NEAR EAST UNIVERSITY

FACULTY OF ENGINEERING

Department of Electrical and Electronic Engineering

REMOTE CONTROL OVER TELEPHONE LINE

Graduation Project
EE- 400

Student: Omar Daban (20001101)

Supervisor: Assist. Prof. Dr. Kadri Bürüncük

Nicosia - 2004

ACKNOWLEDGMENTS

__IN THE NAME OF ALLAH, MOST GRACIOUS, MOST MERCIFUL.__

Firstly I wish to express my deep appreciation to my God how stood beside me all the time and how supported me in all my achievements and who has given me the power and patience to finish my college studies successfully.

Secondly I offer special thanks to my parents, who encouraged me in every field of life and try to help whenever I needed. They enhanced my confidence in myself and they trust in my behavior. I am also grateful to my father who supported me and my mother whose prayers to accomplish every stage in my study journey and reach this point of education.

I would like to present my special appreciation to my supervisor Assist. Prof. Dr. Kadri Bürüncük for his help, advises, comments and effort in preparation for this project. His trust in my work and me. His friendly behavior with me and his words of encouragement kept me doing my project.

Last but not least I would thank Mohannad Yasin, Mohammad Al-jahraw and Yaza Al-kilani for their contribution to achieve this project. Also I will never forget my wonderful times that I spent in Cyprus and Near East University with my good friends who helped me incorporeally during my studying in college.

This project is dedicated to Palestine.

ABSTRACT

The remote control over telephone line project designed to meet the need of remotely controlling the electronic devices by following a few simple procedures, in addition the project circuit has to be compact, low cost to construct and requires ordinary equipments to operate, however the remote control over telephone line circuit requires a telephone at the commander position and another one at the destination position, with the telephone line at the destination connected to one terminal of the project circuit and to the other terminals are the desired electronic devices required to be controlled.

This is achieved by building the project using both hardware and software, it is possible to build the same project using electronic components only, but a large number of components would be needed, so more power supply consumption, additional cost expenses and increased possibility for faults, but the software achieves this task with the help of a limited number of electronic components in order to lower the power consumption, decrease the project budget and the circuit to be immune for possible errors that may occur during the control process.

Because of the need to use the telephone to pass the commands in a hand, and electronic components with a software to build the project at the other hand, the basic knowledge about telecommunications, microcontrollers and electronic design are very important in order to plan and build this project.

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INTRODUCTION

The remote controlling occupies an important position in the control systems design because of the demands to control the electronic devices from a remote positions. The remote control device can be used for many applications, such as TV & VCR remote control that uses IR (infrared light) for a few meters distance control, also there are remote controls for a short and long distance using RF (radio frequency), and some other remote controls using the networks to pass the control commands[7].

The communication devices can communicate the data and also the control commands, whether wired or wireless communication systems can be used for control purpose. Telecommunication systems and techniques can contribute to remote controlling because it is communicating local and remote parties, in remote control over telephone line project the PSTN (public switching telephone network) is used to handle the commands from a remote or local position by using the DTMF (dual tone multi frequency) signaling principle, that is used to encode and decode the pressed buttons of the telephone keypad[3].

The processing of the DTMF signals needs a qualified system to handle, therefore the microcontrollers used in this project to achieve this job with the help of a DTMF decoder circuit and some other interfacing circuits. Microcontrollers need to be programmed at the low level language programming that the CPU of the microcontrollers understand.

Such a project has many applications depending on the user demands, for example controlling the home appliances such as heaters, air conditions and gates, and may be used in a factory or a big company that need to control some devices from many positions in the site[7].

CHAPTER ONE
TELEPHONE NETWORK

1.1 Introduction

The Telephone System has been developed over many years and has gone through many incremental evolutionary steps. The traditional service of the Telephone Network has been for Voice Communication and only recently the network has been used to support the high amount of data transfer we are currently experiencing. For cost effective voice communications it has been identified that the Humans can communicate at frequencies between 300Hz to 3500Hz. Though we can hear and speak at higher and lower frequencies, voice communications between 300Hz and 3500Hz are clear and efficient for the telephone network to transmit and receive, look at figure 1.1. A Voice Channel goes from 0Hz to 4000Hz and was developed to avoid any overlapping to any other adjacent voice channels.

Figure 1.1 voice channel bandwidth.

Through the Telephone Network development it has been discovered that it is more efficient to transmit a voice channel in a digital form. In a digital form the voice channel can be routed to its destination with very low noise, higher reliability and more cost effectiveness. To do this the Voice Channel is converted to Digital by an Analog to Digital Converter (A/D Converter or ADC) at the Central Office (see figure 1.2). The Central Office (CO) is the location in your area that the 2-Wires that come from your

house called Tip & Ring are terminated. Once the Voice Channel has been digitized it is transmitted over the network to the CO of the number you called. At this remote CO the Voice Channel is reconstructed back into an analog form so that remote person can understand it. This transformation back to analog is done by a Digital to Analog Converter also know as a D/A Converter or DAC.

One important point to know is that the A/D Converter samples the Voice Channel at twice the frequency of the voice channel; that is at 8Khz. The reason the sampling is done at twice the original signals frequency is due to a law called the Nyquest Rate, which states to digitize a waveform and have enough information to reconvert it back to the an analog waveform one must sample the original waveform at no less than twice the frequency of the original waveform. Also each sample is identified by 8 unique bits that represent 256 different states. 8 Bits sampled 8000 times a second is equal to 64Kbps. Figure 1.2 shows the traditional communication process[4].

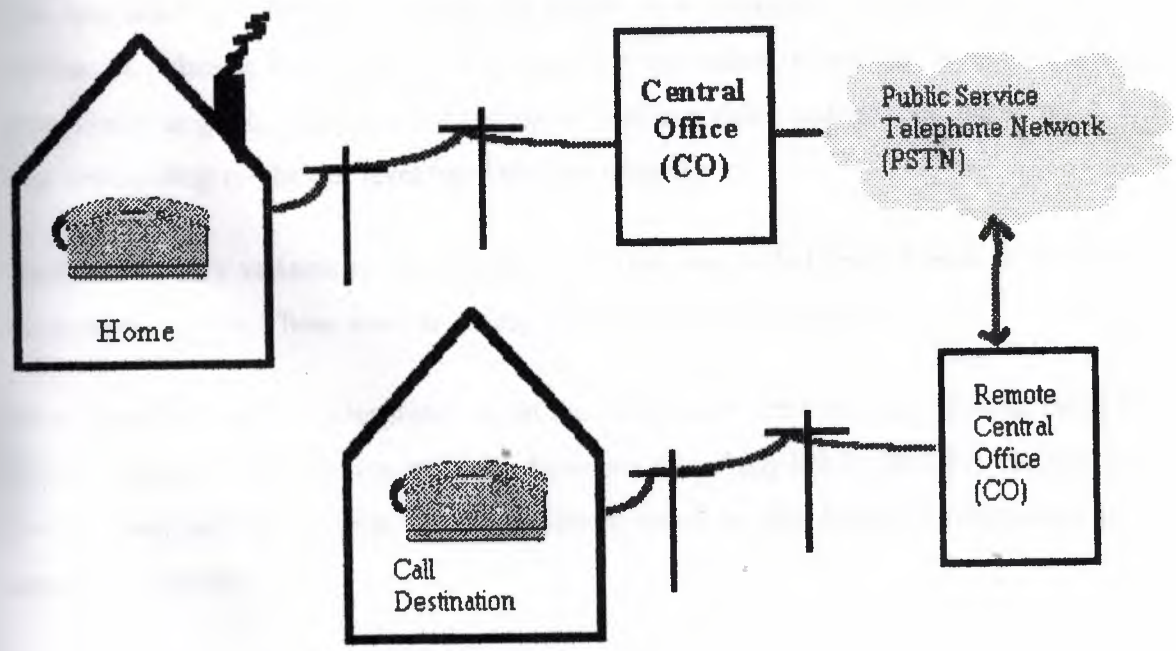

Figure 1.2 communication process.

1.2 Local Central Office Switching Systems

A brief overview of the history and operation of local central office telephone switching systems.

1.2.1 Step by Step (Strowger or SxS)

The Step by Step switching system was invented in the early 1890's by Almon Strowger, an undertaker in Kansas City, MO USA. Rumor has it that the local switchboard operator was diverting calls from his business to others. He wanted to have some way to have calls automatically routed to the destination without the intervention of an operator.

His invention had changed throughout the years, but the basic concept has remained the same. The system operates by using rotating "selectors" that can select a "level" (or step) by the number of pulses that are interrupted on a line (using a "rotary" telephone). This instructed a solenoid to move this wiper to a particular position based on the number of pulses it had received. This then put the caller on another level and again pulse dialed to get to another level and so on until the caller had reached the subscriber they were calling on the last level (or digit they dialed).

There were many variants to the step by step. One was called the XY switch made by Stromberg-Carlson. These were in widespread use in many rural areas.

These systems were in widespread use in the North American system well into the late 1980's (and some into the late 1990's!). There are few if any left in the USA or Canada, but they may still be in use in some third-world countries and former Soviet controlled countries in Europe.

1.2.2 Panel Switching Systems

The "Bell System" (Western Electric) never liked the idea of allowing customers to dial their own calls since they specialized in "customer service" and had operators complete all calls. By the late 1910's, they started to realize that this may not be a good idea and

started to use step by step systems (as described above). However, it did not work out well in large cities where there a large number of calls in progress at an one time. They developed a new system called a Panel switch. Their first switch was called a Panel switch because of the large vertical panels. It followed the same basic concept of the Step by Step but used a method of where the selector system operated on a concept of "ladders" where the selector would rise up on a panel by the number of pulses it had received.

The first one was installed in the early 1920's in Omaha, Nebraska and others were installed throughout the 20's and 30's in most metropolitan areas in the USA (except for Los Angeles, which was step by step).

Panel systems had a system of "revertive pulsing" where it communicated with another Panel switch by the receiving switch pulsing BACK to the originating switch! Instead of sending dial pulses out TO the terminating switch, it received them FROM the terminating switch.

Most Panel offices were removed from service by the late 1970's, though I hear the last one was located in northern New Jersey and was removed in the early 1980's.

1.2.3 Crossbar Systems

Crossbar systems were developed in the late 1930's (with technology from LM Ericsson) by Western Electric (the Bell System). The Crossbar had a significant advantage over either Panel or Step by Step since it was able to use a "store and forward" concept where it would take incoming digits, store them, and then process the call. It was also able to do call routing and determine where a call should be sent to by doing "translations" of the incoming digits and deciding how to send the call.

The concept of the crossbar is the crossbar mechanism operates on a matrix concept where dialed digits would change the position of the crossbar and connect to other parts of the matrix. It was still the same "level" concept of the step by step and panel but using more efficient and smaller equipment.

The first crossbar system was the Number 1 Crossbar (or #1XB for short) developed in 1938. These were in widespread use in large metropolitan areas (like Panel) where step by step equipment would not work as well.

In the late 1940's, Western Electric developed the Number 5 crossbar (or #5XB) where it was the same concept as the #1XB but an improved design. The #5XB was put into widespread use in the US by the late 1950s and 1960s. The last #5XB was installed in 1969.

The #5XB was also significant since it was later modified to handle customer dialed Dual Tone Multi-Frequency dialing in the early 1960's. This is of course something we all know as "Touch Tone".

Other crossbar systems were developed by independent telephone company manufacturers. One popular one called the NX-1 crossbar (and its smaller counterpart, the NX-2) developed by North Electric (not to be confused with Northern Electric). It was used by companies such as United Telephone and others.

Northern Electric made their own crossbar system based on the #5 crossbar concept of Western Electric.

Most of not all crossbar systems have been removed by the late 1980's and definately by the late 1990's in the North American system. There may still be some crossbar in countries outside the US and Canada.

1.2.4 Early Computer Controlled Analog Switching Systems

The first "electronic" switch that was used in the public telephone network was an experimental switch in Morris, IL in 1960. See the Telephone History pages for more details.

In the mid 1960s, Western Electric developed a computer controlled analog switch called the Number 1 Electronic Switching System or #1ESS for short. It was basically a computer controlled Number 5 Crossbar (#5XB) but using "reed relays" instead of

physical crossbars. This considerably reduced the size of the switch, improved its reliability, and made it easier to make "translation" modifications (how switches route calls) by changing software, not hardware.

The switch was also unique because of new inventions called Custom Calling Features - Call Waiting, Three-Way Calling and Speed Dialing.

Independent switch manufacturers also made computer controlled analog telephone switching systems. Automatic Electric made the #1EAX and #2EAX (Electronic Automatic eXchange) in the early and late 1970's respectively. Other companies such as Stromberg-Carlson made the ESC switch in the early 1970s.

In 1976, Western Electric made advancemnets in the #1ESS technology and produced the #1AESS system. Older #1ESS systems were modified to #1AESS and new ones made after 1976 had the new technology.

Western Electric also made other computer controlled switching systems that were an improvement on the #1ESS/#1AESS. These switches were used in areas with a small amount of customers. These switches included the #2ESS for suburban use (1970), the #2BESS (1976) and the #3ESS for rural use (1976) switches.

There are no more #2ESS/#2BESS or #3ESS switches in service, nor any old AE or Stromberg-Carlson early electronic switches in service. However, there are still a number of #1AESS switches still in use, though many are scheduled to be replaced by sometime in the early 2000s.

1.2.5 Fully Digital Switching Systems

Though the #1ESS/#1AESS switch was computer controlled, it still was an analog switch. Technology had advanced enough by the late 1970s where 100% digital systems were being developed for use in the telephone industry. Digital systems "sample" the analog signal and handle telephone calls internally as binary digits, then convert them back to analog to be compatible with regular telephones. This again made the switch smaller and more reliable. It also allowed new technologies to be added faster by using

modular techniques (adding systems to the original system without complete redesign/reinstallation) and by using advanced computer software.

The first totally digital system (for end offices) was not developed by Westen Electric but by a company called Vidar. Their first switch was developed in 1978. Northern Telecom (formerly Northern Electric, now Nortel) developed the DMS series of digital switches. In 1979, the DMS-10 was first produced. Later they started producing the DMS-100 as a local end office switch.

Western Electric developed their fully digital switch in 1982. It was called the Number 5 Electronic Switching System or #5ESS for short. It is a fully digital switch that did everything the #1AESS did and more. Though it is primarily used as a local central office, the #5ESS can be used as an operator services switch or as a low to medium traffic volume tandem. In 1984, Western Electric was absorbed into AT&T as part of divestiture. In 1996, the hardware group of AT&T was spun off into its own company - Lucent Technologies. The #5ESS switch is still made to this day by Lucent.

Other independent companies developed their own digital switches. Automatic Electric developed the #5EAX (or better known as the GTD-5) switch. Others such as Stromberg Carlson (now a unit of Siemens) developed the DCO (Digital Central Office), and Siemens made the EWSD (Electronic Worldwide Switch Digital) switch

The GTD-5 switch is no longer being made though it is still in widespread use. The #5ESS, the DMS series, DCO and the EWSD switches are still in production are in widespread use in the US, Canada - as well as many countries worldwide.

1.2.6 Packet Switching Systems

The Internet and the technologies that are used with the Internet changed the idea of how communications are handled. Traditional telephone company technologies were direct poiont-to-point systems where dedicated "trunks" (toll circuit routes) were used. Once a trunk was allocated, the trunk stayed up for the entire connection. If a trunk was not being used, it sat idle - underutilizing limited resources.

The Internet (TCP/IP and Ethernet in particular) uses a completely different pardigm (methodology). In the TCP/IP world, all systems share a common connection. Each system (usually a computer) was connected to this common system and shared "bandwidth" with other people. Resources were not allocated for just one user, but all users can share resources.

Another advantage of using TCP/IP technology is the switching "matrix" is also different and more efficient. Instead of using a dedicated switching matrix - with a limited amount of circuits in a traditional telephone switch, TCP/IP routers (traffic handlers) could be used. This increases the amount of switching "lines" and can do it more efficiently than traditional switching systems since all the switching is done on a "virtual" scale in software rather than in hardware.

Telephone systems have been slowly adopting the "IP" (Internet Protocol) technology over the last 8 to 10 years. Early experiments proved that voice can be converted to digital "packets" and sent over the Internet. The packets would be collected and converted back to analog voice. The quality of the calls was not great but it showed that it could be done. The major problem was somethig called "packet loss" which is common with TCP/IP connections.

By the early 2000s, the IP telephony (or "VoIP" - Voice over Internet Protocol) technology had improved. Using "classes" of service, reliable connections could be obtained and packet loss reduced to minimum levels. Business systems started using VoIP technology in their PBX (Private Branch Exchange) switches. The telephones themselves were almost like small computers that had their own analog/digital conversion systems and TCP/IP networking technology all the the same system. The phone could "piggy-back" on their existing computer network system. Hence having voice AND data traffic over the same wires!

Telephone companies - both local and toll - are also handling data traffic at alarming rates. AT&T is handling 5 to 10 times more data traffic than it handles for traditional voice traffic. To handle both efficiently - it would be cost effictive to have switching systems that handle data switching on a high volume scale - that can serve both the voice market AND the data market at the same time.

Lucent Technologies and Nortel Networks are both making switching systems that are known as "packet switches" that take the traditional switching system to a higher level. These systems can handle both data AND voice at the same time. These are add-ons to existing voice switches and are located at traditional central office locations.

Lucent's packet switching system is known as the 5E-XC. It builds upon the #5ESS switch and handles both data and voice.

Nortel's packet switching technology is known as the Succession family of switches. The Succession Server 2000 is now being used as a replacement or in addition to the traditional DMS series of digital switching systems. The local arm of Sprint is now installing Succession 2000 switches in various places nationwide as of the fall of 2003. This includes cities such as Las Vegas, NV.

1.3 Telephone Signalling System Technologies

A brief overview of the history and operation of telephone signalling technologies.

1.3.1 Pulses

Pulses (interrupting current on toll trunk lines), especially on inter-office trunk lines, were common in the days of step by step and crossbar systems. These were used to pulse out dialed digits, either directly or via pulse senders from a central office switch, to the destination switch. Though not in common use today, these may still do exist in rare instances outside of North America.

1.3.2 DC Voltage and DC Polarity

Voltage changes and polarity changes (as well as pulses) were used in some inter-office signalling routines. These were sometimes found in Panel systems when communicating to Panel "tandem" systems. Never was in common use and are not used anymore[1].

1.3.3 Multi-Frequency Signalling (DTMF)

Dual-tone multi-frequency (DTMF), also known as Touch Tone® is used for telephone signaling over the line in the voice frequency band to the call switching center. DTMF is an example of a multi-frequency shift keying (MFSK) system. Today DTMF is used for most call setup to the telephone exchange, at least in the Western world, and trunk signalling is now done out of band using the SS7 signaling system. The trunk signalling tones were different than the tones known as touch tone with a triangular matrix being used rather than a square matrix. See: blue box for more details on the switching tones.

Prior to DTMF the phone systems had used a series of clicks on the phone line to dial numbers, a system known as pulse dialing. The clicks were actually the connection of the calling party's phone line being made and broken, like flicking a light switch on and off. This was useful only as far as the local end office where the wires stopped, requiring operator intervention for long distance dialing.

DTMF was developed at Bell Labs in order to allow dialing signals to dial long-distance numbers, potentially over non-wire links such as microwave links or satellites. Encoder/decoders were added at the end offices that would convert the standard pulse dialing clicks into DTMF tones and play them down the line to the remote end office. At the remote site another encoder/decoder would decode the tones and turn out a series of clicks. It was as if you were connected directly to that end office, yet the signaling would work over any sort of link. This idea of using the existing network for signaling as well as the message is known as in-band signaling.

It was clear even in the late 1950s when DTMF was being developed that the future of switching lay in electronic switches, as opposed to the mechanical crossbar systems currently in use. In this case pulse dialing made no sense at any point in the circuit, and plans were made to roll DTMF out to end users as soon as possible. Various tests of the system occurred throughout the 1960s where DTMF became known as Touch Tone.

The Touch Tone system also introduced a standardized keyboard layout. After testing 18 different layouts, they eventually chose the one familiar to us today, with 1 in the upper-left and 0 at the bottom. The adding-machine layout, with 1 in the lower-left was also tried, but at that time few people used adding machines, and having the 1 at the

"start" (in European language reading order) led to fewer typing errors. In retrospect, many people consider that this was a mistake. With the widespread introduction of computers and bank machines, the phone keyboard has become "oddball", causing mistakes.

The engineers had also envisioned phones being used to access computers, and surveyed a number of companies to see what they would need for this role. This led to the addition of the pound (#) and star (*) keys, as well as a group of keys for menu selection, A, B, C and D. In the end the lettered keys were dropped from most phones, and it was many years before the # and * keys became widely used, primarily for certain vertical service codes such as *67 to suppress caller ID. Many non-telephone applications still use the alphabet keys, such as Amateur Radio repeater signaling and control.

The US military also used the letters, relabled, in their Autovon phone system. Here they were used before dialing the phone in order to give some calls priority, cutting in over existing calls if need be. The idea was to allow important traffic to get through every time. Pressing C, Immediate, before dialing would make the switch first look for any free lines, and if all lines were in use, it would hang up any non-priority calls, and then any Priority calls. While the Autovon phone system no longer exists, their original names were Flash Override (A), Flash (B), Immediate (C), and Priority (D). Pressing one of these keys gave your call priority, over-riding other conversations on the network. Flash Override is the highest priority.

Present-day uses of the A, B, C and D keys on telephone networks are few, and exclusive to network control. For example, the A key is used on some networks to cycle through different carriers at will (thereby listening in on calls). Their use is probably prohibited by most carriers.

The DTMF keypad is laid out in a 4 x 4 matrix with each row representing a low frequency, and each column representing a high frequency. Pressing a single key such as '1' will send a sinusoidal tone of the two frequencies 697 and 1209 hertz (Hz). The two tones are the reason for calling it multi-frequency. These tones are then decoded by the switching center in order to determine which key was pressed.

The frequencies were initially designed with a ratio of 21/19, which is slightly less than a whole tone, to avoid harmonics or natural occurring frequencies that could occur when the two tones are sent. The frequencies may not vary more that +/- 1.5% from their nominal frequency, or the switching center will ignore the signal. The high frequencies may be the same volume or louder as the low frequencies when sent across the line. The loudness difference between the high and low frequencies can be as large as 3 decibels (dB) and is referred to as twist.

Table 1.1 DTMF Keypad Frequencies.

1	2	3	A	**697 Hz**
4	5	6	B	**770 Hz**
7	8	9	C	**852 Hz**
*	0	#	D	**941 Hz**
1209 Hz	**1336 Hz**	**1477 Hz**	**1633 Hz**	

DTMF can be regarded as a simple form of orthogonal frequency division multiplexing. Synonyms include multifrequency pulsing and multifrequency signaling[3].

1.3.4 Common Channel Inter-Office Signaling / Signaling System 7

CCIS (Common Channel Inter-Office Signalling) started in the late 1970s as a way to send more information between tandems and switches in the network. Sending a call via MF takes up to 15 seconds on a domestic phone call. Also, MF lead itself to toll fraud since making MF tones is fairly easy to do.

CCIS was invented to send information "out of band" on a data circuit parallel to the voice circuit. This way the call can be set up and completed in a shorter amount of time, be able to send more information, and avoid toll fraud - all at the same time.

The early commonly deployed versin of CCIS was version 6. The modern version of CCIS is version 7 - commonly called SS7 (Signalling System 7) where there are 7

"layers" of the signalling network. SS7 is in widespread use in the USA, Canada and many industrialized countries in the world including the UK, Austrailia and others.

1.4 Telephone Transmission Technologies

A brief overview of the history and operation of telephone transmission technologies.

1.4.1 Open Wire Carrier

Open wire carrier was developed to carry multiple calls over a pair of copper wires simultaneously. It uses a method of frequency division multiplexing (FDM) where calls are sent on different frequencies. The frequencies are around 100Khz at about 4 Khz per channel.

Open wire carrier has long ago been removed in all areas. The last open wire carrier system I am aware of was located in rural New Mexico and removed in August of 1997.

1.4.2 Coax Cable

Coax cable was used for cross country communications. Coax cable was also used in the transmission of radio and television programming. It was also used for use by government and defense department purposes. Many repeaters were involved and many distribution facilities as well.

1.4.3 Microwave (Radio) Towers

Developed in the 1940s and 1950s, microwave transmission became a widespread telephone call transmission medium. Many microwave towers were erected in many countries worldwide. Microwave again uses frequency division multiplexing.

The microwave system is called as such because of the short wavelengths of the frequencies involved - micrometers! In terms of frequency, the range is in the gigahertz.

There are a few microwave systems in use today, but those that are still in use are converted to digital transmission techniques. Many of the old towers have either been decomissioned or are now used as cellular/PCS towers.

(Off-site Link - "The Latest Word in Communications") In 1947, AT&T inaugurated an experimental microwave radio link, connecting Boston and New York City. This brochure was published by the Long Lines Department to describe the system's technology and facilities.

1.4.4 Satellites

Most people think of satellites for television transmission. But Bell Labs/AT&T invented communications satellites for long distance communications. The first satellite was Telstar in 1962. The major drawback was the 1/2 second delay because of the distance from the Earth to the geostationary orbit over 22,000 miles from Earth and return. Satellites are still used today for far remote places.

1.4.5 Fiber Optics

Ultra pure glass optical fibers using amplitude modulated infrared light, commonly known as Fiber Optics, revolutionized telecommunications transmission techniques. Developed in the 1970s and implemented in the 1980s and beyond - fiber optics is the high bandwidth and high quality transmissions medium that is in widespread use in the telephone industry today. Almost all telephone companies in North America use fiber optics, as do many telephone companies in the industrialized world.

1.5 Telephone Tandem Switching Systems

Telephone tandems are used to "bridge" many central offices together, or used to route telephone traffic within a long distance network. Some tandem switches are dedicated pieces of equipment, while other switches are multi-function (toll and local).

1.5.1 Step by Step Tandem

A step by step switch can be used as a tandem - either as a dedicated switch or as part of a switch that also does local central office switching functions. For example, Carolina Telephone (now Sprint local) in North Carolina used specially dedicated step by step switches as a tandem to several step by step local end offices.

Step tandems were common in rural areas. Step tandems were removed when most step switches were removed in the 1980s.

1.5.2 Crossbar Tandems

A crossbar tandem was based on crossbar switch technology. Western Electric developed the "Crossbar Tandem" or XBT in 1941 for use in urban areas to handle primarily local toll traffic. Later, WECO developed the Number 4 Crossbar Toll Switch (commonly referred to as the #4XB) in 1943 for use as a full service short and long haul toll switch. The first #4XB was installed in Philadelphia.

The first series of #4XB tandems had routing "translations" hard coded in the system. In the early 1950s, new #4XB toll switch had "card translator" boxes installed to do routings that could be changed with new cards. These new tandems were called "4A" toll switches, the A for Advanced. Older #4XB toll switches were modified, calling them 4M for Modified. In 1969, new #4XB toll switch had electronic translator systems or ETS for short. Some older ones were modified to use ETS while others used card translators until they were removed from service.

The last #4XB toll switch was installed in 1976. #4XB toll switches were in service well into the late 1980s. There were over 200 4XB toll switches made between the 1940s and the 1970s. A historical chronological list of installation dates and locations can be found in our #4XB Toll Switch List.

Other crossbar switches were used as tandems, but many were also used as regular central office switches. #5XB switches from Western Electric were sometimes used as

tandems.

1.5.3 Electronic Switching System Tandems

AT&T #4ESS

In 1976, Western Electric developed the Number 4 ESS (#4ESS) (Electronic Switching System) toll tandem switch to replace the #4XB as the toll tandem used in the toll network. The first #4ESS was installed in Chicago. In later years, the #4ESS was manufactured by AT&T and later Lucent Technologies.

Primarily used in the long haul toll network, some #4ESS switches are used in regional toll traffic as well. AT&T still uses the #4ESS with over 140 #4ESS switches used in the US and Canada. Some local telephone companies use the #4ESS as a regional tandem. Similar in design to the #5ESS switch (which can be used as a medium traffic volume tandem) is a fully digital switch. The #4ESS switch can do local end office functions as well and is used as such in some areas where AT&T is operating as a CLEC (Competative Local Exchange Carrier).

The last #4ESS switch was installed in June 1999 in suburban Atlanta, Georgia. Lucent has determined that the #4ESS, while good at switching voice circuits, is not very good at switching data circuits. Emphasis will now go to the Lucent #5ESS and switches from other vendors for future tandem switching. However, the existing #4ESS switches that are in use will continue to do so for a number of years to come.

Nortel DMS tandem series

Other switch manufacturers also make digital electronic style tandem switches. The most popular are the Nortel (Northern Telecom) DMS series of switches.

DMS-200 - Low-volume/regional tandem (can be combined with TOPS (Toll Operation Position Station) for operator services). Usually as part of a DMS-100 local switch.

DMS-250 - High-volume/long-haul tandem

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ABSTRACT

The remote control over telephone line project designed to meet the need of remotely controlling the electronic devices by following a few simple procedures, in addition the project circuit has to be compact, low cost to construct and requires ordinary equipments to operate, however the remote control over telephone line circuit requires a telephone at the commander position and another one at the destination position, with the telephone line at the destination connected to one terminal of the project circuit and to the other terminals are the desired electronic devices required to be controlled.

This is achieved by building the project using both hardware and software, it is possible to build the same project using electronic components only, but a large number of components would be needed, so more power supply consumption, additional cost expenses and increased possibility for faults, but the software achieves this task with the help of a limited number of electronic components in order to lower the power consumption, decrease the project budget and the circuit to be immune for possible errors that may occur during the control process.

Because of the need to use the telephone to pass the commands in a hand, and electronic components with a software to build the project at the other hand, the basic knowledge about telecommunications, microcontrollers and electronic design are very important in order to plan and build this project.

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INTRODUCTION

The remote controlling occupies an important position in the control systems design because of the demands to control the electronic devices from a remote positions. The remote control device can be used for many applications, such as TV & VCR remote control that uses IR (infrared light) for a few meters distance control, also there are remote controls for a short and long distance using RF (radio frequency), and some other remote controls using the networks to pass the control commands[7].

The communication devices can communicate the data and also the control commands, whether wired or wireless communication systems can be used for control purpose. Telecommunication systems and techniques can contribute to remote controlling because it is communicating local and remote parties, in remote control over telephone line project the PSTN (public switching telephone network) is used to handle the commands from a remote or local position by using the DTMF (dual tone multi frequency) signaling principle, that is used to encode and decode the pressed buttons of the telephone keypad[3].

The processing of the DTMF signals needs a qualified system to handle, therefore the microcontrollers used in this project to achieve this job with the help of a DTMF decoder circuit and some other interfacing circuits. Microcontrollers need to be programmed at the low level language programming that the CPU of the microcontrollers understand.

Such a project has many applications depending on the user demands, for example controlling the home appliances such as heaters, air conditions and gates, and may be used in a factory or a big company that need to control some devices from many positions in the site[7].

CHAPTER ONE
TELEPHONE NETWORK

1.1 Introduction

The Telephone System has been developed over many years and has gone through many incremental evolutionary steps. The traditional service of the Telephone Network has been for Voice Communication and only recently the network has been used to support the high amount of data transfer we are currently experiencing. For cost effective voice communications it has been identified that the Humans can communicate at frequencies between 300Hz to 3500Hz. Though we can hear and speak at higher and lower frequencies, voice communications between 300Hz and 3500Hz are clear and efficient for the telephone network to transmit and receive, look at figure 1.1. A Voice Channel goes from 0Hz to 4000Hz and was developed to avoid any overlapping to any other adjacent voice channels.

Figure 1.1 voice channel bandwidth.

Through the Telephone Network development it has been discovered that it is more efficient to transmit a voice channel in a digital form. In a digital form the voice channel can be routed to its destination with very low noise, higher reliability and more cost effectiveness. To do this the Voice Channel is converted to Digital by an Analog to Digital Converter (A/D Converter or ADC) at the Central Office (see figure 1.2). The Central Office (CO) is the location in your area that the 2-Wires that come from your

house called Tip & Ring are terminated. Once the Voice Channel has been digitized it is transmitted over the network to the CO of the number you called. At this remote CO the Voice Channel is reconstructed back into an analog form so that remote person can understand it. This transformation back to analog is done by a Digital to Analog Converter also know as a D/A Converter or DAC.

One important point to know is that the A/D Converter samples the Voice Channel at twice the frequency of the voice channel; that is at 8Khz. The reason the sampling is done at twice the original signals frequency is due to a law called the Nyquest Rate, which states to digitize a waveform and have enough information to reconvert it back to the an analog waveform one must sample the original waveform at no less than twice the frequency of the original waveform. Also each sample is identified by 8 unique bits that represent 256 different states. 8 Bits sampled 8000 times a second is equal to 64Kbps. Figure 1.2 shows the traditional communication process[4].

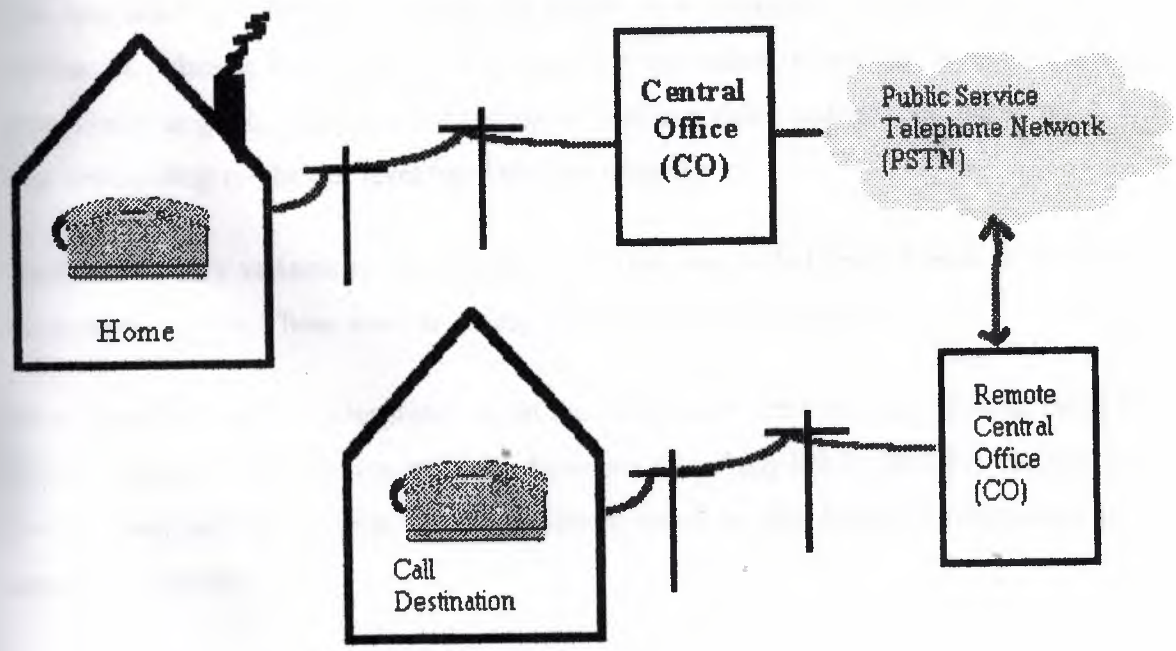


Figure 1.2 communication process.

1.2 Local Central Office Switching Systems

A brief overview of the history and operation of local central office telephone switching systems.

1.2.1 Step by Step (Strowger or SxS)

The Step by Step switching system was invented in the early 1890's by Almon Strowger, an undertaker in Kansas City, MO USA. Rumor has it that the local switchboard operator was diverting calls from his business to others. He wanted to have some way to have calls automatically routed to the destination without the intervention of an operator.

His invention had changed throughout the years, but the basic concept has remained the same. The system operates by using rotating "selectors" that can select a "level" (or step) by the number of pulses that are interrupted on a line (using a "rotary" telephone). This instructed a solenoid to move this wiper to a particular position based on the number of pulses it had received. This then put the caller on another level and again pulse dialed to get to another level and so on until the caller had reached the subscriber they were calling on the last level (or digit they dialed).

There were many variants to the step by step. One was called the XY switch made by Stromberg-Carlson. These were in widespread use in many rural areas.

These systems were in widespread use in the North American system well into the late 1980's (and some into the late 1990's!). There are few if any left in the USA or Canada, but they may still be in use in some third-world countries and former Soviet controlled countries in Europe.

1.2.2 Panel Switching Systems

The "Bell System" (Western Electric) never liked the idea of allowing customers to dial their own calls since they specialized in "customer service" and had operators complete all calls. By the late 1910's, they started to realize that this may not be a good idea and

started to use step by step systems (as described above). However, it did not work out well in large cities where there a large number of calls in progress at an one time. They developed a new system called a Panel switch. Their first switch was called a Panel switch because of the large vertical panels. It followed the same basic concept of the Step by Step but used a method of where the selector system operated on a concept of "ladders" where the selector would rise up on a panel by the number of pulses it had received.

The first one was installed in the early 1920's in Omaha, Nebraska and others were installed throughout the 20's and 30's in most metropolitan areas in the USA (except for Los Angeles, which was step by step).

Panel systems had a system of "revertive pulsing" where it communicated with another Panel switch by the receiving switch pulsing BACK to the originating switch! Instead of sending dial pulses out TO the terminating switch, it received them FROM the terminating switch.

Most Panel offices were removed from service by the late 1970's, though I hear the last one was located in northern New Jersey and was removed in the early 1980's.

1.2.3 Crossbar Systems

Crossbar systems were developed in the late 1930's (with technology from LM Ericsson) by Western Electric (the Bell System). The Crossbar had a significant advantage over either Panel or Step by Step since it was able to use a "store and forward" concept where it would take incoming digits, store them, and then process the call. It was also able to do call routing and determine where a call should be sent to by doing "translations" of the incoming digits and deciding how to send the call.

The concept of the crossbar is the crossbar mechanism operates on a matrix concept where dialed digits would change the position of the crossbar and connect to other parts of the matrix. It was still the same "level" concept of the step by step and panel but using more efficient and smaller equipment.

The first crossbar system was the Number 1 Crossbar (or #1XB for short) developed in 1938. These were in widespread use in large metropolitan areas (like Panel) where step by step equipment would not work as well.

In the late 1940's, Western Electric developed the Number 5 crossbar (or #5XB) where it was the same concept as the #1XB but an improved design. The #5XB was put into widespread use in the US by the late 1950s and 1960s. The last #5XB was installed in 1969.

The #5XB was also significant since it was later modified to handle customer dialed Dual Tone Multi-Frequency dialing in the early 1960's. This is of course something we all know as "Touch Tone".

Other crossbar systems were developed by independent telephone company manufacturers. One popular one called the NX-1 crossbar (and its smaller counterpart, the NX-2) developed by North Electric (not to be confused with Northern Electric). It was used by companies such as United Telephone and others.

Northern Electric made their own crossbar system based on the #5 crossbar concept of Western Electric.

Most of not all crossbar systems have been removed by the late 1980's and definately by the late 1990's in the North American system. There may still be some crossbar in countries outside the US and Canada.

1.2.4 Early Computer Controlled Analog Switching Systems

The first "electronic" switch that was used in the public telephone network was an experimental switch in Morris, IL in 1960. See the Telephone History pages for more details.

In the mid 1960s, Western Electric developed a computer controlled analog switch called the Number 1 Electronic Switching System or #1ESS for short. It was basically a computer controlled Number 5 Crossbar (#5XB) but using "reed relays" instead of

physical crossbars. This considerably reduced the size of the switch, improved its reliability, and made it easier to make "translation" modifications (how switches route calls) by changing software, not hardware.

The switch was also unique because of new inventions called Custom Calling Features - Call Waiting, Three-Way Calling and Speed Dialing.

Independent switch manufacturers also made computer controlled analog telephone switching systems. Automatic Electric made the #1EAX and #2EAX (Electronic Automatic eXchange) in the early and late 1970's respectively. Other companies such as Stromberg-Carlson made the ESC switch in the early 1970s.

In 1976, Western Electric made advancemnets in the #1ESS technology and produced the #1AESS system. Older #1ESS systems were modified to #1AESS and new ones made after 1976 had the new technology.

Western Electric also made other computer controlled switching systems that were an improvement on the #1ESS/#1AESS. These switches were used in areas with a small amount of customers. These switches included the #2ESS for suburban use (1970), the #2BESS (1976) and the #3ESS for rural use (1976) switches.

There are no more #2ESS/#2BESS or #3ESS switches in service, nor any old AE or Stromberg-Carlson early electronic switches in service. However, there are still a number of #1AESS switches still in use, though many are scheduled to be replaced by sometime in the early 2000s.

1.2.5 Fully Digital Switching Systems

Though the #1ESS/#1AESS switch was computer controlled, it still was an analog switch. Technology had advanced enough by the late 1970s where 100% digital systems were being developed for use in the telephone industry. Digital systems "sample" the analog signal and handle telephone calls internally as binary digits, then convert them back to analog to be compatible with regular telephones. This again made the switch smaller and more reliable. It also allowed new technologies to be added faster by using

modular techniques (adding systems to the original system without complete redesign/reinstallation) and by using advanced computer software.

The first totally digital system (for end offices) was not developed by Westen Electric but by a company called Vidar. Their first switch was developed in 1978. Northern Telecom (formerly Northern Electric, now Nortel) developed the DMS series of digital switches. In 1979, the DMS-10 was first produced. Later they started producing the DMS-100 as a local end office switch.

Western Electric developed their fully digital switch in 1982. It was called the Number 5 Electronic Switching System or #5ESS for short. It is a fully digital switch that did everything the #1AESS did and more. Though it is primarily used as a local central office, the #5ESS can be used as an operator services switch or as a low to medium traffic volume tandem. In 1984, Western Electric was absorbed into AT&T as part of divestiture. In 1996, the hardware group of AT&T was spun off into its own company - Lucent Technologies. The #5ESS switch is still made to this day by Lucent.

Other independent companies developed their own digital switches. Automatic Electric developed the #5EAX (or better known as the GTD-5) switch. Others such as Stromberg Carlson (now a unit of Siemens) developed the DCO (Digital Central Office), and Siemens made the EWSD (Electronic Worldwide Switch Digital) switch

The GTD-5 switch is no longer being made though it is still in widespread use. The #5ESS, the DMS series, DCO and the EWSD switches are still in production are in widespread use in the US, Canada - as well as many countries worldwide.

1.2.6 Packet Switching Systems

The Internet and the technologies that are used with the Internet changed the idea of how communications are handled. Traditional telephone company technologies were direct poiont-to-point systems where dedicated "trunks" (toll circuit routes) were used. Once a trunk was allocated, the trunk stayed up for the entire connection. If a trunk was not being used, it sat idle - underutilizing limited resources.

The Internet (TCP/IP and Ethernet in particular) uses a completely different pardigm (methodology). In the TCP/IP world, all systems share a common connection. Each system (usually a computer) was connected to this common system and shared "bandwidth" with other people. Resources were not allocated for just one user, but all users can share resources.

Another advantage of using TCP/IP technology is the switching "matrix" is also different and more efficient. Instead of using a dedicated switching matrix - with a limited amount of circuits in a traditional telephone switch, TCP/IP routers (traffic handlers) could be used. This increases the amount of switching "lines" and can do it more efficiently than traditional switching systems since all the switching is done on a "virtual" scale in software rather than in hardware.

Telephone systems have been slowly adopting the "IP" (Internet Protocol) technology over the last 8 to 10 years. Early experiments proved that voice can be converted to digital "packets" and sent over the Internet. The packets would be collected and converted back to analog voice. The quality of the calls was not great but it showed that it could be done. The major problem was somethig called "packet loss" which is common with TCP/IP connections.

By the early 2000s, the IP telephony (or "VoIP" - Voice over Internet Protocol) technology had improved. Using "classes" of service, reliable connections could be obtained and packet loss reduced to minimum levels. Business systems started using VoIP technology in their PBX (Private Branch Exchange) switches. The telephones themselves were almost like small computers that had their own analog/digital conversion systems and TCP/IP networking technology all the the same system. The phone could "piggy-back" on their existing computer network system. Hence having voice AND data traffic over the same wires!

Telephone companies - both local and toll - are also handling data traffic at alarming rates. AT&T is handling 5 to 10 times more data traffic than it handles for traditional voice traffic. To handle both efficiently - it would be cost effictive to have switching systems that handle data switching on a high volume scale - that can serve both the voice market AND the data market at the same time.

Lucent Technologies and Nortel Networks are both making switching systems that are known as "packet switches" that take the traditional switching system to a higher level. These systems can handle both data AND voice at the same time. These are add-ons to existing voice switches and are located at traditional central office locations.

Lucent's packet switching system is known as the 5E-XC. It builds upon the #5ESS switch and handles both data and voice.

Nortel's packet switching technology is known as the Succession family of switches. The Succession Server 2000 is now being used as a replacement or in addition to the traditional DMS series of digital switching systems. The local arm of Sprint is now installing Succession 2000 switches in various places nationwide as of the fall of 2003. This includes cities such as Las Vegas, NV.

1.3 Telephone Signalling System Technologies

A brief overview of the history and operation of telephone signalling technologies.

1.3.1 Pulses

Pulses (interrupting current on toll trunk lines), especially on inter-office trunk lines, were common in the days of step by step and crossbar systems. These were used to pulse out dialed digits, either directly or via pulse senders from a central office switch, to the destination switch. Though not in common use today, these may still do exist in rare instances outside of North America.

1.3.2 DC Voltage and DC Polarity

Voltage changes and polarity changes (as well as pulses) were used in some inter-office signalling routines. These were sometimes found in Panel systems when communicating to Panel "tandem" systems. Never was in common use and are not used anymore[1].

1.3.3 Multi-Frequency Signalling (DTMF)

Dual-tone multi-frequency (DTMF), also known as Touch Tone® is used for telephone signaling over the line in the voice frequency band to the call switching center. DTMF is an example of a multi-frequency shift keying (MFSK) system. Today DTMF is used for most call setup to the telephone exchange, at least in the Western world, and trunk signalling is now done out of band using the SS7 signaling system. The trunk signalling tones were different than the tones known as touch tone with a triangular matrix being used rather than a square matrix. See: blue box for more details on the switching tones.

Prior to DTMF the phone systems had used a series of clicks on the phone line to dial numbers, a system known as pulse dialing. The clicks were actually the connection of the calling party's phone line being made and broken, like flicking a light switch on and off. This was useful only as far as the local end office where the wires stopped, requiring operator intervention for long distance dialing.

DTMF was developed at Bell Labs in order to allow dialing signals to dial long-distance numbers, potentially over non-wire links such as microwave links or satellites. Encoder/decoders were added at the end offices that would convert the standard pulse dialing clicks into DTMF tones and play them down the line to the remote end office. At the remote site another encoder/decoder would decode the tones and turn out a series of clicks. It was as if you were connected directly to that end office, yet the signaling would work over any sort of link. This idea of using the existing network for signaling as well as the message is known as in-band signaling.

It was clear even in the late 1950s when DTMF was being developed that the future of switching lay in electronic switches, as opposed to the mechanical crossbar systems currently in use. In this case pulse dialing made no sense at any point in the circuit, and plans were made to roll DTMF out to end users as soon as possible. Various tests of the system occurred throughout the 1960s where DTMF became known as Touch Tone.

The Touch Tone system also introduced a standardized keyboard layout. After testing 18 different layouts, they eventually chose the one familiar to us today, with 1 in the upper-left and 0 at the bottom. The adding-machine layout, with 1 in the lower-left was also tried, but at that time few people used adding machines, and having the 1 at the

"start" (in European language reading order) led to fewer typing errors. In retrospect, many people consider that this was a mistake. With the widespread introduction of computers and bank machines, the phone keyboard has become "oddball", causing mistakes.

The engineers had also envisioned phones being used to access computers, and surveyed a number of companies to see what they would need for this role. This led to the addition of the pound (#) and star (*) keys, as well as a group of keys for menu selection, A, B, C and D. In the end the lettered keys were dropped from most phones, and it was many years before the # and * keys became widely used, primarily for certain vertical service codes such as *67 to suppress caller ID. Many non-telephone applications still use the alphabet keys, such as Amateur Radio repeater signaling and control.

The US military also used the letters, relabled, in their Autovon phone system. Here they were used before dialing the phone in order to give some calls priority, cutting in over existing calls if need be. The idea was to allow important traffic to get through every time. Pressing C, Immediate, before dialing would make the switch first look for any free lines, and if all lines were in use, it would hang up any non-priority calls, and then any Priority calls. While the Autovon phone system no longer exists, their original names were Flash Override (A), Flash (B), Immediate (C), and Priority (D). Pressing one of these keys gave your call priority, over-riding other conversations on the network. Flash Override is the highest priority.

Present-day uses of the A, B, C and D keys on telephone networks are few, and exclusive to network control. For example, the A key is used on some networks to cycle through different carriers at will (thereby listening in on calls). Their use is probably prohibited by most carriers.

The DTMF keypad is laid out in a 4 x 4 matrix with each row representing a low frequency, and each column representing a high frequency. Pressing a single key such as '1' will send a sinusoidal tone of the two frequencies 697 and 1209 hertz (Hz). The two tones are the reason for calling it multi-frequency. These tones are then decoded by the switching center in order to determine which key was pressed.

The frequencies were initially designed with a ratio of 21/19, which is slightly less than a whole tone, to avoid harmonics or natural occurring frequencies that could occur when the two tones are sent. The frequencies may not vary more that +/- 1.5% from their nominal frequency, or the switching center will ignore the signal. The high frequencies may be the same volume or louder as the low frequencies when sent across the line. The loudness difference between the high and low frequencies can be as large as 3 decibels (dB) and is referred to as twist.

Table 1.1 DTMF Keypad Frequencies.

1	2	3	A	**697 Hz**
4	5	6	B	**770 Hz**
7	8	9	C	**852 Hz**
*	0	#	D	**941 Hz**
1209 Hz	**1336 Hz**	**1477 Hz**	**1633 Hz**	

DTMF can be regarded as a simple form of orthogonal frequency division multiplexing. Synonyms include multifrequency pulsing and multifrequency signaling[3].

1.3.4 Common Channel Inter-Office Signaling / Signaling System 7

CCIS (Common Channel Inter-Office Signalling) started in the late 1970s as a way to send more information between tandems and switches in the network. Sending a call via MF takes up to 15 seconds on a domestic phone call. Also, MF lead itself to toll fraud since making MF tones is fairly easy to do.

CCIS was invented to send information "out of band" on a data circuit parallel to the voice circuit. This way the call can be set up and completed in a shorter amount of time, be able to send more information, and avoid toll fraud - all at the same time.

The early commonly deployed versin of CCIS was version 6. The modern version of CCIS is version 7 - commonly called SS7 (Signalling System 7) where there are 7

"layers" of the signalling network. SS7 is in widespread use in the USA, Canada and many industrialized countries in the world including the UK, Austrailia and others.

1.4 Telephone Transmission Technologies

A brief overview of the history and operation of telephone transmission technologies.

1.4.1 Open Wire Carrier

Open wire carrier was developed to carry multiple calls over a pair of copper wires simultaneously. It uses a method of frequency division multiplexing (FDM) where calls are sent on different frequencies. The frequencies are around 100Khz at about 4 Khz per channel.

Open wire carrier has long ago been removed in all areas. The last open wire carrier system I am aware of was located in rural New Mexico and removed in August of 1997.

1.4.2 Coax Cable

Coax cable was used for cross country communications. Coax cable was also used in the transmission of radio and television programming. It was also used for use by government and defense department purposes. Many repeaters were involved and many distribution facilities as well.

1.4.3 Microwave (Radio) Towers

Developed in the 1940s and 1950s, microwave transmission became a widespread telephone call transmission medium. Many microwave towers were erected in many countries worldwide. Microwave again uses frequency division multiplexing.

The microwave system is called as such because of the short wavelengths of the frequencies involved - micrometers! In terms of frequency, the range is in the gigahertz.

There are a few microwave systems in use today, but those that are still in use are converted to digital transmission techniques. Many of the old towers have either been decomissioned or are now used as cellular/PCS towers.

(Off-site Link - "The Latest Word in Communications") In 1947, AT&T inaugurated an experimental microwave radio link, connecting Boston and New York City. This brochure was published by the Long Lines Department to describe the system's technology and facilities.

1.4.4 Satellites

Most people think of satellites for television transmission. But Bell Labs/AT&T invented communications satellites for long distance communications. The first satellite was Telstar in 1962. The major drawback was the 1/2 second delay because of the distance from the Earth to the geostationary orbit over 22,000 miles from Earth and return. Satellites are still used today for far remote places.

1.4.5 Fiber Optics

Ultra pure glass optical fibers using amplitude modulated infrared light, commonly known as Fiber Optics, revolutionized telecommunications transmission techniques. Developed in the 1970s and implemented in the 1980s and beyond - fiber optics is the high bandwidth and high quality transmissions medium that is in widespread use in the telephone industry today. Almost all telephone companies in North America use fiber optics, as do many telephone companies in the industrialized world.

1.5 Telephone Tandem Switching Systems

Telephone tandems are used to "bridge" many central offices together, or used to route telephone traffic within a long distance network. Some tandem switches are dedicated pieces of equipment, while other switches are multi-function (toll and local).

1.5.1 Step by Step Tandem

A step by step switch can be used as a tandem - either as a dedicated switch or as part of a switch that also does local central office switching functions. For example, Carolina Telephone (now Sprint local) in North Carolina used specially dedicated step by step switches as a tandem to several step by step local end offices.

Step tandems were common in rural areas. Step tandems were removed when most step switches were removed in the 1980s.

1.5.2 Crossbar Tandems

A crossbar tandem was based on crossbar switch technology. Western Electric developed the "Crossbar Tandem" or XBT in 1941 for use in urban areas to handle primarily local toll traffic. Later, WECO developed the Number 4 Crossbar Toll Switch (commonly referred to as the #4XB) in 1943 for use as a full service short and long haul toll switch. The first #4XB was installed in Philadelphia.

The first series of #4XB tandems had routing "translations" hard coded in the system. In the early 1950s, new #4XB toll switch had "card translator" boxes installed to do routings that could be changed with new cards. These new tandems were called "4A" toll switches, the A for Advanced. Older #4XB toll switches were modified, calling them 4M for Modified. In 1969, new #4XB toll switch had electronic translator systems or ETS for short. Some older ones were modified to use ETS while others used card translators until they were removed from service.

The last #4XB toll switch was installed in 1976. #4XB toll switches were in service well into the late 1980s. There were over 200 4XB toll switches made between the 1940s and the 1970s. A historical chronological list of installation dates and locations can be found in our #4XB Toll Switch List.

Other crossbar switches were used as tandems, but many were also used as regular central office switches. #5XB switches from Western Electric were sometimes used as

tandems.

1.5.3 Electronic Switching System Tandems

AT&T #4ESS

In 1976, Western Electric developed the Number 4 ESS (#4ESS) (Electronic Switching System) toll tandem switch to replace the #4XB as the toll tandem used in the toll network. The first #4ESS was installed in Chicago. In later years, the #4ESS was manufactured by AT&T and later Lucent Technologies.

Primarily used in the long haul toll network, some #4ESS switches are used in regional toll traffic as well. AT&T still uses the #4ESS with over 140 #4ESS switches used in the US and Canada. Some local telephone companies use the #4ESS as a regional tandem. Similar in design to the #5ESS switch (which can be used as a medium traffic volume tandem) is a fully digital switch. The #4ESS switch can do local end office functions as well and is used as such in some areas where AT&T is operating as a CLEC (Competative Local Exchange Carrier).

The last #4ESS switch was installed in June 1999 in suburban Atlanta, Georgia. Lucent has determined that the #4ESS, while good at switching voice circuits, is not very good at switching data circuits. Emphasis will now go to the Lucent #5ESS and switches from other vendors for future tandem switching. However, the existing #4ESS switches that are in use will continue to do so for a number of years to come.

Nortel DMS tandem series

Other switch manufacturers also make digital electronic style tandem switches. The most popular are the Nortel (Northern Telecom) DMS series of switches.

DMS-200 - Low-volume/regional tandem (can be combined with TOPS (Toll Operation Position Station) for operator services). Usually as part of a DMS-100 local switch.

DMS-250 - High-volume/long-haul tandem

DMS-300 - International gateway

DMS-500 - A combination of DMS-100 (local switch) with a DMS-200 (medium volume/regional tandem) and DMS-250 (high-volume/long-haul tandem).

1.6 Telephone Elements

In 1876, Alexander Graham Bell applied for a patent for the telephone. The first, simple application consisted of two battery-powered devices placed in separate rooms and connected by one direct line. By turning a crank to generate a current in one of the devices, the user caused a signal to buzz in the other device. One day, Bell's assistant heard not only that signal but also the first words spoken over a telephone:

Today, the telephone is powered by the local exchange. The schematic diagram in Figure 1.3 illustrates the principle of the standard version of the telephone. Somewhat simplified, it can be said to consist of four units:

- the bell and a series capacitor;
- the hook switch;
- the keypad (or dial); and
- the speech circuit with the receiver and microphone.

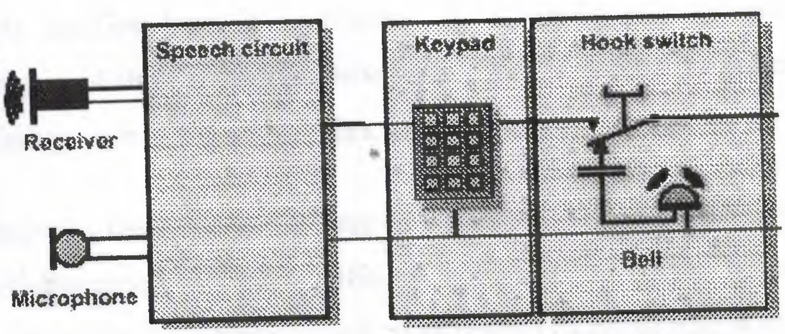


Figure 1.3 Schematic diagram of a keypad telephone.

1.6.1 The bell

The bell is connected via the capacitor when the receiver is resting in its cradle (on hook). When a call is placed to the B-subscriber, the bell is energised via the capacitor by an alternating voltage (approximately 90 V, 25 Hz), producing a ringing signal that notifies the subscriber of the incoming call.

1.6.2 The hook switch

When the A-subscriber lifts the receiver to place a call, the speech circuit and keypad are connected (and the bell is disconnected) via the hook function. This alerts the local exchange that a number is about to be dialled: the B-subscriber number. When the B-subscriber lifts the receiver to answer, the hook switch disconnects the bell in his telephone and instead connects the speech circuit and keypad. Since this closes the subscriber line, current from the local exchange can be fed to the line - an indication that the B-subscriber has answered. The parties can commence their conversation.

1.6.3 keypad

The keypad of a modern telephone is connected to a tone generator, an electronic circuit that translates keyed inputs to tone codes. Each of the digits and each of the "star" (*) and "hash" (#) function keys is represented by a combination of two tones. The frequency of the oscillators is selected whenever a key is pressed to generate the dual-tone combination unique to the digit or function in question.

Figure 1.4 illustrates the principle of keypad signalling. The standard is referred to as dual-tone multi-frequency (DTMF). Different combinations of the seven frequencies (the tones) represent the 12 symbols found on an ordinary keypad telephone.

Some modern telephones also have a function key marked with an "R" (register button). Its function (register recall) is to generate a single pulse.

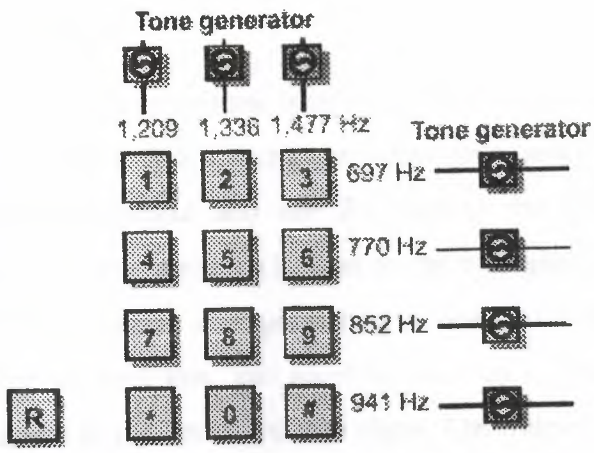


Figure 1.4 Schematic diagram of a keypad and its frequencies.

1.6.4 The Dial

Older telephones have dials instead of keypads. Although still common in many countries, these telephones represent just a few percent of all telephones sold today. The principle of the dialling function is of historical interest, so we will briefly discuss it.

The dial creates a pulse train (signals) containing information to the local exchange. The circuit connecting the exchange and the telephone is closed during the entire digit-sending process, but a contact disconnects the speech circuit during each pulse sequence. (The pulses would otherwise be heard as interference, as "clicks", in the receiver.)

The contact connected to the dial consists of a toothed wheel and two contact tongues. When the dial is released (after being wound up), the wheel starts to rotate, alternately breaking and closing the circuit. Every break results in a pulse, and the number of pulses indicates the digit dialled by the subscriber. Each of the digits forms a pulse train that is detected by the local exchange. Interestingly, Sweden is the only country that has zero as the first digit on the dial. The dials of other countries have zero following the nine.

1.6.5 The Speech Circuit

The primary function of the speech circuit is to adapt the sound level of incoming voice, outgoing voice and sidetone. The circuit comprises two amplifier blocks (one for amplifying the microphone current and one for feeding the receiver) and a bridge connection that separates voice signals to be sent to the microphone and to the receiver. Since the degree of amplification is regulated by a control circuit, transmission and reception distortion can be kept low, and amplification can be maintained constant for subscriber line resistances in the interval 0-900 ohms. Line impedance and the sidetone produced by the caller's voice are adapted by the balance circuit.

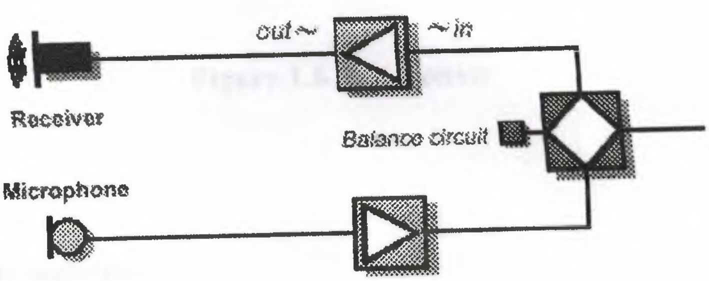


Figure 1.5 Speech circuit.

The speech circuit of older telephones was constructed in a simpler fashion, consisting in principle of only a microphone (usually a carbon-type microphone) and a dynamic receiver. Modern speech circuits provide numerous advantages.

- Sound-level attenuation over long-distance connections is counteracted by line-current-controlled regulation of the speech circuit amplifier.
- Accurate bridge balance and speech circuit impedance enhance sidetone characteristics and optimise the impedance of the apparatus.
- Transmission distortion is negligible.

1.6.6 The receiver

In principle, the design of the receiver is still based on traditional techniques. The current generated by the incoming speech passes through an electromagnet that is constructed around a permanent magnet and connected to a membrane. The oscillations, or movement, of the membrane are converted to sound waves that are perceived by the ear.

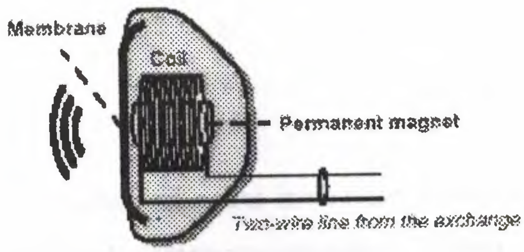


Figure 1.6 The receiver.

1.6.7 The Microphone

The old carbon microphones are being increasingly replaced by electret microphones. The material upon which these new microphones are based consists of a thin plastic film, similar to Teflon, that is exposed to a strong electrical field. The film retains its negative and positive charges after the external electrical field is removed - somewhat analogous to the poles of a magnet.

The principle of operation of the electret microphone is illustrated in Figure 1.7. The Teflon film (electret material) is stretched over a fixed electrode.

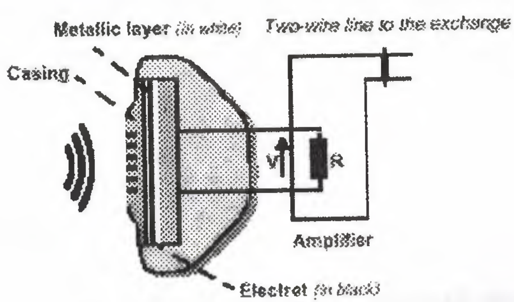


Figure 1.7 Electric microphone.

Irregularities in the surface of the fixed electrode cause a number of small air gaps to arise between the electret and the fixed electrode. The electret microphone can therefore be said to consist of a number of small parallel-connected microphones. The electrical field existing in each of the air gaps is generated by the electret's charge. The movements of the membrane change the size of the air gaps and hence their capacitance. These capacitance variations result in voltage variations that appear across the load resistor[2].

1.7 Standards

Many international PSTN recommendations have been developed during the past years, notably by the International Telecommunication Union - Telecommunications Standardization Sector (ITU-T, previously the CCITT). They are not all-inclusive. Traditionally, the PSTN has also been the subject of national standardization efforts in a number of areas:

- signalling, both in an operator's network and to subscribers;
- the billing and pricing of services;
- service offering and service procedures; and
- the physical access interface.

These local standards have led to a multitude of product versions, so vendors of switching equipment must design their systems to allow for extensive parameterization. Some parameters can be set to their default values. Nevertheless, operators and vendors must exchange a good deal of information to adapt the equipment for delivery.

For modern systems and networks, like GSM and N-ISDN, the need for parameterization is significantly less.

1.8 Network hierarchy

In the following subsections we will mention some network elements and functions that are relevant to PSTN connection set-up.

1.8.1 Central and remote subscriber stages

As mentioned earlier, the trend is towards increasingly larger local exchanges. Subscribers are often connected via remote subscriber stages which, from a switching point of view, perform exactly the same functions as those performed by centrally located subscriber stages.

Signals are sent to the control function in the local exchange, even in the case of a remote subscriber stage. However, a function is ordinarily provided for handling internal calls, should there be a break on the links to the main exchange. When these links are down, information about which services the various subscribers have access to is lost and their use cannot be charged for.

The remote subscriber stage is becoming an increasingly common component in access networks.

1.8.2 Alternative routing

In the traditional PSTN exchange hierarchy, traffic has been routed to direct links (high-congestion routes) and if these links have been busy, then the next higher level in the hierarchy (low-congestion routes) has been used. New routing functions are now available thanks to SS7 and the TUP protocol. One example is the possibility of preventing rerouting further on in the network and instead trying an alternative route all the way from the originating exchange. Another example is placing subscribers in different categories; emergency services, for example, could have access to a number of alternative routes or even routes designated for their exclusive use.

1.8.3 Semi-Permanent Connections

A connection which must not be congested and which must have good transmission characteristics can be set up through the group switch using commands. Such connections are referred to as semi-permanent connections and can utilize different paths through the exchange hierarchy.

Semi-permanent connections are used to connect SS7 signalling terminals with their dedicated time slots. These connections run either from one local exchange to another or to an exchange higher up the hierarchy that serves as a signal transfer point (STP). Semi-permanent connections are also used to create an internal network for a company by setting up leased lines between the company's PBXs. Leased lines can also be handled exclusively by the transport network.

The transmission quality of modem data connections can be guaranteed through the use of leased lines. Avoiding the time-consuming connection set-up phase in data transmission is an added advantage of leased lines.

Leased lines not only run from one exchange to the next but can connect many exchanges at different levels in a hierarchy; for example, to link the offices of a single company in several different countries.

1.9 Synchronization

In the early suppressed carrier systems, it was necessary that the frequency at the demodulator should match that of the carrier signal so that the baseband signals could be recovered without exhibiting an excess frequency shift. Tests performed as late as the 1940's showed that any shift of less than 2 Hz was acceptable for J and K carrier systems (both multiplexed 12 channels).[4]

When Pulse Code Modulation (PCM) and Time Division Multiplexing (TDM) in the form of T1 carrier was implemented, a new set of problems emerged. Now the channels of the system were separated by time with a sample for each channel repeating every 125us.

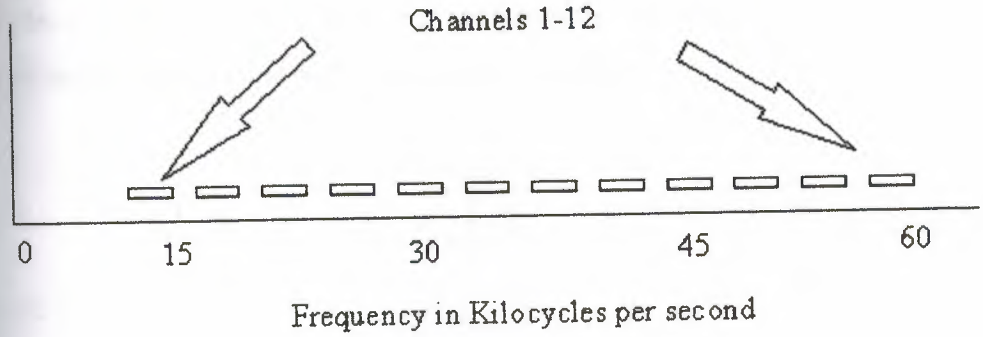


Figure 1.8 Carrier system.

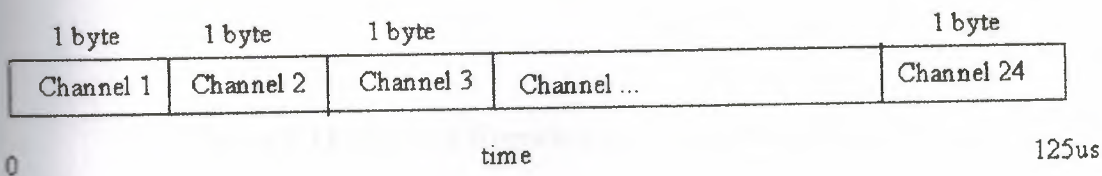


Figure 1.9 Carrier System.

T1 requires a method of phase control so that the receiver can tell the difference between channels for each frame. If the receiver becomes out of phase with the sending signal, data samples could be applied to the wrong channel. Frequency control is also required between the sender and receiver so that the receiver can differentiate each bit for the channels. If the signals are not in proper sequence, data will be lost caused by what is known as **slip.** Figure 1.10 shows an example of a receiver whose frequency is higher than that of the sender. Notice that the receiver checks for samples too often and ends up with the wrong data.

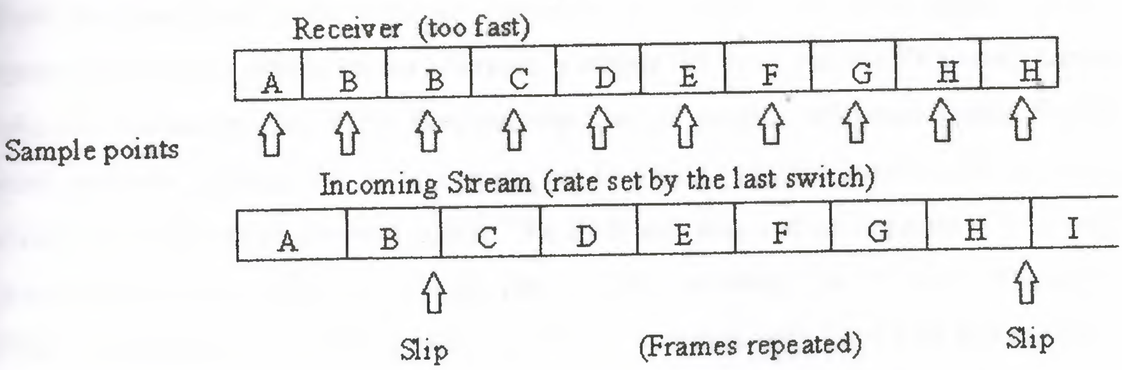


Figure 1.10 Receiver frequency higher than the sender.

When the receiver frequency is slower than that of the incoming stream, the receiver does not sample enough and as a result data is lost[3].

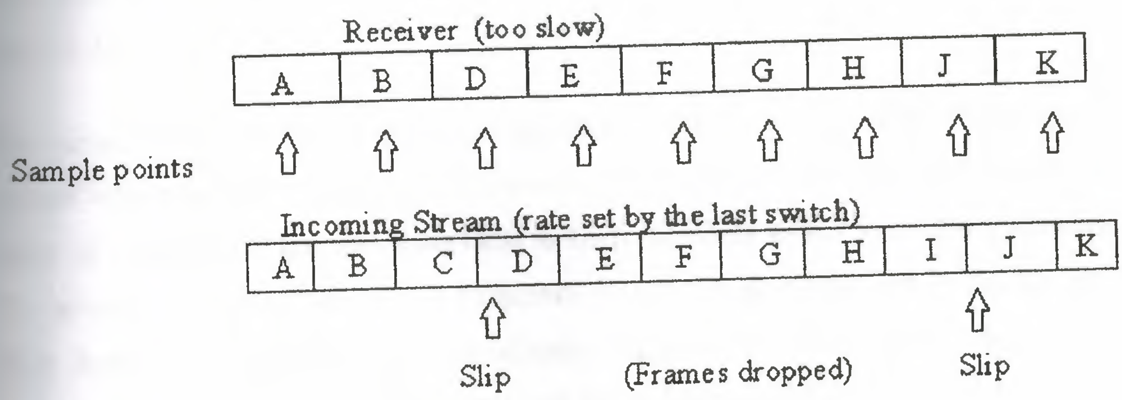


Figure 1.11 Receiver frequency is slower than the sender.

1.9.1 Synchronization Implementation

PFS-1

Before coaxial cable was introduced into the PSTN (pre 1935) modulation frequencies were held to less than 0.5Mhz. Independent tuning fork controlled oscillators were used and able to supply accurate and stable carriers having a frequency accuracy of about 1 in 10^6. [2] However with the introduction of coaxial cable came a corresponding jump in usable frequencies along with a need for higher frequency accuracies. Better accuracy was accomplished by transmitting a separate synchronizing 64 kHz pilot over the line from the originating office terminal. The pilot was merely a 64 kHz signal that was transmitted to each offices primary frequency supply (PFS) so that the PFS could have a reference frequency. The PFS's function was then to provide reference signals for the local multiplex carriers and to regenerate the incoming synchronization pilot so that it could be passed on to the next office. The PFS was adjusted as required if its locally generated frequency did not match that of the incoming 64kHz pilot frequency. PFS-1 was the early implementation of PFS and operated with the L1 (5 groups of 12 channels inside super groups of 10) coaxial system using a 64kHz pilot. This system was capable of maintaining an accuracy of less than 7 parts in 10^7. However as the multiplexing supergroups grew with the introduction of L3 (3 master groups of 600 channels), the PFS-1 system was adapted to use a 308 kHz pilot. New high quality temperature controlled oscillators were required to maintain accuracy to within a few parts in 10^8.

The errors stated are based on frequency offset relative to each other rather than a system wide absolute frequency. The system wide frequency accuracy was only on the order of 1 part in 10^6 but was acceptable to keep the pilots within their filter passbands. To maintain absolute frequency, a frequency standard was established in Murray Hill, New Jersey. This standard was periodically adjusted to match that of the national standard at the US Bureau of Standards and Navy. Murray Hill provided a 4 kHz signal to the Long Lines building in New York where the synchronization pilot was generated and transmitted to all other offices. The Long Lines supply was able to maintain an accuracy with Murray Hill on the order of a few parts in 10^9.

PFS-2

The PFS-1 method worked well for a while, but as the system expanded with the growing population, a pilot could be regenerated as many as 20 times. PFS-1 also used mechanical servo motors which moved a variable capacitor to correct pilot frequency differences at the terminal offices. A faster response and more accurate system was necessary. PFS-2 became the successor and utilized a phase locked loop which ensured zero frequency offset between the incoming and regenerated pilot. If the incoming pilot were to be lost the PFS-2 would then run free at the frequency of the local crystal oscillator. PFS-2 was widely installed the 1960's and early 1970's. Soon new problems began to arise as a result of the zero offset. Facility switching or maintenance could temporarily interrupt pilots and introduce transients into the system.

These transients would be propagated through the PFS carrier supplies where modulation would place the transients in the signal paths. The older slower PFS-1 systems had not been quick enough to respond to these pilot changes and were thus unaffected. While these transients were of little effect on speech, they could cause errors with data transmission.

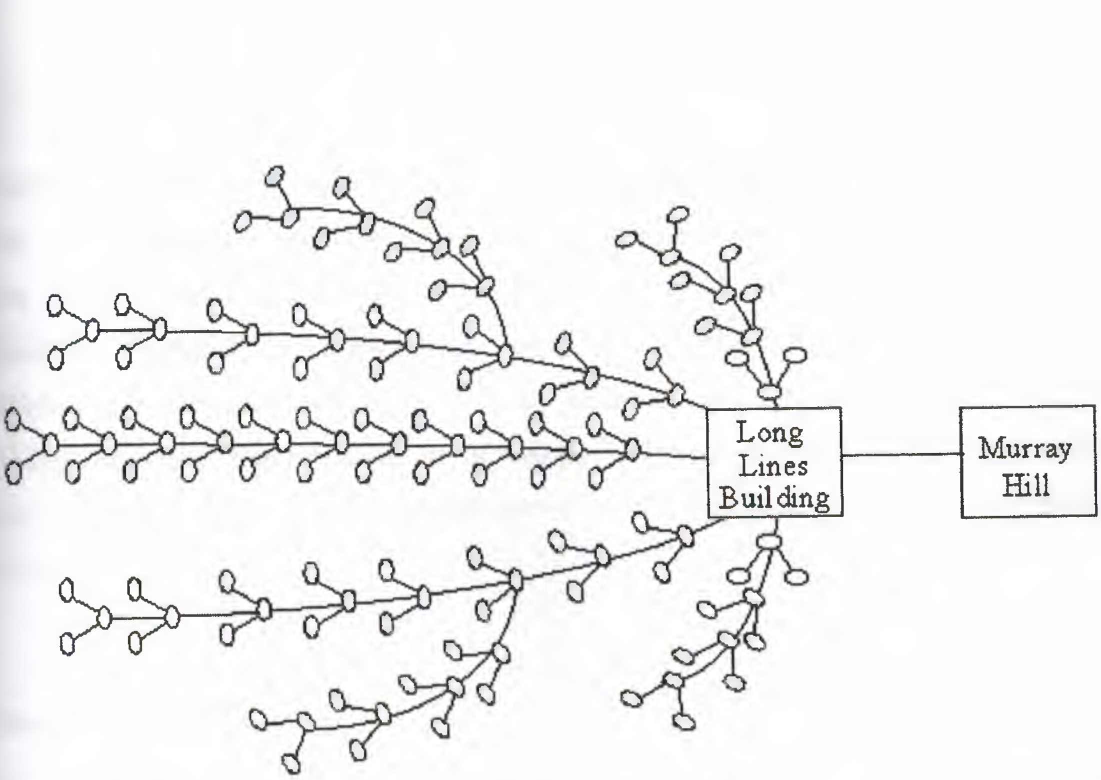


Figure 1.12 Conceptual view of the network branching of the 64kHz pilot from the Long Lines Building.

To compound this, PFS-2 was redesigned to operate with the new L4 system which had a top frequency of 17.5Mhz. The error between two L4 PFS's had to be less than one part in 10^7 and the PFS-2 could not maintain this if the synchronization pilot was lost and the PFS was free running. Also the number of consecutive PFS's continued to grow. Finally with the new L5 systems the frequency accuracy requirements could no longer be met by the PFS's and it was apparent a new system was needed.

JFS

The jumbo frequency supply (JFS) was the approach taken for synchronization for L5 systems around 1974. It became a reference supply for different regions in the country. It consisted of three crystal oscillators which when free running only had a drift of 1 part in 10^{10} per day. This allowed it to run for several weeks without adjustment. The oscillator handled transients from the incoming reference signal by quasi-frequency lock. Quasi-frequency lock (also refered to as plesiosynchronous) means that the compared frequency signals are maintained nearly synchronous. Cycles that were different would be counted with no correction made. When the count reached 256 a correction of only 2 parts in 10^{10} was made in the proper direction. When very large differences between the local and incoming signal occurred, the regional supply would run free. Regional supplies would normally run within 3 parts in 10^{10} .[1] The JFS also needed an improved reference signal. A new Bell System reference frequency standard was implemented using three Cesium atomic clocks located in Hillsborough, Missouri which maintain an accuracy of a few parts in 10^{12} with the national standard. The reference signal was transmitted to the JFS's using coaxial cable and microwave radio. The JFS's then passed on the reference signal to the traditional PFS's.

Stratum Levels

The North American network is now modeled on four stratum levels. Each level refers to the accuracy of the oscillator in it. A clock in a stratum is able to phase lock with any clock in the same or superior (lower numbered) stratum. The primary reference source (PRS) is Stratum 1 and is the highest level with the best accuracy (1 part in 10^{11}). This accuracy can only be met by a Cesium clock either onsite, via Loran-C or via global positioning system (GPS).

Table 1.2 strata requirements.

Stratum Levels	Accuracy	Minimum Stability
Stratum 1	1×10^{-11}	N/A
Stratum 2	1.6×10^{-8} (.0025 Hz at 1.544 MHz)	1×10^{-10}/day
Stratum 3	4.6×10^{-6} (7 Hz at 1.544 MHz)	< 255 slips on any connecting link during the initial 24 hours
Stratum 4	32×10^{-6} (50 Hz at 1.544 MHz)	N/A

The list of strata requirements as stated in ANSI T1.101 -1987 are contained in Table 1.[3]

By ensuring that all Stratum 1 oscillators are extremely accurate and are matched to the same reference (a world standard) different networks that contain separate Stratum 1 sources can be connected without frequency synchronization problems. The CCITT Rec. G.811 recommended that a primary reference clock be used for international switching centers. The clock should not have a longterm frequency departure of greater than 1×10^{-10} and should use Coordinated Universal Time (UTC) as its reference. Using this method the theoretical slip rate on any 64 kbps channel should not be greater than one in 70 days. (Note that this slip rate is based on undisturbed conditions.) In this way connection between separate networks would not require transferring timing information. Each network controlled by its own Stratum 1 clock should ensure that their connection together is synchronous because they are timed to the same reference clock. This is also the method used to connect different networks (eg ATT and MCI).

Timing information is distributed through this network using the T1 carrier signal. The timing information can be derived from the framing rate or bit rate of the signal since it is known to have a 1.544 Mbps transfer rate. It can be framed as all 1's or carry traffic but either way is traceable back to the Stratum 1 clock signal[4].

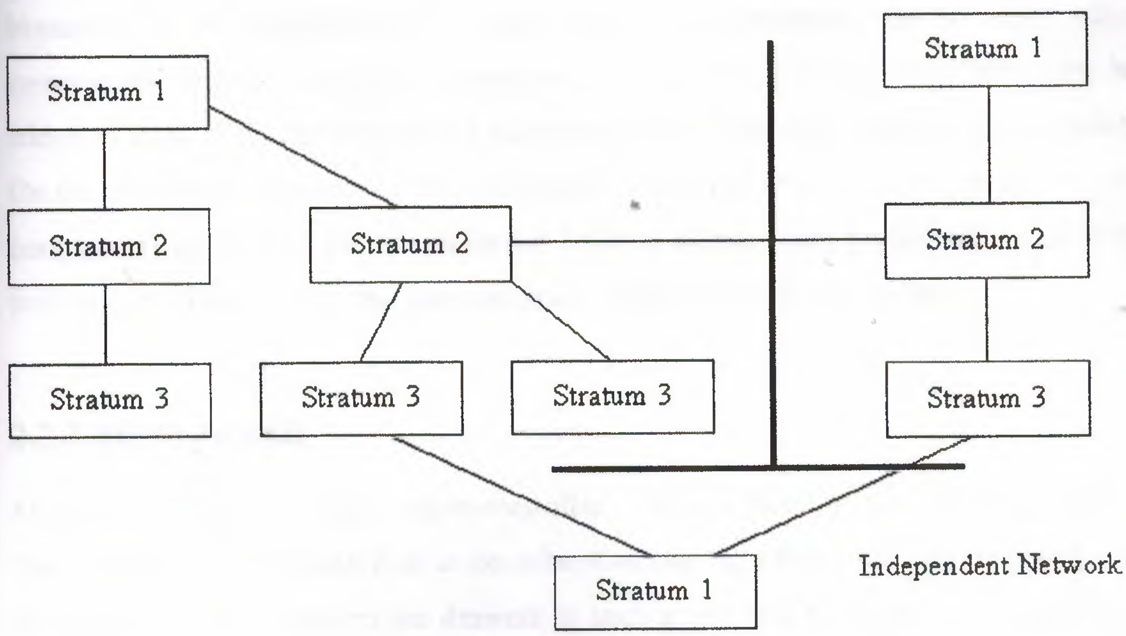


Figure 1.13 Stratum level network.

CHAPTER TWO
PIC MICROCONTROLLERS

2.1 Introduction to Microcontrollers

Circumstances that we find ourselves in today in the field of microcontrollers had their beginnings in the development of technology of integrated circuits. This development has made it possible to store hundreds of thousands of transistors into one chip. That was a prerequisite for production of microprocessors , and the first computers were made by adding external peripherals such as memory, input-output lines, timers and other. Further increasing of the volume of the package resulted in creation of integrated circuits. These integrated circuits contained both processor and peripherals. That is how the first chip containing a microcomputer , or what would later be known as a microcontroller came about.

2.2 Microcontrollers versus Microprocessors

Microcontroller differs from a microprocessor in many ways. First and the most important is its functionality. In order for a microprocessor to be used, other components such as memory, or components for receiving and sending data must be added to it. In short that means that microprocessor is the very heart of the computer. On the other hand, microcontroller is designed to be all of that in one. No other external components are needed for its application because all necessary peripherals are already built into it. Thus, we save the time and space needed to construct devices.

2.2.1 Memory unit

Memory is part of the microcontroller whose function is to store data. The easiest way to explain it is to describe it as one big closet with lots of drawers. If we suppose that we marked the drawers in such a way that they can not be confused, any of their contents will then be easily accessible. It is enough to know the designation of the drawer and so its contents will be known to us for sure.

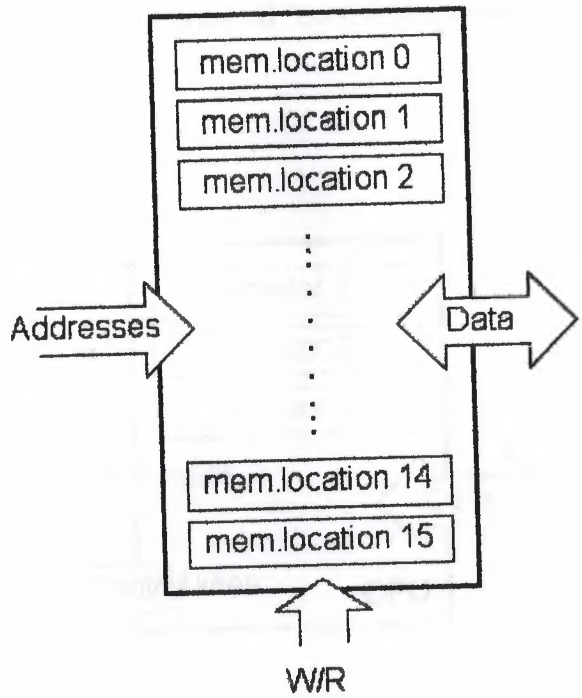


Figure 2.1 Example of simplified model of a memory unit.

Memory components are exactly like that. For a certain input we get the contents of a certain addressed memory location and that's all. Two new concepts are brought to us: addressing and memory location. Memory consists of all memory locations, and addressing is nothing but selecting one of them. This means that we need to select the desired memory location on one hand, and on the other hand we need to wait for the contents of that location. Beside reading from a memory location, memory must also provide for writing onto it. This is done by supplying an additional line called control line. We will designate this line as R/W (read/write). Control line is used in the following way: if r/w=1, reading is done, and if opposite is true then writing is done on the memory location. Memory is the first element, and we need a few operation of our microcontroller .

2.2.2 Central Processing Unit

Let add 3 more memory locations to a specific block that will have a built in capability to multiply, divide, subtract, and move its contents from one memory location onto another. The part we just added in is called "central processing unit" (CPU). Its memory locations are called registers.

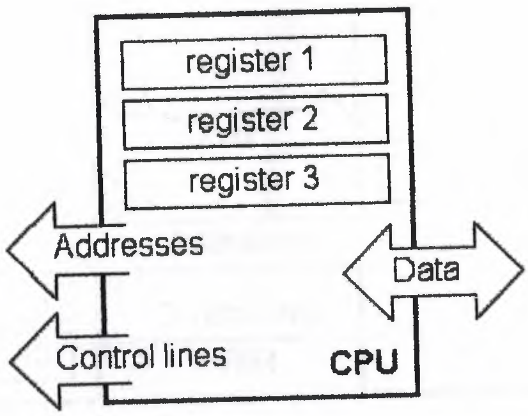


Figure 2.2 Example of simplified central processing unit with three registers.

Registers are therefore memory locations whose role is to help with performing various mathematical operations or any other operations with data wherever data can be found. Look at the current situation. We have two independent entities (memory and CPU) which are interconnected, and thus any exchange of data is hindered, as well as its functionality. If, for example, we wish to add the contents of two memory locations and return the result again back to memory, we would need a connection between memory and CPU. Simply stated, we must have some "way" through data goes from one block to another.

2.2.3 Bus

That "way" is called "bus". Physically, it represents a group of 8, 16, or more wires There are two types of buses: address and data bus. The first one consists of as many lines as the amount of memory we wish to address, and the other one is as wide as data, n our case 8 bits or the connection line. First one serves to transmit address from CPU emory, and the second to connect all blocks inside the microcontroller.

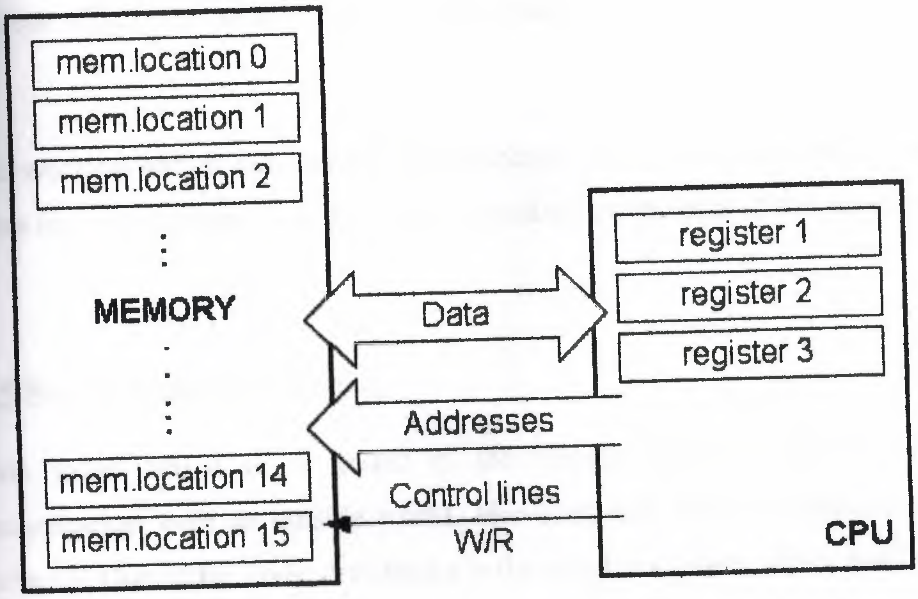


Figure 2.3 connecting memory and central unit.

As far as functionality, the situation has improved, but a new problem has also appeared: we have a unit that's capable of working by itself, but which does not have any contact with the outside world, or with us! In order to remove this deficiency, let's add a block which contains several memory locations whose one end is connected to the data bus, and the other has connection with the output lines on the microcontroller which can be seen as pins on the electronic component.

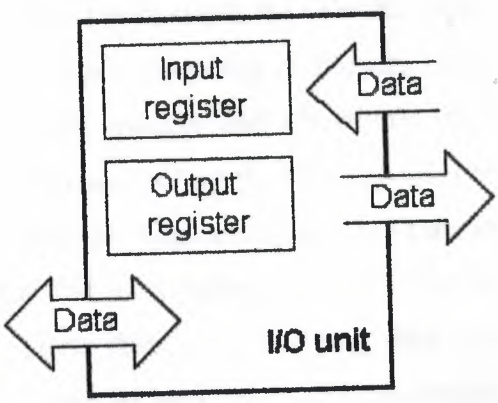


Figure 2.4 example of a simplified input-output unit provides communication.

2.2.4 Input-Output unit

Those locations we've just added are called "ports". There are several types of ports : input, output or bidirectional ports. When working with ports, first of all it is necessary to choose which port we need to work with, and then to send data to, or take it from the port.

When working with it the port acts like a memory location. Something is simply being written into or read from it, and it could be noticed on the pins of the microcontroller.

2.2.5 Serial Communication

Beside stated above we've added to the already existing unit the possibility of communication with an outside world. However, this way of communicating has its drawbacks. One of the basic drawbacks is the number of lines which need to be used in order to transfer data. What if it is being transferred to a distance of several kilometers? The number of lines times number of kilometers doesn't promise the economy of the project. It leaves us having to reduce the number of lines in such a way that we don't lessen its functionality. Suppose we are working with three lines only, and that one line is used for sending data, other for receiving, and the third one is used as a reference line for both the input and the output side. In order for this to work, we need to set the rules of exchange of data.

These rules are called protocol. Protocol is therefore defined in advance so there wouldn't be any misunderstanding between the sides that are communicating with each other. For example, if one man is speaking in French, and the other in English, it is highly unlikely that they will quickly and effectively understand each other. Let's suppose we have the following protocol. The logical unit "1" is set up on the transmitting line until transfer begins. Once the transfer starts, we lower the transmission line to logical "0" for a period of time (which we will designate as T), so the receiving side will know that it is receiving data, and so it will activate its mechanism for reception. Let's go back now to the transmission side and start putting logic zeros and ones onto the transmitter line in the order from a bit of the lowest value to a bit of the highest value. Let each bit stay on line for a time period which is equal to T, and in the end, or after the 8th bit, let us bring the logical unit "1" back on the line which will mark the end of the transmission of one data. The protocol we've just described is called in professional literature NRZ (Non-Return to Zero).

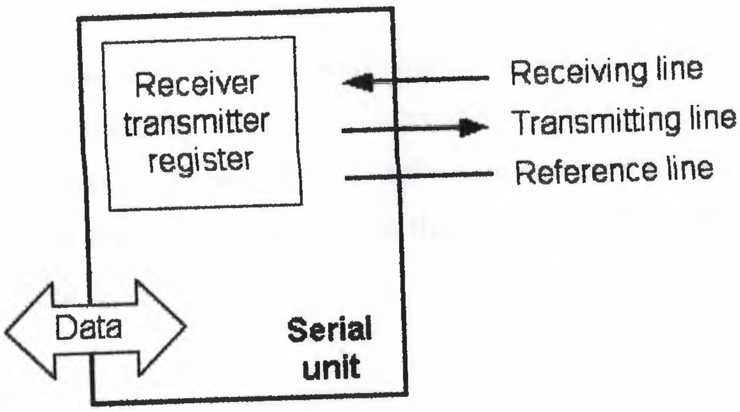


Figure 2.5 serial unit used to send data, but only by three lines.

As we have separate lines for receiving and sending, it is possible to receive and send data (info.) at the same time. So called full-duplex mode block which enables this way of communication is called a serial communication block. Unlike the parallel transmission, data moves here bit by bit, or in a series of bits what defines the term serial communication comes from. After the reception of data we need to read it from the receiving location and store it in memory as opposed to sending where the process is reversed. Data goes from memory through the bus to the sending location, and then to the receiving unit according to the protocol.

2.2.6 Timer Unit

Since we have the serial communication explained, we can receive, send and process data. However, in order to utilize it in industry we need a few additionally blocks. One of those is the timer block which is significant to us because it can give us information about time, duration, protocol etc.

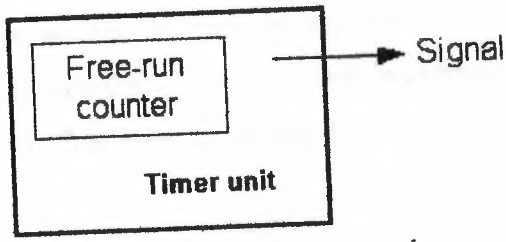


Figure 2.6 timer unit generates signals in regular time intervals.

The basic unit of the timer is a free-run counter which is in fact a register whose numeric value increments by one in even intervals, so that by taking its value during periods T1 and T2 and on the basis of their difference we can determine how much time has elapsed. This is a very important part of the microcontroller whose understanding requires most of our time.

2.2.7 Watchdog

One more thing is requiring our attention is a flawless functioning of the microcontroller during its run-time. Suppose that as a result of some interference (which often does occur in industry) our microcontroller stops executing the program, or worse, it starts working incorrectly.

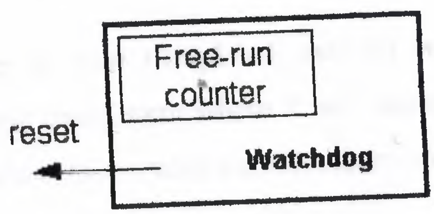


Figure 2.7 watchdog reset.

Of course, when this happens with a computer, we simply reset it and it will keep working. However, there is no reset button we can push on the microcontroller and thus solve our problem. To overcome this obstacle, we need to introduce one more block called watchdog. This block is in fact another free-run counter where our program needs to write a zero in every time it executes correctly. In case that program gets "stuck", zero will not be written in, and counter alone will reset the microcontroller upon achieving its maximum value. This will result in executing the program again, and correctly this time around. That is an important element of every program to be reliable without man's supervision.

2.2.8 Analog to Digital Converter

As the peripheral signals usually are substantially different from the ones that microcontroller can understand (zero and one), they have to be converted into a pattern which can be comprehended by a microcontroller. This task is performed by a block for analog to digital conversion or by an ADC. This block is responsible for converting an information about some analog value to a binary number and for follow it through to a CPU block so that CPU block can further process it.

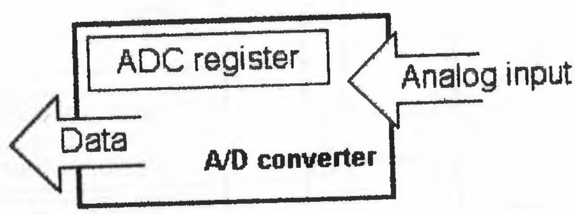


Figure 2.8 block for converting an analogue to a digital form.

Finally, the microcontroller is now completed, and all we need to do now is to assemble it into an electronic component where it will access inner blocks through the outside pins. The picture below shows what a microcontroller looks like inside.

Thin lines which lead from the center towards the sides of the microcontroller represent wires connecting inner blocks with the pins on the housing of the microcontroller so called bonding lines. Chart on the following page represents the center section of a microcontroller.

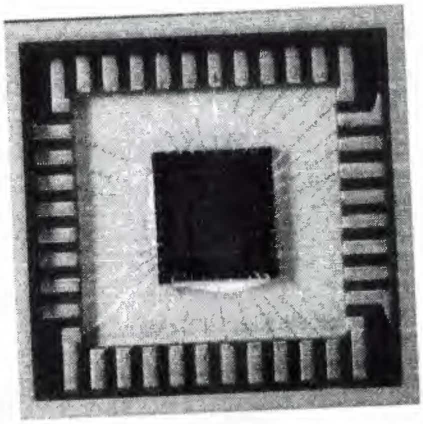


Figure 2.9 Physical configuration of the interior of a microcontroller.

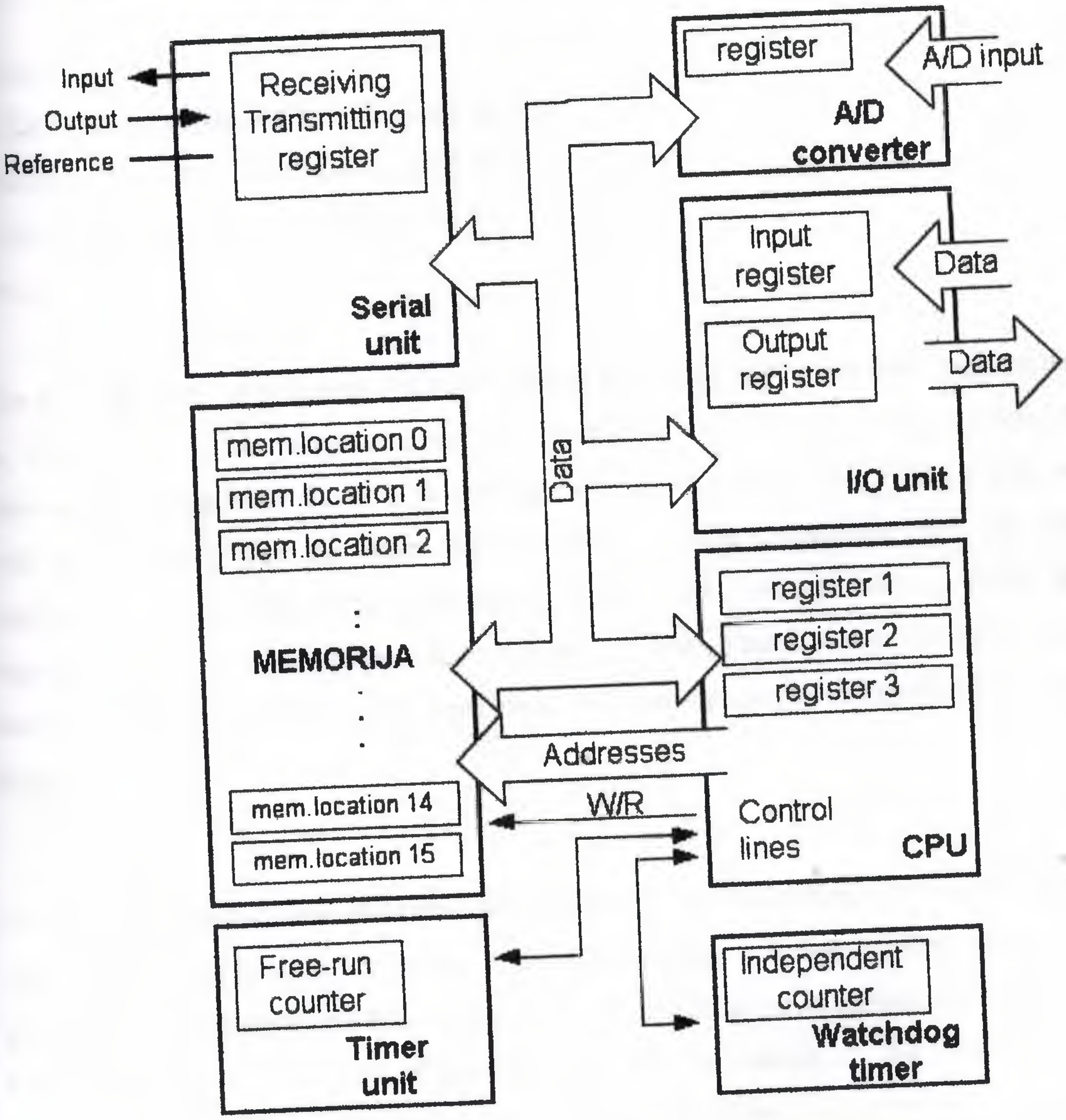


Figure 2.10 Microcontroller outline with its basic elements and internal connections

For a real application, a microcontroller alone is not enough. Beside a microcontroller, we need a program that would be executed, and a few more elements which make up a interface logic towards the elements of regulation.

2.2.9 Program

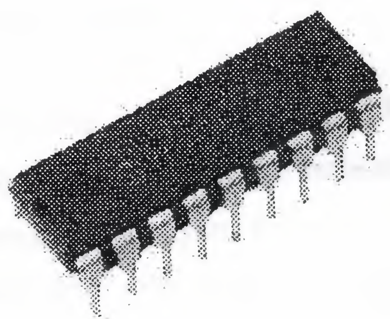
Program writing is a special field of work with microcontrollers and is called "programming". Try to write a small program in a language that we will make up ourselves first and then would be understood by anyone.

```
START
REGISTER1=MEMORY LOCATION_A
REGISTER2=MEMORY LOCATION_B
PORTA=REGISTER1 + REGISTER2
END
```

The program adds the contents of two memory locations, and views their sum on port A. The first line of the program stands for moving the contents of memory location "A" into one of the registers of central processing unit. As we need the other data as well, we will also move it into the other register of the central processing unit. The next instruction instructs the central processing unit to add the contents of those two registers and send a result to port A, so that sum of that addition would be visible to the outside world. For a more complex problem, program that works on its solution will be bigger.

Programming can be done in several languages such as Assembler, C and Basic which are most commonly used languages. Assembler belongs to lower level languages that are programmed slowly, but take up the least amount of space in memory and gives the best results where the speed of program execution is concerned. As it is the most commonly used language in programming microcontrollers it will be discussed in a later chapter. Programs in C language are easier to be written, easier to be understood, but are slower in executing from assembler programs. Basic is the easiest one to learn, and its instructions are nearest a man's way of reasoning, but like C programming language it is also slower than assembler. In any case, before you make up your mind about one of these languages you need to consider carefully the demands for execution speed, for the size of memory and for the amount of time available for its assembly. After the program is written, we would install the microcontroller into a device and run it. In order to do this we need to add a few more external components necessary for its work. First we must give life to a microcontroller by connecting it to a power supply (power needed for operation of all electronic instruments) and oscillator whose role is similar to the role that heart plays in a human body. Based on its clocks microcontroller executes instructions of a program. As it receives supply microcontroller will perform a small check up on itself, look up the beginning of the program and start executing it. How the device will work depends on many parameters, the most important of which is the skillfulness of the developer of hardware, and on programmer's expertise in getting the maximum out of the device with his program[5].

2.3 PIC Microcontroller



PIC(Peripheral Interface Controller) is the IC which was developed to control the peripheral device, dispersing the function of the main CPU.

When comparing to the human being, the brain is the main CPU and the PIC shares the part of which is equivalent to the autonomic nervous.

PIC has the calculation function and the memory like the CPU and is controlled by the software. However, the throughput, the memory capacity aren't big. It depends on the kind of PIC but the maximum operation clock frequency is about 20 MHz and the memory capacity to write the program is about 1K to 4K words. The clock frequency is related with the speed to read the program and to execute the instruction. Only at the clock frequency, the throughput can not be judged. It changes with the architecture in the processing part. As for the same architecture, the one with the higher clock frequency is higher about the throughput.

I used the WORD for the capacity of the program memory. It represents the one instruction as being the 1 word. It often uses the BYTE to show the capacity of the memory. The 1 byte shows the 8 bits. The bit is the atomicity which shows 1 or 0. The instruction of the PIC16F84A is composed of the 14 bits. It is 1 x 1,024 x 14 = 14,336 bits when converting the 1K words to the bit. It is 14,336/(8 x 1,024) = 1.75K bytes when converting this to the byte.

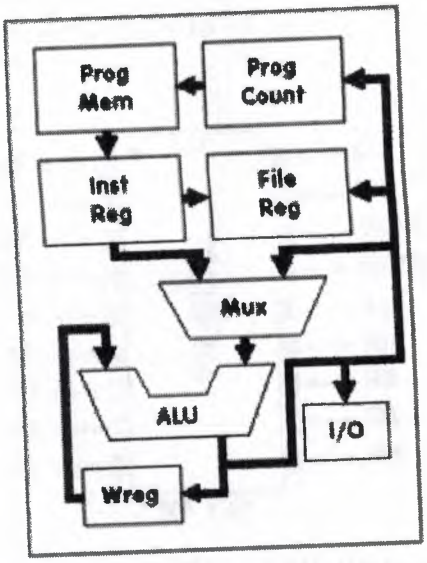


Figure 2.11 PIC microcontroller structure.

At the memory capacity, it is the 1G bytes = 1,024M bytes, 1M bytes = 1,024K bytes, 1K bytes = 1,024 bytes. It is not 1000 times. This is because it calculates in the binary. The point which the PIC is convenient for is that the calculation part, the memory, the input/output part and so on are incorporated into one piece of the IC. The efficiency, the function are limited but can compose the control unit only by the PIC even if it doesn't combine the various ICs. So, the circuit can be compactly made.

2.3.1 Pin Diagram

OSC1/CLKIN : Oscillator crystal input.
External clock source input.

OSC2/CLKOUT : Oscillator crystal output.Connects to crystal or resonator in crystal oscillator mode.

MCLR(inv) : Master clear(reset)input.Programming voltage input. This pin is an active low reset to the device.

RA0 - RA3 : Bi-directional I/O port.

RA4/T0CKI : Bi-directional I/O port. Clock input to the TMR0 timer/counter.

RB0/INT : Bi-directional I/O port. External interrupt pin.

RB1 - RB7 : Bi-directional I/O port.

V_{SS} : Ground

V_{DD} : Positive supply(+2.0V to +5.5V)

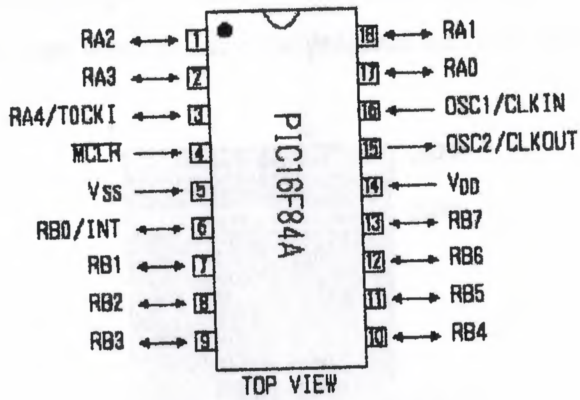


Figure 2.12 PINs of PIC19F84A.

2.3.2 Hardware of PIC16F84A

Flash Program Memory

The flash memory is used for the memory which stores the program. The 1 word is composed of the 14 bits and 1,024 words (the 1K words) are installed. Even if it switches off the power supply, the contents which is stored in the flash memory don't disappear. The contents of the flash memory can be rewritten using the writer. But, the rewritten number of times is limited. It is the about 1000 times.

Reset Vector (0000h)

When the reset is executed by the turning on, WDT(Watchdog Timer) time-out, the other factor, the program starts after the reset from this address.

Peripheral Interrupt Vector (0004h)

When there is the time-out interruption of the timer(TMR0), the interruption from outside and so on, the program starts from this address.

Configuration word (2007h)

The basis operation of the PIC is specified by this memory. The enable bit of the Power-up timer, the enable bit of the Watchdog timer, the Oscillator Selection bits can be set. This area is behind usual program area and can not set by the program. It is necessary to be written using the writer when writing the program into the flash memory.

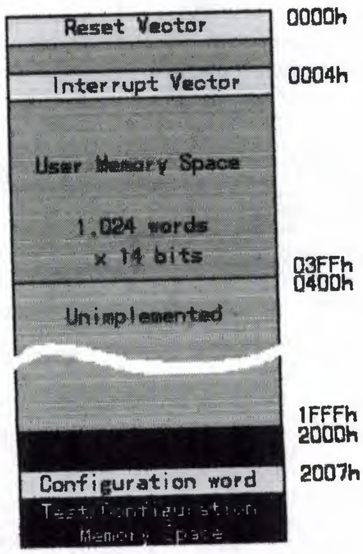


Figure 2.13 flash program memory.

RAM(Random Access Memory) File Registers

The management structure, the bank, is adopted to this memory. The memory capacity is the 80 bytes(00h-4Fh) per bank. In case of the PIC16F84A, there are two banks. This memory is used, dividing into the two areas. The first 12 bytes(00h-0Bh) of each bank are called SFR(Special Function Registers) and are used to record the operating states of the PIC, the conditions of the input/output ports, the other conditions. Each use is decided.

NEAR EAST UNIVERSITY

FACULTY OF ENGINEERING

Department of Electrical and Electronic Engineering

REMOTE CONTROL OVER TELEPHONE LINE

Graduation Project
EE- 400

Student: Omar Daban (20001101)

Supervisor: Assist. Prof. Dr. Kadri Bürüncük

Nicosia - 2004

ACKNOWLEDGMENTS

<u>IN THE NAME OF ALLAH, MOST GRACIOUS, MOST MERCIFUL.</u>

Firstly I wish to express my deep appreciation to my God how stood beside me all the time and how supported me in all my achievements and who has given me the power and patience to finish my college studies successfully.

Secondly I offer special thanks to my parents, who encouraged me in every field of life and try to help whenever I needed. They enhanced my confidence in myself and they trust in my behavior. I am also grateful to my father who supported me and my mother whose prayers to accomplish every stage in my study journey and reach this point of education.

I would like to present my special appreciation to my supervisor Assist. Prof. Dr. Kadri Bürüncük for his help, advises, comments and effort in preparation for this project. His trust in my work and me. His friendly behavior with me and his words of encouragement kept me doing my project.

Last but not least I would thank Mohannad Yasin, Mohammad Al-jahraw and Yaza Al-kilani for their contribution to achieve this project. Also I will never forget my wonderful times that I spent in Cyprus and Near East University with my good friends who helped me incorporeally during my studying in college.

This project is dedicated to Palestine.

ABSTRACT

The remote control over telephone line project designed to meet the need of remotely controlling the electronic devices by following a few simple procedures, in addition the project circuit has to be compact, low cost to construct and requires ordinary equipments to operate, however the remote control over telephone line circuit requires a telephone at the commander position and another one at the destination position, with the telephone line at the destination connected to one terminal of the project circuit and to the other terminals are the desired electronic devices required to be controlled.

This is achieved by building the project using both hardware and software, it is possible to build the same project using electronic components only, but a large number of components would be needed, so more power supply consumption, additional cost expenses and increased possibility for faults, but the software achieves this task with the help of a limited number of electronic components in order to lower the power consumption, decrease the project budget and the circuit to be immune for possible errors that may occur during the control process.

Because of the need to use the telephone to pass the commands in a hand, and electronic components with a software to build the project at the other hand, the basic knowledge about telecommunications, microcontrollers and electronic design are very important in order to plan and build this project.

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INTRODUCTION

The remote controlling occupies an important position in the control systems design because of the demands to control the electronic devices from a remote positions. The remote control device can be used for many applications, such as TV & VCR remote control that uses IR (infrared light) for a few meters distance control, also there are remote controls for a short and long distance using RF (radio frequency), and some other remote controls using the networks to pass the control commands[7].

The communication devices can communicate the data and also the control commands, whether wired or wireless communication systems can be used for control purpose. Telecommunication systems and techniques can contribute to remote controlling because it is communicating local and remote parties, in remote control over telephone line project the PSTN (public switching telephone network) is used to handle the commands from a remote or local position by using the DTMF (dual tone multi frequency) signaling principle, that is used to encode and decode the pressed buttons of the telephone keypad[3].

The processing of the DTMF signals needs a qualified system to handle, therefore the microcontrollers used in this project to achieve this job with the help of a DTMF decoder circuit and some other interfacing circuits. Microcontrollers need to be programmed at the low level language programming that the CPU of the microcontrollers understand.

Such a project has many applications depending on the user demands, for example controlling the home appliances such as heaters, air conditions and gates, and may be used in a factory or a big company that need to control some devices from many positions in the site[7].

CHAPTER ONE

TELEPHONE NETWORK

1.1 Introduction

The Telephone System has been developed over many years and has gone through many incremental evolutionary steps. The traditional service of the Telephone Network has been for Voice Communication and only recently the network has been used to support the high amount of data transfer we are currently experiencing. For cost effective voice communications it has been identified that the Humans can communicate at frequencies between 300Hz to 3500Hz. Though we can hear and speak at higher and lower frequencies, voice communications between 300Hz and 3500Hz are clear and efficient for the telephone network to transmit and receive, look at figure 1.1. A Voice Channel goes from 0Hz to 4000Hz and was developed to avoid any overlapping to any other adjacent voice channels.

Figure 1.1 voice channel bandwidth.

Through the Telephone Network development it has been discovered that it is more efficient to transmit a voice channel in a digital form. In a digital form the voice channel can be routed to its destination with very low noise, higher reliability and more cost effectiveness. To do this the Voice Channel is converted to Digital by an Analog to Digital Converter (A/D Converter or ADC) at the Central Office (see figure 1.2). The Central Office (CO) is the location in your area that the 2-Wires that come from your

house called Tip & Ring are terminated. Once the Voice Channel has been digitized it is transmitted over the network to the CO of the number you called. At this remote CO the Voice Channel is reconstructed back into an analog form so that remote person can understand it. This transformation back to analog is done by a Digital to Analog Converter also know as a D/A Converter or DAC.

One important point to know is that the A/D Converter samples the Voice Channel at twice the frequency of the voice channel; that is at 8Khz. The reason the sampling is done at twice the original signals frequency is due to a law called the Nyquest Rate, which states to digitize a waveform and have enough information to reconvert it back to the an analog waveform one must sample the original waveform at no less than twice the frequency of the original waveform. Also each sample is identified by 8 unique bits that represent 256 different states. 8 Bits sampled 8000 times a second is equal to 64Kbps. Figure 1.2 shows the traditional communication process[4].

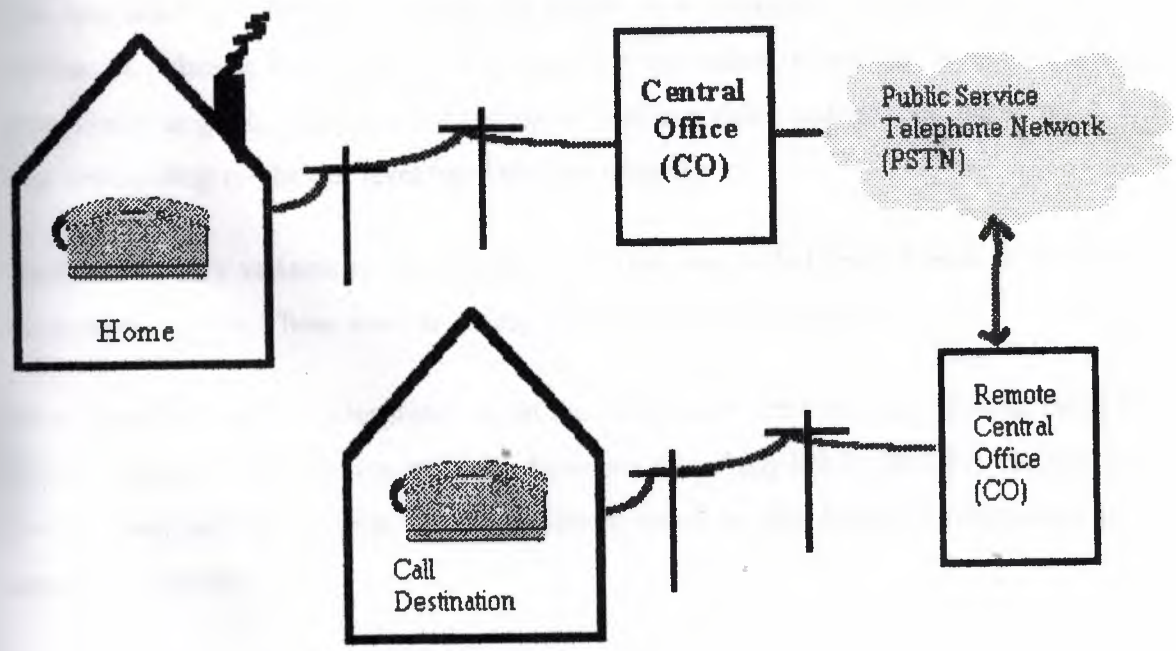


Figure 1.2 communication process.

1.2 Local Central Office Switching Systems

A brief overview of the history and operation of local central office telephone switching systems.

1.2.1 Step by Step (Strowger or SxS)

The Step by Step switching system was invented in the early 1890's by Almon Strowger, an undertaker in Kansas City, MO USA. Rumor has it that the local switchboard operator was diverting calls from his business to others. He wanted to have some way to have calls automatically routed to the destination without the intervention of an operator.

His invention had changed throughout the years, but the basic concept has remained the same. The system operates by using rotating "selectors" that can select a "level" (or step) by the number of pulses that are interrupted on a line (using a "rotary" telephone). This instructed a solenoid to move this wiper to a particular position based on the number of pulses it had received. This then put the caller on another level and again pulse dialed to get to another level and so on until the caller had reached the subscriber they were calling on the last level (or digit they dialed).

There were many variants to the step by step. One was called the XY switch made by Stromberg-Carlson. These were in widespread use in many rural areas.

These systems were in widespread use in the North American system well into the late 1980's (and some into the late 1990's!). There are few if any left in the USA or Canada, but they may still be in use in some third-world countries and former Soviet controlled countries in Europe.

1.2.2 Panel Switching Systems

The "Bell System" (Western Electric) never liked the idea of allowing customers to dial their own calls since they specialized in "customer service" and had operators complete all calls. By the late 1910's, they started to realize that this may not be a good idea and

started to use step by step systems (as described above). However, it did not work out well in large cities where there a large number of calls in progress at an one time. They developed a new system called a Panel switch. Their first switch was called a Panel switch because of the large vertical panels. It followed the same basic concept of the Step by Step but used a method of where the selector system operated on a concept of "ladders" where the selector would rise up on a panel by the number of pulses it had received.

The first one was installed in the early 1920's in Omaha, Nebraska and others were installed throughout the 20's and 30's in most metropolitan areas in the USA (except for Los Angeles, which was step by step).

Panel systems had a system of "revertive pulsing" where it communicated with another Panel switch by the receiving switch pulsing BACK to the originating switch! Instead of sending dial pulses out TO the terminating switch, it received them FROM the terminating switch.

Most Panel offices were removed from service by the late 1970's, though I hear the last one was located in northern New Jersey and was removed in the early 1980's.

1.2.3 Crossbar Systems

Crossbar systems were developed in the late 1930's (with technology from LM Ericsson) by Western Electric (the Bell System). The Crossbar had a significant advantage over either Panel or Step by Step since it was able to use a "store and forward" concept where it would take incoming digits, store them, and then process the call. It was also able to do call routing and determine where a call should be sent to by doing "translations" of the incoming digits and deciding how to send the call.

The concept of the crossbar is the crossbar mechanism operates on a matrix concept where dialed digits would change the position of the crossbar and connect to other parts of the matrix. It was still the same "level" concept of the step by step and panel but using more efficient and smaller equipment.

The first crossbar system was the Number 1 Crossbar (or #1XB for short) developed in 1938. These were in widespread use in large metropolitan areas (like Panel) where step by step equipment would not work as well.

In the late 1940's, Western Electric developed the Number 5 crossbar (or #5XB) where it was the same concept as the #1XB but an improved design. The #5XB was put into widespread use in the US by the late 1950s and 1960s. The last #5XB was installed in 1969.

The #5XB was also significant since it was later modified to handle customer dialed Dual Tone Multi-Frequency dialing in the early 1960's. This is of course something we all know as "Touch Tone".

Other crossbar systems were developed by independent telephone company manufacturers. One popular one called the NX-1 crossbar (and its smaller counterpart, the NX-2) developed by North Electric (not to be confused with Northern Electric). It was used by companies such as United Telephone and others.

Northern Electric made their own crossbar system based on the #5 crossbar concept of Western Electric.

Most of not all crossbar systems have been removed by the late 1980's and definately by the late 1990's in the North American system. There may still be some crossbar in countries outside the US and Canada.

1.2.4 Early Computer Controlled Analog Switching Systems

The first "electronic" switch that was used in the public telephone network was an experimental switch in Morris, IL in 1960. See the Telephone History pages for more details.

In the mid 1960s, Western Electric developed a computer controlled analog switch called the Number 1 Electronic Switching System or #1ESS for short. It was basically a computer controlled Number 5 Crossbar (#5XB) but using "reed relays" instead of

physical crossbars. This considerably reduced the size of the switch, improved its reliability, and made it easier to make "translation" modifications (how switches route calls) by changing software, not hardware.

The switch was also unique because of new inventions called Custom Calling Features - Call Waiting, Three-Way Calling and Speed Dialing.

Independent switch manufacturers also made computer controlled analog telephone switching systems. Automatic Electric made the #1EAX and #2EAX (Electronic Automatic eXchange) in the early and late 1970's respectively. Other companies such as Stromberg-Carlson made the ESC switch in the early 1970s.

In 1976, Western Electric made advancemnets in the #1ESS technology and produced the #1AESS system. Older #1ESS systems were modified to #1AESS and new ones made after 1976 had the new technology.

Western Electric also made other computer controlled switching systems that were an improvement on the #1ESS/#1AESS. These switches were used in areas with a small amount of customers. These switches included the #2ESS for suburban use (1970), the #2BESS (1976) and the #3ESS for rural use (1976) switches.

There are no more #2ESS/#2BESS or #3ESS switches in service, nor any old AE or Stromberg-Carlson early electronic switches in service. However, there are still a number of #1AESS switches still in use, though many are scheduled to be replaced by sometime in the early 2000s.

1.2.5 Fully Digital Switching Systems

Though the #1ESS/#1AESS switch was computer controlled, it still was an analog switch. Technology had advanced enough by the late 1970s where 100% digital systems were being developed for use in the telephone industry. Digital systems "sample" the analog signal and handle telephone calls internally as binary digits, then convert them back to analog to be compatible with regular telephones. This again made the switch smaller and more reliable. It also allowed new technologies to be added faster by using

modular techniques (adding systems to the original system without complete redesign/reinstallation) and by using advanced computer software.

The first totally digital system (for end offices) was not developed by Westen Electric but by a company called Vidar. Their first switch was developed in 1978. Northern Telecom (formerly Northern Electric, now Nortel) developed the DMS series of digital switches. In 1979, the DMS-10 was first produced. Later they started producing the DMS-100 as a local end office switch.

Western Electric developed their fully digital switch in 1982. It was called the Number 5 Electronic Switching System or #5ESS for short. It is a fully digital switch that did everything the #1AESS did and more. Though it is primarily used as a local central office, the #5ESS can be used as an operator services switch or as a low to medium traffic volume tandem. In 1984, Western Electric was absorbed into AT&T as part of divestiture. In 1996, the hardware group of AT&T was spun off into its own company - Lucent Technologies. The #5ESS switch is still made to this day by Lucent.

Other independent companies developed their own digital switches. Automatic Electric developed the #5EAX (or better known as the GTD-5) switch. Others such as Stromberg Carlson (now a unit of Siemens) developed the DCO (Digital Central Office), and Siemens made the EWSD (Electronic Worldwide Switch Digital) switch

The GTD-5 switch is no longer being made though it is still in widespread use. The #5ESS, the DMS series, DCO and the EWSD switches are still in production are in widespread use in the US, Canada - as well as many countries worldwide.

1.2.6 Packet Switching Systems

The Internet and the technologies that are used with the Internet changed the idea of how communications are handled. Traditional telephone company technologies were direct poiont-to-point systems where dedicated "trunks" (toll circuit routes) were used. Once a trunk was allocated, the trunk stayed up for the entire connection. If a trunk was not being used, it sat idle - underutilizing limited resources.

The Internet (TCP/IP and Ethernet in particular) uses a completely different pardigm (methodology). In the TCP/IP world, all systems share a common connection. Each system (usually a computer) was connected to this common system and shared "bandwidth" with other people. Resources were not allocated for just one user, but all users can share resources.

Another advantage of using TCP/IP technology is the switching "matrix" is also different and more efficient. Instead of using a dedicated switching matrix - with a limited amount of circuits in a traditional telephone switch, TCP/IP routers (traffic handlers) could be used. This increases the amount of switching "lines" and can do it more efficiently than traditional switching systems since all the switching is done on a "virtual" scale in software rather than in hardware.

Telephone systems have been slowly adopting the "IP" (Internet Protocol) technology over the last 8 to 10 years. Early experiments proved that voice can be converted to digital "packets" and sent over the Internet. The packets would be collected and converted back to analog voice. The quality of the calls was not great but it showed that it could be done. The major problem was somethig called "packet loss" which is common with TCP/IP connections.

By the early 2000s, the IP telephony (or "VoIP" - Voice over Internet Protocol) technology had improved. Using "classes" of service, reliable connections could be obtained and packet loss reduced to minimum levels. Business systems started using VoIP technology in their PBX (Private Branch Exchange) switches. The telephones themselves were almost like small computers that had their own analog/digital conversion systems and TCP/IP networking technology all the the same system. The phone could "piggy-back" on their existing computer network system. Hence having voice AND data traffic over the same wires!

Telephone companies - both local and toll - are also handling data traffic at alarming rates. AT&T is handling 5 to 10 times more data traffic than it handles for traditional voice traffic. To handle both efficiently - it would be cost effictive to have switching systems that handle data switching on a high volume scale - that can serve both the voice market AND the data market at the same time.

Lucent Technologies and Nortel Networks are both making switching systems that are known as "packet switches" that take the traditional switching system to a higher level. These systems can handle both data AND voice at the same time. These are add-ons to existing voice switches and are located at traditional central office locations.

Lucent's packet switching system is known as the 5E-XC. It builds upon the #5ESS switch and handles both data and voice.

Nortel's packet switching technology is known as the Succession family of switches. The Succession Server 2000 is now being used as a replacement or in addition to the traditional DMS series of digital switching systems. The local arm of Sprint is now installing Succession 2000 switches in various places nationwide as of the fall of 2003. This includes cities such as Las Vegas, NV.

1.3 Telephone Signalling System Technologies

A brief overview of the history and operation of telephone signalling technologies.

1.3.1 Pulses

Pulses (interrupting current on toll trunk lines), especially on inter-office trunk lines, were common in the days of step by step and crossbar systems. These were used to pulse out dialed digits, either directly or via pulse senders from a central office switch, to the destination switch. Though not in common use today, these may still do exist in rare instances outside of North America.

1.3.2 DC Voltage and DC Polarity

Voltage changes and polarity changes (as well as pulses) were used in some inter-office signalling routines. These were sometimes found in Panel systems when communicating to Panel "tandem" systems. Never was in common use and are not used anymore[1].

1.3.3 Multi-Frequency Signalling (DTMF)

Dual-tone multi-frequency (DTMF), also known as Touch Tone® is used for telephone signaling over the line in the voice frequency band to the call switching center. DTMF is an example of a multi-frequency shift keying (MFSK) system. Today DTMF is used for most call setup to the telephone exchange, at least in the Western world, and trunk signalling is now done out of band using the SS7 signaling system. The trunk signalling tones were different than the tones known as touch tone with a triangular matrix being used rather than a square matrix. See: blue box for more details on the switching tones.

Prior to DTMF the phone systems had used a series of clicks on the phone line to dial numbers, a system known as pulse dialing. The clicks were actually the connection of the calling party's phone line being made and broken, like flicking a light switch on and off. This was useful only as far as the local end office where the wires stopped, requiring operator intervention for long distance dialing.

DTMF was developed at Bell Labs in order to allow dialing signals to dial long-distance numbers, potentially over non-wire links such as microwave links or satellites. Encoder/decoders were added at the end offices that would convert the standard pulse dialing clicks into DTMF tones and play them down the line to the remote end office. At the remote site another encoder/decoder would decode the tones and turn out a series of clicks. It was as if you were connected directly to that end office, yet the signaling would work over any sort of link. This idea of using the existing network for signaling as well as the message is known as in-band signaling.

It was clear even in the late 1950s when DTMF was being developed that the future of switching lay in electronic switches, as opposed to the mechanical crossbar systems currently in use. In this case pulse dialing made no sense at any point in the circuit, and plans were made to roll DTMF out to end users as soon as possible. Various tests of the system occurred throughout the 1960s where DTMF became known as Touch Tone.

The Touch Tone system also introduced a standardized keyboard layout. After testing 18 different layouts, they eventually chose the one familiar to us today, with 1 in the upper-left and 0 at the bottom. The adding-machine layout, with 1 in the lower-left was also tried, but at that time few people used adding machines, and having the 1 at the

"start" (in European language reading order) led to fewer typing errors. In retrospect, many people consider that this was a mistake. With the widespread introduction of computers and bank machines, the phone keyboard has become "oddball", causing mistakes.

The engineers had also envisioned phones being used to access computers, and surveyed a number of companies to see what they would need for this role. This led to the addition of the pound (#) and star (*) keys, as well as a group of keys for menu selection, A, B, C and D. In the end the lettered keys were dropped from most phones, and it was many years before the # and * keys became widely used, primarily for certain vertical service codes such as *67 to suppress caller ID. Many non-telephone applications still use the alphabet keys, such as Amateur Radio repeater signaling and control.

The US military also used the letters, relabled, in their Autovon phone system. Here they were used before dialing the phone in order to give some calls priority, cutting in over existing calls if need be. The idea was to allow important traffic to get through every time. Pressing C, Immediate, before dialing would make the switch first look for any free lines, and if all lines were in use, it would hang up any non-priority calls, and then any Priority calls. While the Autovon phone system no longer exists, their original names were Flash Override (A), Flash (B), Immediate (C), and Priority (D). Pressing one of these keys gave your call priority, over-riding other conversations on the network. Flash Override is the highest priority.

Present-day uses of the A, B, C and D keys on telephone networks are few, and exclusive to network control. For example, the A key is used on some networks to cycle through different carriers at will (thereby listening in on calls). Their use is probably prohibited by most carriers.

The DTMF keypad is laid out in a 4 x 4 matrix with each row representing a low frequency, and each column representing a high frequency. Pressing a single key such as '1' will send a sinusoidal tone of the two frequencies 697 and 1209 hertz (Hz). The two tones are the reason for calling it multi-frequency. These tones are then decoded by the switching center in order to determine which key was pressed.

The frequencies were initially designed with a ratio of 21/19, which is slightly less than a whole tone, to avoid harmonics or natural occurring frequencies that could occur when the two tones are sent. The frequencies may not vary more that +/- 1.5% from their nominal frequency, or the switching center will ignore the signal. The high frequencies may be the same volume or louder as the low frequencies when sent across the line. The loudness difference between the high and low frequencies can be as large as 3 decibels (dB) and is referred to as twist.

Table 1.1 DTMF Keypad Frequencies.

1	2	3	A	**697 Hz**
4	5	6	B	**770 Hz**
7	8	9	C	**852 Hz**
*	0	#	D	**941 Hz**
1209 Hz	**1336 Hz**	**1477 Hz**	**1633 Hz**	

DTMF can be regarded as a simple form of orthogonal frequency division multiplexing. Synonyms include multifrequency pulsing and multifrequency signaling[3].

1.3.4 Common Channel Inter-Office Signaling / Signaling System 7

CCIS (Common Channel Inter-Office Signalling) started in the late 1970s as a way to send more information between tandems and switches in the network. Sending a call via MF takes up to 15 seconds on a domestic phone call. Also, MF lead itself to toll fraud since making MF tones is fairly easy to do.

CCIS was invented to send information "out of band" on a data circuit parallel to the voice circuit. This way the call can be set up and completed in a shorter amount of time, be able to send more information, and avoid toll fraud - all at the same time.

The early commonly deployed versin of CCIS was version 6. The modern version of CCIS is version 7 - commonly called SS7 (Signalling System 7) where there are 7

"layers" of the signalling network. SS7 is in widespread use in the USA, Canada and many industrialized countries in the world including the UK, Austrailia and others.

1.4 Telephone Transmission Technologies

A brief overview of the history and operation of telephone transmission technologies.

1.4.1 Open Wire Carrier

Open wire carrier was developed to carry multiple calls over a pair of copper wires simultaneously. It uses a method of frequency division multiplexing (FDM) where calls are sent on different frequencies. The frequencies are around 100Khz at about 4 Khz per channel.

Open wire carrier has long ago been removed in all areas. The last open wire carrier system I am aware of was located in rural New Mexico and removed in August of 1997.

1.4.2 Coax Cable

Coax cable was used for cross country communications. Coax cable was also used in the transmission of radio and television programming. It was also used for use by government and defense department purposes. Many repeaters were involved and many distribution facilities as well.

1.4.3 Microwave (Radio) Towers

Developed in the 1940s and 1950s, microwave transmission became a widespread telephone call transmission medium. Many microwave towers were erected in many countries worldwide. Microwave again uses frequency division multiplexing.

The microwave system is called as such because of the short wavelengths of the frequencies involved - micrometers! In terms of frequency, the range is in the gigahertz.

There are a few microwave systems in use today, but those that are still in use are converted to digital transmission techniques. Many of the old towers have either been decomissioned or are now used as cellular/PCS towers.

(Off-site Link - "The Latest Word in Communications") In 1947, AT&T inaugurated an experimental microwave radio link, connecting Boston and New York City. This brochure was published by the Long Lines Department to describe the system's technology and facilities.

1.4.4 Satellites

Most people think of satellites for television transmission. But Bell Labs/AT&T invented communications satellites for long distance communications. The first satellite was Telstar in 1962. The major drawback was the 1/2 second delay because of the distance from the Earth to the geostationary orbit over 22,000 miles from Earth and return. Satellites are still used today for far remote places.

1.4.5 Fiber Optics

Ultra pure glass optical fibers using amplitude modulated infrared light, commonly known as Fiber Optics, revolutionized telecommunications transmission techniques. Developed in the 1970s and implemented in the 1980s and beyond - fiber optics is the high bandwidth and high quality transmissions medium that is in widespread use in the telephone industry today. Almost all telephone companies in North America use fiber optics, as do many telephone companies in the industrialized world.

1.5 Telephone Tandem Switching Systems

Telephone tandems are used to "bridge" many central offices together, or used to route telephone traffic within a long distance network. Some tandem switches are dedicated pieces of equipment, while other switches are multi-function (toll and local).

1.5.1 Step by Step Tandem

A step by step switch can be used as a tandem - either as a dedicated switch or as part of a switch that also does local central office switching functions. For example, Carolina Telephone (now Sprint local) in North Carolina used specially dedicated step by step switches as a tandem to several step by step local end offices.

Step tandems were common in rural areas. Step tandems were removed when most step switches were removed in the 1980s.

1.5.2 Crossbar Tandems

A crossbar tandem was based on crossbar switch technology. Western Electric developed the "Crossbar Tandem" or XBT in 1941 for use in urban areas to handle primarily local toll traffic. Later, WECO developed the Number 4 Crossbar Toll Switch (commonly referred to as the #4XB) in 1943 for use as a full service short and long haul toll switch. The first #4XB was installed in Philadelphia.

The first series of #4XB tandems had routing "translations" hard coded in the system. In the early 1950s, new #4XB toll switch had "card translator" boxes installed to do routings that could be changed with new cards. These new tandems were called "4A" toll switches, the A for Advanced. Older #4XB toll switches were modified, calling them 4M for Modified. In 1969, new #4XB toll switch had electronic translator systems or ETS for short. Some older ones were modified to use ETS while others used card translators until they were removed from service.

The last #4XB toll switch was installed in 1976. #4XB toll switches were in service well into the late 1980s. There were over 200 4XB toll switches made between the 1940s and the 1970s. A historical chronological list of installation dates and locations can be found in our #4XB Toll Switch List.

Other crossbar switches were used as tandems, but many were also used as regular central office switches. #5XB switches from Western Electric were sometimes used as

tandems.

1.5.3 Electronic Switching System Tandems

AT&T #4ESS

In 1976, Western Electric developed the Number 4 ESS (#4ESS) (Electronic Switching System) toll tandem switch to replace the #4XB as the toll tandem used in the toll network. The first #4ESS was installed in Chicago. In later years, the #4ESS was manufactured by AT&T and later Lucent Technologies.

Primarily used in the long haul toll network, some #4ESS switches are used in regional toll traffic as well. AT&T still uses the #4ESS with over 140 #4ESS switches used in the US and Canada. Some local telephone companies use the #4ESS as a regional tandem. Similar in design to the #5ESS switch (which can be used as a medium traffic volume tandem) is a fully digital switch. The #4ESS switch can do local end office functions as well and is used as such in some areas where AT&T is operating as a CLEC (Competative Local Exchange Carrier).

The last #4ESS switch was installed in June 1999 in suburban Atlanta, Georgia. Lucent has determined that the #4ESS, while good at switching voice circuits, is not very good at switching data circuits. Emphasis will now go to the Lucent #5ESS and switches from other vendors for future tandem switching. However, the existing #4ESS switches that are in use will continue to do so for a number of years to come.

Nortel DMS tandem series

Other switch manufacturers also make digital electronic style tandem switches. The most popular are the Nortel (Northern Telecom) DMS series of switches.

DMS-200 - Low-volume/regional tandem (can be combined with TOPS (Toll Operation Position Station) for operator services). Usually as part of a DMS-100 local switch.

DMS-250 - High-volume/long-haul tandem

DMS-300 - International gateway

DMS-500 - A combination of DMS-100 (local switch) with a DMS-200 (medium volume/regional tandem) and DMS-250 (high-volume/long-haul tandem).

1.6 Telephone Elements

In 1876, Alexander Graham Bell applied for a patent for the telephone. The first, simple application consisted of two battery-powered devices placed in separate rooms and connected by one direct line. By turning a crank to generate a current in one of the devices, the user caused a signal to buzz in the other device. One day, Bell's assistant heard not only that signal but also the first words spoken over a telephone:

Today, the telephone is powered by the local exchange. The schematic diagram in Figure 1.3 illustrates the principle of the standard version of the telephone. Somewhat simplified, it can be said to consist of four units:

- the bell and a series capacitor;
- the hook switch;
- the keypad (or dial); and
- the speech circuit with the receiver and microphone.

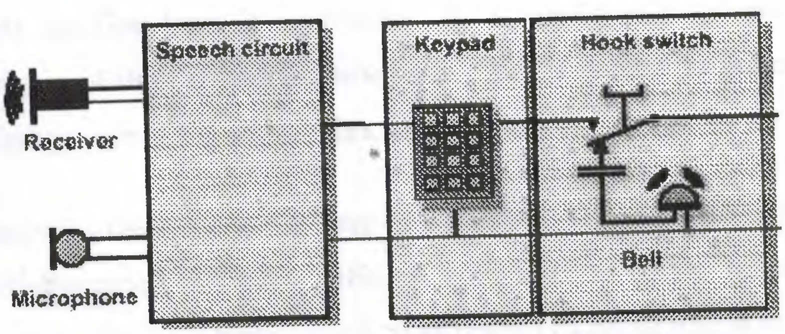


Figure 1.3 Schematic diagram of a keypad telephone.

1.6.1 The bell

The bell is connected via the capacitor when the receiver is resting in its cradle (on hook). When a call is placed to the B-subscriber, the bell is energised via the capacitor by an alternating voltage (approximately 90 V, 25 Hz), producing a ringing signal that notifies the subscriber of the incoming call.

1.6.2 The hook switch

When the A-subscriber lifts the receiver to place a call, the speech circuit and keypad are connected (and the bell is disconnected) via the hook function. This alerts the local exchange that a number is about to be dialled: the B-subscriber number. When the B-subscriber lifts the receiver to answer, the hook switch disconnects the bell in his telephone and instead connects the speech circuit and keypad. Since this closes the subscriber line, current from the local exchange can be fed to the line - an indication that the B-subscriber has answered. The parties can commence their conversation.

1.6.3 keypad

The keypad of a modern telephone is connected to a tone generator, an electronic circuit that translates keyed inputs to tone codes. Each of the digits and each of the "star" (*) and "hash" (#) function keys is represented by a combination of two tones. The frequency of the oscillators is selected whenever a key is pressed to generate the dual-tone combination unique to the digit or function in question.

Figure 1.4 illustrates the principle of keypad signalling. The standard is referred to as dual-tone multi-frequency (DTMF). Different combinations of the seven frequencies (the tones) represent the 12 symbols found on an ordinary keypad telephone.

Some modern telephones also have a function key marked with an "R" (register button). Its function (register recall) is to generate a single pulse.

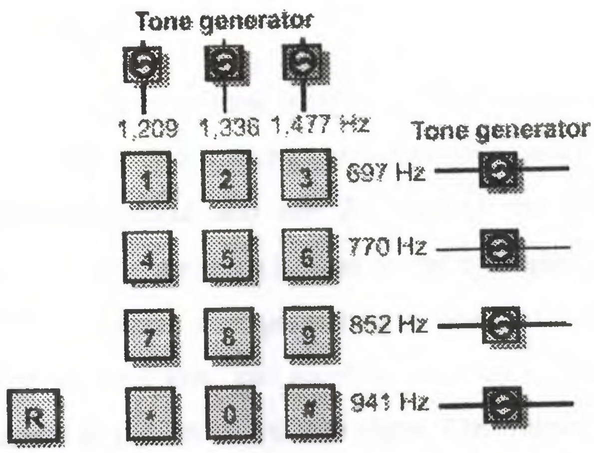


Figure 1.4 Schematic diagram of a keypad and its frequencies.

1.6.4 The Dial

Older telephones have dials instead of keypads. Although still common in many countries, these telephones represent just a few percent of all telephones sold today. The principle of the dialling function is of historical interest, so we will briefly discuss it.

The dial creates a pulse train (signals) containing information to the local exchange. The circuit connecting the exchange and the telephone is closed during the entire digit-sending process, but a contact disconnects the speech circuit during each pulse sequence. (The pulses would otherwise be heard as interference, as "clicks", in the receiver.)

The contact connected to the dial consists of a toothed wheel and two contact tongues. When the dial is released (after being wound up), the wheel starts to rotate, alternately breaking and closing the circuit. Every break results in a pulse, and the number of pulses indicates the digit dialled by the subscriber. Each of the digits forms a pulse train that is detected by the local exchange. Interestingly, Sweden is the only country that has zero as the first digit on the dial. The dials of other countries have zero following the nine.

1.6.5 The Speech Circuit

The primary function of the speech circuit is to adapt the sound level of incoming voice, outgoing voice and sidetone. The circuit comprises two amplifier blocks (one for amplifying the microphone current and one for feeding the receiver) and a bridge connection that separates voice signals to be sent to the microphone and to the receiver. Since the degree of amplification is regulated by a control circuit, transmission and reception distortion can be kept low, and amplification can be maintained constant for subscriber line resistances in the interval 0-900 ohms. Line impedance and the sidetone produced by the caller's voice are adapted by the balance circuit.

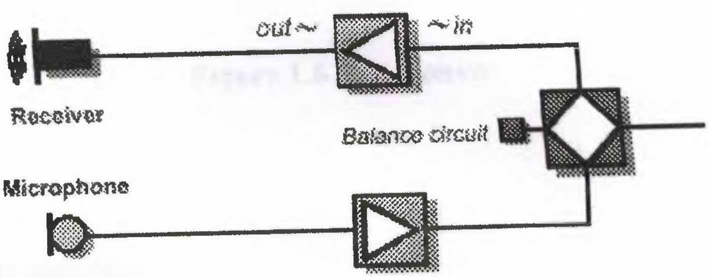


Figure 1.5 Speech circuit.

The speech circuit of older telephones was constructed in a simpler fashion, consisting in principle of only a microphone (usually a carbon-type microphone) and a dynamic receiver. Modern speech circuits provide numerous advantages.

- Sound-level attenuation over long-distance connections is counteracted by line-current-controlled regulation of the speech circuit amplifier.
- Accurate bridge balance and speech circuit impedance enhance sidetone characteristics and optimise the impedance of the apparatus.
- Transmission distortion is negligible.

1.6.6 The receiver

In principle, the design of the receiver is still based on traditional techniques. The current generated by the incoming speech passes through an electromagnet that is constructed around a permanent magnet and connected to a membrane. The oscillations, or movement, of the membrane are converted to sound waves that are perceived by the ear.

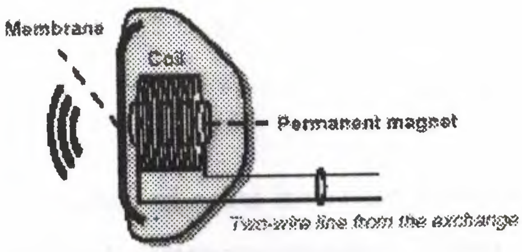


Figure 1.6 The receiver.

1.6.7 The Microphone

The old carbon microphones are being increasingly replaced by electret microphones. The material upon which these new microphones are based consists of a thin plastic film, similar to Teflon, that is exposed to a strong electrical field. The film retains its negative and positive charges after the external electrical field is removed - somewhat analogous to the poles of a magnet.

The principle of operation of the electret microphone is illustrated in Figure 1.7. The Teflon film (electret material) is stretched over a fixed electrode.

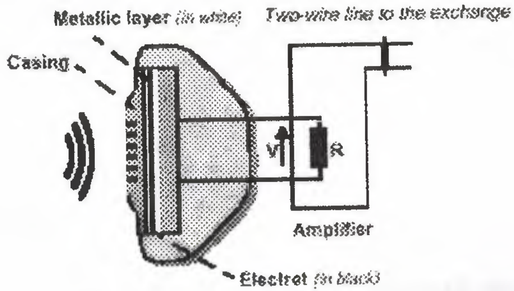


Figure 1.7 Electric microphone.

irregularities in the surface of the fixed electrode cause a number of small air gaps to arise between the electret and the fixed electrode. The electret microphone can therefore be said to consist of a number of small parallel-connected microphones. The electrical field existing in each of the air gaps is generated by the electret's charge. The movements of the membrane change the size of the air gaps and hence their capacitance. These capacitance variations result in voltage variations that appear across the load resistor[2].

1.7 Standards

Many international PSTN recommendations have been developed during the past years, notably by the International Telecommunication Union - Telecommunications Standardization Sector (ITU-T, previously the CCITT). They are not all-inclusive. Traditionally, the PSTN has also been the subject of national standardization efforts in a number of areas:

- signalling, both in an operator's network and to subscribers;
- the billing and pricing of services;
- service offering and service procedures; and
- the physical access interface.

These local standards have led to a multitude of product versions, so vendors of switching equipment must design their systems to allow for extensive parameterization. Some parameters can be set to their default values. Nevertheless, operators and vendors must exchange a good deal of information to adapt the equipment for delivery.

For modern systems and networks, like GSM and N-ISDN, the need for parameterization is significantly less.

1.8 Network hierarchy

In the following subsections we will mention some network elements and functions that are relevant to PSTN connection set-up.

1.8.1 Central and remote subscriber stages

As mentioned earlier, the trend is towards increasingly larger local exchanges. Subscribers are often connected via remote subscriber stages which, from a switching point of view, perform exactly the same functions as those performed by centrally located subscriber stages.

Signals are sent to the control function in the local exchange, even in the case of a remote subscriber stage. However, a function is ordinarily provided for handling internal calls, should there be a break on the links to the main exchange. When these links are down, information about which services the various subscribers have access to is lost and their use cannot be charged for.

The remote subscriber stage is becoming an increasingly common component in access networks.

1.8.2 Alternative routing

In the traditional PSTN exchange hierarchy, traffic has been routed to direct links (high-congestion routes) and if these links have been busy, then the next higher level in the hierarchy (low-congestion routes) has been used. New routing functions are now available thanks to SS7 and the TUP protocol. One example is the possibility of preventing rerouting further on in the network and instead trying an alternative route all the way from the originating exchange. Another example is placing subscribers in different categories; emergency services, for example, could have access to a number of alternative routes or even routes designated for their exclusive use.

1.8.3 Semi-Permanent Connections

A connection which must not be congested and which must have good transmission characteristics can be set up through the group switch using commands. Such connections are referred to as semi-permanent connections and can utilize different paths through the exchange hierarchy.

Semi-permanent connections are used to connect SS7 signalling terminals with their dedicated time slots. These connections run either from one local exchange to another or to an exchange higher up the hierarchy that serves as a signal transfer point (STP). Semi-permanent connections are also used to create an internal network for a company by setting up leased lines between the company's PBXs. Leased lines can also be handled exclusively by the transport network.

The transmission quality of modem data connections can be guaranteed through the use of leased lines. Avoiding the time-consuming connection set-up phase in data transmission is an added advantage of leased lines.

Leased lines not only run from one exchange to the next but can connect many exchanges at different levels in a hierarchy; for example, to link the offices of a single company in several different countries.

1.9 Synchronization

In the early suppressed carrier systems, it was necessary that the frequency at the demodulator should match that of the carrier signal so that the baseband signals could be recovered without exhibiting an excess frequency shift. Tests performed as late as the 1940's showed that any shift of less than 2 Hz was acceptable for J and K carrier systems (both multiplexed 12 channels).[4]

When Pulse Code Modulation (PCM) and Time Division Multiplexing (TDM) in the form of T1 carrier was implemented, a new set of problems emerged. Now the channels of the system were separated by time with a sample for each channel repeating every 125us.

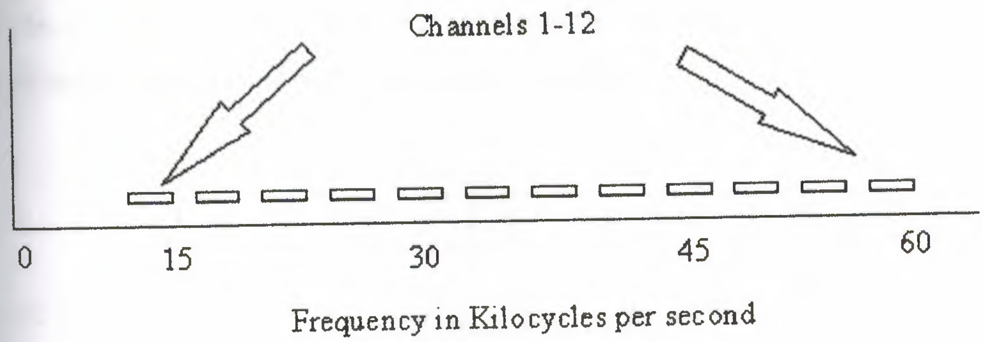


Figure 1.8 Carrier system.

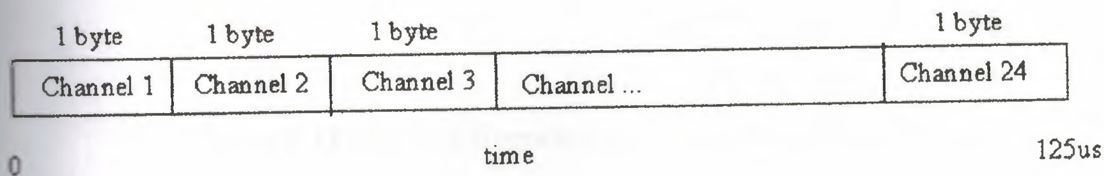


Figure 1.9 Carrier System.

T1 requires a method of phase control so that the receiver can tell the difference between channels for each frame. If the receiver becomes out of phase with the sending signal, data samples could be applied to the wrong channel. Frequency control is also required between the sender and receiver so that the receiver can differentiate each bit for the channels. If the signals are not in proper sequence, data will be lost caused by what is known as **slip.** Figure 1.10 shows an example of a receiver whose frequency is higher than that of the sender. Notice that the receiver checks for samples too often and ends up with the wrong data.

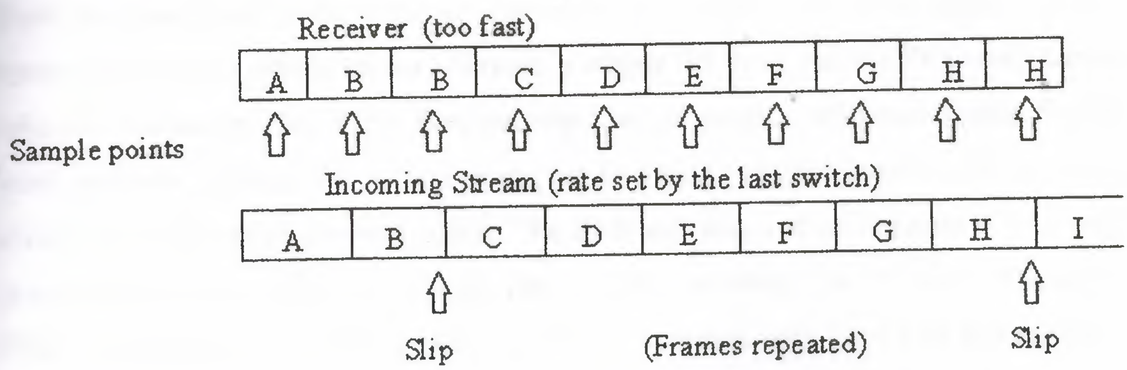


Figure 1.10 Receiver frequency higher than the sender.

When the receiver frequency is slower than that of the incoming stream, the receiver does not sample enough and as a result data is lost[3].

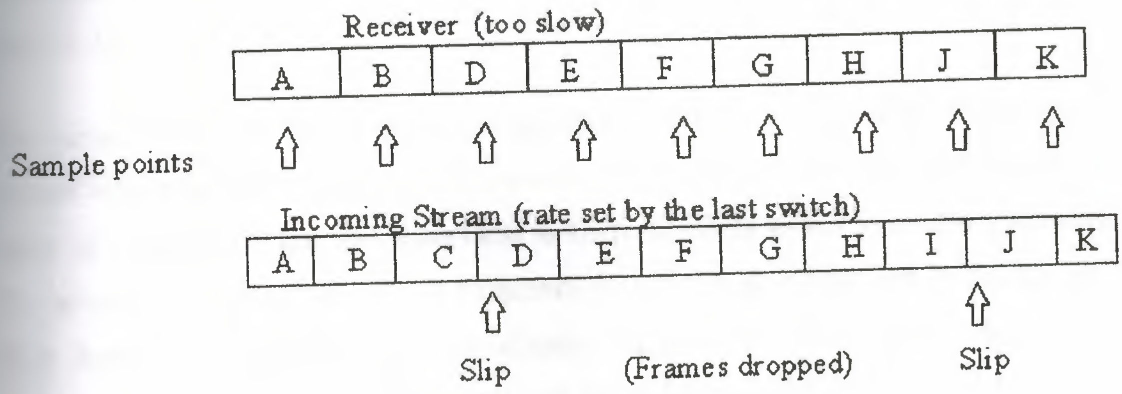


Figure 1.11 Receiver frequency is slower than the sender.

1.9.1 Synchronization Implementation

PFS-1

Before coaxial cable was introduced into the PSTN (pre 1935) modulation frequencies were held to less than 0.5Mhz. Independent tuning fork controlled oscillators were used and able to supply accurate and stable carriers having a frequency accuracy of about 1 in 10^6. [2] However with the introduction of coaxial cable came a corresponding jump in usable frequencies along with a need for higher frequency accuracies. Better accuracy was accomplished by transmitting a separate synchronizing 64 kHz pilot over the line from the originating office terminal. The pilot was merely a 64 kHz signal that was transmitted to each offices primary frequency supply (PFS) so that the PFS could have a reference frequency. The PFS's function was then to provide reference signals for the local multiplex carriers and to regenerate the incoming synchronization pilot so that it could be passed on to the next office. The PFS was adjusted as required if its locally generated frequency did not match that of the incoming 64kHz pilot frequency. PFS-1 was the early implementation of PFS and operated with the L1 (5 groups of 12 channels inside super groups of 10) coaxial system using a 64kHz pilot. This system was capable of maintaining an accuracy of less than 7 parts in 10^7. However as the multiplexing supergroups grew with the introduction of L3 (3 master groups of 600 channels), the PFS-1 system was adapted to use a 308 kHz pilot. New high quality temperature controlled oscillators were required to maintain accuracy to within a few parts in 10^{8}.

The errors stated are based on frequency offset relative to each other rather than a system wide absolute frequency. The system wide frequency accuracy was only on the order of 1 part in 10^6 but was acceptable to keep the pilots within their filter passbands. To maintain absolute frequency, a frequency standard was established in Murray Hill, New Jersey. This standard was periodically adjusted to match that of the national standard at the US Bureau of Standards and Navy. Murray Hill provided a 4 kHz signal to the Long Lines building in New York where the synchronization pilot was generated and transmitted to all other offices. The Long Lines supply was able to maintain an accuracy with Murray Hill on the order of a few parts in 10^9.

PFS-2

The PFS-1 method worked well for a while, but as the system expanded with the growing population, a pilot could be regenerated as many as 20 times. PFS-1 also used mechanical servo motors which moved a variable capacitor to correct pilot frequency differences at the terminal offices. A faster response and more accurate system was necessary. PFS-2 became the successor and utilized a phase locked loop which ensured zero frequency offset between the incoming and regenerated pilot. If the incoming pilot were to be lost the PFS-2 would then run free at the frequency of the local crystal oscillator. PFS-2 was widely installed the 1960's and early 1970's. Soon new problems began to arise as a result of the zero offset. Facility switching or maintenance could temporarily interrupt pilots and introduce transients into the system.

These transients would be propagated through the PFS carrier supplies where modulation would place the transients in the signal paths. The older slower PFS-1 systems had not been quick enough to respond to these pilot changes and were thus unaffected. While these transients were of little effect on speech, they could cause errors with data transmission.

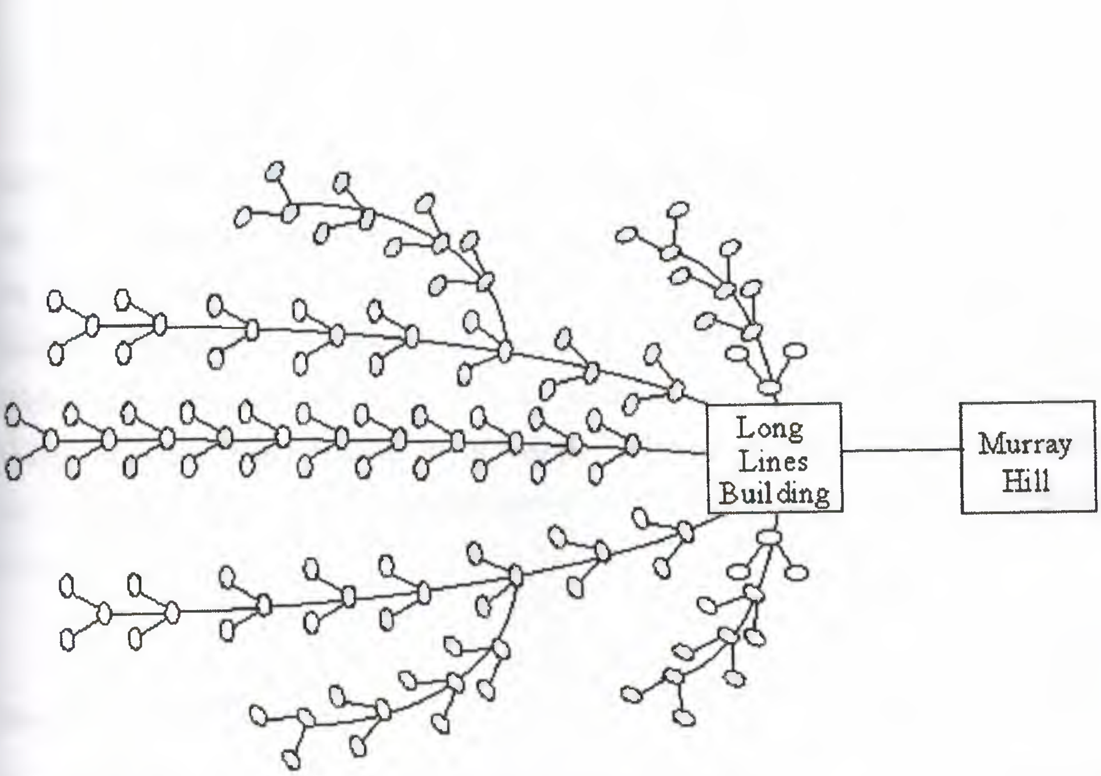


Figure 1.12 Conceptual view of the network branching of the 64kHz pilot from the Long Lines Building.

To compound this, PFS-2 was redesigned to operate with the new L4 system which had a top frequency of 17.5Mhz. The error between two L4 PFS's had to be less than one part in 10^7 and the PFS-2 could not maintain this if the synchronization pilot was lost and the PFS was free running. Also the number of consecutive PFS's continued to grow. Finally with the new L5 systems the frequency accuracy requirements could no longer be met by the PFS's and it was apparent a new system was needed.

JFS

The jumbo frequency supply (JFS) was the approach taken for synchronization for L5 systems around 1974. It became a reference supply for different regions in the country. It consisted of three crystal oscillators which when free running only had a drift of 1 part in 10^{10} per day. This allowed it to run for several weeks without adjustment. The oscillator handled transients from the incoming reference signal by quasi-frequency lock. Quasi-frequency lock (also refered to as plesiosynchronous) means that the compared frequency signals are maintained nearly synchronous. Cycles that were different would be counted with no correction made. When the count reached 256 a correction of only 2 parts in 10^{10} was made in the proper direction. When very large differences between the local and incoming signal occurred, the regional supply would run free. Regional supplies would normally run within 3 parts in 10^{10} .[1] The JFS also needed an improved reference signal. A new Bell System reference frequency standard was implemented using three Cesium atomic clocks located in Hillsborough, Missouri which maintain an accuracy of a few parts in 10^{12} with the national standard. The reference signal was transmitted to the JFS's using coaxial cable and microwave radio. The JFS's then passed on the reference signal to the traditional PFS's.

Stratum Levels

The North American network is now modeled on four stratum levels. Each level refers to the accuracy of the oscillator in it. A clock in a stratum is able to phase lock with any clock in the same or superior (lower numbered) stratum. The primary reference source (PRS) is Stratum 1 and is the highest level with the best accuracy (1 part in 10^{11}). This accuracy can only be met by a Cesium clock either onsite, via Loran-C or via global positioning system (GPS).

Table 1.2 strata requirements.

Stratum Levels	Accuracy	Minimum Stability
Stratum 1	1×10^{-11}	N/A
Stratum 2	1.6×10^{-8} (.0025 Hz at 1.544 MHz)	1×10^{-10}/day
Stratum 3	4.6×10^{-6} (7 Hz at 1.544 MHz)	< 255 slips on any connecting link during the initial 24 hours
Stratum 4	32×10^{-6} (50 Hz at 1.544 MHz)	N/A

The list of strata requirements as stated in ANSI T1.101 -1987 are contained in Table 1.[3]

By ensuring that all Stratum 1 oscillators are extremely accurate and are matched to the same reference (a world standard) different networks that contain separate Stratum 1 sources can be connected without frequency synchronization problems. The CCITT Rec. G.811 recommended that a primary reference clock be used for international switching centers. The clock should not have a longterm frequency departure of greater than 1×10^{-10} and should use Coordinated Universal Time (UTC) as its reference. Using this method the theoretical slip rate on any 64 kbps channel should not be greater than one in 70 days. (Note that this slip rate is based on undisturbed conditions.) In this way connection between separate networks would not require transferring timing information. Each network controlled by its own Stratum 1 clock should ensure that their connection together is synchronous because they are timed to the same reference clock. This is also the method used to connect different networks (eg ATT and MCI).

Timing information is distributed through this network using the T1 carrier signal. The timing information can be derived from the framing rate or bit rate of the signal since it is known to have a 1.544 Mbps transfer rate. It can be framed as all 1's or carry traffic but either way is traceable back to the Stratum 1 clock signal[4].

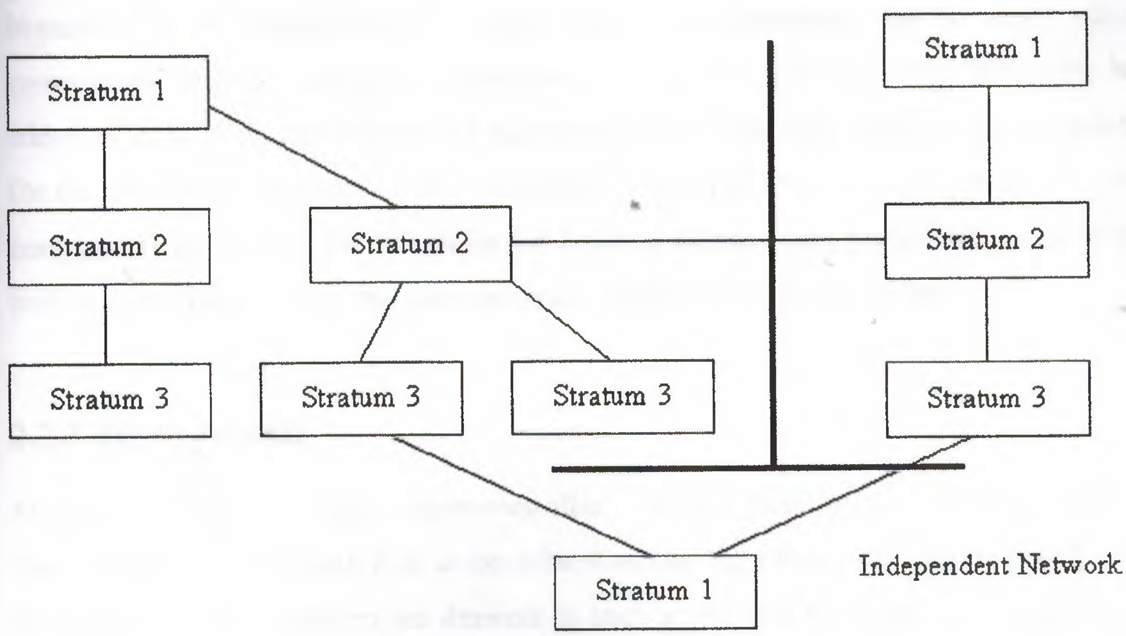


Figure 1.13 Stratum level network.

CHAPTER TWO
PIC MICROCONTROLLERS

2.1 Introduction to Microcontrollers

Circumstances that we find ourselves in today in the field of microcontrollers had their beginnings in the development of technology of integrated circuits. This development has made it possible to store hundreds of thousands of transistors into one chip. That was a prerequisite for production of microprocessors , and the first computers were made by adding external peripherals such as memory, input-output lines, timers and other. Further increasing of the volume of the package resulted in creation of integrated circuits. These integrated circuits contained both processor and peripherals. That is how the first chip containing a microcomputer , or what would later be known as a microcontroller came about.

2.2 Microcontrollers versus Microprocessors

Microcontroller differs from a microprocessor in many ways. First and the most important is its functionality. In order for a microprocessor to be used, other components such as memory, or components for receiving and sending data must be added to it. In short that means that microprocessor is the very heart of the computer. On the other hand, microcontroller is designed to be all of that in one. No other external components are needed for its application because all necessary peripherals are already built into it. Thus, we save the time and space needed to construct devices.

2.2.1 Memory unit

Memory is part of the microcontroller whose function is to store data. The easiest way to explain it is to describe it as one big closet with lots of drawers. If we suppose that we marked the drawers in such a way that they can not be confused, any of their contents will then be easily accessible. It is enough to know the designation of the drawer and so its contents will be known to us for sure.

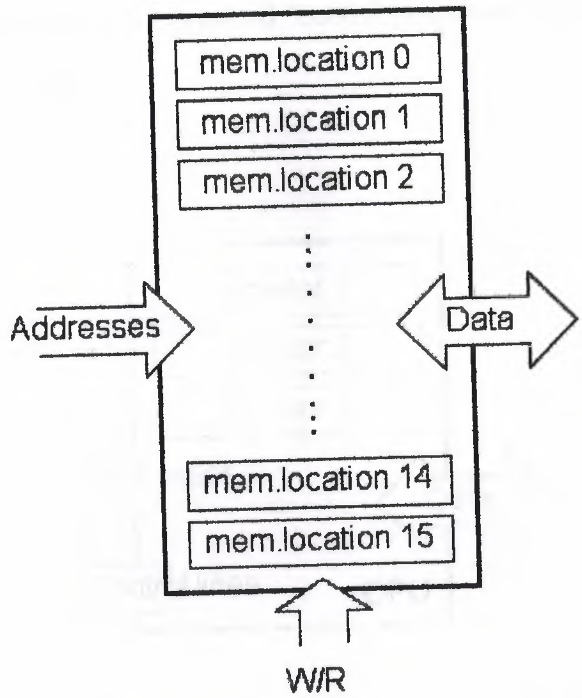


Figure 2.1 Example of simplified model of a memory unit.

Memory components are exactly like that. For a certain input we get the contents of a certain addressed memory location and that's all. Two new concepts are brought to us: addressing and memory location. Memory consists of all memory locations, and addressing is nothing but selecting one of them. This means that we need to select the desired memory location on one hand, and on the other hand we need to wait for the contents of that location. Beside reading from a memory location, memory must also provide for writing onto it. This is done by supplying an additional line called control line. We will designate this line as R/W (read/write). Control line is used in the following way: if r/w=1, reading is done, and if opposite is true then writing is done on the memory location. Memory is the first element, and we need a few operation of our microcontroller .

2.2.2 Central Processing Unit

Let add 3 more memory locations to a specific block that will have a built in capability to multiply, divide, subtract, and move its contents from one memory location onto another. The part we just added in is called "central processing unit" (CPU). Its memory locations are called registers.

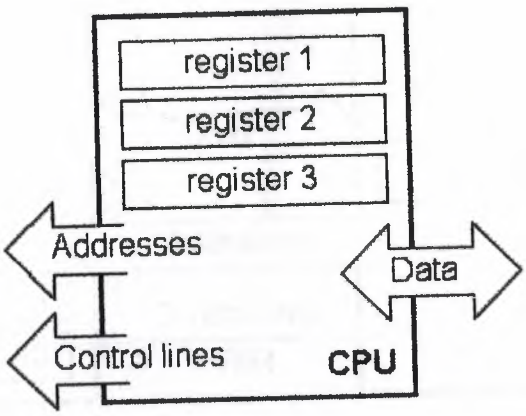


Figure 2.2 Example of simplified central processing unit with three registers.

Registers are therefore memory locations whose role is to help with performing various mathematical operations or any other operations with data wherever data can be found. Look at the current situation. We have two independent entities (memory and CPU) which are interconnected, and thus any exchange of data is hindered, as well as its functionality. If, for example, we wish to add the contents of two memory locations and return the result again back to memory, we would need a connection between memory and CPU. Simply stated, we must have some "way" through data goes from one block to another.

2.2.3 Bus

That "way" is called "bus". Physically, it represents a group of 8, 16, or more wires There are two types of buses: address and data bus. The first one consists of as many lines as the amount of memory we wish to address, and the other one is as wide as data, in our case 8 bits or the connection line. First one serves to transmit address from CPU memory, and the second to connect all blocks inside the microcontroller.

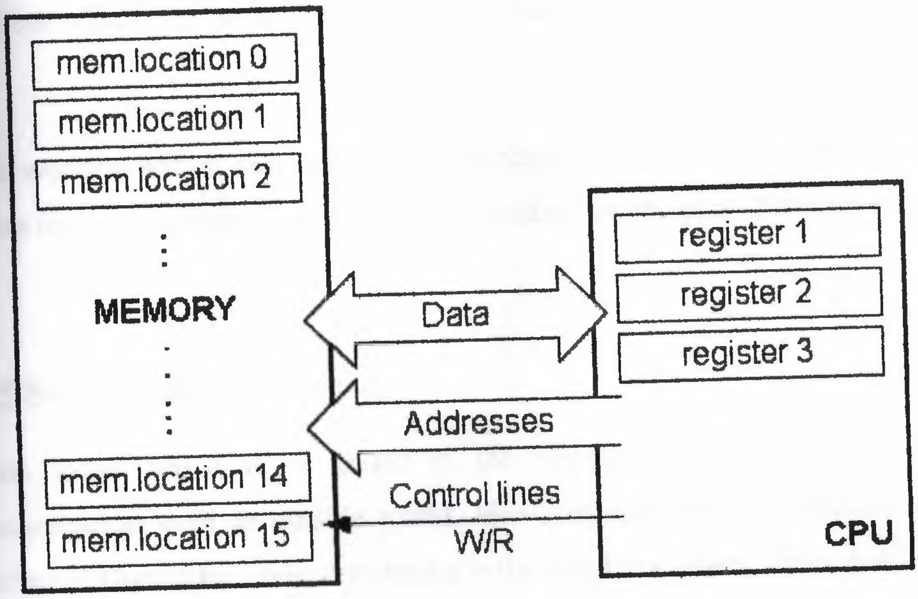


Figure 2.3 connecting memory and central unit.

As far as functionality, the situation has improved, but a new problem has also appeared: we have a unit that's capable of working by itself, but which does not have any contact with the outside world, or with us! In order to remove this deficiency, let's add a block which contains several memory locations whose one end is connected to the data bus, and the other has connection with the output lines on the microcontroller which can be seen as pins on the electronic component.

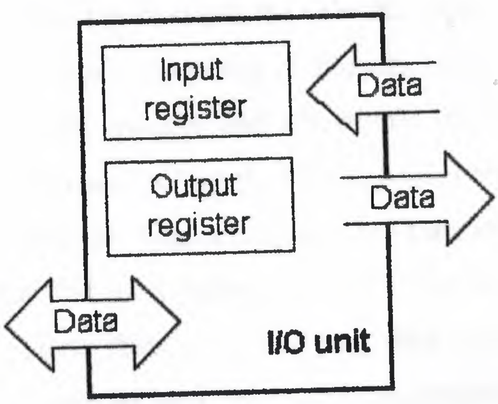


Figure 2.4 example of a simplified input-output unit provides communication.

2.2.4 Input-Output unit

Those locations we've just added are called "ports". There are several types of ports : input, output or bidirectional ports. When working with ports, first of all it is necessary to choose which port we need to work with, and then to send data to, or take it from the port.

When working with it the port acts like a memory location. Something is simply being written into or read from it, and it could be noticed on the pins of the microcontroller.

2.2.5 Serial Communication

Beside stated above we've added to the already existing unit the possibility of communication with an outside world. However, this way of communicating has its drawbacks. One of the basic drawbacks is the number of lines which need to be used in order to transfer data. What if it is being transferred to a distance of several kilometers? The number of lines times number of kilometers doesn't promise the economy of the project. It leaves us having to reduce the number of lines in such a way that we don't lessen its functionality. Suppose we are working with three lines only, and that one line is used for sending data, other for receiving, and the third one is used as a reference line for both the input and the output side. In order for this to work, we need to set the rules of exchange of data.

These rules are called protocol. Protocol is therefore defined in advance so there wouldn't be any misunderstanding between the sides that are communicating with each other. For example, if one man is speaking in French, and the other in English, it is highly unlikely that they will quickly and effectively understand each other. Let's suppose we have the following protocol. The logical unit "1" is set up on the transmitting line until transfer begins. Once the transfer starts, we lower the transmission line to logical "0" for a period of time (which we will designate as T), so the receiving side will know that it is receiving data, and so it will activate its mechanism for reception. Let's go back now to the transmission side and start putting logic zeros and ones onto the transmitter line in the order from a bit of the lowest value to a bit of the highest value. Let each bit stay on line for a time period which is equal to T, and in the end, or after the 8th bit, let us bring the logical unit "1" back on the line which will mark the end of the transmission of one data. The protocol we've just described is called in professional literature NRZ (Non-Return to Zero).

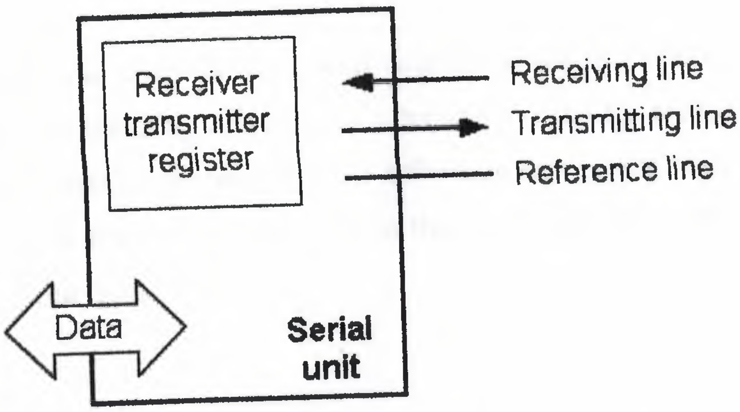


Figure 2.5 serial unit used to send data, but only by three lines.

As we have separate lines for receiving and sending, it is possible to receive and send data (info.) at the same time. So called full-duplex mode block which enables this way of communication is called a serial communication block. Unlike the parallel transmission, data moves here bit by bit, or in a series of bits what defines the term serial communication comes from. After the reception of data we need to read it from the receiving location and store it in memory as opposed to sending where the process is reversed. Data goes from memory through the bus to the sending location, and then to the receiving unit according to the protocol.

2.2.6 Timer Unit

Since we have the serial communication explained, we can receive, send and process data. However, in order to utilize it in industry we need a few additionally blocks. One of those is the timer block which is significant to us because it can give us information about time, duration, protocol etc.

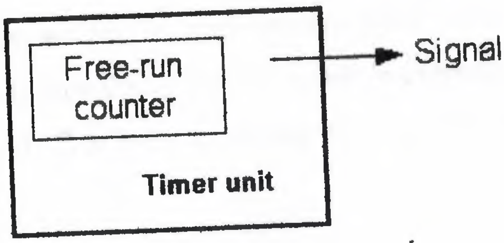


Figure 2.6 timer unit generates signals in regular time intervals.

The basic unit of the timer is a free-run counter which is in fact a register whose numeric value increments by one in even intervals, so that by taking its value during periods T1 and T2 and on the basis of their difference we can determine how much time has elapsed. This is a very important part of the microcontroller whose understanding requires most of our time.

2.2.7 Watchdog

One more thing is requiring our attention is a flawless functioning of the microcontroller during its run-time. Suppose that as a result of some interference (which often does occur in industry) our microcontroller stops executing the program, or worse, it starts working incorrectly.

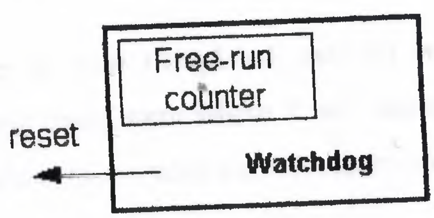


Figure 2.7 watchdog reset.

Of course, when this happens with a computer, we simply reset it and it will keep working. However, there is no reset button we can push on the microcontroller and thus solve our problem. To overcome this obstacle, we need to introduce one more block called watchdog. This block is in fact another free-run counter where our program needs to write a zero in every time it executes correctly. In case that program gets "stuck", zero will not be written in, and counter alone will reset the microcontroller upon achieving its maximum value. This will result in executing the program again, and correctly this time around. That is an important element of every program to be reliable without man's supervision.

2.2.8 Analog to Digital Converter

As the peripheral signals usually are substantially different from the ones that microcontroller can understand (zero and one), they have to be converted into a pattern which can be comprehended by a microcontroller. This task is performed by a block for analog to digital conversion or by an ADC. This block is responsible for converting an information about some analog value to a binary number and for follow it through to a CPU block so that CPU block can further process it.

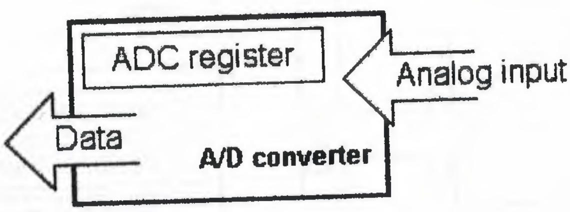


Figure 2.8 block for converting an analogue to a digital form.

Finally, the microcontroller is now completed, and all we need to do now is to assemble it into an electronic component where it will access inner blocks through the outside pins. The picture below shows what a microcontroller looks like inside.

Thin lines which lead from the center towards the sides of the microcontroller represent wires connecting inner blocks with the pins on the housing of the microcontroller so called bonding lines. Chart on the following page represents the center section of a microcontroller.

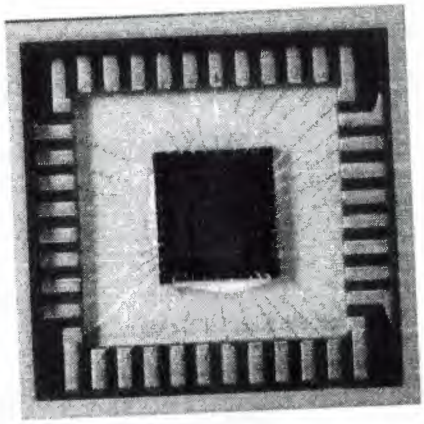


Figure 2.9 Physical configuration of the interior of a microcontroller.

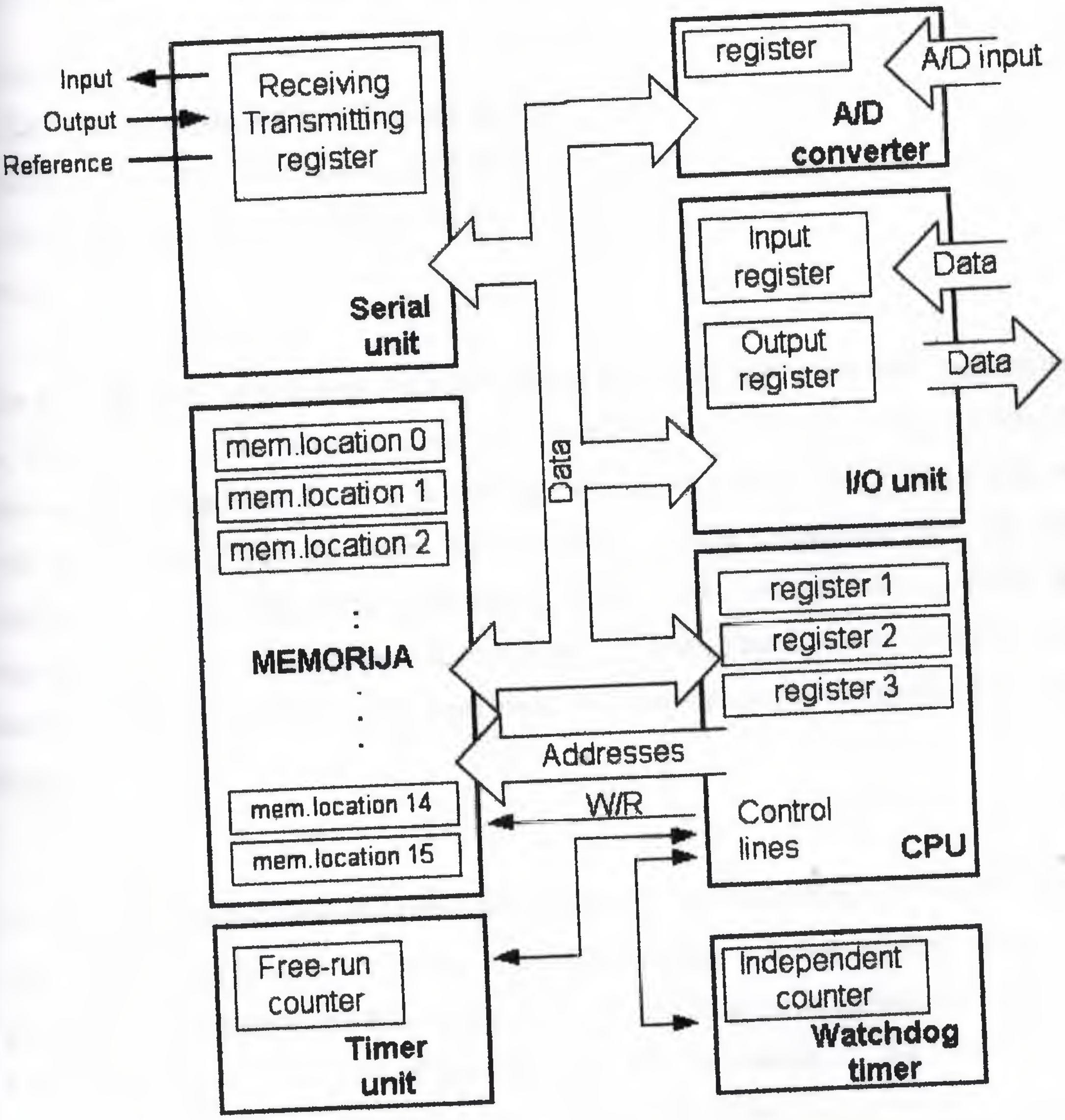


Figure 2.10 Microcontroller outline with its basic elements and internal connections

For a real application, a microcontroller alone is not enough. Beside a microcontroller, we need a program that would be executed, and a few more elements which make up a interface logic towards the elements of regulation.

2.2.9 Program

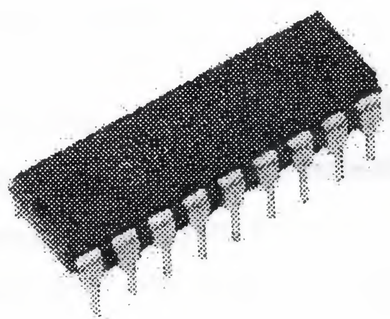
Program writing is a special field of work with microcontrollers and is called "programming". Try to write a small program in a language that we will make up ourselves first and then would be understood by anyone.

```
START
REGISTER1=MEMORY LOCATION_A
REGISTER2=MEMORY LOCATION_B
PORTA=REGISTER1 + REGISTER2
END
```

The program adds the contents of two memory locations, and views their sum on port A. The first line of the program stands for moving the contents of memory location "A" into one of the registers of central processing unit. As we need the other data as well, we will also move it into the other register of the central processing unit. The next instruction instructs the central processing unit to add the contents of those two registers and send a result to port A, so that sum of that addition would be visible to the outside world. For a more complex problem, program that works on its solution will be bigger.

Programming can be done in several languages such as Assembler, C and Basic which are most commonly used languages. Assembler belongs to lower level languages that are programmed slowly, but take up the least amount of space in memory and gives the best results where the speed of program execution is concerned. As it is the most commonly used language in programming microcontrollers it will be discussed in a later chapter. Programs in C language are easier to be written, easier to be understood, but are slower in executing from assembler programs. Basic is the easiest one to learn, and its instructions are nearest a man's way of reasoning, but like C programming language it is also slower than assembler. In any case, before you make up your mind about one of these languages you need to consider carefully the demands for execution speed, for the size of memory and for the amount of time available for its assembly. After the program is written, we would install the microcontroller into a device and run it. In order to do this we need to add a few more external components necessary for its work. First we must give life to a microcontroller by connecting it to a power supply (power needed for operation of all electronic instruments) and oscillator whose role is similar to the role that heart plays in a human body. Based on its clocks microcontroller executes instructions of a program. As it receives supply microcontroller will perform a small check up on itself, look up the beginning of the program and start executing it. How the device will work depends on many parameters, the most important of which is the skillfulness of the developer of hardware, and on programmer's expertise in getting the maximum out of the device with his program[5].

2.3 PIC Microcontroller



PIC(Peripheral Interface Controller) is the IC which was developed to control the peripheral device, dispersing the function of the main CPU.

When comparing to the human being, the brain is the main CPU and the PIC shares the part of which is equivalent to the autonomic nervous.

PIC has the calculation function and the memory like the CPU and is controlled by the software. However, the throughput, the memory capacity aren't big. It depends on the kind of PIC but the maximum operation clock frequency is about 20 MHz and the memory capacity to write the program is about 1K to 4K words. The clock frequency is related with the speed to read the program and to execute the instruction. Only at the clock frequency, the throughput can not be judged. It changes with the architecture in the processing part. As for the same architecture, the one with the higher clock frequency is higher about the throughput.

I used the WORD for the capacity of the program memory. It represents the one instruction as being the 1 word. It often uses the BYTE to show the capacity of the memory. The 1 byte shows the 8 bits. The bit is the atomicity which shows 1 or 0. The instruction of the PIC16F84A is composed of the 14 bits. It is 1 x 1,024 x 14 = 14,336 bits when converting the 1K words to the bit. It is 14,336/(8 x 1,024) = 1.75K bytes when converting this to the byte.

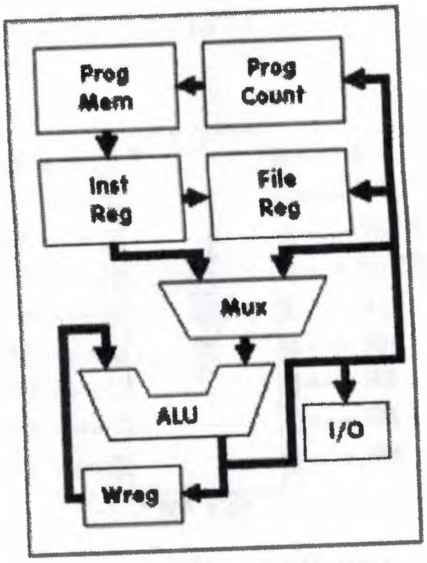


Figure 2.11 PIC microcontroller structure.

At the memory capacity, it is the 1G bytes = 1,024M bytes, 1M bytes = 1,024K bytes, 1K bytes = 1,024 bytes. It is not 1000 times. This is because it calculates in the binary. The point which the PIC is convenient for is that the calculation part, the memory, the input/output part and so on are incorporated into one piece of the IC. The efficiency, the function are limited but can compose the control unit only by the PIC even if it doesn't combine the various ICs. So, the circuit can be compactly made.

2.3.1 Pin Diagram

OSC1/CLKIN : Oscillator crystal input.
External clock source input.

OSC2/CLKOUT : Oscillator crystal output.Connects to crystal or resonator in crystal oscillator mode.

MCLR(inv) : Master clear(reset)input.Programming voltage input. This pin is an active low reset to the device.

RA0 - RA3 : Bi-directional I/O port.

RA4/T0CKI : Bi-directional I/O port. Clock input to the TMR0 timer/counter.

RB0/INT : Bi-directional I/O port. External interrupt pin.

RB1 - RB7 : Bi-directional I/O port.

V_{SS} : Ground

V_{DD} : Positive supply(+2.0V to +5.5V)

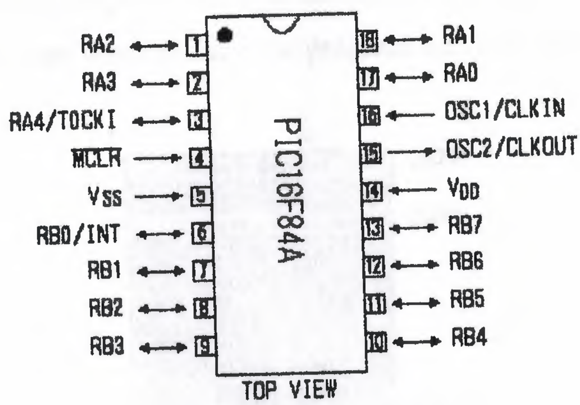


Figure 2.12 PINs of PIC19F84A.

2.3.2 Hardware of PIC16F84A

Flash Program Memory

The flash memory is used for the memory which stores the program. The 1 word is composed of the 14 bits and 1,024 words (the 1K words) are installed. Even if it switches off the power supply, the contents which is stored in the flash memory don't disappear. The contents of the flash memory can be rewritten using the writer. But, the rewritten number of times is limited. It is the about 1000 times.

Reset Vector (0000h)

When the reset is executed by the turning on, WDT(Watchdog Timer) time-out, the other factor, the program starts after the reset from this address.

Peripheral Interrupt Vector (0004h)

When there is the time-out interruption of the timer(TMR0), the interruption from outside and so on, the program starts from this address.

Configuration word (2007h)

The basis operation of the PIC is specified by this memory. The enable bit of the Power-up timer, the enable bit of the Watchdog timer, the Oscillator Selection bits can be set. This area is behind usual program area and can not set by the program. It is necessary to be written using the writer when writing the program into the flash memory.

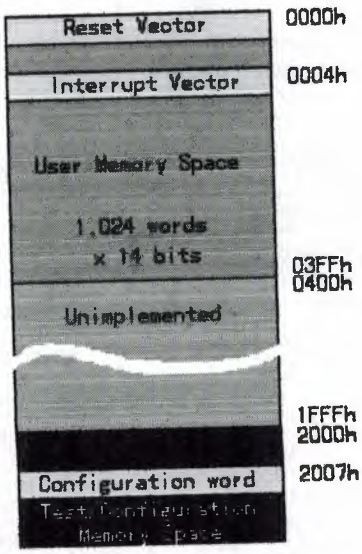


Figure 2.13 flash program memory.

RAM(Random Access Memory) File Registers

The management structure, the bank, is adopted to this memory. The memory capacity is the 80 bytes(00h-4Fh) per bank. In case of the PIC16F84A, there are two banks. This memory is used, dividing into the two areas. The first 12 bytes(00h-0Bh) of each bank are called SFR(Special Function Registers) and are used to record the operating states of the PIC, the conditions of the input/output ports, the other conditions. Each use is decided.

The 68 bytes(0Ch-4Fh) of the 13rd byte since then are called GPR(General Purpose Registers) and are possible to make record the results and the conditions on the way which executes the program temporarily.

The contents which depend on each bank are managed and there are 16 kinds of registers in SFR. But, the part of SFR is common on the bank, so, it is not the 24 kinds. Even if it changes the bank, because the contents in the GPR area are the same completely, they are the capacity of 68 bytes of the substance.

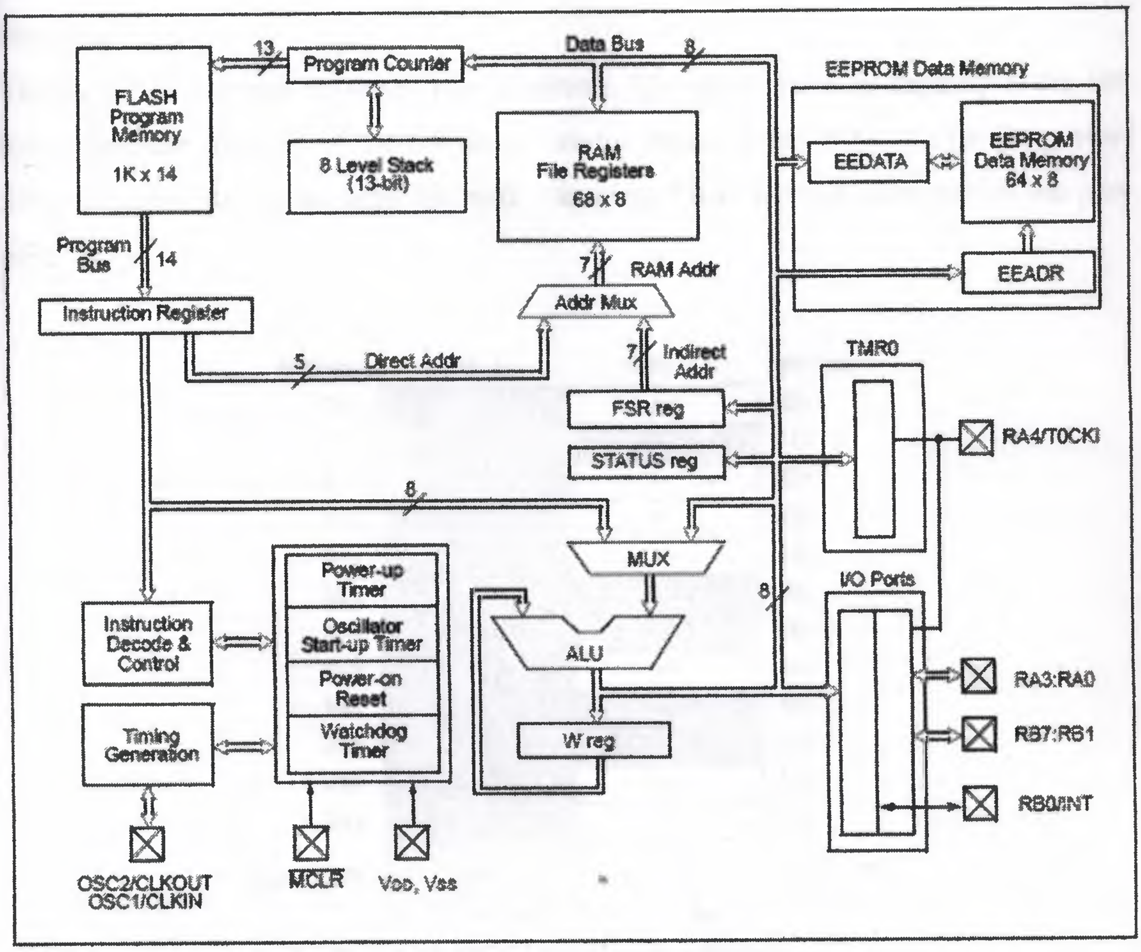


Figure 2.14 block diagram of PIC16F84A.

When the power supply is switched off, the contents are lost. There is not limitation on the number of times to rewrite.

EEPROM(Electrically Erasable Programmable Read Only Memory)

This memory is the type which maintains the contents even if it switches off the power supply. The contents of this memory can be rewritten by the program. The memory capacity is the 64 bytes. As for this memory, the rewriting number of times is limited. It is the about one million time. So, it isn't possible to use to store the data on the processing way and so on. It is used to store the data with few change frequencies. This memory is said to be able to maintain the memorized contents for the about 40 years.

SFR Registers

SFR(Special Function Registers) can specify 16 kinds of the registers by the bank changing.

The figure 2.15 shows the RAM File Registers. The whole memory capacity is the 160 bytes. But, the contents of the left arrow are the same on any bank. As for the part of SFR, the contents change with the bank changing. There are not memories in the gray part.

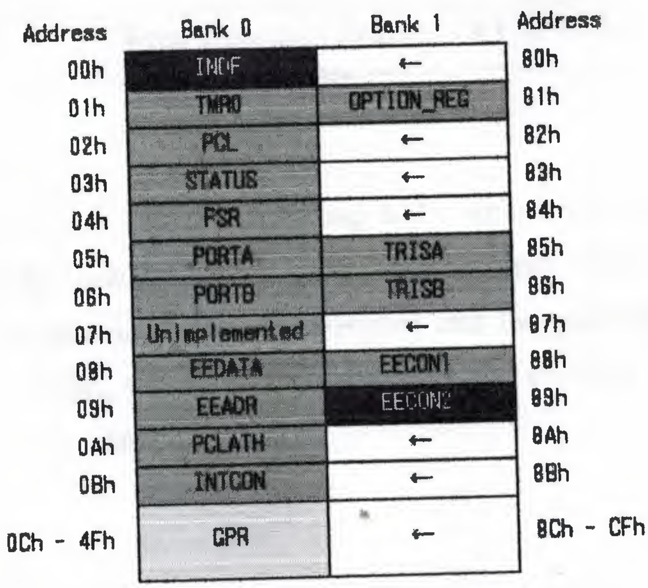


Figure 2.15 special function registers.

Each SFR has a function as listed in table 2.1.

Table 2.1 SFR functions.

INDF	Data memory contents by indirect addressing
TMR0	Timer counter
PCL	Low order 8 bits of the program counter

STATUS	Flag of the calculation result
FSR	Indirect data memory address pointer
PORTA	PORTA DATA I/O
PORTB	PORTB DATA I/O
EEDATA	Dtata for EEPROM
EEADR	Address for EEPROM
PCLATH	Write buffer for upper 5 bits of the program counter
INTCON	Interruption control
OPTIN_REG	Mode set
TRISA	Mode set for PORTA
TRISB	Mode set for PORTB
EECON1	Control Register for EEPROM
EECON2	Write protection Register for EEPROM

Program Counter

This is the counter which shows the reading address(Fetch address) of the program which is written in the flash memory. It is the 13 bits counter. Generally, the one count rises every time the one instruction is executed and the position of the following instruction is shown. But, when JUMP is executed, the contents of this counter are rewritten to the jumping address.

8 Level Stack

The stack is the memory which stores the return address of the program. For example, when doing the same processing at more than one, it makes the processing in the form of the subroutine. At the end of the subroutine, the instruction of RETURN is written. The memory is saved in making the processing the subroutine. At the program which uses the subroutine, it jumps to the subroutine using the instruction of CALL. At this time, the return address is stored in the stack. This operation is sometimes called the PUSH. When the processing by the subroutine ends and the instruction of RETURN is executed, it jumps to the return address which is written at e stack. This operation is sometimes called the POP. If being in this way, even if ALL to the subroutine is written at more than one part, the processing can be returned the original program which jumped to the subroutine. he fact that there are eight stacks can do the subroutine the eight times at the serial. Then doing the subroutine the nine times, the return address has been written at the rst stack. Because the contents of the first stack are rewritten, the processing can not e returned to the original program. The eight times can not be exceeded. he subroutine must always return the processing to the called original program using he instruction of RETURN. Don't return by the instruction of JUMP.

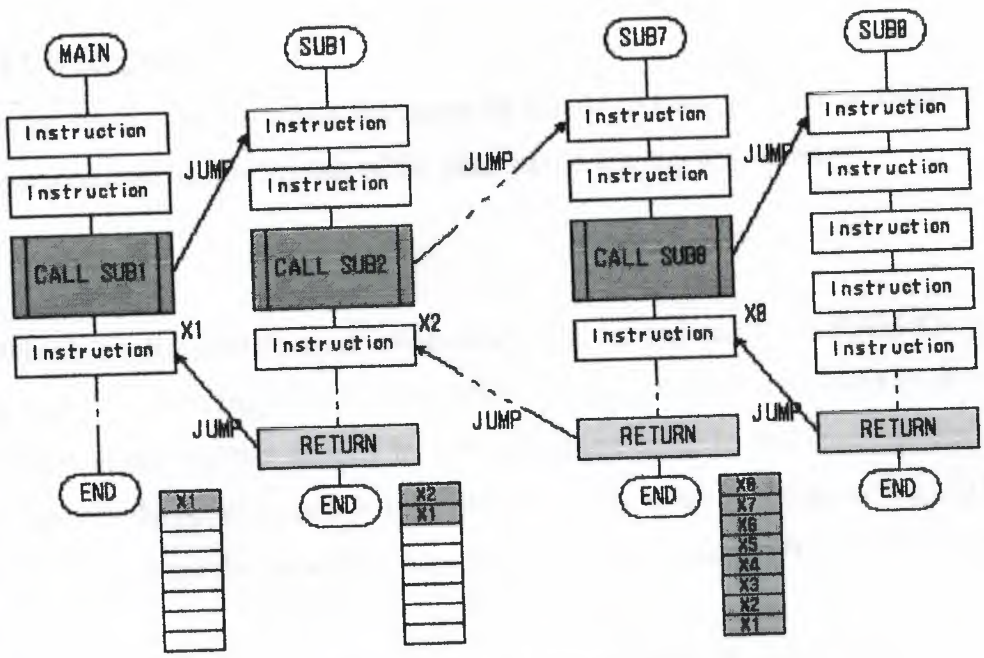


Figure 2.16 call instruction.

Instruction Register

The instruction of the program which is specified by the program counter is read to this register. This operation is called the FETCH.

Instruction Decode & Control

The instruction which was fetched to the instruction register is analyzed here and the operation according to the contents is done.

Iultiplexer and Arithmetic Logic Unit

he calculation operation is done by the Multiplexer and the Arithmetic Logic Jnit(ALU). It is not the computer when there are not these.

V Register

This is the work register. It is used to keep the calculation result of the ALU emporarily. For the calculation operation, it is the indispensable register. The contents of this register are stored in the various register and are utilized. It is used for the output control of the input-output port, too.

STATUS Register

This is the register which stores the result of the ALU(Zero, positive or negative), the time-out condition, the indication of the bank of the file register, so on.

FSR Register

FSR(File Select Register) is used when specifying the address of the RAM file register by the indirect addressing method. The direct address method is the way of specifying the address of the register directly by the instruction code. In this case, the addressing bit can specify the address from 0 to 127 by the 7 bits. This specification range is for the one bank. To change the bank, it is necessary to combine with RP0 bits of the STATUS register.

Because FSR is the 8 bits, it is possible to be specified at once including the bank specification. In PIC16F84A , the memory is not installed at the address of 80(50h) to 127(7Fh).

It is convenient when using the FSR for the addressing when making the data area which was continued at the file register. The processing is simplified when inclement the FSR when doing the writing or the reading continuously.

Address Multiplexer

It distinguishes the indirect addressing or the direct address.

EEDATA

This is the register to use when writing or reading the data to or from the EEPROM.

EEADR

This is the register which specifies the address of the EEPROM. Because are composed of the 8 bits, the address from 0 to 255 can be specified. In PIC16F84A, the EEPROM is only 64 bytes installed. When setting the data of the EEPROM by the source coding, it specifies 2100h as the memory address. When writing data to the EEPROM by the processing of a program, it is necessary to do the processing to set 55h and AAh to the EECON2 register in the order.

Timer

PIC16F84A has the one timer(TMR0) with the 8 bits. It is in the time-out when the count becomes 256 and the T0IF bit of the INTCON register of SFR becomes "1". It is possible to make the interrupt occur when being in the time-out. The interrupt is to make process the time-out, making the processing which goes at the point stop. To make the interrupt occur, the GIF bit and the T0IE bit of the INTCON register of SFR must be made "1".

I/O Ports

There are 13 pins with individual direction control. The mode (the input or the output) of each pin can be set by the program. The 13 pins are divided into the two groups. They are the five pins as the A port and the eight pins as the B port. There is limitation on the timing of the control but each of the 13 pins can be controlled.

Timing Generation

This circuit generates the clock pulse which fixes the operation speed. the oscillation operation is done by putting the capacitors the crystal(or ceramic) oscillator outside. When having the oscillation with the high stability, it uses the crystal. Generally, the circuit becomes simple when using the resonator which incorporated the ceramic and the capacitors into the one module. The clock pulse can be inputted from outside, too. The PIC16F84A execute the one instruction (the 1 cycle) by the four clock pulses using pipeline architecture. But, in case of the JUMP to change the program address, the 2 les are necessary. In the execution time of usual instruction, it is the 200 oseconds because the pulse period of 20MHz is 1/(20MHz) = 50 nanoseconds. The 00,000 instructions can be executed within the 1 second.

itialization circuits

e PIC16F84A has the various initialization circuits.

)WON mer	: This is the timer to limit the operation until the voltage is stable in case of the turning on.
SC tartTimer	: This is the timer to limit the operation until the clock is stable in case of the turning on.
OW ON eset	: This initializes the inner circuit of the PIC in case of the turning on.
Vatchdog Timer	: This is the timer to watch over the normal operation of the software of PIC. This timer must be regularly cleared by the software. When the timer does in the time-out, the PIC returns to the condition immediately after the turning on. This timer is used to recover the extraordinary operation when the software has the defect(the bug). Even if it is initialized, the bug doesn't pass away.

2.3.3 Radix

In the software of the computer, hexadecimal is often used.

Decimal

In our daily life, 10 numbers from "0" to "9" are used. In case of the count-up, after 9, the carry is done and becomes 10. We are using properly but this is the count method of the number which the human being decided. This seems to depend on that the fingers of the hand of the person is 10.

nary

ly two values of "0" and "1" are used to express a condition by the digital world ich included a computer. These are sometimes expressed by "Low level " and "High el ", too. Like 0 →1 →10, after 1, it is 10.

exadecimal

the condition to be handling with the computer, it is a binary number but it is difficult understand for the person who is using the decimal. 163(Decimal) → 100011(Binary). Therefore, a hexadecimal is used as the expression which it is easy r the person to understand. As for the hexadecimal, 16 numbers are used for 1 digit. It from 0 to 9. So, six of the remainder are expressed by the alphabet. 10 →A, 11→ B, 2 →C, 13 →D, 14 →E, 15 →F. The figure has begun with 0. Therefore, 10 of the figure ows the 11th and 15 shows the 16th. 16 kinds of conditions are expressed by 4 bits in e binary. Oppositely, the hexadecimal is used because that it is possible to express 16 y 4 bits. There is the octal which is expressed by 3 bits. In case of the hexadecimal, 1 yte is expressed by 2 digits. Also, it puts "h" to distinguish the hexadecimal from the ecimal. h is the initial of hexadecimal(16). It shows in 00h or H'00' or 0x00. However, n expression isn't unified.

The correspondence of radix

The change of Binary, Decimal and Hexadecimal can make simple if you use the unction electronic calculator which is attached to the Windows. When changing without using the calculator, it is possible to do in the following way.

B → H : It is possible to change simply if dividing 4 bits. It is easier if learning a binary pattern to Fh from Ah.

EX. 111000100100010000100 →1C4884h

H → B : It is possible to change 1 digit of the hexadecimal into the binary in the order.

EX. 5F37Bh →1011111001101111011

B → D : This is troublesome a little. First, you write a decimal value every bit like the following figure. Total the decimal value of the bit of "1".

512 256 128 64 32 16 8 4 2 1 EX.512 + 256 + 128 + 8 + 4 + 1 = 909

1 1 1 0 0 0 1 1 0 1

→ B : This is terrible a little, too. Subtract the maximum number of power of two(1,2,4,8,16,32,64,128,256,512,1024,...) which can be subtracted from the decimal number. It makes the bit which corresponds to the number of power of two which could be subtracted to "1". Subtract the number of power of two which could be subtracted from the remainder more. Hereinafter, repeat similar subtraction until the remainder passes away. The row of "1" and "0" by above result is a binary number.

EX. "582" 582 - 512 = 70, 70 - 64 = 6, 6 - 4 = 2, 2 - 2 = 0

512 256 128 64 32 16 8 4 2 1

1 0 0 1 0 0 0 1 1 0

D → H : Change a decimal into the binary first and change a result into the hexadecimal more.In case of the example(D →B) which was shown above, 582 = 1001000110 = 246h.

When changing directly, there is a way of dividing by the value of 4 bits.

582 / 256 = 2 remainder 70

70 / 16 = 4 remainder 6

The result is 246h.

H → D : Change a hexadecimal into the binary first and change a result into the decimal more.

In case of the example(B →D) which was shown above, 38Dh = 1110001101 = 909

When changing directly, there is a way of multiplying the value of 4 bits.

3 x 256 = 768

8 x 16 = 128

The result is 768 + 128 + 13(Dh) = 909.

Table 2.2 The correspondence of Binary, Decimal and Hexadecimal.

Decimal	Binary	Hexdecimal
0	0	0h
1	1	1h
100	1100100	64h
127	1111111	7Fh

2	10	2h	128	10000000	80h
3	11	3h	200	11001000	C8h
4	100	4h	255	11111111	FFh
5	101	5h	256	100000000	100h
6	110	6h	300	100101100	12Ch
7	111	7h	400	110010000	190h
8	1000	8h	500	111110100	1F4h
9	1001	9h	511	111111111	1FFh
10	1010	Ah	512	1000000000	200h
11	1011	Bh	600	1001011000	258h
12	1100	Ch	700	1010111100	2BCh
13	1101	Dh	800	1100100000	320h
14	1110	Eh	900	1110000100	384h
15	1111	Fh	1000	1111101000	3E8h
16	10000	10h	1023	1111111111	3FFh
17	10001	11h	1024	10000000000	400h
18	10010	12h	2000	11111010000	7D0h
19	10011	13h	2047	11111111111	7FFh
20	10100	14h	2048	100000000000	800h

Still, it is easy to calculate using the function electronic calculator.

2's Complement

The 2's complement is the one which shows negative numerical value. For example, "-1" of the decimal number is 11111111 when showing by the binary with te.

is as follows when confirming.

```
00000001   (1)
    +
11111111   (-1)
--------
00000000   (0)
```

he overflow occurs but the numerical value becomes zero. binary addition is done from the lower rank figure like the decimal number. When here is a carry, it calculates a higher rank figure including it.

To use a negative value, there is a condition. The numerical value which it is possible to show at the byte is 256 kinds of 0 to 255. However, when using the negative value, it becomes 255 kinds of - 127 to +127. The reason why the numbers are few is because 10000000 isn't used. The row of these bits shows -0 but in the calculation, it can not use. Most significant bit 7 is used as the sign bit which shows negative or positive. The type with numerical value must be considered when processing it. For example, it is 10000001 when showing -127 in the binary number. It becomes 129, supposing that this is only plus numerical value.

A change into the 2's complement is done as follows. I attempt to change 56 into -56 as the example.

(1) Subtract 1 from the value   56 - 1 = 55

(2) Change this into the binary   55 →00110111

(3) It makes 0 and 1 opposite   00110111 →11001000

11001000 is the binary number which shows -56.

It is as follows when confirming.

```
(+56) → 00111000
            +
(-56) → 11001000
        --------
        00000000
```

The answer became zero.

4 General format for instructions

format for instructions of PIC16 series is the following three kinds. The instructions written in the program memory and one instruction is composed of 14 bits. These 14 are called a word.

te-oriented file register operations

instructions of this format are the instructions which processes a byte unit

ECODE : The code to distinguish a instruction is written.

(Destination It specifies the register which stores the execution result of the
ect) : instruction.

d=0 : It specifies working register (W reg).

d=1 : It specifies file register which is specified by f.

In case of the assembler language, d is written in W or F.

(EX)	ADDWF	COUNT,W	(d=0 when writing W)
	ADDWF	COUNT,F	(d=1 when writing F)

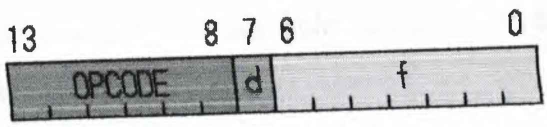


Figure 2.17 Byte-oriented file register operations.

(Register file) : It specifies the address of the register which is dealt with for the instruction.

f can specify an address from 0(00h) to 127(7Fh) because it is 7 bits.

In case of PIC16F84A, because the register memory is 80 bytes including SFR, it is possible to be specified by f if being 7 bits. When writing by the assembler language, the label is put to the register and uses.

(EX) ADDWF COUNT,F (COUNT is the label of the register)

t-oriented file register operations

e instructions of this format are the instruction which processes a bit unit

Figure 2.18 Bit-oriented file register operations.

PECODE : The code to distinguish a instruction is written.

it address): It specifies the bit position of the register file. Because the register file is a byte, it can specify all bit positions with 3-bit b.

Register e): It specifies the address of the register which is dealt with for the instruction.
f can specify an address from 0(00h) to 127(7Fh) because it is 7 bits. In case of PIC16F84A, because the register memory is 80 bytes including SFR, it is possible to be specified by f if being 7 bits. When writing by the assembler language, the label is put to the register and uses.

Literal and control operations

The instructions of this format do the processing which used the fixed number (k) which was written in the instruction.
There are two instruction types and fixed number (k) is 11 bits about GOTO and CALL instruction.

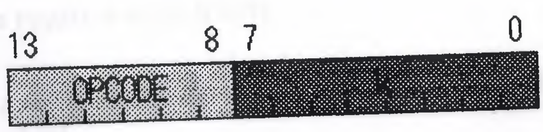


Figure 2.19 GOTO and CALL instructions.

case of GOTO and CALL instruction:

ECODE : The code to distinguish a instruction is written.

Literal (ld): This is the fixed number to use for the calculation.
It specifies by the numerical value or the label.
Because it is a byte except JUMP and CALL instruction, it is the range of 0(00h) to 255(FFh).
Because GOTO or CALL instruction is 11 bits, it is the range of 0(00h) to 2047(7FFh).

nstruction set of PIC16 series

n PIC16 series, RISC(Reduced Instruction Set Computer) is adopted and the number of he instructions to use is 35 kinds. When clicking the mnemonic of each instruction, you an jump to the instruction specification.

The terminology explanation

Mnemonic : The assembler language which made an operation code plain

Operands : The specification part except the operation code

MSb : Most Significant bit

LSb : Least Significant bit

Flag : The field of the STAUS register

Table 2.3 instruction set of PIC16 series

Mnemonic Operands		Operation explanation	Instruction code MSb LSb	Flag
Byte-oriented file register operations				
ADDWF	f, d	Add W and f	00 0111 dfff ffff	C, DC, Z
ANDWF	f, d	AND W with f	00 0101 dfff ffff	Z
CLRF	f	Clear f	00 0001 1fff ffff	Z

CLRW	-	Clear W	00 0001 0xxx xxxx	Z
COMF	f, d	Complement f	00 1001 dfff ffff	Z
DECF	f, d	Decrement f	00 0011 dfff ffff	Z
DECFSZ	f, d	Decrement f, Skip if 0	00 1011 dfff ffff	
INCF	f, d	Increment f	00 1010 dfff ffff	Z
INCFSZ	f, d	Increment f, Skip if 0	00 1111 dfff ffff	
IORWF	f, d	Inclusive OR W with f	00 0100 dfff ffff	Z
MOVF	f, d	Move f	00 1000 dfff ffff	Z
MOVWF	f	Move W to f	00 0000 1fff ffff	
NOP	-	No Operation	00 0000 0xx0 0000	
RLF	f, d	Rotate Left f through Carry	00 1101 dfff ffff	C
RRF	f, d	Rotate Right f through Carry	00 1100 dfff ffff	C
SUBWF	f, d	Subtract W from f	00 0010 dfff ffff	C, DC, Z
SWAPF	f, d	Swap nibbles in f	00 1110 dfff ffff	
XORWF	f, d	Exclusive OR W with f	00 0110 dfff ffff	Z
Bit-oriented file register operations				
BCF	f, b	Bit Clear f	01 00bb bfff ffff	
BSF	f, b	Bit Set f	01 01bb bfff ffff	
BTFSC	f, b	Bit Test f, Skip if Clear	01 10bb bfff ffff	
BTFSS	f, b	Bit Test f, Skip if Set	01 11bb bfff ffff	

Literal and control operations				
ADDLW	k	Add literal and W	11 111x kkkk kkkk	C, DC, Z
ANDLW	k	AND literal with W	11 1001 kkkk kkkk	Z
CALL	k	Call subroutine	10 0kkk kkkk kkkk	
CLRWDT	-	Clear Watchdog Timer	00 0000 0110 0100	$\overline{TO}$, $\overline{PD}$
GOTO	k	Go to address	10 1kkk kkkk kkkk	
IORLW	k	Inclusive OR literal with W	11 1000 kkkk kkkk	Z
MOVLW	k	Move literal to W	11 00xx kkkk kkkk	
RETFIE	-	Return from interrupt	00 0000 0000 1001	
RETLW	k	Return with literal in W	11 01xx kkkk kkkk	
RETURN	-	Return from Subroutine	00 0000 0000 1000	
SLEEP	-	Go into standby mode	00 0000 0110 0011	$\overline{TO}$, $\overline{PD}$
SUBLW	K	Subtract W from literal	11 110x kkkk kkkk	C, DC, Z
XORLW	K	Exclusive OR literal with W	11 1010 kkkk kkkk	Z

X : Don't care[6]

2.3.5 PIC programmer

The programmer is used to download the hex file of the assembled program into the PIC chip, there are many types of PIC programmers designs, some of them fairly working, some with high protection, some programming limited models, some difficult to use and there is a special one that is very accurate and programming all the models sold by MICROCHIP company but its expensive. Me specially, I chose a simple to use, programming wide rang of models, low cost and simple to construct, I built it my self d downloaded its software debugger from internet, this programmer is called PIC-G2B serial port programmer, look at figure 2.20.

IC-PG2B serial port programmer

IC-PG2B is low cost serial port programmer for 8, 18, 28 and 40 pin PIC icrocontrollers. The programmer doesn't need external power supply and takes all ecessary signals and power from RS232 port .

IC-PG2B works with ICPROG software. The programmer can be used to program I2C erial EEPROM memory devices from 24Cxx series.

CPROG installation

Setup the Hardware settings as "JDM programmer" with direct IO access if you are using Windows 95/98 and Windows API if you are working with Windows NT .Please note that the programmer is powered from the RS232 port, so before you put or take off your device disconnect the programmer from the RS232 port!

Your RS232 cable must provide the following signals for properly operation of PIC-PG2: Tx ,Rx, CTS, DTR, RTS and GND.

Supported devices

Current supported devices by ICPROG are:

12C508, 12C508A, 12C509, 12C509A, 12CE518, 12CE519, 12C671, 12C672, 12CE673, 12CE674, 16C61, 16C62A, 16C62B, 16C63, 16C63A, 16C64A, 16C65A, 16C65B, 16C66, 16C67, 16C71, 16C72, 16C72A, 16C73A, 16C73B, 16C74A, 16C76, 16C77, 16C84, 16F83, 16F84, 16F84A, 16C505, 16C620, 16C621, 16C622, 16C622A, 16F627, 16F628, 16C715, 16F870, 16F871, 16F872, 16F873, 16F874, 16F876, 16F877, 16C923, 16C924.

PIC16F84A connection

Look at figure "programmer schematic" and join the following terminals:

PGC to PIN 12

PGD to PIN 13

VDD to PIN14

LR to PIN 4

D/PGM to PIN 5. [8]

Figure 2.30 prog[illegible]

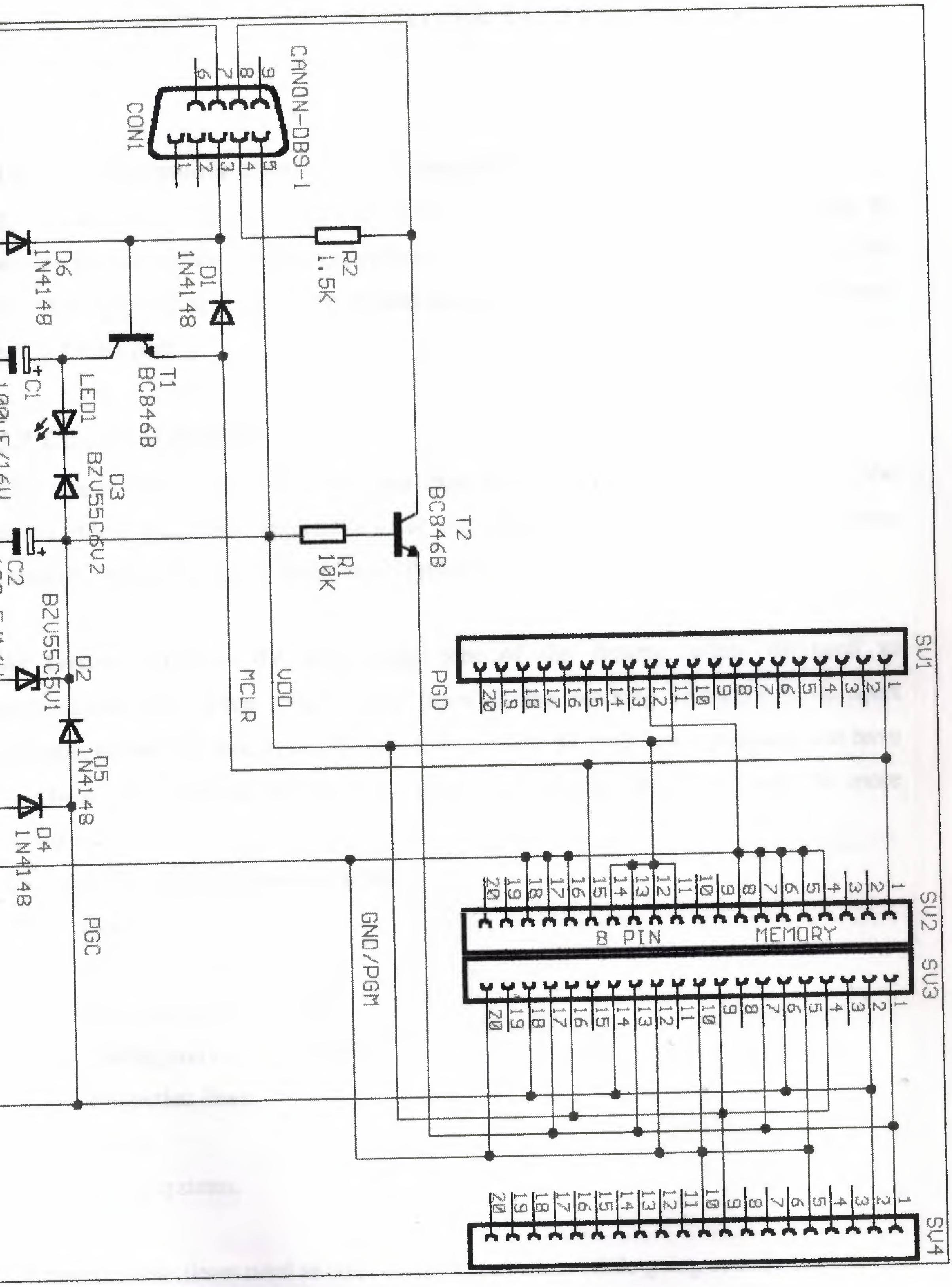


Figure 2.20 programmer schematic diagram.

CHAPTER THREE

REMOTE CONTROL OVER TELEPHONE LINE

.1 What is remote control over telephone line?

t is an electronic device using the DTMF signal of the ordinary telephone line, to eceive the commands from any telephone anywhere around the world, and switch ON nd OFF up to any five electronic devices connected to it, it is built up from a hardware nd software parts.

.2 Circuit applications

This circuit can receive the commends from any telephone keypad anywhere in the world, there are many applications for the circuit when need to control devices remotely; below are some examples of applications:

At factory: Due to the large sight size of the factory, when we need to activate/inactivate some things many times during working hours from an apart position, or/and we need to be able to activate/inactivate from many positions that have telephone line (internal or external), instead of wiring, this circuit will be more economic.

The controlled electric devices can be:

- Gates.
- Air compressors.
- Air conditioning systems.
- Heating and cooling systems.
- Production lines.
- Lights.
- Alarm systems.

At home: Some times need to control home appliances while going outside, usually use telephone to check if some body at home may help doing this job, but if there is no body at home or unable body, such a circuit may help.

u can control appliances or check them, while going outside, such as:

- Main gate.
- Lights.
- Water heater.
- Electric cooker.
- Ear condition.
- Alarm system
- Sprinklers.

3 Project development

ow, is explanation on how could accomplish this project, how the first idea came and ow could be developed and got better and better.

hile I am surfing internet pages of electronic devices shopping markets, the remote ontrol device took my attention, it was a device using the telephone line to control the ome appliances remotely.

Of course because the website was a shopping site, there was no schematic for that levice, there was only the control procedures and the applications of it, but after surfing he net more and more for the same kind of device, realized that there is a lot of such a levice offered for sale on internet and every one has a different usage procedures, number of controlled outputs and different features..

The search started for a ready schematic for such a device, and after a lot of search could find one seem ok, and the following is the explanation and the schematic that was presented in that website:

Remote control using telephone

Here is a teleremote circuit which enables switching 'on' and 'off' of appliances through telephone lines. It can be used to switch appliances from any distance, overcoming the limited range of infrared and radio remote controls.

The circuit described here can be used to switch up to nine appliances (corresponding to ie digits 1 through 9 of the telephone key-pad). The DTMF signals on telephone istrument are used as control signals. The digit '0' in DTMF mode is used to toggle etween the appliance mode and normal telephone operation mode. Thus the telephone an be used to switch on or switch off the appliances also while being used for normal onversation.

The circuit uses IC KT3170 (DTMF-to-BCD converter), 74154 (4-to-16-line demultplexer), and five CD4013 (D flip-flop) ICs. The working of the circuit is as follows. Once a call is established (after hearing ring-back tone), dial '0' in DTMF mode. IC1 lecodes this as '1010,' which is further demultiplexed by IC2 as output O10 (at pin 11) of IC2 (74154). The active low output of IC2, after inversion by an inverter gate of IC3 (CD4049), becomes logic 1. This is used to toggle flip-flop-1 (F/F-1) and relay RL1 is energised. Relay RL1 has two changeover contacts, RL1(a) and RL1(b). The energised RL1(a) contacts provide a 220-ohm loop across the telephone line while RL1(b) contacts inject a 10kHz tone on the line, which indicates to the caller that appliance mode has been selected. The 220-ohm loop on telephone line disconnects the ringer from the telephone line in the exchange. The line is now connected for appliance mode of operation.

If digit '0' is not dialed (in DTMF) after establishing the call, the ring continues and the telephone can be used for normal conversation. After selection of the appliance mode of operation, if digit '1' is dialed, it is decoded by IC1 and its output is '0001'. This BCD code is then demultiplexed by 4-to-16-line demultiplexer IC2 whose corresponding output, after inversion by a CD4049 inverter gate, goes to logic 1 state. This pulse toggles the corresponding flip-flop to alternate state. The flip-flop output is used to drive a relay (RL2) which can switch on or switch off the appliance connected through its contacts. By dialing other digits in a similar way, other appliances can also be switched 'on' or 'off.'

Once the switching operation is over, the 220-ohm loop resistance and 10kHz tone needs to be removed from the telephone line. To achieve this, digit '0' (in DTMF mode) is dialed again to toggle flip-flop-1 to de-energise relay RL1, which terminates the loop on line and the 10kHz tone is also disconnected. The telephone line is thus again set free receive normal calls.This circuit is to be connected in parallel to the telephone nstrument.[7]

schematic was attached to the website, look at figure 3.1.[7]

rom this point, the project development and problems facing story began.

.3.1 DTMF receiver

he DTMF receiver stage of the circuit has been built firstly to check whether it will vork nice or not before building the other stages of the circuit, unfortunately, the TMF receiver was very bad, weak in receiving the signals from the remote telephone nd there was repeatedly errors of detecting the right number pressed.

fter checking the built circuit many times along with the components but the problem till there, then realized that my work is ok and the problem may be in the receiver esign, so searched for the datasheet of the receiver chip KT3070, and after found it, the chematic in the datasheet was the same of the one that built, but found equivalent chips isted in the datasheet, so bought one of them it was HM9270D, and it was wondering hat it could work better.

ut still one problem, the detection of the signals was weak and some time a pressed umber not being detected, there was a thought that the signals reaching the circuit are veak, therefore replaced the 220 OHM resistor with a coil, because the coil is dropping he DC voltage as the resistor do but don't drop the AC signals as much as the resistor lo, and that replacement solved the problem.

.3.2 Hook off stage

n the circuit explanation, its said that the circuit will hook off when you press the utton star "*" is pressed, while I am experimenting the receiver circuit, neither the star utton neither any other buttons could be detected while the telephone is ringing, there vas no way to make such a reception to hook off and start the control process.

Therefore, its needed to make the circuit hook off by the counting of the ring tones as nswer machine do and then hook off.

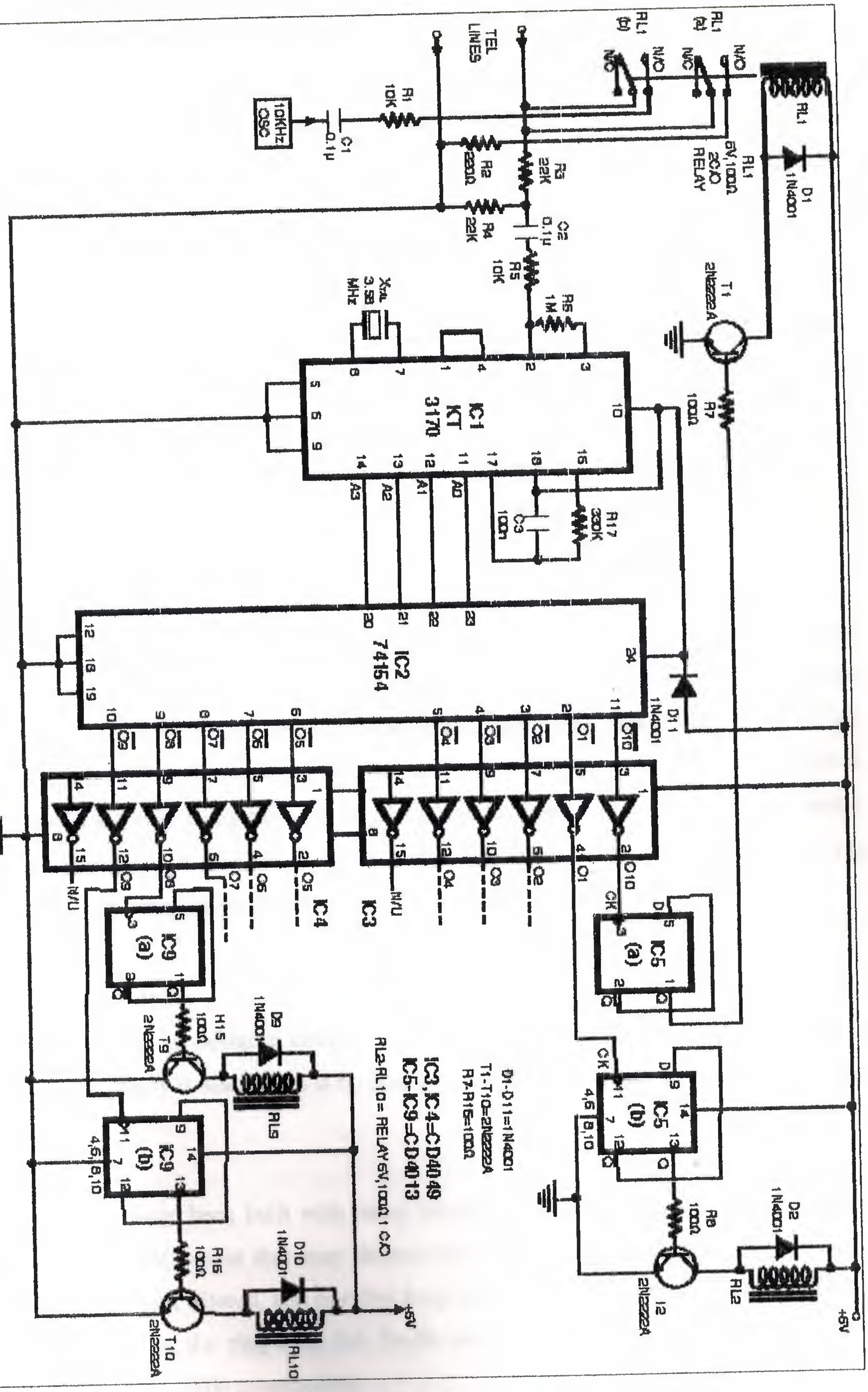


Figure 3.1 remote control over telephone line schematic.

.3 Controller stage

ong with the hook problem, the project features are not sufficient for me, it was ought to make this project more advanced, can utilize the skills of parallel port ogramming of the PC by interfacing the telephone line to the port to be able to count e ring tones and connect the DTMF receiver to do the advanced options.

it the PC attachment to the project will add more expenses to it and make it bigger in ze, the better idea is to use a microcontroller as a small and cheap PC, I hadn't any owledge about microcontrollers at that time, therefore I started to search internet for icrocontroller tutorials and could find many of them for PIC microcontrollers, I ownloaded three tutorials and start to read, and built a cheap programmer for it and ought the software CD and the PIC chip mentioned in the tutorial, and did my training n them.

tarted to plan what are the inputs and the outputs of the controller will be in the binary orm of course, as the decoded DTMF signal four inputs, detection of DTMF signal nput, ring tone input, user informing tone output and hook on/off output, so the lift umber of PINs on the chip was five to be the control outputs of the circuit, then started o program the chip using the assembly language defined for this type of chips, used a imulator software to run my program and check errors, after that the program was eady to be downloaded to the chip, and to build the other circuit stages.

.3.4 Ring stage

The mission was to design a circuit utilizing the 90V 20HZ ring tone to output zero voltage as a binary 0 when there is no ring tone and 5V as a binary 1 when the ring tone present.

many circuits have been built with many mistakes like a circuit not protected against exceeding the 5V output that may destroy the controller port, another one has a 5V output but not well filtered, and one that keep the present of its 5V output for long time after the absent of the ring tone, but finally could build a circuit with protection and filtered well with a little components.

3.5 Informing tone stage

e mission was to design a circuit that inject the square pulses tone from the controller the telephone line and block the DC current and the 90V the ring tone to pass the ntroller and destroy it.

erefore built an amplifier transistor circuit, but the problem of it was that the nnection of the telephone line polarity should be considered in order to block the DC ltage, but this problem was solved later and the connection way of the telephone line the circuit became not important.

.3.6 University telephones

fter the completion of the circuit it worked ok at the home, an experiment made at niversity and every thing worked but the circuit begin hook off it self without a ring ne, and the telephone line DC voltage was below the normal, and my advisor told me at this telephone is connected to a switching board.

seems that the telephone line of the university is disturbed and got a strong noise, that assing the ring counting circuit to the controller and being counted as a ring, therefore odified the program of the microcontroller to ignore any signal hasn't the ring tone uration, and again the circuit experimented there, and the result was the problem liminated.

.3.7 Turkcell network

After experimenting the circuit calling from the three telephone networks of Cyprus, the PSTN Telcom company, and the GSM Telsim and Turkcell companies, the first two vorked well, but using Turkcell telephone mobile couldn't make the circuit receive the DTMF signals well, you should press the number many times for the circuit to response pecially first three numbers 1, 2 and 3, another thing, the 500hz tone injected to the line by the circuit is not heard correctly.

These problems led to understand that the Turkcell network don't pass the frequency correctly specially the low frequency, that mean the bandwidth BW is not wide enough.

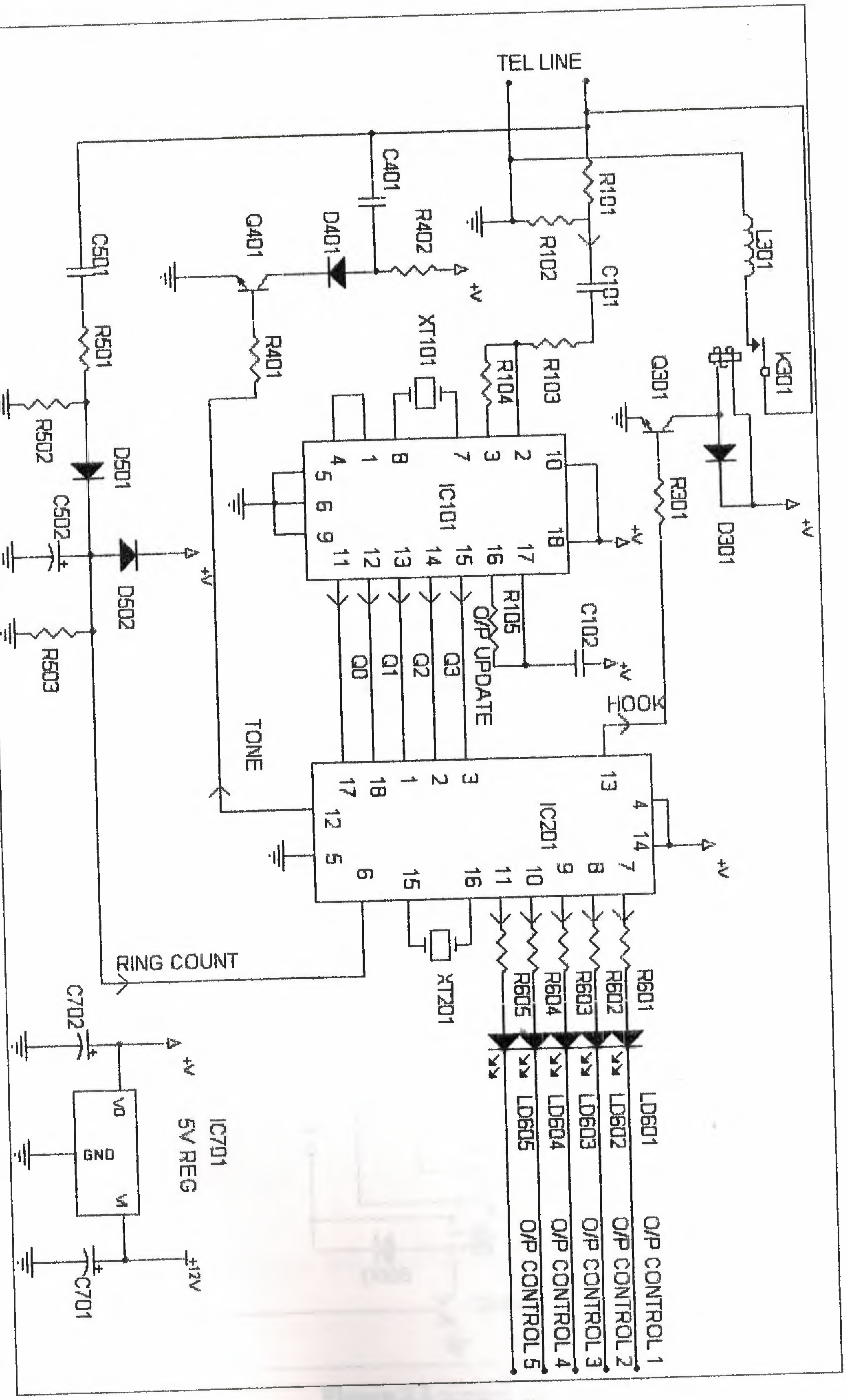


Figure 3.2 Main schematic.

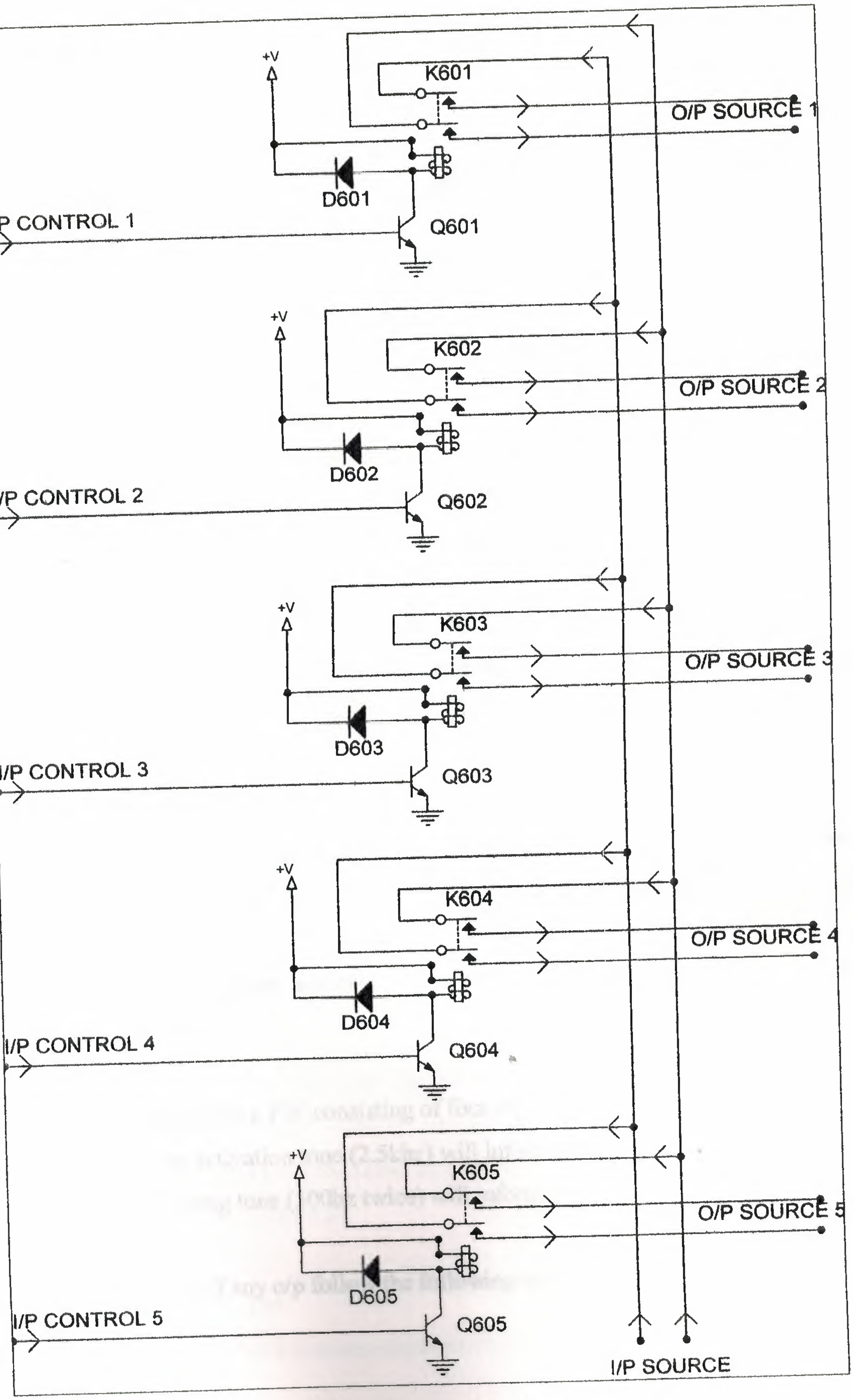


Figure 3.3 control schematic.

Circuit usage guide

1 Circuit installation

) A 12V DC adapter should be connected to the circuit with paying attention to the polarity.

2) Two wires from telephone line should be connected to the circuit, without considering the polarity.

.2 Control establishing

re are three situations for the circuit to start to operate:

1) Calling the destination telephone, and some body hook off.

2) Calling the destination telephone, and the circuit will hook off automatically after a certain numbers of rings.

3) Hook off the destination telephone, and dial the PW.

.3 Control procedures

ien you call the destination telephone and the circuit hook off, you will hear the ivation tone; other wise the circuit will not affect the telephone any way unless the rect PW is typed.

pressing the following numbers sequence on the key pad of the telephone the cuit will proceed:

First you should enter a PW consisting of four digits 0-9.

PW is right, the activation tone (2.5khz) will inform you.

not right, the wrong tone (500hz twice) will inform you.

To switch on or off any o/p follow the following steps to toggle the o/p:

for o/p 1

for o/p 2

for o/p 3

4 for o/p 4

5 for o/p 5

fter each step you will hear the activation tone after switching on, and the inactivation 500hz) tone after switching off or the wrong tone.

To check the situation of any o/p:

for o/p 1

for o/p 2

for o/p 3

for o/p 4

for o/p 5

After each step you will hear the activation tone for switched on o/p, and the nactivation tone for switched off o/p or the wrong tone.

* To change the PW:

)- four digits PW - same four digits PW

After that you will hear the activation tone if the two times typed PW is the same other wise will hear the wrong tone.

* To change the rings number:

9- a digit from 1 to 8

After that you will hear the activation tone or the wrong tone.

3.5 Schematic diagram

Remote control over telephone line circuit schematic drawn using ORCAD software without simulation for the design because the DTMF decoder and the PIC microcontrollers chips are not included, so its only used to draw the schematic.

Look at figure 3.2 and figure 3.3.

3.5.1 Components list

The circuit consist of the following electronic components, Look at figure 3.2 and figure 3.3 :

esistors:

101, R102 = 22K OHM

103, R104, R501, R503 = 100K OHM

105 = 220K OHM

301, R401, R402, R502 = 10K OHM

601, R602, R603, R604, R605 = 470 OHM

apacitors:

101, C102, C501 = 0.1uF

401 = 40nF

502 = 10uF

701, C702 = 47uF

iodes and LEDs:

301, D401, D501, D502, D601, D602, D603, D604, D605 = 1N4001

D601, LD602, LD603, LD604, LD605 = 1.5V

rystals:

XT101 = 3.579545 MHZ

XT201 = 4 MHZ

Transistors:

Q301, Q401, Q601, Q602, Q603, Q604, Q605 = 2N2222

Relays:

K301, K601, K602, K603, K604, K605 = 5V, 1A

Chips:

IC101 = HM9270D

IC201 = PIC16F84A

IC701 = 78M05

; Circuit stages

e following are explanations of each stage of the remote control over telephone line cuit schematic, the explanation referring to the schematic components and nnections, there are some data sheets in the index part of this book, you can look at em.

6.1 DTMF Receiver

ie DTMF receiver stag has the function of receiving the DTMF signals and stinguish them, decode them, output the suitable binary bits that represent the DTMF gnal detected, and output acknowledgement binary bit confirming the output update, is stage is denoted by the code 100, and consist of:

- IC 101 is the DTMF decoder chip, has a one input for the signals, 4 bits binary outputs to represent the DTMF detected signals in the binary form and 1 bit binary output goes one whenever a DTMF signal is detected and the 4 bits binary outputs are updated, PIN 4 is a reference voltage connected to the non-inverting input PIN 1 of the internal OP-AMP, and PIN 10 is the enable input of the 4 bits output, so its connected to the VCC to give binary one and keep the outputs enabled all the time.
- R101 and R102 are a voltage divider that drops the input voltage the half to protect the OP-AMP of IC101.
- C101 is to block the DC current and passing the AC signals.
- R103 is the input source resistor of the OP-AMP of IC101, and works with C101 as a high pass filter to block the ring tone (20hz) and other unwanted noise and pass the DTMF signals that are higher in frequencies.
- R104 is the feed back resistor of the OP-AMP.
- XT101 is a crystal connected to IC101 to generate 3.56Mhz frequency to the chip as a clock signal.
- R105 and C102 are used to control the update of the outputs.

5.2 Controller

e controller stag is the brain of the circuit, its functions are to receive the binary presentation of the DTMF signal and/or the ring tone detection and analyze them, ok on/off the telephone line, send feedback tone through the telephone line and tivate/inactivate the chosen outputs, this stage is denoted by the code 200, and consist :

- IC201 is the microcontroller chip, to process the stored written assembly program.
- XT201 is a crystal connected to the microcontroller to generate the clock signal of the chip.

.6.3 Hook off/on

Iook on/off stage has the function of interrupt the telephone line (close circuit) to start nd maintain the control process, and open the circuit of the telephone line (hook on) vhenever the process session is finished, this stage is denoted by the code 300, and onsist of:

- L301 is an inductor that is connected in parallel to the telephone line whenever the relay is switched on to drop the DC voltage of the line and don't affect the AC signals too much.
- K301 is a relay used to connect L301 to the telephone line when enough current passing its coil.
- Q301 is a skinning transistor for the relay coil.
- R301 is providing the proper switching current to Q301 from PIN 13 of IC201.
- D301 is used to protect the other stages from the opposite discharge current of the relay coil.

3.6.4 Feedback tone

Feedback tone stage has a function of injecting the tone signal generated by the controller stage and increase the impedance between the telephone line and the troller stage to protect the controller stage from the excessive current that may pass n the telephone line, this stage is denoted by the code 400, and consist of:

- Q401 is an inverting amplifier transistor used to isolate the controller from the telephone line.
- R401 is providing the proper input current from the controller stage to Q401.
- R402 is biasing resistor for Q401.
- D401 is used to prevent any current passing from the telephone line.
- C401 is used to block the DC current from the telephone line and pass the tone signal.

6.5 Ring detector

ng detector stage is suitable to detect the ring tone and output a binary one whenever e tone is represented to inform the controller stage, and its designed to protect the ntroller stage by weaken the coming ring tone to not output any excessive voltage, is stage is denoted by the code 500, and consist of:

- C501 is used to block the DC current of the telephone line and pass the ring tone signal.
- R501 and R502 are a voltage divider, used to drop the voltage the half to protect the circuit.
- D501, C502 and R503 are a half wave rectifier, used to supply the binary one voltage.
- D502 is used to protect the circuit from the excess voltage may pass, when the voltage on the anode is more than 5.7V, the diode will be forward biased and will pass the current to the power source 5V, and so maintain the voltage below 5.7V.

.6.6 Outputs control

utputs control stage is the interface of the outputs of the controller stage to the source estination, using LEDs to indicate the control status of each output, and has an elective iput source to be switched to the outputs destinations, this stage is denoted by the code 00, and consist of:

- LD601, LD602, LD603, LD604 and LD605 are the on/off LEDs indicators.
- R601, R602, R603, R604 and R605 are voltage divider for the LEDs, also, these resistors along with the LEDs deriving the proper currents from the controller stage to the switching transistors.
- Q601, Q602, Q603, Q604 and Q605 are the switching transistors used as skinning source for the relays coils.
- K601, K602 ,K603, K604 and K605 are the relays that switching the input source to its destination.
- D601, D602, D603, D604 and D605 are the protective diodes from the back current (discharge) of the relays coils.

3.6.7 Power Supply

power supply stage has the function of supplying a steady 5V DC to all stages in the circuit from any 12V DC source, this stages denoted by the code 700, and consist of:

- IC701 is a 5V regulator chip.
- C701 is the filtering capacitor for the 12V source.
- C702 is the filtering capacitor for the 5V source.

3.7 Control Program

Control program is the assembly program downloaded into the microcontroller, to do the jobs of analyzing the decoded DTMF inputs, update the outputs, command the hook circuit and generate the informing tone signal. If you want to look at the source code of the program look at the index of this book.

Figure 3.4 has the main flowchart of the control program, and the blocks that have bold border has also subprograms, look at their figures. This program will never reach the end, the end word of the program is just put as a programming requirement. If at any internal error happened any time, the program flow will be reset to just to the point after the LOAD PW&RING.

ure 3.5 illustrates the DTMF control flowchart, this subprogram keep checking for DTMF detection beside a timing counter for about 15 seconds to reset the program ase no DTMF detection occurred during this period, but if it detected, the program l wait for the detection to stop again to take the sample of the received number, but avoid the not ideal contact of the pressed button, the program will assure detection p for 0.1 second period to ignore the pulses may occur.

gure 3.6 illustrates the ring counter, this subprogram start counting when any ring e detected, will count each ring after it stop, and will ignore the noise by assure that detected signal will hold for 0.5 second, there is also a duration tolerance 5 seconds in between received rings if exceeded the program will reset.

gure 3.8 illustrates the activation tone generator, it is a 2.5khz square waves generated indicate the activation event.

gure 3.9 illustrates the inactivation tone generator, it's a 500hz square waves nerated to indicate the inactivation event., this signal is just a 1 and 0 pulses quence.

igure 3.7 illustrates the error tone generator, it is a two bursts of 500hz signal to dicate the error event.

igure 3.10 illustrates the subprogram that check the inserted password and match it vith the saved password, to give the result whether right or wrong.

Figure 3.11 illustrates the reception of the new password that will replace the old one, here is matching check of the first interred password and the confirmation interred password, if they are the same it will be saved in the EEPROM other wise it will not, with an informing tone of the result.

The following are some abbreviations clarification:

- **LOAD PW & RING**: the saved password and rings number will be loaded from the EEPROM into the RAM.

- ?: this mark mean a check of a condition belongs input, output or data and the result will be yes or no.
- **T:** a timer.
- **TIME OUT**: the specified time for a process is running out.
- **A.S**: a 2.5khz frequency square pulses, the activation signal injected to the line.
- **I.S**: a 500hz frequency square pulses, the inactivation signal injected to the line.
- **E.S**: a two bursts of 500hz frequency square pulses, the error signal injected to the line.
- **O/P ON?**: means, is the output has the binary 1?
- **SQ?, 1?, 2?, 3?, 4?, 5?, 6?, 7?, 8?, 9?**: mean, is this number present at the input of the decoded DTMF signals?
- **1 TO 8?**: means, is the DTMF number detected between 1 and 8?
- **TOGGLE O/P**: means, invert the binary state of the output.
- **NUM**: the DTMF number detected.

3 Project Features

e project remote control over telephone line circuit designed carefully to meet the manded features of the user and to minimize and expect the errors that may occur, e following are the summary of the feature of this project:

- **Compact and cheap**, it contains only the necessary number of electronic components, and all the components available in the market with cheap price.

- **Careless polarity**, the polarity connection of the telephone line to the circuit is not important, so it can be connected in either ways independent of the DC voltage polarity.

- **Protected**, the excessive DC or AC current that may pass from the telephone line will or the adapter not affect the circuit, because there is blocking and absorption of such currents.

- **Low power**, the circuit don't consume high current or voltage and may work on battery or DC adapter.

- **Sensitive**, well designed to drop the DC voltage of the telephone line when hook-off without too much dropping to the voice and DTMF signals, so detecting the pressed number and not affecting the conversation.

- **Interrupts**, any internal interrupt in the program may occur, will not cause the CPU to stuck but to reset and start from the beginning of the program.

- **Time-out**, while using the circuit, if the process of the control is not ended by the user, the program will reset after a certain time is passed, so if the user didn't hook-on after the control process, the circuit will reset after 15 seconds.

- **Buttons contact,** when the keypad buttons pressed by the user, the contact will not be ideal, therefore the project ignore the short negative pulses and will not operate unless a long period negative pulse occurs.

- **Ring count**, while the circuit counting the ring tones received, no way for the program to stuck if the ring counting didn't complete, the program wait 5 seconds for the next tone to come, other wise, the program will be reset, and if a noise presented, it will not be counted because the program confirming that the tone will hold for 0.5 second minimum.

- **Interface**, the circuit outputs can be interfaced from the binary form to any other form using the proper equipment, for example, when want to switch on/off a 220v device, a relay with a switching transistor can be used, if have a device that based on binary, no interface will be required.

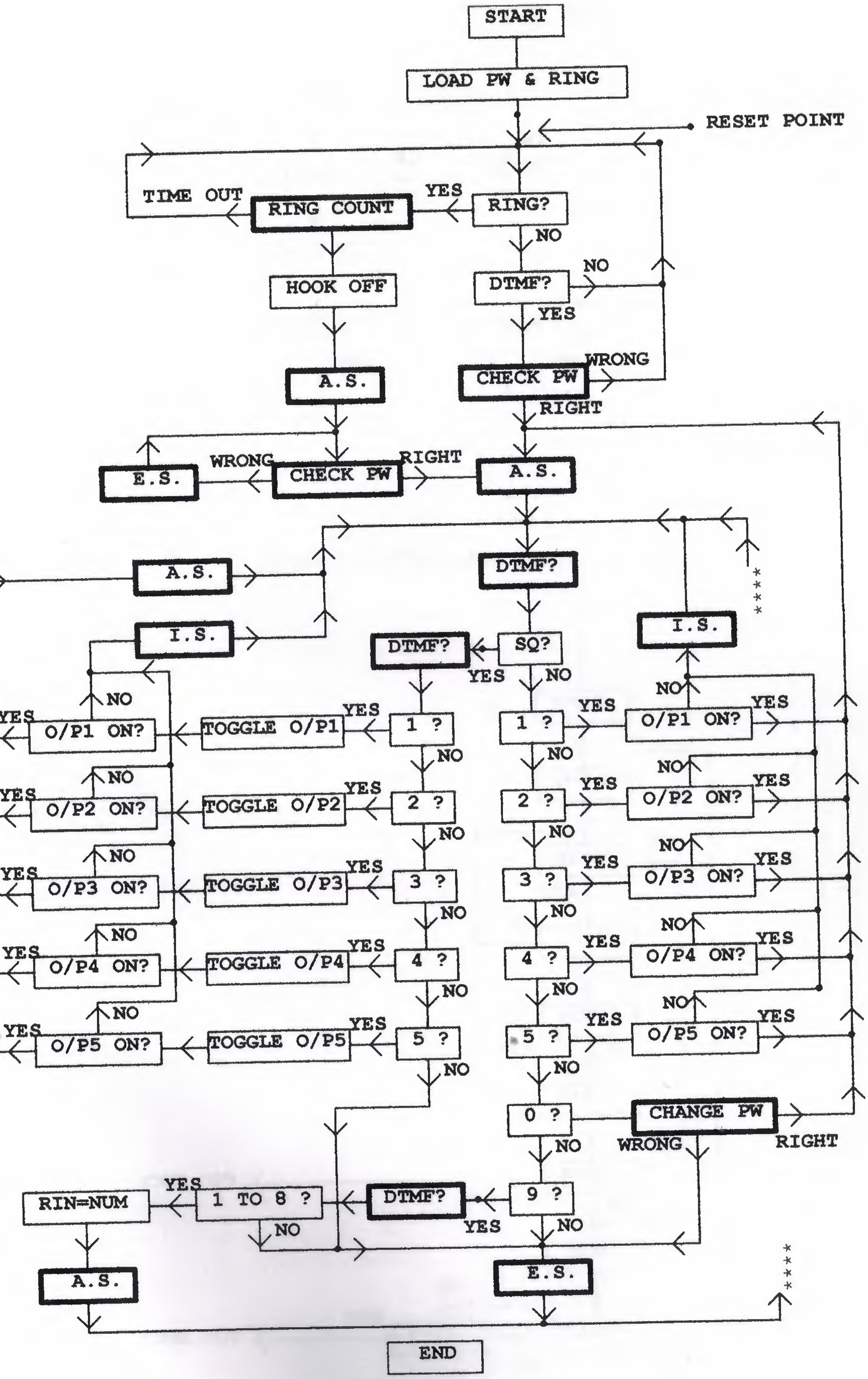


Figure 3.4 Main flowchart.

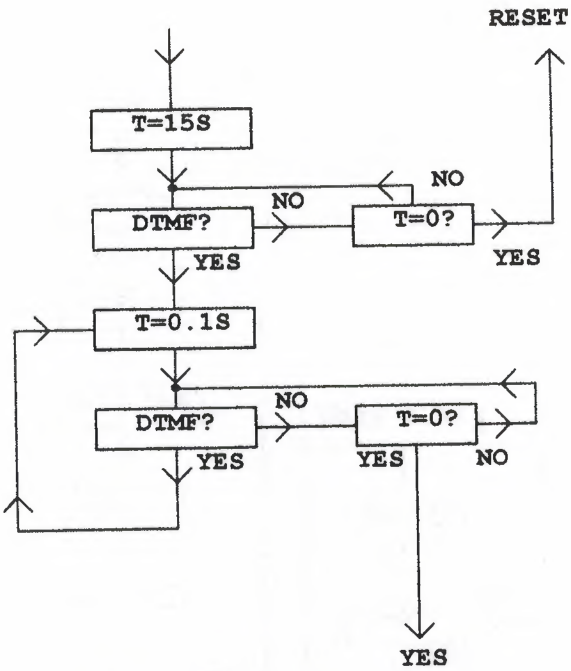


Figure 3.5 DTMF subprogram.

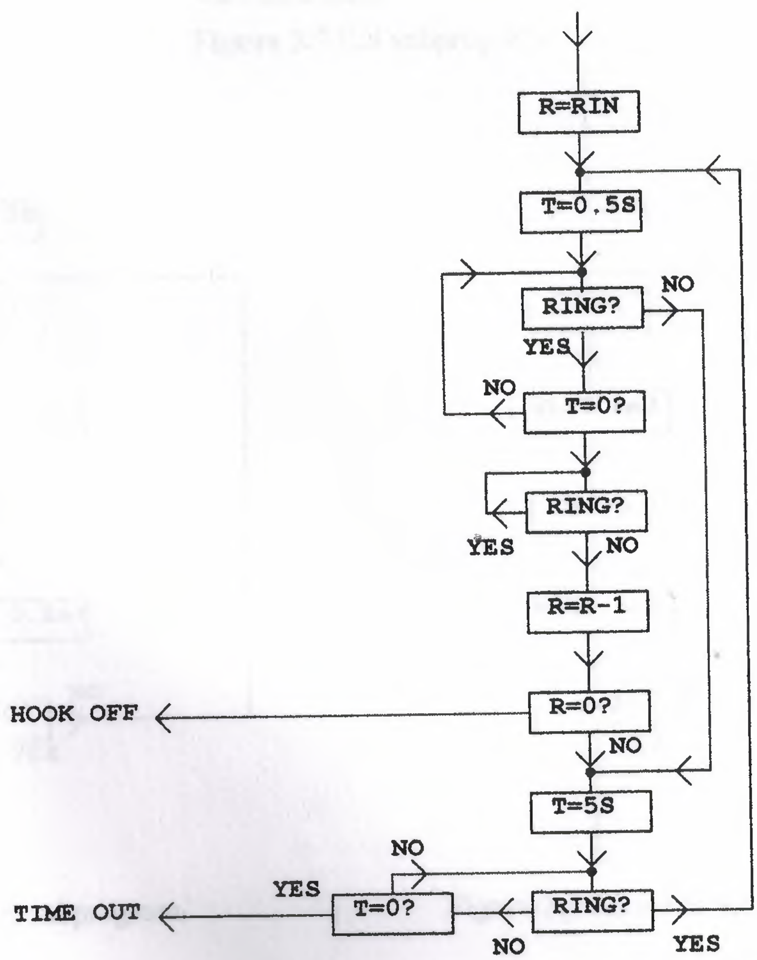


Figure 3.6 RING COUNT subprogram.

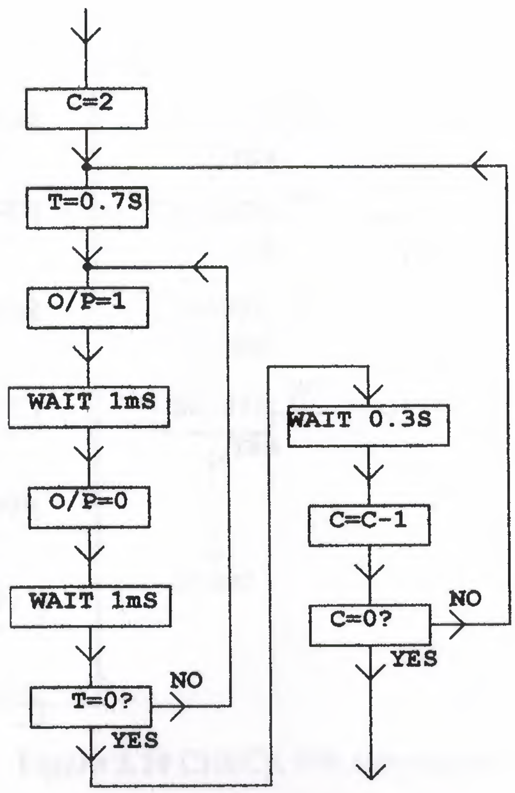


Figure 3.7 E.S subprogram.

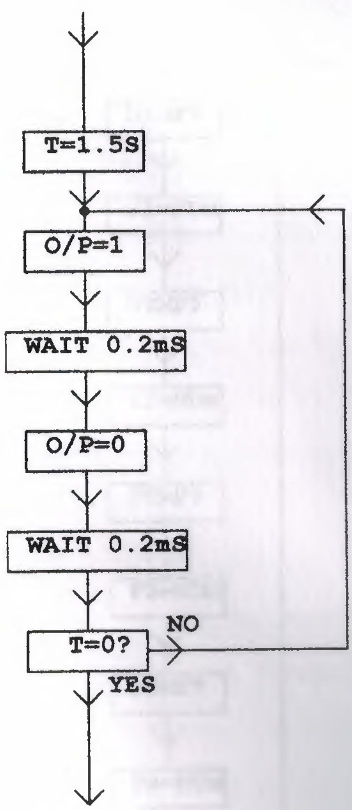


Figure 3.8 A.S subprogram.

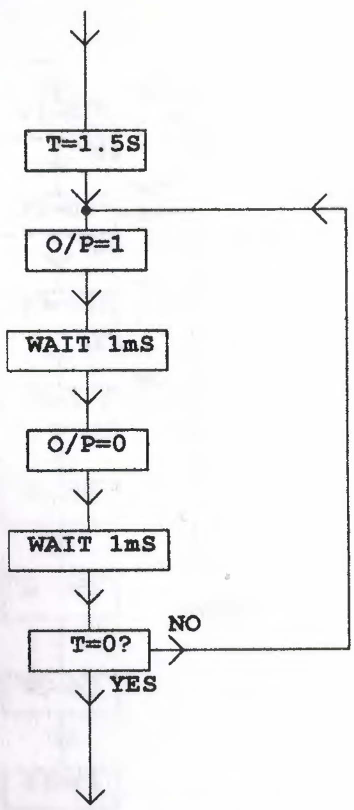


Figure 3.9 I.S subprogram

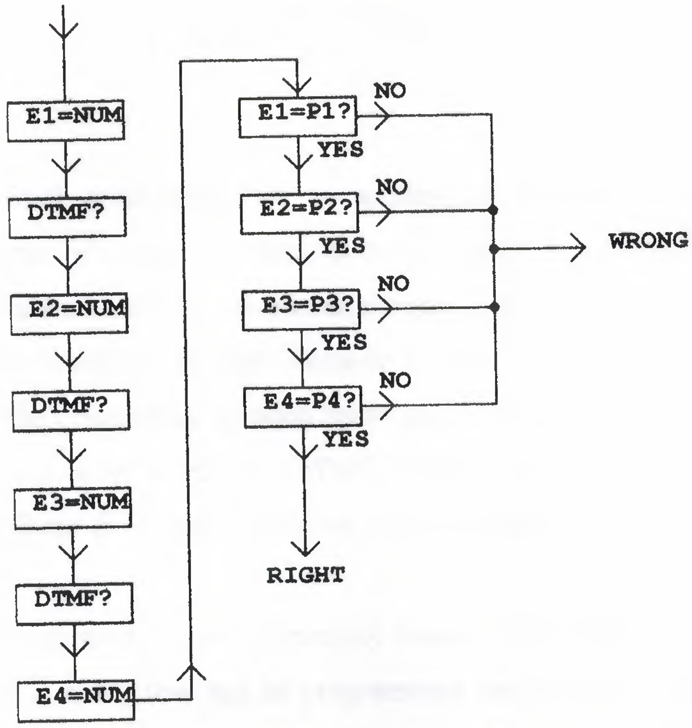


Figure 3.10 CHECK PW subprogram.

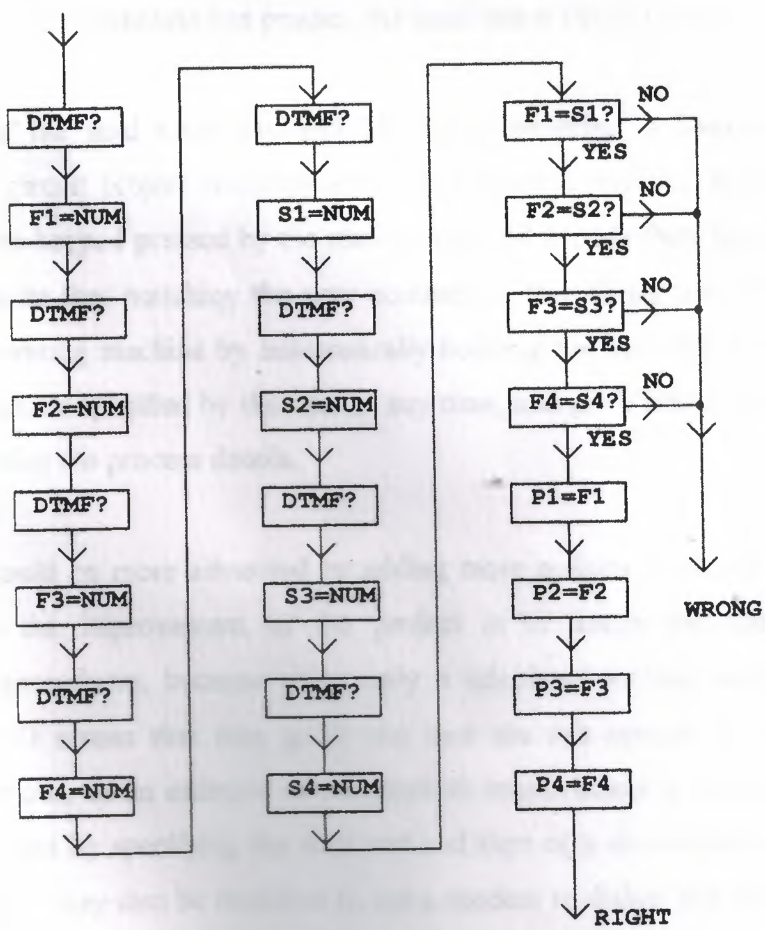


Figure 3.11 CHANGE PW subprogram.

CONCLUSION

'he remote control commands can be exchanged between systems using many echniques, in case of using the remote control over telephone line, the ordinary elephone line (public switching telephone network, PSTN) has been used to achieve his mission. This technique is used because of the availability of the telephones in oday's life and the connection between them along the streets, cities and nations, this nade the control process using the DTMF dialing signals a choice to build such a ystem and control our devices from anywhere in the world.

nother invention contributed to this project design is the microcontrollers, a small in ize and cheap in cost PC that can be programmed and installed in any circuit project, his invention adds more possible improvement to the project and reduced the electronic omponents needed. There are many kinds and models of microcontrollers but in emote control over telephone line project, the used one is PIC16F84A microcontroller.

s a result of the land telephone and PIC microcontroller, a remote control over elephone line circuit project could be achieved, based on receiving the DTMF signals f the telephone keypad pressed by the remote user and decode them then analyzing and utput the results that matching the user commands, this circuit uses a password and cts as an answering machine by automatically hooking the line after a certain number f rings that can be specified by the user at any time, and gives informing tone signal to he user indicating the process details.

'his project could be more advanced by adding more options to it, but the only thing hat opposite the improvement of the project is to insure the simplicity of the ommanding procedures, because using only a telephone keypad without additional uttons or LCD screen that may guide the user are not enough to represent more ontrolling options, as an example of the possible improvement is to add timers to the ontrolled outputs by specifying the start and end time of a chosen output. As a future vork, the project may also be modified to use a modem to dialup and proceed using the nternet connection or computers network.

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APPENDIX

Datasheet of PIC16F84A

.ECTRICAL CHARACTERISTICS

›solute Maximum Ratings

ıbient temperature under bias..-55°C to +125°C

›rage temperature .... -65°C to +150°C

›ltage on any pin with respect to VSS........ .... -0.3V to (VDD + 0.3V)

›ltage on VDD with respect to VSS .... -0.3 to +7.5V

›ltage on MCLR with respect to VSS(1) ....-0.3 to +14V

›ltage on RA4 with respect to VSS .... -0.3 to +8.5V

›tal power dissipation(2)..800 mW

aximum current out of VSS pin ....150 mA

aximum current into VDD pin ....100 mA

put clamp current, IIK (VI < 0 or VI > VDD)..................................... ± 20 mA

utput clamp current, IOK (VO < 0 or VO > VDD) .... ± 20 mA

laximum output current sunk by any I/O pin...25 mA

laximum output current sourced by any I/O pin ....25 mA

laximum current sunk by PORTA...80 mA

laximum current sourced by PORTA...50 mA

laximum current sunk by PORTB...150 mA

laximum current sourced by PORTB ....100 mA

lote 1: Voltage spikes below VSS at the MCLR pin, inducing currents greater than 80 ıA, may cause latch-up.

'hus, a series resistor of 50-100W should be used when applying a "low" level to the ICLR pin rather than

ulling this pin directly to VSS.

: Power dissipation is calculated as follows: Pdis = VDD x {IDD - ∘ IOH} + ∘ {(VDD-/OH) x IOH} + ∘(VOl x IOL).

Datasheet of HM9270D

n Description

n	Sym.	Function
1	IN+	Non-Inverting input
2	IN-	Inverting Input, Connections to the front-end differential amplifier.
3	GS	Gain select. Gives access to output of front-end differential amplifier for connection of feedback resistor.
4	VREF	Reference voltage output,nominally VDD/2. May be used to bias the inputs at midrail
5	INH	Inhibit (input) logic high inhibit the detection of 1633Hz internal built-in pull down resistor.
6	PWDN	Power down (input). Active high power down the device and inhibit the oscillator internal built-in pull down resistor.
7	OSC1	Clock Input
8	OSC2	Clock Output, Clock 3.579545 MHz crystal connected between these pins completes internal oscillator.
9	VSS	Negative power supply, normally connected to 0V.
10	TOE	3-state data output enable (input). Logic high enables the outputs Q1-Q4. Internal pull-up.
11	Q1	3-state data outputs. When enabled by TOE, provide the code corresponding to the last valid tone-pair received.
12	Q2	
13	Q3	
14	Q4	
5	StD	Delayed steering output. Presents a logic high when a received tone-pair has been registered and the output latch updated; returns to logic low when the voltage on St/GT falls below VTSt.
6	ESt	Early steering output. Presents a logic high immediately when the digital algorithm detects a recognizable tone-pair (signal condition). Any momentary loss of signal condition will cause ESt to return to a logic low.
7	St/GT	Steering input/guard time output (bi-directional). A voltage greater than VTSt detected at St causes the device to register the detected

		tone-pair and update the output latch. A voltage less than VTSt frees the device to accept a new tone-pair. The GT output acts to reset the external steering time-constant; its state is a function of ESt and the voltage on St
18	VDD	Positive power supply, +5Volts.

bsolute Maximum Ratings

Parameter	Min.	Max.	Units
Power Supply /oltage, VDD - VSS		6	V
Voltage on any pin	VSS - 0.3	VDD+ 0.3	V
Current at any pin		10	mA
perating temperature	-40	+85	oC
Storage temperature	-65	+150	oC
Package power dissipation		500	mW

) Program Source Code

OUNT0 EQU 0X0C		EEADR EQU 0X09
OUNT1 EQU 0X0D		EEPGD EQU 7
0 EQU 0X0E		RD EQU 0
C1 EQU 0X0F		WR EQU 1
		WREN EQU 2
EQU 2		GIE EQU 7
TATUS EQU 0x03		INTCON EQU 0X0B
P0 EQU 5		EECON2 EQU 0X89
RISA EQU 0x85		OPTION_REG EQU 0X81
RISB EQU 0x86		TMR0 EQU 0X01
ORTA EQU 0x05		TOIF EQU 2
ORTB EQU 0x06		TOIE EQU 5
EECON1 EQU 0X88		N1 EQU 0X10
EEDATA EQU 0X08		N2 EQU 0X11

EQU 0X12		ORG 0x00
EQU 0X13		GOTO HERE
)B EQU 0X14		
EQU 0X15		ORG 0X04
EQU 0X16		GOTO HERE
R EQU 0X17		
J1 EQU 0X18		HERE
J2 EQU 0X19		BSF STATUS,RP0
EQU 0X1A		MOVLW 0XFF
EQU 0X1B		MOVWF TRISA
EQU 0X1C		CLRF TRISB
EQU 0X1D		BSF TRISB,0
EQU 0X00		BCF OPTION_REG,5
EQU 0X01		BCF OPTION_REG,3
EQU 0X02		BSF OPTION_REG,0
EQU 0X03		BSF OPTION_REG,1
R EQU 0X04		BSF OPTION_REG,2
ONSTANT STAR=D'11'		BCF STATUS,RP0
ONSTANT ONE=D'1'		CLRF PORTB
ONSTANT TWO=D'2'		
ONSTANT THREE=D'3'		BCF STATUS,RP0
ONSTANT FOUR=D'4'		MOVLW D1
ONSTANT FIVE=D'5'		MOVWF EEADR
ONSTANT SIX=D'6'		BSF STATUS,RP0
ONSTANT SEVEN=D'7'		BSF EECON1,RD
ONSTANT EIGHT=D'8'		BCF STATUS,RP0
ONSTANT SQUARE=D'12'		MOVF EEDATA,W
ONSTANT ZERO=D'10'		MOVWF N1
ONSTANT NINE=D'9'		BCF STATUS,RP0
		MOVLW D2

OVWF EEADR		BTFSC PORTA,4
SF STATUS,RP0		GOTO DTM
SF EECON1,RD		BTFSS PORTB,0
CF STATUS,RP0		GOTO START
OVF EEDATA,W		GOTO RING
OVWF N2		
CF STATUS,RP0		DTM
OVLW D3		CALL PW_CHECK
IOVWF EEADR		XORLW 0XFB
SF STATUS,RP0		BTFSC STATUS,Z
SF EECON1,RD		GOTO CLOSE
CF STATUS,RP0		MOVF BOB,W
IOVF EEDATA,W		XORLW 0X0F
IOVWF N3		BTFSC STATUS,Z
CF STATUS,RP0		GOTO CALL_ES
IOVLW D4		GOTO START
IOVWF EEADR		
SF STATUS,RP0		CHECK
SF EECON1,RD		CALL DTMF
CF STATUS,RP0		XORLW 0XFB
IOVF EEDATA,W		BTFSC STATUS,Z
IOVWF N4		GOTO CLOSE
CF STATUS,RP0		MOVF PORTA,W
IOVLW DR		XORLW SQUARE
IOVWF EEADR		BTFSC STATUS,Z
SF STATUS,RP0		GOTO CHANGE
SF EECON1,RD		MOVF PORTA,W
CF STATUS,RP0		XORLW ZERO
MOVF EEDATA,W		BTFSC STATUS,Z
MOVWF NR		GOTO PW_SET
		MOVF PORTA,W
TART		XORLW ONE

TFSC STATUS,Z		BTFSC STATUS,Z
OTO S1		GOTO FIRST
IOVF PORTA,W		MOVF PORTA,W
ORLW TWO		XORLW TWO
TFSC STATUS,Z		BTFSC STATUS,Z
OTO S2		GOTO SECOND
IOVF PORTA,W		MOVF PORTA,W
ORLW THREE		XORLW THREE
TFSC STATUS,Z		BTFSC STATUS,Z
OTO S3		GOTO THIRD
IOVF PORTA,W		MOVF PORTA,W
ORLW FOUR		XORLW FOUR
TFSC STATUS,Z		BTFSC STATUS,Z
OTO S4		GOTO FOURTH
IOVF PORTA,W		MOVF PORTA,W
ORLW FIVE		XORLW FIVE
TFSC STATUS,Z		BTFSC STATUS,Z
OTO S5		GOTO FIFTH
IOVF PORTA,W		CALL RS
ORLW NINE		GOTO CHECK
TFSC STATUS,Z		
OTO RING_CHANGE		FIRST
ALL RS		BTFSC PORTB,1
OTO CHECK		GOTO FIRST0
		GOTO FIRST1
HANGE		FIRST0
ALL DTMF		BCF PORTB,1
ORLW 0XFB		CALL DS
TFSC STATUS,Z		GOTO CHECK
OTO CLOSE		FIRST1
IOVF PORTA,W		BSF PORTB,1
ORLW ONE		GOTO CALL_ES

		GOTO CHECK
ECOND		FOURTH1
TFSC PORTB,2		BSF PORTB,4
OTO SECOND0		GOTO CALL_ES
OTO SECOND1		
ECOND0		FIFTH
CF PORTB,2		BTFSC PORTB,5
ALL DS		GOTO FIFTH0
OTO CHECK		GOTO FIFTH1
ECOND1		FIFTH0
SF PORTB,2		BCF PORTB,5
OTO CALL_ES		CALL DS
		GOTO CHECK
HIRD		FIFTH1
TFSC PORTB,3		BSF PORTB,5
OTO THIRD0		GOTO CALL_ES
OTO THIRD1		
HIRD0		CLOSE
CF PORTB,3		BCF PORTB,7
ALL DS		GOTO START
OTO CHECK		
HIRD1		DTMF
SF PORTB,3		MOVLW D'231' ;15S
OTO CALL_ES		MOVWF CCR
		A7
OURTH		CLRF TMR0
TFSC PORTB,4		CLRF INTCON
OTO FOURTH0		DECFSZ CCR,F
OTO FOURTH1		GOTO T7
OURTH0		UI
CF PORTB,4		RETLW 0XFB
ALL DS		T7

FSC PORTA,4		MOVWF COUNT0
OTO DTMF_0		YES
FSS INTCON,TOIF		MOVLW D'100'
OTO T7		MOVWF COUNT1
OTO A7		AGAIN
MF_0		BSF PORTB,6
TFSC PORTA,4		CALL DELAY
OTO DTMF_0		BCF PORTB,6
		CALL DELAY
OVLW D'3'		DECFSZ COUNT1
OVWF CCR		GOTO AGAIN
7		DECFSZ COUNT0
RF TMR0		GOTO YES
RF INTCON		RETURN
ECFSZ CCR,F		
OTO T77		DELAY
OTO RE		MOVLW D'100'
7		MOVWF CC1
TFSC PORTA,4		RR
OTO ZZ		DECFSZ CC1
TFSS INTCON,TOIF		GOTO RR
OTO T77		RETURN
OTO A77		
E		DS
OVF PORTA,W		MOVLW D'16'
ORLW STAR		MOVWF COUNT0
TFSC STATUS,Z		Y
OTO UI		MOVLW D'100'
ETURN		MOVWF COUNT1
		AGA
S		BSF PORTB,6
OVLW D'32'		CALL DELAY

ALL DELAY		PP
CF PORTB,6		CALL DELAY
ALL DELAY		DECFSZ CU1
ALL DELAY		GOTO PP
ECFSZ COUNT1		DECFSZ CU2
OTO AGA		GOTO PO
ECFSZ COUNT0		MOVLW D'7'
OTO Y		MOVWF COUNT0
ETURN		YUU
		MOVLW D'100'
S		MOVWF COUNT1
IOVLW D'7'		AGAUU
IOVWF COUNT0		BSF PORTB,6
IOVLW D'21'		CALL DELAY
IOVWF CU2		CALL DELAY
U		BCF PORTB,6
IOVLW D'100'		CALL DELAY
IOVWF COUNT1		CALL DELAY
GAU		DECFSZ COUNT1
SF PORTB,6		GOTO AGAUU
ALL DELAY		DECFSZ COUNT0
ALL DELAY		GOTO YUU
CF PORTB,6		RETURN
ALL DELAY		
ALL DELAY		CALL_ES
ECFSZ COUNT1		CALL ES
OTO AGAU		GOTO CHECK
ECFSZ COUNT0		
OTO YU		CALL_RS
O		CALL RS
IOVLW D'100'		GOTO CHECK
IOVWF CU1		

S1		XORLW 0XFB
BTFSC PORTB,1		BTFSC STATUS,Z
GOTO CALL_ES		GOTO CLOSE
CALL DS		MOVF PORTA,W
GOTO CHECK		MOVWF P1
S2		CALL DTMF
BTFSC PORTB,2		XORLW 0XFB
GOTO CALL_ES		BTFSC STATUS,Z
CALL DS		GOTO CLOSE
GOTO CHECK		MOVF PORTA,W
		MOVWF P2
S3		
BTFSC PORTB,3		CALL DTMF
GOTO CALL_ES		XORLW 0XFB
CALL DS		BTFSC STATUS,Z
GOTO CHECK		GOTO CLOSE
		MOVF PORTA,W
S4		MOVWF P3
BTFSC PORTB,4		
GOTO CALL_ES		CALL DTMF
CALL DS		XORLW 0XFB
GOTO CHECK		BTFSC STATUS,Z
		GOTO CLOSE
S5		MOVF PORTA,W
BTFSC PORTB,5		MOVWF P4
GOTO CALL_ES		
CALL DS		CLRF BOB
GOTO CHECK		CALL DTMF
		XORLW 0XFB
PW_SET		BTFSC STATUS,Z
CALL DTMF		GOTO CLOSE

MOVF PORTA,W		MOVF BOB,W
XORWF P1,W		XORLW 0X0F
BTFSC STATUS,Z		BTFSS STATUS,Z
BSF BOB,0		GOTO CALL_RS
		MOVF P1,W
CALL DTMF		MOVWF N1
XORLW 0XFB		MOVF P2,W
BTFSC STATUS,Z		MOVWF N2
GOTO CLOSE		MOVF P3,W
MOVF PORTA,W		MOVWF N3
XORWF P2,W		MOVF P4,W
BTFSC STATUS,Z		MOVWF N4
BSF BOB,1		
		BCF STATUS,RP0
CALL DTMF		MOVLW D1
XORLW 0XFB		MOVWF EEADR
BTFSC STATUS,Z		MOVF N1,W
GOTO CLOSE		MOVWF EEDATA
MOVF PORTA,W		BSF STATUS,RP0
XORWF P3,W		BCF INTCON,GIE
BTFSC STATUS,Z		BSF EECON1,WREN
BSF BOB,2		MOVLW 0X55
		MOVWF EECON2
CALL DTMF		MOVLW 0XAA
XORLW 0XFB		MOVWF EECON2
BTFSC STATUS,Z		BSF EECON1,WR
GOTO CLOSE		BSF INTCON,GIE
MOVF PORTA,W		A1
XORWF P4,W		BTFSC EECON1,WR
BTFSC STATUS,Z		GOTO A1
BSF BOB,3		BCF EECON1,WREN

BCF STATUS,RP0		BSF INTCON,GIE
MOVLW D2		A3
MOVWF EEADR		BTFSC EECON1,WR
MOVF N2,W		GOTO A3
MOVWF EEDATA		BCF EECON1,WREN
BSF STATUS,RP0		
BCF INTCON,GIE		BCF STATUS,RP0
BSF EECON1,WREN		MOVLW D4
MOVLW 0X55		MOVWF EEADR
MOVWF EECON2		MOVF N4,W
MOVLW 0XAA		MOVWF EEDATA
MOVWF EECON2		BSF STATUS,RP0
BSF EECON1,WR		BCF INTCON,GIE
BSF INTCON,GIE		BSF EECON1,WREN
A2		MOVLW 0X55
BTFSC EECON1,WR		MOVWF EECON2
GOTO A2		MOVLW 0XAA
BCF EECON1,WREN		MOVWF EECON2
		BSF EECON1,WR
BCF STATUS,RP0		BSF INTCON,GIE
MOVLW D3		A4
MOVWF EEADR		BTFSC EECON1,WR
MOVF N3,W		GOTO A4
MOVWF EEDATA		BCF EECON1,WREN
BSF STATUS,RP0		
BCF INTCON,GIE		BCF STATUS,RP0
BSF EECON1,WREN		GOTO CALL_ES
MOVLW 0X55		
MOVWF EECON2		PW_CHECK
MOVLW 0XAA		CLRF BOB
MOVWF EECON2		CALL DTMF
BSF EECON1,WR		XORLW 0XFB

TFSC STATUS,Z		BTFSC STATUS,Z
OTO C_PW		BSF BOB,3
OVF PORTA,W		MOVLW ONE
OVWF P1		XORWF P1,W
ORWF N1,W		BTFSS STATUS,Z
TFSC STATUS,Z		GOTO NOO
SF BOB,0		MOVLW NINE
ALL DTMF		XORWF P2,W
ORLW 0XFB		BTFSS STATUS,Z
TFSC STATUS,Z		GOTO NOO
OTO C_PW		MOVLW SEVEN
OVF PORTA,W		XORWF P3,W
OVWF P2		BTFSS STATUS,Z
ORWF N2,W		GOTO NOO
TFSC STATUS,Z		MOVLW EIGHT
SF BOB,1		XORWF P4,W
ALL DTMF		BTFSS STATUS,Z
ORLW 0XFB		GOTO NOO
TFSC STATUS,Z		MOVLW 0X0F
OTO C_PW		MOVWF BOB
OVF PORTA,W		GOTO NOO
OVWF P3		C_PW
ORWF N3,W		RETLW 0XFB
TFSC STATUS,Z		NOO
SF BOB,2		RETURN
ALL DTMF		
ORLW 0XFB		RING
TFSC STATUS,Z		MOVF NR,W
OTO C_PW		MOVWF CR
OVF PORTA,W		R1
OVWF P4		CALL DELAY
ORWF N4,W		CALL DELAY

IOVLW D'8'		OP
IOVWF CCR		BSF PORTB,7
H		CALL DELAY
LRF TMR0		CALL DELAY
LRF INTCON		CALL ES
IH		FFF
TFSS PORTB,0		CALL PW_CHECK
OTO TC		XORLW 0XFB
TFSS INTCON,TOIF		BTFSC STATUS,Z
OTO MH		GOTO CLOSE
ECFSZ CCR		MOVF BOB,W
OTO KH		XORLW 0X0F
H		BTFSC STATUS,Z
TFSC PORTB,0		GOTO CALL_ES
OTO JH		CALL RS
ECFSZ CR		GOTO FFF
OTO TC		
OTO OP		RING_CHANGE
C		CALL DTMF
IOVLW D'83'		XORLW 0XFB
IOVWF CCR		BTFSC STATUS,Z
C1		GOTO CLOSE
LRF TMR0		MOVF PORTA,W
LRF INTCON		XORLW ONE
C2		BTFSC STATUS,Z
TFSC PORTB,0		GOTO R_OK
OTO R1		MOVF PORTA,W
TFSS INTCON,TOIF		XORLW TWO
OTO TC2		BTFSC STATUS,Z
ECFSZ CCR		GOTO R_OK
OTO TC1		MOVF PORTA,W
OTO CLOSE		XORLW THREE

BTFSC STATUS,Z	MOVWF EEDATA
GOTO R_OK	BSF STATUS,RP0
MOVF PORTA,W	BCF INTCON,GIE
XORLW FOUR	BSF EECON1,WREN
BTFSC STATUS,Z	MOVLW 0X55
GOTO R_OK	MOVWF EECON2
MOVF PORTA,W	MOVLW 0XAA
XORLW FIVE	MOVWF EECON2
BTFSC STATUS,Z	BSF EECON1,WR
GOTO R_OK	BSF INTCON,GIE
MOVF PORTA,W	AR
XORLW SIX	BTFSC EECON1,WR
BTFSC STATUS,Z	GOTO AR
GOTO R_OK	BCF EECON1,WREN
MOVF PORTA,W	BCF STATUS,RP0
XORLW SEVEN	GOTO CALL_ES
BTFSC STATUS,Z	
GOTO R_OK	END
MOVF PORTA,W	
XORLW EIGHT	
BTFSC STATUS,Z	
GOTO R_OK	
CALL RS	
GOTO CHECK	
R_OK	
MOVF PORTA,W	
MOVWF NR	
BCF STATUS,RP0	
MOVLW DR	
MOVWF EEADR	
MOVF NR,W	