document processing.

4. Speed Dial Key

Press before dialing pre-programmed coded dial numbers.

5. Poll Key

Press to poll other faxes or to allow other units to poll your fax. In the Function mode, use this key to move the LCD cursor to the right.

6. Original Key

Press to select the proper settings if your original documents are too dark or too light. In the Function mode, use this key to change current settings, registrations or to answer negatively.

7. Resolution Key

Press to increase the clarity of documents transmitted or to send photographs. In the Function mode, use this key to change current settings, registrations or to answer affirmatively.

8. Function Key

Use in combination with other keys to perform settings, registrations or other functions. In the Function mode, use this key to move the LCD cursor to the left.

9. Speaker Key

Press to dial with the receiver on the hook. Also, press this key when using the fax line as a telephone when not sending/receiving documents.

10. Redial Key

Press to redial the last telephone number called. Also use this key to insert a pause (a two-second delay in dialing between numbers) when dialing or when programming fax/telephone numbers. Redial is automatic when a document has been inserted for transmission to make a fax call.

11. Stop/Clear Key

Press to stop, clear an opreation or to eject original documents.

12. Start/Copy Key

Press to start when copying, sending or receiving documents or to eject original documents.

13. One Touch Dialing Keypad

Press to dial pre-programmed One Touch Dialing numbers.

14. Alpha-Numeric Keypad

Use these keys to dial and program information. They operate just like the keys on a tone-dialing telephone.

 # Key : In the function mode, use this key to display alpha characters A - Z (forward) and to display the next character.

*** Key :** In the function mode, use this key display alpha characters Z - A (backward) and to display the previous character.

15. Release Lever

Use this lever to open the fax in order to replace paper, correct paper jams, etc. (located 1" from front end of unit on right side).

INITIAL CHECKS

Turn power switch on (the Date and Time appear in the display window).

Initial Memory Clear
Before continuing with checks any invalid data in memory must be cleared.

Internal Test

Adjustment of Output level

Programming

All programming the setting according to Operating Manual.

On-Line test
Perform On-line send and receive operation in customer network.

DISASSEMBLY / REASSEMBLY

Assembly is assumed to be in the reverse order of disassembly.

HANDSET

a. Warning
Turn off the ON/OFF switch & remove the power cord from the wall outlet.
b. Remove the modular jack from the main board assembly and/on handset.
c. Remove the rubber cap & screw.
d. Push the center of the handset.
e. Lift up.

OPE. PANEL ASSEMBLY

a. Warning
Turn off the On/Off switch and remove the power cord from the wall outlet.
b. Pull the locker to open the OPE panel.
c. Remove 2 screws.
d. Lift up the OPE panel.

Note:
1. There are two hooks on the rear edge of the OPE panel that secure it to the chasis.
2. If not remove the OPE panel and/or OPE PBA, don't remove the OPE PBA screw. Put the OPE Panel Assembly on the Chasis after remove it from Chasis's hook.

OPE PANEL & OPE PBA

a. Warning
Turn off the ON/OFF switch and remove the power cord from the wall outlet.
b. Remove the OPE PANEL Assembly.
c. Remove the 15 screw that secure OPE Panel & OPE PBA.

TOP COVER ASSEMBLY

a. Warning
Turn off the ON/OFF switch and remove the power cord from the wall outlet.
b. Remove four screws on bottom and rear side of the machine.
c. Lift up with push the top cover, attention to the outlet on the back side.

POWER ASSEMBLY

a. Warning
Turn off the On/Off switch and remove the power cord from the wall outlet.
b. Remove Top Cover Assembly
c. Remove 3 Screws
d. Lift Up.
e. Disconnect from main board.

ADF ROLLER ASSEMBLY & DOC SENSOR ASSEMBLY

a. Warning
Turn off the ON/OFF switch and remove the power cord from the wall outlet.
b. Remove Top Cover Assembly.
c. Remove OPE Panel Assembly.
d. Remove upper screw of the Latch Wire.

FRAME ASSEMBLY & MAIN BOARD ASSEMBLY

a. Warning
Turn off the ON/OFF switch and remove the power cord from the wall outlet.
b. Remove Top Cover Assembly.
c. Remove OPE Panel Assembly.
d. Remove Power Assembly.
e. Disconnect from Main Board.
f. Remove 2 Screws
g. Lift up the Frame Assembly
h. Remove 3 screws. Lift the Main Board Assembly (Notice the 2 Snap fit).

RX, TX MOTOR & PLATTEN ROLLER

a. Warning
Turn off the ON/OFF switch and remove the power cord from the wall outlet.
b. Remove Top Cover Assembly.
c. Remove OPE Panel Assembly.
d. Remove Power Assembly.
e. Disconnect from Main Board.
f. Lift up Frame Assembly.
g. Remove E-Ring & Gears.
h. Remove 2 screws.
i. Release the TX Bracket Assembly.
j. Remove 2 screws.
k. Release TX Motor.
l. Remove 2 screws.
m. Release RX Motor.
n. Remove 2 E-Ring & Bushiny.
o. Release Platten Roller.

C. I. S. ASSEMBLY & DOOR SENSOR & PAPER SENSOR

a. Warning
   Turn off the ON/OFF switch and remove the power cord from the wall outlet.
b. Remove Top Cover Assembly.
c. Remove OPE Panel Assembly.
d. Remove Power Assembly.
e. Remove Base Assembly.
f. Remove the upper screw of the Latch Wire.
g. Remove RX Motor.
h. Remove 2 special screws.
i. Lift up CIS Assembly.
j. Remove Screw.
k. Pull the Door Sensor.
l. Pull the paper sensor on the bottom of the frame assembly.

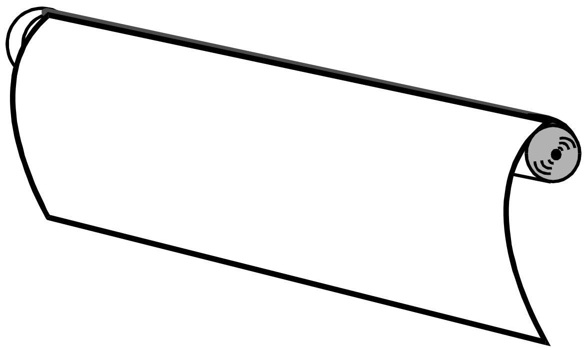
CUTTER ASSEMBLY

a. Warning
b. Remove Top Cover Assembly.
c. Open OPE Panel Assembly.
d. Remove 2 screws.
e. Lift up Cutter Assembly.

OPERATION

LOADING PAPER ROLL

1. Place the paper in the trough.
   (Note : Paper must unwind over the top)
2. Pull paper forward allowing 1" to extend out from unit and close the cover firmly.
   (Note : When it is time to change the paper, the alarm will sound and the LCD will show PAPER END).



DOCUMENTS

Paper Type

This fax will process documents printed on either standard bond or copier paper.

NEVER attempt to transmit documents which are:

- Wet with water, ink, or paste, wrinkled, curled, torn, or folded.
- Too thin (onionskin, air, mail paper etc.), card stock, newsprint, coated (glossy), cloth, metal, or carbon-coated.

To transmit these types of documents, make a photocopy or place the document in the carrier sheet.

Document Size

Below are the acceptable widths and lengths of documents you can transmit. If you want to send a document of outside these dimensions reduce it or enlarge it to an acceptable size.

Attended:	5.8” - 8.5” Wide 5”-40” Long
Unattended :	5.8” - 8.5” Wide 6” -12” Long (Automatic feeder)
Effective Scanning Width:	8.3”
Effective Recording Width:	8.3”

LOADING DOCUMENTS

1. Remove all clips, staples, and similar objects from the document. (If the document is not standard bond or copier paper, be sure to make a photocopy or place it in an optional document cover).
2. Place document face down with the top edge at the entrance. Adjust the sliding document guides to ensure that the document is centered as it enters the machine.
3. Slide document into the Automatic Document Feeder (ADF) slot until the ADF takes document and loads it for copying or transmission. When properly loaded, the display will read DOCUMENT SET.
4. ADF allows you to load up to five documents at once. When loading more than one document, be sure the bottom document loads first. (Note: When different types of paper are stacked, the ADF may not work properly. In this case, feed documents into machine one at a time while transmitting. For delayed transmission of such documents, make photocopies to transmit.)

USING THE TELEPHONE

1. You can use the fax as a regular telephone, simply pick up the handset to take calls or dial outgoing calls using the alphanumeric keypad.
2. For on-hook dialing, press [Speaker]. When you hear the dial tone, enter the number you wish to reach on the keypad without picking up the handset. When the party you

are calling answers, pick up the handset and begin talking. The speaker allows you only to hear the dial tones and the party you are calling. It does not act as a microphone to transmit you voice.

Note: If [Speaker] or [Stop/Clear] is pressed throughout any of the above steps, the machine will return to standby state. If [Start/Copy] is pressed while using the telephone, transmission or receipt of document(s) will begin and conversation will no longer be possible.

Note: When the fax rings, pressing [Speaker] allows you to monitor the voice or fax tone of the incoming call.

MAKING COPIES

Fax's copy function is convenient for desktop copying, and can also be used to see what a document will probably look like at the receiving station. When making copies, this fax automatically sets Resolution to "FINE".

1. Load document(s).
2. Press [Start/Copy] to begin copying.
3. When making copies of more than five documents, insert additional pages while at least one page is still in the Automatic Document Feeder.
4. To stop the copy process, or eject document before copying begins, press [Stop/Clear].

Note: When the Handset is off the hook, it is impossible to make copies.

CIRCUIT DESCRIPTION

1. Ring Detector Circuit

Capacitor and Resistor control ringer impedance. Photocoupler detects the AC ringing signal and pulses drive the ring detector.

2. Off-hook Detector Circuit

Internal telephone OFF-hook detector circuit monitors the OFF-hook handset by circuit flow about 30mA current and photo coupler detects this current.

3. Dial Pulse Out Circuit

Dial pulse is controlled by programming the unit. When programmed for Dial pulse, DP out relay is set to ON. The CPU supplies the make/break ratio.

4. TX / RX Circuit

The TX / RX circuit exchanges data between a public telephone network and a facsimile machine.

Transmission of data to the subscriber line is through the MODEM and transmission gain control circuit by CPU.

Also incoming data from the subsciber line enters the CPU through the Receiver.

AMP and MODEM. The data is printed by the TPM by means of decoding.

5. Input Sensor Control Circuit

This circuit has 8 input ports, polling is from CPU to sensor port. In this case, input data reads the data bus through the ASIC.

6. Auto Cutter circuit

A cutting pulse is generated pulse of 150MB at ASIC. This enables the paper cutter motor by energising relay.

7. Thermal Print Interface Circuit

Printing utilizes the thermal recording method when a fax is received and/or a copy is made.

8. Document reading circuit

Document is read by the contact image sensor, using the following principle. The ST, CLK, DIS receive input from the ASIC.

9. Operator panel interface circuit

Main CPU and one chip microprocessor of the operator panel exchanges data by TX port and RX port.

10. Battery Back-up circuit

This circuit is used for data back-up of SRAM and RTC when power is turned off.

MAINTENANCE

1. **Paper Jams**
   Do not register documents which have paper clips or staples still attached. If a document is accidentally registered with paper clips or staples, pull forward on the release lever. This will release the document in an emergency and, if done in time, prevent damage to the fax machine's sensitive internal parts. This can also be used to release jammed paper.

2. **Paper Dust**
   A can of filtered air can be used to blow out paper dust brought into the machine by

documents and fax paper.
After disconnecting the wall plug, blow air on the recording head (8½" glass-locking device). To further improve the printing performance, clean the recording head with a lint-free soft cloth dampened with 98% isopropyl alcohol or 70% ethyl alcohol (rubbing alcohol). Never use oil based cleaners (paint thinners, gasoline etc.) and never touch the recording head with a hard object.

3. **Fax Paper**
If the fax paper is allowed to become loosely rolled, the roll will become egg-shaped causing the paper to jam. Always roll the paper tightly when installing the paper roll.

4. **Paper Replacement**
a. Pull release lever forward to open the top cover.
b. Place the paper roll in the paper trough.
Note : Paper roll must unwind over the top.
c. Pull the paper forward allowing 1" to remain out and close the lid.
Note : The alarm will sound when it is time to change the paper and the display window will read "Paper End".

TROUBLESHOOTING

1. **Cannot Dial Out (no dial tone)**
Check the telephone line connections. If they are OK, plug a telephone into the fax line to test for a dial tone. If you can't hear the dial tone, the telephone line is defective.
If there is a dial tone and you do not have tone service, program or re-program for PULSE 20PPS or PULSE 10PPS (depending on your service) rotary dial mode instead of tone.

2. **Document Feeds Askew**
Straighten the document either before or as it feeds through the machine.

3. **Fax Machine Does Not Operate**
Check the power line connection. Be sure the power switch is in the ON position. Connect another appliance to the AC outlet to check if the outlet is in proper working condition.

4. **Fax Paper will not Feed**
Check the fax paper roll. It must be tightly rolled and rounded in order for it to feed. Paper rolls that are damaged will cause paper to jam.

5. **Print Quality Differs Between Sending and Receiving**
Use the fax to make a copy of the document and compare it with the original. If there is a difference, clean the recording head as outlined in the Maintenance section.

6. **Error Code Number Appears**
Check the Error Code List to determined fault.

7. Reset Procedure

When you encounter any unusual problem with you fax, you can reset the machine by turning the power switch off and waiting 10 seconds before turning it back on.

OTHER TROUBLESHOOTING POINTS

Dead	:	1. OPE Wire assembly Check. 2. Power wire assembly check. 3. Power supply change 4. Main board change
LCD Display dead	:	1. OPE Wire assembly check. 2. Power wire assembly check. 3. Power supply change. 4. Main board change.
Document jam	:	1. Chassis check. 2. Tx Bracket check. 3. Drive Roller. 4. ADF roller Assembly check.
Drive Roller not moving	:	1. Chassis Check. (Roller movement check)
Copy not Proper	:	1. TPH wire connector check. 2. Main PCB change.
Tx Motor not working	:	1. Tx motor wire assembly check. 2. Power Supply check. 3. Sensor change. 4. Tx motor change. 5. Main board change.
Rx motor not working	:	1. Rx motor wire assy. check 2. Power supply check 3. Rx motor change. 4. Main board change.
Copy black	:	1. CIS Wire Assy. check. 2. T.P.H wire assy. check. 3. Main PCB change. 4. Power supply change

Copy white	:	1.	CIS wire check.
		2.	TPH wire check
		3.	Main PCB change.
		4.	Power supply change.
Copy very light or dark.	:	1.	VR adjust
		2.	VR change
Paper out continuous	:	1.	DOC sensor check.
		2.	Main PCB change
Copy Stop	:	1.	Power Supply change.
		2.	Rx. motor check.
		3.	Main board change.
Cutter jam	:	1.	Cutter gear and cutter Assy. check.
		2.	Motor start check.
		3.	Cutter change
		4.	Main PCB change
DOC jam	:	1.	DOC sensor change.
		2.	Tx Motor check.
		3.	Main Board change.
Over heat	:	1.	Power supply change
		2.	Check main board.
No speaker sound	:	1.	Speaker wire check
		2.	Speaker change.
		3.	Main PCB change.
Distorted speaker	:	1.	Speaker change.
		2.	Main PCB change.
Speaker sound low	:	1.	Speaker change
		2.	Power supply change
		3.	Main board change.
Door not locked	:	1.	Locker shaft check
		2.	Adjust door locks.

THE FOLLOWING FEATURES EXIST IN FACSIMILE MACHINE

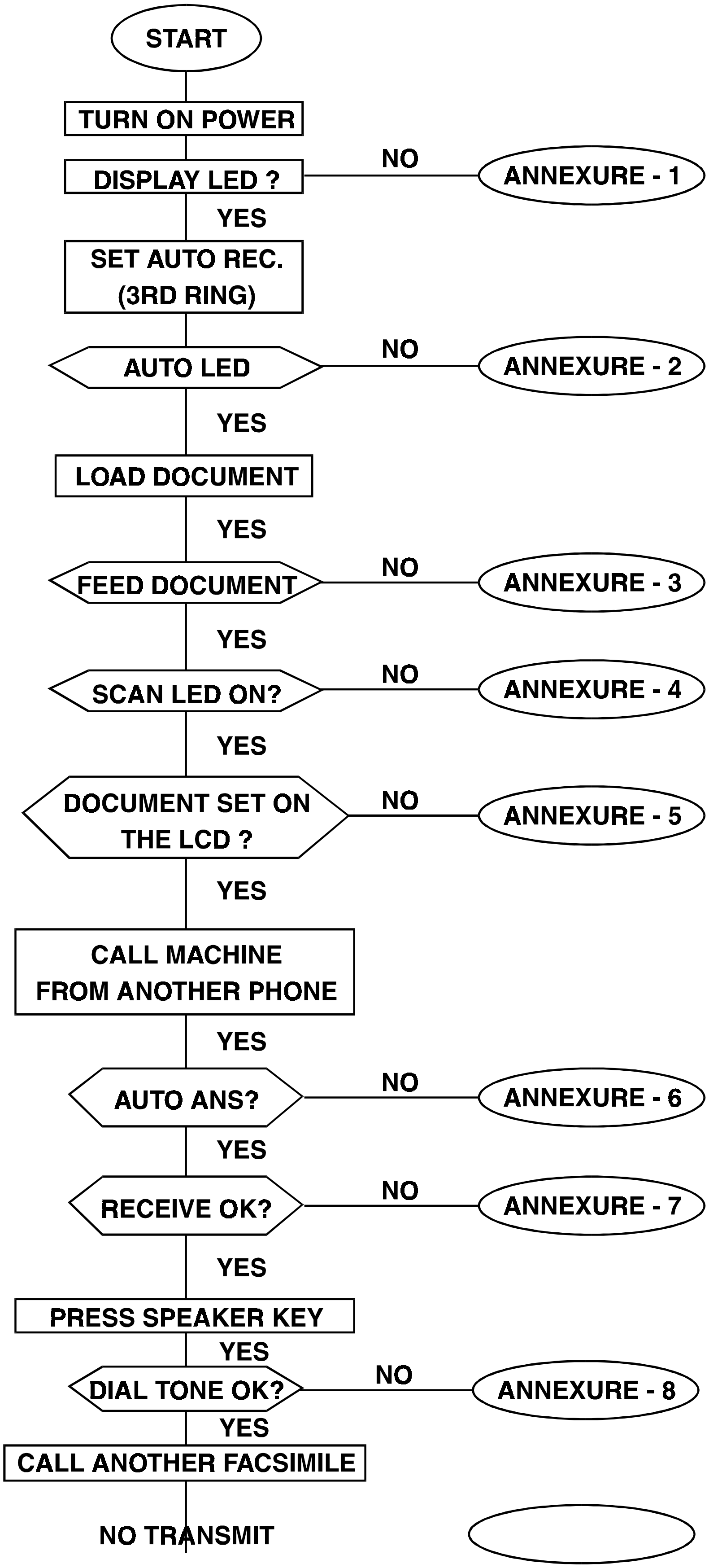
1. **Built in dial :** Enter the numbers you send faxes to as individuals and for groups you will send the same message to. This is a time saver and you'll avoid wrong numbers.
2. **One touch Dialing :** These allow items associated with each destination, such as telephone number, resolution, and transmission time, to be stored and recalled by one button. A different set of transmitting conditions can be associated with each number on the list.
3. **Speed Dialing:**
4. **Automatic Re-dialing :** The phone numbers of fax machines called can be stored in memory and used for sending fax transmissions by pressing one or two buttons.
5. **Alternate Number Dialing :** The document may be sent to an alternative fax machine is in use. When setting up the list of telephone numbers for fax machines to be called, an alternative number may be entered just after listing the regular number. Some fax machines are programmed to try the regular number one time, then if busy, try the alternative number. If the alternative number is busy, the regular number is tried again a specified number of times at a specified interval.
6. **Delayed Transmission :** A timer in the fax machines can be programmed to delay calling until the telephone rates are low or a particular person will be in the receiving office. This is particularly useful for large time zone differences where no one is at the receiving end during business hours at the sending station.
7. **Automatic Voice/Data Switch :** A received call rings the telephone if a voice call is received or the fax machine if a fax call is received. An automatic transfer switch built into the fax machine determines whether the incoming call is from a fax machine or is a voice call. If the call is from an automatic calling fax machine, the CNG beeps between ring signals are detected and the call is switched to the fax machine. The CNG tone may not be present for automatic fax calls from some PC-fax cards. For a manual call, the logic used may differ between different models. On design inserts CNG tone beeps when the sender's START button is pressed. On a manual fax call, the receiving fax machine may automatically answer after a preset number of rings. If the call is answered on the telephone, the fax sender is told to press the START button to start the receiving fax machine.
8. **Document Feeder :** Use this feature to send more than one page and you won't waste time waiting for each page to be sent. If the pages you want to send are on onion-skin paper or dog-eared, make an office copy to send and avoid the possibility of jamming your original in the fax machine. Some fax units has a five-page document feeder. To avoid paper-jam, don't exceed the limit or put in thick sheets.
9. **Dual Access**
10. **Automatic Paper Cutting :** This very desirable feature automatically cuts received pages and stacks them in the receive tray. Without a cutter, a long banner of paper spills on the floor if documents are received while the fax machine is unattended. The time saved can easily pay for the extra cost. If, however, only a few pages a day we received, a cutter is probably not needed.
11. **Multiple Paper Rolls**
12. **Copier Option**
13. **Automatic Fallback**

14. **Automatic Error Correction :** This standardized group 3 option produces error-free fax copies even though there are some errors in the received data stream. Error-correction mode (ECM) breaks the picture signal into HDLC blocks and automatically retransmits
15. **Transmit Terminal Identification**
16. **Broadcast Transmission :** Many fax machines with this feature allow documents to be scanned rapidly and stored in memory for sending sequentially to many different locations. Other fax machines require leaving the pages to be sent in the automatic document feeder. Some fax machines use special dialing cards to allow different documents to be sent to different fax machines use special dialing cards to allow different documents to be sent to different fax machines. Some units allow documents entered into memory during the day to be automatically sorted for delayed sending of all documents for one destination in one phone call. Different combinations of the same documents can be programmed for different destinations.
17. **Polling :** Use a polling code when leaving documents in the fax machine to be sent later upon a polling command from someone else. Otherwise they could be sent to the wrong party.
18. **Delayed Transmission and Timer Transmission**
19. **Contrast Control**
20. **Half Tone Transmission**
21. **RS-232 Port or Interface**
22. **Transmission Report**
23. **Communication Journal**
24. **Power Backup :** A separate power circuit for the fax machine is recommended. This prevents putting the fax out of business when the circuit breaker is tripped by overload from an electric heater or other power-hungry device. Keep the fax power turned on 24 hours, seven days a week, so it is always ready to receive. This allows use across time zones or when the fax sender is working late. Standby power is usually about 15W. A plug-in power and telephone line filter like those sold for computers is a good investment, but the power and telephone lines should be unplugged during severe thunderstorms, if possible.

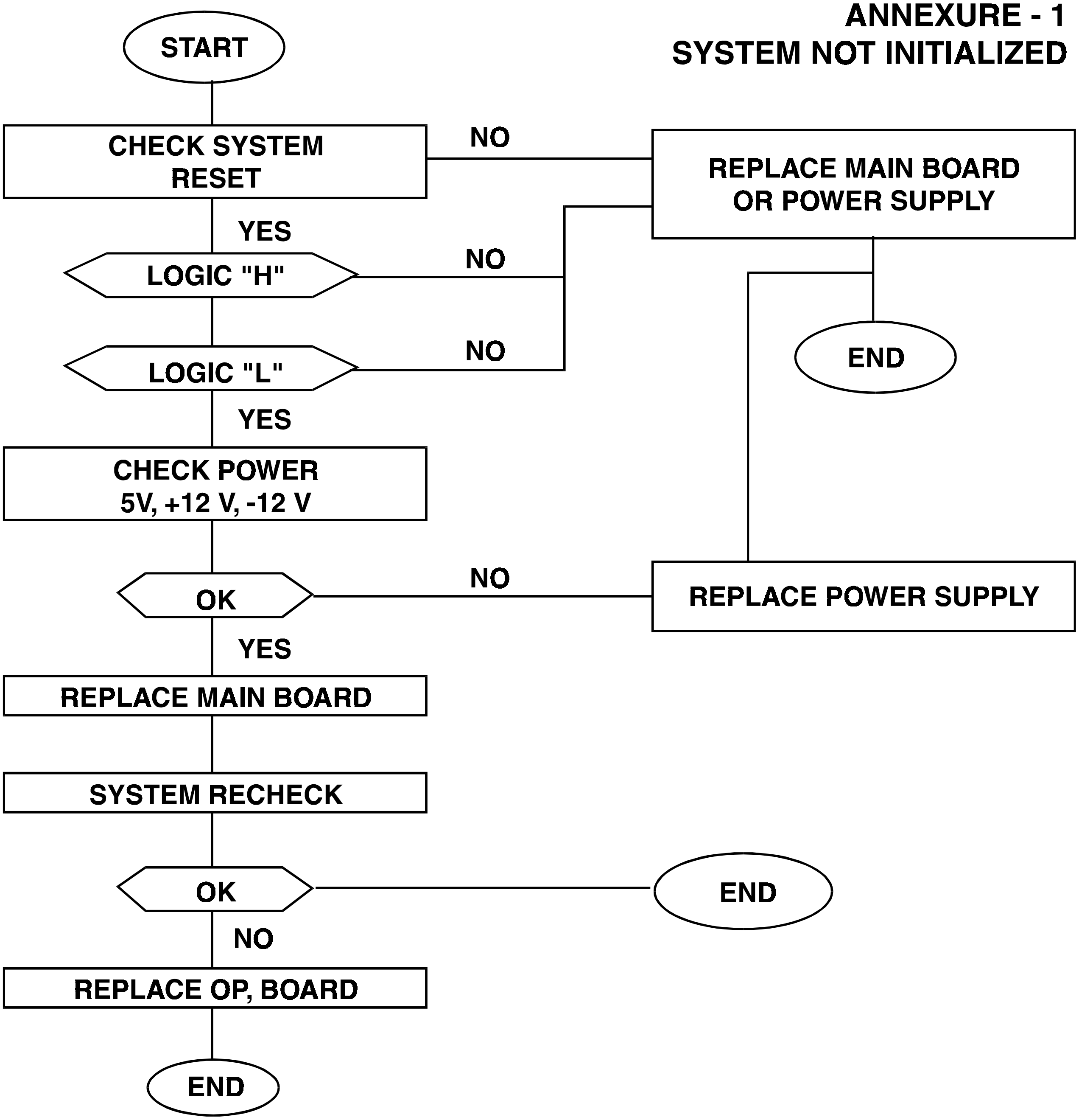
ASSIMILATION EXERCISE

Q.1 What are the steps involved in a facsimile process?
Q.2 How a FAX message is transmitted from one machine to other?
Q.3 What are the different groups of FAX specified by CCITT?
Q.4 What is the difference between the G3 &G4 FAX machine?
Q.5 Which features are commonly available in a FAX machine?

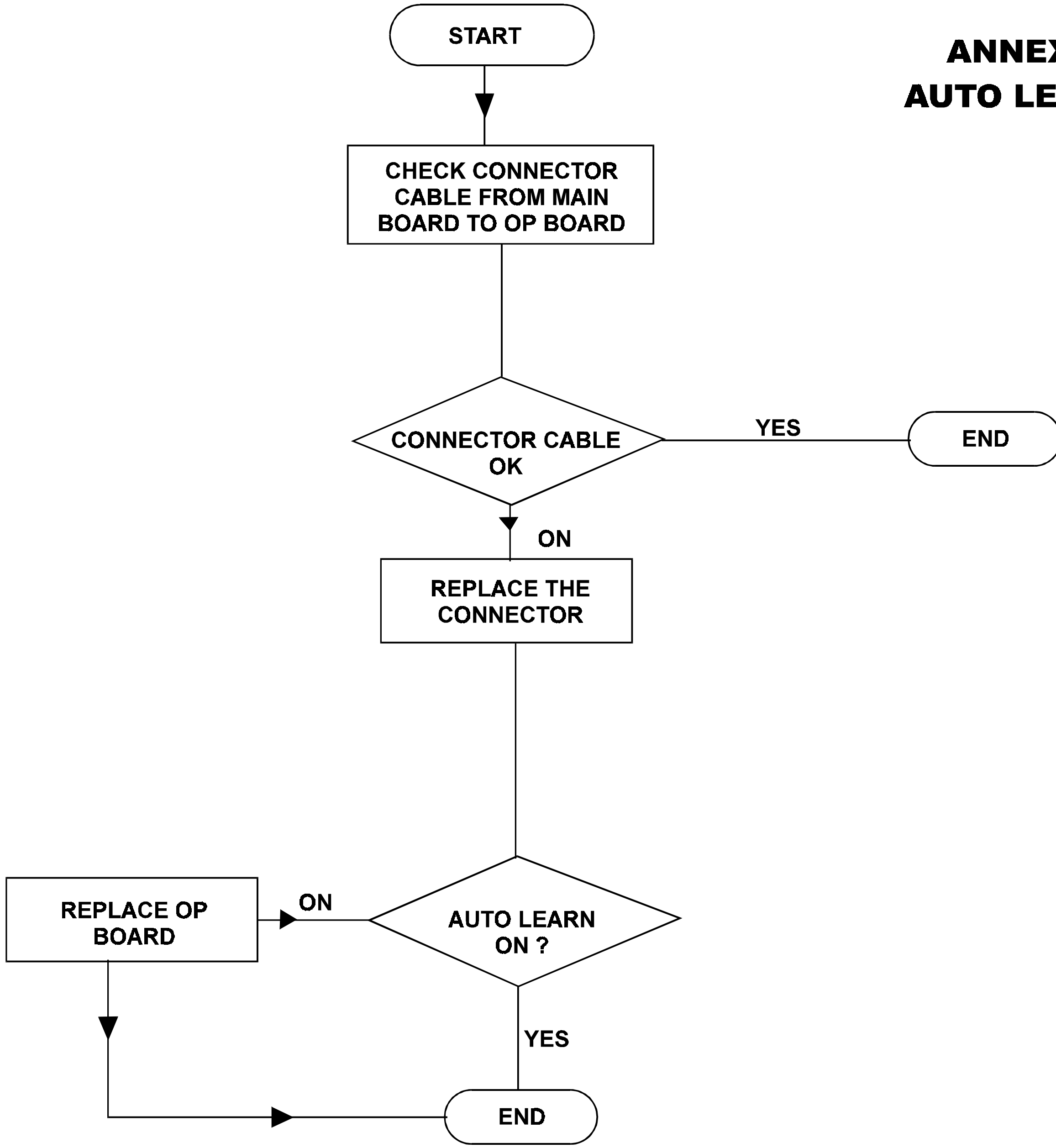
FACSIMILE TROUBLESHOOTING



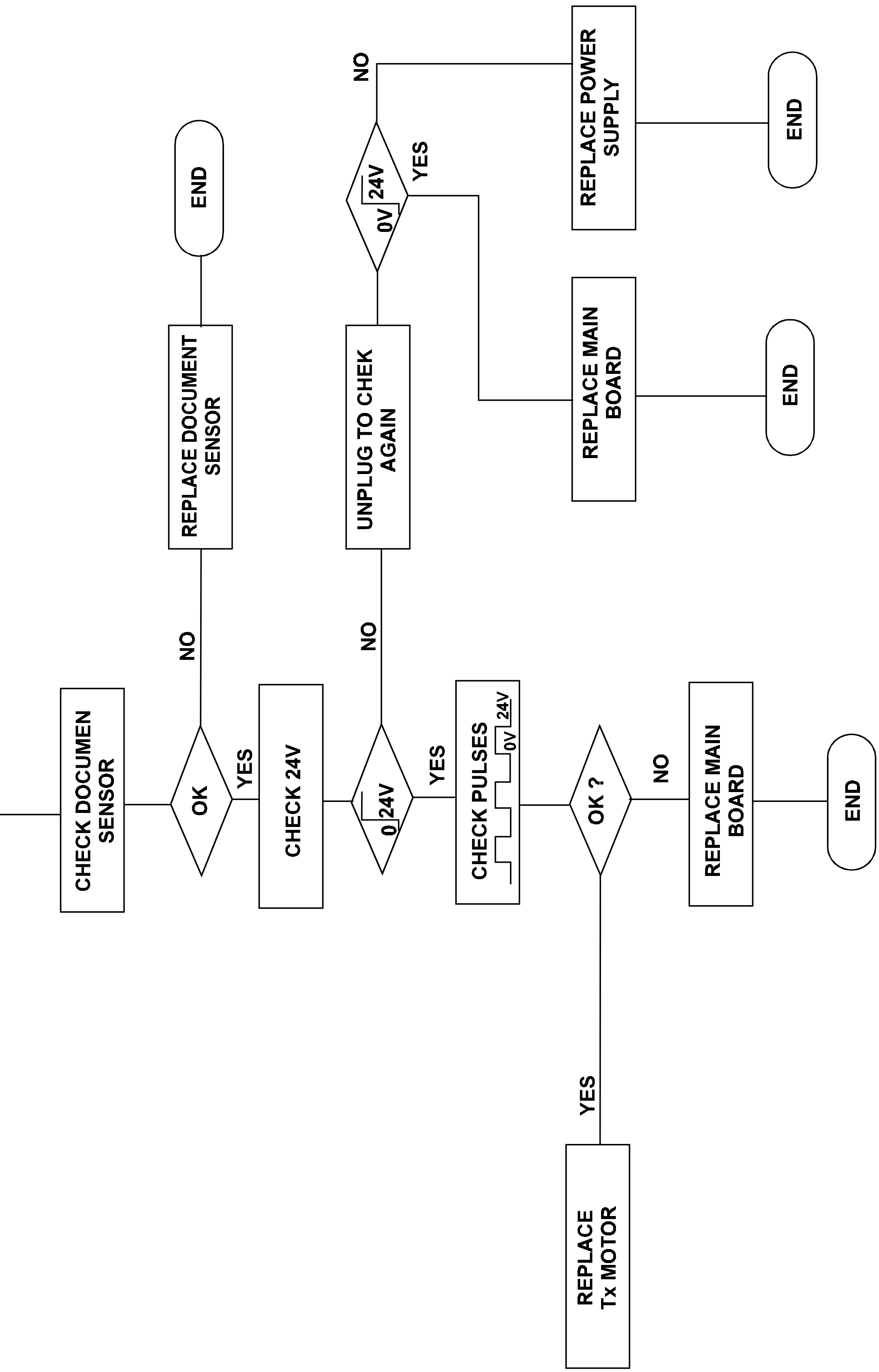
ANNEXURE - 1
SYSTEM NOT INITIALIZED



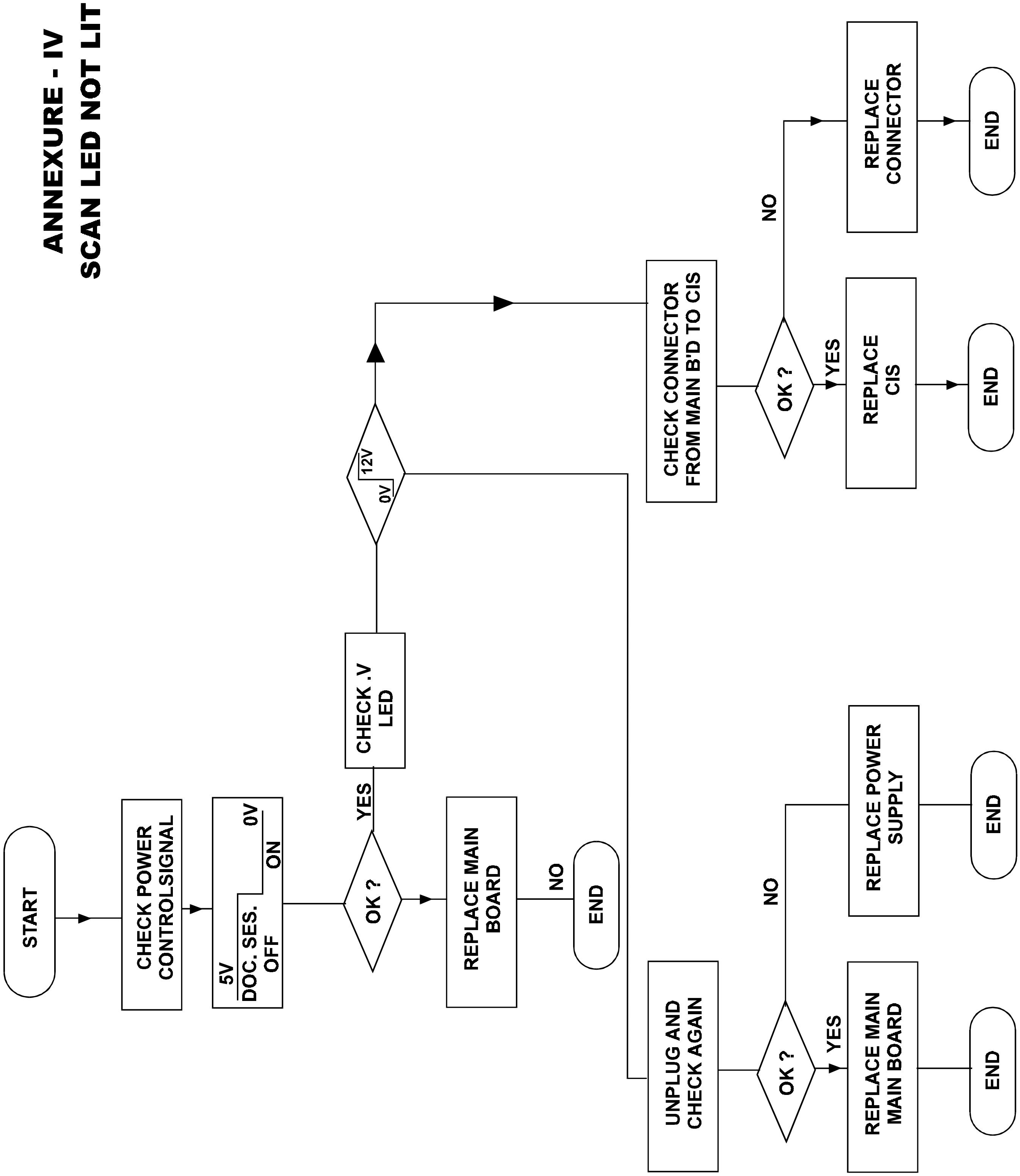
ANNEXURE -II
AUTO LED NOT LIT



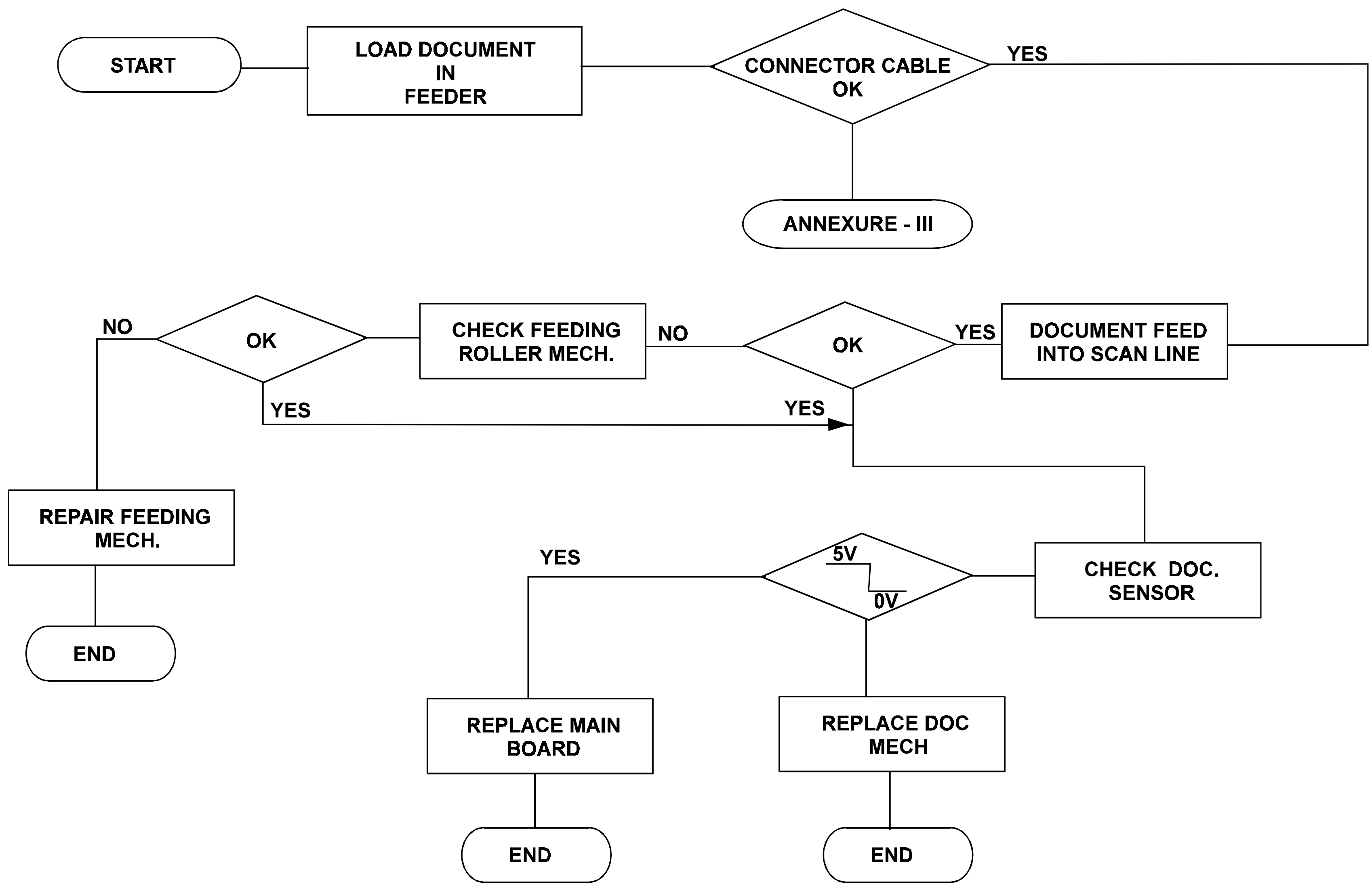
ANNEXURE - III
DOCUMENT NOT FEED



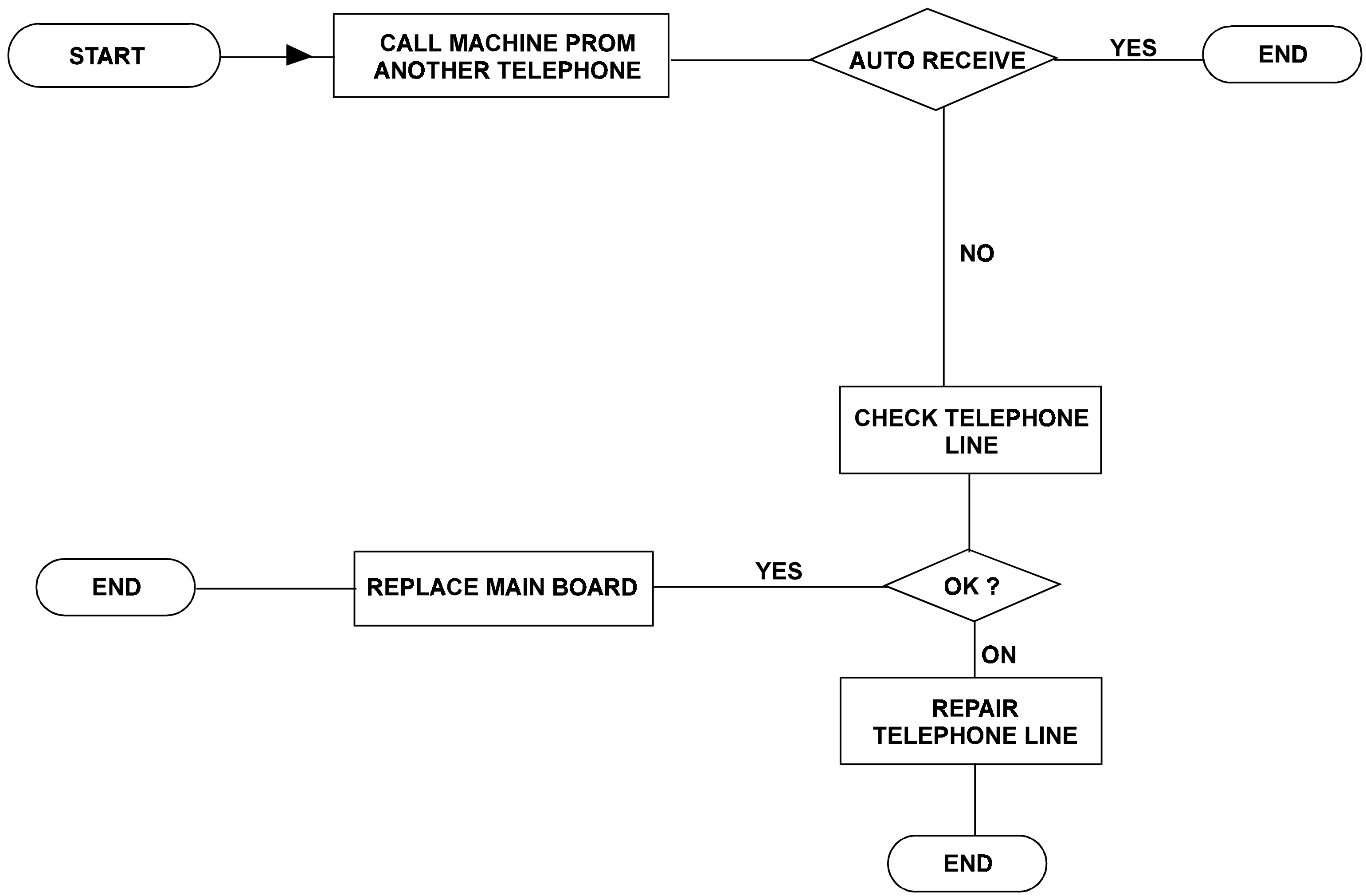
ANNEXURE - IV
SCAN LED NOT LIT



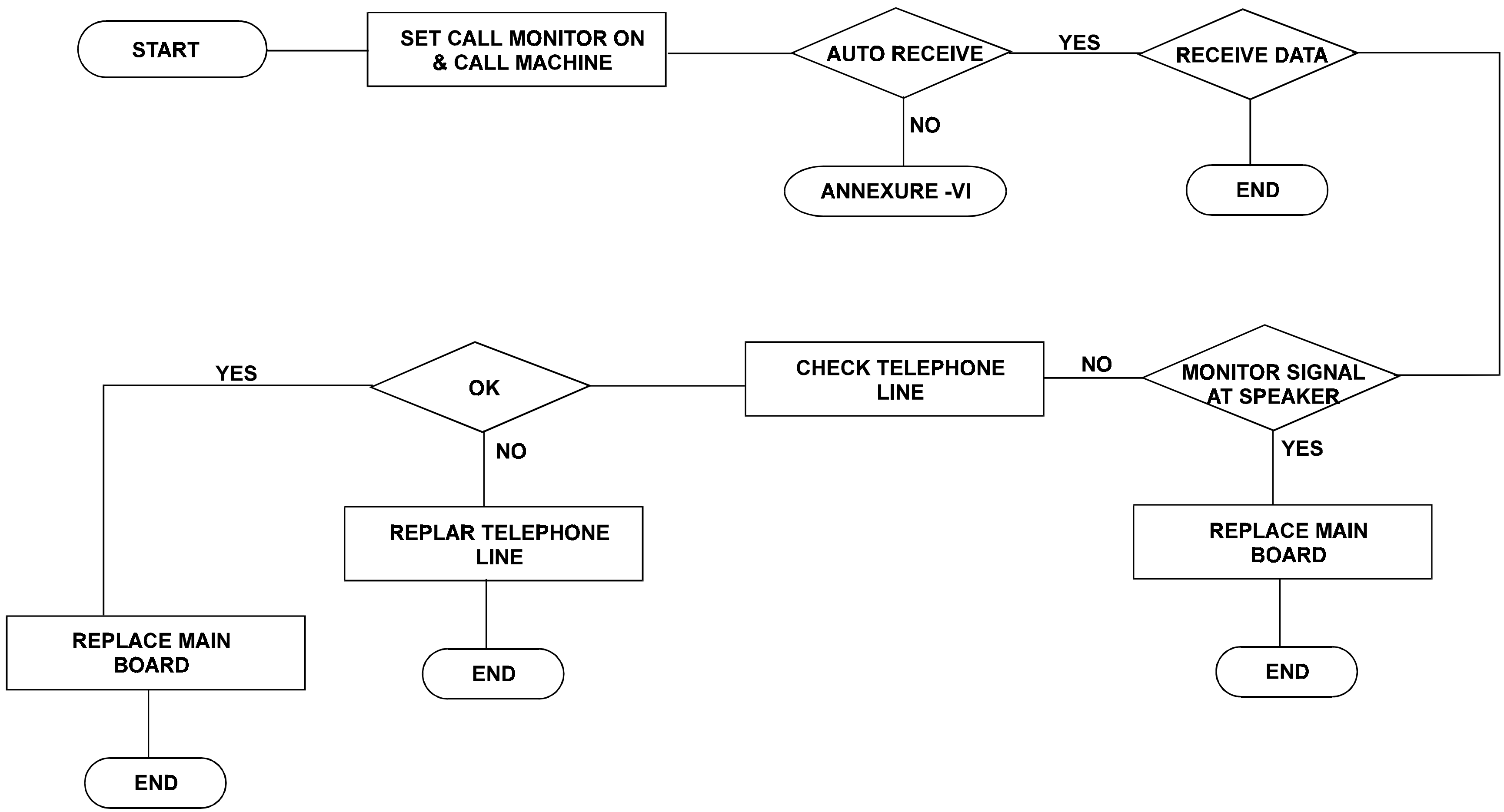
ANNEXURE V
NO INDICATION "DOCUMENT SET" ON THE LCD



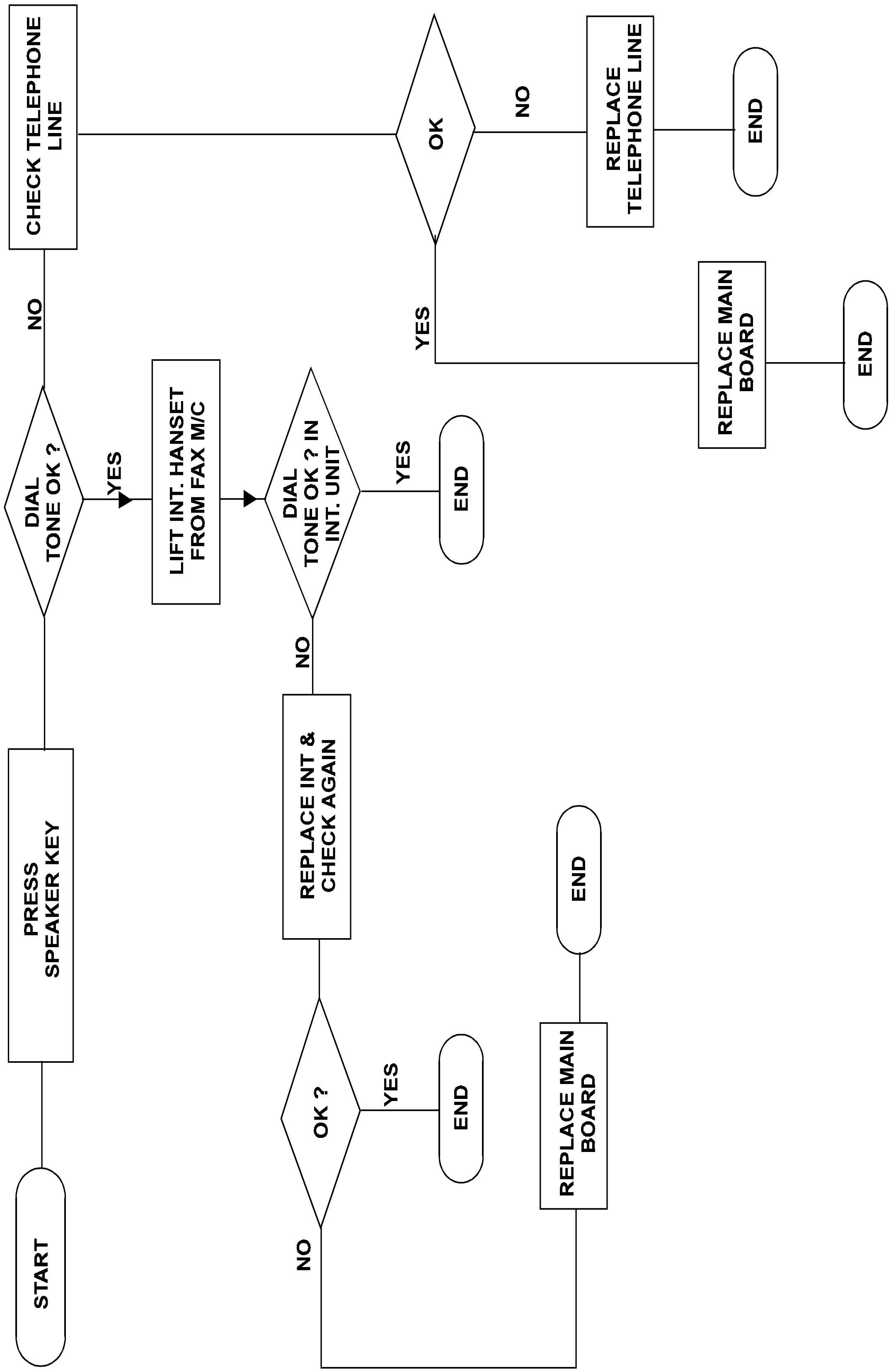
ANNEXURE VI
AUTO RECEIVE PROBLEM



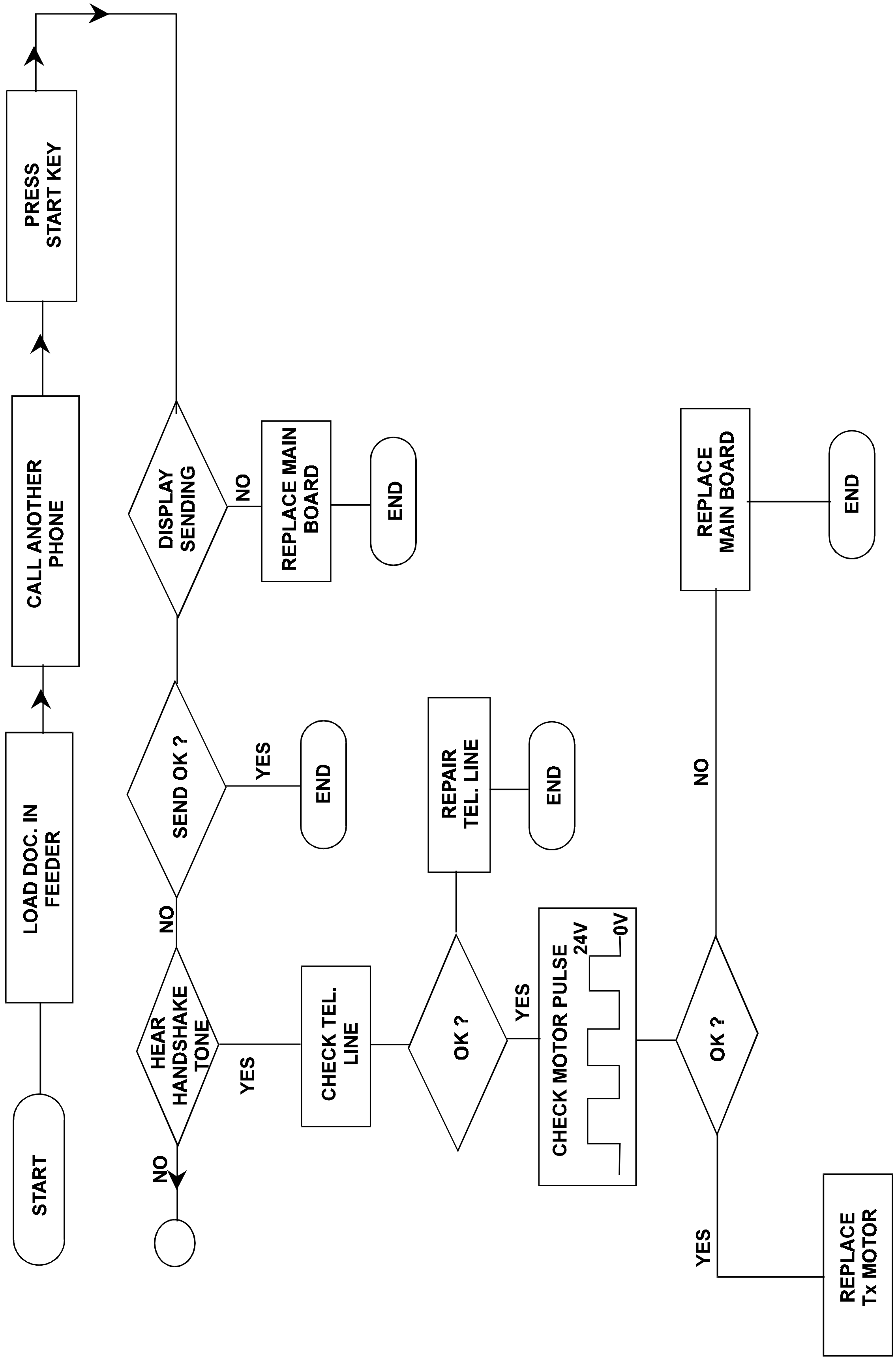
ANNEXURE VII
NO RECEIVE



ANNEXURE VIII
NO DIAL TONE



ANNEXURE-IX
NO TRANSMIT



MODEMS

COMPETENCY OBJECTIVES

The objective of this Chapter is to introduce the students to:-

- Introduction
- Digital Modulation
- Theory of Operation
- Interface
- Problems
- Troubleshooting

Chapter 2

MODEMS

INTRODUCTION

Modems derived from modulators and demodulator for digital signals, have been in use since 1920. Earlier amplitude modulation was used and later frequency shift-keying modulation. High speed modems are using phase-shift keying, quadrature phase shift keying, quadrature amplitude modulation. In United states, the first modems that used the PSTN for data transmission appeared in 1950, with the introduction of modems by AT&T and other. They were controlled by binary control signals, each control wire affected one function. These modems had the ability to go off-hook and establish connection with a distant modem and answer incoming calls automatically. The usual practice was to accomplish the dialing function of call set up by means of a separate device - a telephone or an automatic calling unit which used a separate cable to the DTE for calling control. The first intelligent modem came in 1980 by Biz Comp . The second modem (300 bps) was introduced by D, C. Hayes Associates in 1981. D.C. Hayes associates are now-a-days recognized by Hayes Microcomputer Products, Inc. Since then dozens of manufacturers have introduced intelligent modems and now-a-days more than 200 companies are manufacturing modems. The first intelligent modems used synchronous communication between DTE and modem. In 1984, Racal Vadic introduced a synchronous DTE modem command language, called synchronous Auto Dial language (SADL) which provided synchronous control of modem using Binary Synchronous Communications (BSC) Synchronous Data Link Control (SDLC) and High-level Data-link control (HDLC) protocols. The International Telegraph and Telephone Consulate Committee (CCITT) Recommendation V.25 got approved in 1984, which had limited command and response set and provision for asynchronous DTE interface and two synchronous DTE interface protocol, BSC and HDLC.

Role of Modem

Modem with software allows you to

a. Print a record incoming information to a disk.
b. Send and receive files.
c. Connect directly to other nearby computers that have modem.
d. Set up communications sessions to be executed auto at a specific time.
e. Manage a phone directory list complete with usage tracking and provision for long distance dialing codes.

So in order to take full advantage of telecommunications you need.

1. An IBM PC, XT AT or PS/2 compatible system.
2. A modem compatible with the Hayes AT command set.
3. A telephone connection/line.

TYPES OF MODEM

Internal Modem - An internal modem is an expansion card that occupies a slot in the PC. Like the other adapter card, it also gets its power from the PC. No cabling is required for the internal type of Modem. What all is needed is the phone cord to connect the modem to phone line.

External Modem - External modems are packaged in enclosures, ranging from very nice aluminium cases to inexpensive plastic ones. For an external modem you require a serial port in the PC, cabling from the port to the modem, and a power supply. Some modems have a separate power adapter that plugs into wall and has wire that plugs into mode. A few modems use an adapter that sits on the floor and has two wire, one that extends to the wall plug and one that goes to the modem.

An external modem connects your serial port **COMI** - designation for first serial port on most machines. There are also modems that are actually add-in-cards. These have serial port built in and are usually configured to connect to COM2 - which is designation of second serial port of the machine COM1- COM2 - It is important to know these designation because telecommunication program, such as PROCOMM, has to be told which COM port has the modem attached. There is no way for it to know this without asking the computer operator. When first PROCOMM is run type "PROCOMM" at the print, after which a series of questions about your modem including its location (COM1, COM2, COM3 etc.) and its speed. Its always good to have the modem documentation handy in case you have to look some thing up.

During installation modems sometimes have problems due to use of wrong cable, wrong switch setting or bad software. The most important thing is to hook the phone wire from the wall into the connector in the modem. RJ11 plugs are required otherwise go to radio shack and get the wiring necessary to modernizes your phone or call the phone company and have them do it. Some modems have two RJ11 jacks, one marked 'phoneline' and one marked 'telephone' so that separate telephone line is installed to makes calls when you aren't using your modem or monitor you modems calls. If you have utility of Borlands side kick, you can have a super memory telephone dialer.

A modem whether internal or external performs the function of converting information from your PC into a form that can be transmitted over a telephone line (Modulation).

It converts the signals from the phone line into information that is sent to your PC (Demonstration). Modems are equipped with speaker that lets you monitor the progress of a call. Volume level on some modems is set with knob, others use a command. External modems status lights to indicate what the modem is doing. Internal modems do not have light but some software packages such as RS-232, make up for this by simulating these lights with a lever screen display.

HOW A MODEM WORKS

The modem is a device to convert the ON/OFF (1/0) electrical pulse generated by a computer into audible sound, and then back to ON/OFF electrical impulses again at the other end. The digital information used by computer is converted to analog signals by the modem for transmission over ordinary telephone lines. After the analog information is transmitted it is converted back to digital. This process is called modulating/demodulating. All modems share common components like a transmitter and receiver. The transmitter modulates the digital signals to analog (tones and sounds) and the receiver demodulates the analog back to digital. When two modems communicate, they exchange continuous audible tones called carrier signals. Each carrier signal has a frequency established by the modem manufacturers or by published standard. If one modem detects the absence of a carrier signal for more than a few milliseconds, the connection is broken and the modems hangs up.

TRANSMITTING DIGITAL INFORMATION USING MODULATION

Atleast two states are required to represent digital information. These states are represented by alteration of the carrier signal to represent the binary digit 0 or 1. Changing the carrier signal is called modulation. Modulation may involve varying any of three attributes of the carrier.

(a) Amplitude - Magnitude of peak voltage level
(b) Frequency - No. of complete oscillations of the signal per unit of time.
(c) Phase - Location at which a signal crosses the zero level related to previous signal.

Modem uses different forms of modulation depending on the speed involved. For instance we have following types of Modulation:

(a) Frequency Shift Keying: FSK is used for speed below 1200 bps . FSK modulation is a two levels technique that represents changes in the binary bit pattern by changes in the frequency of an audio tone. The line is assumed to be in a steady binary 1 state when idle, represented by one particular tone frequency.

The modem changes to another tone frequency when the data bit value 0 is sent. These tone changes cause a unique musical effect during transmission. Although higher speed modems do include support for FSK transmission, the lower speed FSK frequency modulation is seldom used. Today 300bps modems are more commonly found at swap meets N flea markets. Although extremely reliable, 300 bps limits transmission to about 30 characters per seconds. Most users can read text displayed on the screen two or three times faster than this and would grow impatient waiting for a 300 bps stream.

(b) Phase Shift Keying : PSK changes the phase of a signal, that is, its timing relationship to a fixed reference, to represent changes in the bit pattern. A reference oscillator is used to measure the phase shift of the incoming signals to determine 0 or 1 bit being transmitted.

(c) Differential Phase Keying *:* DPSK is used in 1200 and 2400 bps PC modems and compares the phase angle of the incoming signal to the previously received signal. One change in phase is interpreted as a binary 0 if the preceding phase has been interpreted as binary 1 and so forth. This method does not require a separate reference wave and needs less electronic circuitry.

(d) Amplitude Modulation : AM is the simplest modulation technique. Waves of high amplitude are denoted by binary 1 and waves of low amplitude are denoted by binary 0. AM is highly susceptible to line interference and in practice is not used by itself.

(e) Quadrature Amplitude Modulation : QAM is the prevailing standard at 9600 bps and higher, and is a combination of PSK and AM. QAM changes both the amplitude (height) and the phase of the wave, allowing twice as much, information to be encoded on one wave as does phase shift keying. QAM is essentially a four phase technique that uses two signals at the same frequency but the signals are 90 out of phase with each other. For each signal four possible levels of amplitude can be applied (A1, A2, A3 and A4). By combining two signals that are 90 out of phase, 16 different conditions can be generated, each representing four bits of information. It is possible to represent 32 conditions with the two signal levels. QAM encodes more information on one wave, achieving greater throughput and resulting in fast data communications. PC users can upgrade to higher speed modems. Higher transmission speeds reduce phone call length and expense, increase computer response time, and reduce charges for on-line services.

BANDWIDTH

Bandwidth is the information carrying capacity of a transmission facility. Bandwidth define a range of frequencies, measured in hertz, that it can accommodate without significant signal degradation. The wider the modem's range of frequencies, the greater the capacity to carry data. Most modems used a 300 to 3400 Hz frequency range in the middle of the bandwidth of a telephone connection. In telecommunications, the average telephone line is not very stable in the high or low frequencies. The modem is limited to the centre of the bandwidth, which is clearest and most able to accurately reproduce the modulation. To move into higher and lower ranges of the bandwidth, modems often use sophisticated, multiple-bit encoding algorithms to squeeze as much data as possible in both directions. Multiple- bit encoding, unfortunately increases data loss during line bits (like static on the line, or a brief voice bleed-through). The goal of effective modem design is to minimize data loss while sending greater amounts of data through a communication link.

TRANSMISSION CHARACTERISTICS : bps and baud

Data rate is denoted in bits per second and it indicates the speed of information transmission. Modems operate either at fixed transmission speed, or at the speed of the sending device restricted to a specified range. Some modems are equipped to operate at any of the several different speeds. The data rate is controlled by modems switch settings, or by changes within the software. The data rate is commonly referred to as the Baud rate. Baud and data rate do not mean the same thing, yet they are often used synonymously. The difference is that the term 'bps' expresses the data signalling rate, Baud is a measurement of modulation rate. A voice line can accommodate 2400 signal changes per second (Baud). Higher speed modems encode two or more data bits in each signal change. The bps rate corresponds to the number of data bits per signal, multiplied by the baud.

Baud - signals changes per second
bps - (data bits group) x baud rate

One-bit group	Two-bit group
0	00 10
1	01 11

For two bit groups, four possible bit patterns (00,01,11) exist. In phase shift modulation, for example, a shift of 90 10 degree could be represented by 01.180 degree by 10, 270 degree by 11 no shift by 00. By transmitting four data bits into each signal a modem can support 9600 bps. Because medium-speed and high-speed modems, all, group data bits for signalling, a modem's bit rate equivalent to its baud.

LINE CONDITIONS

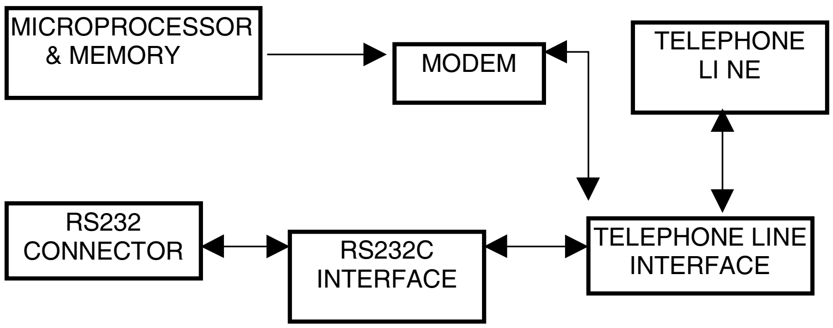
The telephone network is not a perfect transmission path, even for voice communication. Increased data speed demands ever more precise signalling methods. Problems such as line failure, electrical interference, and random noise may interrupt transmission. Modems must be able to adjust to the properties of the communications line to prevent the signal from getting changed due to propagation of low and high frequencies. This condition, called phase jitter, severely affects high-speed modems whose modulation techniques use phase shift to represent bit pattern. The newer modems now offer a useful feature called fallback. Fallback is the ability to detect poor line conditions and adjust to lower transmission speed (bps) to prevent data errors. For example, 9600 bps modems often offer fallback to 7200 bps or 4800 bps when the line conditions are unfavorable.

THEORY OF OPERATION

The modem consists of five main blocks namely

1. Telephone line interface
2. Modem

3. Microprocessor & Memory
4. RS 232C interface
5. Power Supply



TELEPHONE LINE INTERFACE

Line Protection

(a) **Impulse Surges -** The two zener diodes provide back to back protection between the tip and the ring of breakdown voltage of 30Volt each, Thus giving effective breakdown voltage of a little more than 30 volt each.

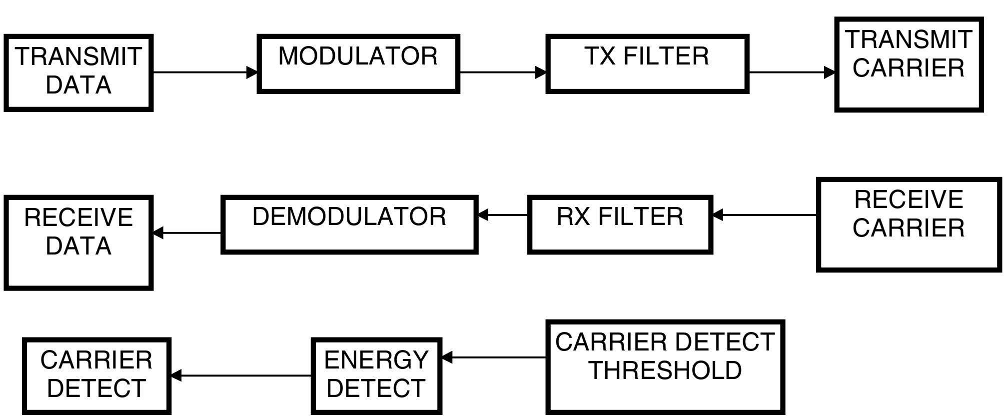
(b) **Surges over a period of time -** A gas type surge volt protector is provided between the tip and ring, having a rating of nominal DC breakdown voltage equal to 350 volt and impulse breakdown maximum voltage V/us = 750V.

(c) **Isolation -** The ring detector circuit is optionally isolated from digital circuit.

MODEM

Modem consists of six major functional blocks:

(a) Modulator
(b) Transmit filter
(c) Receive filter
(d) Demodulator
(e) Energy detect circuit
(f) Timing, control and handshake logic



MICROPROCESSOR AND MEMORY

Microprocessor control the entire operation of modem and provides all the features with respect to user communication software. The Zilog Z80 microprocessor includes all the circuits necessary to build high performance micro-computer system with virtually no other logic and a minimum number of low cost standard memory elements. The Z80 is third generation single chip microprocessor with unrivalled computational power. This increased computational power results in higher systems through put and more efficient memory utilization when compared to second generation microprocessors. In addition the Z80 is very easy to implement into a system because of single voltage requirement plus all output signals are fully divided and timed to control standard memory or peripheral circuits. The circuit is implemented using a N-channel, ion implanted, silicon gate MOS process. The various features of Z80 are

(a) Single chip, N-channel silicon gate CPU.
(b) 158 instructions-includes all 78 instructions of 8080 a microprocessor with total software compatibility. The new instructions include 4-bit, 8-bit and 16-bit operations with more useful addressing modes such as indexed, bit and relative mode.
(c) 17 internal registers.
(d) Three modes of fast interrupt response plus a non-maskable interrupt.
(e) Directly interface standard speed, static or dynamic memories with virtually no external logic.
(f) 1.0 microsecond instruction execution speed.
(g) Single 5 volt DC supply and single-phase 5 volt clock.
(h) Out-performs any other single chip microcomputer in 4-bit, 8-bit or 16-bit applications.
(i) All pins TTL compatible.
(j) Built in dynamic RAM refresh circuitry.

RS 232 INTERFACE

The Electronics Industry Association (EIA) standard Rs 232C covers the electrical specifications for bit serial handshaking signals used to control standard telephone connection equipment and standard modems. Electrically the standard uses nominal plus and minus 12 volt pulses to effect information transfer the RS232c specifies a 25 pin connector. We will not be using all the 25 plus to carry signals but only 9 of the 25 plus are actually used to carry signals.

Mnemonic	Name	Source	Purpose
TxD	Transmitted Data	PC	Carries Characters from PC to Modem
RxD	Received Data	Modem	Carries character from Modem to PC
RTC	Request to send	PC	Used for Flow control
CTS	Clear to	Modem	Used for Flow control
DSR	Data set ready	Modern	Used for Flow control
SG	Signal ground	N/A	An electrical reference point for the other signal
CD	Carrier Detect	Modem	The Modem make this signal active when it is connected to other modem. Some s/w programs monitor this signal to know when they are on line.
DTR	Data Terminal Ready	PC	Some programs make this signal active when they are about to place a call or are in local mode.
RI	Ring Indicator	Modern	Becomes active when a ring is detected. Can be important when answering calls.

25 Pin PC Port (DTE) to Modem (DCE)

Signal	PC Pin	Modem Pin
TxD	2	2
RxD	3	3
RTS	4	4
CTS	5	5
DSR	6	6
SG	7	7
CD	8	8
DTR	20	20
RI	22	22

8-Pin AT Port to a 25-Pin Modem

Signal	PC Pin	Modem Pin
TxD	3	2
RxD	2	3
RTS	7	4
CTS	8	5
DSR	6	6
SG	5	7
CD	1	8
DTR	4	20
RI	9	22

CONNECTING MODEMS TO TELEPHONE LINES

There are two ways to hook a modem to a telephone line. 1. Acoustically and 2. Directly. Acoustic coupling involves pushing a telephone and set into the set of rubber cups that hold the modem speaker and receiver. Direct connect modems connect to telephone lines by means of the familiar RJ 11 modular telephone jack. They are less sensitive to noise and are easy to connect.

CONNECTING A MODEM TO A PC

After successfully connecting your modem to the telephone systems, the next step is to make the connections between the modem and your Personal Computer Free standing or external modems are connected by either a DB-25 or DB-9 cable to the computer’s RS-232 serial communications port. Board or internal modems are connected by plugging a card into one of the computer’s expansion slots.

Two PCs connected to each other via phone line and modems cannot communicate without communications software, sometimes called terminal emulation. Some internal or board modems have communications software built in which is known as Firmware and is permanently stored in ROM chips.

THE INTERFACE MODE

There are two standard interfaces used in PCs.

1. Parallel interface
2. Serial interface

PARALLEL INTERFACE

Eight data channels in parallel can transmit an 8-bit code that represents a single piece of information. Internally, computers transfer in the parallel format using a data bus. With the parallel port, the data is presented at the interface as 8-bit channels or data lines. An additional line is also provided, called a clock or strobe line. The data is clocked or strobed out, and the receiving devices then uses the clock to synchronize itself to the parallel data being sent. Most PCs are equipped with unidirectional parallel ports capable only of sending data. Since parallel ports are intended for use with the printers, this limitation is not a problem. IBM PS/2 and some other computers have bidirectional parallel ports.

With a bidirectional ports, the same 8-lines are used for incoming or outgoing data. An interesting feature of the parallel interface is its ability to pass data at fairly high rates of speed. Unlike a serial interface with predetermined data rates, a parallel interface can transferred data at the maximum rate of the device to which it is connected.

SERIAL INTERFACE

This leads us to a discussion of the dreaded serial interface, one of the most efficient and cumbersome methods of communicating. Serial interface was nearly orphaned by the industry in the 1970s. The serial interface appears at first to be less complex than parallel interface. Data is transferred in a serialized form, that is , the eight data bits are carried over the serial channels one after the other, in a specific order. With the parallel interface requires a minimum of 11 wires, not including control signals, where as the serial interface requires only three wires and also not including control signals. This limitation is one reason why parallel interfacing is not more widely used.

PROBLEMS FACED BY MODEMS

ECHOES

For short-distance communication, a two wire line goes in and out of the phone company offices without major transformations. Irregularities inherent in transmission causing a portion of the signal energy to be reflected back toward the originating end. This is referred to as “talkers echo” when using a phone, you can hear your own voice on the receiver but at a much lower level. If phone line irregularities become prominent enough to generate loud echoes, they will interfere with data. Loud echoes are obvious, so connecting a handset to a line and listening for them is easy.

DISTORTION

Distortion occurs when the voltage loss or delay of a transmission line varies as a function of frequency. If you transmit a test signal at one end, and monitor the receive level at the other end, you will see a sharp loss when you attempt to transmit through the bad area.

OVERSEAS CALL

When you call a very long distance, coast to coast or overseas, the call may be routed via satellite. Satellites are poor data communications facilities. There are massive delays in transmission, horrible echoes, and something called the Doppler Effect. The Doppler effect is simply a reference to the fact that the satellite is in motion. During prolonged transmissions, this motion can cause the carrier to become distorted.

IMPULSE NOISE

Sporadic, low-frequency voltage spikes typically caused by older phone company equipment. There are still many parts of the country with non-state-of-the-art phone equipment. A VF line monitor or telephone handset can pick up impulse noise. It sounds like a pop, or a crackle.

INTERFERENCE

Outright interference is a real problem. It comes from many sources sneaking in at the phone company office, along the telephone line route, or from the wiring in your house. Military CW, radio stations foreign exchange lines and engine noise may interfere with data communications.

BACKGROUND NOISE

It is present in every circuit, but usually filtered to such a degree that it is rendered harmless. Background noise becomes harmful when its signal strength increases to a point where it can compete with your carrier. When you amplify your data signal, background noise is also amplified. If the data signal is weak and the phone company tries to compensate with a simple increase of the circuit, noise level will increase as well.

IMPROPER IMPEDANCE

Improper impedance will adversely affect the data communications. You cannot control impedance in most cases. More advanced modems and telephone equipment do provide such options, remember that commercial modems are far more superior to modems designed for the home PC.

RULES AND REGULATIONS

As a last note on telephone lines, it is illegal to connect an unregistered device to phoneline. Use of homemade filters, equalizers, or amplifiers is subject to severe penalties. Should you desire to connect an unregistered device to a phone line you request the phone company to instal Data Access Arrangement (DAA) or its equivalent. This device protects the phone line if your equipment does not perform as you expect.

Hence, these are the problems faced by the Modems which can be overcome by the owner of the Modem, if they are kept in mind and these problems should be given top priority while using MODEM.

MODEM TROUBLESHOOTING

If your modem doesn’t seem to work the way it should when you first hook it up, there are some things that you can check. Most problems can be traced to just a few types of snags that are fairly easy to fix.

A list of the most common problems follows as given below:

1. You forgot to cancel CALL Waiting.
2. An external modem cable connection is bad or loose.
3. The phone line is disconnected at wall or modem.
4. The Comm program is set up incorrectly.
5. The modem switches or jumper plugs are set up incorrectly.
6. The modem is competing with another device for a COM port.
7. You have the wrong type of cable.
8. Switches are set incorrectly in computer (not enough serial ports).
9. There is a problem with the computer Basic Input/Output System (BIOS).
10. The Telecom program is not compatible with modem.
11. Your modem is defective or broken.

PROCEDURES FOR DIAGNOSING MODEM PROBLEMS

Cancel Call Waiting *:* Call Waiting is a telephone company service that notifies you during a call that another caller is trying to reach you. This is a fine service most of the time. The trouble is that the Call Waiting signal consists of a brief cut off of all sound followed by a high beep. Your modem interprets this as a lost connection and hangs up. Most telephone companies that offer Call Waiting service also provide a way to cancel Call Waiting for times when you don’s want the interruption.

Check Your Cables *:* Before you tear into the machine, take a moment to check your cable and phone connections, the two simplest things to fix.

Check For A Signal: If you suspect your modem isn’t dialling out, listen for a dial tone or for your modem’s dial-out tones, if your modem has a speaker. If you hear anything when you tell the modem to dial out, the modem probably dials out alright.

Check Your Hardware Layout: If bad connections aren’t the answer and you get nothing back when you try to call one of the network numbers, step back and think for a moment. It this the first time you have used a modem with this computer? If so, it’s a good idea to make sure your system is set up the way you thought it was.

(a) External Modems: Do you have a serial port for an external modem to tie into? Most AT's, PS/2's, and brand-name compatibles come equipped with at least two serial ports, one for a mouse and one for a modem or serial printer. Many can handle up to four serial ports in all, if you set the switches correctly. Serial ports come in two styles, small ones with 7 pins designed for a mouse, and larger ones, the standard RS232C port with up to 25 pins. External modems usually connect with an RS232C serial port. If you see only one connection, it may be a parallel printer port rather than a serial port. Printers use the same size connector.

(b) Internal Modems: Internal modems, on the other hand, plug into an expansion slot inside the computer and provide their own modulator telephone jacks to connect with a standard phone cord. They don't need a separate RS232C serial port because the modem itself becomes a new serial port.

Most internal modems these days come set as COM2. If your computer already has two or more serial ports it may get confused when you add a third port. Reset the modem to operate as COM3 or COM4 if it can do it. Some modems will only operate on COM1 or COM2. This means you may have to remove or disable one of your serial ports to make the modem work. One way to handle this is to unplug the device plugged into the serial port and then plug in the modem temporarily.

Use The Right Phone Jack : Most modems accept standard RJ11-series phone jacks. All modems designed for use in the United States accept a standard RJ11 telephone line jack. The RJ11 is the little plastic clip on the end of the wire that you plug into the phone, the wall, telephone jacks, and wall boxes. An RJ11 has at least two wires and usually has four. It is designed to accommodate two separate phone lines but some manufacturers only put in two wires, on the theory that most people only have a single phone line. Some modems require all four wires, so check your RJ11 plug.

Locate Existing Serial Ports: If loose or improper cables or connectors aren't the problem, the next thing to do is figure out how many serial ports your machine has. Copy a short file to various serial ports such as (COM1, COM2, COM3, COM4). If the computer considers the destination to be a serial port, DOS will issue an error statement or just sit there doing nothing. When the computer thinks it has run out of ports, however, DOS stops giving error messages. Instead it creates a file named COMn (where n is the number where the machine ran out of ports).

To install an internal modem on our sample machine, we can turn off the existing COM2 and replace it with the modem. We can also reset the modem to function as COM3 and plug it in to see if the host computer allows it.

Use The Right Modem Cable : If your modem cable came with your external modem, chances are it's the right cable. Standard RS-232 modem cables look much like printer cables and null modem cables, but there are ways to tell them apart. It's easy to tell a printer cable from a modem cable. The printer cable has ends that are not mirror-image male and female versions of each other. The ends of a modem cable will plug into each other and the ends of printer cable won't.

Check Your Software Settings: If you seem to be fine in hardware department, the next place to look is your Comm program. Specifically, you need to match your Comm program settings to the computer you're trying to call and to correct COM port number. You also need to make sure the number you're calling is expecting the call-that the modem on the other end of the line is set to answer calls.

COM PORT NUMBER - If your modem is set up as COM2 and your Comm program is sending your data to COM1, the signal won't ever get out to your modem. Make sure your software is set to the same COM port as your modem. Let's say your Comm program is set to 2400bps while the answering modem is set to 1200bps. You'll hear what sounds like the correct "fuzzy" answering tone but your modem won't respond. Changing the setting to 1200bps will solve the problem.

If someone needs to call you by modem but your phone won't go any further. Check your modem to set to Auto Answer, and then use your Comm program to send the instructions to the modem. On modems that use the AT command set, enter AT SO=1 than hit ENTER to make your modem pick up the phone on the first ring. It you need to call another modem that can't Auto Answer the phone, you may still be able to establish contact. If your modem can communicate solely in the answer mode carrier frequencies, someone on the other end can simply tell their system to go on line after you call. For modems that use the Hayes command set, add the letter R after the phone number. For example if the phone number is 671259 then we feed the number as "ATDT671259R". This may solve your problem.

Finally if you are on a 'True Blue' 4.77 MHz machine and using a Comm program written in BASIC, you can run into trouble at communication speeds greater than 300bps. You may need to change to a compiled BASIC program or to one written in a faster language, such as C, in order to communicate successfully.

ASSIMILATION EXERCISE

Q.1 What are the various types of Modem?

Q.2 Define bandwidth.

Q.3 List the blocks of a modem.

Q.4 What is the procedure of connecting a modem to a PC?

Q.5 Define synchronous & asynchronous communication.

ELECTRONIC PRIVATE AUTOMATIC BRANCH EXCHANGE

COMPETENCY OBJECTIVES

The objective of this Chapter is to introduce the students to:-

- ❖ Introduction
- ❖ Types of EPABX
- ❖ Architecture and Interface
- ❖ History
- ❖ Circuit Details
- ❖ Features
- ❖ Troubleshooting

Chapter 3

ELECTRONIC PRIVATE AUTOMATIC BRANCH EXCHANGE

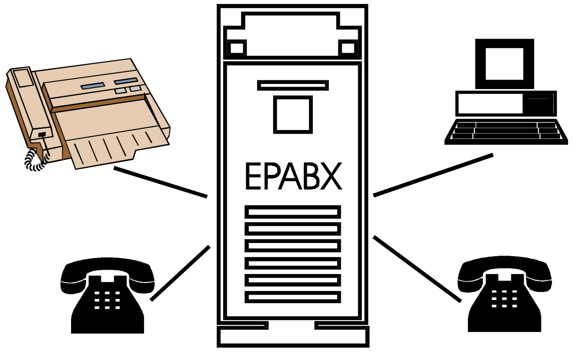
INTRODUCTION

Modern business requires effective communication facilities. The communication requirements of a small business can be met by a single telephone. On the other hand, larger business requires a private automatic branch exchange to provide connections between a large number of extensionsand the PSTN.

EPABX stands for Electronic Private Automatic Branch exchange. It is a circuit switch that serves a community of stations.EPABX switches the telephone calls within a building which can be an office, hotel, hospitals etc.It basically performs the following two functions.

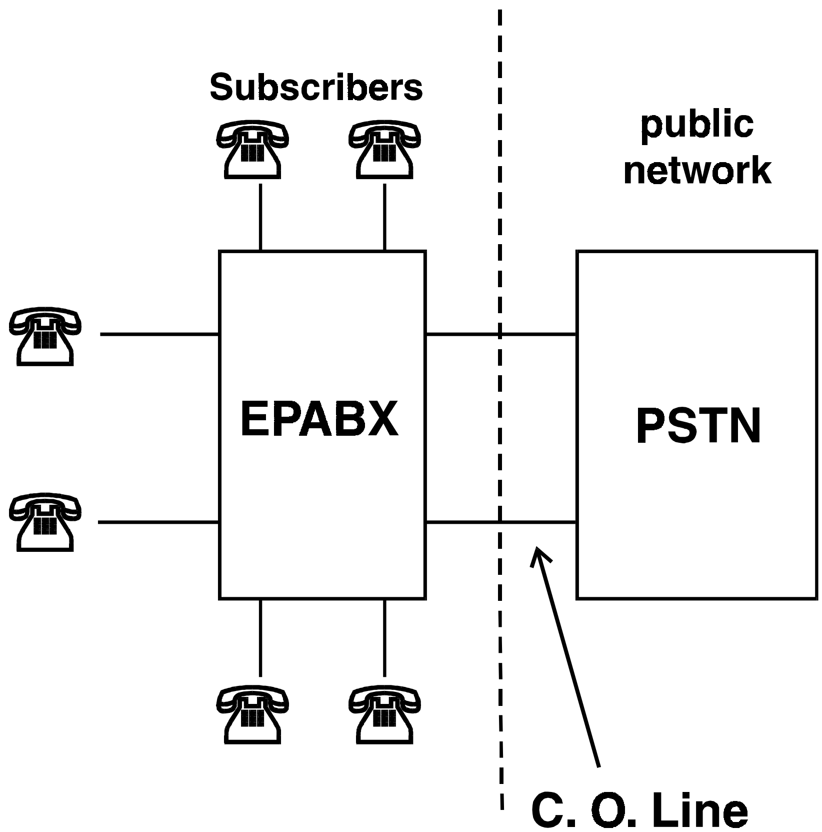
i) The internal communication within the subscriber's own organisation.
ii) The external communication with the PSTN in a time sharing mode among the users of the private switch.

When a call comes into the EPABX from an outside trunk line, it is either routed directly to a telephone extension or to a console. The phone/console rings which is answered by an attendant. The incoming call is connected to the required destination extension by dialing the number.



EPABX is capable of producing line voltages and DTMF (Dual Tone Multi Frequency) - dial tone - that can support single line type equipment which can be the telephone in your home, fax machines, modems etc as shown in the figure.

Worldwide, EPABXs are manufactured by many companies. PBXs range in size from as few as 10s lines to 1000s of lines.They are very similar to public exchanges except that they include only a few of the operational and network management functions.on the other hand, it offers lot more features which are not available in the PSTN. Before we get into the details of the EPABX, let us first define few telecom terms which will be most commonly used in this chapter.



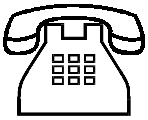
PSTN : Public Switched Telephone Network (PSTN) is the telephone subscribers network which is connected to automatic public telephone exchanges interconnected by transmission circuits.

A Central Office (CO) is the facility to which the telephones in a public telephone network are connected. It is the gateway to the rest of the telephone system; dial tone, telephone ringing, connection to other telephones, long distance carriers or outside trunks, is done here.

C.O.Line : Central Office Line (C.O.Line) connects a subscriber to the Local Public exchange or popularly known as DOT (department of Telecommunication)excange .

Off Hook : Off Hook is a state when the receiver/handset of the telephone instrument is off the cradle/hook in a busy position.

On Hook : On Hook is a state when the receiver /handset of the telephone instrument is on the cradle in the idle position.



DTMF : Dual Tone Multi-Frequency (DTMF) is used in dialling a telephone number in tone mode. Bursts of a combination of two frequencies out of eight is generated when a button is pressed to enter a digit.

Subscriber : Subscriber is a user of the services offered by the local exchange to which it is connected directly e.g status supervision, call billing, send signalling tones etc.

Signalling : Signalling can be defined as the communication that takes place between the exchange and the subscribers required to setup the call between the two subscribers.

Switch : A “switch” is a general term referring to facilities where telephone traffic is routed from one destination to another.

EPABX STANDARDS

The EPABX is a Private Exchange and certain variations exist in the terminology and other technical parameters from EPABX to EPABX. But as far as the interface of EPABX to PSTN is concerned, it has to confirm certain regulated standards. The telecom standards are internationally goverened by ITU. These standards are modified when implemented in different countries depending on their structure. In India DoT sets the standards, which the telecom equipment to be installed has to confirm.

ITU is the International Telecommunication Union, the Geneva-based United Nations agency dealing with international telecommunications standards. CCITT (the French acronym for the International Telegraph and Telephone Consultative Committee) is the former telecommunications standards body of the ITU. CCITT is now known as the ITU Telecommunication Standardization Sector (ITU-T) effective 1 March 1993.

HISTORY OF EPABX

EPABX has evolved through a number of distinct technological, generations. First generation was termed PBX which was completely manual. The attendant manually connected the call by physically inserting a cord into the socket of that particular extension. The operator continuously scans the board for the appearance of new calls and plugs into the calling line with a cord circuit, throwing the cord circuit speech key to speak to the caller. After the operator receives the caller's verbal instructions, the operator sets up the wanted connection by plugging the other end of the cord circuit into the wanted line or trunk and passing on instructions . Finally, ringing is applied to the called line and on answer, the operator retires from the connection. But monitors the indicators for a release signal in order to unplug the connection and make out charge bill for the call.

Then came PABX, which were entirely electromechanical, in which the switching takes place using the electromechanical relays.As the name signifies PABXs are automatic but switching takes place using the electromechanical relays. With the introduction of microprocessors/ computers, Stored Program Control (SPC) exchanges presently known as EPABX replaced PABXs. EPABXs are completely automatic and switching is electronically implemented.

SPC

Stored Program Control means the telecommunication system controlled by the computer or the microprocessor. SPC based systems provide a high system reliability and it supports the necessary processing within the system to meet the need of the relevant networks applications. The exchange is controlled, monitored and programmed through the programs stored in the system memory.

The EPABX's switching is either based on space division switching or time division switching.

In space division switching a dedicated path is established between calling and the called subscriber for the entire duration of the call. In time division switching sampled values of speech signals are transferred at fixed intervals. The sampling of the voice signals is based on the nyquist sampling theory this show that to retain the integrity of the input signal the sampling rate must be at twice the frequency of the highest frequency signal to be transmitted for speech telephony, whose bandwidth is 300Hz to 3400Hz a sampling rate of 8khz is therefore adequate. This represents a sample of the same channel once every 125us.

SPACE SWITCH

It has a matrix of crosspoints, which are the electronic gates. These gates are activated / deactivated by a stored, decoded signal. This is decoded as per the address select bits send externally by the control unit. The space switching implies switching between subscriber lines distributed in space and is conceptually same as crossbar switching, except that switching crosspoints are now controlled through electronic gates rather than electromechanical controls due to bar movements.

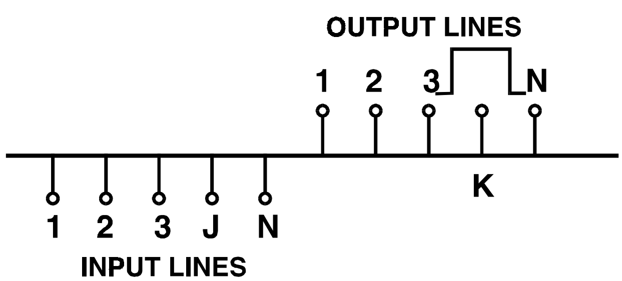
The simplest form of space switching is activation of NXN cross-point array. Such a configuration is called non-blocking because this call is not blocked if the output line is free. The number of crosspoints required are NxN. Now the lines to be switched can be connected in single stage crosspoint array on multistage crosspoint array. If we compare the two in multistage, there is significant saving in number of crosspoints required to connect N number of input & N output lines.

In multistage switching multiple cross-points participate as compared to single cross-point in the single stage switch. This may degrade the quality of transmission throughout the switch. The time to establish call also increases. Also same cross-point can be used to establish connection between a number of input/output lines. Hence the cross-points are used more efficiently.

TIME-DIVISION MULTIPLEXING

It is well known that the same channel can be used to transmit many signals using time-division multiplexing (TDM) i.e. several signals are separated in time. The same concept is carried over to switching.TDM is a technique used for transmitting several message signals over a communication channel by dividing the time frame into slots, one slot for each message signal.

There are two basic modes of operation for TDM - one that repeatedly assigns a portion of the transmission capacity as and when needed. The former mode is known as synchronous TDM and the latter as asynchronous TDM.



All the N input and out lines are attached to the bus. Suppose the input line J is to be connected to the output line K. Then a time gate opens for lines J & K simultaneously and other lines are inhibited to use that time slot. Thus during that time slot only J and K are connected. Thus by opening time gates for different input-output line pair connections can be implemented, although separated in time. The information in conversation is not lost during the interval the slot is not available to a particular input/output pairs if voice samples (or the corresponding group of digits) occur at Nyquist rate or higher and time slots are available to all successive samples (or the corresponding group of digits). This can be further explained by the following illustration.

Any single channel system can be shared by several communicators, each with his own message, by allotting certain times to each communicator. The overall system is illustrated in Fig. A switch in the transmitter is synchronized to a similar switch in the receiver. The switches start in position 1 and stay there for a period of t seconds, during which time Source 1 is transmitting and Destination 1 is receiving the information. The switches go to position 2 for another period of t seconds, during which time Source 2 and Destination 2 are communicating. The switches go to positions 3 for t seconds, connecting Source 3 to Destination 3. The switches repeat the sequence starting at position 1 allowing the three sets of communicators to time-share the same bandwidth, the overall bandwidth of the system is the same regardless of how many pairs share the channel. The standard type of coommunication systems can be used from the source switch inputs to the destination switch outputs. But each communicator must wait his turn, which in the case of outputs. But each communicator must wait his turn, which in the case of figure would mean a wait of 3t seconds between the start of transmissions. Usually this delay is not a problem except in systems requiring very high speed transfer of information.

TYPES OF EPABX

Today in the market we find a wide range of EPABXs starting from the minimum capacity of four extensions and offering to the maximum of multiples of thousand lines. The total range of EPABX is segmented on the basis of technology which is broadly covered under three standard switching systems explained below. By EPABX, we mean automatic and these automatic exchanges are SPC microprocessor based systems. The technology mainly differs in terms of switching technology which is very clearly classified below:

1. Space Division Switching
2. Time Division Switching
3. Space Time Division Switching

The higher end exchanges are basically based on space time division multiplexing in order to have a non blocking system. The switches based on space division switching are popular as **analogue exchanges**. Whereas the switches based on time division switching are termed as **digital exchanges** based on nyquist sampling theory.

Comparison of Digital and Analogue EPABX

	Analogue	**Digital**
Technology	SDM	PCM/TDM
Cost	Proves to be economical for smaller range and vice-versa	economical for higher range and vice-versa
Quality	SNR relatively low	SNR is high

DIGITAL EPABX

In an Analogue EPABX the human voice, as we all know is analogue in nature, is directly switched through the SDM switch to another extension when the speech path has been established. In the case of a Digital EPABX, the human voice is first converted into digital in the SLIC and Trunk Line Interface Circuit portion of an EPABX using a speech CODEC (PCM based) and then it is switched using TDM to another extension.

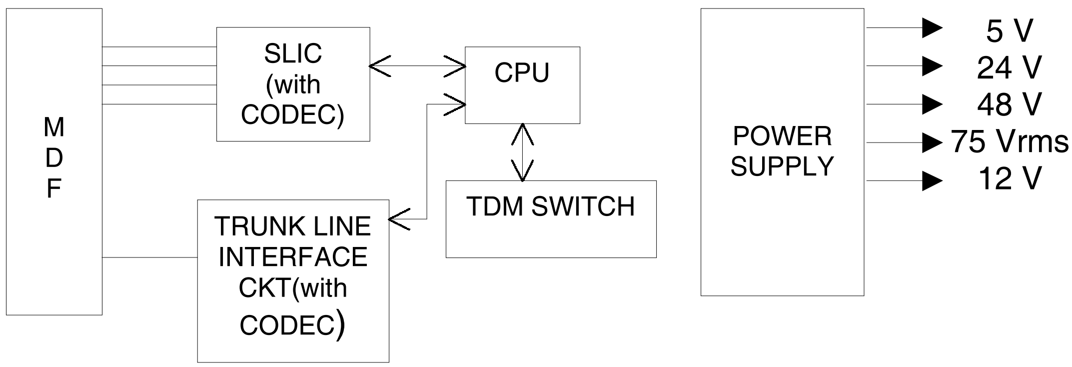


Fig.: Block diagram of Digital EPABX

The switching techniques can be classified in two ways-one according to the nature of the signal i.e. analog and digital and the other according to the distribution technique i.e. space or time. When the signals in and out of exchange are analog then the switching is analog electronic switching and when the in and out signals are digital then the switching is digital electronic switching. However, it is possible that the switching is digital and in and out signals are analog. In the above case analog signals are converted to digital signals before switching and back to analog after switching. The space switching implies switching between subscriber lines distributed in space.

Boradly, we can say, in Digital EPABX SLIC is performing an additional function of converting analog speech to digital Strain. Also the switch is based on TDM where the digital speech is time switched. However, time switching is a little involve process and requires storage and retrieval of input & output signals in the desired order timewise. Subscriber lines are terminated in line cards and one of the functions of these cards is to convert analog signals to digital signals.

PULSE CODE MODULATION

Pulse Code Modulation is used to obtain a time-divided sample of an analog signal**,** encode each sample into digital fom, and transmit a digital bit stream representing the numeric values of the series of encoded samples.The process is reversed at the receiving end : the signal is demultiplexed , decoded and integrated to regain the analog signal .

PCM involves the following three steps :

i) **Sampling :** Sampling is the first stage in the conversion of analogue signal to digital. The sampling of the analogue signal is done as per the sampling theorem.The sampling theorem (Nyquist theorem) states that, in order to reconstruct the original signal with minimum distortion , the sampling rate should be minimum twice the bandwidth or maximum signal frequency. For speech applications the sampling rate should be minimum 8000 samples per second.The audio frequency bandwidth for telephony purposes is 4KHz and it can be seen that the sampling rate is twice the bandwidth.

ii) **Quantisation :** It would be difficult and expensive in signalling bandwidth to send all the discrete sample values.By a process called quantisation ,the level actually sent at any sampling time is the nearest standard level. If the signal amplitude at the sampling time is 6.8V, it is not sent as 6.8 V,but simply as 7V because 7V is the standard amplitude nearest to 6.8 V.

iii) **Coding :** The quantised signal values are converted into a binary number which is known as coding.The sampled and quantised value 7V is converted into binary 0111. If the sampling is as per the nyquist rate there is not much difference from the original signal.

ANALOG OR DIGITAL?

This is a question of cost-effective solution rather than technology. Systems up to capacity of 70 ports can be designed very efficiently using analog switching scheme.

Beyond 100 ports, analog switching becomes unwieldy and digital switching makes more sense. That is why all over the world, small capacity EPABXes are based on analog and medium-high capacity systems employ digital scheme. Almost all the facilities and features supported by both types are identical as they are software-based and not dependent on switching scheme.

CONFIGURATION OF AN EPABX

The configuration of an EPABX defines the connecing capacity in terms of the maximum number of extensions and the C.O lines. The lower end EPABX configuration is normally given by the model number , for example COX308 tells it has a maximum capacity of 3 trunk lines and 8 extension lines and COX is abbreviated model name given to a series of EPABXs of a particular company.Whereas for higher end EPABXs altogether different terminology is used. In this case the configuration is represented in terms of the number of ports. Ports are the total number of extension lines, C.O lines, Signalling tones(DTMF, dial tone etc.) and other miscellaneous hardware input/output devices with in which the switching is required to be done.

ARCHITECTURE AND INTERFACE OF AN EPABX

Generally an EPABX is constitued of five major blocks as listed below

i) SLIC - Subscriber Line Interface Circuit

ii) C.O Line interface circuit

iii) Switch and Tone Generation Unit (TGU)

iv) CPU

v) Main Distribution Frame (MDF)

vi) Power Supply

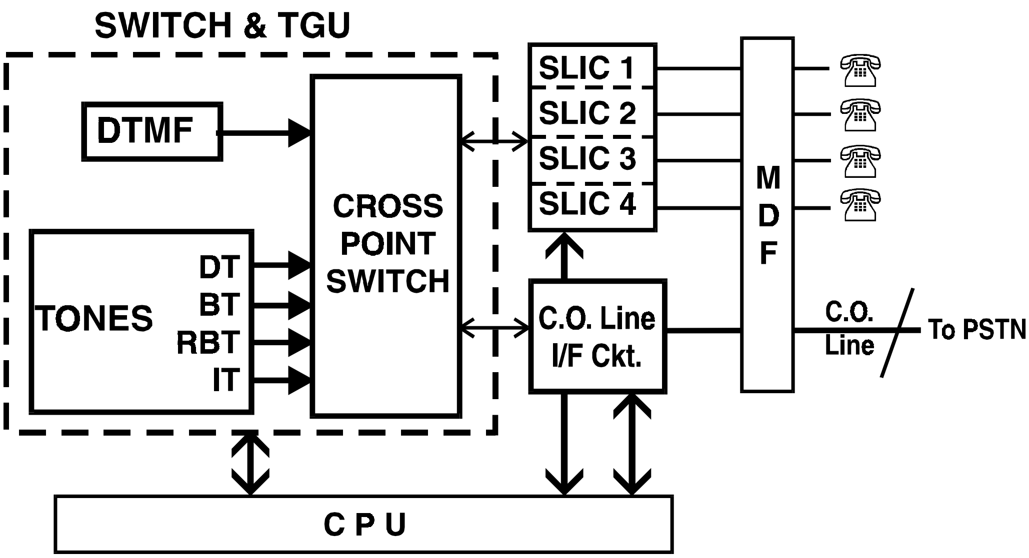


Fig.: Analog EPABX Block Diagram
(4-Extension lines and 1 C.O. Line)

SLIC (**S**ubscriber **L**ine **I**nterface **C**ircuit) : The subscriber is directly interfaced to EPABX through this circuit . There will be as many circuits as the number of subscriber lines in an EPABX. In 308 configuration,there will be three SLICs. Signalling and voice transmission on the subscriber lines requires that the exchange performs a set of functions.These functions are performed by this circuit.The complete set of functions are as follows :

B - Battery feed circuit is a constant current source and provides DC power to the telephone instrument.

This circuit is basically a constant current source with maximum current 60mA. As and when the subscriber goes off Hook,the circuit gets complete which results in the flow of current which is known as loop current. The two lines on which the telephone instrument is connected is known by the terms Tip (T) and Ring (R). Tip (T) is usually positive charge with respect to the Ring (R). Ring is typically at -48 Volts (subject to voltage losses). Tip (T) is then at ground when no current is flowing. The actual voltages may differ in PBX/Key system situations (where 24 volt systems can be found) or higher voltages can be used for situations where there are long distances among the subscribers and the EPABX. Two wires normally suffice to complete a connection between a telephone and the EPABX ; any extra wiring would be for purposes such as grounding.

O - Overvoltage protection protects the circuits from the spikes caused by atmospheric noise e.g lighting and other causes.The copper wire is like an antenna which can pick up noise and effect the circuitry. The EPABX is protected from these high voltage signals by using Varistor, Zener diodes. This is highly significant in the case when the subscriber is at a distance and is located outside the EPABX building premises.

R - Ringer relay which connects the ringer signal from the power supply to the subscriber instrument . This ringer relay is activated by the CPU and is physically present in the SLIC .The ringer signal is 75 V,25 Hz which is normally generated in the power supply section of an EPABX.

S - Supervisory features such as hook status is performed by SLIC.The Off hook and On hook status is sensed by this circuit when the loop current flows through it. Practically 15 mA and above loop current is taken as off hook status. At the detection of loop current , the SLIC reports the off hook status of that particullar subscriber to the CPU.

H - Hybrid, Isolating the voice signals from the SLIC to switching unit and from switching to SLIC is passed through the Hybrid transformer.It is doing the four/two wire conversion**.**Amplifiers are most commonly used in the communication system to compensate for the attenuation of a transmission path. Most of the amplifiers are unidirectional, hence it is required that there is a separate path for transmit and receive. This will result in the ‘four-wire circuit’ in a system. At the end of the communication system (EPABX), for transmission this four wire circuit is required to be connected to two wires. If connected directly, it will result in the indefinite oscillations. The two wire line is connected to the four wire circuit by hybrid transformer.

Note : *In case of Digital EPABX SLIC functions are BORSCH whereas C stands for Coding. In SLIC the analog speech is converted into a PCM stream using a CODEC.*

C. O. LINE INTERFACE CIRCUIT

The CO Line is directly interface to EPABX through this circuit. This circuit is basically performing the following three functions:

1. Ring detection for incoming C O line call : The incoming ring is detected by this circuit at which it generates the ring detect signal which is fed to CPU. The ringer signal is $75V_{rms}$, 25Hz which is sent from PSTN to the EPABX on the C O line.

2. Overvoltage Protection : The overvoltage protection in this circuit is very important because the C O line is travelling a long distance from PSTN premises to the EPABX premises. The chances of the wire picking up noise are very high specially during the rainy seasons. Again very reliable overvoltage protection circuit is used to protect the EPABX C O line interface circuit from high voltage surges appearing across it.

3. Dialing out the pulses in case of outgoing C O line call. In this section, the number dialled in pulse mode by the subscriber is repeated using a relay which is activated and deactivated by the on/off state of the pulse.

CPU

The Central Processor Unit is based on 8/16/32 bit microprocessor. It has system memory in a RAM, programmable memory in EPROM, battery back up circuit for RAM.It has decoding logic which enables/disables various I/O ports by selecting that particular address. The control unit receives the digital information from line and trunk interface unit and keeps the record within.

SWITCH AND TONE GENERATION UNIT (TGU)

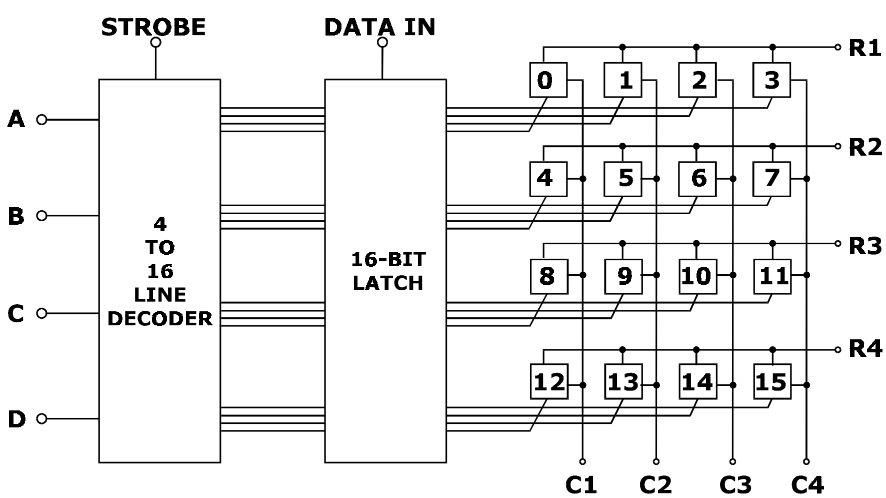
We will discuss this block separately for analog and digital EPABX.

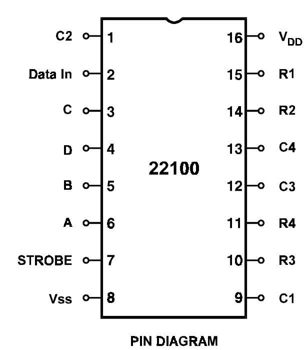
In an analog EPABX the switch is based on cross point which is further explained in the following paragraph. These cross points are now-a-days available in an integrated circuit replacing discrete ones. The various tones required for signalling between an EPABX and subscriber are generally in the TGU part.

In digital EPABX the switch is based on TGU in which there are two RAMs generally known as SPM (Speech Path Memory) and HM (Hold Memory). The tones are not generated within an EPABX but are available in TONE ROM which is available in the market. In Tone ROM the tones are stored in digital form.

CROSS POINT SWITCH

22100 - 4x4 cross point switch with control memory





IC 22100 is a 18 pin IC and is commonly found in the switch section of an analog EPABX for SDM switch.

The 22100 consists of 16 crosspoint switches (analog transmission gates) organised in 4 rows and 4 columns. Both devices have 16 latches, each of which controls the state of a particular switch. Any of 16 switches can be selected by applying its address to device and pulse to strobe input. The selected crosspoint will turn off it during strobe, data in was 1 and will turn off if during strobe, data in was a 0. In addition this IC will reset all non selected switches in the same row as selected switch. Other switches are unaffected in the IC, an internal power on reset turns off a switch as power is applied. This has following features:

* Internal latches control state of switches.
* Power on reset
* Low on resistance
* Large analog range (V_{DD} - V_{SS})
* All pins are diode protected
* High CMOS noise immunity

A	B	C	D	Select
0	0	0	0	C1R1
1	0	0	0	C2R1
0	1	0	0	C3R1
1	1	0	0	C4R1
0	0	1	0	C1R2
1	0	1	0	C2R2
0	1	1	0	C3R2
1	1	1	0	C4R2
0	0	0	1	C1R3
1	0	0	1	C2R3
0	1	0	1	C3R3
1	1	0	1	C4R3
0	0	1	1	C1R4
1	0	1	1	C2R4
0	1	1	1	C3R4
1	1	1	1	C4R4

SIGNALLING TONES

Dial Tone (DT) : Dial tone is a continuous signal of 400 Hz frequency. When the subscriber goes off hook, EPABX sends the Dial tone as a go ahead signal to the subscriber to dial the Number.

Busy Tone (BT) : Busy tone is also 400 Hz tone with some off/on time. This tone is fed to calling subscriber by an EPABX when the called subscriber is in busy condition.

Ring Back Tone (RBT) : Ring back tone is 400 Hz modulated with 33 Hz or 50Hz. This tone is fed to the calling subscriber. This tone is fed to the calling subscriber is in when the called subscriber is in On-Hook state.

Music on Hold (MoH) : This tone is fed to the calling party when it is put on hold at the time of call transfer from one extension to another.

Intrusion / Barge in Tone (IT) : This is an EPABX feature tone and fed to the subscriber's in conversation. This tone is an indication to go on hook as some other important call is in queue.

Confirmation Tone (CT): This is again a feature tone which is fed to a subscriber when it registers a particular feature by dialling feature code. This tone is a confirmation from EPABX to subscribe that it has registered the feature.

MDF (Main Distribution Frame)

All the subscriber and C O line connections are terminated from the circuit in an EPABX on a MDF. It is from MDF only that the wire is picked up for installation of telephone instruments within the building premises and PSTN. All the connection changes are made on MDF only.

MDF shall provide for

i) Accessing line or exchange side independently for test purposes.
ii) Protection against overvoltages and overcurrents for each extension and trunk.
iii) Connection flexibility between exchange and line terminals.

POWER SUPPLY

The Power Supply consists of a main transformer, which coverts 220 volts. A.C. to 240 A.C. and 9V A.C. from these AC voltage following D.C. voltages are drawn.

24V Regulated D.C.
15V Regulated D.C.
5V Regulated D.C.
75V RMS A.C.

24V Regulated D.C.
The 24V AC is converted into D.C. by bridge, rectified and fed to regulator IC, which is biased to produce regulated 24V DC is the line voltage for the extensions.

15V Regulated DC
Bridge converts the 24V AC into DC which on rectification is fed to regulator IC. It is biased to produce regulated 15V output, which drives the cross point switches (IC 22100) and another 15V is generated to drive the other analog components in the circuit like operational amplifier and 555 timer.

5V Regulated DC
Bridge converts 9V AC into DC and is fed to regulator IC after rectification. It is biased to give 5 volt at its output. This 5V drives optocouplers on subscriber card, microprocessors and some other IC'S on CPU card.

75V RMS A.C.
(Ringer voltage for the extension) The 15 volt supply is fed to the primary of ferrite transformer.

The signal thus generated is shifted below ground and a 25 Hz signal is then superimposed to produce 75V RMS below ground level .

CALL PROCESSING

Call processing is the procedure to set up a call between two subscribers. It is carried out in two stages: (i) Signalling, and (ii) Speech Path Established. In the first stage, an EPABX proceeds as per the program stored in EPROM to activate/deactivate different hardware to set up a call which is explained below.

When hand set of the station telephone is picked, it sets the I/P When sensed by CPU appropriate switches are activated to give dial tone to the station and hence a link. The dialed number or code is taken by CPU through touch tone receiver. It is station to station call, it gives ring enable signal to that particular station and ring back tone to the caller, if the calling party is busy/OFHK, then the busy tone is put on the callers line by activating the appropriate switch. If it is out station call, it searches for free line. If busy/OFHK busy tone is given or else the particular relay is switched on and loop is established. Then depending on the status of the line i.e. tone or pulse the no is dialled either by pulsing the relay or directly O/P the analog (tone) signal from the station. The LED is made to glow for the particular line when engaged.

If the user is using pass word the feature programmed are directly put into RAM. In case of incoming call from P&T line, according to service group programmed for that particular No. i.e. junction line, the ring enable signal is given to the programmed station and thus the ring hand set. Relay operators and link is established. Whenever there is incoming ring, the ring is AC coupled to optocoupler which sets the signal there by changing the status of the line. This signal is sensed by the CPU and accordingly flickers the LED for visual indication.

FEATURES OF EPABX

The entire range of features are classified into following groups

SYSTEM FEATURES

Class of Service

This feature allows the users to be combined into different groups, each with access to different features, and each with different external calling capabilities.

Direct Outward Dialling (DoD)

An extension user can dial external calls directly without going through the operator.

Paging

It provides a PABX connection to the loudspeaker so that users can page someone by dialling special code and speaking into the telephone caller.

Tie Line Networking
The basic objective of Tie-Line Networking is to provide direct communication between two or more PABXs without operator intervention.

Call Privacy
Full privacy of conversation is ensured for all calls whether established directly or via attendant. The Intrusion is normally indicated with a tone.

Line Out Of Service
If an extension is in dial tone state for more than 20 sec it is put into the Line lock out state. Declaring the Extension as LLO means out of service and any attempt like dialling digit etc. will be ignored.

Direct Inward Dialling (DID)
External subscriber can use direct inward dialling to reach an extension. The external subscriber is fed a tone from pabx when it reaches PBX through central office. This tone is an indication for him to start dialling extension number on DTMF basis.

Emergency
In case of an emergency, any extension reporting user can call up a set of predetermined emergency reporting extensions.

Distinctive Ringing
Different ring patterns can be provided for different calls. For example, internal call may be indicated by a single ring, external calls by two quick rings.

Trunk Queuing
Users placing an external call when all trunks are busy will be queued or put in line, for a trunk. With some systems the user can hang up, and the system calls back when the trunk is available.

EXTENSION FEATURES

An EPABX offers a set of features to the extension users that are described below. An extension user can invoke these features using accesses code or feature code.

Extension To Central Office
An extension can access central office (CO) through Junction by dialling two accesses codes. One of these access code is used when an extension wishes to dial a local number, and second one when he wishes to dial STD/ISD number.

Local Call
An extension dial 'O' after receiving dial tone on going off-hook, to get a free junction. If class of service permits the extension to make an outgoing call & free CO junction is available, the extension gets connected to the junction. The extension now receives the CO dial tone and proceeds to dial further digits. If all trunks are busy, extension gets busy tone. If class of

service does not permit an outgoing CO call from extension, he gets NUT. Call is terminated when extension goes on-hook.

STD-ISD Call

If CO Trunks are divided into STD & non STD categories, an extension with STD allowed would have to dial access code to get a CO trunk of STD Category.

Extension To Extension Call

Extension receives dial tone on going off-hook and required extension no. Ring is extended to the called party, if free, and ring back tone to the caller. The caller gets busy tone if called party is busy or NU tone if called extension is not listed in the directory or has been put out of service by operator due to some fault detected.

Extension To tie Line

Extension user can access tie line by dialling ‘8’ .A free tie line is searched and given to extension.

Automatic Call Back-Busy

An extension user can invoke automatic call back feature by dialling feature code on encountering a busy condition on a called extension. You can cancel the feature by the feature code.

Call Forwarding

The feature allows an extension to forward all his incoming calls to any other extension by invoking the feature code followed by the extension number to which call is to be forwarded.

Three types of CALL FORWARDING are available.

1) **Call Forwarding always :** Once an extension has invoked this feature, all the calls meant for that extension are forwarded to the other extension.

2) **Call Forwarding on Busy :** When this feature is invoked, all the calls for this extension are forwarded only when the extension is found busy.

3) **Call Forwarding on no answer :** Once this feature is invoked, then call forwarding is done after ring time out on this extension.

Call Consultation

An extension engaged in an external call can hold this call and dial another extension for private conversation.

Call Pickup

An incoming call due to which an extension is ringing can be picked up by nearby extension user, by dialling feature code and the ringing extension number.

Call Transfer

An incoming or outgoing call can be transferred to any other extension.

Executive Override

This feature allows a priority extension marked" Executive Class" to intrude into a busy extensions call. 'Executive Override' gets activated automatically when a user defined as Executive category tries to contact an extension busy in conversation. Initially to is returned to the calling extension and an intrusion tone is fed to the parties in conversation. After some time gap the executive gets connected to the called party and the third party is put on hold.

Do Not Disturb

This feature facilitates an extension to block all incoming calls, internal or external, to it. This feature is activated by dialling the feature code. When this is activated, system will route the calls to the available operator.

Call Parking

Allows user to put calls in progress on hold at another person's extension. For example call answered at your desk can be answered at your colleagues desk. You can then go to your colleagues desk. You can then go to your colleagues office, get some information and pick up his, phone, continuing your conversation.

DIFFERENCE BETWEEN EPABX AND KTS

What is KTS ?

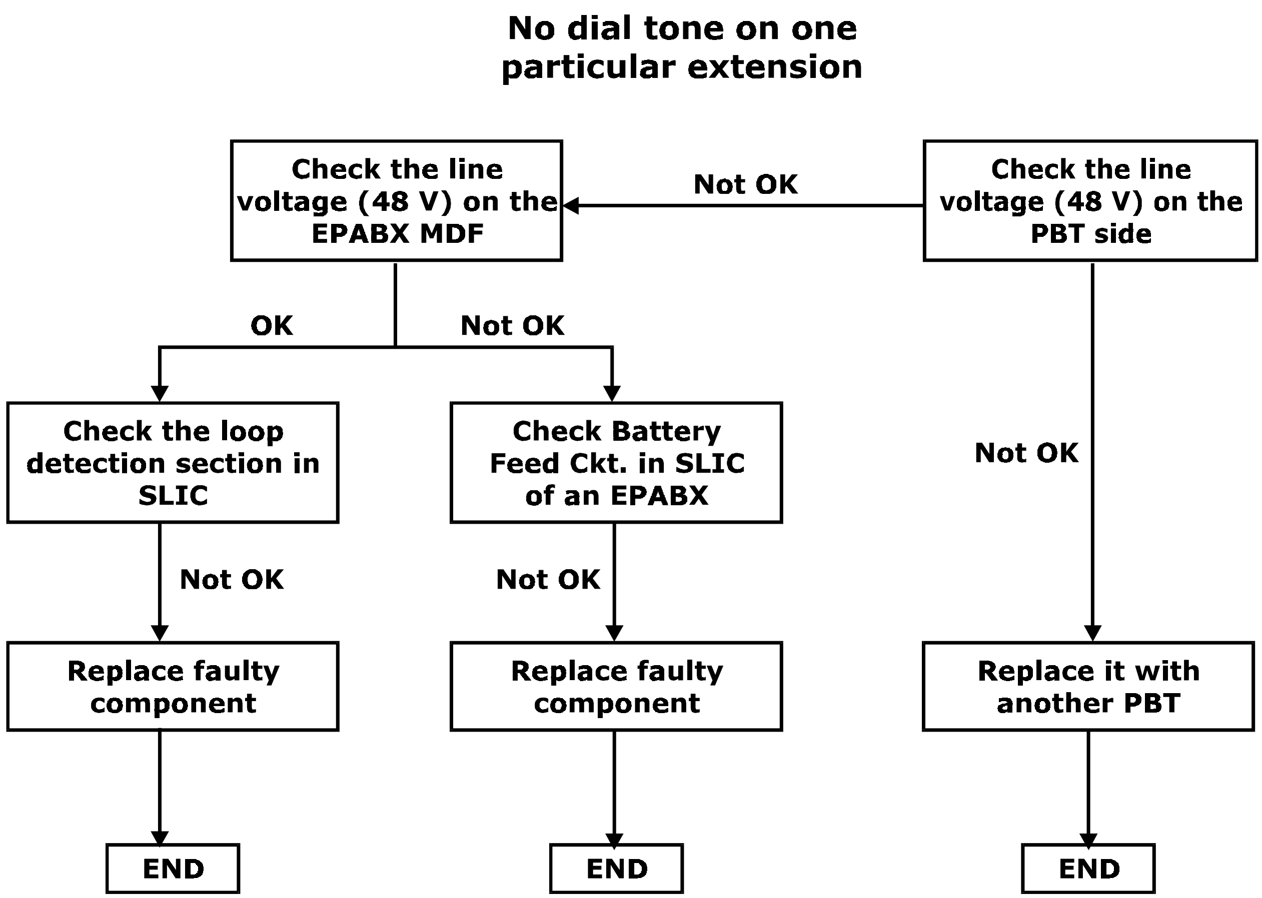
The discussion on EPABX would have been incomplete without KTS. KTS stands for Key Telephone System which is also used for private communication like EPABX. The KTS has certain differences over EPABX which are discussed below :

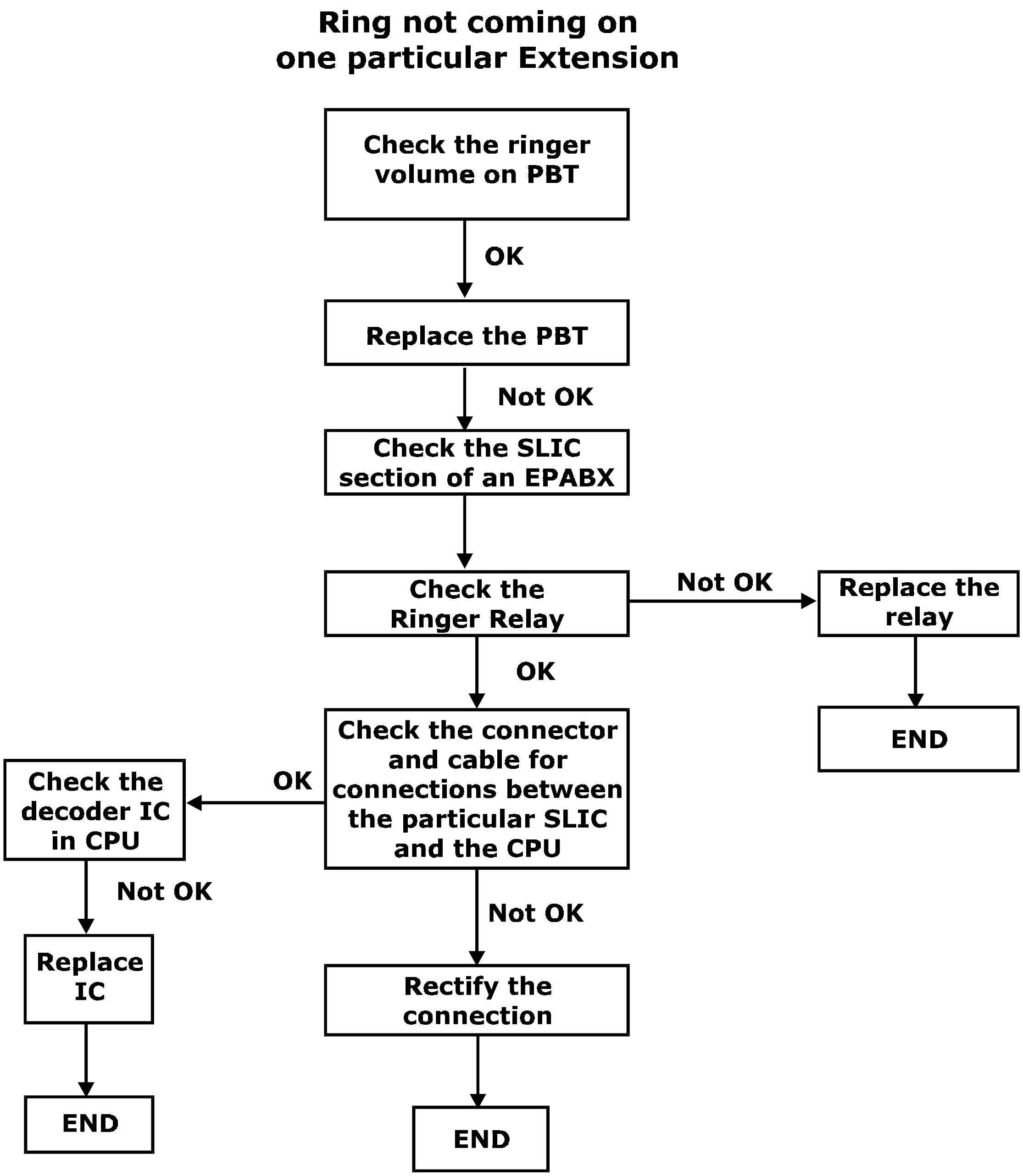
KTS	EPABX
It is possible for all users to have an individual status of all the trunk lines in the system. These keys can be accessed by one touch operation. If required, a group of lines can also be assigned to one key	In most cases, it is not possible to assign a trunk line to a key and have line status. In some cases, it is possible to assign lines to the keys but there is a limitation to the number of keys that can be assigned
The main advantage in a KTS is the logic of distributed intelligence to the extensions, i.e., the key phones	The logic of centralized intelligence is used, whereby the control is only limited to the operator consoles
It is easily possible to assign any-key-to-any-system feature. The keys can be assigned as any of the the following: a. Line key b. Extension key c. Feature key like forward, lock, DND, and message waiting d. One-touch calling for external numbers	The programming of keys is limited to certain features of system. In many cases, the feature keys are pre-programmed and cannot be changed
It is possible to have a status of all extensions in the system. In case, the keys on the phone are not sufficient, then an expansion set can be connected to the key phone without the need of any additional hardware in the system	The number of keys on the phone limit the number of keys that can be assigned for one-touch calling. The addition of an expansion console requires expensive additional hardware
Enables operator-less operations as the incoming calls can be distributed to the key phones	It is always necessary to have at least one operator console for incoming call distribution
The display on the key phone can be used for message text transfer, feature activation status, etc. Thus enabling user-friendly operations	The display is limited to certain fixed messages only
Hierarchical flexibility, this means that there can be any combination of key phones and single-line telephones	At least one operator console is required, thus depriving the user the convenience of the key phone

ASSIMILATION EXERCISE

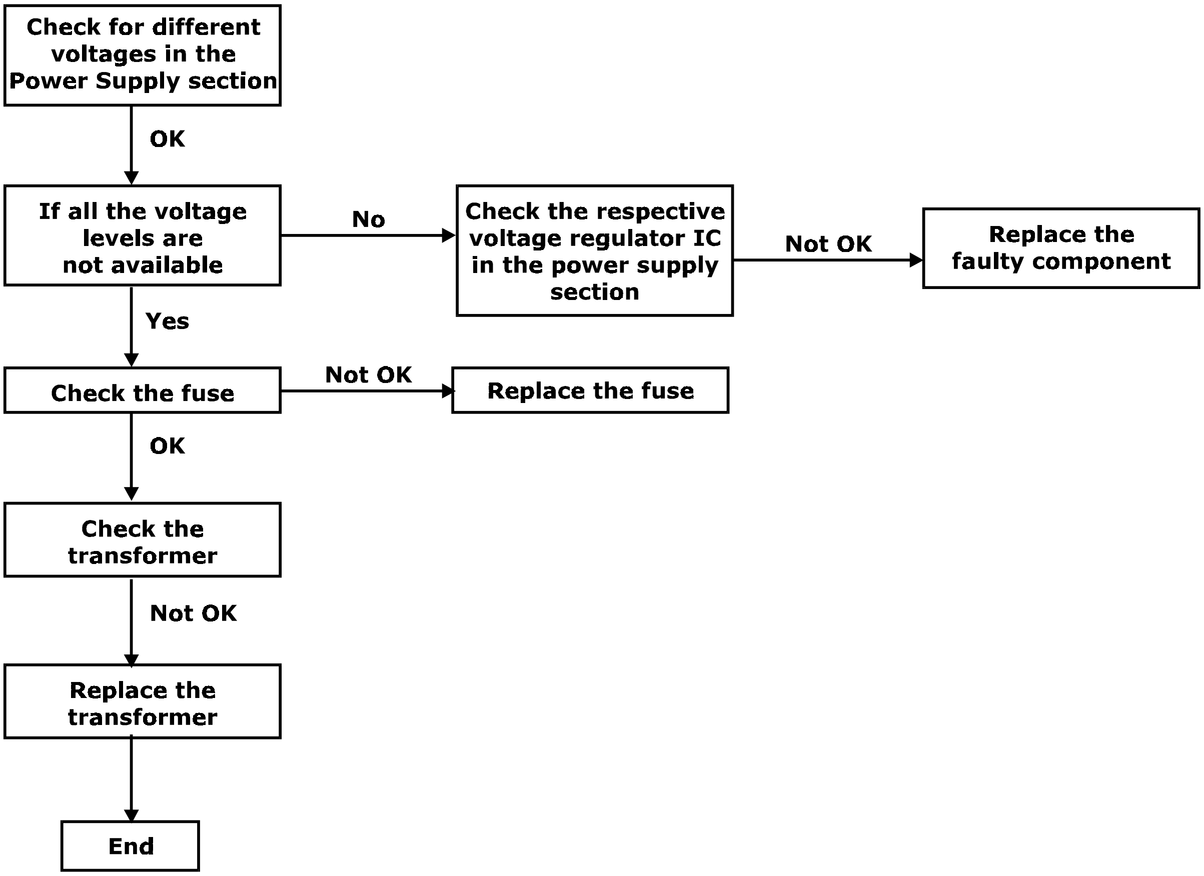
Q.1 Expand EPABX?
Q.2 What are different types of an Epabx?
Q.3 What is the difference between the analogue & digital EPABX?
Q.4 Draw the block diagram of an EPABX.
Q.5 List the functions of SLIC block.
Q.6 What is the ringer voltage?
Q.7 Explain the following features
- a) Camp on Busy
- b) Do not Disturb
- c) Executive Override
- d) Call forwarding
- e) Class of service

TROUBLESHOOTING

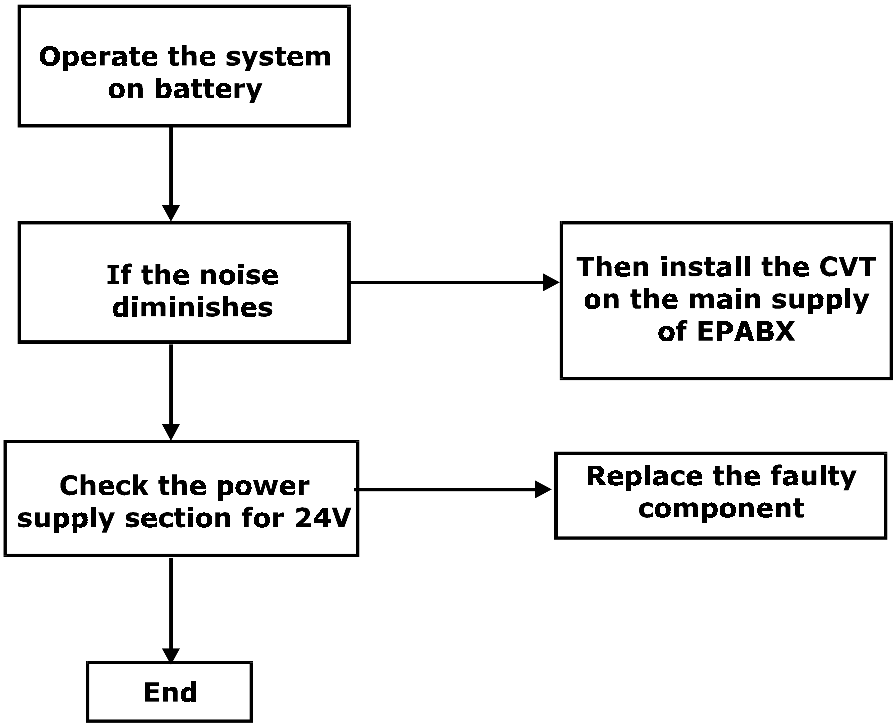



Ring not coming on
one particular Extension
Check the ringer volume on PBT
OK
Replace the PBT
Not OK
Check the SLIC section of an EPABX
Check the Ringer Relay
Not OK
Replace the relay
END
OK
Check the connector and cable for connections between the particular SLIC and the CPU
OK
Check the decoder IC in CPU
Not OK
Replace IC
END
Not OK
Rectify the connection
END

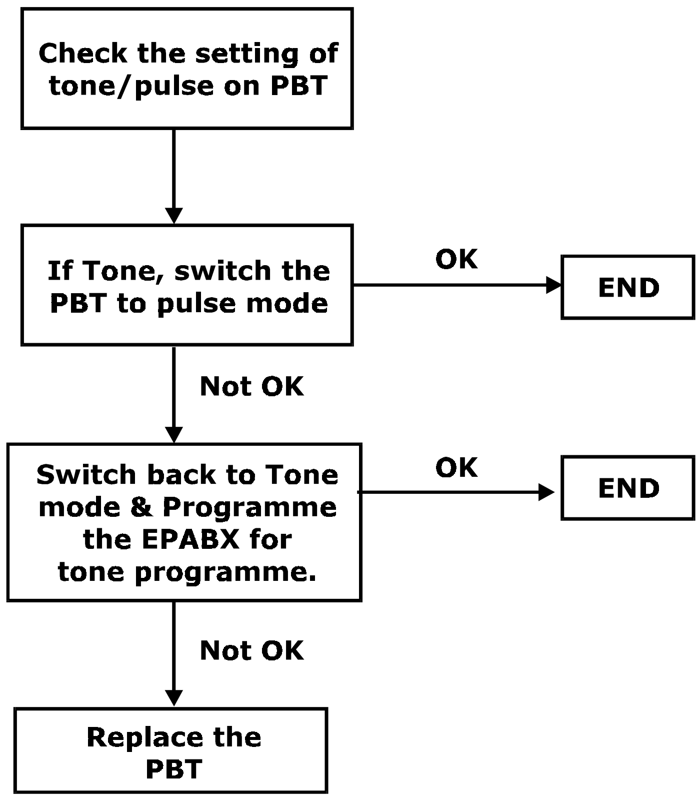
EPABX is completely DEAD



Humming Noise on the Extensions



Dial tone does not disconnect on Extension line while dialing



CELLULAR TELEPHONY

COMPETENCY OBJECTIVES

The objective of this Chapter is to introduce the students to:-

- Introduction
- Types of EPABX
- Architecture and Interface
- History
- Circuit Details
- Features
- Troubleshooting

Chapter 4

CELLULAR TELEPHONY

INTRODUCTION

Cellular telephony involves dividing a large service area into regions called “cells”. Each cell has the equipment to switch ,transmit and receive calls from any subscriber located within the borders of its radio coverage area.

If cells were not used the transmitter would need to use very high power to cover a large area .Using cells ,means that the area covered by a single transmitter is reduced ,thereby reducing the need for high powered transmission.
Cells are conventionally regarded as being hexagonal ,but in reality they are irregularly shaped .The cell shape is determined by the nature of the surrounding area e.g hills ,tall buildings etc.The cells also overlap one another.

The current frequency spectrum is very full with only a narrow bandwidth available for cellular telecommunications .This means that the bandwidth we have must be used very efficiently .Each cell in the cellular Network requires an RF carrier to be allocated to it.

An RF carrier refers to a pair of radio frequencies ,one is used in each direction(transmit and receive) so that information may be passed in both directions simultaneously .The transmit and receive frequencies are separated by a set distance of 45 MHz to avoid interference.

There are not enough frequencies available for every cell to have a different RF carrier.There are 124 RF carriers in the GSM frequency band ,therefore if each carrier can carry seven phone calls a maximum of 868(124*7) calls may be made .This is clearly not enough and therefore the frequencies must be reused.

CELL SIZE

The number of cells in any geographic area is determined by the number of mobile subscribers who will be operating in that area and the geographical layout of the area(hills, lakes , buildings etc.)

Large Cells

Present cellular systems use UHF frequencies in the bands 872-960 MHz at these frequencies the normal maximum cell radius is about 20 miles(32 Km) Large are generally used in remote areas where there are very few subscribers.

* Remote Areas
* High Transmission Power
* Few Subscribers

Small Cells

The minimum cell size is determined by the number of mobile subscribers who will be using the cell .In an urban area where a large number of people are located ,very small cells will be used which are about half a mile across (0.8 Km).Small cells have the advantage of being able to use a low transmit power this means that small lightweight hand portable phones may be used.

GSM - GLOBAL SYSTEM FOR MOBILE

GSM is a cellular system standard which was developed to meet the need for a common standard for mobile telecommunications .Europe GSM is a second generation standard using the digital modulation techniques .Before this standard the Europe was using different analogue standards for different places for which it was not possible to have a common subscriber unit .Presently it is the most popular standard .In India we are using the same standard for the cellular telephony application .GSM first came in the year 1991, in Europe.

GSM uses the 900 MHz bandwidth time division multiple access(TDMA) with eight time slots (i.e transmitting for one eighth of the time).GSM 's radio channel separation is 200 KHz . The frequency band used is 890-915 MHz (mobile transmit) and 935-960 MHz (base transmit).

FEATURES OF GSM

FLEXIBILITY AND INCREASED CAPACITY

With the analogue air interface , every connection between a mobile subscriber and a cell site requires a separate set of RF hardware at the cell site .Therefore to expand the capacity of a cell site by a given number of channels ,an equivalent quantity of RF hardware must be added to the cell site equipment .GSM uses the available radio spectrum more efficiently .Eight simultaneous conversations can now be carried out on one RF carrier .This means that separate RF hardware is only required for every eight subscribers .The system is also more versatile and it is possible to move capacity from one part of the network to another by reconfiguring the system database.

NOISE ROBUST

In the analogue cellular telephone system the mobile unit communicates with the cell site by means of analogue radio signals.Although this technique can provide an excellent audio quality ,it is vulnerable to noise as anyone who has tried to receive broadcast stereo with a poor aerial will testify.
The noise which interferes with the current system can come from any of the following sources.

* A powerful or nearby external source
* Another transmission on the same frequency
* Another transmission “breaking through “ from a nearby frequency
* Background radio noise intruding because the wanted signal is too weak to exclude it.

The signals passed over a digital air interface can be protected against errors caused by noise .This protection comes from encoding the signal .This enables the errors in a signal to be detected and also corrected .The end result is a much more robust air interface.

IMPROVED SECURITY AND CONFIDENTIALITY

Security figures high on the list of problems encountered by some operators of analogue systems .GSM offers high speech and data confidentiality both inherent in the method of transmission over the air interface and the manner in which traffic is processed prior to transmission .The GSM system provides high degree of confidentiality for the subscriber ,calls will be digitized ,encoded and then ciphered when sent over the air .This will make listening in to some ones call virtually impossible.

SUBSCRIBER IDENTIFICATION

With the analogue systems a mobile subscriber is identified by a telephone number which is associated with their mobile equipment .This number is held in the mobile Equipment of that subscriber ,therefore if the subscriber wishes to make or receive calls he must take the mobile equipment with him .With the GSM system the subscriber and the mobile equipment are identified separately .The subscriber is Identified by means of a “Smart Card” known as a SIM(Subscriber Identification Module).

ISDN COMPATIBLE

ISDN (Integrated Services Digital Network) is a standard that most developed countries are committed to implement .The GSM network has been designed to operate with the ISDN system and provide features which are compatible with it.

GSM NETWORK COMPONENTS

The principle main component groups of a GSM Systems are:

THE MOBILE STATION (MS)

This consists of the mobile telephone ,fax machine etc.

The Base Station Subsystem (BSS)

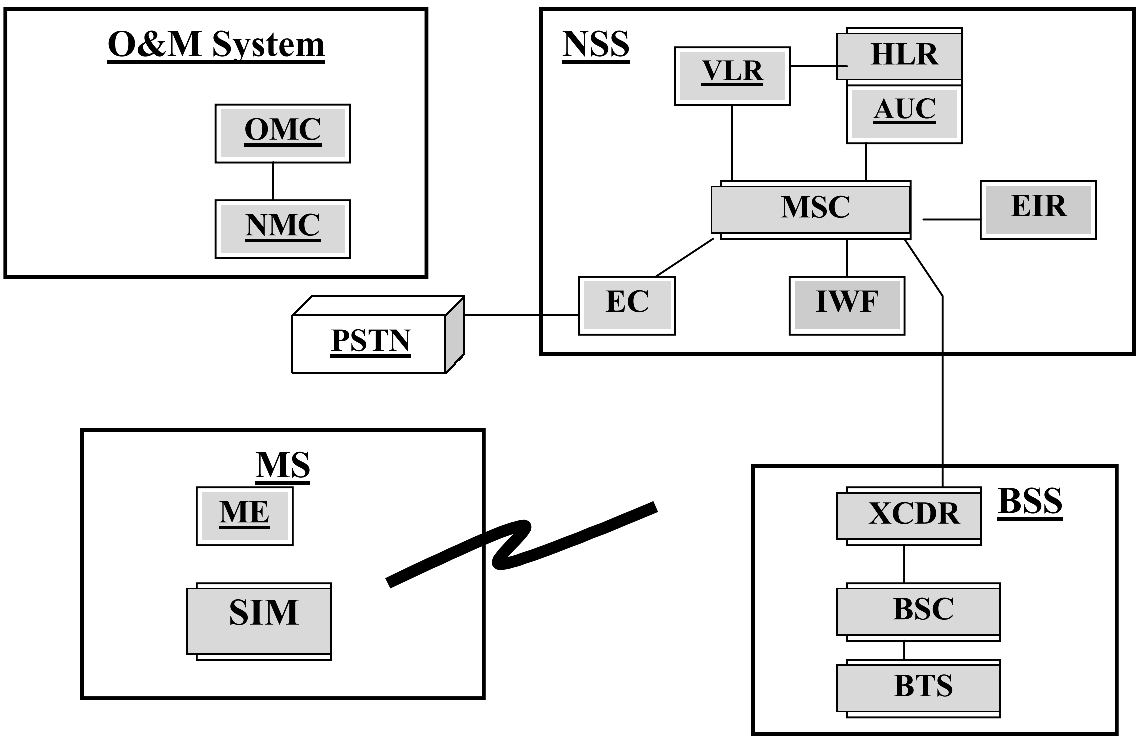
This is the part of the network which provides the radio inter connection from the MS to the land based switching equipment.

The Network Switching Sub-System (NSS)

This consists of the Mobile Services Switching Centre(MSC) and its associated system control databases and processors together with the required interfaces .This is the part which provides for interconnection between the GSM network and Public switched Telephone Network(PSTN).

The Operations and Maintenance System

This enables the network operator to configure and maintain the network from a central location.



MOBILE STATION - MS

The Mobile Station consists of two parts ,the Mobile equipment (ME) and an electronic ‘smart card’ called a Subscriber Identity Module (SIM).

The Mobile Equipment is the hardware used by the subscriber to access the network.This may be a telephone ,fax machine ,computer etc .The hardware has an identity number associated with it which is unique for that particular device and permanently stored in it.The Mobile Equipment is the only part of the GSM network which the subscriber will really see.

The SIM is a card which plugs into the Mobile equipment .This card identifies the mobile subscriber and also provides other information regarding the service that subscriber should receive. Without the SIM inserted the mobile equipment will only be able to make emergency calls.

BASE STATION SUBSYSTEM

The GSM Base Station Sub-system is the equipment found at a cell site ,it comprises of a combination of digital and RF equipment .The BSS provides the link between the mobile equipment and the Mobile Services Switching Centre.

The BSS consists of three major hardware components these are detailed below.

The Base Station Controller - BSC

The BSC as its name implies provides the control for the BSS .The BSC communicates directly with the MSC.The BSC may control single or multiple BTSs.

The Base Transceiver Station - BTS

The BTS contains the RF components that provide the air interface for a particular cell .The Antenna is included as part of the BTS.

THE TRANSCODER - XCDR

The Transcoder is used to compact the signals from the mobile station so that they be more efficiently sent over the terrestrial interfaces .Although the Transcoder is considered to be a part of the BSS it is very often located closer to the MSC.

NETWORK SWITCHING SUBSYSTEM

The Network switching system includes the main switching functions of the GSM Network .It also contains the databases required for subscriber data and mobility management .Its main function is to manage communications between the GSM network and other telecommunication networks.

The components of the Network Switching system are listed below :

* Mobile Services Switching Centre-MSC
* Home Location Register-HLR
* Visitor Location Register-VLR
* Equipment Identity Register-EIR
* Authentication Centre-AUC
* Inter-Working Function-IWF
* Echo Canceller-EC

OPERATIONS AND MAINTENANCE SYSTEM

The operations and maintenance sub-system provides a capability to manage the GSM network remotely .The Operations & Maintenance System comprises of two parts.

Network Management Centre-NMC
Operations and Maintenance Centre-OMC

ASSIMILATION EXERCISE

Q.1 What is GSM?

Q.2 Write the specificatons of a GSM system.

Q.3 Define a cell.

Q.4 What are components of the GSM network?

Q.5 What is the function of a SIM card in a Mobile Equipment?

PAGING

COMPETENCY OBJECTIVES

The objective of this Chapter is to introduce the students to:-

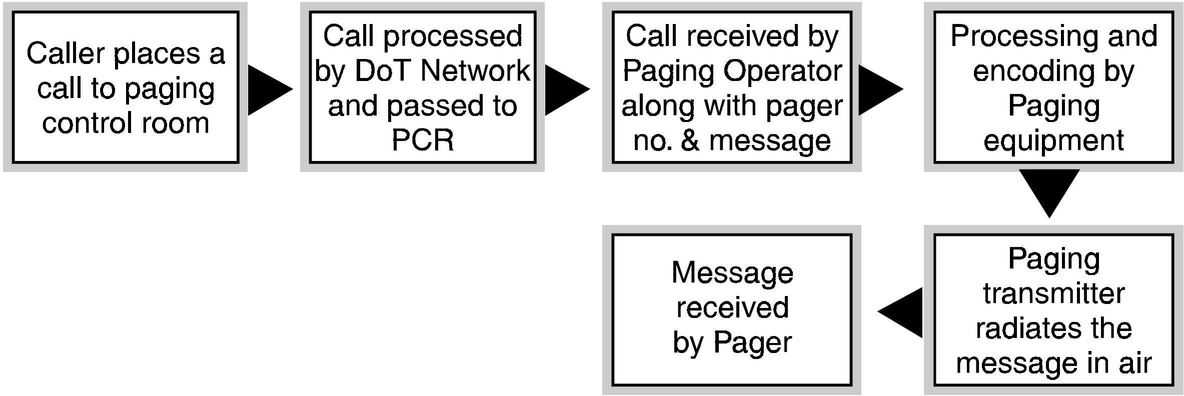
- ❖ Introduction
- ❖ Types of EPABX
- ❖ Architecture and Interface
- ❖ History
- ❖ Circuit Details
- ❖ Features
- ❖ Troubleshooting

Chapter 5

PAGING

INTRODUCTION

Paging is one way, receive only mode of mobile communication .Paging ,as a concept has emerged as the best ,efficient and most economical way of communication for the people who are generally on the move .Paging systems are designed as extensions of the telephone network without the speech facilities. Following is the Signal Flow Diagram for Paging System.



TYPES OF PAGING

There are two types of paging

Operator Assisted Paging
Call is received & processed by the paging operator.

Auto Paging
Caller directly lands on to the paging terminal without intervention of operator.

ELEMENTS OF PAGING CONTROL ROOM

Paging Control Terminal (PCT)
It is the central controller of the infrastructure.This is microprocessor based equipment having interface for operator assisted as well as auto paging.It consists of the complete subscriber database and subscriber information.

Paging Transmitter

Paging transmitter is used for transmitting paging messages.Paging Tx is synthesised device operating in VHF/UHF band.

Call Distribution System/EPABX

This is used for distributing calls to paging operators via EPABX.

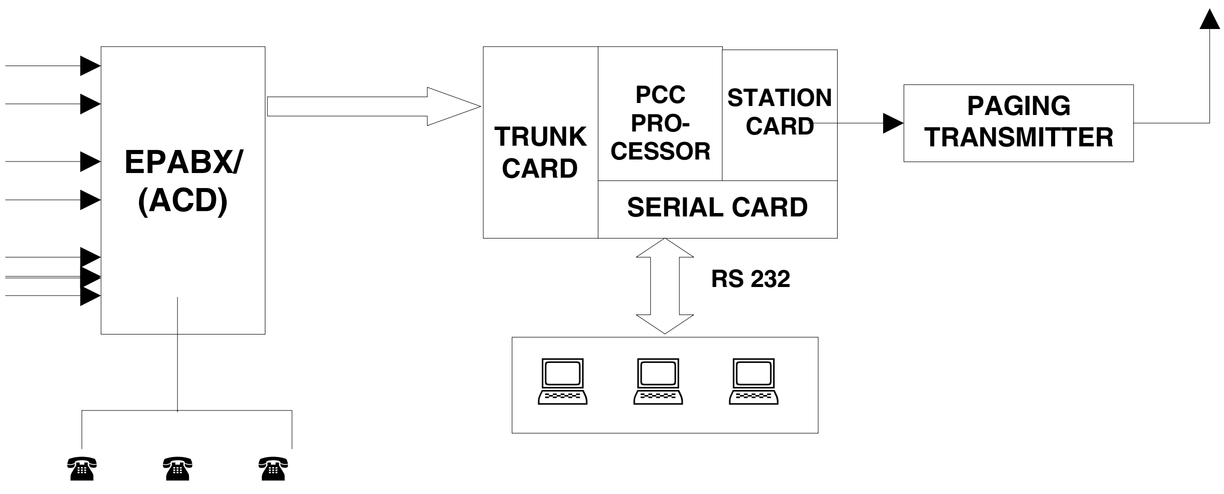
Operator Terminal

Operator Terminals are used for entering messages to paging terminal.

Computers, Modems & Peripherals

These devices are used for controlling various equipments,for remote logging,billing etc.

Uninterrupted Power System



Block Diagram of Paging Control Room

PAGING STANDARDS

There are three types of paging standard being used worldwide namely

Analogue Paging Standard

Analogue standards are wide area paging standard being used since 1960's.An example of this kind of analogue standard is 2-tone and 5-tone Paging.

FM-RDS Paging Standard

Digital Paging Standard

Presently ,Digital paging standards are most popular worldwide.Existing digital paging standards are

a) POCSAG Format:POCSAG stands for Post Office Code Standardisation Advisory Group.
b) GOLAY Format: GOLAY stands for GOLAY Sequential coding
c) NEC Format : NEC digital format is mainly used in JAPAN

These formats use 2-level FSK.The difference between POCSAG and GOLAY is the rate of data transmission.GOLAY was designed to operate at 300 bps.POCSAG has been designed to operate at 512bps,1200 bps and 2400 bps(APOC).It has got the advantage of accomodating more no. of subscribers on one channel due to high speed.It is the most popular digital format in use today, worldwide.

PAGERS

Pager is a small hand-held receiver used for receiving paging messages.The Pager size is typically 2"x 3" and weight is approximately 100 grams (including battery).The display is based on LCD (Liquid Crystal Display).The receiver is powered by a single AAA size alkaline battery and operates in the VHF frequency(138-174 MHz) range.

Standard features of PAGER

1. Attractive and miniature style.
2. Power back-up.
3. Automatic Power on/off.
4. Message protection by a user's password.
5. Time Stamping.
6. Alarm
7. Contrast selectable LCD module
8. Long battery life by full custom decoder IC.
9. Low battery life logo
10. Out of range logo display
11. Memory full logo display
12. Duplicate Message check
13. User selectable font size
14. Information receiving function

ASSIMILATION EXERCISE

Q.1 What are the elements of a paging control room?
Q.2 What are the various Paging standards?
Q.3 Expand POCSAG standard and also explain it.
Q.4 What is a PAGER?
Q.5 What are the features of a PAGER?