Design & Fabrication Of Electronic Private Automatic Branch Exchange

Capt Hammad Zaka,

Capt Umar Shahzad

CHAPTER I

INTRODUCTION

The desire to communicate is as old as a legend. All the Prophets communicated with Allah through one way or another. The importance of communication is obvious from the fact that it is one of the basic instincts of every intelligent life form in present age. Modern scientific world owes its existence to different and better modes of communication evolved during the progress of human civilization. It is because of these fast means of communication, that world has now become a "Global Village".

1.1. ESSENTIALS OF TELECOMMUNICATIONS

There are four essentials for effective information transfer between two points, all of which are provided in well-designed telecommunications systems:

(i) A transmitting device

(ii) A transport mechanism

(iii) A receiving device

(iv) The fourth requirement is that the transmitter transmits only information, which is compatible and comprehensible to the receiver. The coding and method of transfer of the information over the transport mechanism is said to be the protocol.

1.2. SIGNAL

Information conveyed over telecommunications systems is termed as a signal. It can either be an analog signal or a digital one. An analog signal is an electrical waveform, which has a shape directly analogous to the information it represents (e.g. speech or a telephone picture). A digital signal involves transmission of

information in a string of binary digits or more precisely, pulses of on” and ‘off’ electrical current. (Fig 1.1)

1.3. TRANSMISSION MEDIUM

The transmission medium is what actually carries a signal from point to point in the network. The signal carried by the medium may be voice or data, network control signals, or combinations of the two. Each medium has its own particular advantages and disadvantages. Although modern transmission mediums can be found in many shapes and sizes they can typically be separated into three categories: wire, radio and fiber optics.

1.3.1. WIRE MEDIUM

Wire is certainly the oldest and most straightforward of all the mediums, yet it still remains the foundation of the network. Its types include the twisted pair cable and the co-axial cable as shown in fig 1.2. These two mediums in conjunction with the open wire lines are common transmission media found not only in the local loop and exchange network, but in the long-haul network as well.

Wire has several disadvantages. It is expensive, heavy and bulky. The cost of installing and repairing long—haul wire is often prohibitive when compared with other new media. Wire is also susceptible to such environmental effect as

Corrosion, noise and voltage spikes.

1.3.2. RADIO MEDIUM

Radio takes into account microwave and satellite communication. It offers many advantages over wire in the exchange and long haul networks. A typical microwave link can handle more than 40,OQO voice channels in an analog system.

The network can be altered or rearranged easily without having to relocate huge amounts of copper or fiber optic cable.

Maintaining radio systems is also less expensive than a wire network. On the negative side, radio systems installation requires huge expenditures for structures, equipment, and real estate. Radio systems are also subject to the problems associated with propagation and atmospheric conditions. Reflection, refraction, diffraction, fading and interference can also work to degrade the transmission of signals through the atmosphere. (Fig 1.3)

1.3.3. OPTICAL FIBER

Optical fibers represent the newest frontier in telecommunications transmission media. Surrounding a high quality glass core with a glass cladding material of a different refractive index makes these fibers. Plastic core and cladding materials are also commonly used. Light introduced to the core is carried down the fiber by continuously reflecting at the core-cladding interface.

Fibers have many advantages. They are thin, light but very strong. A typical fiber can carry signals for great distances with less than 1 dB per km of attenuation. Fibers have a wide signal bandwidth. Some systems can carry over 30,000 voice channels. Repeaters required are spaced well apart — 35-80 km.

Optical fibers carry no electrical current and are free from noise, ground loop effects, cross talk and interference. This makes optical fiber the ideal choice for long haul installations. Splicing problems, Micro- and Macro-bend losses are the main constraints in the way of making multiple taps from the same piece of fiber, so its use is usually limited to direct point-to-point applications (Fig 1 .4)

1.4.COMMUNICATION MODES

Communication systems may operate in either a one-way (simplex) or a two-way (Duplex) mode.

1.4.1. SIMPLEX MODE

Transmission, which is confined to one direction, is known as simplex operation. Examples are broadcast systems and Tele-text services. A simplex terminal can send or receive data, but it cannot do both at the same time. Simplex operation is limited in terms of its operational capability but it is simple to implement since little is required by way of a protocol. it has a major limitation in that a receiver cannot directly indicate to a transmitter that it is experiencing any difficulty in reception. (Fig 1.5 a)

1.4.2. DUPLEX MODE

Communication in which the device at either end of the transmission channel can transmit and receive is known as duplex communication. Two possibilities exist for this two-way communication.

- Half Duplex Transmission

♦ Full Duplex transmission

Half-Duplex Transmission allows transmission in either direction, but only one way at a time. Hence, some form of protocol is necessary to ensure that one station is in transmit mode and other is in the receive mode at any one time as well as to determine when stations would change state e.g. wireless communication. (Fig 1 .5 b)

Full—Duplex transmission allows transmission in both directions at once. It can support simultaneous two-way communication by using two separate channels, one for each direction of transmission. This is more costly but is simpler to operate. Examples are telephone transmission, data transmission using the Internet. (Fig 1.5 c)

1.5. COMMON TYPES OF TELECOMMUNICATION SYSTEMS

In order to meet differing communications needs, a number of different types of telecommunications equipment have been developed over time. These include in chronological order:

- Telegraph, telex
- Telephone
- Data networks—using either circuit-switched' or packet-switched' conveyance,
- Computer local', metropolitan' and wide area networks' (LANs, MANs and WANs),
- Integrated voice and data networks

1.5.1. TELEGRAPH AND TELEX

Electric telegraphy was the earliest common method of digital transmission. It began in the early part of the nineteenth century, and became the forerunner of modern telecommunications. Samuel Morse, inventor of the Morse code, developed a working telegraph system in 1 835 based on the Morse code". 1837 established a practical system from Euston railway station in London to Camden, a mile away.

The transmission of information by telegraphy is a process of signalling by sonic form of code. The letters and characters which formed the message" are first of all translated, at the transmitter, into telegraph signals in the appropriate code: the coded signals are then transmitted to the receiving point where decoding takes place.

Telex and correspondingly the Murray code, another digital code gradually replaced telegraphy. Unlike the Morse code, its successor, the Murray code was especially designed for automatic working. The evolution from the Morse to the Murray code took place in a number of stages, first to a manual and then an automatic teleprinter system, finally to the widespread telex system. The output of the early teleprinter devices was a narrow strip of paper, which could be stuck on to a receiving telegram form. The telex system by contrast prints direct on to a roll of ordinary paper.

1.5.2. TELEPHONE

A telephone system is an arrangement of transmitters and receivers with electrical interconnection whereby the communication of speech can be carried on between any two points, which may be close together, or remote. Telephone is one of the most important means of communication. More details on this communication mode are given in the next chapter.

1.5.3. DATA NETWORKS

There are three types of real networks that are used for the conveyance of data:

1. Point to point data networks.
2. Circuit switched data networks.
3. Packet switched data networks.

Point to point networks involves the simple interconnection between two pieces of equipment and it is relatively simple to establish. The distinguishing property of a circuit switched connection is the existence through out the communication of the call, of an unbroken physical and electrical path between origin and destination points. In packet switched network, an entire physical path from origin to destination will not generally be established at any time during communication instead, the total information to be transmitted is broken down into a number elemental “packets” each of which are sent in turn.

1.5.4. LOCAL, METROPOLITAN AND WIDE AREA NETWORKS

A local area network is a communication network that interconnects a variety of devices and provides a method for information exchange among those devices. This data communication network is used to interconnect a community of digital devices distributed over for a moderate size geographic area up to, and more than l0sqkm devices may be office workstations, mini- and in microcomputers, intelligent instrumentation equipment, and so on.

A MAN is optimized for a larger geographical area than a LAN, ranging from several blocks of buildings to entire cities i.e., a network that links a set of LANs that are physically distributed around a town or city .It depends on communications channels of moderate-to-high data rates. Error rates and delay may be slightly higher than that obtained on a LAN. A MAN might be owned and operated by a single organization, but usually will be used by many individuals and organizations. MANs might also be owned and operated as public utilities

Any form of network private or public that covers a wide geographical area is termed as WAN. Wide area networks have been developed to support voice, data and multimedia communications over long distance networks. Typically, a WAN consists of a number of interconnected switching nodes, transmission from any one device is routed through these internal nodes to the specified destination device.

1.5.5. INTEGRATED SERVICES DIGITAL NETWORK (ISDN)

ISDN is a network evolved from the telephony integrated digital network (IDN) that provide end to end digital connectivity to support a wide range of services, including voice and non-voice services, to which the users have access by a limited set of standards multi-purpose customer interfaces.

ISDN has two generations. The first generation, sometimes referred to as narrow band ISDN, is based on the use of a 64-kbps channel as the basic unit of switching and has a circuit switching orientation. The second generation referred to as broadband. ISDN, support very high data rates (100 of Mbps) and has a packet switching orientation

1.6. COMMUNICATION NETWORK PLANNING

The following list is a brief summary of some of the factors which require consideration while telecommunication network planning.

- Transmission plan

A plan laying out guidelines according to which appropriate transmission media may be arranged in order that adequate end-to-end conveyance of information is achieved. The plan will include safeguards to ensure that the received signal is loud enough, clear enough and free of noise and interference.

♦ Numbering and routing plan

The numbering plan is crucial to the ability of the network to deliver communications to the appropriate destination. Much like reading an address on an envelope, it is the inspection of the destination number that permits the determination of destination and appropriate route within the network.

♦ Usage monitoring plan

A usage-monitoring plan is needed in order to ensure continuing and future suitability of the network. Adequate measurements of network performance are required in order that early steps may be taken, as necessary, to adjust the network design, or expand its overall throughput capacity to meet demand.

♦ Charging and accounting plan

Public telecommunication operators (PTOs) need reimbursement for their services. Specific equipment may be required to monitor individuals' use of the network, in order that bills can he generated.

♦ Maintenance plan

A maintenance plan is needed to guarantee continuation of service without a break.

Any of these items may be appropriate to any type of telecommunications net regardless of what type of information it is carrying.

CHAPTER 2

TELEPHONY

Telephony is the science of extending the radius of communication by translating the variations in air pressure into corresponding variations in electrical conductors and then translating them back into variations of air pressure to be intelligible to the listener.

The word telephone (which incidentally is derived from two Greek words. TELE" meaning distance and 'PHONE" meaning speech) was in use some considerable time before Bell's patents and was according to some authorities first used by Sir Charles Wheat Stone to describe a mechanical arrangement, which he had devised to transmit speech from one room to another. In modern usage, the true literal derivation is somewhat lost and the word telephone almost invariably implies the transmission of speech over a distance by electric current.

There exists two main forms of telephony and are classified according to the medium over which the information is sent.

- Line telephony: invented by Bell in 1875/76.
- Radio telephony: first used by Fesseden in 1906.

2.1. TELEPHONE SYSTEM

A telephone system is an arrangement of transmitters and receivers with electrical interconnection; thereby the communication of speech can be carried on between two points, which may be closed together or remote.

Telephone arrived as a practical instrument over a century ago in 1875 and outgrowth experiments on a device to send multiple telegraph signals over a single wire.

2.2. ESSENTIALS OF A TELEPHONE SYSTEM

There are five essentials of effective telephone communication.

- Subscriber Interface — Telephone Set
- Central Office
- Local Loop
- Trunk
- Channel Bandwidth requirements

2.2.1. SUBSCRIBER INTERFACE - TELEPHONE SET

Telephone Set is the interface between the subscriber and the central office. It is used to originate and receive calls and is simple in appearance and operation, yet performs a surprising number of functions. The most important ones are:

- It indicates the use of the telephone system when the handset is lifted.
- It indicates that the system is ready for use by receiving a tone called the dial tone.
- It sends the number of telephone to be called to the system.
- It indicates the state of a call in progress by receiving tones indicating the status (ringing, busy, etc)
- It indicates an incoming call to the called telephone by ringing bells to other audible tones.
- It changes speech of a calling party to electrical signals for transmission to a distant party through the system. It changes electrical signals received from distant party to speech for the called party.
- It automatically adjusts for changes in the power supplied to it.

- It signals the system that a call is finished then a caller" hangs up" the handset.

All the functions of a telephone set can be realized by employing a telephone circuit as shown in Block Diagram 2. 1.

2.2.1.a. TRANSMITTER

The part of a telephone set into which a person talks is called a transmitter. It converts speech (acoustical energy) into variations in an electric current (electrical energy) that can be transmitted through the transmission system to the receiver of the called telephone. The most common telephone transmitter in use today is in principle like the one invented about hundred years ago by Thomas Edison.

The transmitter consists of a small two-piece capsule filled with thousands of carbon granules as shown in figure 2.2. The front and back are metallic conductors and are insulated from each other. One side of the capsule is held fixed by a support that is part of the handset housing. The other side is attached to a diaphragm, which vibrates in response to the air pressure variation caused by speaking into it. The vibrations of the diaphragm vary the pressure on the carbon granules. If the granules are forced together more tightly, the electrical resistance across the capsule decreases. Conversely, if the pressure on the granules is reduced, they move a part and the resistance increases. The current flowing through the transmitter capsule varies because of the varying resistance; thus, the varying air pressure representing speech is converted to a varying electrical signal for transmission to the called party.

2.2.1.b. RECEIVER

The receiver converts the varying electrical current representing the transmitted speech signal to variations in air pressure usable by the human ear.

A simple receiver consists of an iron diaphragm and electromagnets as shown in fig 2.3. An electromagnet is an electrical device which acts like a magnet when current passes through its winding and tends to attract the iron diaphragm accordingly. The strength of the attraction depends upon the magnitude of the current flowing in the coil of the electromagnet (i.e. upon the telephone circuit current). The varying electrical signal created by the microphone results in a varying force of attraction on the diaphragm. This vibrates the surrounding air and recreates the original sound wave.

2.2.I.c. RINGER

When a call has been connected through to the central office serving the called subscriber's local loop, the central office must send a signal to the called party that a call is waiting to be answered. This signalling function is called ringing. An AC voltage producing a corresponding alternating magnetic field actuates the ringer mechanism. The field interacts with the permanent magnet fields, causing a pivoted armature to vibrate. The AC ringing signal (90 Arms at a frequency of l6 Hz) is generated at the central switching office. Multiple ringing mechanisms run simultaneously, this helps to more evenly distribute the load. (Fig 2.4)

2.2.I.d. ANTI -TINKLE AND SPEECH MUTING

High voltage spikes are produced each time the dial pulsing contacts interrupt the flow of loop current. These spikes of increased voltage can cause the bell of the ringer to sound as the pulses are generated. The ringing is fairly soft, like a tinkle, thus the circuit to prevent is called an anti-tinkle circuit. This circuit is often combined with the speech muting circuit, which is also added to the receiver network. A mute works in conjunction with the speech network's dialler interface to shut down (or mute) the receiver during dialling operations. This prevents loud tones or clicks from reaching the receiver. The mute control signal is generated by the external dialling circuit.

2.2.1.e. SWITCH HOOK

- **ON HOOK**

When the handset is on the cradle, the telephone circuit is open circuit and does not draw any power. This condition is called “on hook” state.

- **OFF HOOK**

Lifting the handset from the telephone cradle loops the telephone-to-exchange access line, thereby completing the circuit and causing loop current to flow, called the “off hook” condition.

Both these states are shown in figure 2.5.

2.2.1.f. DIALING CIRCUITRY

The number to be dialled is relayed to the exchange by one of the following two signalling methods.

- Pulse dialling (Loop-Disconnect dialling)
- Tone dialling (Dual Tone Multi Frequency)

- **PULSE DIALING**

In this type of dialling, the digits are send by connecting and disconnecting the local exchange accesses line, or “loop”. The rotating dial produces a series of pulsed interruptions in the current flow. Opening the loop circuit interrupts the loop current flow of 20-120 mA and closing the circuit permits the loop current to flow again. The pulses are used to indicate the digits of the destination number, one pulse for a digit of value ‘1’, two pulses for value ‘2’ etc, up to ten pulses for the digit ‘0’. The pulses are repeated at a frequency of ten per second. Between each digit a longer gap of 700 milliseconds is left, in order that the exchange may differentiate between the sequences of pulses representing consecutive digits of the

overall numbers. The gap is called the" inter digit pause (IDP)". The IDP may range from 600- 900 Milliseconds depending on system (Fig 2.6). Dial pulsing requires a 60 milliseconds break and a 40 milliseconds make period for each dial pulse for a total time of 100 milliseconds per dial pulse.

- **TONE DIALING**

In the tone dialling each digit is represented by a combination of two pure frequency tones. There is a low frequency tone for each row and high frequency tones for each column, hence the name dual. The use of two tones in combination reduces the risk of mis-operation if other interfering noises are present in the line. These telephone sets are equipped with a push-button keypad with twelve keys which represent number 0- and the symbols * and # (fig. 2.7).

- **TIME COMPARISON OF PULSE DIALING AND TONE DIALING**

DTMF Dialling is much faster in principle and practice than dial pulsing. It also reduces the post dialling delay. Using DTMF, the time required to recognize any digit tone is only 50 milliseconds with an inter-digit pause of another 50 milliseconds. Thus, the total time to send any digit is about 100 milliseconds. In contrast, dial pulsing requires a 60 milliseconds break and a 40 milliseconds make period for each dial pulse for a total time of 100 milliseconds per dial pulse. Thus, for dial pulsing, each higher digit number requires more time because the number of pulses per digit increases. Also, the inter digit interval is of about 700 milliseconds.

For example, by using the number 555-555-5555, an average time for pulse dialling a long distance call can be obtained as follows:

5 pulses per digit x 100 ms per pulse x 10 digits= 5 seconds

Inter digit interval x (number of digits-1) = 700 ms x 9= 6.3 seconds

Total time for dial pulsing = 5 + 6.3 = 11.3 seconds

DTMF dialling for the same number takes:

Number of digits x 100ms per digit = 10 x 100 ms =1 second

- **ADVANTAGES OF TONE DIALING**

  - Tone dialling is gradually replacing pulse dialling because it:
  - Decreases dialling time
  - Uses solid state electronic circuits
  - Can be used for end-to-end signalling after the call is connected (low speed data transmission)
  - Reduces local exchange equipment requirements
  - Is more compatible with electronically (stored program) controlled exchanges.

2.2.1.g. LOOP LENGTH COMPENSATION

The telephone circuit works adequately at short line length, but if too long a line were to be used, then because the electrical resistance of the microphone would be substantially smaller then the resistance of the line itself, the microphone would not be able to stimulate much fluctuation in the circuit current. The result is faint sound reproduction. To overcome this, modern telephone sets have automatic compensation circuits. For this a varistor is inserted in the circuit to automatically by pass excess current flow around the transmitter, regulating current flow for a wide range of telephone line lengths. Thus, the varistor automatically adjusts the speech level so that a relatively constant speech level will appear at the exchange regardless of the distance to the telephone.

2.2.1.h. HYBRID CIRCUIT (INDUCTION COIL)

The hybrid is a multiple winding transformer. The heart of the hybrid system is a multiple winding transformer. By electromagnetic coupling, signals are transferred between windings. Where coupling results in opposing fields, signal cancellation occurs. Thus, two super imposed AC signals can be recovered (Fig 2.8). The function of a hybrid is to interface a 2-wire circuit to 4-wire circuit to permit full duplex operation. 2-wire circuits are used for the millions subscriber local loop because they are cheaper. 4-wire circuits, which have 2-wires for each direction, are used for all other circuits in the network. A hybrid is used at the central office to interface the local loop to trunks and between trunks at some older offices where only 2-wire switching is available.

2.2.l.i. LINE BALANCING NETWORK

A signal from the microphone is delivered to the telephone line after amplification by a factor based upon the gain and the ratio of decoupling of the hybrid. The imbalance of the hybrid results in a fraction of this signal being fed through the receiver amplifier, to the receiver. It is called the side tone. The electrical signal to the receiver is converted into the sound and fed to the user’s ear. However, a fraction of this acoustical signal will be fed back to the microphone via the air. This acoustical feedback can be positive and add to the original signal to cause howling if the level of the feedback is high enough. Line balancing networks are used to reduce this acoustical feedback and this sort of equalization allows the speech network to work well on local loops of any length (fig 2.9).

2.2.2. CENTRAL OFFICE

A telephone exchange (central office) is a switched telecommunication link used to connect different telephone terminals with each other, to form an cc and efficient system, winch enables a subscriber to communicate with one another. Main functions performed by a telephone exchange are:

- To advise the calling subscriber when the equipment is ready for him to dial.
- To ring up the telephone of the called subscriber, indicating to him that some other subscriber wants to talk with him.
- Giving an indication to caller that the required telephone is ringing by connection of ring back tone to the caller.
- If called subscriber picks up the telephone (goes off hook), then making communication link between the called person and the caller.
- To return a distinctive signal to the calling party if the required line is engaged (Connecting the engage tone).
- To continuously monitor the line and clear the line when call is finished.

2.2.3. LOCAL LOOP

Each subscriber telephone is connected to a central office that contains switching equipment, signalling equipment, and batteries that supply direct current to operate the telephone as shown in figure 2.10. Each phone is connected to the central office through a local loop of two wires called a wire pair. One of these wires is called a T (for Tip) and the other is called R (for Ring), which refers to the Tip and Ring part of the plug used in the early manual switchboards.

Switches in the central office respond to the dial pulses or tones from the telephone to connect the calling phone to the called phone. When the connection is established, the two telephones communicate over transformer-coupled loops using the current supplied by the central office batteries.

2.2.4. TRUNK

Functionally, the communication channels between switching systems are referred to as trunks. These channels are implemented with a variety of facilities including: pairs of wire, coaxial cables, point to point microwave radio links; and most recently optical fibers. These facilities are almost exclusively implemented with dedicated wire pairs. These wire pairs are commonly referred to as station loops, subscriber loops, or customer loops.

2.2.5. CHANNEL BANDWIDTH REQUIREMENTS

The telephone circuits are designed to pass a limited bandwidth. This permits the transmission of the voice frequencies and limits unwanted circuit noises. In order, to eliminate unwanted signals (noise) that could disturb conversations or cause errors in control signals, the circuits that carry the telephone signals are designed to pass only certain frequencies. The ranges of frequencies that are passed are said to be in the pass band. For a telephone system voice channel (a VF channel) the pass band is 0-4000 Hz. Sometimes this band is called a message channel. The Bandwidth is the difference between the upper and lower limits of the pass band; thus, the bandwidth of the VF channel is 4000 Hz. However, not the entire VF channel is used for the transmission of speech. The voice pass band is restricted to 300-3000 Hz and is known as in-band signal as shown in Figure 2.11.

2.3. CALL SETUP

The establishment of a physical connection across a circuit switched network relies not only upon the availability of a appropriate topology of exchanges and transmission links between the two end points, but also upon the correct functioning of a logical ‘call setup’ and ‘clear down’ procedures.

2.3.1. INI’IIATING A CALL

2.3.1.a. OFF HOOK DETECTION

When the handset of the telephone is resting in its cradle, weight of the handset holds the switch hook buttons down and the switches are open. This is called the on-hook condition. The circuit between the telephone handset and the central office is open; however, the ringer circuit in the telephone is always connected to the central office. The ringer circuit presents high impedance to speech signals so it has no effect on them.

When the handset is removed from its cradle, the spring-loaded buttons come up and the switch hook closes. This completes the circuit to the exchange and the current flows in the circuit. This is called the off-hook condition. The off-hook signal tells the exchange that someone wants to make a call and action needs to be taken.

2.3.1.b. CALLING LINE IDENTITY

On receipt of the off-hook signal the exchange has to establish what is called the calling line identity, i.e. which particular telephone of the many connected to the exchange has generated the signal. The exchange’s control system has to know this in order to identify which access line ‘termination’ requires onward cross-connection. This information also serves to monitor customer's network usage, and shows how much to charge them.

2.3.1.c. CODE RECEIVER AND REGISTER ALLOCATION

After identifying the calling line, the exchange's next job is to allocate and connect equipment, ready for the receipt of dialled digits from the customer. This equipment normally consists of two main parts, the 'code receiver', which recognizes the values of the digits dialled, and the 'register', which stores the received digit values, ready for analysis.

2.3.1.d. DIAL TONE

Once the exchange has prepared code receivers and a register, it announces its readiness to receive digits, and prompts the customer to dial the directory number of the desired destination. This it does by applying dial tone, which is the familiar noise heard by customers on lifting the handset to their ear. The whole sequence of events (the 'off-hook' signal, the preparation of code-receiver and register, and the return of dial tone) is normally almost instantaneous; the customer usually hears dial tone before the earphone reaches the ear. Noticeable delays occur in exchanges where there are insufficient code receivers and registers to meet the call demand. The remedy lies in providing more of them.

2.3.2. SENDING A NUMBER

On hearing the dial tone, the customer dials the directory number of the desired destination. There are two signalling systems by which the digit values of the number may be indicated to the exchange: 'Loop Disconnect dialling' and 'Multi Frequency Signalling (DTMF)'.

2.3.3. NUMBER ANALYSIS

Number analysis is the process by which the exchange is able to determine the appropriate onward routing for the call, and the charge per minute to be levied for the call. During the digit analysis, the exchange compares the dialled number held by the register with its own list of permitted numbers. The permitted numbers are held permanently in routing tables with in the exchange. The routing tables give the exchange numbers identity of the outgoing rout required to reach the ultimate destination.

2.3.4. RINGING THE CALLED PHONE

The central office has various switches and relays that automatically connect the calling and the called customers. Once the connection has been completed, ringing tone will be sent to ring the destination telephone, and simultaneously ring back tone is sent to the calling customer.

2.3.5. ANSWERING THE CALL

As soon as the called customer answers the telephone by lifting the handset, an answer signal is transmitted back along the connection. This has the effect of tripping the ringing and ring back tone, and commencing the process of charging the customer for his call.

2.3.6. CONVERSATION

Conversation or the equivalent phase of communication may continue for as long as required until the calling customer replaces the telephone handset to signal the end of the call.

2.3.7. CLEAR DOWN SEQUENCE

When the calling customer replaces the telephone handset to end the call, a clear signal is generated and it act in the reverse manner to the off-hook signal, breaking the access line loop. This signal is passed to each exchange along the connection, releasing all the equipments and terminating the call charging.

2.4. UNSUCCESSFUL CALLS

A call is known as a 'successful call', when the caller gets through the sequence of events and a conversation follows. But our calls do not always succeed, and when a call fails, perhaps because of network congestion, or because the called party is busy or fails to answer, the network has to tell the caller what has happened, and then it has to clear the connection so as to free the network for more fruitful use.

When it is a case of network congestion or called customer busy, the caller usually hears either a standard 'advisory tone', or a recorded announcement of the form 'all lines are busy—please try later', which tells that an invalid number has been dialled.

A caller who hears one of the call unsuccessful advisory announcements, or a prolonged ringing tone, usually gives up and clears the connection by replacing the handset, to try again later. When, however, the caller fails to do this, the network has to 'force' the release of the connection. 'Forced release', if needed, is put in hand between 1 and 3 minutes after the call has been dialled, and it is initiated if there has been no reply from the called party during this period, whatever the reason. Forced release once initiated—normally by the originating exchange even though the handset of the calling telephone is left off-hook—forces the calling telephone into a ‘number obtainable’ or ‘park’ condition. To normalize the condition of a telephone, the handset must be returned to the cradle.

Release is also forced in a number of other ‘abnormal’ circumstances, as when a calling party fails to respond to dial tone (by dialling digits), or when too few digits are dialled to make up a valid number. In such cases, the calling subscriber may be entirely disconnected from the local exchange, silencing even its dial tone. This has the advantage of freeing code receivers, registers, and other common equipment for more worthwhile use on other customers’ calls. The calling customer who is a victim of forced release may hear either silence or number unobtainable’ tone. To restore dial tone, a new ‘off-hook’, signal must be generated by replacing the handset and then lifting it off again.

It is clearly important for exchanges to be capable of ‘forced release’ so that their common equipment is not unnecessarily ‘locked up’ by a backlog of unsuccessful calls.

2.5. DEVELOPMENT OF A TELEPHONE SYSTEM

Following steps are involved in the development of a simple telephone system.

STEP #1: A BASIC (SIMPLEX) TELECOMMUNICATION SYSTEM

The physical elements of a simple telecommunication system include the transmitter, the receiver and the transport mechanism as shown in fig 2.12 (a).

Transmitter transforms and encodes the information in such a way to produce the electromagnetic signals that can be transmitted over some sort of transport mechanism. The transmitter in case of telephony is the microphone, a part of the telephone hand set into which the speaker speaks. It converts the voice

signals (20Hz-20KHz) into a form that can be transmitted over a transport medium.

The transport mechanism may be a single transmission line or complex network connecting the source and destination, which is a telephone line in this particular case.

The receiver accepts the signal from the transmission system and converts it into a form that can be handled by a destination device. An earphone serves this purpose in a simple telecommunication circuit as shown in fig 2. 12 b.

STEP #2: A DUPLEX TELECOMMUNICATION SYSTEM

A general duplex telecommunication system is shown in fig 2. 13.a. This system is capable of transmitting information in both directions. The corresponding particular circuit that connects the equivalent of two complete telephone handsets (capable of two way communication) is shown in fig 2.13 (b). The use of transformer reduces the effect of line resistance and increases the signal strength.

STEP #3: PROVISION OF ANTI SIDE-TONE & ECHO CANCELLER CIRCUIT

Side-tone is the acoustical signal resulting from a portion of the transmitted voice signal being coupled back to the receiver of the same handset. An anti-side tone circuit is normally incorporated into the telephony circuitry to control the level of side tone. It does so by restricting the volume of speech and other noises picked up by the microphone from being heard in the speaker’s own earphone. The anti-side tone circuit also eases the difficulty of hearing weak incoming signals. This circuit has the further benefit that it controls the undesirable effect of echo.

A simple anti-side tone circuit is shown in fig 2. 14. In this circuit, if the impedance Z is made equal to the impedance of the line and the balanced transformers have equal impedances of the windings T1 and T2 then the current I1 flowing in the winding T1 induces an emf in the winding T3 in one direction and the current I2 flowing in the winding T2 induces an emf in the other direction so that the net emf in the winding T3 is zero and hence no signal in the winding T3. Since there is no signal in the winding T3 so there is no sound in the receiver and hence no side tone.

STEP #4: CENTRAL BATTERY SYSTEM

The common (or central) battery system is normally used to meet the power requirements of customer's handsets. A large battery, or power system, is positioned in each exchange and provides the necessary power for all customers' lines terminated at that exchange. The use of a CB system has significant benefits in simplifying the job of maintenance in preference to individual line batteries.

A central battery configuration is shown in fig 2.15. Note that a high impedance coil has been added to the circuit. This forms part of another normal circuit feature called the 'transmission bridge'. Without the transmission bridge coil the speech currents from one telephone would be shorted out by the battery and not heard in the receiving telephone. An inductive 'coil' prevents this 'shorting'. A high resistance can be used but this has the disadvantage of causing conversation 'overhearing' in all the other telephones connected to the same central battery.

In addition to these facilities, overload protection, polarity protection, number dialling mechanisms are also provided to form a complete telephone network.

2.6. TYPES OF EXCHANGES

In general, the telephone uses two types of exchange mechanisms:

1. Manual Control
2. Automatic Control

2.6.1 MANUAL CONTROL

If an operator at a manual type switchboard affects the desired switching manually, the exchange is known as Private Manual Branch Exchange (P.M.B.X).

Early telephone switchboards were operated manually using a jack for each line and two plugs on a long flexible wire, called a cord pair for making the connection as shown in fig 2.16. The cord pairs appeared in rows on a shelf in front of the operator, and the jacks (called line appearances) were mounted on a vertical panel. To make a connection, the operator picked up a cord, plugged it into the jack corresponding to the line requesting service, obtained from the calling party the name or number of the desired party, then plugged the other end of the cord pair into the correct outgoing line jack. There are many thousands of cord switchboards still in operation, providing the versatility and ease of programming of the control system.

2.6.2. AUTOMATIC CONTROL

If the desired switching is accomplished automatically, the exchange is known as Private Automatic Branch Exchange (P.A.B.X).

Automatic exchanges can be further divided into 4 categories:

- Strowger Exchange (Progressive Control)
- Crossbar Exchange

- Reed relay Exchange
- Digital Exchange

2.6.2.a. STROWGER EXCHANGE

If the desired switching relies on a progressive step-by-step operation of several series switches operating in tandem, the exchange is known as a progressive or Strowger exchange.

The Strowger exchange responds to the dial pulses generated by the calling telephone. Fig 2.17 represents a simplified schematic diagram of such an exchange. The telephone line shown in fig is actually a pair of wires. When the calling telephone goes off hook, current flowing in the local loop operates a relay in the exchange, causing the first switch in the train (the line feeder) to search for the active line by stepping vertically until the vertical contact is connected to the off hook line. The line feeder then steps horizontally until it finds a first selector that is not use on another call. This is the next switch in the train. When a free first selector is connected, a dial tone is returned to the calling party. The first selector switch waits for the first digit to be dialled, and then steps vertically one step for each dial pulse received. When it has taken in one digit, it steps horizontally until a free second selector is found, and the process is repeated. Thus, the first switch in the train (the line feeder) takes in no digit, the second and third takes in one digit each, and the last switch in the train (called a connector) takes in the final two digits. A 10,000-line exchange requires four digits to be dialled (0000 through 9999) and requires four switches for each connected call (line feeder, first selector, second selector and connector).

The Strowger switching system has the following significant limitations:

- Since several switches are operated in tandem and the switches (except for the first one) are shared among many incoming lines, it is possible for a call

- to become blocked part way through the dialling sequence, even though the called line is free.

- It is not possible to use tone-dialling (DTMF) telephones directly. They may be used if the central office is equipped with a conversion device.

- The switch requires the successful sequential (step-by-step, time related) operation of several relays and a sizable voltage and current is switched each time a switch is stepped. Consequently, the mechanical reliability of the switches is low, they require large amounts of maintenance by skilled people, and they generate large amounts of electrical and mechanical noise.

- Since the switching network is hard-wired, it is difficult to make changes in the switching arrangement.

2.6.2.b. CROSS BAR EXCHANGE

Cross bar, as the name implies, depends on the crossing or intersection of two points of a switching matrix being managed by an external common control to make a connection. The switching matrix (also known as cross point array) is shown in fig 2.18. Its operation depends on energizing a vertical line and a horizontal line and the point where they intersect represents the connection made. Therefore, any one of the input lines can be connected to any one of the output lines by energizing a particular input line and a particular output line. The cross bar matrix is controlled by common control. Control signals from transmission lines are detected and used to control the matrix to connect the proper lines for the path from the calling telephone to the called telephone.

2.6.2.c. REED RELAY EXCHANGE

The reed relays exchanges make use of the reed relays to make the connections. The reed relay is a small glass-encapsulated, electromechanical switching device as shown in fig 2.19. These devices are actuated by a common control which selects the relays to be closed in response to the number dialled, and sends pulses through coils wound around the relay capsules. The pulses change the polarity of magnetization of plates of magnetic material fitted alongside the glass capsules. The contacts open or close in response to the direction of magnetization of the plates, which is controlled by the positive or negative direction of the pulse sent through the windings. Since the contacts latch, no holding current is required for this type of cross point, but separate action is required by the common control to release the connection (unlatch or reset the relay) when one party or the other hangs up.

Reed relays have improved the reliability and maintainability of switches. Cross bar switches still provide much of the switching for long distance or long haul telephone calls. In addition, reed relays are an important part of stored program control led electronic switching systems.

All of the step-by-step, cross bar and reed relay switching is called space division switching because each telephone conversation is assigned a separate physical path through the telephone system.

2.6.2.d. DIGITAL EXCHANGE

A digital exchange is the most efficient and most advanced version of exchange system. In digital exchange, voice is converted into digital form and this signal from different users is Time Division Multiplexed on a single line. The circuit switching is performed by microprocessor by controlling the digital switching circuit. The multiplexed data is demultiplexed and passing through D/A converter is fed to the desired user. The digital exchanges are always

recommended for large number of lines, as they offer cost effective solution for that situation. However for small number of lines their selection does not offer cost effective solution because they have three different cards for switching, controller and speech path. For this reason, the electronic exchanges are the best option for smaller number of lines.

CHAPTER 3

EPABX - HARDWARE DESIGN

Communication is one of the basic instincts of every intelligent life form. Modern scientific world owes its existence to different and better modes of communication evolved during the process of human civilization. Nowadays, the progress of country depends upon the methods and speed with which information is spread amongst its masses. Most of the data and speech is transferred via telephone lines, thus making it the most important media of concern.

Normally the requirement of telephone facilities is limited in residences, small shops and certain small-scale businesses. In such cases a single telephone instrument is usually enough to meet the communication requirement. In somewhat larger organizations, it may be desirable to make the exchange line available to several different persons by the provision of extensions, which are conveniently located in the various offices. In still larger businesses organizations, the volume of telephone traffic may justify the provision of a number of lines to the public exchange, and these lines must be accessible to a larger number of telephones using staff. Moreover, as a size of the firm increases, it becomes more and more desirable to provide telephone facility between the various departments in addition to the normal exchange line services. The switching of a number of extension points and the provision of communication facilities between these extension points, required the provision of some more or less complex switching arrangement at the subscriber’s location. If the desired switching is accomplished by means of an automatic switching plant, the installation is known as Electronic Private Automatic Branch Exchange (EPABX).

The provision of automatic equipment at the subscriber premises does off course; materially increase the capital cost of the installation. Automatic equipment also requires more maintenance. The annual charges of the equipment at the subscriber's premises are, however, comparatively small when compared with the charges for the external line plant.

PABX installation can be broadly classified as:

- **UNATTENDED EPABX**

In this case, there is no manual switchboard at the subscriber's premises. Calls between extensions are completed automatically on the dialling basis through the local automatic equipment. Similarly, extensions can obtain outward calls via the public exchange by the dialling a special code that gives access to the exchange line. The equipment can be designed so that incoming calls are automatically routed to one or more selected extensions and these extensions can, if necessary transfer the call to other extension by dialling the required local number.

- **ATTENDED EPABX**

In this case, a manual switchboard is provided for receiving incoming calls to the subscribers. The attendant can then extend the call to any desired extension either through the local automatic equipment or by means of plugs and cords. In some cases all outgoing calls from the extensions to the public exchange may be routed through the EPABX operator while in some other cases, the subscriber may wish to provide automatic access to the public exchange for local calls but to filter all outgoing trunk calls to the EPABX operator.

We have implemented an unattended EPABX. The PC along with certain hardware circuitry will serve this purpose. In an unattended exchange, calls between various extensions are completed automatically on a dialling basis through the local automatic equipment. The equipment can be designed so that the incoming calls are automatically routed to one or more selected extensions, and these extensions can, if necessary, transfer the calls to other extensions by dialling the required local number.

3.1. SALIENT FEATURES

The salient features of our exchange are:

- Number of subscribers: 4
- Number of trunks: 1
- Dialling mechanism: DTMF
- Direct Inward Station Access (DISA): To present the line status information to a subscriber accessing a limited availability line exchange, DISA is used, which plays recorded messages like

  "Welcome, if you know the extension of the person, you would like to contact, dial now or wait for the operator's assistance".

  "All lines are busy, please call later".

  "The number you dialled is not listed please call again"

- Call transfer facility: Each extension has been provided with the facility to transfer the call to another extension.
- Billing and charging facility: Once the loop between the calling and the called party is complete, the software monitors the customers' network usage to determine, how much to charge him.
- Music facility: During the handover from trunk to required extension of the calling party music will be played.
- Software: The software of the exchange is written in Turbo C++.

3.2. BLOCK DIAGRAM

The block diagram of the exchange implemented by us is shown in figure 3.1 Block diagram is composed of the following blocks.

- Block # 1: Control Module

- ♦ Block # 2: Address Decoding Module
- ♦ Block # 3: Communication Interface
- ♦ Block # 4: Switching Module
- ♦ Block # 5: Extension Module
- ♦ Block # 6: Number Analysis Module
- ♦ Block # 7: Interrupt and signalling Module
- ♦ Block#8:Trunk
- ♦ Block # 9: Additional Facilities

3.3. EXPLANATION OF BLOCK DIAGRAM

3.3.1. BLOCK # 1: CONTROL MODULE

This module is the brain of the exchange and is a general-purpose computer that runs the EPABX. The PC is interfaced to the exchange by IBM prototype card that is inserted into 8-bit ISA slot of the PC. When a telephone is picked up or a terminal powered on, the control module gets an interrupt from the appropriate line module. The control unit then connects the digit of the number called, and sets up the switch to create a conversation channel between calling and the called party and monitor the call in progress. The IBM card is also used to buffer the data bus and the two least significant address bits. It also provides other control signals from ISA bus to the Exchange such as I/O read, I/O Write, power supply and reset etc. The control module also provides call-billing facility to this stored program control (SPC) exchange.

3.3.2. BLOCK #2: ADDRESS DECODING MODULE

In order to interface devices to the microprocessor it is necessary to decode the addresses to create partition of the memory map so as to enable one IC amongst several connected ICs. This removes the severe problem of data collision, which would have taken place if all the ICs sent their data simultaneously on the data bus. The outputs of the ISA slot, on the motherboard of the PC, along with a special purpose PAL performs the address decoding tasks of our exchange.

3.3.2.a. ISA BUS DESCRIPTION

The Industry Standard Architecture (ISA) is a 62-pin slot on the motherboard, which controls the flow of information to and from the PC. Any card from the very first PC will plug into and function in any of the most modem Pentium Pro based computers. We plugged our exchange card into 8-bit ISA slot of PC. Fig 3.2 illustrates the 8-bit ISA connector found on the main board of all PCs.

3.3.2.b. ISA PIN CONFIGURATION

DATA BUS (D0-D7): It is a bus, which controls the data flow between PC and other modules of the exchange pin # 2-9 (on component side of ISA).

CONTROL SIGNALS

I/OW: It is a Memory Write signal, which is connected to WR control pin of PAL, 8255 and 8253.

I/OR: It is a Memory Read signal, which is connected to RD Control pin of PAL, 8255 and 8253.

Address Enable (AEN): It is connected to pin # I of Programmable Array Logic (PAL), which is a 20 pin IC.

Interrupt Request (IRQ 3): It is input from pin#10 of 8253(Timer).

ADDRESS BUS (A2_ A9): Pin # 22_ 28 on the component side of ISA bus are connected to pin # 4_ 9 of PAL, which are used for address decoding.

3.3.2.c. PAL DESCRIPTION

PAL belongs to PLD. PLD stands for Programmable Logic Device. Logic designers have a wide range of ICs available to them with numerous logic functions and logic circuits arrangements on a chip. In addition, these ICs are available from many manufacturers at a reasonable cost. For these reasons, designers have been interconnecting standard ICs to form an almost endless variety of different circuits and systems and will continue to do so in definite future.

PAL actually stands for Programmable Array Logic. PAL consists of AND and OR arrays. In PAL the inputs to the AND gate are programmable while inputs to the OR gate are hard wired. This means that every AND gate can be programmed to generate any desired product of four input variable and their complements. The PAL used by us is PAL 16L8, which is a 20 pin IC. The description of each IC pin is shown in table 3.1. It has 10 fixed inputs, two fixed outputs and 6 pins that are programmable as inputs or outputs. Each output pin is generated from a seven input OR gate that has an AND gate attached to each input. The outputs of the OR gates pass through a three-state inverter that defines each out as an AND/NOR function. Initially all the fuses connect the AND and OR array, programming is accomplished by blowing the fuses. The wired AND function is performed at each input connection that allows a product term of up to l6 inputs. A logic expression using the PAL 16L8 can have seven product terms with 16 inputs NORed together to generate the output expression.

3.3.2.d. REASON FOR USING PAL

There are some problems with circuit and system design that use only standard ICs. Some system designs might require thousands of these ICs. This large number of ICs requires a considerable amount of circuit board space, and a great deal of time and the cost in inserting, soldering and testing of ICs. Reducing

PIN NUMBER	PIN DESCRIPTION
1	AEN
2	IOR
3	IOW
4	A_2
5	A_3
6	A_4
7	A_5
8	A_6
9	A_7
10	GND
11	A_8
12	S_6
13	A_9
14	S_1
15	S_2
16	S_3
17	S_4
18	S_5
19	NC
20	V_{CC}

Table 3.1. Pin Description of PAL

the number of ICs used in a design can have several advantages, less board space, fewer circuit boards and smaller enclosure, low power requirement, faster and less costly assembly, high reliability, since there are fewer ICs and circuit connections that can fail and easier troubleshooting.

3.3.2.e. PROGRAMMING OF PAL

Address bus coming from the ISA slot has several address lines among which only 10 address lines are used, two of them are connected directly to 8255A and 8253. The remaining 8 lines are inserted in a PAL for addressing decoding, to select one IC at a time. The remaining three inputs of the PAL are fed from the address enable AEN, IOR and IOW of the ISA slot. 5 outputs of PAL are connected to the chip select of various ICs. It also contains sixth output which is the OR operation of all other 5 outputs, this output is used for buffer enable and connected to the buffer lC74245. Therefore at a specific address, given by the computer, one of the IC is selected. The programming of PAL is accomplished with special equipment called Universal Segup Programmer. A program is written in this equipment and the fuses of PAL are burned for the required function. The program of PAL designing is given below:

PROGRAM

CHIP DECODER PAL 16L8

Pins:	1	2	3	4	5	6	7	8	9	10
	AEN	IOR	IOW	A2	A3	A4	A5	A6	A7	GND
Pins	11	12	13	14	15	16	17	18	19	20
	Ag	S	A	S	S	S	S	S	NC	Vcc

/S1= /AEN */IOR */A2 */A3 */A4 */A5 */A6 */A7 *A8 *A9 +

/AEN */IOW */A2 */A3 */A4 */A5 */A6 */A7 *A8 *A9

/S2= /AEN */IOR *A2 */A3 */A4 */A5 */A6 */A7 *A8 *A9 +

/AEN */IOW *A2 */A3 */A4 */A5 */A6 */A7 *A8 *A9

/S3= /AEN */IOR */A2 *A3 */A4 */A5 */A6 */A7 *A8 *A9 +

/AEN */IOW */A2 *A3 */A4 */A5 */A6 */A7 *A8 *A9

/S4= /AEN */IOR *A2 *A3 */A4 */A5 */A6 */A7 *A8 *A9 +

/AEN */IOW *A2 *A3 */A4 */A5 */A6 */A7 *A8 *A9

/S5= /AEN */IOR */A2 */A3 *A4 */A5 */A6 */A7 *A8 *A9 +

/AEN */IOW */A2 */A3 *A4 */A5 */A6 */A7 *A8 *A9

/S6= /S1 +/S2 +/S3 +/S4 +/S5

3.3.3. BLOCK #3: COMMUNICATION INTERFACE

The interconnection or linkage of the parts within a microprocessor system is referred to as interfacing. It includes synchronization, direction of data transmission, and sometimes the adjustment of the signal levels or modes. Speed of the CPU is high as compared to other equipment used, so to achieve compatibility between CPU and exchange ICs. 8255 is used as an interface; it latches data from the computer to the exchange and vice versa.

3.33.a. FEATURES

- MCS- 85 TM compatible 8255A-5
- 24 programmable 1/0 pins
- Completely TTL Compatible
- Fully compatible with Intel Microprocessor families
- Improved timing characteristics
- Direct bit Set/Reset capability easing control application interface
- Improved DC driving capability
- 40-pin DIP package

3.3.3.b. DESCRIPTION

The Intel 8255A is a general-purpose programmable I/O device designed for use with Intel microprocessors. Its function is that of a general purpose I/O component to interface peripheral equipment to the microcomputer system bus. The system software programs the functional configuration of the 8255A so that normally no external logic is necessary to interface peripheral devices or structures. It has 24 1/0 pins, which may be individually programmed in 2 groups of 12 and used in 3 major modes of operation. In the first mode (MODE 0), each group of 12 I/0 pins may be programmed in sets of 4 to be input or output. In MODE 1, the second mode, each group, may be programmed to have 8 lines of input or output. Of the remaining 4 pins, 3 are used for handshaking and interrupt control signals. The third mode of operation (MODE 2) is a bi-directional bus mode that uses 8 lines for a bi-directional bus, and 5 lines, borrowing one from the other group, for handshaking.

3.3.3.c. PIN DIAGRAM & FUNCTIONS

The pin diagram of Programmable Peripheral Interface (PPI) 8255A is shown in fig 3.3. Table 3.2 is an explanation of the function of each IC pin.

DATA BUS BUFFER

This 3-state bi-directional 8-bit buffer is used to interface the 8255A to the system data bus. Data is transmitted or received by the buffer upon execution of input or output instruction by the CPU. Control words and status information are also transferred through the data bus buffer.

READ/WRITE AND CONTROL LOGIC

The function of this block is to manage all of the internal and external transfers of both data and control or status words. It accepts inputs from the CPU address and Control busses and in turn, issues commands to both of the Control Groups.

CHIP SELECT (CS')

A "low" on this input pin enables the communication between 8255A and the CPU.

READ (RD')

A '1ow" on this input pin enables the 8255A to send the data or status information to the CPU on the data bus. In essence, it allows the CPU to "read from" the 8255A.

PIN NAME	**PIN DESCRIPTION**
D_7- D_0	**DATA BUS (BI- DIRECTIONAL)**
RESET	**RESET INPUT**
CS	**CHIP SELECT**
RD	**READ INPUT**
WR	**WRITE INPUT**
A0, A1	**PORT ADDRESS**
PA7- PA0	**PORT A (BIT)**
PB7- PB0	**PORT B (BIT)**
PC7- PC0	**PORT C (BIT)**
V_{CC}	**+5 VOLTS**
GND	**0 VOLTS**

Table 3.2. Pin Description of 8255A

WRITE (WR)

A “low” on this input pin enables the CPU to write data or control words into the 8255A.

PORT SELECT 0 AND PORT SELECT 1: (A0 and A1)

These input signals, in conjunction with the RD and WR inputs, control the selection of one of the three ports or the control word registers. They are normally connected to the least significant bits of the address bus (A0 and A1). The basic operation of 8255A under the effect of these inputs is shown in table 3.3.

RESET

A “high” on this input clears the control register and all ports (A, B, C) are set to the input mode.

GROUP A AND GROUP B CONTROLS

The system software programs the functional configuration of each port. In essence, the CPU “outputs” a control word to the 8255A. The control word contains information such as mode “bit set”, “bit reset”, etc, that initializes the functional configuration of the 8255A.

Each of the control blocks (Group A and Group B) accepts “commands” from the Read/Write Control Logic, receives “control words” from the internal data bus and issues the proper commands to its associated ports.

Control Group A----Port A and Port C upper (C7---C4)

Control Group B----Port B and Port C lower (C3---CO)

The Control Word Register can only be written into. No read operation of the Control Word Register is allowed.

A_1	A_0	RD	WR	CS	INPUT OPERATION (READ)
0	0	0	1	0	PORT A→ DATA BUS
0	1	0	1	0	PORT B→ DATA BUS
1	0	0	1	0	PORT C→ DATA BUS
					OUTPUT OPERATION (WRITE)
0	0	1	0	0	DATA BUS→ PORT A
0	1	1	0	0	DATA BUS→ PORT B
1	0	1	0	0	DATA BUS→ PORT C
1	1	1	0	0	DATA BUS→ CON
					DISABLE FUNCTION
X	X	X	X	1	DATA BUS→ 3-STATE
1	1	1	1	0	ILLEGAL CONDITION
X	X	1	1	0	DATA BUS→ 3-STATE

Table 3.3. Basic Operation of 8255A

PORTS A, B AND C

The 8255A contains three 8-bit ports (A, B and C). All can be configured in a wide variety of functional characteristics by the system software but each has its own special features or “personality” to further enhanced the power and flexibility of the 8255A.

PORT A: One 8-bit data output latch/buffer and one 8-bit data input latch.

PORT B: One 8-bit data input/output latch/buffer and one 8-bit data input buffer.

PORT C: One 8-bit data output latch/buffer and one 8-bit data input buffer (no latch for input). This port can be divided into two 4-bit ports under the mode control. Each 4-bit port contains a 4-bit latch and it can be used for the control signal outputs and status signal inputs in conjunction with ports A and B.

For our project port A was used as the input port and B and C as output ports. To initialize this configuration, control word 90H was used.

D_7	D_6	D_5	D_4	D_3	D_2	D_1	D_0
1	**0**	**0**	**1**	**0**	**0**	**0**	**0**

3.3.3.d., REASON FOR USING 8255A

The 8255A PPI is a very popular low-cost interfacing component found in many applications. The 8255A can interface any TTL comp 110 devices to the microprocessor. Because I/O devices are inherently slow, wait states used during I/O transfers do not impact significantly upon the speed of the system. Amongst many available ICs, the 8255A is particularly chosen because it is fully compatible

with the most modem Intel’s Pentium microprocessors. According to our requirement, it has three output ports, each of 8-bit that could be programmed either output or input ports. Although, our requirement was individual bit manipulation at the output and input ports, a feasible option was there to employ more expensive but simple in use " bit addressable latches" but we have achieved the same objective by software masking using this IC at a reasonable cost.

3.3.4. BLOCK #4: SWITCHING MODULE

Telephone networks are circuit switched networks characterized by the existence of a physical path, or a circuit between its points of origin and its destination, throughout the establishment of the call. Three particular attributes needed in all circuit switched exchanges are:

- The ability not only to establish and maintain (or 'hold') a physical connection between the 'caller' and the 'called party' for the duration of the call but also to disconnect (CLEAR) it afterwards.

- The ability to connect any circuit carrying an incoming call (a so-called 'incoming) circuit to one of a multitude of other (so-called 'outgoing') circuits. Particularly important is the ability to select different outgoing circuits when subsequent calls are made from the same incoming circuit. During the set-up period of each call the exchange must determine which outgoing circuit is required usually by extracting it from the dialled number. This makes it possible to put through calls to a number of other network users.

- The ability to prevent new calls intruding into circuits, which are already in use. To avoid this the new call must either be diverted to an alternative circuit, or it must temporarily be denied access in which case the caller will hear a 'busy' or 'engaged' tone.

For these reasons, exchanges are usually designed as an "array" or "matrix" of "switched cross-points" known as a cross-point array. The cross-point array used by us is MITEL MT8816.

3.3.4.a. FEATURES

- Internal control latches and address decoder
- Short set-up and hold times
- Wide operating voltage: 4.5V to 13.2V
- 12Vpp analog signal capability
- R_{ON} 65 Ω max. V =12V, 25°C
- $\Delta R_{ON} \leq 10\Omega$ @ V_{DD} =12V, 25°C
- Full CMOS switch for low distortion
- Minimum feed through and cross talk
- Separate analog and digital reference supplies
- Low power consumption ISO-CMOS technology

3.3.4.b. DESCRIPTION

The MITEL MT8816 is fabricated in MITEL's ISO-CMOS technology providing low power dissipation and high reliability. The device contains an 8 x 16 array of cross point switches along with a 7 to 128 line decoder and latch circuits. Any one of the 128 switches can be addressed by selecting the appropriate seven address bits. The selected switch can be turned on or off by applying a logical one or zero to the DATA input. Vss is the ground reference of the digital inputs. The range of the analog signal is from V_{DD} to V_{EE} Chip Select (CS) allows the cross point array to be cascaded for matrix expansion.

3.3.4.c. PIN DIAGRAM & FUNCTIONS

Pin diagram of MT8816 is shown in fig 3.4 along with the pin description table 3.4. The MT8816 is an analog switch matrix with an array size of 8 x 16. The switch array is arranged such that there are 8 columns by 16 rows. The columns are referred to as the Y inputs/outputs and the rows are the X inputs/outputs. The cross point analog switch array will interconnect any X I/O with any Y I/O when turned on and provide a high degree of isolation when turned off. The control memory consists of a 128-bit write only RAM in which the bits are selected by the address inputs (AYO-AY2, AXO-AX3). Data is presented to the memory on the DATA input. Data is asynchronously written into memory whenever both the CS (Chip Select) and STROBE inputs are high and are latched on the falling edge of STROBE. A logical “1” written into a memory cell turns the corresponding cross point switch on and a logical “0” turns the cross point off. Only the cross point switches corresponding to the addressed memory location are altered when data is written into memory. The remaining switches retain their previous states. Establishing appropriate patterns in the control memory can interconnect any combination of X and Y inputs/outputs. A logical “1” on the RESET input will asynchronously return all memory locations to logical “0” turning off all cross point switches regardless of whether CS is high or low. Two voltage reference pins (V_{SS} and V_{EE}) are provided for the MT8816 to enable switching of negative analog signals. The range for digital signals is from V_{DD} to V_{SS} while the range for analog signals is from V_{DD} to V_{EE}. V_{SS} and V_{EE} pins can be tied together if a single voltage reference is needed.

ADDRESS DECODE

The seven address inputs along with the STROBE and CS (Chip Select) are logically ANDed to form an enable signal for the reset able transparent latches. The DATA input is buffered and is used as the input to all latches. To write to a location, RESET must be low and CS must go high while the address and data are set up. Then the STROBE input is set high and then low causing the data to be

PIN NO	NAME	DESCRIPTION
1	Y3	Y3 Analog (Input/ Output): this is connected to Y3 column of the switch array.
2	AY2	Y2 address line (Input)
3	RESET	Master Reset (Input): this is used to turn off all switches regardless of the condition of CS: High
4, 5	AX3, AX0	X3 and X0 Address Lines (Inputs)
6,7	X14, X15	X14 and X15 Analog (Inputs/ Outputs): these are connected to X14 and X15 rows of switch array.
8- 13	X6- X11	X6- X11 Analog (Inputs/ Outputs): these are connected to X6- X11 rows of the switch array.
14	NC	No Connection
15	Y7	Y7 Analog (Input/ Output): this is connected to Y7 column of the switch array.
16	V_{SS}	Digital Ground Reference
17	Y6	Y6 Analog (Input/ Output): this is connected to Y6 column of the switch array.
18	STROBE	STROBE (Input): enables function selected by address and data. (Active High)
19	Y5	Y5 Analog (Input/ Output): connected to Y5 column of the switch array.
20	V_{EE}	Negative power supply.
21	Y4	Y4 Analog (Input/ Output): connected to Y4 column of the switch array.

PIN NO	NAME	DESCRIPTION
22, 23	AX1, AX2	X1 and X2 Address Lines (Inputs)
24, 25	AY0, AY1	Y0 and Y1 Address Lines (Inputs)
26, 27	X13, X12	X13 and X12 Analog (Inputs/ Outputs): connected to X13 and X12 rows of the switch array.
28- 33	X5- X0	X5- X0 Analog (Inputs/ Outputs): connected to X5- X0 rows of the switch array.
34	NC	No Connection.
35	Y0	Y0 Analog (Input/ Output): connected to Y0 column of the switch array.
36	CS	Chip Select (Input): used to select the device: Active High
37	Y1	Y1 Analog (Input/ Output): connected to Y1 column of the switch array.
38	DATA	Data (Input): a logic high input will turn on the selected switch and a logic low will turn off the selected switch. Active High.
39	Y2	Y2 Analog (Input/ Output): this is connected to Y2 column of the switch array.
40	V_{DD}	Positive Power Supply.

Table 3.4. Pin Description of MT8816

latched. The data can be changed while STROBE is high, however, the corresponding switch will turn on and off in accordance with the DATA input. DATA must be stable on the falling edge of STROBE in order for correct data to be written to the latch.

The three tones (dial tone, ring tone and ring back tone) are applied on X6, X7 and X8. Music 1C output is given to X5. Pins X12 and X13 are used for DISA connection. At X12 is the microphone and the speaker is at X13. The trunk connection is at Y3.

3.3.4.d. REASON FOR USING MT8816

The MITEL MT8816 provides low power dissipation and high reliability. It has built in latches and decoder. The latches store the data and software controls the switching of appropriate cross points. The total number of switches needed is 45. 9 links are needed for connection to the rows (three for the three DTMFs' junctors, three for various tones; two for DISA and one for the Music IC) and 5 links for connection to the columns (4 for SL1Cs' junctors and one for the trunk). Other ICs provided either too less or too many a number of switches leading to inefficiency or the wastage of resources.

3.3.5. BLOCK #5: EXTENSION MODULE

The extension module provides a complete interface between the telephone line and the switch via a subscriber line interface circuit, commonly known as a SLIC. The basic functions of a SLIC are usually described by using the acronym "BORSCH" which stands for Battery, Over voltage ringing, Signalling, Coding and Hybrid (two-two wire). The SLIC used in our project is MITEL MH88500 (20 pin IC), which interconnects the telephone line to the speech switch requiring only single bi-directional switch or cross points. Each extension needs a SLIC; hence four SLICs are used for four extensions in this project.

3.3.5.a. FEATURES

- Differential to single ended conversion
- No transformers required
- Minimum installation space
- Off hook detection and LED indicator drive
- Relay drive output
- Battery and ringing feed to line
- Logic interface: MUTE’, OFHK’, RC
- Mute of incoming audio
- Dial pulse detection
- Voltage surge protection

3.3.5.b. DESCRIPTION

The MITEL MH88500 Subscriber Line Interface Circuit provides a complete interface between the telephone line and a speech switch requiring only single bi—directional switch per cross point. The functions provided by the MH88500 include bi-directional differential to single ended conversion in the speech path, line battery feed, ringing feed and loop and dial pulse detection. The device is fabricated as a thick film hybrid in a 20-pin single in line’ package allowing optimum circuit board packing density.

3.3.5.c. PIN DIAGRAM & FUNCTIONS

The pin diagram of, MH88500, the SLIC used by us, is shown in figure 3.5. The pin description table is given as Table 3.5.

PIN NO	NAME	DESCRIPTION
1	**TIP**	Tip Lead. Connects to the Tip lead of the tel wire.
2	**V_{A1}**	Positive line feed supply voltage. Normally connected to V_{A2}
3	**RING**	Ring Lead. Connected to ring lead of the tel line.
4	**RING FEED**	Negative line feed voltage and ringing input. Normally connected to ring relay.
5	**IC**	Internal connection. Leave open cct. Use for testing only.
6	**V_{C1}**	Sense Input. Normally connected to negative line feed voltage supply.
7	**GND**	Analog Ground. Internally connected to pin 13.
8	**V_B**	Negative Analog Voltage Supply.
9	**LED**	LED Drive Output. Drives LED directly. Off hook condition, logic low.
10	**OFHK**	Logic low output. Indicates closed loop condition.
11	**THRESH ADJ**	Allows adjustment of OFHK detection threshold.

12	**V_{A2}**	Positive power supply voltage. Connected to V_{A1}.
13	**GND**	Analog Ground. Internally connected to pin 7.
14	**V_{C2}**	Loop detector voltage supply. Connected to negative line feed voltage supply.
15	**MUTE**	Input mutes the incoming audio. Active low.
16	**JUNCTOR**	Receive/ Transmit audio speech path.
17	**RD**	Relay drive output.
18	**RGND**	Ground for relay drive cct.
19	**RC**	Ring control input. Active high.
20	**CD**	Clamping diode. Connected to relay positive voltage.

Table 3.5. Pin Description of MH88500

SPEECH CIRCUIT

Like speech circuit converts the bi—directional TIP and RING line pair to a bi-directional single ended junctor line. Fig 3.6a illustrates a typical connection between two SLICs through two cross point switches. This configuration gives optimum Tran hybrid loss as seen from fig 3.6b given that the output impedance of the junctor line is 604Ω.

The 'MUTE' input mutes signals coming from TIP and RING to the junctor line while allowing the signal from the junctor to the tip-ring pair to be transmitted.

LOOP DETECTION

The loop detection circuit determines whether low enough impedance is across TIP and RING to be recognized as an off hook condition. (Threshold impedance=5.4KΩ with no adjustment). This threshold level can be adjusted by the use of external resistors. 'OFHK' has low output drive capability so it may drive CMOS operating with different power supplies.

LINE FEED/RING FEED CIRCUIT

The line feed circuit provides loop current and the ability to apply ringing onto TIP and RING. The impedance from RING FEED to GND is 600Ω that give the loop current as:

I_L = (Voltage at RING FEED pin)/(Telephone impedance + 600)

The positive supply for the line feed circuit is V_{A1} though the loop current is determined from RING FEED and GND.

RELAY DRIVE CIRCUIT

The relay drive circuit switches ringing on RING FEED. The diode is present to suppress voltage transients during relay switching caused by the inductive coils of the relay. Ringing voltage includes AC ringing (90V typically and DC line feed voltage (-24V typically).

3.3.5.d. REASON FOR USING MH88500

We have used the M1TEL MH88500 SL1C because it provides a complete interface between the telephone line and a speech switch requiring only single bi directional switch per cross point. Moreover, this IC provides the additional function of line battery feed. It asserts voltage on the loop to alert the subscriber of an incoming call. It is also able to detect whether the telephone is off hook or a dialling is performed. A protective circuit is also provided to save internal circuitry against short circuit. SLIC also ensures a constant feed current to the line. The SLIC provides a transformer less hybrid circuit. Circuits using transformers for hybrid conversion are expensive, bulky and occupy more space. SLICs serve the same purpose resulting in a smaller and compact system.

3.3.6. BLOCK # 6: NUMBER ANALYSIS MODULE

Having identified the calling line, the exchange’s lob is to allocate and connect equipment ready for the receipt of dial digits from the customer. This equipment normally consists of two main parts the code receiver, which recognizes the values of the digits dialled, and the register that stores the received digit values ready for analysis, a process known as digit analysis. By digit analysis the exchange is able to determine the appropriate onward routing for the call and the charge per minute to be levied for the call. During digit analysis, the exchange compares the dialled number held by the register with its own list of permitted numbers.

There are two prevalent signalling systems by which the digit values of the number may be indicated to the exchange; ‘loop disconnect’ and ‘Dual Tone Multi Frequency (DTMF)’ signalling. We adopt the latter being more accurate and time effective, so the receiver used is M1TEL 8870 (18 pin IC). Three 8870 ICs are used to provide call transfer as well as call waiting facility. As a result three junctors are available, one from each 8870. These junctors are connected to X9,

X 10 and X11 of 8816 respectively.

3.3.6.a. FEATURES

- -3.6volt operation
- Complete DTMF receiver
- Low power consumption
- Internal gain setting amplifier
- Adjustable guard time
- Central office quality
- Power-down mode
- Inhibit mode
- Functionally compatible with M1TEL’s MT887OD

3.3.6.b. DESCRIPTION

The MT88L70 is a complete 3-Volt, DTMF receiver integrating both the band split filter and digital decoder functions. The filter section uses switched capacitor techniques for high and low group filters; the decoder uses digital counting techniques to detect and decode all 16 DTMF tone-pairs into a 4-bit code.

External component count is minimized by on chip provision of a differential input amplifier, clock oscillator and latched three-state bus interface.

Separation between the low-input and high group tones is achieved by applying the DTMF signal to the inputs of the two sixth-order switched capacitor band-pass filters, the bandwidths of which correspond to the low and high group frequencies. When the detector recognizes the presence of two valid tones, the ‘Early Steering (ESt)’ output will go to an active state. The number latching on the output of the decoder (Q1, Q2, Q3, Q4) is indicated by change of state of the Std pin from high to low. The three Std pins of 8870 are connected to pins PA6, PA5, PA7 (port A) of 8255. 8255 detects this change of state and informs it to the computer. The duration of this high-to-low transition of the 8870 is controlled by RC combination in such a way that it enables the 8255 to respond to this state change during Interrupt Service Routine (ISR) execution. The control module enables the CS of 8870 via the PAL and reads the number. It then performs call routing operation accordingly.

3.3.6. c. PIN DIAGRAM & FUNCTIONS

Pin diagram and pin function description of the IC MT88L70 is shown in figure 3.7 and table 3.6a respectively.

The MT88L70 monolithic DTMF receiver offers small size, low power consumption and high performance, with 3-volt operation. Its architecture consists of a band split filter section, which separates the high and low group tones, followed by a digital counting section that verifies the frequency and duration of the received tones before passing the corresponding code to the output bus.

PIN NO	NAME	DESCRIPTION
1	**IN+**	Non- inverting Op- Amp (Input).
2	**IN-**	Inverting Op- Amp (Input).
3	**GS**	Gain select. Gives access to output of front- end differential amplifier for connection of feed back resistor.
4	**V_{REF}**	Reference voltage (Output). Normally $V_{DD}/2$ is used to bias inputs at mid- rail.
5	**INH**	Inhibit (Input). Logic high inhibits the detection of tones representing characters A, B, C and D. This pin is internally pulled down.
6	**PWDN**	Power down (Input). Active High. Power down the device and inhabits the oscillator. This pin input is internally pulled down.
7	**OSC1**	Clock (Input).
8	**OSC2**	Clock (Output). A 3.579545 MHZ crystal connected between pins OSC1 and OSC2 completes the internal oscillator cct.
9	**V_{SS}**	Ground.

10	**TOE**	Three state output enable (Input). Logic high enables the outputs Q1- Q4. This pin is pulled up internally.
11- 14	**Q1- Q4**	Three state data (Output). When enabled by TOE, provide the code corresponding to the last valid tone- pair received. When TOE is logic low, the data outputs are high impedence.
15	**StD**	Delayed steering (Output). Presents a logic high when a received tone pair has been registered and output latch updated; returns to logic low when the voltage on St/ GT falls below V_{TSt}.
16	**ESt**	Early steering (Output). Presents logic high once the digital algorithm has detected a valid tone pair. Any momentary loss of signal condition will cause Est to return to logic low.
17	**St/ GT**	Steering Input/ Guard time (Output) Bi- directional. A voltage greater than V_{TSt} detected at St causes the device to register the detected tone pair and update the output latch.
15	**V_{DD}**	Positive power supply (Input). +3V typical.

Table 3.6.a Pin Description Table of MT88L70

FILTER SECTION

Separation of the low-group and high group tones is achieved by applying the DTMF signal to the inputs of two sixth-order switched capacitor band pass filters, the bandwidths of which correspond to the low and high group frequencies. The filter section also incorporates notches at 350 and 440 Hz for exceptional dial tone rejection. A single order switched capacitor filter section, which smoothes the signals prior to limiting follows each filter output. Limiting is performed by high-gain comparators, which are provided with hysteresis to prevent detection of unwanted low-level signals. The outputs of the comparators provide full rail logic swings at the frequencies of the incoming DTMF signals.

DECODER SECTION

Following the filter section is a decoder employing digital counting techniques to determine the frequencies of the incoming tones and to verify that they correspond to standard DTMF frequencies. A complex averaging algorithm protects against tone simulation by extraneous signals such as voice while providing tolerance to small frequency deviations and variations. This averaging algorithm has been developed to ensure an optimum combination of immunity to talk-off and tolerance to the presence of interfering frequencies (third tones) and noise. When the detector recognizes the presence of two valid tones (this is referred to as the “signal condition” in some industry specifications) the “Early Steering” (ESt) output will go to an active state. Any subsequent loss of signal condition will cause ESt to assume an inactive state (see “Steering Circuit” Fig 3. 8a).

STEERING CIRCUIT

Before registration of a decoded tone pair, the receiver checks for a valid signal duration (referred to as character recognition condition). This check is performed by an external RC time constant driven by ESt. Logic high on ESt causes V_C (see Figure 3.8a) to raise as the capacitor discharges. Provided signal condition is maintained (ESt remains high) for the validation period t_{GTP} , V_C reaches the threshold V_{TSt} of the steering logic to register the tone pair, latching its corresponding 4-bit code (see Table 3.6b) into the output latch. At this point the GT output is activated and drives V_C to V_{DD}. GT continues to drive high as long as ESt remains high. Finally, after a short delay to allow the output latch to settle, the delayed steering output flag (Std) goes high, signalling that a received tone pair has been registered. The contents of the output latch are made available on the 4-bit output bus by raising the three-state control input (TOE) to logic high.

The steering circuit works in reverse to validate the inter-digit pause between signals. Thus, as well as rejecting signals too short to be considered valid, the receiver will tolerate signal interruptions (dropout) too short to be considered a valid pause. This facility, together with the capability of selecting the steering time constants externally, allows the designer to tailor performance to meet a wide variety of system requirements.

GUARD TIME ADJUSTMENT

In many situations not requiring selection of tone duration and inter digit pause, the simple steering circuit shown in Figure 3.8a is applicable. Component values are chosen according to the formula:

$$t_{REC} = t_{DP} + t_{GTP}$$

$$t_{ID} = t_{DA} + t_{GTA}$$

Digit	TOE	INH	ESt	Q4	Q3	Q2	Q1
ANY	L	X	H	Z	Z	Z	Z
1	H	X	H	0	0	0	1
2	H	X	H	0	0	1	0
3	H	X	H	0	0	1	1
4	H	X	H	0	1	0	0
5	H	X	H	0	1	0	1
6	H	X	H	0	1	1	0
7	H	X	H	0	1	1	1
8	H	X	H	1	0	0	0
9	H	X	H	1	0	0	1
0	H	X	H	1	0	1	0
*	H	X	H	1	0	1	1
#	H	X	H	1	1	0	0
A	H	L	H	1	1	0	1
B	H	L	H	1	1	1	0
C	H	L	H	1	1	1	1
D	H	L	H	0	0	0	0
A	H	H	L	Undetected , the output code will remain the same as the previous detected code.			
A	H	H	L				
A	H	H	L				
A	H	H	L				

Table 3.6.b Functional Decode Table of MT 88L70

The value of t_{DP} is a device parameter and t_{REC} is the minimum signal duration to be recognized by the receiver. A value of C of 0.01μf is recommended for most applications, leaving R to be selected by the designer.

CRYSTAL OSCILLATOR

The internal clock circuit is completed with the addition of an external 3.579545 MHz crystal and is connected as shown in Figure 3.8b (Single-ended Input Configuration).

3.3.6.d. REASON FOR USING MT88L70

Our basic requirement for the exchange is a DTMF receiver, which is fulfilled by the use of MT88L70. This IC effectively latches the number given on the data bus and detects that whether the number has been latched or not. There are several ICs available serving the same purpose but these ICs are transceivers (containing both the transmitter and receivers). Employing these ICs will be wastage of resources.

This IC efficiently meets system specifications which place both accept and reject limits on both tone duration and inter digit pause. Guard time adjustment allows the designer to tailor system parameter such as talk off and noise immunity. Increasing the minimum signal duration to be recognized by the receiver t_{REC} improves talk off performance, since it produces the probability that tones stimulated by speech will maintain signal conditions long enough to be registered. Alternatively, a relatively short t_{REC} with a long inter digit pause reject time t_{DO} would be appropriate for extremely noisy environments, where acquisition time and immunity to tone drop outs fast are required.

3.3.7. BLOCK#7: INTERRUPT AND SIGNALLING MODULE

Interrupt is a method of informing the microprocessor that an I/O device is ready and action needs to be taken. The interrupt signal causes the microprocessor unit to finish executing the current instruction, suspend the normal operation and jump to a special group of instructions in its monitor program that handles the data input. The special subroutine that performs this action is called interrupt service routine (ISR). The addresses of ISR are located in an interrupt vector table.

We have used Intel's 8253 (Programmable interrupt Timer) in our project to interrupt the microprocessor after every 50 milliseconds and for the generation of various tones.

3.3.7.a. FEATURES

- Compatible with all Intel and most other microprocessors
- Handles inputs from DC to 10 MHz
- Status read back command
- Six programmable counter modes
- Three independent 16-bit counters
- Binary or BCD counting
- Single +5V supply

3.3.7.b. DESCRIPTION

The Intel 8253 is a programmable interval counter/timer device designed to solve the common timing control problems in microcomputer system design. It is a general purpose, multi-timing element that can be treated as an array of I/0 ports in the system software. It provides three independent 16-bit counters, each capable of handling clock inputs up to 10 MHz. All modes are software programmable. This IC uses HMOS technology and comes in a 24-pin plastic package.

The 8253 solve one of the most common problems in any microcomputer system, the generation of accurate time delays under the software control. Instead of setting up timing loops in software, the programmer configures the 8253 to match his requirements and programs one of the counters for the desired delay. After the desired delay, the 8253 will interrupt the CPU. It can also be used for the implementation of several other functions as: real time clock, event counter, digital one-shot, programmable rate generator, square wave generator, binary rate multiplier etc.

3.3.7.c. PIN DIAGRAM & FUNCTIONS

The pin diagram of Intel 8253 is shown in fig 3.9 and description of each pin is given in table 3.7.

BLOCK DIAGRAM

Block diagram showing the various functions of 8253 is given in fig 3.10.

DATA BUS BUFFER

This 3-state bi-directional, 8-bit buffer is used to interface the 8253 to the system bus.

READ/WRITE LOGIC

The Read/Write logic accepts inputs from the system bus and generates control signals for the other functional blocks of 8253. A1 and A0 select one of the three counters or the Control Word Register to be read from/written into. A “low” on the ‘RD’ input tells the 8253 that the CPU is reading one of the counters. A “low” on the ‘WR’ input tells the 8253 that the CPU is writing either a Control Word or an initial count. CS’ qualifies both ‘RD’ and ‘WR’. ‘RD’ and ‘WR’ are ignored unless the 8253 have been selected by holding ‘CS’ low.

SYMBOL	PIN NO	TYPE	DESCRIPTION
D7-D0	1-8	I/O	Data: Bi-directional three state data bus lines, connected to system data bus.
CLK0	9	I	Clock 0: Clock input of counter 0.
OUT0	10	O	Output 0: Output of counter 0.
GATE0	11	I	Gate 0: Gate input of counter 0.
GND	12		GROUND: Power supply connection
V_{CC}	24		POWER: +5 V power supply connection
WR'	23	I	WRITE CONTROL: This input is low during CPU writing operations.
RD'	22	I	READ CONTROL: This input is low during CPU read operations.
CS'	21	I	CHIP SELECT: A low on this input enables the 8253 to respond to RD' and WR' signals.
A1, A0	20-19	I	ADDRESS: Used to select one of three counters or the control word register for read or write operations.
CLK2	18	I	Clock 2: Clock input of counter 2.
OUT 2	17	O	Output 2: Output of counter 2.
GATE 2	16	I	Gate 2: Gate input of counter 2.
CLK 1	15	I	Clock 1: Clock input of counter 1.
GATE 1	14	I	Gate 1: Gate input of counter 1.
OUT1	13	O	Output 1: Output of counter 1.

Table 3.7: Pin Description Table of INTEL 8253

CONTROL WORD REGISTER

The Control word Register as shown in fig 3.11a is selected by the Read /Write Logic when A1, A0= 11. If the CPU then does a write operation to the 8253, the data is stored in the control word register and is interpreted as a control word used to define the operation of the counters. The control word register can only be written to status information is available with the Read-Back Command.

COUNTER 0, COUNTER 1, COUNTER 2

These three functional blocks are identical in operation but fully independent. Each counter may operate in a different mode. The control word register is not the part of counter itself, but its contents determine how the counter operates. The status register when latched contains the current contents of the control word register and status of the output and input flag.

The counter 2 is used for interrupting the microprocessor, whereas the counter1 is used for tone generation, in our project.

8253 SYSTEM INTERFACE

The 8253IC is a component of the Intel microcomputer system and interfaces in the same manner as all other peripherals of the family. It is treated by the system’s software as an array of peripheral I/O ports; three are counters and the fourth is a control register for MODE programming. Basically, the select inputs A0, A1 connect to the A0, A1 address bus signals of the CPU. The CS can be derived directly from the address bus using a linear select method or it can be connected to the output of a decoder, such as an Intel 8205 for larger systems.

PROGRAMMING THE 8253

After power up, the state of 8253 along with the mode, count value and outputs of all counters are undefined. Each counter must be programmed before it can be used. Writing a control word and then an initial count programs counters. The control words are written into the control word register which is selected when A1, A0= 11. The control word itself specifies which counter is being programmed. By contrast, initial counts are written into the counters, not the control word register. The A1, A0 inputs are used to select the counter to be written into.

In order to program 8253, we have used a software command given as:

outportb(CONT_WORD_ 8254, 0x76)

D_7	D_6	D_5	D_4	D_3	D_2	D_1	D_0
SC1	**SC0**	**RW1**	**RW0**	**M2**	**M1**	**M0**	**BCD**
0	**1**	**1**	**1**	**0**	**1**	**1**	**0**

This performs the following functions:

SC1, SC0= 0, 1	Selects Counter 1
RW1, RW0= 1,1	Read/ Write Least Significant Byte first, then Most Significant Byte.
M2, M1, M0= 0,1,1	Mode 3 (Square Wave Mode)
BCD=0	Binary Counter (16 bits)

Square wave mode, as selected by the control word, is typically used for baud rate generation. This mode functions like a divide by N counter. The Duty Cycle of OUT is such that it is high for half a cycle. When half the initial count has expired, OUT goes low for the remainder of the count. This mode is periodic: the sequence is repeated indefinitely resulting in a square wave with a period of N CLK cycles. We have used a crystal of 1843 MHz connected between pin number 15 and 18. it is required to generate a tone of 455 kHz., dividing 1843M by 455k yields 4050 which is sent as D2H (LSB) and OFH (MSB). The commands given for this purpose are

```
outportb(COUNTER ,0xd2)

outportb(COUNTER I ,0x0f)
```

3.3.7.d. REASON FOR USING INTEL 8253

Intel 8253 programmable interval timer is used to generate the dial tone and interrupt signal. Although the 555 timer can be used to perform the same function but the reason for using this particular IC is that it contains three counter cum timers, which are software programmable. The 555 timer IC is hardware controllable. The requirement of our project is to use two timers. The Intel 8253 contains three built in timers cum counters. So, only one IC is used — a task that would have been accomplished by the use of two 555 timers otherwise.

3.3.8. BLOCK #8: TRUNK

Functionally, the communication channels between switching systems are referred to as trunks. These channels are implemented with a variety of facilities including: pairs of wire, coaxial cables, point to point microwave radio 1inks and most recently optical fibers. These facilities are almost exclusively implemented with dedicated wire pairs. These wire pairs are commonly referred to as station loops, subscriber loops, or customer loops.

3.3.8.a. FUNCTIONAL BLOCK DIAGRAM & EXPLANATION

The functional block diagram of trunk is shown in fig 3.12. It is divided into three blocks:

- Control Block
- Hybrid Block
- Gain Block

CONTROL BLOCK

The control block incorporates both the ring detection and the off hook detection circuitry. Its operation can be realized by considering the following explanation.

The TIP and RING from any other exchange (PTCL in our case) is connected to the hybrid transformer through a RC combination. Under on hook condition, the hybrid transformer is isolated from the TIP/RING wire pair through a large I0kΩ resistor. This large value of R does not allow any current to pass, resulting in an open circuit. Since the loop is not complete, it indicates an on hook condition.

The PTCL alerts the local exchange (PC in our case) of a call by sending a voltage of 90 V rms. A blocking capacitor of 1μF blocks any DC component present in the ringing tone and applies the resultant to the input of the bridge rectifier. The function of the bridge rectifier is to convert the incoming AC to a raw DC. This pulsating DC is applied to an optocoupler. The optocoupler is a combination of an LED 4N26 and a silicon phototransistor 4N25, which are electrically isolated but optically coupled. The pulsating DC input is applied to the

LED and output is taken from the phototransistor. The optocoupler converts this DC into +5 V and GND to be detected by the computer.

Under off hook condition, logic high is provided by the local exchange, as a result, the control transistor saturates, energizing the relay, hence bypassing the RC combination. The loop completes, current flows and an off hook condition is detected.

HYBRID BLOCK

The hybrid block consists of a hybrid transformer and two operational amplifiers.

The basic function of the hybrid transformer is to convert the TIP and RING on the incoming wire pair into speech signals at the junctor and vice versa.

The operational amplifiers are arranged in such a way that one is used for transmission and other for reception. These op-amps are provided with resistances whose values can be adjusted to control the gain.

MUTE

A combination of a transistor and a relay is used to mute the incoming signals from the TIP/RING wire pair. We have implemented this function in our Exchange to provide music & DISA to the calling subscriber by completely isolating the incoming signals via the mute pin.

3.3.8.b. REASON FOR USING TH1SCIRCUIT

Instead of using this complicated trunk circuit, an option is to use the IC MH88510. Main problem in using this IC is that its gain is not controllable. Hence, signal strength cannot be altered for intelligible conversation, if it is found to be too weak or too loud. The trunk circuit removes this problem effectively.

3.3.9. BLOCK #9: ADDITIONAL FACILITIES

Additional facilities supplied in our exchange include the DISA and the provision of music during waiting periods.

3.3.9.a. DIRECT INWARD STATION ACCESS (DISA)

To present the line status information to a subscriber accessing a limited availability line exchange, DISA is used, which plays the recorded messages like

1. "Welcome, if you know the extension of the person, you would like to contact, dial now or wait for the operator's assistance".
2. "All lines are busy, please call later".
3. "The number you dialled is not listed please call again"

Two ICs provide direct inward station access.

1. ISD 2560 (Single chip voice record/Play back device)
2. DTMF receiver (MT88L70)

3.3.9.b. ISD 2560 (Single chip voice record/Play back device)

FEATURES

The main features of the IC ISD 2560 (Single chip voice record! Play back device) are as follows:

- ♦ Easy to use single chip voice record/play back solution
- ♦ High quality, natural voice/audio reproduction
- ♦ Manual switch or micro controller compatible play back can be edge- or level activated.
- ♦ Single chip duration of 60 seconds
- ♦ Automatic power down (Push- button mode)
    - Standby current 0.5µA
- ♦ Zero power message storage

- Eliminates battery backup circuits
- ♦ Fully addressable to handle multiple messages
- ♦ 100-year message retention
- ♦ 100,000 record cycle
- ♦ On chip clock source
- ♦ Programmable support for play- only application
- ♦ Single +5 V power supply

DESCRIPTION

Information storage device ISD 2560 Chip Coder provides high quality, single chip and record/playback solution to short duration messaging applications. This CMOS device includes an on chip oscillator, microphone preamplifier, automatic gain control, antialiasing filter, smoothing filter and speaker amplifier, and high-density multilevel storage array.

Recordings are stored in on chip non-volatile memory cells, providing zero power message storage. Voice and audio signals are stored directly into memory in their natural form providing high quality, solid-state voice reproduction.

ISD 2560 is operated at 8KHz sampling frequency. The speech samples are stored directly into on chip non-volatile memory with out the digitisation and compression. Direct analog storage provides a very true, natural sounding reproduction of voice, music, tone and sound effects not available with most solid-state digital solutions.

♦ **EEPROM STORAGE**

One of the benefits of ISD’s Chip coder technology is the use of on chip non-volatile memory, providing zero power message storage. The message is retained for up to hundred years typically without power. In addition, the device can be re- recorded typically over 100,000 times.

PIN DIAGRAM AND FUNCTIONS

Pin diagram of ISD 2560 is shown in figure 3.13.

VOLTAGE INPUTS (V_{CCA}, V_{CCD})

Analog and digital circuits in the ISD 2560 use separate power buses to minimize noise on the chip. These power buses are brought out to separate pins on the package and should be tied together as close to the supply as possible. In addition, these supplies should be decoupled as close to the package as possible.

GROUND INPUT (V_{SSA}, V_{SSD})

The ISD2500 series of devices utilizes separate analog and digital ground buses to minimize noise. These pins should be connected separately through a low impendence path to power supply ground.

POWER DOWN INPUT (PD)

When not recording or playing back, the PD pin should be pulled high to place the part in a very low power mode.

CHIP ENABLE INPUT (CE)

The CE pin is taken low to enable all play back and record operations. The address inputs and play back/ record input (P/R) are latched by the falling edge of CE.

PLAYBACK/RECORD INPUT (P/R)

The P/R input is latched by the falling edge of the CE pin. A high level selects a playback cycle while a low level selects a record cycle. For a record cycle, the address inputs provide the starting address and recording continues until PD or CE is pulled high or an overflow is detected (i.e. the chip is full). When pulling PD or CE high terminates a record cycle, an end of message marker

(EOM) is stored at the current address in memory. For a Playback cycle, the address inputs provide the starting address and the device will play until an EOM marker is encountered

MICROPHONE INPUT (MIC)

The microphone input transfers its signal to the on-chip preamplifier. An on-chip automatic gain control (AGC) circuit controls the gain of this preamplifier from -15 to 24 dB. An external microphone should be AC coupled to this pin via a series capacitor. The capacitor value, together with the internal 10KΩ resistance on this pin, determines the low frequency cut-off.

MICROPHONE REFERENCE (MIC REF)

The MIC REF input is the inverting input to the microphone preamplifier. This provides a noise cancelling or common mode rejection input to the device when connected differentially to a microphone.

AUTOMATIC GAIN CONTROL (AGC)

The AGC dynamically adjusts the gain of the preamplifier to compensate for the wide range of microphone input levels. The AGC allows the full range of sound, from whispers to loud sounds, to be recorded with minimal distortion. The 'attack' time is determined by the time constant of a 5KΩ internal resistance and an external capacitor connected from the AGC pin to V_{SSA} analog ground. The 'release' time is determined by the time constant of an external resistor and an external capacitor connected in parallel between the AGC pin and V_{SSA} analog ground. Nominal values of 470KΩ and 4.7μf give satisfactory results in most cases.

ANALOG OUTPUT (ANA OUT)

This pin provides the preamplifier output to the user. The voltage level at the AGC pin determines the voltage gain of the preamplifier.

ANALOG INPUT (ANA IN)

The ANA IN pin transfers the input signal to the chip for recording. For microphone inputs, the ANA OUT pin should be connected via an external capacitor to the ANA IN pin. This capacitor value, together with the 3KΩ input impedance of ANA IN, is selected to give additional cut-off at the low frequency end of the voice pass band. If the desired input is derived from a source other than a microphone the signal can be fed, capacitive coupled into the ANA IN pin directly.

EXTERNAL CLOCK INPUT (XCLK)

The external clock input for the ISD 2560 has an internal pull-down device. These devices are configured at the factory with an internal sampling clock frequency that guarantees its minimum nominal record playback time of 60 seconds. The sampling frequency is then maintained to a variation of ±2.25% over the commercial temperature and operating voltage ranges, while still maintaining the minimum specified recording duration. The internal clock has a ±5% tolerance over the industrial temperature and voltage range. If XCLK is not used, this input should be connected to ground.

SPEAKER OUTPUTS (SP+, SP-)

The device includes an on chip differential speaker driver, capable of driving 50mW into 16Ω from AUX IN. The speaker outputs are held at V_{SSA} levels during record and power down. A single output may be used alone (including a coupling capacitor between SP pin and the speaker).

ADDRESS/ MODE INPUTS (AX/MX)

The address/mode inputs have two functions, depending upon the level of the two most significant bits (MSB) of the address(A8 and A9). If either or both of the two MSBs are LOW, the inputs are all interpreted as address bits and are used as start address for the current record or playback cycle. The address pins are inputs only and do not output internal address information as the operation progresses. Address inputs are latched by the falling edge of CE. If both MSBs are high, the Address/ Mode inputs are interpreted as Mode bits.

The second IC providing the direct inward station access in conjunction with single chip record/play back device is MT88L70 (DTMF receiver), which is already explained above.

3.3.9.c. MUSIC IC

The music IC M66T is a three pin IC, which behaves like an oscillator, in order to generate the non-uniform waveform (sounds like music). The output is a weak signal, which is then passed through an operational amplifier having a gain of 10. This strengthened output is then applied to X5 of MT8816, so that music can be played to the calling subscriber during wait periods.

CHAPTER 4

EPABX - SOFTWARE DESIGN

The algorithm of our exchange is implemented in Turbo C++. Turbo C+- is an object oriented programming language, which offers greater flexibility and programming power, while offering a diverse set of tools designed to suit every programming style. The Standard C++ Library is a comprehensive framework of classes and functions as defined by the International Standards Organization (ISO) and the American National Standards Institute (ANSI) that provides enormous convenience while programming.

The concept of 'less hardware and extensive software' has been implied to make more compact, low cost and easily maintainable system in comparison to another commercial systems. In software, an unusual way of programming "Implementation of State Diagram through Software Flow Chart" has been adopted to create structured and modular software, in which debugging the problem becomes easy. This technique requires all the states to be programmed, but during one scan only five states are executed (four of extensions and fifth of trunk), hence reducing the run time. It is very important to divide all the states into almost equal time slices of the time slot available, so that the state processor does not remain in one state so long, that it cannot sense the changes occurring in other states of different telephone and trunk. In this fashion hardware and software is integrated to make a functional exchange.

4.1. STATE DIAGRAM

State diagram is a standard way of solving the digital circuit problems. In this technique, hardware states are defined through which a system can go. In our case, we have defined different states of extensions and trunk, due to which software has become large, but the portion of the program that will be executed in one single processing block is very small. For example a total of 12 states for each telephone and 12 for trunk are defined, but in one single loop the states which are executed are only 5, resulting in a small run time of program.

4.2. VARIOUS STATES OF OUR EXCHANGE

The function of our exchange can be aptly studied by studying its state diagram. State diagram representation of a system is analogous to the software flow charts, the only difference being that in the former, operation of the exchange is explained in terms of various states of the system.

4.2.1. STATE DIAGRAM OF EXTENSION

The extension's state diagram of our project is shown in figure 4.1. The functions performed during each state are listed below.

4.2.l.a. STATE 1

In state 1 the system continuously monitors the off hook pin of each extension. If any extension goes off hook then dial tone and DTMF receiver is connected to it and its busy flag is set, and its state is changed to state 2, otherwise the state is not changed.

4.2.l.b. STATE 2

In state 2, the system checks whether any extension has dialed a number or not. If dialed, dial tone and DTMF receiver is disconnected and the control is transferred to various cases of the state 2. All the cases analyze the dialed number. If the dialed number is busy, the case switches the state 2 to state 9. If the dialed number is not busy the control is transferred to the respective states (states 3,4 and 5) of each extension and respective extension's busy flag is high. If the dialed number is not found to be valid after analysis (i.e. the number is not listed) the state is changed to

state 11. If extension requires the trunk, system off hooks the trunk and connects the junctor of both lines and transfers the control to state 12.

4.2.1.c. STATE 3,4,5

These states generate ring on the respective extension and ring back tone to the calling party. It also monitors the called party's off hook pin in order to assign it the stale 10 and connects the speech path of two speakers and transfers the control to state 6,7,8 respectively.

4.2.1.d. STATE 6,7,8

In these states, the system monitors the off hook of the calling party. When the calling party wants to finish the call, it on hooks and at that time, the system is reset and both the extensions are assigned the state 1. It must be noted that only the calling party has the facility to clear down the call.

4.2.1.e. STATE 9

This state generates the busy tone, which is generated from the make-break of the dial tone. This state also monitors the off hook pin of extensions in order to reset the system.

4.2.1.f. STATE 10

This is idle state and used when the call is in progress between the two extensions. It is that state, which is assigned to the called party that does not, has any reset option, so it cannot disconnect an incoming call.

4.2.1.g. STATE 11

This state connects the repeated message to the calling subscriber when the number dialed by him is not valid.

4.2.l.h. STATE 12

When the call between an extension and the trunk is in progress, this state continuously monitors the off hook pin of the extension in order to reset the system and release the extension from trunk.

4.2.2. STATE DIAGRAM OF TRUNK

The state diagram of trunk is shown in fig 4.2. The function performed by each state is listed below.

4.2.2.a. STATE 1

In state 1, the system continuously monitors the ring detector pin of trunk. When this pin goes high, the system starts counting the number of bells. When this number is increased from 5, the system off hooks the trunk and the busy flag of trunk is made high and welcome message is given to it. After wards the DTMF receiver is connected to it and the state is transferred to state 2, other wise the state is not changed.

4.2.2.b. STATE 2

In state 2 the system checks whether trunk has dialed a number or not. If dialed, DTMF receiver is disconnected and the control is transferred to various cases of the state 2. If not, then the system automatically transfers the trunk to one among the various cases (i.e. operator). Rest of the cases ana1yz the dialed number. If the dialed number is busy, the case switches the state 2 to state 9. If the dialed number is not busy the control is transferred to the respective states (states 3.4,5 and 6) of each extension and respective extension’s busy flag is high. If the dialed number is not found to be valid after analysis (i.e. the number is not listed) the state is changed to state 11.

4.2.2.c. STATE 3,4,5,6

All these states generate ring on the respective extension and also generate ring back tone to the trunk. Meanwhile they transfer the state of the called party to state 12 and start ringing the extension and keep on monitoring the off hook pin of extension. If the called party off hooks the telephone then the system connects the speech path of the two speakers and transfers the trunk control to state 7. If the called subscriber does not off hook, then after 25 bells, the system automatically reset the trunk i.e. trunk is assigned state 1 and busy flag is lowered.

4.2.2.d. STATE 7

This is idle state and used when the trunk makes a call to any extension. Upon the completion of call setup, the trunk resides in an idle state, which means it cannot clear down the call. Therefore, the state assigned to the extension is state 12 that has reset option, so it can only disconnect the call between trunk and extension.

4.2.2.e. STATE 8

This state connects the repeated message to the trunk when the number dialed is not listed.

4.2.2.f. STATE 9

This state generates the busy tone for a period of 20 seconds and then reset the trunk i.e. trunk is assigned state 1 and busy flag is lowered.

4.2.2.g. STATE 10

It continuously monitors the off hook of the extension in order to reset the system and release the extension from trunk.

4.3. FLOW CHART

Flow chart is a graphical method of representation of software. The flow chart of our software is shown in fig 43.

4.4. SOFTWARE

The software of our exchange is given in Appendix A.

DISA

In order to record the messages on ISD 2560 single chip voice record playback IC, we have developed a separate program. This program has the advantage of allowing the operator or the vendor to change the messages when required. The ISD 2560 is capable of recording any message of sixty seconds duration. We have divided this 60 seconds duration into further four portions, by applying different addresses on the address input of ISD 2560. The program is executed in such a way that the system first asks the user either to record or play the message. After getting the required option, the system then inquires about the number of the message (1,2,3 or 4). Each message is of the duration of fifteen seconds. In this way, the vendor can change the message depending upon the subscriber's request.

The source code is programmed with the switch keyword, whose function is to route to required message among the four. Value of the variable 'a' is given by the user, which permits the system to record or play the message IC.

This program is given the name of DISA.exe and is given in Appendix B. It can either be executed at the time of installation of the exchange or when the message is to be changed during the operation of exchange.

4.5. SYSTEM DEFINED FUNCTIONS AND KEYWORDS

The explanation of various functions and keywords used in our software program which have already been defined in the libraries of Turbo C++ are listed below:

4.5.1. inportb, outportb <DOS.H>

This function provides a method of accessing the processor through the medium of input/output ports. The commands used for this purpose are inport and outport e.g.

a. inportb reads a byte from a hardware port

b. outportb outputs a byte to a hardware port

DECLARATION: unsigned char inportb (int portid);

void outportb(int portid, unsigned char value);

REMARKS: a. inportb works just like the 80x86 instruction IN. It reads the byte from portid.

b. outportb works just like the 80x86 instruction OUT. It writes the byte of value to portid.

ARGUMENTS: Int portid: It is either an integer value or in the form of hex or decimal value of the address of the port on which/from data is written/retrieve.

Unsigned character value: This is the value of data to be written on data bus only in case of outportb.

RETURN VALUE: a. inportb return the value read.

b. outportb does not return any value.

4.5.2. SWITCH STATEMENT

The SWITCH is one of the most commonly used keyword in C++ programming. It is similar to ‘if else’ statement with the advantage that it is much cleaner to look at the whole bunch of ‘if else’ statements. We have used this keyword in our software for branching. Details of the switch statement are given below.

This switch uses switch, case, and default keywords.

SYNTAX

- Switch (<expression>) <statement
- Case <constant expression>
- Default:

Switch causes control to branch to one of a list of possible statements in the block defined by <statement>. The branched-to statement is determined by evaluating <expression>, which must return an integral type. The list of possible branch points within <statement> is determined by preceding sub statements with case <constant expression>: where <constant expression> must be an int and must be unique. Once a value is computed for <expression>, the list of possible < constant expression> values determined from all case statements is searched for a match. If a match is found, execution continues after the matching case statement and continues until a break statement is encountered or the end of <statement> is reached.

Default: If a match is not found and the “default:” statement prefix is found within <statement>, execution continues at this point. Otherwise, <statement> is skipped entirely.

4.5.3. STRUCTURE

Structures are used to hold different amounts of any type of variable, even arrays or other structures i.e. it groups variables into a single record. The keyword used in C language for this purpose is struct.

SYNTAX

struct [<struct type name>]

{

[<type> <variable-name[, variable-name,….]>];

[<type> <variable-name[, variable-name,….]>];

………………

} [< structures variables>];

A struct, like a union, groups variables into a single record.

<struct type name> An optional tag name that refers to the structure type

<structure variables> The data definitions, also optional.

Though both <struct type name> and <structure variables> are optional, one of the two must appear. Elements in the record are defined by naming a <type>, followed by one or more <variable-name> (separated by commas). Different variable types can be separated by a semicolon. To access elements in a structure, you use a record selector (.). To declare additional variables of the same type, you use the keyword “struct” followed by the <struct type name>, followed by the variable names.

4.5.4. ARRAY

Array is the collection of similar data variables, placed contiguously in Memory.

SYNTAX

Variable type; array name [elements of array]

Variable type= char, integer, long, float.

Array name= user dependent

Elements of array=user dependent

4.6. USER DEFINED FUNCTIONS AND KEYWORDS

4.6.1. MACROS

Macros provide a mechanism for token replacement with or without a set of formal, function-line parameters, thus avoids writing this phrase again and again, hence, and consumes labor and time.

The macro, which is used by us in our software program, is:

#define PORTA_ 8255 0x300

0x300 → PORT A of 8255 (input port)

0x301→ PORT B of 8255(output port)

0x302 → PORT C of 8255(output port)

0x303 →Control Word of 8255

0x304 → COUNTER 0 of 8253(not used)

0x305 → COUNTER 1 of 8253(dial tone)

0x306 → COUNTER 2 of 8253(interrupt generator)

0x307 → Control Word of 8253

0x308~ 0x30B→ Control of message IC

0x30C~ 0x30F→ Enabling Dtmf receiver I and 2

0x310~ 0x313→ Enabling Dtmf receiver 3

4.6.2. VARIABLES

- **STRUCT EXCHANGE PHONE [x]**

The identity given to the four extensions and one trunk is named as ‘phone’, which is basically a ‘structure’ named as ‘exchange’. The variables of the structure exchange’ are

- state
- busy
- dial tone
- dtmf
- counter.

The array ‘phone’ has 5 elements and is defined as phone[5]. The declaration of this variable is struct exchange phone[5]; (in C)

Or

exchange phone[5]; (in C++)

And the elements of array along with the variable of the structure ‘exchange’ are defined as

a. phone[0].state
b. phone[0].busy
c. phone[0].dialtone
d. phone[0].dtmf
e. phone[0].counter

The above example is for trunk and if the connection to an extension is required, then instead of ‘0’ we use either 1,2,3 or 4. From now onwards, the variable ‘x’ will be used which corresponds to 0,1,2,3 or 4, just for the explanation.

The function of each variable is defined below.

- **phone[x].state**

This variable as its name indicates is used to point the state of the extensions (including trunk). The word state means all the possible cases (defined in state diagram) at which the extensions could stuck in order to do its job i.e. having a dial tone to press a destination number, ring at an extension or hear a message etc. The values given to this variable are all unsigned; therefore we have assigned an unsigned integer value to this variable. Their values range from 1 to 12. The initial or reset value is ‘1’. Since our exchange is unattended (no operator is there to handle the exchange) therefore for developing software, then a technique must be adopted to ensure that any non-listed value will not be interacted. If this happens, that particular extension will be hanged forever until the system is reset manually. The user of that particular extension will suffer from disconnection. Therefore we have to take into account an ultimate care in assigning the values of these variables in order to remove the case of disconnection.

- **phonelx].busy**

Phone[x].busy is one of the simplest but extremely useful variable. This variable can make the user's extension 'dead'. 'Dead' is the term, which is used occasionally and it corresponds to an extension without a dial tone, if such an extension is dialed, a busy tone will be encountered, even the extension is on hook. This variable is unsigned and its possible values are '0' and '1' i.e. it behaves as a flag which is either low or high. When a person picks up the phone, the flag is high and when on hooks, it is low. If somehow this remains high even after the call ends up, the extension becomes dead, and only manual resetting of the system can solve this problem. Therefore care should be taken in assigning the value to this variable.

- **phone[x].dialtone**

The possible values that this variable can take are 1,2 and 3. Since we have made 3 dial tones with the help of INTEL 8253, therefore whenever a dial tone is connected to the extension, the number of that particular dial tone must be assigned to the extension with the help of this variable. This variable is used at the time of disconnection of dial tone. When the system connects the dial tone, it also records the number of dial tone, which is helpful in disconnecting the same dial tone; otherwise there is a possibility of disconnecting the dial tone, which is currently being used by other subscriber. This variable is extremely useful with another array of variables named as dialtones[x], which will be discussed later.

- **phone[x].dtmf**

The possible values, which this variable can take, are 1,2 and 3 (unsigned variable). Since we have used 3 DTMF receivers, therefore whenever a DTMF receiver's junctor is connected to the extension, the number of this DTMF (1,2 or 3) is assigned to the extension with this variable. This variable is used in two ways; one way is to get the latched number from the corresponding DTMF output and other is at the time of disconnection of DTMF. When the system connects a DTMF, it records the number of DTMF, which is helpful in disconnecting the same DTMF otherwise it could disconnect the DTMF of other user. This variable when used in conjunction with dtmf[x] is very useful in various respects, which will be discussed later.

- **phone[x].counter**

Phone[x].counter is a variable, which is defined as a long integer. But even with this definition, this variable cause overflow. This overflow can be avoided by refreshing the counter in software. This counter performs multiple operations. It is used in counting the duration of connecting and disconnecting the ring, ring back tone and busy tone. As it is mentioned earlier, the system monitors all the states in milliseconds, therefore we cannot generate the connection and disconnection of tones by using the 'delay( )' function of Turbo C++. This will make the system lethargic at the time of ringing and generation of tones. To encounter this situation, this counter is initialized at the time of busy and ringing state and incremented each time the system passes through. Its resetting is done before overflow or after jumping to the new state.

- **dialtones[x] AND dtmf[x]**

  These two variables are defined as unsigned integers and contain the flags of each dial tone and DTMF receiver. There are 3 elements of array. When a dial tone is used by any extension, the flag of that particular dial tone is switched to high so that none of the other extension can use it. When a disconnect dial tone function is called, the release of this flag is carried out, thus indicating that it is free to use. Same is the case with dtmf variable. Initially resetting should be done otherwise its garbage value may cause problems. Further explanation is given in pref_dial tone; connect_ dialtone and disconnect_dialtone.

4.6.3. FUNCTIONS

The functions, which are generated and used by us, are:

4.6.3.a. connect_ ext x_ ext y( )

The main job of this function is to connect two junctors. The various junctors connected by this function are

- Extension1 and Extension 2.
- Extension1 and Extension 3.
- Extension1 and Extension 4.
- Extension2 and Extension 3.
- Extension2 and Extension 4.

- Extension3 and Extension 4.
- Trunk and Extension1.
- Trunk and Extension2.
- Trunk and Extension3.
- Trunk and Extension 4.
- Music and Extension1.
- Music and Extension2.
- Music and Extension3.
- Music and Extension4.
- Music and Trunk.
- Message and Trunk.
- Message and Extension1.
- Message and Extension2.
- Message and Extension3.
- Message and Extension4.

ARGUMENTS: The above-mentioned function does not contain any argument.

RETURN VALUE: The above-mentioned function does not return any value.

CALLING A FUNCTION: connect extl_ext2 ();

4.6.3.b. int pref_dialtone( ) AND int pref_dtmf()

We have created three dial tones named as dialtones[x] and DTMF receivers named as dtmf[x]. Therefore preference function is required in order to assign the dial tone and DTMF in sequence. When the dial tone is assigned to a particular extension, the busy flag is set i.e. dialtones[x]= 1. This function also has the overload facility which connects the message " all lines are busy please call later" to a calling extension.

ARGUMENTS: There is no argument in the above-mentioned function.

RETURN VALUE: It generates a return value. The return value contains the number of dial tones, free at that moment.

CALLING A FUNCTION: This function is called by the connect_dialtone and connect_ dtmf functions which is to be discussed later e.g. int b= pref_dialtone();

4.6.3.c. connect_dialtone(int) AND disconnect_dialtone(int)

This function is used to connect and disconnect the dial tone The main job of this function is to first connect the dial tone in conjunction with int pref_dialtone function in sequence and secondly stores the respective dial tone number to the associated extension no in phone[x].dialtone, which enables the disconnection of dial tone. It also lowers the flag of the released dial tone.

ARGUMENT: It takes an argument that which extension wants the dial tone and record the dial tone number in phone[x].dialtone, which enables the disconnection process of dial tone.

RETURN VALUE: It does not have a return value.

CALLING A FUNCTION: This function is called by using connect_dialtone(l); which shows that extension 1 wants the dial tone e.g. if dial tone is free then it is assigned to extension 1, phone[1].dialtone=2, shows that dial tone 2 is assigned to extension 1.

The function disconnect dialtone(1); shows that extension 1 wants to release the dial tone. In order to disconnect the second dial tone from extension 1, phone[1].dialtone=2 is used.

4. 6.3.d. connect_dtmf(int) AND disconnect dtmf(int)

This function is used to connect and disconnect DTMF. The main job of this function is to first connect the DTMF in conjunction with int pref_dtmf function in sequence and secondly stores the respective DTMF number to the associated extension number in phone[x].dtmf which enables the disconnection of DTMF. It also resets the flag of the released DTMF.

ARGUMENT: Its argument is the extension number that requires the DTMF. It records the DTMF number in phone[x].dtmf, which enables the disconnection process of DTMF.

RETURN VALUE: It does not have a return value.

CALLING A FUNCTION: This function is called by using the statement connect_ dtmf(l); which shows that extension 1 wants the DTMF. If dtmf[2] is free then it is assigned to extension 1 which is shown by the statement phone[1].dtmf=2.

The function disconnect dtmf(1); shows that extension 1 wants to release the DTMF. The variable phone[1].dtmf=2 is used to disconnect second DTMF from extension 1 and dtmf[2] flag is reset.

4.6.3.e. void offhookx(int)

This is a multipurpose function, which is executed only when the extension is off hook and it is present in almost all the state diagrams of the system. Four extensions require same number of these functions.

1. void offhookl( );

2. void offhook2( );

3. void offhook3( );

4. void offhook4();

The main purpose of this function is to off hook the extensions properly. The word properly is used due to the fact that when an extension undergoes a call setup procedure, it passes through different cases. Off hook could be required by the extension at any moment, that moment could be state 1,2,3,4,……... Using five cases solves this problem. Among these five cases, three are used to off hook the calling party's extension when having a conversation or generating a ring or calling a called party's extension i.e. it releases the busy flag of both extension, switches off the ringing, disconnect the junctor and reset the associated counters.

Fourth case deals with the off hook of the extension when having a call setup or conversation with the trunk. It on hooks the extension, releases the busy flags of both lines, disconnects the junctor and resets the associated counters. The fifth case is used to off hook the extension on hearing a message in loop i.e.

resetting the DISA IC and disconnecting the junctor between DISA and extension. This function without arguments only perform the general off hook procedure i.e. releases the busy flag and disconnects dial tone and DTMF.

ARGUMENTS: It contains arguments in the form of number i.e. 1,2,3,4,5 which points to the number of cases of this function as mentioned earlier.

RETURN VALUE: This function does not have any return value.

CALLING A FUNCTION: The various cases of this function are contained by almost all the states of the software. Therefore this function plays an important role in the software and system monitors any of these functions at a time. For example offhook l(2) means off hook of extension 1 with case 2.

4.6.3.f. SET_RESET_XPT()

This function is used to set and reset the cross points of MT8816. The strobe pin of the cross point is connected to 8255 IC. The job of this function is to enable the internal decoder of the cross point for a while and then disable it. During this process, what ever the input to the decoder is, it is decoded and corresponding cross point is switched on or off, depending upon the data pin of the decoder. The reason for using this function is that the ports of 8255 are latches. They store the data until power is switched on. When we want to change the state of the cross point, we first give its address and then enable the decoder of MT8816, otherwise the previously latched data of port is taken as an address. In other words, the reason of this function is to synchronize the process of switching on or off the cross point.

ARCUMEINTS: There is no argument in the above-mentioned function.

RETURN VALUE: This function does not have any return value.

CALLING A FUNCTION: The calling of this function is carried out in almost all the connect and disconnect functions explained earlier.

e.g outport(0x302,0x45)

set_ reset_ xpt( );

4.6.3.g. int getno(int)

As its name indicates, this function is used to get the number dialed by an extension. This function is defined with arguments and returning values to perform its job. This function performs two operations:

1. The main job of this function is to extract a word of four bits from a word of a byte (8 bits). Since we have used three DTMF receivers each having all output of 4 bits. The output of these DTMF receivers is retrieved by an 'inportb' function. They give the number of 8 bits. These eight bits are extracted in such a way that the required digit is present in the lower nibble of a byte. So some sort of filtering is required in order to get only last four bits of a byte. The C language operator modular division (%) is used to get the required results.

2. The arguments and the returning values play an important role in this function. The argument shows the particular value dialed by any extension and returning value returns that particular number in the form of integer. Argument value provides the information about that particular extension which is dialing at that moment in addition to the DTMF receiver (either 1,

2 or 3), which is currently in use. Therefore by substituting the extension number (argument) in variable phone[x].dtmf we can have the information about the number and address of DTMF receiver. Any further calculation is carried out in a normal manner.

ARGUMENTS: The arguments of this function can take any value either 0,1,2,3 or 4 depending upon the extension, which is dialing a digit.

RETURN VALUE: The returning value could be 0~9 depending upon the number dialed by user. Full number is returned after all sort of filtering that has already been explained above.

CALLING A FUNCTION: This function is called when an STD pin of DTMF receiver is high and is detected by 8255 IC. The calling is carried out as:

int dial = getno(2); which shows that extension 2 is requiring a number dialed by its user and it should be stored in a variable ‘dial’.

CHAPTER 5

EPABX OPERATION

Chapter 3 is the hardware explanation of the exchange while Chapter 4 is the software explanation. Chapter 5 cumulates the properties of both, in this chapter, the working of the exchange will be explained completely involving both the hardware and the software modules.

The hardware card is inserted into the ISA slot. Power supply is given to it and 5 pairs of cable come out of it. Among these 10 cables, 4 are of extensions and one is of trunk. These 4 cables of extensions are given to each subscriber and they are allotted a number as 6, 7, 8, 9. If a subscriber wants the exchange to become attended instead of unattended, he has the facility to alter the extension 6 as the operator's extension but only when requested. Along with this, he also has the facility to record 4 messages as he wishes. All the ICs used are easily available, and can be changed in the case of damage, because of their insertion in the sockets.

In software, an unusual way of programming has been adopted to create structured and modular software. This technique requires all the states to be programmed, but during one scan only five states are executed (four of extensions and fifth of trunk), hence reducing the run time. All the states are divided into almost equal time slices of the time slot available, so that the state processor does not remain in one state so long, that it cannot sense the changes occurring in other states of different telephone and trunk.

5.1. CASE 1:CALL FROM AN EXTENSION

Suppose extension A wishes to call. Before he picks up the telephone, the software is monitoring the state 1 of this extension. The state 1 has the functions which always check the off hook pin of this extension's SLIC through 8255's port A. As the subscriber picks up the telephone, the off hook pin goes low. At the same time this extension is given a busy status, so that any other extension or the trunk cannot access it. The change of off hook pin from high to low is sensed by state 1, latched by 8255's port A. In response to this change, state 1 tries to connect the dial tone and DTMF receiver's input by calling

connect_ dialtone(A); connect_dtmf(A);

These two functions scan the free dial tone and DTMF by calling

int pref_dialtone( ); int pref_dtmf ();

respectively. The number of the dial tone and DTMF is saved along with the allocation of the dial tone and DTMF. At the same time, the flags of this dial tone and DTMF receiver go high, so that any other telephone cannot use them, in order to avoid cross talk. The voice output of the message recorder IC is given to X13 of 8816. The junctors of the extensions are given to Y5, Y6, Y7 and Y8. In case of a situation where all the dial tones and DTMF are fully occupied, the respective column from Y5 to Y8 is connected to X13 and the DISA's second message

"All lines are busy, please call later"

will be given by enabling the P/R' pin of the message IC, through

outportb(0x308, 0x33)

State 1 monitors this entire sequence of events, and after allocating the dial tone and DTMF; it transfers the control to state 2 by the command:

Phone[A].state= 2;

The function of state 2 starts with monitoring the Std pin of the associated DTMF receiver, and waits for the Std pin to become high which shows pressing of a number. Meanwhile, if a subscriber wants to change his mind of making a call, this state also has a facility to off hook the extension. On the other hand, if he presses a number, the DTMF receiver whose junctor is already connected with the extension takes in the frequency of the corresponding number entered and latches it. After latching the number, the Std pin goes high. This is sensed by the 8255's port A and given to the software, which:

- Breaks the connection between the dial tone, DTMF receiver's output, and the SLIC's junctor. At the same time, it lowers the associated flags, thus freeing up the dial tone and DTMF receiver.

- ♦ Generates the inportb function, which will retrieve the number from the data bus by enabling 8870. This number will be stored into a variable named as ‘dial’ with the help of the function known as int getno(A); this scan doesn’t change the state of the system.

In the next scan, this number will be analyzed and compared with 6, 7, 8, 9 or 0. There come three situations.

- ♦ An invalid number is dialed
- ♦ A valid extension number is dialed
- ♦ The trunk is accessed

5.1.1. SITUATION 1: AN IN VALID NUMBER IS DIALED

If the dialed digit is not equal to any of the digits 6,7, 8, 9 or 0, then the system is transferred to state 11, where the calling subscriber will hear the third message of the DISA. Connecting the voice output of the IC 1SD2560 to the respective SLIC’s junctor does this and the following message is heard:

“The number you dialed is not listed please call again”

State 11 repeatedly complements the P/R pin so that this message is repeated again and again and the subscriber has no other option except to on hook the telephone — thus disconnecting the loop. This process releases up the junctor of this telephone as well as the message output of the message IC.

5.1.2. SITUATION 2: A VALID EXTENSION NUMBER IS DIALED

In case if the number pressed is a valid one, (6, 7, 8 or 9), excluding the extension number of the calling customer, the software checks the busy flag of the particular extension. If the busy flag is high, the variable phone[A].state equals 9. In the next scan, the software runs state 9. In state 9 busy tone — a make break sequence of the dial tone. This connects the dial tone in such a way that the counter increments from 0 to 7^{th} scan of this software — a time during which the connection is made. During, 7^{th} to 15^{th} scan, the connection is broken_ a period recognized as the silent period. After the 15th scan, the counter is reset. This busy tone can also be generated by the DOS.H function “delay”. However, this function results in a slow program execution. During this state, the subscriber has no other option except to on hook the telephone, causing all the telephones to be reset.

If he dials up another extension, the telephone switches its state from 2 to 3, 4 or 5, depending upon the extension dialed. With the start of this state, the called party is sent from state 1 to state 10, which is an idle state. This state 3, 4 or 5 of the calling party generates a ring tone for the destination and a ring back tone for the source. Ring tone is generated for the calling party by providing a high pulse on the pin number 19 (Ring Control Pin) of the 8255. The principle of the ring back tone is similar to the busy tone, with the counter incrementing from 0 to 15th scan of this software — a period recognized as the active period. During, 15th to 30th scan, the connection is broken — a period recognized as the silent period. After the 30th scan, the counter is reset. Running concurrently is the monitoring of the off hook pin of the called party. The subscriber will remain in this state, until the called party off hooks the telephone or the calling party on hooks the telephone. In case, where the called party is not at home, the calling party has the facility to on hook the telephone.

If the called party picks up the telephone, depending upon the destination, the calling party jumps to state 6, 7 or 8. At this time, the ring tone and ring back tones are disconnected and junctors of both the calling and the called party are connected. The called party remains in state 10. Conversation will continue as long as desired. It must be noted that in this state only the calling and not the called party has the capability to on hook the telephone. When the calling party on hooks the telephone, the junctors of both the telephones are disconnected and the system is reset.

5.1.3. SITUATION 3: THE TRUNK IS ACCESSED

In the third case, the subscriber dials up a valid telephone number — either of another extension or of the trunk. If he dials up the trunk, by pressing '0', the 8255's outport off hooks the trunk. The trunk's flag gets busy, and it goes from its wait state state1, to the idle state — state 7. The subscriber hears the dial tone and the case is handed over to the PTCL with the variable phone[A].state assigned state 12. This state — state 12, has the ability to on hook the trunk, since it cannot on hook itself.

In case the trunk's busy flag is already high, the calling extension jumps to state 9 and a busy tone is heard. This busy tone is similar to the one experienced in situation 2.

5.2. CASE 2:CALL FROM TRUNK

This case directs trunk call to any of the extension or the operator. Initially when the software runs, the trunk is recognized as phone 0. The phone 0 is resting in state 1. During state 1, in each scan, the ring detection output of the trunk module is checked, via 8255. When a call comes from other exchanges, the trunk's ring detector output oscillates between logic high and low, informing our exchange of a call from other exchanges. Here comes the role of phone0.counter, which starts counting the incoming rings, so that the telephone is picked up after exactly 5 rings. After the exchange picks up the telephone, the junctor of the trunk and the output of the speaker are connected through an outportb command. The following DISA message is then heard:

"We1come, if you know the extension of the person, you would like to contact, dial now or wait for the operator's assistance"

The operator in this case is the extension 1 by default. Its role comes in situations, where the trunk accesses the exchange by pulse dialing. This extension then has a capability to transfer calls to the required destination. The trunk in state 1 hears this DISA message.

After giving the D1SA message, a DTMF is assigned to the trunk's junctor and the trunk goes in state 2. The state 2 continuously monitors the Std pin of the assigned DTMF receiver. When a number is latched, this pin goes high. When this occurs, the DTMF receiver is released and the number is taken in by system by the function getno( ). This number is stored in an integer named as 'dial'. In the next scan this number is analyzed. There are two cases to be considered.

- An invalid number is dialed
- A valid number is dialed

5.2.1. SITUATION 1:AN INVALID NUMBER IS DIALED

If the dialed digit is not equal to any of the digits 6, 7, 8, 9, then the calling party will hear the third message of the DISA. Connecting the voice output of the IC ISD 2560 to the trunk's junctor does this and the following message is heard:

"The number you dialed is not listed please call again"

This message is heard once, after which the trunk is automatically on hooked. This process disconnects the loop and releases up the junctor of the trunk as well as the message output of the message IC.

5.2.2. SUTUATION 2: A VALID NUMBER IS DIALED

In case, if the number dialed is a valid one, (6, 7, 8 or 9), the software checks the busy flag of the particular extension. If the busy flag is high, the variable phone[0].state equals 9. In the next scan, the software runs state 9. In state 9 busy tone — a make break sequence of the dial tone — is heard. This connects the dial tone in such a way that the counter increments from 0 to 7^{th} scan of this software — a time during which the connection is made. During, the 7^{th} to 15^{th} scan, the connection is broken — a period recognized as the silent period. After the 15^{th} scan, the counter is reset. This busy tone can also be generated by the function "delay" defined in DOS.H. However, this function results in a slow program execution. The subscriber will hear this busy tone 10 times. This count is done by a variable ‘count’. After 10 busy tones, the trunk is on hooked and both the count variable and the busy flags are reset with the trunk-reassigned state 1.

If he dials up an. extension (not busy), the trunk switches its state from 2 to 3, 4, 5 or 6, depending upon the extension dialed. With the start of this state, the called party is sent from state 1 to state 12, which has the capability of on hooking the trunk and itself, because trunk do not have this facility. This state 3, 4, 5 or 6 of the trunk generates a ring tone for the destination and a ring back tone for the source. Ring tone is generated for the calling party by providing a high pulse on the pin number 19 (Ring Control Pin) of the 8255. The principle of the ring back tone is similar to the busy tone, with the counter incrementing from 0 to 15th scan of this software — a period recognized as the active period. During, the 15th to 30th scan, the connection is broken — a period recognized as the silent period. After the 30th scan, the counter is reset. Running concurrently is the monitoring of the off hook pin of the called party. This state will generate a ring for the extension 15 times, with the same ‘count’ variable. After 15 bells, it is considered that the called party is not at home, thus initializing both the states and resetting both the busy flags and on hooking the trunk.

If the called party picks up the telephone, depending upon the destination, the trunk jumps to state 7, which is an idle state. At this time, the ring tone and ring back tones are disconnected and junctors of both the calling and the called party are connected. The called party remains in state 12. Conversation will continue as long as desired. It must be noted that in this state only the called party and not the trunk has the capability to on hook both the telephones. When the called party on hooks the telephone, the trunk is on hooked, the junctors of both the telephones are disconnected and the busy flags are reset.

5.3. BILLING FACILITY

The billing facility is a complementary one. In case the exchange is installed in commercial premises and the network usage is to be monitored, the time, date, duration and charges of the calls made are logged in a text file. After the usage of a month, the operator or the owner of the exchange, prints these four files (each of a subscriber).

CHAPTFR 6

CONCLUSIONS & FUTURE SUGGESTIONS

Our exchange is a one trunk and four extensions one. This exchange incorporates a minimal of hardware components being controlled by software. The software approach, "Implementation of State Diagram through Program Flow Chart" is a modular one, so that a variation in software without altering the hardware saving the labor, aids in debugging of the problem.

6.1. OBJECTIVES TO BE OBTAINED

When we started to develop this exchange, we had certain objectives in our mind. All these comprised facilities, which were never present in an exchange, implemented in university level. These facilities are:

- Inclusion of DISA
- Provision of music, during 'wait' period
- Billing Facility
- Call transfer
- Calling Line Identity
- Programming in Visual C++

Out of these six facilities, we achieved the earlier four, whereas due to the lack of time and resources, we were unable to accomplish the facilities comprising the lowest pair. These facilities are explained below.

6.1.1. INCLUSION OF DISA

To present the 'line status' information to a subscriber accessing a limited availability line exchange, DISA is used, which plays the recorded messages like

1. "Welcome, if you know the extension of the person, you would like to contact, dial now or wait for the operator's assistance ".
2. "All lines are busy, please call later".

3. “The number you dialed is not listed please call again”.

6.1.2. MUSIC PROVISION

During the handover from trunk to required extension of the calling party music will be played. This objective is achieved by the use of a three-pin oscillator type Music IC M66T, which is a cheap and readily available IC. It generates a non-uniform waveform, which sounds like music. This music is provided during the wait period, to entertain the caller.

6.1.3. BILLING FACILITY

The basic theme of our exchange is the control of hardware via software. The software approach is a modular one, so that, additional features can be added or edited only via variations in software — thus adding flexibility to the system. One of such facilities is billing. In order to monitor the network usage, the customer is charged. The Central Office in a commercial exchange is equipped with switching call transfer and billing facilities. The formation of our exchange is such that it can be used in offices and other such areas. Hence, it is equipped with charging facility, so that the network usage of every customer can be catered for.

6.1.4. CALL TRANSFER

In situations, where the trunk accesses the exchange by pulse dialing, one extension, by default, serves as an operator. This extension then has a capability to transfer calls to the required destination.

6.1.5. CALLING LINE IDENTITY

Pre-detection of the calling party is known as “Calling Line Identity”. On receipt of the off hook signal, the exchange establishes the calling line identity. This is because, the exchange’s control system has to monitor network usage. This facility can also be provided on the subscriber side before the telephone is picked up to inform him/her about the calling party. The later type can be provided on our exchange. This facility being a new one was and is still not well known. Hence, we did not have any proper guideline in this regard. Also, in order to establish the calling line identity at the subscriber telephone set the PTCL's main connection (trunk in our case) has to be CLI encoded. Since, our university does not have this facility, it would have impossible to demonstrate this achievement.

6.1.6. VISUAL C++

Visual C++ is a windows based oriented approach providing a Graphical User interface (GUI). Programming in this language would have given a visual display of the software allowing the user to interact with it graphically. With the start of this project, we decided to implement the software in Visual C++. Since, we had less time we cancelled our idea and decided to program in Turbo C++ a language we had earlier mastered. Turbo C++ is one of the most comprehensive and sophisticated development environments. It provides a high level of programming power and convenience, while offering a diverse set of tools designed to suit almost every programming style.

6.2. FUTURE PROSPECTS

The crux of a microprocessor-based system is the design of control strategy. This control strategy manifests through software, which manipulates the control system. The advantage of using such a strategy lies in ease of gaining added benefits just by altering the program or a part of it — without a thorough change of physical components is required. Some additional facilities outlined below can be included in future, if required.

6.2.1. EXPANSION OF THE EXCHANGE

Our exchange is a 1+4 line one, which means that it has 1 trunk and 4 extensions. In large offices or buildings, use of small exchanges can result in network congestion. Hence, suiting customer requirements, this exchange can be expanded. Use of the same types of ICs can lead to much more extensions over the single trunk in order to fulfill the user requirements. So this 1+4 exchange gives the basic idea for understanding and establishing an augmented form.

6.2.2 INCLUSION OF OPERATOR

An operator can be incorporated in this exchange, in instances, where this exchange is expanded by fusion of much more SLICs and hence addition of extensions. With the operator come some certain security measures, which are:

- Some business subscribers prefer that the operator at the EPABX can be debarred from entering an established connection either local or via the main exchange. However, adoption of secrecy measures facility on calls through the exchange line tends to increase the difficulties of monitoring and supervisory work.

- In most cases, it is desirable to give the EPABX operator facilities to enter an established connection in order to offer a trunk call, if trunk offering facilities are required on an installation designed to give secrecy, then it is necessary to arrange for the equipment to give a warning tone to the parties concerned, whenever the operator makes use of the trunk offering access to an established connection.

REFERENCES

- John Bellamy. Digital Telephony. John Wiley & Sons, 1982.
- Atkinson, J. Telephony. Pitman Publishing, 1982.
- W. Fraser. Telecommunication. CBS Publishers, 1985.
- Barry B. Bray. The Intel Microprocessors. Prentice Hall International, 1998.
- Martin P. Clark. Network and Telecommunications. John Wiley & Sons, 1991.
- Michele Duck, Bishop, Richard Read. Data Communications for Engineers.Addison Wesley Longman, 1996.
- Stephen J. Bigelow. Understanding Telephone E1ectronics. SAMS Publishers, 1997.

- Ghazanfar- ud - Din Arshad. Design of Telephone Exchange (Two Lines).Texila: Department of Electric Engineering , 1983.
- Encarta Encyclopedia Deluxe, 2000.
- Encyclopedia Britannica, 1999.
- http://www. mitel.com.
- http://www.ti.com
- http://www.icmaster.com
- http://www. Intel.com
- hltp://www. Beyondlogic.org
- http://www. chessworth. com/ pv