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A RESEARCH PAPER
Integrated Voice/Data Through
A Digital PBX

by
Ian James Schofield

A Thesis

Submitted to the Faculty of the
Graduate School of Computer Science

at

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In partial fulfillment of the
Requirements for the
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DEDICATION

I would like to dedicate
this thesis to my
Mom & Dad
who made most of this possible.

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The digital voice/data PBX is finally reaching it's anticipated potential and becoming a major factor when considering the total communications picture for many businesses today. The digital PBX has always been the choice for voice communications but has lagged behind the LAN industry when it comes to data transfers. The pendulum has begun to swing with the enhanced data capabilities of third and fourth generation PBXs. The battle for the total communication market is quite fierce between the LAN and PBX vendors now.

This research thesis looks at the history, evolution, and architecture of voice/data PBXs. It traces development of PBXs through the present fourth generation architectures. From the first manual switches introduced in the late 1800's through the Strowger switch, step-by-step switching, stored program control, common control, digital switches, dual bus architectures, and finally what is anticipated in the future. A detailed description of of the new fourth generation dual bus architectures is presented. Lastly, speculations on the future direction PBX architectures will take is explored. A description of the mechanics of a possible Wave Division PBX is presented based on a fiber optic transport system.

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CHAPTER I

1.0. INTRODUCTION

The local area network (LAN) based on packet switching technology has received all the publicity in the battle for the data communications market since the beginning of the information generation in the mid seventies. This is quickly changing with the introduction of sophisticated, digital switching systems using telephone based technologies. The Private Automated Branch Exchange (referred to hereafter as just PBX) had received comparatively little attention to this point, but is now being viewed by many office communication managers as the probable answer to their data communications problems. Once some current problems are overcome, digital PBXs may become the ultimate solution to data sharing in both the local and wide area networks. This is becoming a reality with the recent introduction of new system architectures based on dual busses and integrated cabling systems. The digital PBX is the most workable and economical solution to the current problem of voice/data communications on a common network.

PBX's and LAN's have developed two very different genealogies. While computer networks tied a CPU to a group of peripheral devices, pc's to pc's, or any combinations of such devices, the telephone networks were implementing dialing facilities between telephones. Users for years have dreamed of having a single switching system that could handle transmission of both voice and data over a common integrated system.

There are numerous reasons why the digital PBX system should be considered the ideal candidate for the integrated voice/data system. First is the existence of an already in place configuration for a star based LAN. Every office in the digital PBX system is already wired to the switch over a two or four wire scheme for voice communications. It would only seem logical to use this same wiring scheme for data transmissions. Integration of voice and data across a common media would save the user the cost of having to install separate sets of wires for each independent voice or data network. A major cost in the installation of a packet switched LAN is the running of the coaxial cable. Secondly, the digital PBX's with their new architectures designed to incorporate the best features of circuit and packet switching are getting very close to the total solution of voice/data handling. The problem of data transmission speeds is finally being addressed with the dynamic allocation of bandwidth which was a limiting factor in earlier PBX systems. The Rolm CBX II has a dynamically configured 74 M bps bus. This can handle asynchronous and synchronous data at speeds up to 19.6 K bps and 64 K bps respectively. This bus will be increased to 294 M bps in the near future, with corresponding increases in data transmission speeds [DATA85]. Northern Telecom and GTE are also offering similar features on the Meridian and Omni S line of digital PBXs. Thirdly, the digital PBX had already established itself as having excellent administrative capabilities for the management function. Also there is an endless list of add on features for the user to select from that are of proven quality.

There are still some problems that must be solved, but the digital PBX has a solid foot in the door as the solution to voice/data integration. There is no reason that circuit and packet switching can not coexist in a single network, each performing the job best suited to it. The PBX designers have at least tried to address the complex problems associated with the unique characteristics of voice and data transmissions. The new hybrid circuit/packet switching PBX is their solution. The remaining problems are ones of technology. The fulfillment of the potential of the PBX for voice/data communications is looming on the near horizon. As the remaining problems are solved, the capability of the PBX will come one step closer to the reality of true voice/data integration. To further understand this we will trace the evolution of the PBX from voice only to voice/data.

CHAPTER II

HISTORY

2.0. EVOLUTION OF PBX SWITCHING TECHNOLOGY

If we look at the development of switching systems, LANs and PBXs, it is quite obvious that the PBX has a long and illustrious history. The telephone is a simplistic device when compared to a computer, but before the turn of the million telephones in a network. To provide quality service the telephone and PBX industries have also developed error tolerances in the range of 1 in 100,000 calls.

The computer industry has only been around for about thirty-five years, and in computer industry has only been around for about thirty-five years, and full networking a few devices, the telephone industry was completing international direct dialing to expand their network. The PBX designers have always been imaginative and futuristic with microprocessor controlling switching technology, software for a host of user options, and the recording of activities with database techniques. Is it not possible that the PBX might hold the key to full integration of voice/data communications. In the following we will explore the important contributions in the area of PBX switching technology.

2.1. FIRST GENERATION PBXs (1876-1973)

The first one hundred years laid the framework for the entire telephone and switching industry. Innovations, such as step-by-step and common control switching, modulation and multiplexing techniques, and stored program control, became the basis of the safe, reliable industry we have today. Transmission and networking of voice systems were implemented to the satisfaction of all classes of users during this time.

2.1.1. Manual Switching

In the beginning:

When the telephone was invented in 1876, no one, had any idea of the enormous influence it was going to exert on society in the coming century. Telephones were originally leased in sets of two, with a single dedicated line connecting the two specific endpoints. The subscriber was responsible for providing his own dedicated wire between these two points. The cost to rent was cheap, \$40/year for a business telephone, and \$20/year for individual use [JOEL84]. Other than a few large businesses, a telephone was typically a luxury only the very rich could afford. In many circles it was considered a toy of the very affluent.

It quickly became apparent as the number of subscribers increased that it would be nice to be able to communicate with more than one location at a time. In early 1878, the first telephone exchange was created in Hartford, Connecticut [CHOR84]. By the end of that year there were about a dozen exchanges in major cities along the eastern seaboard. We must remember that these were not public exchanges. They

were owned by an individual or group of businesses for their private use. They were usually set up for communication within a building, or possibly to connect to a remote branch office. In either case they did not cover any area more than a few city blocks in size. These were the first commercial uses of PBXs. A public exchange for residential use was not to appear until 1898 [JOEL84].

Each of these early exchanges was nothing more than a manual switching center. A single wire connected each telephone location through the central switch. In 1892 the introduction of a central power supply arrived on the scene [BHUS85]. At the same time two-wire cable appeared in an effort to reduce the noise induced by competing electric power and streetcar lines, which began to appear on the same poles that were once dedicated to telephone lines. Because the majority of telephone systems at that time were used by businesses, these early PBXs, although manual in nature, were the forefathers of the massive telephone switching systems used today to route thousands of calls [JOEL84]. Some manual switchboards are still in use today.

2.1.2. Step-By-Step Switching

Manual switching systems were made obsolete in 1891 when a Kansas mortician, Almon B. Strowger, patented the Strowger switch. This switching system eliminated the need for manual operator. It seems that Mr. Strowger was afraid the local town switch board operator was being paid to route potential business to his competitors. Removing the manual operator from this function seemed a logical way to resolve this. Thus, a device designed to address one man's ambition, turns out

to have molded the course of modern telephony [MART76].

Publicized as the girl-less switching system, the Strowger switch used a technique called step-by-step, direct, or progressive switching. The Strowger switch, electro-mechanical in nature became the workhorse of the budding telephone industry. In a step-by-step switching system, the call progresses one step at a time as the telephone user dials each successive digit of the number. This system is also called direct control because each switching function is directly responsive to each digit dialed in the phone number. Each digit dialed initiates a vertical move to a corresponding layer of new Strowger switches and then a rotation to find the first free one. This is done for all seven digits one at a time with the last set of switches either making the connection, or if the line is busy it relays a busy signal back to the calling party [MART76]. The Strowger switch was the basis of the "701" PBX, introduced in 1928 by the new division of the Bell Telephone System, Bell Labs [BELL79]. There are still hundreds of PBXs from the 700 series in use today on the North American continent.

Although, the step-by-step switch was simple, economical, and completely modular in nature, it had some major disadvantages. It would not be stretching the truth to say that compared to today's switches it was quite bulky, similar to the now extinct dinosaur. Because of its size it was very labor intensive and required a high degree of maintenance. There was also a high degree of switching delay involved, no economies of scale, and it was entirely unsuitable for text and data transfer because of its instability and noise problems. This impulse noise inhibited error correction; thus reducing the throughput in a

given amount of time. Also, there was no way to backtrack and look for alternative paths when all the available switches at a given level were not free. A busy signal could be sent back to the caller if the switching path was interrupted by reaching a dead-end at any point in the switching sequence. This could very well happen even if the destination of the call was not in use. Another major problem was the whole switching system equipment was tied up until a call was complete because of the nature of the switches at each level in the step-by-step method. Because of its step-by-step configuration, the system was also limited by the number of telephone numbers that could be assigned to a particular central office. Lastly, any logic necessary for analyzing a phone number prior to sending it to the switching system was either totally missing or lacking for all practical purposes [JOEL84], [MART76].

2.1.3. Common Control

The next important step forward in the development of switching technology occurred in Sweden with the introduction of the crossbar switch by the Ericsson Corp. in 1932 [BELL79]. This corporation is a European giant in the telecommunications industry similar to our own AT&T. The crossbar switch was much smaller than the Strowger switch. Because of its electromechanical nature it was also faster and less prone to impulse noise. Although, the first crossbar switches did use the step-by-step technique, it soon became apparent that this new switch was to be a milestone in the establishment of a new type of switching called common control.

Step-by-step switching could be compared to a person on a treasure hunt with the first digit of the telephone number representing the first clue in the search. If that clue is correctly deciphered, it is destroyed and another clue, the second digit, is received. This continues until the whole series of clues leads us to the treasure, or we are stopped at some phase and can not continue either forward or backward because a clue has not been solved. On the other hand common control gives us the whole set of clues and lets us use our intuitive senses to reach our destination. We are allowed to handle the clues in any logical manner we deem appropriate in order to find the treasure.

In 1933 Kempster B. Miller published a series of books, Telephone Theory and Practice [MILL33]. The majority of the three-volume set was devoted to common control switching technology, both automatic and manual. The automatic systems were some of the most sophisticated of their time including new circuitry known as "lockout circuits" similar to contention logic today. Automatic switches were beginning to get away from the old step-by-step method of switching. The new kid on the block, common control switching, began to take its place as the next leading innovation [MIL33], [JOEL84].

Common control uses logic circuitry to complete a phone call. Address digits generated by the caller first pass through a part of the switch called a marker which recognizes it as a user request for service and forwards the number dialed to an originating register for processing. This information is then sent to a translator. The translator converts the dialed number into switching instructions, some instructions are used by the marker to control the switching matrix and other

instructions are used by the sender to route digits to outside trunks if necessary, ~~for~~ central office processing. Common control allows the system to flexibly adapt itself to either an internal call or to establish a connection to the outside world. This is accomplished by storing the entire number being called before the connection processing begins [MART79].

Common control equipment made it possible to adapt flexible numbering schemes to meet user requirements. Also the addition of new features could be easily incorporated because it needed only to be applied to the circuitry. Common control equipment was also much easier to maintain and required less space. The 755A PBX, introduced in 1938, by AT&T was the first PBX to utilize a crossbar switch, and it also used a limited form of common control [JOEL82], [BELL79]. As circuitry design techniques progressed, the add on features of PBXs increased.

2.1.4. Stored Program Control

The explosion in computing technology was to pave the way for one of the most significant contributions to future PBX development. As a result of the ESSEX PROJECT by Bell Labs in 1958-1959, a new concept in switching control was to emerge [CHOR84]. Stored Program Control (SPC) enabled the switching matrix to be controlled by a coded program, resident on a computer. This allowed for very flexible design, even more so than common control. Changes and modification, new features, and number allocations could be easily changed in software without any major hardware revisions. The first commercial application of SPC happened in the public switching system with the introduction of the now

famous ESS switching series. The 101 ESS switching system was released in 1963 by Bell Labs, and was the first switch designed to use stored program control [BELL79].

2.1.5. First Generation Summary

The years after World War II were to become the golden era for switching development. Many of the electronic experts who had been focusing their attention on the war effort were now looking to peddle their skills in the commercial market. They became obsessed with the idea of increasing both the speed and power, while at the same time decreasing the size of the switching systems offered. The problem was that the majority of the new work being done was for the public exchange system, and very little of the new innovations were carrying over into the development of PBX systems at this time. Both manufacturers and users of PBX systems seemed content to sit back, happy with the tried and tested systems already developed and in use. They seemed to be waiting for some major new innovation in switching technology

Even though some remarkable accomplishments had been achieved in PBX design, everything from the first mechanical PBX in 1928 until the early 1970s was classified as first generation. These were all characterized by hard wired or electromechanical switches, space-division multiplexing, step-by-step or common control, and analog loops. In 1973, the beginning of the generation leaps was to begin in PBX design [MOOR85]. It took nearly a century for the first era in switching to come to an end. In the next fifteen years three new generations were to unfold.

2.2. SECOND GENERATION PBXs (1973 - still in use)

-

Second generation PBX's are characterized by a number of new features. They are computer controlled, programmable switches, using SPC. They have a variety of new features, but are still limited because of their blocking architectures. Although they have a digital switching matrix, they still have analog loops. The analog is only changed to digital once it gets to the switch. It is then converted back to analog when it leaves the switch for its destination.

2.2.1. Background

The PBX market was ready to take off. The personal computer revolution that was beginning to get underway was the nudge that was needed. As a result of the increase in computing power along with a proportional decrease in cost, the PBX was taking on a new look. Conversely, the physical size of the unit was also decreasing drastically. With the introduction of integrated circuits economy of size finally became a reality. It was now possible to place a 16-bit microprocessor on a desk top much like a typewriter. A PBX with SPC features that could handle a thousand lines could be set in a corner similar to an office filing cabinet [MART76].

The PBX designers also saw a new niche for their product. Data in digital form was becoming an important factor in the everyday functioning of a business. Also the volume of voice transmissions was reaching astronomical levels in the office. Would it be possible to incorporate both of these with increased throughput and still maintain the reliability of

the old analog switching technology?

There are two distinct methods that can be used for the transmission of information over common telephone wire, analog and digital. To understand what was to evolve in the next ten years we will take a quick look at how communications are handled by both these transmission methods.

2.2.2. *Analog, the Old Established King*

Sound, as we should all be aware, travels in a wave format. Distinct variations in the sound we hear (loudness, pitch, etc,) is created by changes in the frequency and amplitude of the wave pattern. Another characteristic of sound waves is their continuous nature. This was applied to PBXs by setting up a distinct physical path for each conversation going through the PBX. This was traditionally a two-wire or four-wire connection. Because each conversation was sent through a dedicated set of wire, the early first generation PBXs all used space division switching, (a dedicated wire for each conversation). This was later replaced by frequency-division multiplexing, which became the standard for analog transmissions. Alexander G. Bell's greatest achievement to telephony was the device that modulated sound waves, the transducer [ANGU84]. This device was to become the backbone of the telephone industry because it converted the amplitude and frequency of the sound wave to an analogous electrical signal. Thus, the term analog was associated with this particular technique, because electric current vibrations are a direct analog of the air pressure variations, increasing and decreasing in proportion.

A view of the electrical signal on an oscilloscope would reveal a continuous wave which would mimic the pressures of the original speech vibrations. The young human ear can recognize sounds in the range of 30 Hz (cycles per second) to approximately 20,000 Hz. The telephone company only transmits in the frequency range of 30 to 3400 Hz per channel, in the name of economy. The standard voice channel adopted by the majority of telephone equipment manufacturers uses a bandwidth of 4 K Hz. This range is more than enough for the voice on a telephone to be recognizable [RODE82].

Amplification is another aspect of analog transmission that is also very important. Amplification is continually necessary to reconstruct the wave in analog transmissions for the signal keep it's integrity. Amplification is the process of regenerating the shape of the original wave once it gets distorted and weak. A amplifier is used to amplify and/or reshape the analog signal. Unfortunately, the amplifier also amplifies the noise on the line because the amplifier is not able to differentiate between noise and the wanted signal. At some point this line noise ultimately submerges the signal in static and makes the true signal unrecognizable. This is one of the problems that must be resolved in any analog transmission system [GOEL83].

2.2.3. *Digital, the Start of Something Big*

If the telecommunications industry were to start up today, digital transmitting with time division multiplexing, not analog transmitting with frequency or space division multiplexing, would be the preferred choice. Unfortunately, the remains of the older analogs systems, the standard

for the first century of telephony, will be with us into the next decade. Because of this mix, there will be many problems that will need to be addressed for a smooth transition as the old analog systems are being replaced with newer digital systems.

The basis for digital transmission was born in the 1930's when the mathematician, Harry Nyquist, proved that it was not necessary to transmit a complete, continuous wave in order to transmit the total information contained in it [KARP80]. If the wave is sampled at a fast enough rate (at least twice the maximum frequency of the originating wavelength then the samples will contain enough information for a receiving device to recreate the signal without any significant loss of information. This is key to the eventual digitalization of sound through the PBX.

Nyquist's sampling technique as applied to analog transmissions was referred to as Pulse Amplitude Modulation (PAM). Once sound waves had been converted into PAM pulses by sampling its wave form, the next step was digital communications. In 1937 A. H. Reeves proposed a continuation of the process which lead to the measurement of the pulse itself, and the transmission of the measurement in binary [JOEL84]. Sound was thus reduced to a series of discrete 1's and 0's. This further refinement to binary was called linear Pulse Code Modulation (PCM). Other forms of PCM have been developed. These include log, differential, and delta. All are variations of wave form sampling [RODE82]. There are also predictive methods which produce synthetic voice using vocoders. A vocoder is a device that operates in the headset of the telephone. The vocoders operation is based upon a parametric description of speech

characteristics rather than actual waveform encoding techniques [KARP80]. -

When we refer to PCM we will assume that we mean linear unless otherwise stated. PCM is the technique most commonly employed when converting an analog signal to a digital one. The analog signal is first converted by a circuit sampling process into PAM pulses (discussed earlier). How often is it necessary to sample the signal in order for it to be reconstructed? The rule of thumb says that for an originating sound to be fully reconstructed it is necessary to sample the sound at twice the highest frequency needed to achieve that sound.

Standard telephone technology samples at a standard of 8000 PAM pulses per second. This sample is then encoded into 8-bit PCM. This means that the digitized voice signal is transmitted at a voice data rate (VDR) of 64 K bps. Seven of these bits are used to record the pulse and the eighth bit is used for error detection. Often literature refers to the data rate as only 56 k bps [MART76], [RONA86]. Seven data bits allows for the possibility of 128 different voice volume levels. Each of the data bits can be combined with other samples using time-division multiplexing for transmission. The higher the sampling rate, the better the quality of sound produced because of the greater number of different voice volume levels possible. Hi-fidelity music uses a 10 bit sampling technique. First the sound is sampled using PAM techniques. The PAM pulses are then changed into a unique set of equal amplitude pulses by second process similar to PCM, but having 512 possible voice volume levels. This results from 9 bits for data and 1 bit for error detection. This is all done by a device called a Codec. The Codec is usually located

in the headset of the telephone, and does the inverse of what a modem does to a digital signal [MART76].

One innovation leads to another. A new technique for multiplexing PCM signals was introduced, time division multiplexing (TDM) [RODE82]. TDM was realized during WW II for the encryption of voice for military applications [JOE84]. TDM techniques adopted quite nicely into the telephone system since the digital signals generated by PCM could be easily transmitted using TDM principles.

2.2.4. *Digital Advantages*

There are many advantages offered users by a digital voice communication system (DVC). Some of these advantages will be realized immediately while other will take time. They are:

(1) *Compatibility with digital networks:*

There is presently a long range plan by the common carriers to phase in digital transmission lines. Although, a long range advantage because of the high cost of switching from analog to digital, it will be beneficial in the future [ROSS83].

(2) *Less degradation:* Information transmitted digitally suffers less degradation for a variety of reasons. First digital signals are easily regenerated via the use of repeaters. Secondly error control is more simply applied to digital signals. As a result analog transmission impairments normally associated with telephone networks such as cross talk and echo can be eliminated. Digital transmissions can then be made distant independent [CHOR84].

(3) *Secure communications:* The military and certain businesses must transmit voice conversations in a secure manner. The most sophisticated analog scrambler does not afford the degree of protection that digital encryption does [KARP80].

(4) *Reduced Bandwidth:* By compressing voice into a digital format, the new digital signal can require a significantly reduced bandwidth for

transmission. It is not uncommon for ten or more digitized voice signals to be carried on a single analog channel of 4 Hz [MOOR85].

- (5) *Voice/data integration:* Once voice has been digitized it can be freely intermixed with digital data traffic. This flexibility relieves the network planner of separate facilities and management for two separate networks [ASEO83].
- (6) *Compatibility with computers:* Speech in digital form can be readily processed, transformed, and stored by computers. Since all PBXs at this time are computer based it would seem desirable to do this. This is one of the primary reasons for the increase in voice related services now being offered such as voice identification and voice mail [ASEO83].

If we want to take full advantages of digital transmission in a voice/data environment, we must rethink the present way PBXs work.

2.2.5. *Second Generation Equipment*

The second generation in PBX switching technology started when Northern Telecom introduced the first fully electronic PABX, a system that was referred to as the PULSE or SG-1. This system eliminated the electro-mechanical switching matrix entirely, processing all calls through solid state components. Northern Telecom's SG-1 system gave practical application to the theoretical works of Nyquist. The PULSE used PAM principles; therefore it was still considered an analog system. Because of the booming market for data transmission in the local plant, the first digital PBX was not long in appearing. In 1974 Digital Telephone Corporation, now a division of Harris Corporation, introduced the D-1200 digital PBX. In that same year American Telephone and Ericsson introduced the FOCUS and PRODIGY PBX systems [DATA85]. All of these used delta modulation. Within a short time other vendors quickly came out with competing PBXs. These being Harris Corporation with its D1200

series, Solid State Systems with the STS-16, and AT&T with its now famous Dimension Series [ANGU84].

These second generation PBX systems were all products that used computer intelligence to route calls, previously mentioned as SPC. They also restricted features by setting up user groups to control intentional or incidental abuses. Because of the growing use of SPC techniques, a variety of new PBX features were introduced. These included Least Cost Routing (LCR), designed to select the lowest cost route for long distance toll calling; Voice Store and Forward (VSF) for message collection and electronic mail; and Call Detail Review (CDR), which allowed managers to review all long distant calls made per extension. All of these add-ons were helping to improve the reputation of the PBX and the Telecommunications department within the business organization. Although these second generation PBXs were analog in nature, they were marketed as having limited data capabilities [GOEL83].

Two of the most popular PBXs marketed appeared in 1975. Northern Telecom introduced the SL-1, and Rolm (now a subsidiary of IBM), presented the CBX. Both of these use PCM for their analog to digital conversions and are fully digital in nature. Northern's SL-1 used a standard telephone industry encoding scheme based on eight bit data byte, 8000 samples per second, effectively giving 64 K bps bandwidth. Rolm used a 12 bit data byte with 12,000 samples per second that required a bandwidth of 144 K bps [ANGU84].

2.2.6. *Second Generation Unresolved Issues*

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What did the digital PBX really mean in the quest for true voice/data systems? Before the introduction of digital systems, the transmission of data through a PBX required modems. These devices converted from digital to analog and back again to digital at the destination. There were several disadvantages to this. At the top of the list was the restrictions on transmission speeds resulting from the limited bandwidth available. Next there was the

introduction of transmission errors resulting from constant digital-to-analog, analog-to digital conversions. While modem technology had made tremendous strides, it was also obvious that data transmission would clearly be better off if it was kept in its original digital form throughout the entire PBX system.

That was the promise, but it was far from reality. The actual development was far more complex. It was quite easy to talk about the integration of voice and data, another thing to implement it. The question of cost effectiveness was even further off in the picture. The second generation digital PBX systems encouraged the fact that they could handle data, but in reality they were still using an architecture designed to carry voice transmissions. There were three basic design issues that had to be addressed before any of the PBX system manufacturers could say a true voice/data digital PBX system had been developed.

- (1) *How will data get from the device connected to the PBX and back?* Although the second generation PBXs introduced in 1972-75 were internally digital, all external connections were still analog. All

information sent to the PBX was initially analog. Telephone connections still worked in their normal manner, and data was transmitted via a ~~modem~~ in analog form until it reached the PBX. After the signal reached the line card at the PBX, it was digitized. As it left the PBX it was changed back to analog for transmission back to the sending device for processing. If the digital PBX was to handle data and make a real contribution to data communications, it had to be digital throughout the local loop as well as in the PBX switching matrix. The architecture had to be designed for digital transmissions, not just adapted as a retrofit to an existing analog system architecture.

- (2) *How will the PBX handle different data speeds?* Each voice connection requires a fixed amount of bandwidth ranging from a low of 32,000 bps to 800,000 bps. Most systems use 64,000 bps so they are totally compatible with the telephone company standard for voice transmissions as discussed earlier. Data communications on the other hand could require from a low of 300 bps to a million bps depending on the kind of data being transmitted. A digital PBX had to be able to adapt to a variety of speeds.
- (3) *How will the PBX handle data's special traffic characteristics?* After more than a century of experience with voice telephony, the characteristics of voice traffic are well known to PBX designers. The average business telephone call is only three to five minutes in duration and the average number of calls handled per hour is two to three calls during peak traffic periods. That is why the PBX worked so well using dedicated circuit switches. Since it is extremely unlikely that all telephones on a system will be in use simultaneously, designers could conserve on the most expensive component of the switching system, the switching matrix. Mathematical techniques made it possible to predict the percentage of calls that would be blocked, denied service at any particular time, under any given load. Reasonable service was given based on these figures. Data connections are inherently different. A data connection could last from a part of a second to hours depending on the amount of data being transferred. The sharing of the switching matrix was designed around the dedicated nature of telephone connections. The algorithms used for this could usually accommodate data transfer, but were really wasting the resources of the switch. A data device connection could tie the resources of the PBX up in a fashion not addressed by the traditional PBX. The original second generation digital PBXs; therefore were not able to handle as many data device connections as they were voice. Also a large number of data device connections could have a negative impact on the way the PBX handled voice calls.

We must remember that digital PBXs are simply digital computers.

They have as a base a 16-bit or 32-bit microprocessor. Also the signals

they process are in binary like those processed by a computer. The introduction of digital systems opened up many new opportunities for voice and data communications over a single system, using an undifferentiated binary bit stream. Hopefully, data communications could be as transparent as voice communications, and multiple wiring schemes could be eliminated.

2.3. *THIRD GENERATION PBXs*

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Third generation PBXs use a non-blocking architecture and are capable of transmitting a digital signal from source to destination. Third generation offerings integrate voice/data by design, not as an add-on. They also offer a distributed architecture to down load the work.

2.3.1. *Almost There*

The three problems stated in section 2.2.6 had to be addressed. Each manufacturer adopted a different approach. The first efforts were nothing more than adding bandages to the already developed second generation digital PBX systems.

Northern telecom was the first to try to address some of the second generation short-comings by trying to improve their existing SL-1 series originally introduced in 1975. In 1979, they introduced the Add-On Data Module (ADM), which plugged into the side of the switching matrix with a companion line card. This allowed any digital data to enter the switching matrix directly without any conversion to analog at the sending device.

The digital signals from data devices could not be sent over the same wire pair used for analog voice signals. Therefore a new method of transferring data to the PBX was necessary. The SL-1 normally uses a three wire pair from the sending device to the switch; one set each for voice, power, and signaling. Since the signaling format is digital to start with, any digital data transmissions must also utilize this same wire pair for digital transmissions. There were three basically different

kinds of signals going into the central switch. An analog signal for voice, which was not digitized until the analog signal reached the switching matrix. The other two signals were for digital data and signaling in digital format originating from each data device. Things seemed to be getting more complicated, not easier.

Northern Telecom also assigned a separate time slot to each data connection coming in. Each time slot supported voice at 56 k bps, which was the maximum data transfer rate. By assigning one time slot to each data connection, Northern Telecom simplified system administration. This was because voice and data were still handled identically, thus there was no need to develop special software for data. This was also easy to do on site because of the modular expansion of the SL-1 time slots. By assigning one time slot to each data connection, the problem of traffic also became manageable. There are limits to the number of simultaneous voice and data connections possible, but this was only a problem with the size of the switching matrix. Over-kill became the norm. Buy a system considerably too big for your needs and there would be no problems with blocking. This was the approach that Northern's PBX vendors took with their customers [ANGU84].

At the same time Rolm Corporation, Northern's biggest competitor in the digital PBX market, adopted an entirely different approach to additions to existing systems. Rolm announced its Data Terminal Interface and Data Line Interface in 1980 for its CBX product line [KASS85]. The Data Terminal Interface was similar to the ADM, except that it required its own two wire connection to the PBX. This was in addition to the wires required by the telephone. Rolm's approach required up to five, two

wire pairs for each telephone/terminal combination. This Data Terminal Interface allowed devices to send data signals to the PBX.

The CBX's architecture forced Rolm to adopt a different method of allocating bandwidth within the CBX itself. The CBX used a single bus with a fixed capacity of 74 megabits/second with a maximum of 165 simultaneous connections. If one time slot were assigned to each data connection, a very small number of these data connections could overload the system. Rolm's engineers got around this by sub-multiplexing. Each time slot of 192,000 bit per second was subdivided, for example, into five 19.2 kilobit connections or forty 2.4 kilobit connections. This required additional software because voice and data are handled differently. This approach allowed the connection of quite a large number of devices with a minimal impact on traffic.

2.3.2. True Third Generation At Last

No sooner had Rolm and Northern Telecom had introduced their new systems, when an emerging company called InterCom ridiculed Northern's and Rolm's retrofit approach to the problems of the second generation. To be true third generation, the architecture was supposed to be designed for both voice and data, not an add-on to an existing voice system. InterCom claimed that their system represented the true definition of third generation. Intercom's third generation products were based on two new technological innovations. These were incorporated into the InterCom IBX.

The first was end-to-end digital communications. Instead of transmitting sound in analog form from the set to the switch, the voice was digitized at the headset. IBX was capable of carrying data at 128,000 bits/sec; 64K for voice, 56K for data, and 8K for signaling and overhead [ANGU84].

The second was a totally non-blocking architecture. The IBX bus could support up to 8,000 devices using every port on the system with no loss of service. By using an overkill method the voice traffic problem was solved. Data traffic was better, but the limiting factor was still speed, limited to 56K or less [DATA86].

The impact of the IBX and other systems such as the Mitel SX 2000, NEC 2400 IMS, GTE OMNI I, and the Stromberg Carlson Business Communications System are good examples of attempts to produce true non-blocking switches [DATA85]. The switches listed above forced the major competitors; AT&T, Northern Telecom, and Rolm to make some substantial improvements to their systems. Rolm announced that by 1985 it would have a full line of digital headsets and a non-blocking architecture by the end of 1985. This resulted in the CBX II Series. Northern Telecom similarly announced that improvements to its SL-1 series data handling capacity were forth coming. They introduced their Meridian Series, most recently the Meridian DV-1, to compete with the new dual bus technologies [DATA86].

2.4. *FOURTH GENERATION*

Fourth generation PBXs incorporate all the features of the third generation and incorporate two important new features. First, the fourth generation PBXs provide the solution to the problem of design using over-capacity for future expansion. The distributed architecture of fourth generation PBXs allows for the incremental growth of the network when necessary. If more capacity is needed, an organization can expand the existing configuration by adding new nodes, rather than trading in equipment for larger versions.

The second feature is the incorporation of a data LAN for high speed transmission of data between PCs, minis, and main frames when necessary. As will be discussed later there is a point where cost performance in handling data transmissions becomes the determining factor in choosing a PBX or LAN.

2.4.1. *The Race for a Fourth Generation PBX*

No one can rest on their laurels in the rapidly changing PBX market. No sooner had the third generation vendors released their latest systems than a new group of companies announced the coming of the fourth Generation. Several older manufacturers also announced new add-ons involving separate busses for both voice and data using packet switching.

In particular, the third generation switches had two critical design features that were not addressed [ZANN85]. First, voice and data were still processed by circuit switching techniques. Each connection was

assigned a fixed amount of bandwidth and a fixed time slot that remained in place for the duration of the connection. While more suitable for voice transmissions, circuit switching wastes bandwidth when used for data connections because of its bursty nature. Furthermore, data connections really need a wide range of speeds to accommodate different kinds of data transmissions. Packet switching was the more appropriate technique for this. The basic topology of third generation systems was still a star configuration with all functions relative to the system controlled by a central processor. What was really needed was a system that allowed for a fully distributed architecture. This would allow for full distributed switching, while intelligence and control would remain within each local node of the distributed PBX network.

Fourth generation PBX systems are attempts to integrate the best of two worlds. They would combine the low cost PBX-based terminal hookups with the high-speed LAN-based computer connections. This is the present state of the art with offerings of these systems introduced during 1985. Ztel's PNX is one of the best examples of Fourth Generation technology although there are others on the market [ZTEL85].

Traditional PBX architectures were monolithic in nature with a single processor and central control. This put limitations on the number of lines that could be connected to the system. To upgrade, a new system had to be purchased with a bigger switching dimensions. Over-kill in PBX size was the rule so that future upgrading could be possible. The building block style exhibited by the distributed architecture of fourth generation PBX's makes this kind of expansion possible for future growth without the purchase of a larger system. All of this is achieved

in a transparent manner to the user.

A number of manufacturers have introduced products that fall under the heading of fourth generation PBXs. The first to introduce such a product was InteCom in May, 1980. Intercom's system was called the Integrated Business Exchange (IBX). Intercom introduced the unique idea of integrating a ethernet-style local area network as part of a digital PBX system. This system was capable of allowing office data devices to communicate at speeds of up to 10 M bps in burst mode over standard twisted pair wire. In addition, the IBX voice/data line circuit card was the first circuit card able to distinguish between voice calls and data transmissions. The IBX had the ability to route different transmission types to different busses; either a circuit bus or high speed data bus.

The Allen Texas firm also offers a device called the LANmark local area interface. LANmark Ethernet allows any Ethernet compatible devices to transmit through the IBX. The aggregate data rate is 10 M bps, the same as Ethernet itself. A single device is limited to a transfer rate of 1 M bps. LANmark 3270 also allows IBM 3270 family terminal devices to transmit through the IBX, also. The IBX has the capability of connecting to various public networks through an X.25 pad. A fiber optic distribution system allows the interconnection of nodes up to a distance of forty thousand feet. [DATA85]

Two of the smaller firms, Ztel and CXC Corp. introduced the dual bus equivalent of an on site Integrated Services Digital Network in 1981. Both used end to end digital, had distributed architectures, and were totally non-blocking up to system limitations. The Ztel PNX uses a

baseband token ring architecture, compatible with IEEE 802 standards. Information is sent around the ring using token passing. Each ring operates at 10 M bps. One ring is for circuit switching, one for data, and one for backup. Any number of rings can be configured for each function. Up to 1500 voice telephones or data terminals can be attached to a PNX node. The PNX can gateway to IEEE 802 local area networks. The PNX also supports the 3270 family of IBM devices [ZTEL85].

The CXC Rose has a more complex networking structure. Both a broadband and baseband LAN system are employed. Circuit switching is done over a 33 M bps circuit switched 802 token ring. There is also a 10 M bps Ethernet baseband system used for signaling, internode communication interfaces, and packet data. The CXC Rose automatically allocates additional bandwidth for all calls. Voice calls are transmitted on the circuit ring at up to 512 K bps. Sony has recently bought into the corporation and has about a 25% interest [CXCC86].

GTE Communication systems was the next to develop a unique dual bus architecture. Their Omni S I - III, announced in 1982, was a series of switches that could handle up to 1,024 telephone and terminal devices. Each increment doubles the size of the previous model. With a PD-200 LinkUp, a packet transport system, 255 terminal devices, can be connected to a single host or public network. Each switching path in the system has access to more than 1 M bps of bandwidth at any time. [MIC285]

The approach taken by the four manufacturers mentioned above insured the development of an architecture from the ground up. Other manufacturers have elected to add packet switching on as an add-on to their

existing line of digital PBXs. This was the approach taken by Northern Telecom and AT&T Information Systems. Northern developed a packet network that could be directly incorporated into its Meridian SL-1 and SL-100 switches. The packet network could be installed with the circuit switch or remote to the main switch, connected to the main switch by a T1 carrier link. As a packet network, this system has a data rate of 40 M bps. Northern has also done something quite unique with this packet system. The system can also function in a stand alone mode. As a stand alone system, the packet network utilizes a dual bus architecture. Through the use of an analog link, the 40 M bps packet bus is divided into two 20 M bps busses, one for circuit and one for packet data. This system is called the Meridian DV-1.

AT&T has developed similar add-on technology for the System 75 and 85 digital PBXs. AT&T also developed separate star shaped, local area network called the Information Systems Network (ISN). Through an interface the System 75, 85 can connect to ISN for packet switching [BCRS86], [ATT285]. The approaches to the solution of incorporating packet switching features in the local architecture of the digital PBX will be expanded in Chapter Four.

2.5. THE FIFTH GENERATION USING OPTICAL TECHNOLOGY

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Fifth Generation PBX Systems are still on the drawing boards at research facilities throughout the world. The major difference between this new generation and the previous one is the use of opto-electronic devices. The Fifth Generation will be based on optical fiber, optical logic, and optical switches. Probably this new age of PBXs will not be introduced until the early or mid 1990s. Fiber optics technology has revolutionized the speed and bandwidth efficiencies of present day PBXs. The use of optical logic and optical switches will do the same for this new generation of PBX.

Bell and GTE Laboratories are the leaders in research in the area of opto-electronics in this country. Bell Labs in association with S. D. Smith of Heriot-Watt, University England, developed the first optical transistor called the transphaser [SMIT85]. Bell Labs most recent research has been in the development of high speed optical circuitry and large scale switching arrays. Felix P. Kapron leads a group at the IT&T Electro-Optical Products Division in Roanoke, Virginia, working on the development of large scale optical bistability switches. Hughes Aircraft Corp. has a working liquid crystal light valve that can be used as a small dimension matrix multiplication switch [KAPR85].

Ian MacDonald has been doing some of the recent leading research on opto-electronic switching using GaAs photoconductors for the Research Branch of the Department of Communications, Toronto, Canada. An interesting find of MacDonald's has been that the power consumption needed to drive a large optical matrix switch does not increase geometr-

ically as in semiconductors switches, but linearly [MACD85].

Japan is a nation that is heavily committed to research in the field of optical communications. NTT Corp. has several research facilities working in this area. K. Hahara and K. Kikucki of the Musashino Electronic Communication Lab are working intensively in the field of optical time division space switching using tree structured couplers [HARA85]. At a competing NTT lab, the Yokosuka Electronic Communication Lab, H. Tobia and K. Inoue are working on high speed (450M bps) frequency division multiplexing techniques using optics [TOBA85]. At the NEC Corp, the First Switching Division is working on advanced broadband switching systems, which is one of the areas vital to Fifth Generation switching development. Advanced optical fiber interfaces are being worked on at the state run Research Institute of Communications at Tohoku University, Japan. The Japanese are striving to be the leader in state of the art developments in the area of opto-electronic product design [NECC85].

From all this research in the area of opto-electronic, we are starting to see total optical models and designs being advanced for computing and switching systems for communications. One of the young pioneers in this area, B. Clymer of Stanford University, has written several recent papers on this topic [CLYM85]. S. A. Collins Jr. of the Ohio State University Electro-Science Lab is working on modeling theory for optical switching networks [COLL85]. E. Abraham and B. Seaton of Bell Labs are also presently very active in optical switching design [ABRA85]. Even the Russians are getting their feet wet in this field. V. M. Egorov of Autometriyo University in the USSR has published a recent article on the design of optical computing systems [EGOR85]. As more

and more of the research in this area is practically implemented, the chances of an all opto-electronic PBX will quickly become a reality.

2.6. THE TECHNOLOGY OF THE FUTURE

The last area where major research is being done has to do with what is called burst and wideband switching. Both are new and different fields that are being explored by GTE Research Corp. and AT&T Bell Labs respectively. Experimental in nature, at this point in time, they are not scheduled for any kind of release until the early or mid 1990s.

2.6.1. Burst Switching

Burst switching, sometimes known as fast packet switching, has been the work of E. Fletcher Haselton and Stanford R. Amstutz at GTE [AMST83]. Burst switching uses elements of both packet switching and circuit switching technology. Burst switching takes advantage of the bursty nature of both voice and data transmissions. All transmissions are handled over a common channel. Since voice has special timing considerations not necessarily true for data, voice is given priority over data. If necessary, (in most instances), data can be queued to some extent without timing loss. Both burst and packet switching use headers to indicate a transmission. But a burst message is not limited to a fixed length. It exists for the duration of the burst. Burst and circuit switching latch channels. However burst switching does not latch for the duration of the call, but only for the duration of the burst. An end of message byte (flag) indicates that the burst is finished and that the switching matrix can be reconfigured.

The variable burst concept matches PCM transmissions and switching within the PCM time frame. This amounts to a switching decision in

125 us, the maximum time frame. As a bit stream enters a link switch, the header is read to see what kind of message is being transmitted. The header signals the switch telling it how to reset for the particular kind of burst being sent. Voice, data, image, text, and control signals are possible burst modes.

The whole concept of burst switching is based on a very large, widely distributed network of small switches (16 ports) housed near nodes. These small switches (link switches) are tied together by a T1 carrier network. Advances in VLSI and VHSI will steadily reduce the cost of this kind of independent, decentralized switching network. Simultaneous switching of more than one transmission is possible. If burst switching reaches its potential as GTE hopes, it will revolutionize the way the Bell Operating Companies (BOCs) design their architectures for switching transmission [MOOR85].

2.6.2. Broadband Switching

The other non-conventional switching method, in development, is broadband switching. This is being done at AT&T Bell Labs under the leadership of Gottfried Luderer, head of Exploratory Switching Research [MOOR85]. Patents in this area have recently been granted to AT&T Bell Labs. By using simple protocol and routing strategies, broadband switching has been able to minimize high overhead and delay problems associated with packet switching. Both voice and data can be switched simultaneously. The reason advocates don't call this technique "fast packet switching" is because of the dynamic allocation of channels and bandwidth. A single channel can be dedicated between two points or any

combination of bandwidths, including different bandwidths for each direction in a transmission. A good deal of software is necessary for a broadband system to work effectively; therefore, centralized architecture seems more appropriate. This architecture is similar to that used in central office design presently by the BOCs. Another reason for centralization is the nature of the multiple switching matrices required for wideband to operate correctly [DAVE85].

2.6.3. Summary of Burst vs. Broadband

Wideband and burst switching seem to be on a collision course because they are going to be offered by competing common carriers at some time in the future. The architectures are inherently different with AT&T takes a centralized approach in wideband switching system design similar to the present telephone technology and GTE is basing their burst concept on a distributed network of switches located closer to the individual subscribers. Both have advantages and disadvantages. AT&T says its system provides for a more flexible allocation of bandwidth which will incorporate itself nicely with the new ISDN technologies. The new ISDN technologies are being introduced by BOCs as enhancements to the present telephone system. Burst switches, on the other hand, are easier to construct and are also less costly. They also provide higher throughput over twisted pair copper wire. Wideband switching is designed to work over optical fiber. Since a complete changeover to optical fiber by the BOCs at all levels in the wiring scheme does not look possible for many years, if ever, burst switching is the best bet to bring digital service to all subscribers. Never-the-less, there is still plenty of time to debate the issue since at the present time, there are

still more questions than answers for the immediate near future of both these systems.

CHAPTER III

HOW A DIGITAL PBX WORKS

3.0. INTRODUCTION

As we have previously discussed, the digital PBX is probably the most important company business systems. This is especially true when both data and voice transmissions must be supported.

In just ten years, we have seen a transition from the second to the fourth generation in digital PBX switching systems. This rapid evolution has left many prospective buyers of telephone communication equipment confused by the numerous offerings and capabilities of increasingly more complex equipment.

The number of digital PBX products on the market today is substantial, with each one offering numerous features and services. In order to select the best digital PBX system, we need to know when a digital PBX should be considered, what a digital PBX is capable of doing, how a basic system operates, and what digital PBX systems are available. This chapter will address the first three issues.

3.1. WHEN SHOULD A DIGITAL PBX BE CONSIDERED FOR VOICE/DATA?

When is the digital PBX the correct solution to the voice/data communications problem? As in the solution to any problem, an analysis of the symptoms would seem the appropriate way to start. The decision to pursue the use of a digital PBX is no different. The answers to the

following six questions will help us to decide if a digital PBX is a candidate solution. Affirmative answers to at least five of these questions would suggest that a PBX is the solution.

What is the voice/data traffic ratio?

When considering voice/data integration, a 50/50 voice/data mix would be the ideal situation. Realistically, a 75/25 ratio in either direction is suitable for PBX consideration. Supporting a digital PBX with 2% data and 98% voice would probably be an inefficient use of the digital PBX, but might be acceptable if future projections show an increase in data traffic. On the other hand, if the system is 90% data and 10% voice, then a LAN or data switch would be more appropriate with a separate system for voice traffic [KAUF83].

How much of the data traffic is asynchronous/synchronous?

The basic theory behind any digital PBX system is shared resources. If transmissions are asynchronous and bursty, then a PBX is strongly suggested. On the other hand, when the data is synchronous and usage is constant, the concept of sharing is lost. For example if data entry people spend all day continually entering information into a data base, then dedicated lines should be used; otherwise the PBX would be burdened with the constant load.

Are the stations distributed or centralized?

When the stations are scattered, it is a lot easier to use existing twisted pair wire, (or run new twisted pair wire) in contrast to coaxial cable. If all the data entry terminals are in a few rooms, a PBX is inap-

appropriate. A conduit with dedicated transmission line would be a more suitable solution. If the organization has many users with voice and data capability at their desk tops, the analysis becomes more complicated. Are the facilities housed in a single building within a few city blocks, or within few square miles? What are the traffic volumes? Twisted pair telephone wire is literally run throughout every building today. Compared to coaxial cable twisted pair wire is relatively cheap, easy to move, and easy and cheap to add on to. The major problem is twisted pair wire is not well suited to all applications. Careful consideration must be given when selecting the transmission medium. Once installed, it is very hard to reverse the decision.

Is the station equipment moved around a lot?

If the answer is yes, consider a PBX. Twisted pair wire is many times cheaper to run if a new facility is connected to the switch. Usually all rooms are wired with twisted pair for phones. Coax, on the other hand, is very hard and expensive to move.

Are the data transmission rates high or low?

Maximum data rates in PBXs vary. Asynchronous is usually 19.6k bps and synchronous at 56 - 64 k bps. A problem can arise when large files need to be transferred through the digital PBX switch. The PBX could become bogged down hindering services to other stations. What if a two megabyte file has to be transferred, and the maximum transfer rate is only 19.6k bps? The amount of time necessary to transfer a large file could be prohibitive. This problem has been addressed by the newer switches on the market with the dynamic allocation of the PBX

bus up to 75 megabytes.

Are the stations point-to-point or multipoint?

Do not use a voice/data PBX if it is not necessary to switch traffic. If a traffic from devices A, B, and C always connect respectively to devices E, F, and G; switching is not necessary. A dedicated path is the solution. If the traffic from A, B, and C goes to several locations, then a PBX is a good solution.

The above questions provide a starting point in deciding if further research into a digital PBX is appropriate. Many articles on selecting the correct PBX installation have been written [LEVI85]. If it is decided to continue the study on the selection of a PBX, an outline similar to Figure 1. will be useful in conducting a needs analysis.

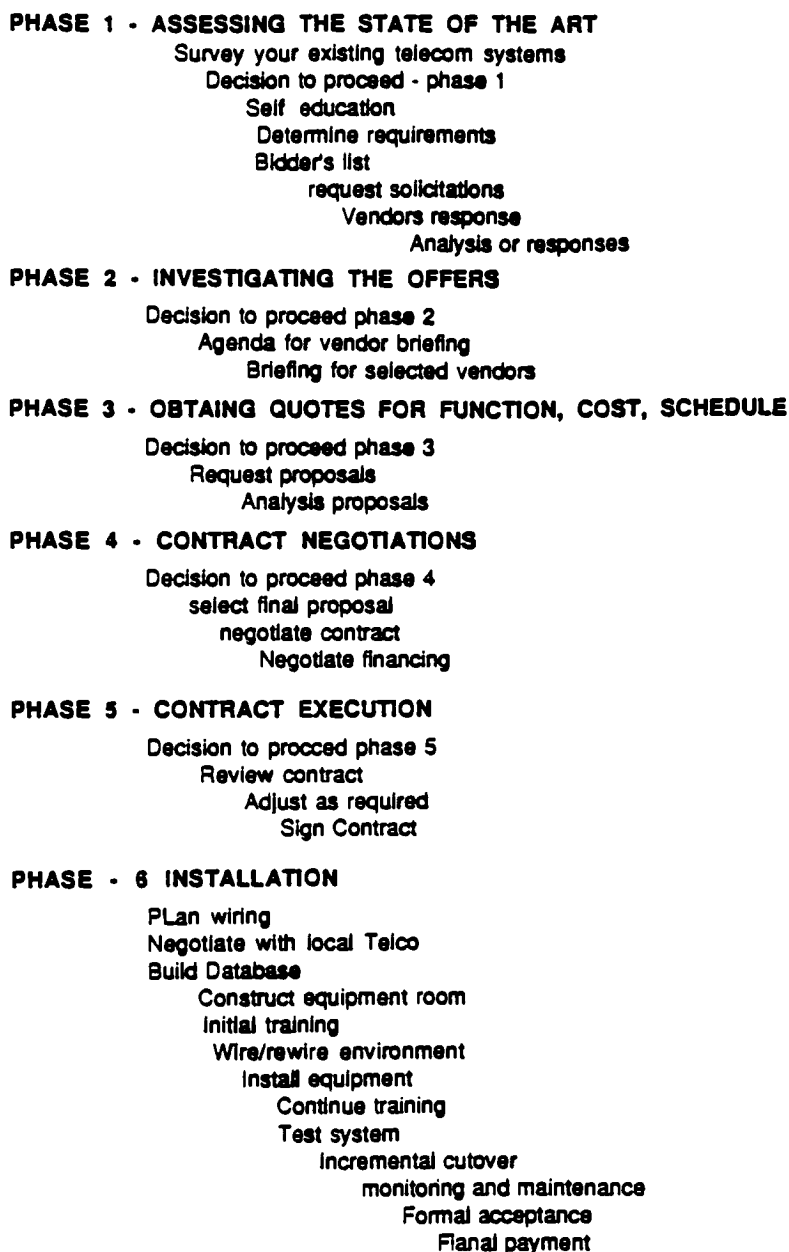


FIGURE 1.
SELECTING A PBX

3.2. WHAT IS AND WHO USES A DIGITAL PBX?

The following scenario could be taken from a typical day in an office environment that is using a digital PBX system:

Fred, the regional sales group coordinator arrives at the office. He keys into the electronic work station provided by his digital PBX vendor to check his electronic and voice mail. A voice message from Sally, the production manager, reminds him of their two o'clock meeting across town. Fred checks his electronic calendar to make sure of the time and place of the meeting. Fred remembers that he needs more information before the meeting, so he decides to call Sally. A few more key strokes gives him the company directory. Using the speed dial feature, he keys in the two digit code for her connection. Her line is busy, so he presses the camp-on button and hangs up. He decides to get a cup of coffee. Upon returning, the system indicates that Sally's line is free. He presses another key to signal the PBX so that if the line is still free, he is ready for the system to connect the call.

Fred expresses his concerns about a part of a recent manufacturing forecast report to Sally. There is some confusion, so he calls up the document on the video screen and adds share screen so that she can also see what is being viewed. They both key in some interactive changes to the document with each seeing the other's changes. When done, the changed document is added to both their files for later editing and printing by their respective secretaries. Sally suddenly receives a message from the system saying she has a call trying to get through. She explains to Fred that she is expecting an important call and puts

him on hold. Fred uses the hands free feature to call up another document for some rough editing while he waits.

Sally is talking to her controller when the topic switches to some sales forecasts. Sally suggests they include Fred, who is on hold waiting for her to return to a previous conversation. Within a few seconds, all three are discussing the subject using a conference feature. Still in the hands free feature, Fred retrieves a file from his desk and distributes the necessary information to the controller. The controller hangs up but before disconnecting Sally, invites Fred and his group to a new equipment demonstration. Fred calls up his calendar and puts in the necessary information. He also uses a group calendar feature to put a memo in the other sales persons' calendars about the demonstration. Sally and Fred both agree that their meeting this afternoon is no longer necessary because everything has been worked out during this conversation. They say good-bye and use the freed up time to work on other pressing matters.

Until recently a private telecommunication system was strictly a phone system. As we can see from the above scenario, it is now possible for both voice and data devices to be connected through the the digital PBX system. A digital PBX is a private system because it is owned by an organization and functions solely for that organization's communication needs. Private systems are also becoming increasingly popular as a shared resource, offered as a service in many new office complexes to the tenants. Whatever the case, a private system allows its users to communicate either internally or externally in a manner that is both time efficient and cost effective.

The primary function of a PBX is quite simple. It manages the job of connecting and disconnecting the various users in the private system to each other or to the public systems. When a number is dialed, the system must recognize the need to process a request for a connection, make sense out of the dialed pulses, and complete or deny the request through some sort of switching device in a timely manner. In today's computer controlled PBX systems, a logical form of common control switching is used with the microprocessor enabling these connections.

A private system has other features that are either not present, or transparent to the users of a public telecommunication system. Because of its smaller size, a PBX has more flexibility when internal changes to its characteristics and configuration are needed.

Who owns, controls, and uses a digital PBX system? The answer is just about every organization that uses telephones and digital data devices. An organization that use a digital PBX systems should also own and operate it. This gives the organization the greatest degree of freedom to utilize the PBX's many capabilities without outside interference. The question of ownership goes hand in hand with the degree of control one has over the system. If the telecommunication system has been designed so that the owner does not have to call the vendor every time he wants to make a change, then the owner has total control within the confines of the system architecture. Ownership and control are not the same concept. One obviously has more control over owned equipment. But there are still gray areas that must be avoided. The major benefit of ownership is that no one within established legal limits can dictate how the system will be used for internal operations. The owner can

replace or modify the PBX system within his budget constraints whenever he wishes and not have his hands tied in equipment selection if he plan the system correctly. Also, as the user of his own system, there may be financial and tax advantages available.

3.2.1. The System Configuration Of A Typical PBX

The system configuration of a private telecommunication system is only slightly different than that of a public Bell Operating Companies. It is composed of five distinct parts; station equipment, other optional and peripheral devices, transmission links, switching, and common control. The station equipment attached to the PBX consists of both voice and data devices. Voice equipment, usually telephones, can transmit using analog and digital transmission modes. Other devices that transmit in digital format such as data terminals, personal computers, printers, and video devices can also be attached. There is also peripheral equipment, and other optional features that can be incorporated into a PBX system. Examples of these features include voice mail, least cost routing, and call detail recording. Peripheral equipment would include such things as modem pools, recording units, and maintenance terminals.

Transmission links can be divided into two basic groups, those internal and those external to the PBX system. The switching system is the means by which the various connections are physically made. Common control in today's digital PBXs is always some form of SPC.

3.2.2. *What You See Is What You Get - The Cabinet*

The typical digital PBX system is no bigger than a rack stereo system and is designed to fit into a closet. An example is the ATTIS System 85 which is being installed here on the RIT campus. The physical specification are 32" D x 29" W x 69" H and it weighs 500 - 700 pounds [DATA85]. It would be fair to say that a total digital PBX system large enough to run a college campus or medium size corporation consisting of the switching unit, consoles, and other peripheral equipment could fit nicely into a 10'x10' room.

The hardware inside the cabinet is what concerns us. Most of the hardware of a PBX consists of printed circuit cards or boards. Each of these cards has a different function. Many duplicates may be needed for both repetition of function and redundancy purposes. For example, the cabinet may be more than half dedicated to circuit cards for station links. These boards are usually constructed to industrial standards. They appear to be stacked similar to books on a bookshelf. They are held in place by plug in connector sockets built into the shelves of the cabinet. Because they are individual cards - moving, exchanging, disconnecting, and replacing them without bringing the entire system down is possible due to their universal nature. This is only true for the newer PBX architectures that are designed for universal board swapping. Usually, there are many slots left in the cabinet for future expansion. The configuration can be easily expanded by just plugging in new boards and adjusting the common control software parameters.

Printed circuit boards are used for interface circuits (predominate use), TDM/SDM circuits, common control computer, fuse panels, and other functions related to the operation of the digital PBX. In order to provide continual service, PBX systems today are usually equipped with a backup power supply board.

3.2.3. Capacity of A Digital PBX

A question that always arises when seriously considering the purchase of a PBX is: where to start when comparing them? Capacity is probably the first criteria. There are three methods that can be used. The first measure of PBX capacity is how many stations, lines, and other options and peripherals can be connected to it. Each connection requires a line circuit card that has a built in number of ports, one for each station device connected through that line circuit card.

A line circuit card is a printed circuit board that allows the connection of various types of station equipment and trunks through ports to the PBX for switching. Line Circuit cards can be either analog or digital. At one time all line circuit cards in a digital PBX were analog. transmissions are devices had to send all their transmissions to the line circuit in analog form even though they were in a supposedly digital PBX. Data devices converted to analog via a modem at the source for transmission. The conversion to digital does not occur until the information reaches the line circuit card using logic circuitry resident on it. The reconverted digital transmission from the data device then progresses through the switch with the reverse process happening for the transmission to the destination data device.

Today's digital PBXs can accommodate both analog and data transmissions. The problem is some devices transmit in analog format and the rest is digital. The ideal situation would be for all devices, both voice and data to transmit in a digital format. The analog line circuit card would not be necessary if this was true. This would relieve the circuit card of having to handle both formats, and only require that the digital format be sent to the appropriate bus. The line circuit cards could then be responsible for separating the analog and digital signals to the appropriate busses. Unfortunately, many older station devices don't have the built in CODEC capability necessary to convert analog to digital at the source. The older PBXs also do not have the resident logic at the line circuit level for this conversion to happen. It is not cost effective to add a CODEC to each analog device in use, or change all line circuit cards in a system. Another reason for analog technology is that many parts of the BOC central offices still only process analog signals. The conversion to total digital systems will be corrected gradually with budget constraints the only stumbling block. Until all station equipment and systems are digital, PBXs will have to support both analog and digital station equipment. Fourth generation digital PBXs have adopted an architecture that will accommodate both transmissions modes.

Two possibilities exist for handling both analog and digital transmissions at the line circuit card level. Based on a particular system design, one technique is to group similar devices together and place them on a line circuit card that handles the format of the transmission. We could have, for example, one hundred, sixteen port line circuit cards

for single line analog telephones; ten, sixteen trunk port line circuit cards for outside connections to the public telephone exchange; and ten, sixteen port line circuit cards for data. There are also speciality line circuit cards for features such as CDR (Call Detail Recording), control consoles, and others for certain maintenance functions used by servicemen.

Usually, there is some trading that can be done at the line circuit card level. Assume a system has fifty single line, fifty trunk, and fifty data line circuit cards in its normal configuration. A user might decide that his particular requirements show a need for more data and single-line connections. We could trade off twenty-five of the trunk line circuit cards and redistribute them to the other two groups as we deemed necessary. Some systems have a limit that maximizes the number of line circuit cards that can be swapped within certain groups. Another restriction on line circuit cards could be console numbers. Many times when consoles are added to a system for monitoring purposes, a sacrifice must be made in the number of line circuit cards from other groups. These decisions are all determined at the time of sale and usually are subject to modification at later dates. The technique of using specific line circuit cards for specific types of station devices is common to third generation PBX systems.

The user is generally only concerned with the line, data, and trunk port capacity of the PBX system that they are buying or using. These system specifications are typically quoted to the user by brochures and company representatives. In fourth generation PBXs, the concept of universally interchangeable line circuit cards has been adopted. This

allows any station device to be attached to any of the line circuit cards in the system. The expense incurred in having universally adaptable line circuit cards is having additional logic built into the them. This additional logic keeps track of the type of signal, analog or digital, arriving from each of the station devices attached. If the signal is analog, the line circuit converts it to digital before the transmission continues.

The second measure of PBX capacity is the number of simultaneous conversations that the system can handle. This effect is referred to as the BLOCKING factor of a system. Blocking usually happens when the switch becomes jammed with calls either totally internal to the system or when all the trunk lines to the public network are being used. In effect, there are two distinctly different types of blocking occurring within a PBX system. The first type of blocking is associated with access to outside lines, trunks. No PBX system designed today has a trunk line number equal to the number of stations the PBX can handle. Therefore there will always be some blocking factor associated with trunking. The second type of blocking relates to internal connections within a PBX system if the station device is free. PBXs can be designed to totally eliminate this second form of blocking.

All systems use routing algorithms determined by user statistics to decide on the best strategy to eliminate these two forms of blocking. An organization can not tolerate a clogged telephone system for any duration of time. Therefore the issue of blocking becomes very critical when designing a PBX system.

When voice was the only transmission over a PBX system, standard predictive measures were developed that reasonably handled the blocking question. Blocking became a critical factor when data capabilities were added to the PBX system. Adding data transmissions to the switching function significantly alters the traffic patterns of a system. Data transmissions were of two types, short and bursty or long and interactive. Such transmissions are hard to predict. A change in the volume or balance of data transmissions in a blocking switch could require some hands on modifications to accommodate it. Resolving blocking issues could lead to enormous maintenance problems. The user interested in both voice/data over an integrated PBX had to consider the blocking issue quite carefully.

The basic design of the switch governs whether the PBX can support simultaneous voice and data without a decline in service levels. All the third and fourth generation switches that we discuss will have a nonblocking architecture (based on manufacture's literature). By definition, this means that the number of simultaneous calls that the PBX can handle should be $1/2$ of the total number of stations that the PBX can configure. This is obviously not true for many of the switches that will be discussed. A non-blocking switch allows all voice and data equipment attached to the PBX to be used simultaneously. This implies that the real time processor that handles the logical switching has been designed to handle maximum volumes of voice/data. Non-blocking should be an integral component of any PBX architecture. Many of the second generation PBXs that have add on data capabilities have what is called "virtual" non-blocking. Virtual non-blocking means there is virtually no

possibility that a call will be blocked, but there is a slim possibility it could happen. The AT&T System 85 is an example of a switch that refers to virtual nonblocking in its literature [DATA86].

The last general measure of PBX capacity is the ability to expand the system. There are two distinct methods by which a PBX system can be expanded. Expansion can be considered based on whether we are looking at a single-switch or multi-switch configuration. In a single-switch design, expansion is the degree to which we can add more ports, i.e. lines for station equipment and peripheral devices. All PBXs on the market today are designed with overkill in this area. A typical suggestion by vendors when purchasing a system is to think present capacity, then double or triple the present requirement for possible future expansion. The additional cost associated with buying an oversized unit is minimal when compared to the cost of upgrading to a new system at some future date. Expansion at the line circuit level is only possible within the confines of the system specifications. Every PBX has a maximum number of lines that it can accommodate.

The second kind of expansion deals with a multiple (distributed) switching configuration. The user now has the possibility for expansion at the switching unit (node) level. Expansion of this form occurs by adding more PBX switching nodes to the network. Every PBX designed today using a node architecture has either a star or ring configuration. Such systems are called distributed PBX systems. Expansion occurs at two levels. First we can add more line circuits to a particular node, or we can add additional nodes to the configuration. Every distributed system has a maximum number of nodes that can be added to it that

allows the architecture to be expanded.

3.3. STATION EQUIPMENT

Station equipment is the group of devices that either originates or terminates some form of user session through the switch. Telephones are probably the largest recognized group of station devices. Telephones can be standard or electronic, dumb or smart, rotary dial or touch tone, and analog or digital. At one time, the telephone was the sole device switched through a PBX. A host of other devices have joined the telephone, forcing the architects of switches to consider many new problems when designing PBXs. These new devices are not limited to only analog transmissions, but also include digital data, CATV, facsimile, and video. Some of the devices that fit into this category are electronic work stations, CAD/CAM systems, data terminals, printers, and answering machines. Examples of two operational systems are Plato used on many university and college campuses, and Videotex, offered by cable companies and used by many home viewers.

3.3.1. Station Interfaces

Contrary to popular belief, the various devices attached to the PBX switching matrix are not always compatible with one another and often not even with the switch itself. Because of this, all devices attached to the PBX go through some kind of Station Interface Device. These interface devices make sure that the incoming transmission from a station has the same format (speed, signaling, and synchronization) as the switching matrix. The reverse process takes place when the information is transmitted to the receiving station. The transmitting and receiving

stations do not have to be using the same equipment, just equipment that can be made compatible by the interfaces because the matrix acts as an intermediary. The interface unit will do the necessary adjustments to make sure the stations are able to communicate through their own specific formats and the format of the switch.

An example is appropriate at this time. What if we want to transmit an analog voice signal from a single station telephone over twisted pair wire to a digital PBX using time division multiplexing. The internal TDM bus of the PBX switch works in parallel. To accomplish this, the analog waveform of the voice transmission is first converted to parallel PCM using a A/D converter. The parallel PCM is then changed to serial PCM using a UART. The conversion from serial to parallel and visa versa is necessary for transmission over the twisted pair wire being used. The device that does this is a CODEC. These conversions can occur through a CODEC built into the headset of the phone or in a separate CODEC attached to the phone. The digital signal in serial form is then transmitted over the twisted pair to the PBX. When the digital signal in serial form reaches the PBX, an Interface Unit must convert the serial signal to parallel PCM and then it is multiplexed onto the TDM bus to the appropriate destination port. The process is then reversed to send the signal back to the receiving station. In the Meridian DV-1, a device called the LANLINK is used as the interface to connect station devices of other vendors in the proper format to the system [DATA86].

Today, there is a concerted effort by manufacturers to reach some common ground in the quest for compatibility. Standardization could reduce or eliminate the need for interfaces. Many users would like to

see standards evolve in all areas so that the interchangeability of products become a universal concept. The concept of Universal Standardization is being addressed by two major groups in the telecommunication and data processing area. On the national front, the Electronics Industrial Association (EIA), a private American enterprise is working toward this goal. The international body is the Consultative Committee for International Telephone and Telegraph (CCITT). Both these groups are doing their best to set some reasonable standards.

Buying equipment that does not conform to some set standards could at some point in time leave a user open to complications if the user ever decides to phase out one vendors equipment and go with another vendor. Sometimes companies tend to ignore the established standards and set their own. This is quite common with IBM, and the procedures they establish often become the "de facto" standards because of their size and impact on the market. Many times, when new technologies are introduced by competing companies who use techniques that are not compatible with one another, new standards must be set. It is often the case that the competing companies are allowed to fight it out in the open market and the victor's standard again becomes the "de facto" standard. Any standard is assigned a name by one of the standard organizations and other companies are allowed to use it on a license or permission basis. Until standards are set and adopted by all, there will always be a need for interface devices. An example of a standardized device would be the RS-232-C connector. The "RS" stands for "registered standard", the "232" for the standard reference number, and the "C" for the version number.

3.3.2. Telephones

When we mention telephones in relation to PBX systems, the average person is really not aware of the broad nature of this statement. Telephones can be single line, feature phones, or electronic. Each of these telephones types has different characteristics associated with them.

- (1) The single line telephone is probably the most well know of all units. This telephone is nothing more then a dump unit, with no programmable features. It is only capable of receiving and transmitting voice transmissions from one place to another. Single line phones can transmit in either analog or digital format. The model that everyone is most familiar with is the standard rotary dial unit. The rotary dial phones, based on DC pulse recognition, are gradually being replaced by the newer Dial Tone Multi Frequency (DTMF) Touch Phone. We usually associate the rotary dial units with the older analog technology and the DTMF units with the newer digital technology. Rotary dialing does not mean that a rotary telephone is limited to analog or a DTMF telephone is digital. The difference is the method of signaling used, dc pulses or frequencies. DTMF units are based on eight independent frequencies rather then DC pulses. DC pulse frequencies are combined in pairs to give 16 different, recognizable signals. The standard touch tone phone has a dial pad with the digits 0 - 9, and the "*" and "#" symbols. Each of the four rows and three columns on the dial pad represents a different frequency. The symbol touched is the sum of these frequencies. Since only twelve of the sixteen possible frequencies are being used there is the possibility of adding other functions to the unused frequencies on the dial pad. All single line telephones are capable of calling up features of a PBX by the use of access codes. Single line telephones can be bought at many locations. These range from your local variety store to the Bell Telephone Centers. Prices vary from \$19.95 to hundreds of dollars depending on how decorative the model is.
- (2) The newer smart telephones are becoming the norm with business users. Called electronic, this kind of telephone is capable of taking advantage of the full range of features of the PBX. The electronic phone does not have its intelligence built in, but has the ability to signal the PBX system to take advantage of the smarts available through the PBX processing unit. The telephone dial pad includes additional buttons that use digital signals for calling up the various PBX special functions. These could include speed dialing, transferring calls, conferencing, voice mail, holding calls, and other multiple functions. The electronic telephones do this by sending digital

signals, not pulses or frequencies, to the PBX processor. An example of stand alone electronic phones are the Meridian M2018, M2112 and M3000. An electronic phone can also be built into a work station that includes a data terminal. An example of this would be the Meridian 4000 Series Integrated Voice/Data Terminal. This workstation has a 12" bit mapped display, keyboard, and telephone headset and dial pad. The electronic telephone is capable of reminding the user if he has any voice messages, alerts if someone is on hold, and can show you the status of all telephones and terminals on the system [NORT85]. Because the electronic telephones have multiple functions, they all require more than the standard twisted pair wire used by a single line telephone. These additional wires are used to accommodate the additional power and signaling that must occur. Electronic phones and work stations are provided by the manufacturers of the various PBX switching system available and are usually not interchangeable with other vendors systems or equipment.

- (3) The third type of phone is the feature phone. The feature phone is a cross between the single line and electronic phone. The difference is the smarts of the feature phone are built into it, and feature phone does not need the PBX microprocessor to perform most functions. An example would be automatic redial. The feature phone also can call up the features of the PBX by using access codes transmitted by pulses or tones, not digital signals. The last difference, and probably the most important, is that the feature phone can universally hook to any private or public switching system since it's functions are not predicated on a particular switch's processor. The feature phone has its own RAM, ROM, and processor chip built into it. ROM features are programmed in by the manufacturer, while the RAM is used by the user to program his own particular features. Features that can typically be purchased resident on the feature phone are clocks, radios, calculators, number storing, autodial, speed dialing, and save and repeat the last number. The cordless phones being advertised on TV all are another example of feature phones. Prices vary again and they can be bought practically anywhere.

3.3.3. Data Terminals

Data terminals are another group of devices that can be attached to stations. Data terminals come in a variety of colors, shapes, and sizes. They are offered by a number of different manufacturers. A data terminal is connected through a line circuit card to the PBX switch. The data terminal now has the capability of connecting to other data terminals,

printers, graphic devices, and host computers using the facilities of the PBX.

We will treat PCs as data terminals for all practical purposes in this section. Data terminals have varying degrees of internal intelligence. A smart unit such as a IBM Pc has a much wider range of functions once connected than a dumb terminal such as a VT 100. Whether a dumb or intelligent terminal, all input data is entered through a keyboard. Output from a data terminal can be either hard copy (printed) or readout (display).

All transmissions to and from the PBX are serial, over twisted pair are in serial. Because the internal architectures of all data terminals transmit in parallel, a device that converts the internal parallel transmissions of the data terminal to serial is necessary before transmissions between the data device and the PBX can occur. An internally built UART accomplishes this conversion. An RS-232 cable connected to a RJ-11 jack acts as the interface from the data terminal to the twisted pair. Because of the character by character input made on the keyboard, asynchronous transmissions are usually standard. This means that each character is represented as a group of single bits, and transmitted independently rather than as a packet in a LAN. The ASCII (American Standard Code Information Interchange) is the most commonly used character set, although some IBM equipment still uses EBCDIC (Extended BCD Interchange Code). If necessary the PBXs can easily handle synchronous transmissions at much faster transmission speeds.

3.3.4. *Digital PBX Station Features*

Anything that a PBX system does beyond the connection of stations to stations or stations to the public network is considered a feature. All of the features in today's PBXs are software packages designed to work through the CPU of the PBX system. Every PBX system comes with some base features with the initial purchase. This initial package contains up to fifty features. Combining a group of features together is called bundling in the trade. No two vendors have the same base packages. Call transfer, call forwarding, conference calling, and hold are common to all basic packages, for example. Many optional features are bundled together to form option groups. Some features can also be bought individually such as Automatic Call Distribution (ACD). The user can purchase additional software packages at any time to upgrade the system. This gives the telecommunications manager the ability to incorporate features that have already been developed or any new ones introduced in the future that would be compatible with their present system.

There are as many packages of feature options as there are vendors. Just because a switch accommodates a particular feature that the vendor has implemented through some software package, does not mean that it is useful in all environment. There is no need to get enticed by a long list of exotic features. First, determine which features are likely to be useful in the local environment and then purchase a base bundle that includes the features wanted. Often to make a sale, a vendor will develop a new bundle of features to suit a buyer's particular needs. If something else is desired add it on later. It is wiser to pay an

additional charge for additional features than to pay for a list of features that will never be used.

The term "feature", as we have discussed, is a broad term that should really be broken into two categories. The two categories available to a PBX system are station and system features. Features that can be requested by the user from his station using some form of signaling are called station features. Features that have been developed for improvement as well as efficiency of the PBX system and are automatically controlled by the system itself are called system features. Direct Inward Dialing, (see page 70) would be an example of a system feature and speed dialing, (see page 67) would be a station feature. The features discussed from here on will be those that are considered station features.

Station features are all accessed by the user through his station device. It is, therefore, necessary for the user to be able to indicate to the system which feature he wishes to access. All features are assigned access codes by the system. When using electronic phones supplied by the vendors, there are usually separate buttons that are dedicated to each feature. By pressing the assigned button, the code for that particular feature is transmitted to the system. If the feature has no assigned button, then that feature code can be manually dialed, or the code programmed into the memory of the phone for speed dialing.

Feature phones because of their universal adaptability, have no pre-assigned feature buttons. All feature codes are programmed into the unit. Feature phones are much cheaper than electronic phones and should be considered when setting up a system. Since many PBX

vendors are offering such a host of different features, it would be unwise not to check both feature phones and electronic phones before purchasing. Often a blend of the two based on use is the appropriate choice.

Many times a feature has to be enabled while either connected to a party or while in the process of ringing another party. Some kind of signal must be sent to the system indicating the need to request a feature. This special kind of signal is called a "flash". A flash signal is used to tell the system that a request is coming. If the feature is built into an electronic phone, the flash precedes the request automatically when the required feature button is pressed. If the feature is dialed by an access code on a feature phone either manually or through some pre-programmed code, the flash button is first depressed immediately before dialing the access code. When a flash is recognized by the system, the conversation presently engaged in is put on momentary hold and a dial tone indicates that the system is ready to receive the code representing a feature. It is possible to signal the PBX from standard dial or tone phones by using a "hook flash". To do this, the user must become very adept at holding the disconnect button down for a length of time short enough to indicate a flash but not long enough to indicate a disconnect signal.

Some of the more commonly used features are listed below. By no means is this a complete list. The generic name will be used. Many features that do the same thing are given different names by different vendors. All of these features are usually included in the basic package.

- (1) **Hold** - a feature common to all PBX systems. Hold lets one party place another party in a temporary disconnect state while some other function is performed. The party placed in the temporary hold state is never really disconnected from the other party. Hold is often incorporated with other features listed below.
- (2) **Transfer** - another of the basic features offered in a most base feature bundles. The basic transfer function allows one party transfer a second party to another station without having to go through the process of hanging up and redialing. This features usually only works within the confines of the PBX system being used.
- (3) **Speed Dialing** - is a feature built into the database associated with the PBX system as a directory of numbers. These numbers can be both internal to the system or to the public network. With the speed dialing feature, an abbreviated code is assigned to each number. For example, a party has an actual assigned number of 1-613-243-2431, with an abbreviated code of 135 assigned to it. A party can dial 135 and the speed dialing function will cross check and dial the appropriate number 1-613-243-2431 associated with that code.
- (4) **Station Call Forwarding** - a feature that gives a party the ability to temporarily route his/her calls to a different extension. The party does this by re-assigning the primary number assigned by the system to their telephone to a temporary secondary access number. When their primary access code is dialed, a pointer or index, diverts the call to the new, temporary location. At any time, the party can cancel the secondary access code and the system will return to the originally assigned access code. The calls then return to the original location. This is a great feature if party plan to be at a different location and want to have all their calls routed to them on a temporary basis.
- (5) **Redialing** - a feature that enables the user to redial the last number called without dialing or looking up the number again. The system software will remember the dialed number and only the push of a redial button is needed to try the number again.
- (6) **Conference Calling** - is by a feature that allows more then two parties to carry on a conversation concurrently. The first connection is made in the familiar fashion. The first party then places the second party on hold and flashes the system through the conference button that another connection is going to be requested. The first party then keys in the correct number to contact the third party. This can be continued until the necessary number of parties are connected or until the system limits are reached. An additional feature called "drop off" allows the last party connected in the sequence to be disconnected in case of a mistake or in the event

the called number is answered by another party.

- (7) **Camp On Queuing** - lets the user upon receiving a busy signal set up a system monitor of the busy line. The system will then signal the party originating the call when the line is free. The system will reset the original request at this time if the party wishes to pursue the call.
- (8) **Automatic Camp On** - lets the calling party give the called party a signal indicating that there is a call waiting to connect. The called party can request no camp on if he likes. Sometimes it is an advantage to be able to signal a party and at other times it could be an inconvenience for the called party. The latter is true if the line is being used as a data connection.
- (9) **Distinctive Ringing** - as the name implies this feature gives different ring signals for different source groups.
- (10) **Do Not Disturb** - makes a parties phone unavailable. The calling party would hear a ring, the phone owner nothing. This feature can usually be incorporated with "call forwarding" so that the call can be automatically forwarded without bothering the owner of the phone.
- (11) **Call Waiting Signals** - are methods by which the PBX system can signal the user that there is an incoming call waiting while you are talking. It is usually a light on the phone, or sometimes the user might hear a beep on the line.
- (12) **Automatic Reminder** - a feature that gives the user the ability to program a wake up or message call for a certain time and/or date. The PBX will then ring back at the specified time and/or date and give the message.
- (13) **Executive Override** - gives the user the right to break in on the conversations of anybody else. Usually this feature is only allowed to a privileged few within an organization. When the "executive override" button is depressed, a beep or other kind of signal would indicate to the parties that they will be joined in the conversation by someone. Within a few seconds the overriding party will be connected to the conversation. The tone is a warning so the parties will not interpret the action as eaves-dropping.
- (14) **Call Pickup** - gives other people the ability to answer a call on another phone in a office without physically walking over and doing so. For instance in a sales office one person can be left to man the office and answer any calls that ring within the immediate area from his desk. The person who plans on answering simply dials the number of the extension ringing. To eliminate unauthorized

persons answering calls, groups can be set up so that not everyone can answer a ringing phone.

- (15) Meet Me - is a feature especially suited to hospital environments. An announcement is placed over a paging system and anyone who calls in the extension announced will be placed in a conference call with the other parties. This is great for a team of doctors wishing to consult on a patients status.
- (16) Private Call - lets the phone user restrict anyone with other features such as "executive override", "camp on", or "call waiting", to disturb a party talking in some private conversation. It also stops the operator from breaking into the connection. This feature is not only important when having a private conversation, but is very important when transmitting data, since data transmissions could never tolerate the beeps and clicks of the various signals associated with the many break in features mentioned.
- (17) Voice Messaging - is the PBXs solution to the telephone answering machine with some additional add-ons. When a party can not connect to another number for various reasons he can leave a voice message on the PBX system. The called party can then check his voice mailbox and retrieve any messages left. In addition to this the calling party can leave multiple messages to groups of users, or a party can forward a message they have receive to other individual parties or groups. A party can also phone into his extension or group and collect any messages left at any time. This requires no operator intervention. The need for answering machines on everyones' desks is eliminated.
- (18) Direct Inward Dialing (DID) - a system feature that reduces the number of wire pairs connected to the central office from the PBX for incoming calls. A direct inward dial call is viewed by the PBX system as an internal call, and consequently does not have to go through the PBX operator. For example, lets say that the 100 phones on the PBX with the most outside activity are chosen. These selected numbers would be keyed into the data base of the local central office of the public telephone company. When the local central office recognizes one of these numbers being called it is directed to one of the 100 DID trunks. The PBX then treats the 100 DID trunks as internal ports and uses the last four digits of the number to connect to the appropriate line. The number of DID trunks does not necessarily have to be one to one. The number of DID trunks is determined by some traffic algorithm so that maximum connection rates are achieved.

Data communications devices are afforded many of the same features as telephone users. A call from a data terminal can be placed to

another terminal or host computer, on site or off site, through the same PBX that is used for telephone connections. A station extension can be dialed from the terminal's keyboard. Redial, speed dial, and call forwarding are three features that are adaptable to data communications. Since data devices transmit digital information, not tones as phones, they have the unique ability to communicate with the switch itself. Instead of just returning a busy signal when a line is busy, the switch can send printed messages to the terminal, giving the reasons for the busy signal. For example, it could print: " too many people are camped on to a line waiting for it to become free." This gives the caller the ability to decide if he wants to add his name to the camp on queue, try later, or use some other technique to get through.

There are many other station features available to PBX system user. The above are a few of the station features that we feel are the most important ones.

3.4. OTHER PERIPHERALS AND OPTIONS

What differentiates a piece of equipment attached to a port from being called station equipment or peripherals and options? Station equipment is that group of devices with which the user has direct hands on contact. Options and peripherals facilitate and optimize system control and system management. Users are usually not aware that these devices are attached to the PBX, let alone exist. They are designed to work transparently when the associated function is called upon. Each option and peripheral device requires a line circuit to function. It is therefore, very important that the user carefully consider his choices in this area. The question of trade-offs in the assignment of line circuits becomes important. The following are examples of some of the optional and peripheral devices that the user can select. All these devices require that a port be dedicated to their use.

- (1) **Modem Pools** - A modem is a device that converts digital to analog. A modem pool is a group of modems configured in some way so that the system automatically selects an available modem and assigns it to a connection that requires a modem connection. The modem pool is capable of converting a number of incoming digital signals and routing them to the appropriate analog lines for transmission. The modem pool is usually associated with connecting digital signals inside the PBX system with the public telephone networks that are in many cases still analog. The PBX has a group of modems to choose from to let a user dial to an outside line from his station. A modem pool is useful when there are more outgoing digital lines than there are analog lines connecting to the PBX. Instead of a modem dedicated to each analog line, a requested connection to an analog line can be switched randomly by the modem pool to a free line. The cost of using a modem pool is substantially less than the cost of individual modems. Even though many parts of the public network have been and are continually being upgraded to digital, the majority are still analog. The digital PBX is transmitting both voice/data in digital form. Problems can arise when the public networks need to be used. No PBX switch designed today has as many trunks to the public network as there stations.

- (2) **Back Up Power** - In case of a power failure any PBX system that works under SPC should have an auxiliary power supply. In most cases this is a battery backup system. Why is the telecommunications system so important that it should be kept going. Answer the following questions and it is easy to see why. What caused the blackout? Is someone hurt? Is emergency help needed? Is communications needed to notify other areas in the building about problems or danger? In many PBXs only essential features will be kept up during a blackout. There is no real need for features such as music during hold when the batteries are running the system. Batteries are only good for a limited span of time. In certain environments, such as a hospital, the telecommunications system is a necessity. Battery backup can be further supplemented by generators. The degree of backup is a cost function that the purchaser of a PBX must decide on based on their own specific requirements.
- (3) **Paging units** - are becoming another very popular option by users of PBX systems. Anyone who can access the system can take advantage of the paging port or group of ports. Paging can be designed to cover a whole building or can be zoned much like heating. Each zone would require its own port as would the all call page. The user could access a particular zone or access all zones by using the "all call" access number. Each zone is accessed by dialing the specific number assigned to that zone. Different zone areas can be established by combining other zones. Other zones can be established that are immune to specific page types except in an emergency. A option called "meet me" lets a person who is being paged have automatic access to the person paging him/her without knowing where the pager is located. All the paged party does is dial a universal access number and they are connected to the correct paging party.
- (4) **Recording devices** - can be attached to a port as a peripheral. Many features such as automatic call distribution, call detail recording, and voice/message systems require a recording device to function. Recording devices can range from a simple cassette recorder to more complex hard disk drives. If we are concerned with the recording of digital vs. analog signal, a floppy or hard disk is the obvious choice. One reason for this is that information on a disk can be accessed in a random fashion, not requiring a sequential search as tape units do. This leads to swifter access of information. It is possible for the system to encode additional information such as the destination and user if stored in digital form.
- (5) **Maintenance Ports** - are probably one of the more important options that a system has. They allow the telecommunication manager total access to the system. The daily operations of the PBX system can be directed from a console or terminal. The telecommunication manager has the ability to renumber extensions and change the class of services provided to the different stations. Most systems have some sort of self diagnostic software built into their operating

systems. If a problem is found an alarm on the console sounds. The manager can then query and test the system through the maintenance ports to locate the source of trouble. The last use of the maintenance ports is the collection of information and statistics about how the systems used. Reports can be generated using these facts for hard copy distribution. These maintenance ports can be both hard wired to the PBX or accessed through a dialup facility using a standard data terminal.

- (6) Other Options and Peripherals - are available for use on the PBX system. Teleconferencing is a option that is becoming very popular with the education and business professions. Security systems using both voice and video surveillance can be controlled by the PBX's processor. Monitoring can be directed to the security guard station or appropriate authority. Another option, environmental monitoring and control can help defer the cost of the PBX system. Sensors in a port can be used to indicate temperature, light, and time. The PBX's computer can control the heating and cooling of the building, turn off lights when left on, and activate sprinkler systems in case of fire. Honeywell Corp. has been a leader in the development of such systems.

3.5. TRANSMISSION LINKS

Transmission links are similar to the veins in our body because they carry the life blood of the PBX. All the various forms of voice and data transmissions are transported from the stations attached through the switch and then to their destinations by these links. There are two classifications for these links: station lines and inter switch links. The station links or lines connect the various devices that constitute the PBX system. The inter-switch links connect to multiple PBX systems together and also connect the PBX to the central office (CO) of the public network. Each of these two classifications of links are distinguished by the volume and type of equipment to which they attach.

Transmission links are named by three different methods. The first method uses the type of equipment to which the various links attach. A station line would connect a station to the PBX. Another example would be an off-premise line, which would connect an off-premise station to the PBX. The second naming method is based on the type of transmission. This method generally use the name analog and data. Sometimes this is ambiguous, since analog lines are capable of carrying data with the use of a modem, and data lines can carry analog using a CODEC. Often the words "voice" and "digital" will be used in the second method. The last method of naming is by function. Loop start, wink start, and ground start are good examples. In many cases, a combination of the methods are used, or the same type of transmission will have more then one name.

3.5.1. Station Lines

Station lines are the largest single group of lines attached to the PBX. Single pair (two-wire twisted) and twisted pair (two single pair) are the most common station lines in any PBX wiring scheme. Up to four-pair wire could be used for some devices but this is very rare. The wire pairs form a loop for full duplex transmissions between the two parties. The wire pairs are twisted to prevent mutual induction (crosstalk) in a neighboring loop. Single pair wire can accommodate analog transmissions well. If we want to have the convenience of both voice and data, twisted pair is used today. In today's wiring scheme for a digital PBX, twisted pair is distributed throughout the premise to every possible location where a phone or data device needs to be located. The major advantage of twisted pair wiring is that it is inexpensive, very compact, easy to work with, and easy to replace or extend to new stations.

In a twisted pair environment, a single line telephone requires one pair of the two for voice transmissions. The other pair might be used to supply power for a light. If an electronic telephone is connected, both pairs are used with one pair carrying the voice signal and the other pair used for signaling and any additional power requirements that the device needs. If the station device is a data device, one pair is used for data transmissions and the other pair for signaling purposes.

CCITT standards call for voice signals over twisted pair to be transmitted at 64k bps digital using PCM over public systems [CCIT73]. There are two forms of modulation commonly used in today's digital PBX

systems. The most common is PCM with some systems using delta modulation. PCM is by far the more prevalent modulation technique because it can directly interface to the public systems without any adjustments. American Telecom's Focus, Ericsson's Prodigy, and United Technology's UTX 1001 are some digital PBX systems that do use delta modulation. None of these really have any substantial number of units sold in the U.S. PCM uses a sampling rate of 8000 samples per second encoded over seven bits which gives 128 possible frequency levels. An eighth bit gives us the 64k bps. Twisted pair using time division multiplexing is another method used to transmit 24B + D (24 voice channels and 1 signaling channel), commonly called T1 span. T1 by CCITT standards [CCJT73] is transmitted at 1.544 m bps. It has been shown that twisted pair can carry in excess of 7 m bps in tests [KUM78].

The twisted or single pair lines distributed from the various station devices are routed in groups of twenty-five pairs to a local distribution frame. The local distribution frames are then merged into one hundred pair cables and brought to the main distribution frame adjacent to the PBX unit itself. The main distribution frame gathers similar station devices into 25 pair connecting blocks. From the connecting block a 50 pin Amphenol connector makes the connection to the appropriate ten group port at the PBX. Number and size of frames, blocks, and ports can vary, but the process is the same.

3.5.2. *Inter-Switch Links*

There are many situations where a PBX must be connected to another switching network. These other switches can be other PBX

systems or a switch located at one of the central office used by the local BOC. Every PBX has some number of ports dedicated for attaching to the CO of the public BOC. These are commonly called CO trunks by the PBX users. Twisted pair is the common choice for this. Each twisted pair line connected to a port on the PBX carries an individual channel from the PBX to the CO switch for outside calls.

Often, an organization has a substantial number of calls placed from their PBX through the public network as long distance toll calls. An organization will often acquire a WATS line (Wide Area Telephone Service). Regular toll calls are billed by time of day, distance from source to destination, and duration of call. A WATS line user is billed by time and zones at a special rate. Both IN WATS and OUT WATS services are available. IN WATS is commonly known to us as toll free 800 dialing.

In certain cases, it is advantageous to have a dedicated connection between two or more PBX systems. This often occurs if an organization has more than one PBX in use. For example, when two locations of an organization are separated by some distance, there are two possible alternatives to choose from. Depending on cost projections, the organization can set up its own facility to handle the connection or lease the service from their local BOC, or one of the other long distance carriers such as AT&T, MCI, SPRINT, etc. If the facilities that house the two PBX units are fairly close, the organization could elect to lay its own twisted pair and set up a T1 circuit. T1 is based on 24 digital voice channels of 56 k bps. Over this T1 line, both voice and data can be multiplexed between PBXs [CCJT73]. If an organization does not have the ability to lay or string its own cable, a T1 line can be leased from the local BOC.

This whole issue is called as "bypass". The issue of bypass is becoming a primary concern of telecommunication managers. This has primarily occurred because of the deregulation of the AT&T Bell telephone system. Many new options are becoming available to users. Customers can now set up their own cable or microwave systems. These systems can be used to bypass the local BOC and directly connect to one of the many long distance carrier that have sprung up. Organizations can directly access their fiber based networks, or go directly between their own PBX systems.

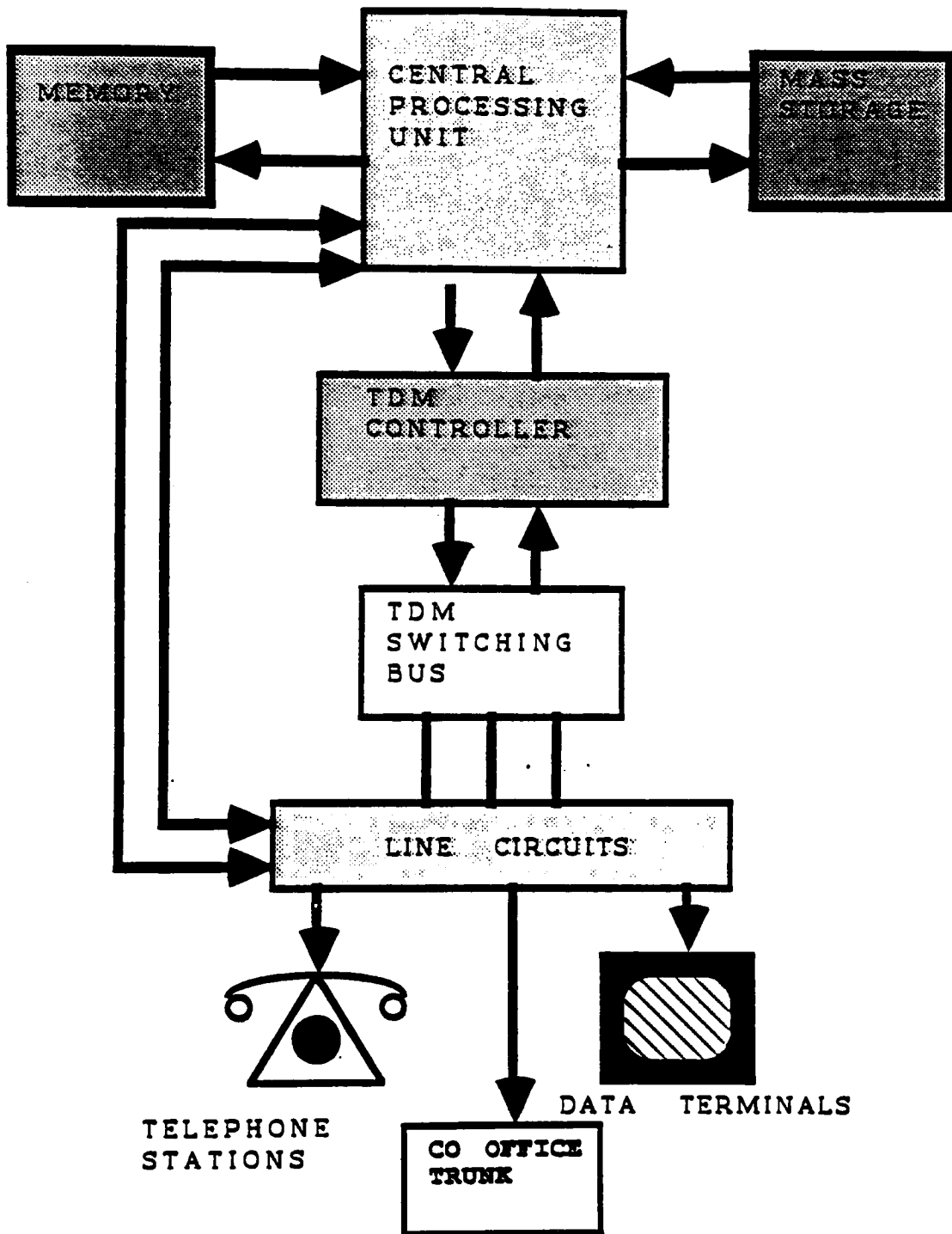


DIAGRAM 3.6.1
COMPONENTS OF A DIGITAL PBX

3.6. ARCHITECTURE OF A PBX SWITCHING UNIT

Throughout this chapter, we have taken the various parts of the digital PBX and described them as separate units. The components of the switching unit can not be described in this fashion. The switching unit is the integration of three closely connected components. The three parts are impossible to describe individually without mentioning the other two. These three components are the line circuit card, TDM controller and TDM bus, and the CPU. To how the switching unit works, we have decided to describe each part in an interrelated fashion. Hopefully, by doing this, the reader will acquire an understanding of how the digital PBX switching unit operates.

The switching unit is the heart of any digital PBX system. All third and fourth generation PBXs in use today use some form of TDM controller that manipulates time slots over a TDM bus for switching. This common transmission format gives the digital PBX the capability of switching both voice and data signals in an integrated fashion, but not simultaneously from the same station. To accomplish this, both voice and data switching requires a common transmission format through the switch. This is accomplished by using a digital transmission stream. Since a variety of analog devices can act as station equipment, all the waveform signals they produce must be converted to digital at some point in the system before reaching the TDM controller and bus. Analog signals can be converted to digital at the source or at the line circuit card. This is not necessary for data devices because the signal from a data device is already in a digital format when it is transmitted from the source.

The point at which the analog waveform signals are converted to digital signals is one of the things that differentiates third from fourth generation PBXs. Regardless of where the conversion takes place, all the waveform signals in a system are converted to digital by a device called a "codec". The codec is a device that does exactly the opposite of a modem. Newer digital telephone units can have the codec built into the headset. A codec can also be a black box that plugs into the side of older analog phones. Dedicating a codec to each station device is expensive. In some digital PBX systems, the codec is resident on the line circuit cards. By placing the codec on the line circuit card, the PBX system allows all the station devices connected to the line circuit cards to share a common codec. This is similar to devices sharing a modem in modem pooling, discussed on page 72. The original analog signal is transmitted in waveform over the twisted pair until it reaches the line circuit card. Digital conversion at the line circuit card is an advantage if we want to use a mix of analog and digital equipment in the system. Whatever the case, once the conversion has occurred, a synchronized, digital bit stream of 64 k bps is transmitted from that point on throughout the digital PBX system. If a data device such as a terminal is used, the output is asynchronous at a speed up to 19.6 k bps, or synchronous at the 64 k bps mentioned above. Newer fourth generation PBXs transmit at much faster speeds and will be discussed later. Whatever the transmission rate, all voice and data transmissions use twisted pair wire, the same TDM controller and bus, and the same control logic. The only difference is that voice transmissions require only one pair of the twisted pair wire, where the data transmissions requires a minimum of three or four wires. Both voice and data can be interlaced

simultaneously over the same wire and separated at the line circuit card. (See diagram 3.6.1 for a typical digital PBX architecture)

The line circuit card plays a very important role in the digital PBX switching unit. The primary function of the line circuit card is to act as the interface between the station devices and the control element via the TDM bus. A line circuit card in today's digital PBX is nothing more than a small microprocessor that uses LSI, Large Scale Integration, to perform a host of tasks associated with it. The line circuit card contains the logic that detects when a station device is requesting service, when a connection has been terminated, what kind of station device is being used, how to communicate with the CPU to set up a connection, and how to interpret control signals for implementation of value added functions.

In order to accomplish these tasks, the line circuit cards used in today's digital PBX systems must have RAM, ROM, buffers, codecs, and logic circuitry capabilities built into them.

None of the larger digital PBX systems have a line circuit card dedicated to each station device attached to the system. The line circuit cards are usually a shared resource with four, six, eight, or sixteen station devices attached to each of them. The line circuit cards are slotted into the cabinet of the PBX, with each slot having an assigned address. Line circuit cards in most newer systems are universally adaptable, fitting any slot in the cabinet. Each station device attached to the line circuit card is also given a specific address by the system, but these addresses can be dynamically reassigned by the system at any time. Dynamic reconfiguration at any time without having to rewire

it allows for easy expansion. The addressing of devices is done at two levels. The two-tiered addressing allows the system CPU to use a two-tiered polling scheme to decide which devices are requesting service. In many digital PBX systems, polling is done using line circuit card addresses first to see if any activity is present, and then at the station device address level if the process needs to be continued.

There are two distinct kinds of signals that the line circuit cards has to interpret. These are voice/data transmissions and control signals. All voice/data transmissions must be able to pass through the line circuit cards with little or no distortion. A control signal is separated at the line circuit card, interpreted, and then sent to the system CPU via the TDM controller along with additional control information. Control signals entering a line circuit card never remain in the same format as when they entered it. We could say that the line circuit card is oblique to control signal transmissions and invisible to voice/data transmissions.

As noted earlier, the line circuit card on one side is constantly monitoring the various station devices for connection requests and termination signals. As the TDM controller receives signals from both the circuit cards and the CPU, these signals are placed in a buffer to await transport to the appropriate destination when timing permits. The control system consists of two parts, the CPU of the system and the TDM controller. There is a constant transmission of control signals between the circuit cards, TDM controller, and the CPU regarding the present state of the system.

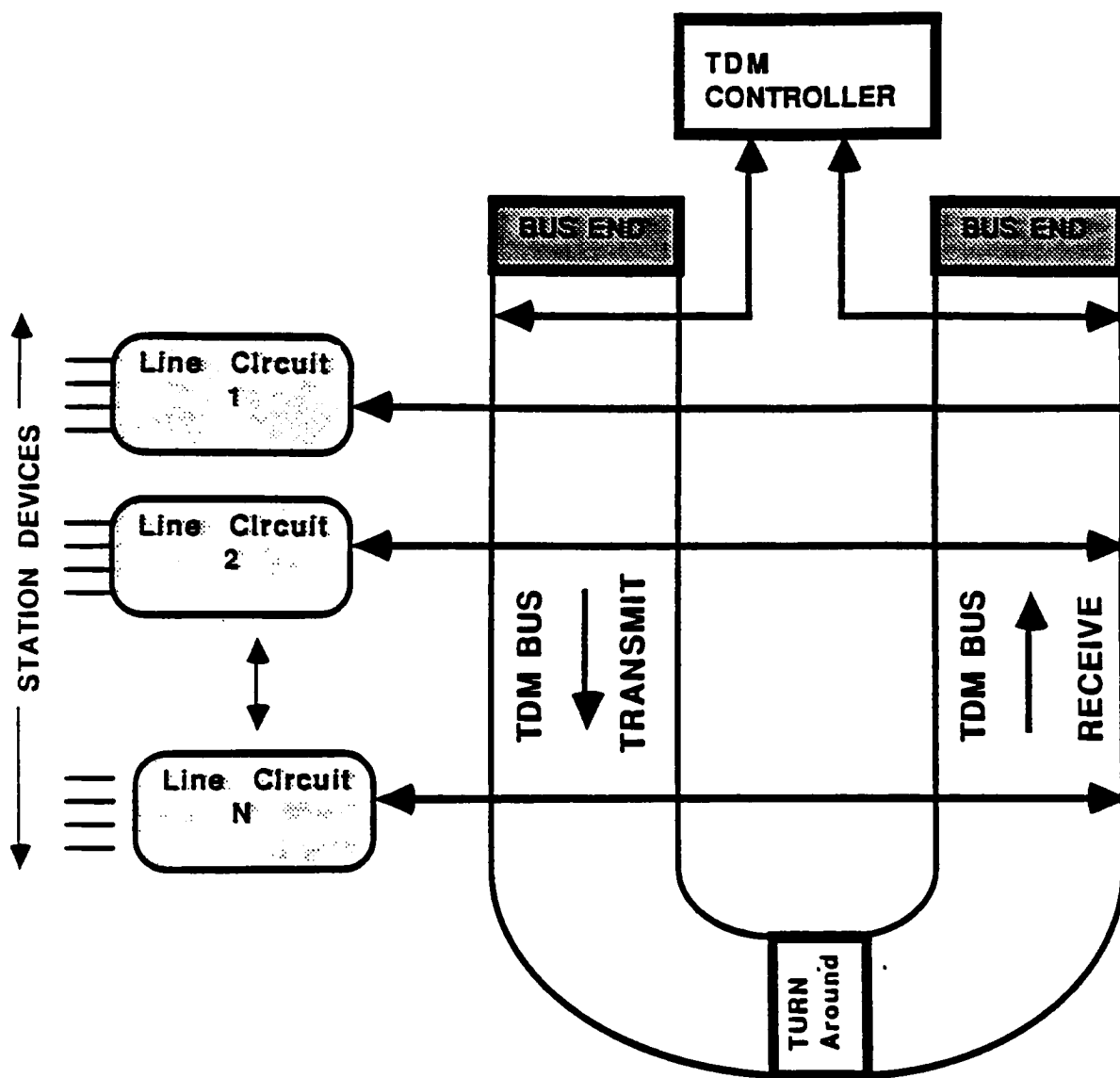


Diagram 3.6.2
TDM BUS & CONTROLLER

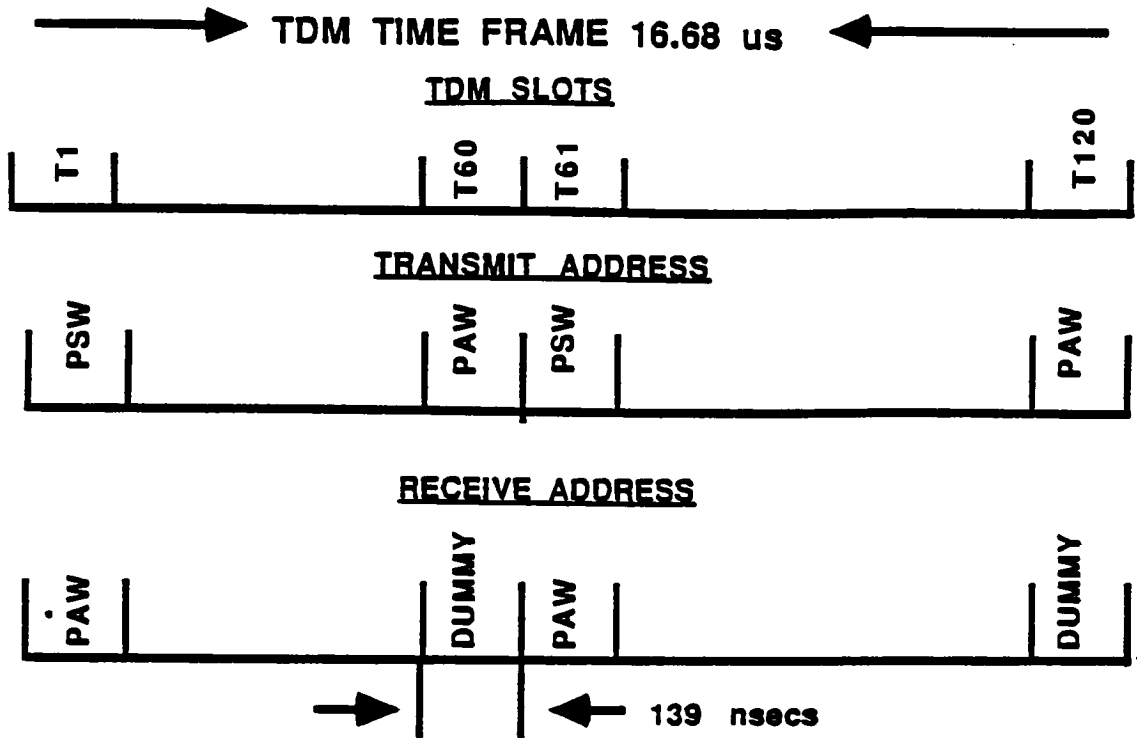
The station devices are constantly transmitting control signals via the TDM controller to the CPU through the line circuit cards. The CPU maintains a table of connections that are presently set up, the present state of the system. The CPU is constantly polling the line circuit cards for new connections. When received by the CPU, it establishes them, checks for associated control signals, is responsible for tone and busy signals, and adds any of the special features that are possible on the digital PBX system.

The TDM bus is the physical medium over which the voice/data transmissions and control signals are transmitted between the line circuits and the TDM controller. (See diagram 3.6.2) The bus is quite simplistic, being just a number of parallel segments similar to a parallel ribbon connector for a printer. The number of segments equals the bit width of the address word transmitted on the bus, usually sixteen bits. Also, there are from two to six additional segments added for synchronization, clock, and timing signals. The speed of transmission on the bus is enhanced significantly because of its parallel structure. For example, if voice is being transmitted in series at 64 k bps over twisted pair, each bit of the sixteen bit word is now transmitted at that same speed but in parallel for a combined speed of 1.024 M bps in a single direction or 2.048 M bps bidirectionally.

The TDM bus in the majority of third and fourth generation switches is unidirectional. Since the transmission can only occur in one direction at a time, two way transmissions are handled in a unique manner. The TDM bus is usually designed in a circular or loop shape. At the half way point is a device called a turn around. One half of the

TDM bus is the transmit side, the other half is the receive side. The turn around acts as a buffer between the two halves and also boosts the signal. All line circuits are capable of monitoring both sides of the bus. The TDM controller also monitors both sides of the bus. Every connection set up between two station devices requires two time slots. One slot for transmission from A to B, the other for the reverse direction. The line circuit cards know when to listen to the appropriate time slots on both sides of the bus. The line circuits do this by synchronizing with the bit and the frame clock signals that are constantly generated by the CPU. These signals are constantly being transmitted on the TDM bus over the additional segments allocated for timing.

The CPU might be the brains behind the operation, but the TDM controller is the device that is "the errand boy", transmitting information between the CPU and the station devices attached to the line circuits, and visa-versa. The TDM controller buffers information sent from the various ports regarding request or termination for service until the CPU can handle them. This is implemented as a First-in, First-out queue (FIFO).



Timing Relationship
Transmit - Receive Time Slots

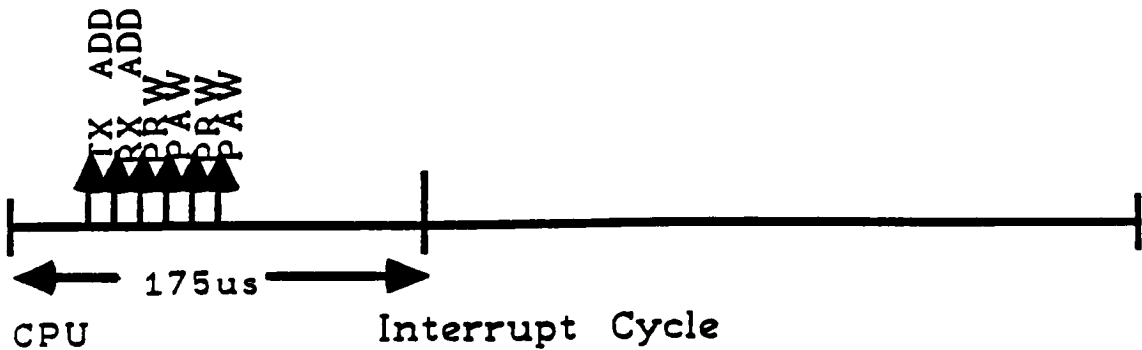
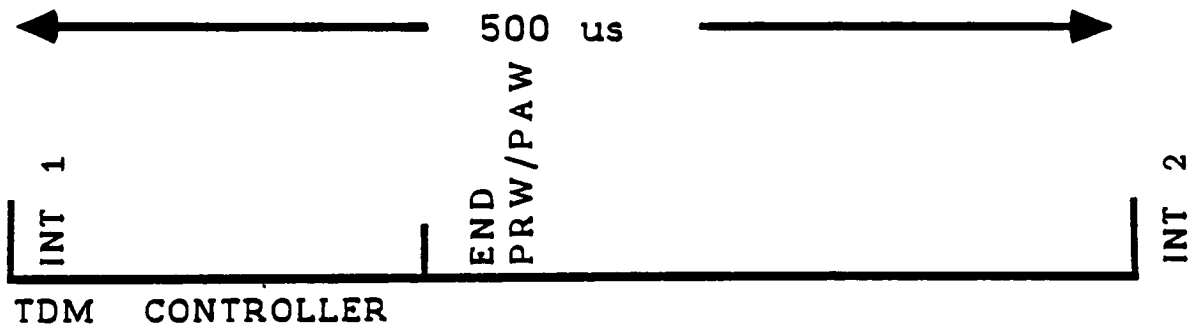


Diagram 3.6.3

The TDM controller is responsible for the actual switching of the transmissions by assigning time slots, and managing the time frame. Typically switching times are measured in nanoseconds. The time slot is the mechanism by which each station device is assigned a sequential turn to transmit over the TDM bus. In a nonblocking digital PBX, the number of time slots is equal to the largest number of possible simultaneous connections times two, plus the number of control slots required by the system, plus the number of time slots required by the additional line circuit cards for features such as tone generation and maintenance ports.

The time frame is composed of a single occurrence of each time slot in some predetermined fashion. The length of the time frame is equal to the length in time of a time slot times the number of individual time slots in a system architecture. Typical time frames are usually measured in microseconds. (See diagram 3.6.3 for a typical synchronization picture.)

A fictitious example of system timing might be a digital PBX designed to handle up to fifty simultaneous calls, tone generation, and six features. Each simultaneous call requires two time slots, one for each direction of communication. This requires a total of one hundred time slots. Four time slots are used by the CPU for scanning, request/termination, and slot assignment purposes. The Tone Generation line circuit card requires three time slots; one for dial tone, one for busy, and one for ring signals. The remaining thirteen time slots can be allocated to the six optional features. The time frame would then consist of one hundred twenty time slots, each time slot appearing once in the

time frame. We will also assume that four station devices or CO trunk lines can be attached to each line circuit. The time frame for such a system could be 16.68 microseconds, making each of the one hundred time slot 139 nanoseconds in duration.

The signaling technique employed by digital PBX systems to recognize, set up, and tear down connections, is based on an interrupt mechanism generated by the CPU. This is performed during what is called an interrupt cycle. Line circuit scanning, line circuit request information, and line circuit time slot assignments are generated during the interrupt cycle to update the present state of the system. The transmission of signaling information occurs between any line circuit and the CPU via the TDM controller on the TDM bus using fixed time slot locations. The CPU communicates with the TDM controller asynchronously and buffers the information either to the line circuit card or from it. Line circuits are scanned sequentially, but can be assigned at liberty by the CPU.

In our fictitious example, we said that four time slots were dedicated to this process. The selected time slots are usually staggered across the time frame for timing purposes. In this case we have four time slots T1, T60, T6, and T120. Our interrupt cycle will be 500 microseconds, giving us the capability of scanning each of the twenty-five line circuits two thousand times per second.

The interrupt cycle begins with the CPU selecting the next two line circuits to be scanned. The addresses of these line circuits are sent to the TDM controller to be placed on the next available time frame in the appropriate time slots. We will use time slots T1 and T61 as our line

circuit scanning slots. A dummy line circuit address can be transmitted to the TDM-controller if the CPU wants to skip the scan of a particular line circuit in the sequence. If the line circuit addressed has active requests for service or termination from the attached station devices, then this information is immediately transmitted during the next available port request slot T60 or T120.

These requests are buffered at the TDM controller in the FIFO buffer described earlier. The CPU is actively processing the buffered requests. If the CPU memory map indicates that the requested station is free, the CPU advises the TDM controller to have the address of the tone generator line circuit time slot connected to the station making the call. The tone generation line circuit has two dedicated time slots transmitting a ring or busy signal respectively. This occurs in the same interrupt cycle on the next time frame with the line circuit assignment slot using the same time slot as the line circuit scanning slot, T1 and T61. Because the ring and busy are transmit only, one time slot handles each of them.

If the on-hook signal is recognized by the CPU from the station trying to be connected, the CPU then sends the necessary information to the TDM controller to set up the connection. The CPU also at this time updates the memory map of the status of all station devices in the system. The TDM controller allocates the two necessary time slots needed from its bank of free time slots and passes this information to the correct line circuit addresses during an interrupt cycle. When the line circuit in the connection senses the appropriate time slots as indicated by the clock signal, it either transmits or receives information. A

disconnect is handled in a similar fashion.

CHAPTER IV

FOURTH GENERATION PBXS

4.0. INTRODUCTION

The primary requirement for business communication systems for the rest of this decade and through the 1990's will be the ability to act as a total communications controller and to be able to incorporate any new technical advances that develop. The fourth generation digital PBX achieves these goals in the most economical and workable fashion. The volume of office communications will increase exponentially over the rest of this decade and continue in the same fashion through the next. Unlike telephones which are used sporadically, terminals and PCs are used less frequently throughout the day. Although attached to the system for long periods of time, data transfers only happen in brief spurts. During these spurts of activity, rate of data transfer is usually the prime concern.

The Fourth Generation PBX possesses sufficient data handling capability to not only adjust to the varying data transfer rates, but also is capable of simultaneously connecting all data devices in a non-blocking fashion. This is the reason fourth generation PBXs are considered the ultimate solution to the voice/data problem. Many applications today will require data speeds in excess of the 9.6 to 19.2 k bps asynchronous or 56 to 64 k bps synchronous that are common to data and voice rates presently handled by most third generation digital PBXs. Data rates across the system should be driven by the application or process being

used, not on the digital PBX limitations. These rates could vary from 300 bps to many hundreds of thousands of bps. To be able to achieve the data rate requirements of today's users, new techniques must be incorporated into the architectural designs of PBXs.

The fourth generation PBXs must also be able to deal with an variety of different vendor's equipment. The current equipment should not have to be discarded, but accommodated and made to look transparent through a unifying communication system using standard interface devices. This should also extend beyond the local loop of the system using X.25 devices to connect to the outside world through the various long distant networks such as Telenet, Tymnet, BX.25, and others that are presently available. All types of transmission mediums should be interconnectable such as coaxial cable, terrestrial microwave, satellite, and fiber optic lines through various kinds of links. (See diagram 4.0.1.) These different linking mediums give the fourth generation PBX greater flexibility to access both public and private data bases, networks, and other PBXs.

In summary, the four generation PBX provides for flexible growth, easy access to a broad spectrum of devices, and integration of a wide variety of applications such as voice, data, text, facsimile, and video. Fourth generation PBXs are distinguished from third generation switches as a matter of degree, not functional differences. Both are totally digital and are capable of handling both voice and data. Third generation handles data as an accommodation to their circuit switched environment.

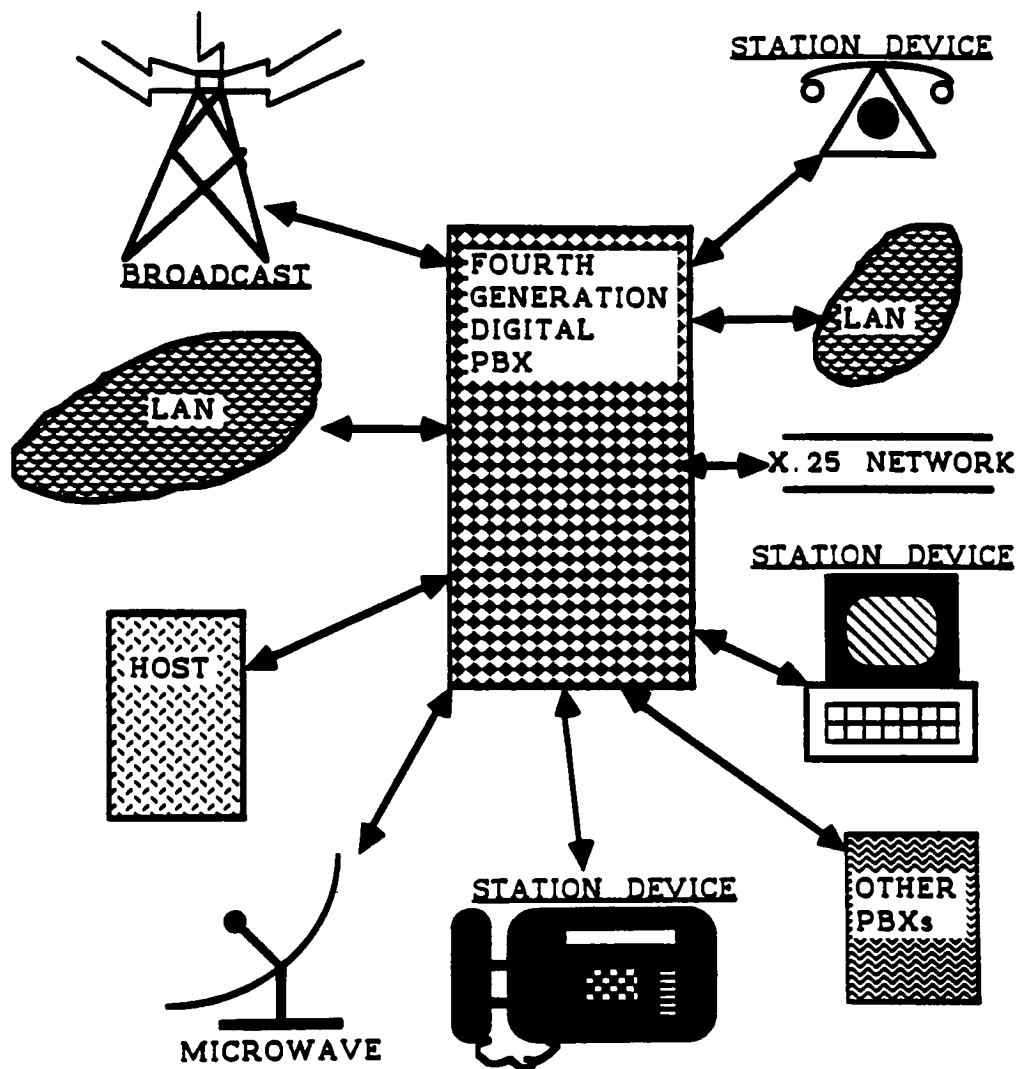


DIAGRAM 4.0.1
THE MULTI-PURPOSE
DIGITAL PBX

Fourth generation digital PBX manufacturers have attempted to incorporate new ideas by making both architectural modifications to existing third generation digital PBX switches, or better yet, starting from the ground up and developed totally new architectures employing the best features of both circuit and packet switching technologies.

4.1. THE FOURTH GENERATION APPROACH TO INTEGRATION

As discussed in Chapter Three, the most cost effective solution to voice/data integration is based on sharing the line circuit of the PBX as well as the local access wiring. When both voice and data are handled simultaneously over the same wire pair, an increase in port capacity is ultimately achieved. A problem inherent to third generation switches is all transmissions are done via circuit switching. If the maximum data rate is 9.6 k bps asynchronous, then only 15% efficiency is achieved through the 64 k bps PCM bit stream common to most digital PBXs. This had been increased to 19.2 k bps asynchronous in many cases, but this still only gave an increase to about 25% efficiency.

The integration of a packet switched bus resident in the digital PBX system architecture is the solution that many manufacturers see to the problem of application driven data rate transfer. The concept of a digital PBX having the capability for both dedicated and virtual connections holds a lot of promise. A virtual connection, associated with packet switching, is a logical connection between two endpoints. The long intervals of idle time so typical to data transmissions do not use switching capacity in a virtual connection. This additional time, when a connection is not actually transmitting data, can be allocated to establishing connections for other users. The reallocation of time greatly increases the capacity of the system to handle data device.

Packet switching inside the digital PBX has several other attractive advantages. Packets do not depend on applications. Packets can represent a universal transport mechanism independent of format, speed,

or content. Packets can be used for voice, data, text, facsimile, and probably other new applications still in the development stages. Packets consume transmission and switching resources only when required. This can substantially reduce the resource requirement for a digital PBX system by increasing the data capacity of a system.

A dedicated connection, on the other hand, is best suited to voice transmissions and select data connections. A broadband bus using frequency division multiplexing seems to be the ideal solution in this case. Bandwidth can be dynamically assigned based on user requirements. By not having fixed requirements for channel size, the transmission rates can be determined by the applications as they should be. An example might be a large file that needs to be transmitted between two hosts. By utilizing the maximum bandwidth allowed, say 20 - 30 M bps instead of the normal 64 k bps allowed by most digital PBXs, the file can be transferred faster and more efficiently.

By providing a noncontention mode bus for voice and a contention mode bus for data, the users have at their fingertips the most appropriate system for handling various transmission requirements. Voice quality is maintained due to its priority handling by dedicated circuits, while the flexibility of virtual connections accommodate data more effectively.

The line circuit card in a fourth generation architecture assumes additional responsibility. When a transmission from the various attached devices reaches a line circuit card, a decision based on the speed, format, and content of the transmission is used to direct it to the appropriate bus. Voice transmissions are handled best by the

noncontention circuit switched bus. Data, on the other hand, do not necessarily always have to be directed automatically to the contention mode packet bus. Deciding what format a data transmission takes has been left up to the individual manufacturers. Some manufacturers have taken the view that all data transmissions should be packetized and all voice transmissions switched in circuit mode. Most have elected to let the user decide by having a variety of line circuit cards, each performing a different routing function. There are line circuit cards for analog voice, both digital voice and data, and others for packet data. The user elects the type of connection needed by a device. However there is no dynamic allocation of transmission format by a device once connected to a specific line circuit card.

Two other goals are also inherent in the new fourth generation architectures. These are distribution of the hardware and the software functions. Distribution gives the fourth generation digital PBX the capability to cover a larger geographical area than third generation PBX systems. Hardware distribution is accomplished by having the digital PBX's architecture based on a node schematic for expansion. Each node has the necessary hardware and software to function as an independent unit if other nodes in the system go down. Distribution is made possible by having the PBX system take advantage of the new high speed transmission links such as fiber optic lines or coaxial cable for inter-node communications.

4.2. *FOURTH GENERATION IMPLEMENTATION*

We have discussed the goals that manufacturers were trying to incorporate into their fourth generation switching architectures. Three different approaches have been taken in implementing these goals. Manufacturers such as Ztel, and CXC Corp. have decided to build an entirely new series of switches predicated on the dual bus architecture. Others such as Northern Telecom and AT&T have elected to go with a proven circuit switch and have added a packet switch option that can be interfaced into the existing architecture of the parent digital PBX. Interfacing a packet system into an existing architecture is similar to adding a new feature onto a PBX system. Two companies, Rolm and GTE have taken a different approach. They have entirely reworked the older model architectures of the CBX and GTD digital PBXs respectively, converting these older models into fourth generation architectures.

4.2.1. *Original Designs For Dual Bus Architectures*

Two companies have adopted the approach of designing an integrated digital PBX system from the ground up. Both Ztel and CXC Corp. were forced to out of necessity, since they were both new companies looking for a foot hold in the blossoming digital PBX marketplace. Looking toward the new frontiers opened up by expanding data emphasis through the digital PBX, Ztel and CXC decided to introduce a unique product based on the best of both worlds, circuit and packet switching combined into a single system architecture.

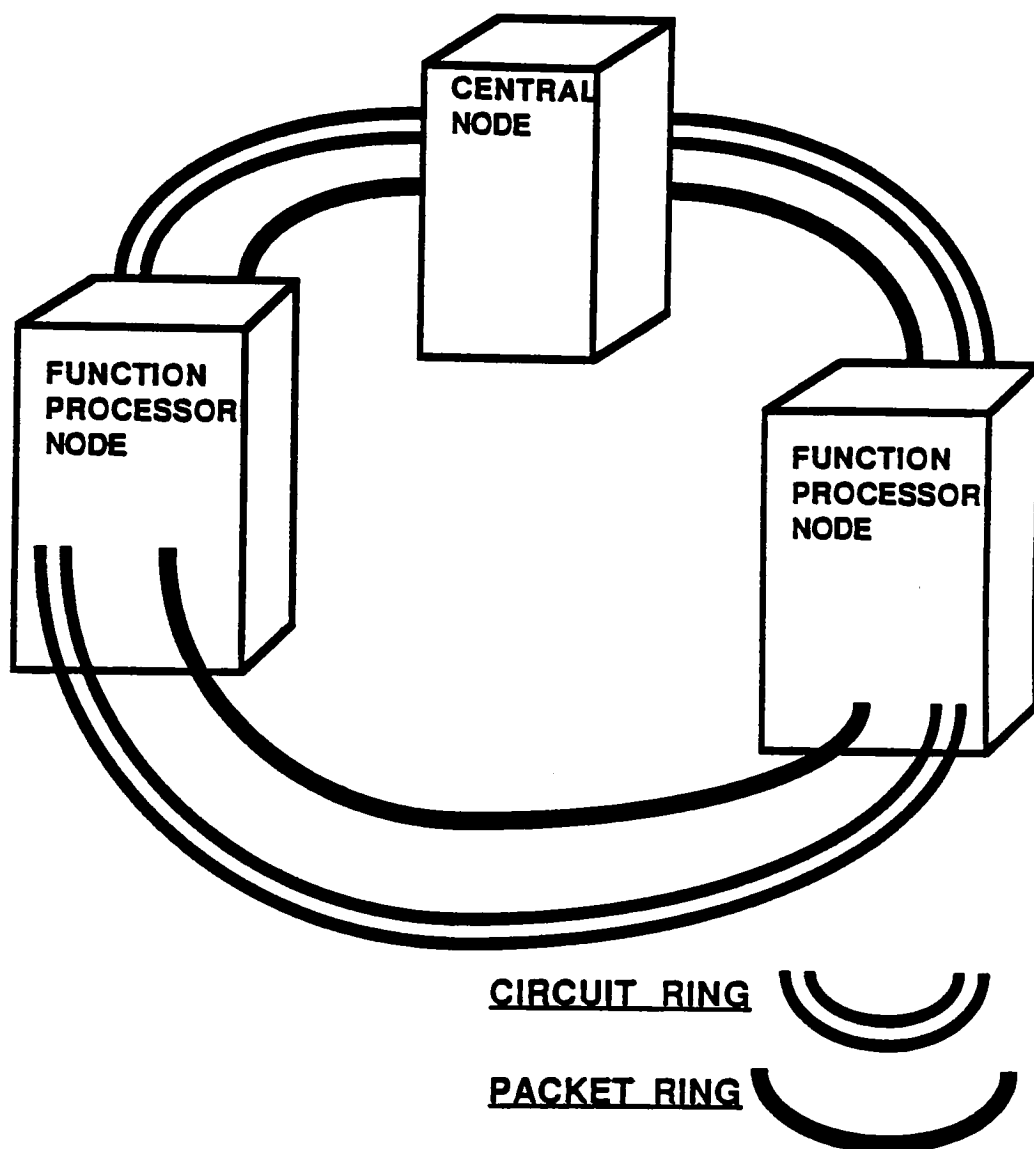
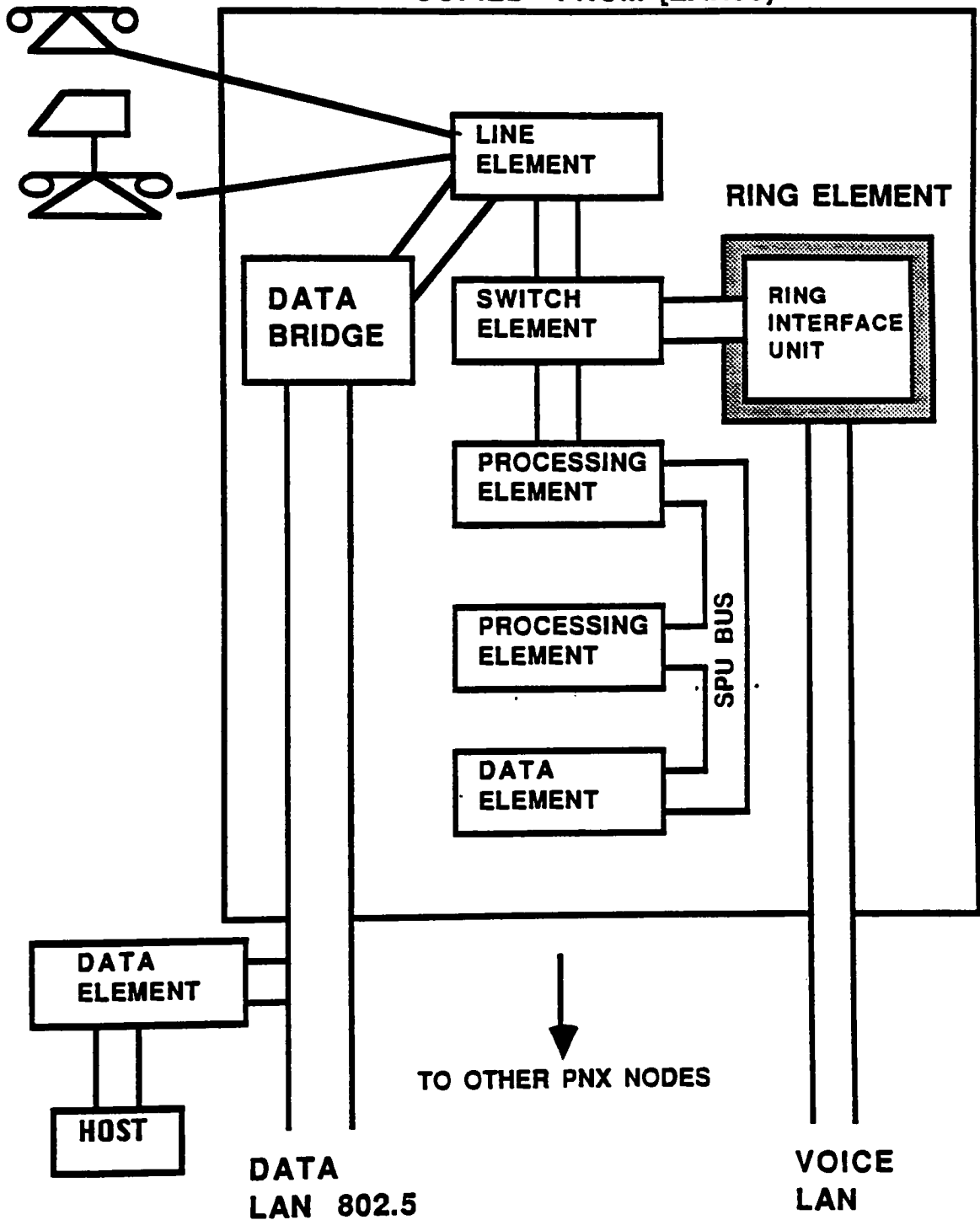


DIAGRAM 4.2.1.1.1
ZTEL PNx FUNCTIONAL PROCESSOR
AND RING ARCHITECTURE

DIAGRAM 4.2.1.1.2.
PNX SYSTEM PROCESSING UNIT
COPIED FROM [ZAN85]



4.2.1.1. Ztel's PNX

Ztel publicly introduced the PNX at the May 1985 ICA Show [DAT86]. This system conforms to IEEE 802.5 token-ring standards. Each Switching Process Unit (SPU) node supports up to 1500 voice and 256 data terminals. Multiple SPUs can be interconnected via two types of rings. The circuit rings can carry all digitized voice traffic at 64 k bps and circuit switched data at up to 56 k bps in a 802.5 compliant token ring network. (See diagram 4.2.1.1.1 for an architectural view of the PNX.) All control signals and packetized data from special teleterminals travel over a 10 M bps Ethernet system. Each of the Ztel rings operates at 10 M bps and multiple rings can be added on demand (See diagram 4.2.1.1.2 for ring overview.) Ztels Distributed Network Operating System (DNOS) coordinates all applications programs, signaling, and systems monitoring [ZTEL85].

Designed within each SPU node is all the necessary switching hardware. Each SPU performs internal digital or packet connecting; external transport of digital and packet signals to other nodes; data base management of the system; protocol conversion; and feature addition to the various data streams. Trunks are also provided to connect to public and private voice and data networks. All station devices are connected to each PNX node through Line Elements analogous to a line circuit. Each Line Element contains 16 ports each of which can handle either digital voice/data or packetized data. All digital or packet conversions are done at the device level using either built in or add-on interface devices. Only digital and packetized data signal can proceed past the Line Element.

When the data streams reach the Line Element, internal circuitry decide the format of the transmission and routes the data stream to the appropriate switching mechanism. All packet data is sent to a device called the Data Bridge which acts as a packet controller for the Data LAN. If the call is internal to the node, the destination device address is checked and the packets go to a Data Element for any necessary protocol conversion before being routed to the destination. An example might be emulation of an IBM 3728 workstation by an asynchronous ASCII terminal [ZANN85].

If any user or system features are needed, the Data Element can then send the packets to a Processor Element for further enhancement on the internal SPU Bus. The packets are placed back on the Data LAN, transmitted back to the destination Line Element, and on to the station device requested. Packet data destined for other PNX nodes goes through the same procedure, but the protocol conversions can be handled in one of three places. As before, a Data Element located in the PNX node where the signal originates can do the conversion, the conversion can be done in a Data Element in the destination node, or a Data Element can be remotely located adjacent to an external host device.

If the Line Element senses a digital bit stream, either (voice or data), the signal is routed to the Switching Element. This standard PCM signal is switched via a TDM bus with the appropriate features added Process Elements. If the address of the destination device for the circuit switched data is in another node, the switch connects to the Ring Element. The Ring Element, using a built in protocol, converts the cir-

cuit switched digital PCM signal into a packet format. The Ring Interface Unit, actually a packet controller, places the packets on the Voice LAN for transport to other PNX nodes. The Voice LAN can handle up to 110 simultaneous connections over this baseband bus using TDM techniques. The distributed architecture of the PNX accommodates wide and local area networks with ease [ZTEL85], [ZANN85].

There are three major problems stand that out immediately in the design of this system. The first is the lack of an Analog Line Circuit. Many Analog devices are still employed in many companies. Ztel expects anyone who utilizes their system to buy add-on modems for all their analog devices if they want to continue using them with the PNX system. It would be much easier if an Analog Line Element were employed at the PNX node so that 16 analog devices could be connected to the switch and be converted at that point to a digital PCM format.

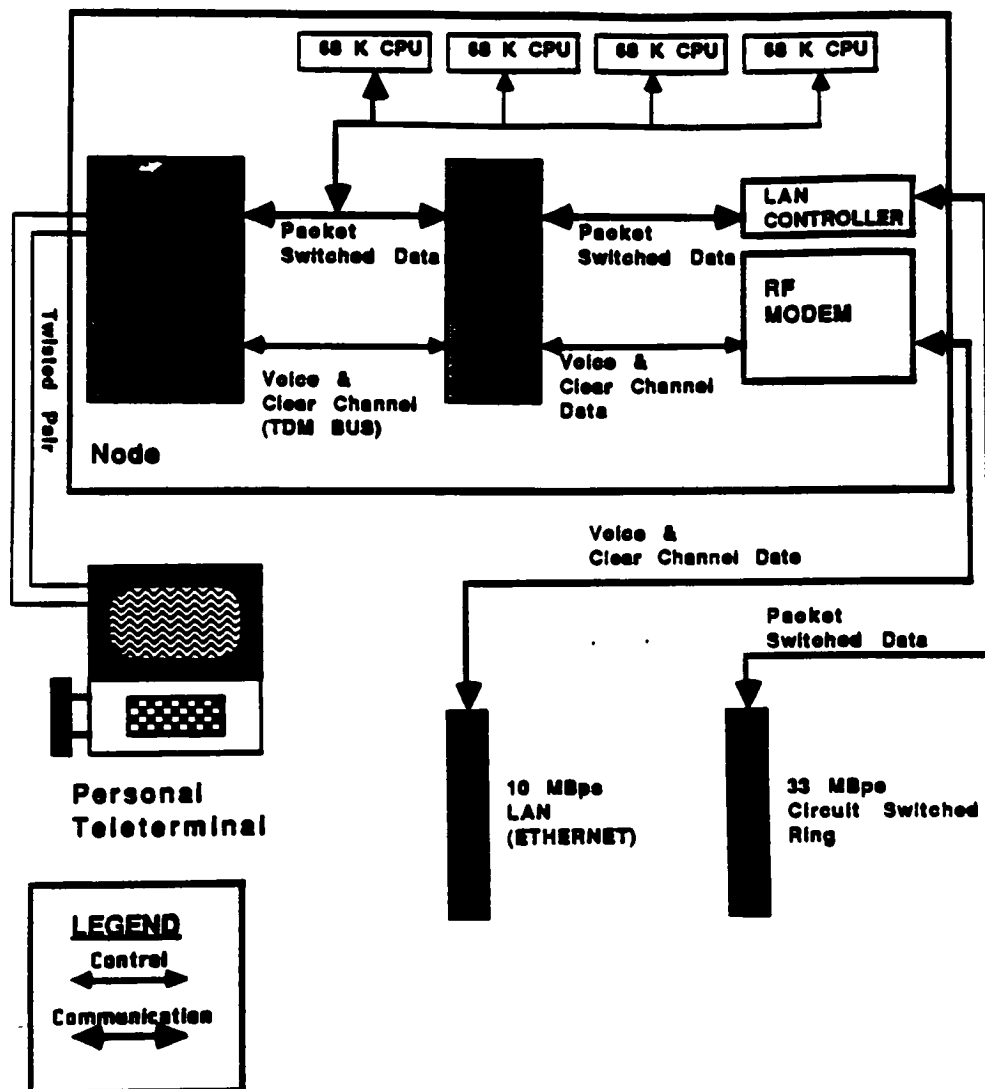
The second major problem the number of different packet formats used within the system. Transport of packetized data from a teleterminal is done in one format. When control and packet data are combined for transport to other PNX nodes via the Data LAN, another format is used. A third format is used when circuit switched digital voice and data are changed to a packet format for transportation to other nodes on the Voice LAN. A single format would make the system more efficient and lessen the number of conversion points for failure.

The last problem is the use of packet technology for transporting digital signals on the Voice Bus. The system would work more effectively if voice and digital data were handled in a different manner. We would like to see either the baseband bus used for the Voice LAN or

replace the baseband bus with a broadband bus that could be dynamically allocated into separate channels for the various circuit switched data signals. Although baseband technology is simpler, cheaper, and the less sophisticated of the two, the nature of a broadband LAN is more conducive to circuit switched technology. The modulation of several channels for digital voice, data, and possibly video allows for the dedicated setup of a channel for the duration of the connection. This is more in line with the philosophy of fourth generation switching design and how a true circuit switch connection should be handled.

The Ztel PNX was the first fourth generation digital PBX that incorporated the notion of a semi-dual bus configuration as a design concept inherent to its architecture from stage one. The major marketing problems that Ztel experienced in releasing the PNX was the lack of working capital to mount a major campaign against the the big three digital PBX manufacturers, and the premature release of the product with capabilities that had not been fully developed. Ztel was a new company founded solely on the PNX. The company was not diversified enough with a product base to sustain itself in this highly competitive area. The Ztel Corporation is presently operating under Chapter Eleven protection and looking for an infusion of capital from some outside source. To date, this has not occurred.

DIAGRAM 4.2.1.2.1
CHC ROSE SYSTEM ARCHITECTURE
 ****Copied from [BCR83]****



4.2.1.2. The CXC Rose

CXC Corporation is another new company that ventured into the lucrative PBX market. They immediately followed Ztel's announcement of a dual bus digital PBX design with one of their own. The CXC Rose was very similar to the Ztel PNX. The Rose also was designed around both a circuit ring and packet bus architecture. The networking architecture of the Rose was more complex than the PNX though. (See Diagram 4.2.1.2.1) The Rose uses a broadband ring for transferring circuit voice and digital data both internally and to other nodes. Using a broadband bus for both circuit voice and digital data did keep in line with the underlying philosophy of fourth generation PBX systems. The broadband bus of the Rose has a capacity of 50 M bps and can be dynamically allocated to accommodate varying transmission speeds. The channels are allocated in 8 k bps bands that can be combined incrementally up to 24 M bps throughput for any one signal.

Channels are setup and held for the duration of a connection using frequency division multiplexing. Bandwidth can be assigned based on application need which is crucial to many processes, i.e. graphics for example. Presently, only 33 M bps of the circuit ring can be used for dynamic allocation, with the rest being reserved for future applications. CXC Corp. has pledged that the Rose will be completely compatible with any of the IBM LANs, and the 17 M bps of reserve is going to be used for this IBM compatible philosophy [CXCC86]. The underlying concept of the Rose is user friendliness. We feel that CXC would be better off developing their dual bus architecture and not concerning itself with what other companies will do, even if it is IBM.

The CXC Rose also has a 10 M bps baseband Ethernet LAN tied into its architecture. This Ethernet LAN is used as a packet bus to transfer packet and control signals between nodes. By using the baseband packet bus for packet data and control signals, the resources of the broadband bus are free for the bulk of the other transmissions.

The only device that can actually generate packet data is the CXC Rose teleterminal. The packet format generated by the CXC Rose teleterminal is capable of transmitting both voice and data simultaneously at 192 k bps. The packet uses 32 k bps for overhead and signaling, 64 k bps for data, and the last 64 k bps for either voice or data. No interface devices have been designed to attach to other asynchronous data devices that take advantage packet format over the packet bus. The baseband Ethernet band is definitely underutilized unless the system is configured with a large number of CXC Rose teleterminals. The thrust of the CXC Rose is the utilization of the broadband bus. There is some talk that the packet bus could be entirely removed since the last 17 M bps reserve on the baseband could replace this function [FRAN83]. If this happens the CXC Rose will mimic the CBX II except for the size of the bus, and will be a step backward in our estimation.

The CXC Rose is expandable in a building block fashion with the node as its distribution backbone. This makes the CXC Rose fully distributed, a prime consideration of fourth generation development. The CXC Rose handles digital station devices such as digital telephones and terminals in the broadband circuit switched mode. CXC Corp. has also developed their own teleterminal that transmits in packet format and utilizes the Ethernet. No analog transmissions are allowed forcing older

analog station devices to digitize their signal or be replaced. This is a definite drawback.

Each node in the system is capable of handling 192 station devices in a nonblocking manner. The various station devices are connected to each CXC Rose node through a Station Interface Module (SIM). Each SIM can accommodate up to 16 ports, one for each station device. Each port can handle packetized transmissions for simultaneous digitized voice and data at up to 192 k bps over standard twisted pair from a teleterminal or 64 k bps digital PCM signals from any other type of station device. The SIM acts as a first level switch. Each port determines the kind of signal arriving. If a digital PCM signal arrives, the port routes it to the broadband bus. If the port senses a packet arriving, the packet is broken up with the digital voice part going to the broadband bus and the data and control signals routed to the packet bus. Other packet control information from the CPUs in packet form joins the packet stream.

Both the broadband and packet busses from the SIMs transport their signals to other devices through the Network Interface Module (NIM). The first job of the NIM is to switch any data streams with destinations within the node to the correct SIM. The second responsibility of the NIM is to repacket any packetized control or data information destined for other nodes before sending it along to the LAN Controller. The NIM also allocates bandwidth in 8 k bps increments for transmission to other nodes of the broadband bus. Up to 64 nodes can be interconnected to give a total system capacity in excess of 12,000 station devices. The system programming is done in Pascal. CXC Corp. stresses the

extended life time of the ROSE due to its modular architecture and flexibility to adjust to future technological introductions in the digital PBX field [CXCC86].

We feel one major problem the CXC Rose has is the use of the broadband bus for the majority of voice and data transmissions. The CXC Rose still handles most data via circuit switching. Packet data is limited if there is a lack of CXC teleterminals. The baseband bus does handle all control signaling, but is this really a full utilization of the capacity of a 10 M bps Ethernet? It seems that the baseband Ethernet has specifically been placed in the architecture to insure compatibility with IBM LAN products [FRAN83]. This could easily change and the present design could take on new looks if IBM's philosophy changes. We feel there is too much fence sitting on the part of CXC Corp with the Rose. CXC Corp. should adopt its own philosophy and not hedge on what others are doing.

Another problem with the CXC Rose is the lack of a single packet format for the entire system. There are two packet formats in this system. We feel this is one too many. There is no reason for one packet format internal to the system and another for transport on the baseband bus. CXC Corp. could eliminate one protocol conversion if they took this approach.

The last problem is the lack of device accommodation. CXC Corp. must develop a more imaginative approach to incorporating a broader range of station devices into the Rose architecture. More interfaces are needed for data devices to gain access to the packet bus. New line card circuitry could be added which would allow a wider variety of station

device transmissions access to the Rose architecture.

CXC Corp. has several good features. They have had more success in getting their product marketed than Ztel. Sony Corp. has backed CXC Corp, buying 20% of CXC's stock. CXC Corp, has also secured a number of multi year distribution contracts with some of the new regional Bell Operating Companies [EDWA86].

4.2.2. Add On Packet Enhancements To Existing Products

While Ztel and CXC Corp. were developing integrated digital PBX systems from the ground up, the two largest digital PBX manufacturers took a different approach. Northern Telecom and AT&T elected to follow the approach of taking an existing proven product and introduce an add-on packet LAN enhancement. Northern Telecom did this by developing a packet network called LANSTAR that could interface with the SL series of digital PBXs, or stand alone as the DV-1. The DV-1 was an entirely new member of the Meridian product line that could act as a stand alone circuit/packet switching system for small users.

AT&T, on the other hand, decided that the market seemed very unclear regarding the acceptance of integrated circuit/packet systems. AT&T decided that they should enter the market with some caution, to test the waters. While AT&T did not want to be outdone by their competition, they also did not want to invest huge sums of money in the develop a new product line if it proved unmarketable. AT&T decided to integrate two of their proven products. These were the System 85 digital PBX and the Information Systems Network (ISN).

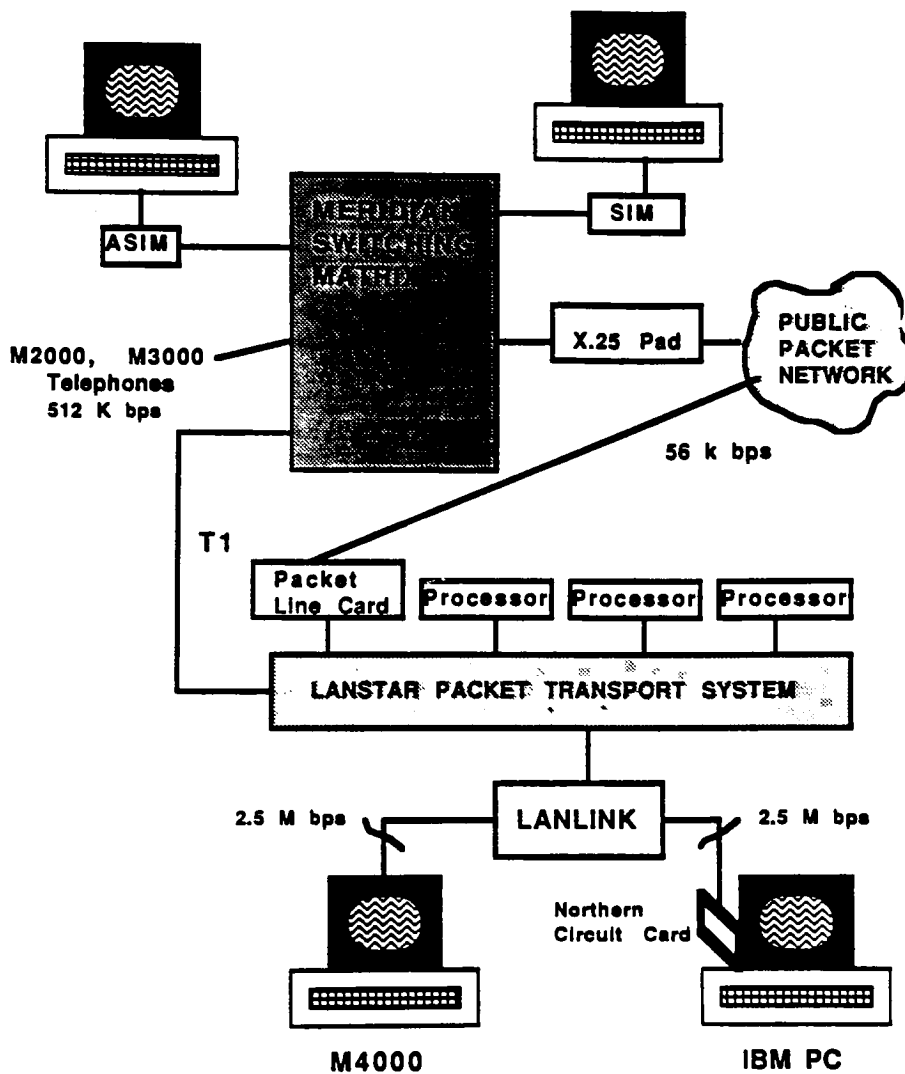


DIAGRAM 4.2.2.1.1
Meridian SL
and LANSTAR

4.2.2.1. Northern Telecom's Meridian Series

In February 1985, Northern Telecom, introduced three new enhancements to its popular SL product series and changed the name to the Meridian. These new enhancements were added as architectural extensions to the SL architecture [DATA86]. Northern's first enhancement was a loosely coupled processor system based on the Motorola 86000, a 32 bit processor. The second was a high speed packet network that used dynamic bandwidth allocation techniques, called LANSTAR. LANSTAR is a packet LAN that interfaces with any of the Meridian digital PBX products via a T1 connection. The third enhancement was a data communication device called the LANLINK that interfaced to the packet network for high speed data transmissions between data terminals and the packet network. All three could be incorporated into the existing architecture of any of the previous SL products [NOR185].

The loosely coupled multiple processors run under a true distributed operating system. This method provides load balancing and optimal response time. As capacity demands increase, processors and memory can be easily added to upgrade the system. If a single point in the system fails, another processor can automatically be assigned the task, based on which processor has spare time.

The packet network called LANSTAR can be added as an extension to any of the Meridian SL PBX products. Unfortunately, we characterize this packet network as nothing more than a packet network connected over T1 transmission lines to a Meridian SL product. The phrase "Meridian over T1" is actually forbidden at Northern Telecoms companies

[BCRS85]. As in the System 85/ISN marriage, the approach taken by Northern Telecom is nothing more than accommodation. The system, however, is a little more complex. (See diagram 4.2.2.1.1 for a system overview)

The LANSTAR packet network has a 40 M bps bus that can accommodate IBM PCs or Northern's M4000 workstations directly through LANLINKS at 2.5 M bps over standard twisted pair wire. Other data devices connect through LANLINK Interface devices and RS-232 connectors at up to 56 k bps. If IBM PCs are used, a special Northern circuit card can be plugged into one of the LANLINK slots. The IBM PC can transmit at the 2.5 M bps over the twisted pair, the same as the M4000 work station. The special card costs about \$700 [DATA86]. A LANLINK is Northern's version of a concentrator. Each LANLINK device can connect from 1 - 12 asynchronous data devices to the LANSTAR packet network.

The 40 M bps LANSTAR consists of a dedicated 10 M bps channel for asynchronous data, in packet format. The remaining 30 M bps is for voice and synchronous data on a priority basis. Channels are allocated dynamically. The 30 M bps portion can be used for additional asynchronous transmissions in packet format on an as-available basis. Voice if transmitted does not use a packet protocol. The packet bus can be installed with the circuit switch or as a remote system connected to the switch via T1 carrier links [NOR384]. The Meridian series of digital PBXs can accommodate data devices at lesser speeds through the circuit mode capabilities of the SL switch. Using an Asynchronous Interface Module (AIMs) or Asynchronous/Synchronous Interface Module (ASIMs), data devices connected through the circuit switch of the digital PBX can

communicate with other data devices connected to the packet bus directly, or via the T1 trunk connection at much slower speeds.

A special packet line card attached to the packet transport system allows 64 data terminals, X.25 access to public or private packet networks at speeds to 56 k bps. A similar X.25 PAD attached to a Meridian series digital PBX can only give access for 10 - 12 devices at 19.2 k bps. The packet transport bus not only gives more users access, but at far greater speeds [NOR185].

A unique aspect of the LANSTAR packet transport bus is its capability to stand alone as a small business communication switching system. This system can support up 35 data users and 100 voice conversations concurrently. [BCRS85] Here the true capabilities of the dual bus architecture are finally realized. When the LANSTAR packet network is used in this fashion, it is called a DV-1. The DV-1 is designed for small system users. No other manufacturer, with the exception of Telenova, has offered a dual bus system aimed toward the small business user. As a stand alone system, the DV-1 can be hooked to CO trunks, used behind Centrex, or behind a Meridian SL product.

The DV-1 can handle all the same data devices that the LANSTAR packet bus can in a similar fashion. By adding an Analog Link, a dedicated voice band for circuit switching is established. In the DV-1, the 40 M bps bus is divided differently. Two equal 20 M bps busses are set up, one for digital circuit voice and circuit data and another for packet data. All voice devices are connected to the voice bus through an analog link, while all data devices are connected via LANLINKS or LANLINK Interface Modules. The same speed requirements apply to the

DV-1 as described for the packet bus enhancement. (See Diagram 4.2.2.1.2 for a schematic of the DV-1.) The DV-1 offers a choice of either Northern Telecom's XMS operating system, or a Unix based operating system running concurrent DOS. A 20 or 40 M byte hard disk is attached with a 200 M byte hard disk available at some future date [EDWA86], [NOR851], [NOR852].

Northern Telecom has clearly introduced some very important enhancements to its line of Meridian products, but nothing that has not been done before. Rolm has featured the capability of dynamic bandwidth allocation for years with the CBX II. AT&T made their Information Systems Network, packet network, completely compatible with the System 75/85 digital PBXs a full year before Northern Telecom introduced the LANSTAR. The one single aspect that does stand out is the DV-1. The stand alone capability and the targeted market is a major first by a recognized PBX manufacturer.

Another thing the Meridian series has shows is the willingness of the largest digital PBX manufacturer to build on its installed base and compete on all levels for PBX market shares. No matter what the level of competition, Northern has shown it will use its vast resources for product development to insure their number one ranking in this area. The small companies will find it very hard to compete for any length of time even in unique niches of the market when this attitude is taken.

ISN ARCHITECTURE

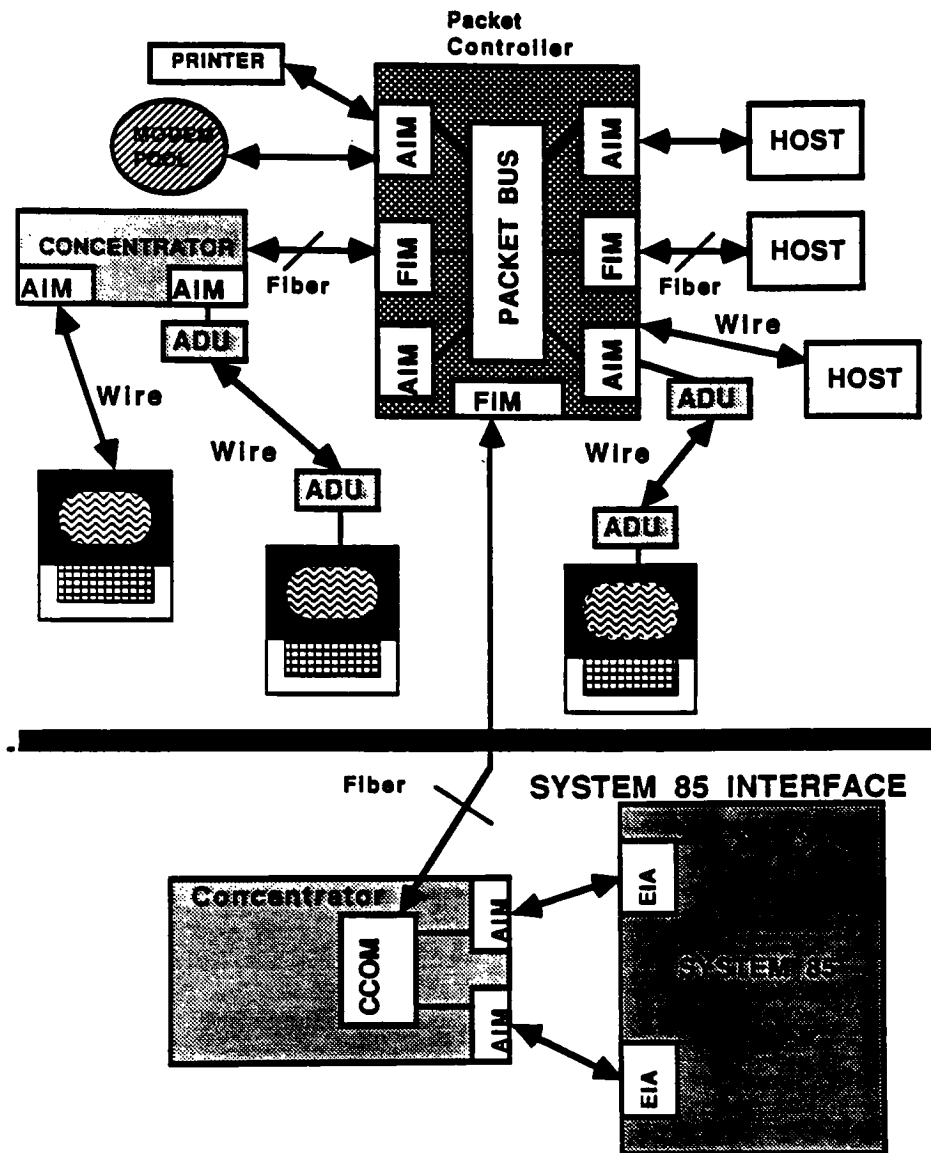


DIAGRAM 4.2.2.2.1
ISN/SYSTEM 85

4.2.2.2. *The AT&T System 85/ISN Marriage*

Release 2 of the System 85 digital PBX was announced in the fall of 1984 [DATA86]. The System 85 was AT&T Information System's solution to the customer based digital switching system that incorporates integrated voice and data in a circuit switched mode. The System 85 uses standard digital processing techniques such as PCM modulation through a TDM bus for switching both voice and data in a circuit switched environment. One of the major new additions to the System 85 announced in 1985, was the capability to interface with ISN.

ISN is a baseband packet LAN, that uses a star topology. It was introduced in the summer of 1984 [DATA86]. The approach that AT&T uses to connect the two systems is to treat the System 85 as just another data device that could interface to ISN. This requires no major architectural changes to either system. (See diagram 4.2.2.2.1 for a schematic of the the ISN architecture and System 85 connection.) The only requirement needed was the development of an interface between the two systems. No architectural changes were needed to either system using this approach due to the component nature used as the basis for the ISN network architecture. If we understand how ISN works, the System 85 interface philosophy is quite simple.

There are five major components to ISN. They are the Packet Controller, Asynchronous Data Units, Concentrators, Asynchronous and Fiber Interface Modules, and various transmission media. Since ISN is a packet network, the device that performs the function of packetizing the incoming asynchronous data stream is very important. The Asynchronous

Interface Module (AIM) performs this packetizing task. No mention is made of a Synchronous Interface Module, but a protocol can be attached to any synchronous data device and then AIM can be used. An AIM is the principle interfaces between the asynchronous hosts and terminals and the Packet Controller [ATT285]. Each AIM has either four or eight ports for data device connections.

All packetizing of information performed by AIMs occurs at the Concentrator or Packet Controller Level. Each AIM receives asynchronous EIA standard signals from the various hosts and terminals and formats these signals into packets, checks their parity, enforces protocols, and buffers the packets for transport to the Packet Controller. The Packet Controller then places these packets on the packet bus. The AIM performs the reverse process of converting packets back to an asynchronous EIA standard signal in order to to transmit back to the the asynchronous terminals and hosts. The AIM is never located at the station device level which consequently requires different transport modes and limits the effective speed of the system [ATT186].

A Concentrator can be used to increase the number of asynchronous devices that can be connected to the Packet Controller via a port. Multiple AIMs are located on the Concentrator to connect up to 40 asynchronous devices to the Packet Controller. The packetized information converted by the AIMs in the Concentrator are multiplexed into a single signal that is transported to the Packet Controller over a fiber optic medium. This is accomplished by combining the signals through an optical multiplexing device called the Connector Common Module (CCOM). The optical fiber link provides fast access to the packet controller at a dis-

tance of up to 3 km and also keeps cabling at a minimum [ATT286].

If a Concentrator is used as the interface to the Packet Controller, an AIM on the Packet Controller must be replaced with a fiber optic port device called a Fiber Interface Module (FIM). The FIM consists of the opto-electronics required to receive the multiplexed signal sent via the optical link from the Concentrator. The FIM demultiplexes the signal, convert it back to a packetized format, and then interfaces the packets with the ISN bus. In the reverse direction the FIM must be able to buffer and multiplex packets from the ISN bus over the optical link to the Concentrator.

If a terminal or host is located more than fifty feet from a Concentrator or Packet Controller AIM, another device the Asynchronous Data Unit, ADU, must be added to increase the transmission distance of the asynchronous data signals sent over the twisted pair. The ADU converts the EIA signals to a different signal with a lower voltage. This lower voltage signal to be transmitted over a greater distance, passes through noisy environments, allows data transmissions over standard house wire carrying both data and analog signals simultaneously, and allows voice and data to share common connect equipment. If an ADU is employed on one end of a twisted pair, it must have a similar ADU at the other end to return the signal back to the EIA format before entering an AIM [ATT386]. One question we might ask is why not just incorporate the logic of the ADU into an AIM similar to an EIA port as discussed below.

The physical connection of the two systems is accomplished through a System 85 trunk to a ISN Concentrator. One or more of the trunk ports associated with System 85 are replaced by an EIA RS-232-C quad

port. Resident on the EIA quad port is a built-in ADU which not seen in diagram 4.2.2.2.1. This is necessary to meet the requirement that ADUs appear in pairs if the Concentrator is located more then 50 feet from the System 85 unit. By using this technique, the System 85 now becomes nothing more then another data device connected to ISN via a Concentrator.

The System 85 and ISN integrate quite easily in both directions. The System 85 functions as a digital PBX connecting all voice and data transmissions in the circuit switched mode. A user can request a connection to ISN by dialing the number of the EIA trunk that connects to the Concentrator. More than one EIA port must be connected to a Concentrator if the system uses multiple baud rates since each EIA trunk is permanently set to 300, 1200, 2400, 4800, 9600, or 19,200 bps. The user must dial the correct trunk to synchronize the speed between the two devices. After the user is connected to ISN, a prompt appears for the specific device number and the connection is complete. A similar set of steps is required if the process is initiated from ISN to a device connected to the System 85 [ATT186].

The first question we must ask ourselves is if this is truly what the integration of LAN and circuit switched technology is about? We believe this is just a convenient way to let asynchronous devices connected to the System 85 also access devices connected to ISN. Connections can be made by devices across the two systems, but two data devices attached to the System 85 can not use the packet facilities of ISN to connect to each other. The same is true for two devices connected to ISN wishing to use the circuit switched mode of the System 85.

Another problem is the multitude of transmission and switching modes required to interface the two systems. A call initiated on the System 85 side starts out at some asynchronous data speed between 300 - 19,200 bps. This signal is changed to 64 bps PCM for switching purposes to the appropriate EIA trunk. Then the signal is redone as a low voltage EIA signal for transport to the Concentrator. At this point the signal is packetized and sent over high speed fiber to the Packet Controller, and then placed on the packet bus. At the destination data device, the packet is finally converted back to an asynchronous data stream possibly going through the same conversion technique mentioned above. By using so many data transmission formats, and protocol conversions, this system leaves multiple points for failure. There is no streamlining of the architecture, a major point in fourth generation design.

The last problem we see is one of cost. Each system is sold separately, and neither is cheap. Unless one already owns both systems and would just like the convenience of interconnection, we wonder if such a system is cost effective. RIT has elected not to interface the two systems. A separate data switch is used to access ISN. We see the approach that AT&T has taken as viable, but it does not really address the true nature of packet/circuit switch technology in a unified PBX system.

4.2.3. The Best Of The Rest

Of all the other digital PBX vendors who have ventured into the integrated fourth generation market, Rolm and GTE stand out as the leaders. Their CBX II and OMNI S switches have been widely accepted

by the telecommunications community if the number installed is used as an indicator. The CBX II was hailed as an innovative leader in PBX design, with the promise of a broad range of applications based on the division of its huge bus bandwidth. The CBX II has lived up to all its expectations as a voice/data circuit switch, but packet applications that were promised have not materialized. The OMNI S with the PD-200 Packet System is the only dual bus architecture we feel has meet or exceeded true fourth generation design expectations. The approach that GTE has taken with the OMNI S has fully utilizes X.25 packet standards and is an ideal networking solution using an integrated digital PBX.

4.2.3.1. *Rolm CBX II*

The Rolm CBX II, (Computer Branch Exchange), was announced in November 1983 [DATA86]. The CBX II features a distributed architecture that uses a fiber optic link system and modular expandability to support from 16 - 10,000 users across a wide range of devices. In addition to the normal analog and digital voice switching provided by a digital PBX, the CBX II's new architecture allows for new data switching capabilities and flexible allocation of bandwidth which opens the door for many new applications. (See diagram 4.2.3.1.1 for the CBX II system architecture.)

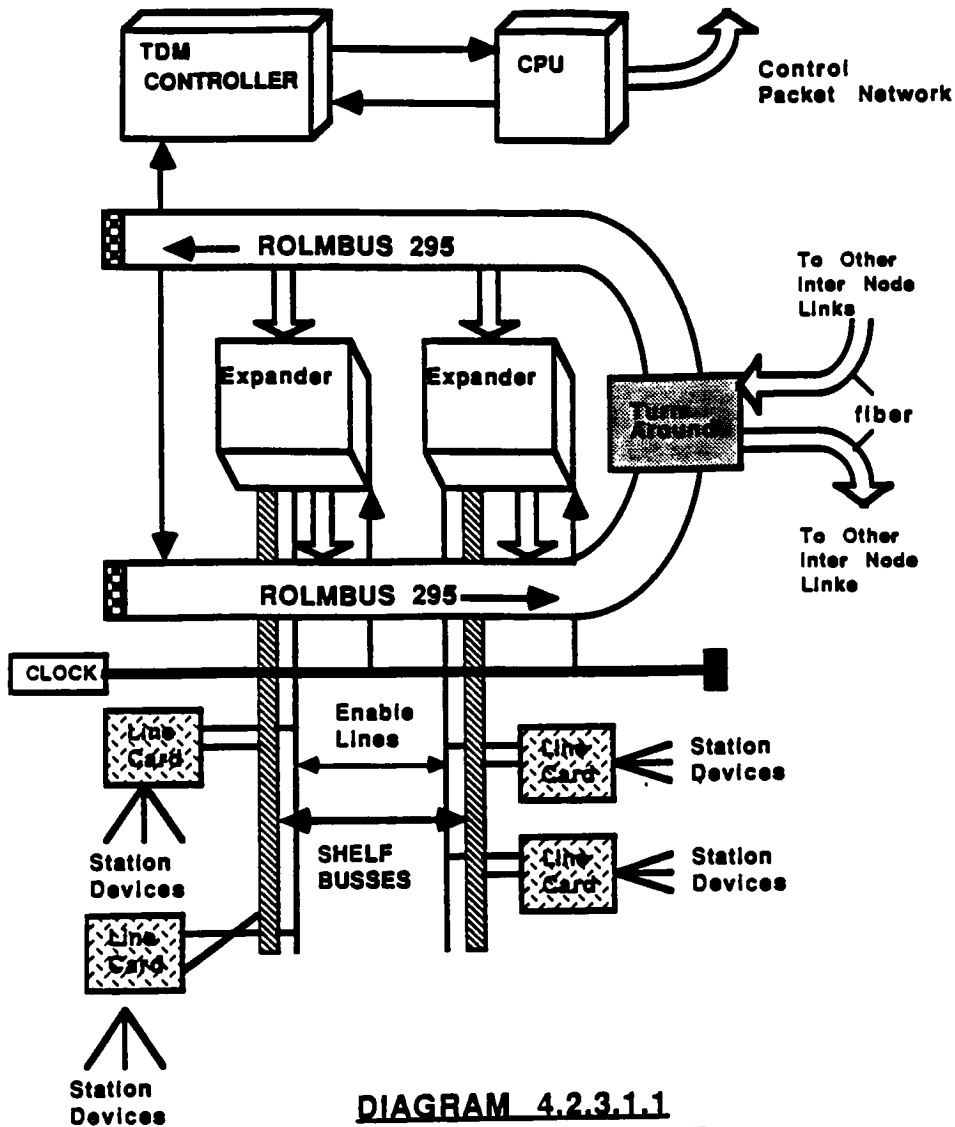


DIAGRAM 4.2.3.1.1
CBX II NODE ARCHITECTURE

Each communications channel can carry data at 192 k bps, and multiple channels can be combined to increase the data rate if needed. These capabilities are built around a high bandwidth bus called the ROLMbus 295. This bus has the capacity of transmitting 295 M bps, and can be configured into a number of noninteracting segments. The CBX II's architecture is also distributed, using a node building block structure. Three new modes of data handling, along with ordinary voice/data connections, upgrade the CBX II. These three new modes are supermultiplexing, shared access, and broadcast [JOHN84].

The first mode, supermultiplexing, lets the ROLMbus 295 be divided into transmission channels of up to 37 M bps. These blocks can be used for main frame to main frame, or LAN to LAN data transmissions. This allows the CBX II to act as a gateway or bridge between geographically disperse LANs over a three - four mile range. The CBX II can also be used to connect to X.25 private networks using the Rolm X.25 interface. Video transmissions, 1.5 - 2.5 M bps, would only take up a small amount of the total bandwidth as would burst transmissions between computer hosts. Because the bandwidth can be torn down or assembled as needed, the possibilities for uses are endless [KASS85].

The second new feature called Broadcast allows a single data device to simultaneously transmit to more than one receiving data device. Up to 74 M bps can be assigned for this purpose. This is done by using non-enable card commands. CBX II line cards are not enabled by signals sent across the TDM bus, but by separate enable lines from the Expanders to the line card. These enable signals can instruct multiple circuit card to listen to the common source band. Broadcast mode is

especially useful for transmitting switch system updates to multiple workstations, global timing signals, automatic program load routines, and mail/document distribution [JOHN84].

The last new mode, shared access, allows a number of station devices to share a common bandwidth using some arbitration process, similar to packet system. This differs from broadcast since only one device can receive transmissions at a time. The arbitration process can be centralized, polling, or token passing. This is achieved by plugging in packet controller into the ROLMbus 295 [DATA85].

The architecture of the CBX II is unique because of the single high bandwidth bus and the distribution of control within each node. All Control of addressing and assignments has been moved one step away from the CPU to the shelf level. (Refer to diagram 4.2.3.1.1.) Each shelf, six per cabinet/three cabinets per node, has its own controller called an Expander. Each shelf has its own backbone bus, the ROLMbus 74. The Expander contains a complete connection table of addresses for all the voice and data devices connected to the shelf as well as the addresses of the other Expanders within the node. Each Expander uses this information to generate addresses for transmission transfers. This eliminates the need for the constant flow of address information between the control CPU and the line circuits typical in a digital PBX. A savings in overhead and bandwidth are realized, leaving more bandwidth for information to be transferred [KASS85].

Line circuits are enabled by enable lines instead of an enable signal transmitted over the bus via an interrupt cycle. This allows any line circuit card to be plugged into any socket on the shelf. For a

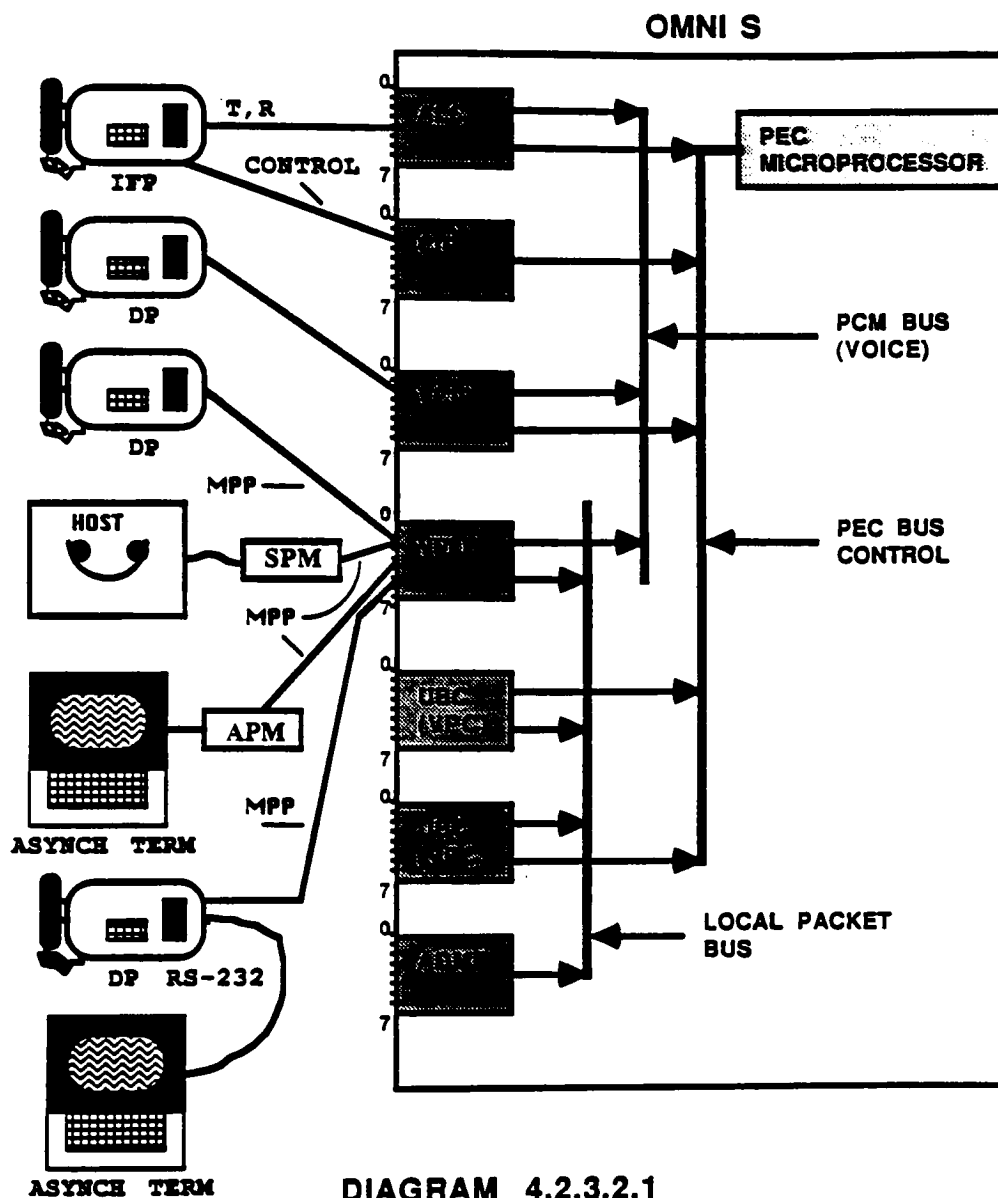
typical data transfer, the shelf Expander selects the appropriate line circuit enable line. The enabled card places the data on the shelf bus which carries it to the Expander. The Expander can either place the data on the same shelf bus if the other device is connected to the same shelf, or place the data on the transmit band of the ROLMbus 295 which acts as an intra shelf or inter node bus [JOHN84].

The ROLMbus 295 still employs the standard loop system common to most digital PBX systems. The CBX II uses a transmit and receive side on the bus loop to establish full duplex conversations. The turnaround in the CBX II has a dual purpose. First, it absorbs the information from the source bus and places it on the destination bus if the connection is within that node. This is the common function of all turnarounds in a loop bus as discussed in chapter three. The second function of the turnaround is to act as a transportation mechanism to other nodes. The turnaround need not place the same information on the destination bus that it received from the source bus. The turnaround is used as a router to other nodes, or as a receiving buffer from other nodes. The turnaround can do this because it contains an address table of other Expanders within that node and the address of other nodes in the system. If the turnaround notes that a piece of information has an address code of another node, the information is placed on a fiber optic or coaxial link through a Time Slot Interchange (TSI), and sent to other nodes [KASS85].

Ordinarily, a loop bus architecture requires two time slots, one for receiving and one for transmitting. If the message is destined for another node, this will effectively remove the need for the destination

side time slot. Consequently an incoming message from another node in the system can be substituted into that now empty time slot. Therefore is a two-way conversation will take up only one time slot if routed to another node in the system. This doubles the efficiency as compared to tie lines [JOHN84].

The philosophy behind the CBX II is a large system capacity, modularity, and new modes of data handling capabilities. The CBX II can grow for present and future application expansion, and presents a viable solution to the problems of integration through a common switching architecture. Rolm has not yet introduced the add-on packet controller that was promised so that part of the ROLMbus 295 can be used as a packet system. Since the buyout by IBM, the announcement of the necessary hardware to implement a packet bus over a part of the ROLMbus 295 has been put on hold. This is because of the indecision of IBM to get involved in competition with its existing LAN technology. The CBX II still is a unique attempt by a major competitor in the digital PBX market to provide a truly integrated fourth generation product. The CBX II has been widely accepted by the business community as a workable PBX architecture. IBM has recently announced that the CBX III will soon be introduced.



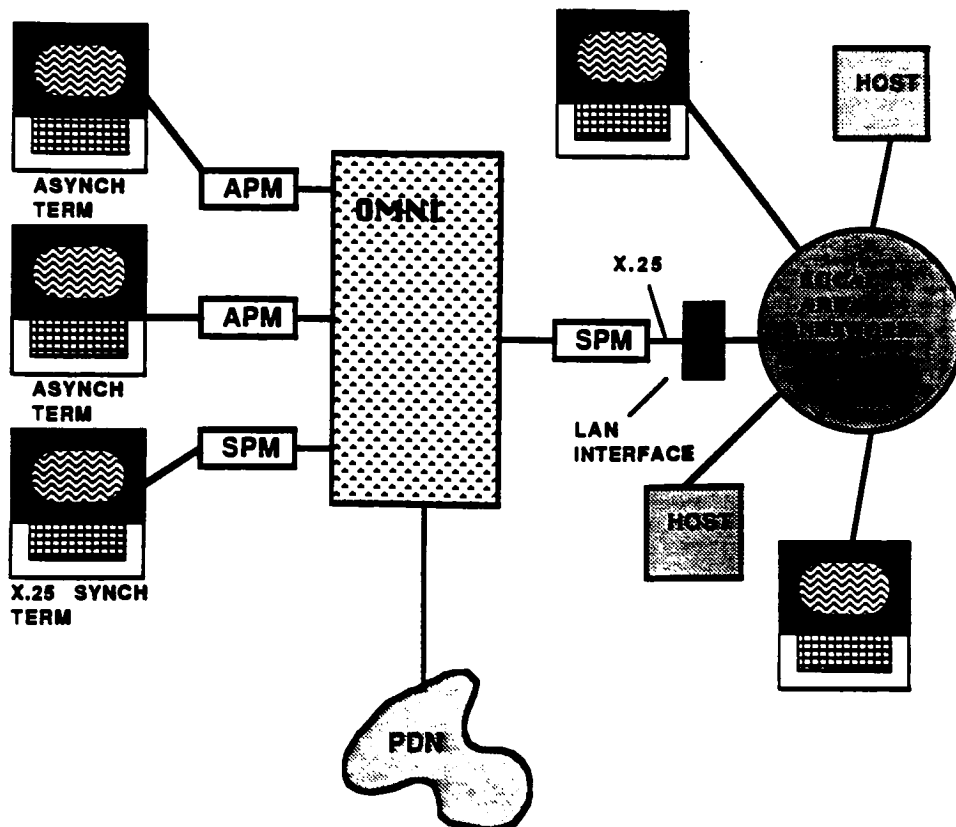


DIAGRAM 4.2.3.2.2
TERMINAL CONNECTION TO LAN

4.2.3.2. OMNI S Series Of Switches

The GTE OMNI S series of digital PBXs is the first legitimate attempt by a major PBX manufacturer to implement a dual bus architecture in a digital PBX product. The OMNI S series of PBX's is the first true implementation that incorporates in a serious manner the objectives of dual bus architecture. The OMNI S series of switches evolved from the earlier GTD digital PBX, which has over three million lines in service. Introduced in late 1982, the first OMNI was installed in early 1984 and over 13,000 have been installed to date [DATA85].

High speed voice and data are transmitted over standard twisted pair wire. The OMNI S provides separate highways for voice and data calls. The voice traffic is handled on a dedicated bus using PCM, and the data traffic is handled on a separate dedicated packet bus using packet switching technology. In essence what GTE has done is incorporate a packet data switching system into the architecture of a digital PBX. Because of the separate nature of the two switching systems, voice calls do not block data calls and data calls do not block voice calls. If either of the two busses goes down, the other remains operational. The two systems can be administered separately for maintenance and control. Engineering constraints due to combined traffic of voice and data can now be dealt with as they occur. The dual bus design supports faster, and controlled growth.

The OMNI S II/III can handle up to 1,024 or 2,048 circuit switched calls respectively. If the PD-200 Packet Data Transport System is active, it can handle up to 255 simultaneous virtual connections. All OMNI S series switches have the necessary backplane circuitry built into

the system to support the PD-200 Packet Transport System. The owner of an OMNI S can bring up the PD-200 Packet Transport System at any time if desired. The backplane for the PD-200 Packet Transport System is built into all OMNI S switches. The unit can be bought initially as a digital voice/data circuit switch and to upgraded to the packet system with the addition of some line circuit cards and the attachment of Asynchronous/Synchronous Packet Managers/Packet Controller [SAG286].

The design concept behind the OMNI S is based on a very simplistic architecture. (See Diagram 4.2.3.2.1 for OMNI system architecture.) A Mini Packet Protocol (MPP) scheme is used that is X.25 compatible. This MPP is an eleven bit packet consisting of two address bits followed by eight data bits and a single cyclical redundancy check (CRC) for error checking. The whole process is done in an end to end fashion by using Asynchronous or Synchronous Packet Managers, APM or SPM respectively. The Packet Managers are nothing more then protocol converter, taking the asynchronous or synchronous input signals and converting them to MPP format. Both the APMs and SPMs are designed as stand alone units that can connect to various vendor data devices and hosts. They are also built into all OMNI S digital phones, terminals, or workstations as part of their hardware. Since the Packet Managers are not part of the OMNI S circuitry, but are located right at the station device level, the system is truly end to end. All voice, data, and control signals are transmitted from the sending devices in MPP format and interpreted at the line circuit level. This allows for true simultaneous voice and data over the same twisted pair. One Asynchronous/Synchronous Packet manager is attached to each data device connection and can be located

up to 3,000 feet from the actual Omni unit [MIC285].

Three busses are employed; the Local Packet Bus (LCB), the PCM Bus, and the Peripheral Equipment Complex Bus (PECB). They respectively transmit data signals in MPP form, voice in standard 64 k PCM, and control signals to all line circuit cards. Also, four line circuits are employed. These give the OMNI S the ability to accommodate a variety of station devices ranging from analog telephones to teleterminals. Standard tip and ring analog telephones are connected to an Analog Line Circuit (ALC) over a single twisted pair for immediate transfer to the PCM Bus. Integrated telephones are accommodated by having one pair attached to an ALC to transmit the analog voice signal, and the other pair to a Control Interface Processor Card (CIPC) to transmit the digital feature signals. Digital telephones are connected directly to a Voice Control Line Circuit. Any digital telephone with an APM attached to it, or OMNI digital telephone unit with a built in APM can connect directly to a Voice Packet Line circuit (VPLC). All transmissions to the VPLCs through APMs use the MPP format. This includes voice, which is translated back to PCM for transmission over the PCM Bus. If the digital phone has a data option and a built in RS-232 connector, an asynchronous data terminal can now transmit through it [MIC187].

We believe that the OMNI S with the PD-200 Packet System is the clear front runner in dual bus architecture on the market today. The first clear advantage is the use of four different line circuit cards designed to handle any station device that a user might need. The correct switching mechanism, packet or circuit is assured for each station device connected to the system. This mechanism is consistent in an

end to end fashion and also transparent to the user.

To achieve this the OMNI S designers have eliminated the need for protocol conversions, by using the MPP format which is entirely X.25 compatible. There is no need for a different packet format for voice, data, and control as exhibited by all the systems discussed to this point. Also the use of the MPP format has made networking design easy and reliable based on industry accepted standards using X.25 technology. (See Diagram 4.2.3.2.2 for a sample network design.) Using a standard building block architecture, there is virtually no application or network solution that can not be solved.

Despite the appealing architecture of the OMNI S for integrated voice/data, GTE has not been able to show a profit in its switching division. Whether this is from poor management or other factors, we do not know. We feel that it is partly due to the fierce competition for shares in the PBX market. Recently GTE merged its switching division with Fujitsu America. Regardless of their economic problems, GTE's Omni S dual bus PBX has come the closest to fulfilling the requirements of a true fourth generation integrated dual bus architecture.

CHAPTER V

THE FUTURE PBX SWITCHING TECHNOLOGY

5.0. INTRODUCTION

We speculate that private communication systems based on optical technologies will be the basis for future PBX systems. We also feel that the term PBX is a misnomer. Private Communication System (PCS), would be much more appropriate due to the variety of services provided. The traditional transmission medium for PBXs has been twisted pair copper wire. Twisted pair has proven to be a safe, reliable, and a cost effective medium for voice and limited data transmissions for many years. The problem that has developed is the inadequacies of twisted pair as it applies to many of the new broadband services being delivery to the office desk top. These include information retrieval systems, graphics applications, video conferencing, word processing, voice mail, closed circuit tv, and facsimile to name a few. We believe that there are five major objectives that the next generation of switches must possess. They are resource sharing, ability to handle a wide range of bandwidths, acceptance of a heterogeneous mix of traffic, being able to adapt to new features, and total flexibility in a range of different environments. We feel optical transmission and switching technology are perfectly suited for the application of these goals.

Any future developments in PCS design will closely parallel research and advancements in optical transmission and switching technology. This will involve a quantum leap from the fourth generation

dual bus architectures presently reaching the marketplace and discussed in Chapter Four. A reasonable time frame for such optical systems to appear will probably be in the latter part of the 1990's. Presently optical transmission technologies using fiber optic lines are well developed and entrenched in the telecommunications industry. Unfortunately optical memories, processors, and switching devices necessary to complement fiber optic transmissions for an all optical switching system have lagged behind in development and are not presently available. The primary challenge facing researchers today is the development of component technologies enabling the enormous data and speed capabilities of optical systems to be exploited. The present state of optical research has been likened to the days of the telegraph. Although optical telecommunication systems are an advanced technology, the full potential of optical telecommunications is just starting to be understood and exploited.

We feel that the optical PCS generation will be defined around optical switching technology, optical megabyte memories, and two way optical transmission using a single optical fiber. Present methods of transmission over optical mediums are unidirectional, requiring a pair of optical fiber cables for two way transmission. The future in the local loop lies in bidirectional transmission using wave or frequency division technology. The transmission capacity of a single optical fiber is limited only by the transmitting and receiving equipment at both ends. System capabilities are constantly increasing because new transmission schemes and equipment are being developed. Once fiber cable is in place, capacity can be increased by upgrading the transmission equipment without necessarily laying new fiber.

The optical age began with the introduction of the laser in 1960. A whole new era in transmission technology had begun [CARL75]. Until the late 1970's, the majority of research work with optics in the telecommunication area focused on long haul transmission systems and how to harness the enormous, transmission bandwidth available using optical fiber. All applications to this point were aimed at point-to-point optical transmission links. Research into two new techniques for transmitting single channel frequencies as either grouped or individual channels took on serious tones in the late 1970s [ISHI84], [ROUS82].

Two techniques, which are more suited for transmissions of 25M bps and less, are wavelength division multiplexing (WDM) and optical-frequency division multiplexing (OFDM). These two techniques are very closely related and many times are mistaken for one in the same. This is really not true. WDM allows modulated radiation from several light sources of clearly distinct wavelengths to be simultaneously transmitted over a single fiber using different formats and different data rates. OFDM, on the other hand, is based on a single wavelength that has been stabilized by an optical oscillator. The wavelength is then divided into subcarrier channels and the various data streams are assigned to the subcarriers. The entire group of signals are then sent as a single wavelength. OFDM requires coherent optical heterodyne procedures while WDM operates independent of coherence [SMIT85].

OFDM research indicates that it is best suited to long haul transmission systems. WDM research while initially directed toward long haul transmission systems has taken a different direction and has shown much promise for application in the local loop [ROUS82]. This research

has lead to new optical devices which affect the telecommunications plant that connects the subscribers telephones, terminals, and other devices that connect to the various switching mechanism. Although the research into WDM techniques is presently being done mostly by telephone companies for central office applications, any advances will ultimately spill over to PCS design [MIDW81], [IWAA83].

5.1. ACTIVE/PASSIVE OPTICAL TECHNOLOGY

Systems or devices that mix both electronic and optical components are called opto-electronic. Any device that converts electronic signals to optical signals or visa versa is considered an active opto-electronic device. On the other hand, if an optical device performs some function on an optical signal, the optical device is considered passive. The transition from opto-electronics to completely passive optical systems will happen as research progresses in developing optical counterparts for the electronic components used in communication systems today. Passive optical solutions are based on developing miniaturized optical components for sources, detectors, modulators, filters, and others optical equivalents for the host of electronic devices in use today in communication and computing systems. The system we are proposing is based on both active and passive optical technologies with the emphasis on as much passive technology as possible. For additional reading on optoelectronics refer to [WILS83].

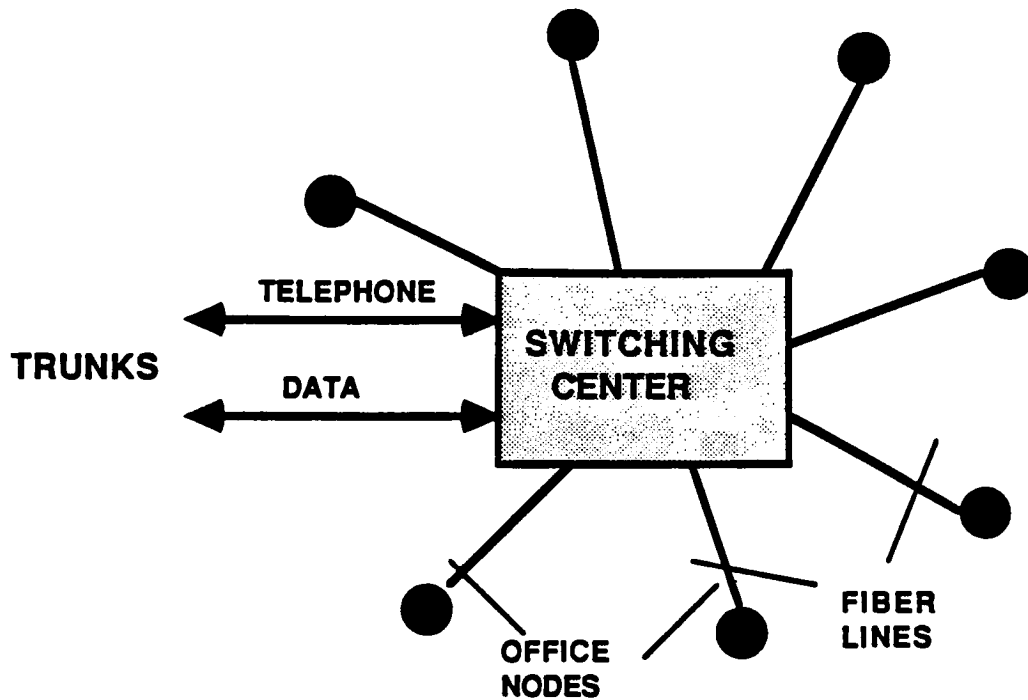


DIAGRAM 5.2.1
STAR CONFIGURATION
OFFICE SWITCHING SYSTEM

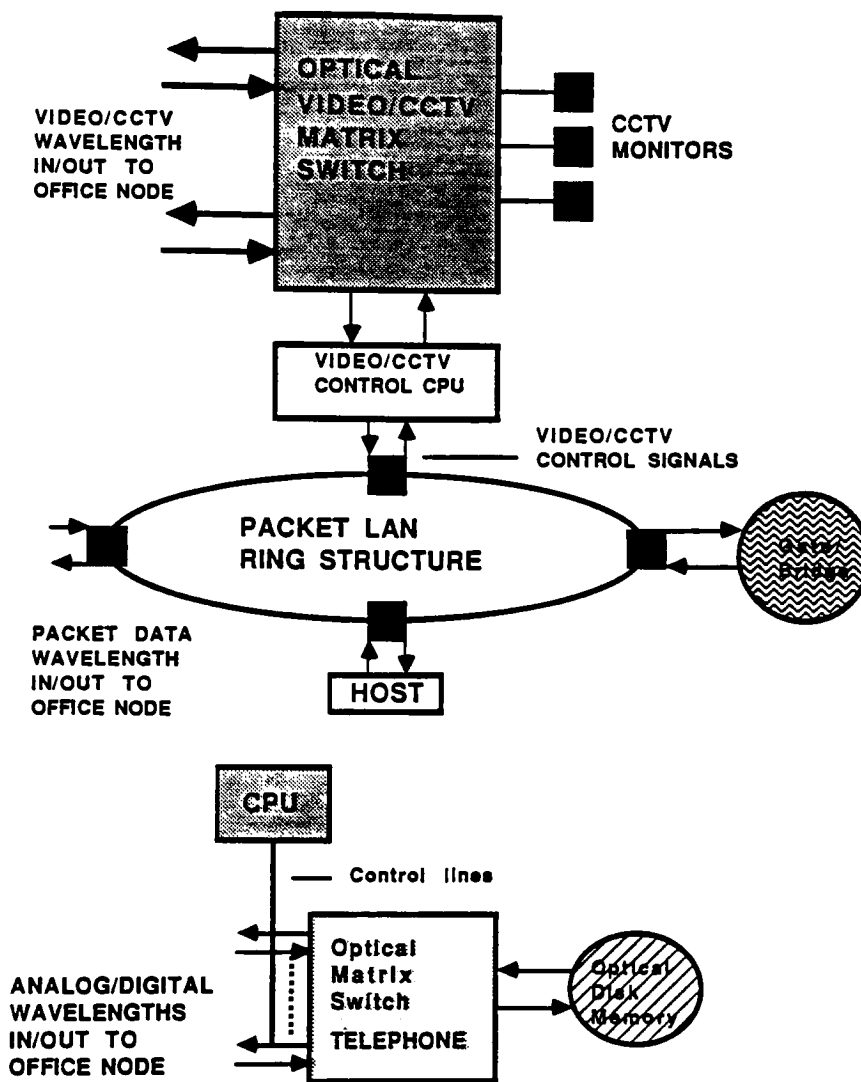


DIAGRAM 5.2.2
SWIYCHING CENTER CONFIGURATION

5.2. DESCRIPTION OF AN OPTICAL COMMUNICATION SYSTEM

The following description is how we envision a private communication system configuration using optical WDM techniques and optical components. We propose a communication system that will effectively carry analog and digital telephone, multiple data formats, fire and energy management, security, and video/CCTV for a small business complex of 75 - 100 offices.

Each transmission will be in the most appropriate format, with no accommodations made. The system will consist of single mode optical fibers for signal transmissions between the stations and the switching centering. All circuits will be routed through a central switching center using single mode optical fibers in place of twisted pair wire. At the switching center, the various signals will be routed to the most appropriate switching mechanisms for connection to their final destinations. Public communication networks are accessed via trunk lines using appropriate interface technology. (See diagram 5.2.1 for a schematic overview of this star shaped system.)

Wave Division Multiplexing techniques will be used to transmit these signals. The switching center will house the necessary optical buffers, memories, and CPU's to accommodate a 10M bps optical baseband LAN, and two optical matrix switches. One of the optical matrix switches will be used for Closed Circuit TV (CCTV) and Video. The other optical matrix switch will be used for telephone, analog or digital, and circuit data switching respectively. (See diagram 5.2.2 for an picture of the switching center.)

All digital data will be switched via the LAN using a packet format. To do this, all digital signals must be gathered and converted to packets at some point in the system. This is best accomplished at the office node in our system. All digital devices will be hardwired to a multiple port, Packet Assembler/Disassembler (PAD) for protocol conversion before being converted to an optical signal. All data terminal, Video/CCTV control signals, fire, energy management, and security signals will be routed through the PAD. Two problems arise from this. Since the PAD has multiple ports, an additional field will need to be added to the packet to clarify address location. The PAD protocol software needs to be adjusted to handle this. The second problem relates to the different data rates of the various signals entering the PAD. When control within the PAD is given to a port, other ports must have sufficient buffering to save signals until their turn for packetizing comes up. This is especially important since slower devices will take longer to assemble or disassemble a packet. Other packets or signals must not be lost while waiting to be processed.

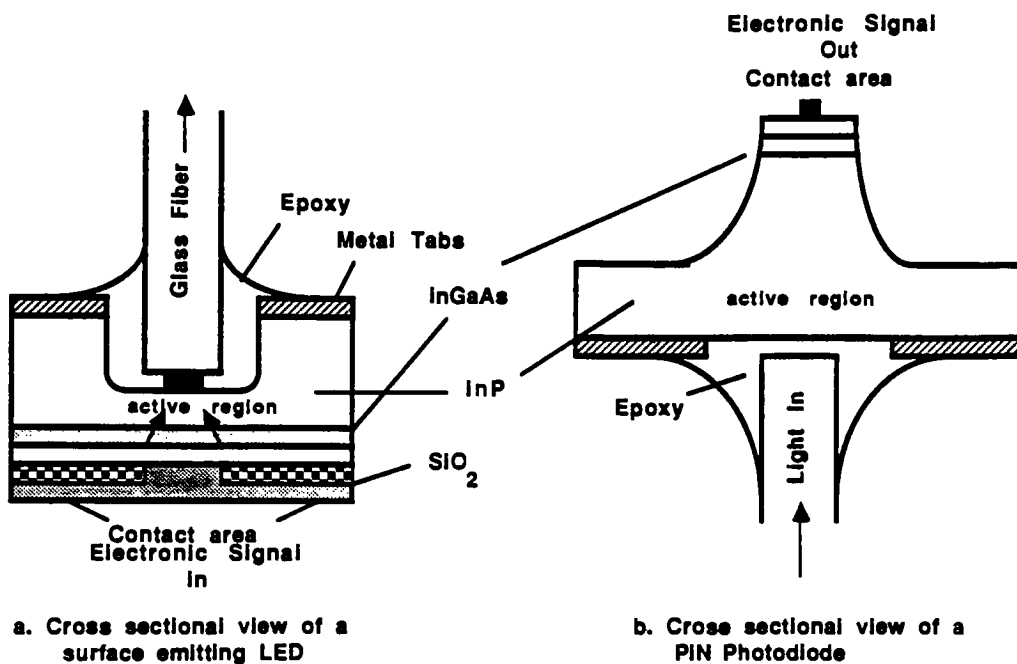


DIAGRAM 5.2.1.1.
SURFACE EMITTING LED & PIN PHOTODIODE

From [KEI83]

5.2.1. *Light Sources and Detectors*

In order to transmit signals across the proposed system the electronic station signals must first be converted to optical wavelengths. Some of the digital signals will first be converted to a packet format. The four electronic channels have to be converted to optical wavelengths.

The principle devices used for sources in optical transmission systems are laser Diodes (LDs), and Light Emitting Diodes (LEDs). Both these devices are suitable for optical transmissions because they have adequate output power to produce a number of different, distinct, light wavelengths across the spectral window. The user can control the output wavelengths by varying the input power to the LD or LED. A complete review of LDs and LEDs is presented in Optical Fiber Communications [KRES81]. In a small business system with bit rates less than 25M bps, LEDs are the most appropriate choice as sources. LDs are better suited to long haul, large bandwidth, coherent optical systems. LEDs require less complex device circuitry than LDs. This is because LEDs require no stabilization circuitry for the output. LEDs are also less expensive to fabricate. The lifetime of LEDs has reached 100,000 to 1,000,000 hrs of service which makes them very useful for our application [ISHI84]. (See diagram 5.2.1.1a for the typical structure of an LED.)

Silicon PIN photodiode devices will be used to convert the returning optical pulse back to an electronic signal. Since the system proposed will be transmitting signals over both the long (LW), and short (SW)

wavelengths, it will need two kind of PIN photodiodes. In the .55 um to 1.0 um SW region, a Silicon PIN produces a sensitivity level of one error bit per one hundred million bits at 140M bps [HOPP81]. In the LW region, 1.0 um to 1.7 um, InGaAs PIN photodiodes are more effective. Complete discussions of both the silicon and InGaAs PIN photodiodes are in Optical Fiber Communications for more detailed reading [KEIS83]. (A diagram of the typical PIN is given in 5.2.1.1b.)

The office wallbox will contain a maximum of four LEDs to convert up to four electronic transmission signals from station devices within the office. Each LED would be assigned to a particular station device transmission. The four transmission wavelengths would be assigned to analog voice (4 Hz) and/or 64 k bps PCM, packet data (1M bps), and Closed Circuit TV or Video (55 MHz). New research into LEDs suggests that A LED does not have to be assigned to each incoming signal. Multiple wavelength LEDs and PIN Photodiode devices are presently in the research stages [CAMP84].

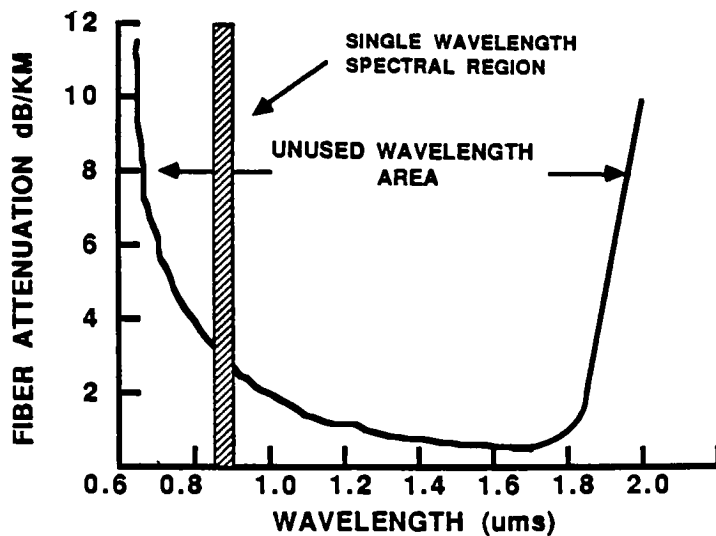


DIAGRAM 5.2.2.1
ATTENUATION LEVELS OF THE
0.6 - 2.0 μM WAVELENGTH WINDOW

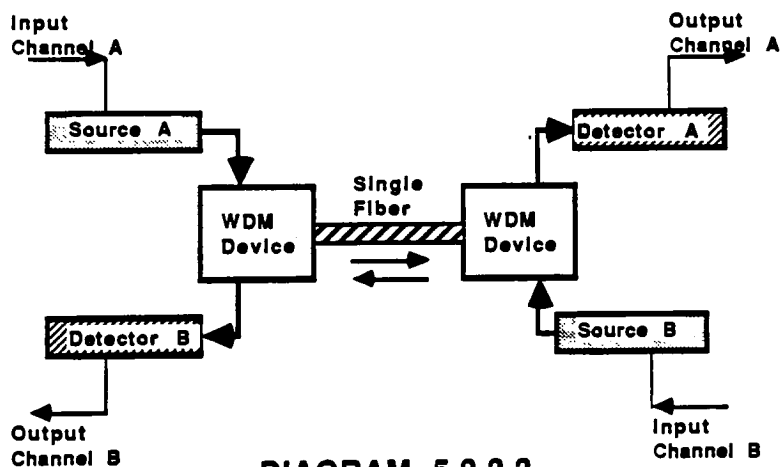


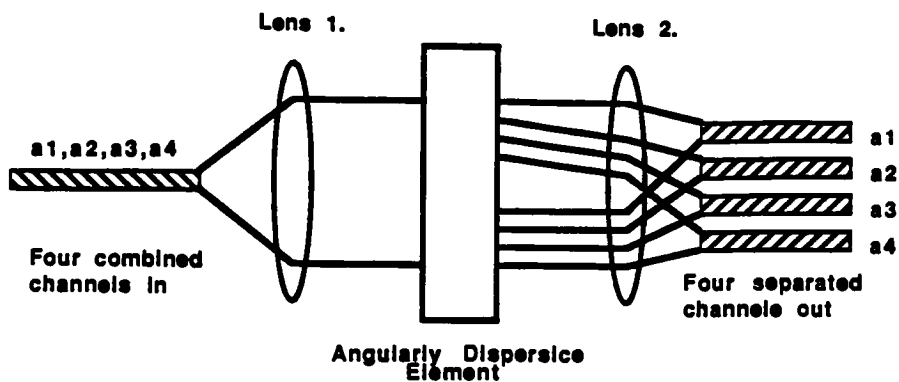
DIAGRAM 5.2.2.2
A TYPICAL BIDIRECTIONAL WDM
TRANSMISSION SYSTEM

5.2.2. Wavelength Division Multiplexing

After the electronic signals are converted to optical signals, these optical signals must be transmitted to the switching center. The system will use WDM technology as the backbone for transmitting the optical signals between the offices and the switching center. Every WDM system must meet three basic performance requirements. These are insertion loss, cross talk loss, and wavelength spacing. Insertion loss is the power loss across the optical fiber that arises from the insertion or removal of an lightwave. It is typically measured in decibels, dBs. A dB is a relative measure of noise level compared to some absolute power level value. Of interest in optical systems is the drop in dBs from receiver to transmitter. Insertion losses in the 1 - 5 dB range are acceptable for WDM systems. Cross talk, measured in dBs, is the inter-channel interference generated by the wavelength spacing. A guard band of open or free space must be placed between the wavelengths to minimize the crosstalk. Cross talk in the -20dB to -30dB range are acceptable norms. Wavelength spacing is the amount of unused wavelength area that must be left so that light wavelengths can travel with minimal amounts of interference. Wavelength spacing is typically measured in micrometers, ums. The guardband between wavelengths varies across the 0.6 - 1.7 um spectral window used for wave transmission. In the short wavelength regions, less then 1.0 (um), wavelength spacings are much less then in the long wavelength, greater then or equal to 1.0 (um). A complete review of these factors can be found in [ISHI84].

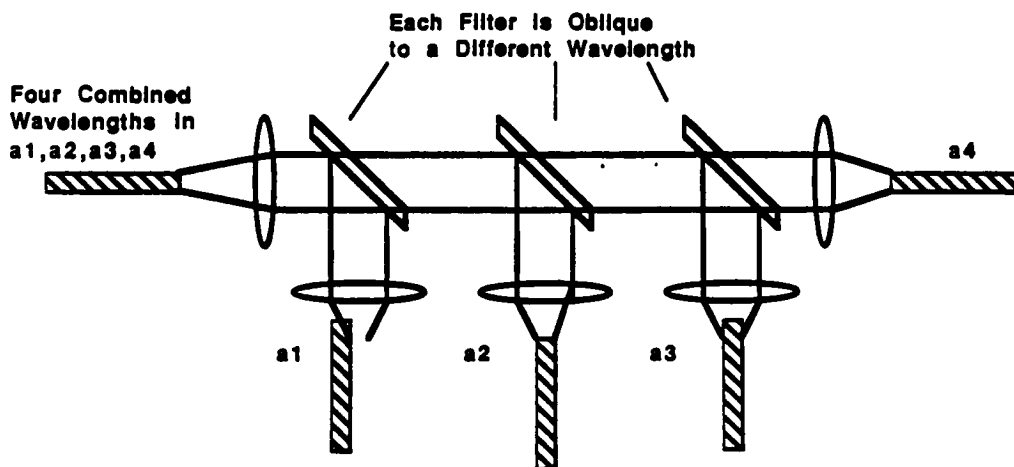
WDM is possible because of the very wide transmission window available in low loss optical fibers. See diagram 5.2.2.1 for this window range. The probable total size of a wavelength window for all practical purposes is in the .6 - 2.0 (μm) range. If we assume that a small business system will always operate at distances of less than two kilometers, any system we propose will not need repeaters and will always operate at less than the -20dB to -30dB crosstalk levels. A single optical transmission has a very narrow spectral width, typically .02 - .03 (μm) in the SW region and .04 to .16 μm s in the LW region [LEEP82]. Consequently, only a very narrow portion of the transmission bandwidth capacity of an optical fiber is used for any one transmission. From diagram 5.2.2.1 it can be seen that it is logically possible to transmit many more simultaneous transmissions by using the additional spectral operating regions.

The system proposed will transmit four wavelengths in one direction in the SW region and four wavelengths in the other direction in the LW region. To date over twenty different distinct wavelength areas have been identified that can be used for WDM transmission purposes [JOUL86]. A schematic of a typical bidirectional WDM transmission system is depicted in diagram 5.2.2.2. The system proposed will differ slightly from this in that there is no need for photodiodes at one end because we will process optical wavelengths, not electronic signals at the switching center.



a)

**SCHEMATIC OF AN ANGULAR DISPERSIVE WDM
DEVICE FOR FOUR WAVELENGTH CHANNELS**



b)

**FILTER TYPE WDM DEVICE
FOR FOUR WAVELENGTHS**

DIAGRAM 5.2.2.3

FILTER & ANGULAR DISPERSIVE WDM DEVICES

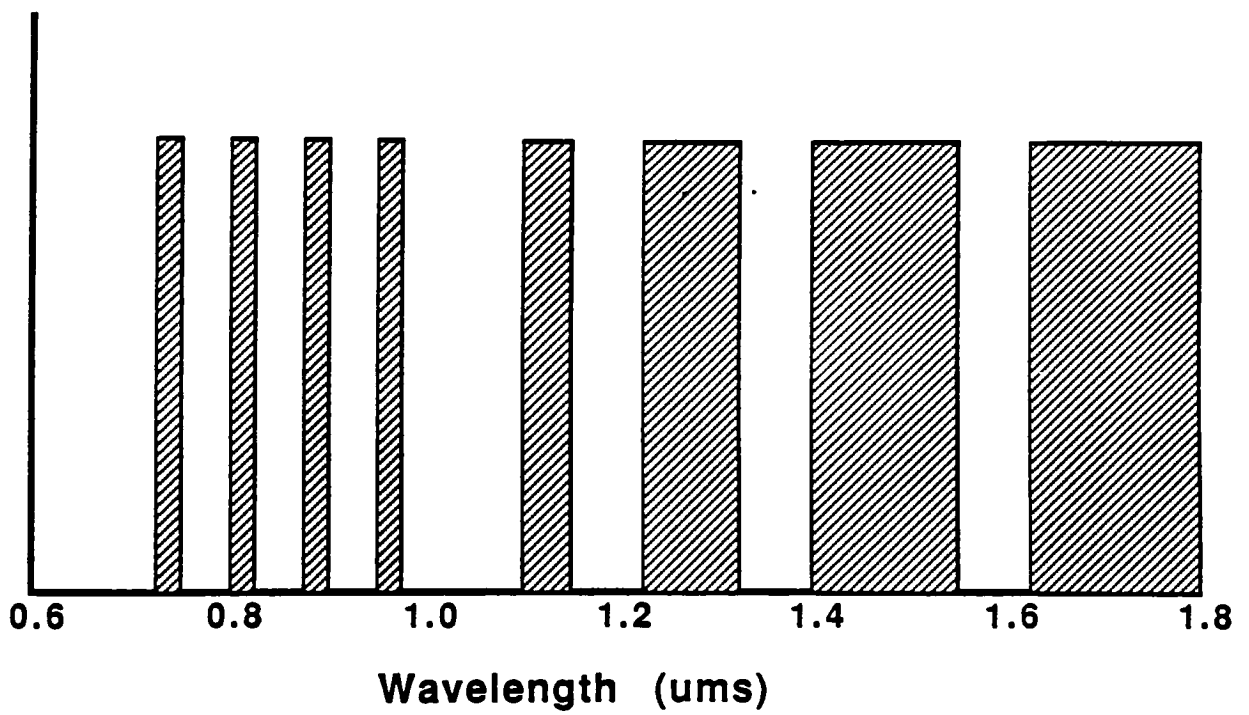


DIAGRAM 5.2.2.4
WAVELENGTH REGIONS IN OUR SYSTEM

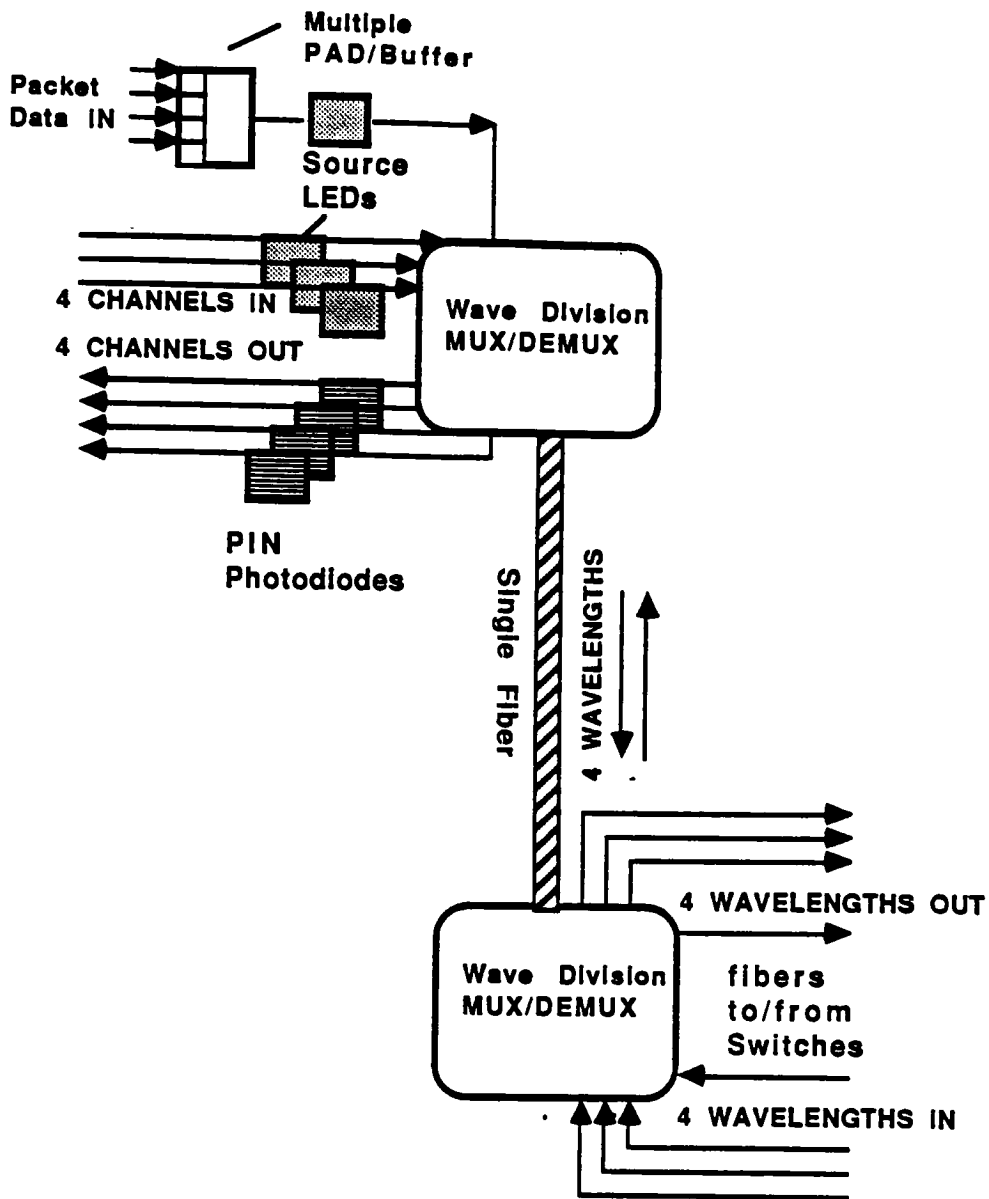


DIAGRAM 5.2.2.5
A BIDIRECTIONAL WDM SYSTEM

***Wavelengths are constant in the SW region
 and gradually widen in the LW region***

There are two basic types of MUX/DEMUX devices that can be used to modulate lightwaves in a WDM system. They are angular dispersive and filter devices. Both devices would be passive in our system. A complete discussion of the two can be reviewed in Optical Multiplexers for Multimode Fiber Transmission Systems, Applications of Grin-Rod Lenses In Optical Fiber Communication Systems, and Optical Devices For Wavelength Multiplexing and Demultiplexing [TOML77], [TOML80], [TOML81]. Filter type devices are more appropriate as multiplexers due to the high insertion losses associated with them. (Diagram 5.2.2.3 shows a schematic for both a four wavelength angular dispersive and a filter type WD device.) WD devices can be used as either a multiplexer or demultiplexer for our purposes. One particular angular dispersive device called the Littrow type has the following performance levels [ISHI84].

| | | |
|-----------------------------|-----------|-----|
| number of possible channels | 3 - 20 | |
| insertion loss | 1 - 4 | dBs |
| cross talk attenuation | 20 - 30 | dBs |
| wave spacing | .20 - .40 | ums |

The system will to employ a Littrow type device with the performance levels falling well within ranges discussed earlier. A WDM system requires additional MUX/DEMUX devices, but when the low insertion and crosstalk levels of such a system are taken into account, WDM systems are no more expensive than conventional space or frequency optical systems [WAT83].

The office wallbox will contain the necessary optical hardware, MUX/DEMUX to transmit and receive a total of eight identified wavelengths across the .6 um to 1.7 um spectral region. The specific

wavelength regions we will use are 0.74, 0.80, 0.87, 0.95, 1.15, 1.3, 1.5, and 1.7 μm .⁻ (Diagram 5.2.2.4 graphically depicts these regions.) Four wavelengths would be transmitted in each direction. Since the system maintains its optical continuity once the signals pass the wallbox, the need for LEDs are virtually not needed at the switching center. All that is needed is a phase shift by the WDM device so that incoming and outgoing signals do not collide. For example, let's say we assign the 0.87 μm wavelength to analog traffic, for telephone transmission purposes from a station to the switching center. The return wavelength for the corresponding signal in the other direction could be in the 1.5 μm region. This assures that there will be no crosstalk interference between the signals going in either direction. By assigning specific wavelengths to specific purposes the demultiplexers can be preset to route specific wavelengths to specific switches or station devices depending on which end of the transmission they are located. (See diagram 5.2.2.5, which is an overview of the office wallbox to switching center connection.)

These eight optical signals would then be modulated by a Wave Division MUX/DEMUX onto assigned wavelengths for transport to the Switching Center. Each station device signal is assigned and modulated onto a specific window, wavelength as described above. Bi-directional transmission is possible since each signal is restricted to its own particular wavelength. At the destination, the optical wavelengths are separated by the demultiplexer. These optical wavelengths could also be converted back to an electronic signal by a photodetector, if desired. The separated optical wavelengths are then routed to the designated appropriate switching device. Optical multiplexers and demultiplexers

are just reciprocals of one another. In an optical WDM system a single optical MUX/DEMUX can be used for both purposes by just reversing its function.

Presently, each input signal from an electronic device requires a separate LED to convert the signal to lightwave. We speculate that as many as eight channels will eventually be modulated on onto eight specific wavelengths by a single LED. Dual wavelength LEDs are being tested. Research is presently being done on a quad LED [EDNS85]. This same technology is presently being applied to PIN photodiodes. If this proves true, the complexity of the office wallbox and switching center port in our system will be greatly reduced. There will be no need for a separate WDM device since the LED will assume the role of the WDM device.

Although each office would have the capability of simultaneously transmitting and receiving over all eight wavelengths, the probability of all eight wavelengths being used is very slim. This is because of the conflicting nature of the sending devices. How often would a person be using an analog and digital phone at the same time, or a packet and circuit switched data terminal. CCTV would be used for night and weekend security and video for daytime purposes. Even though these traffic situations would probably be true, the wavelength spacing and equipment could easily handle full traffic within the parameters stated earlier [KAUF83].

5.2.3. *The Switching Center*

Every optical fiber from each office terminates at a WDM device located at the switching center. (Refer to diagram 5.2.2.) The WDM device located at the switching center could be considered a first level port for the system. At this level, the various wavebands are separated by the WDM device and routed to the appropriate switching mechanisms. Each of the three switching mechanisms, the optical LAN, and the two optical matrix switches have a different control signal structure. Therefore no signal control should be exercised at the WDM device level because of the heterogeneous mix of traffic. Control signals are interpreted at each of the switching devices.

From each of the WDM ports at the switching center, four distinct optical fibers are used as lines to route the separated wavelengths from the stations to a specific switching mechanism. Since only a single wavelength is transmitted over the fiber, it could be said that this method is analogous to space division switching. Each WDM device at the switching center also has four output lines at an offset wavelength to transmit back to the office station.

The optical LAN switches any of the various packet wavelength signals that are routed to it. These include all energy, fire, security, and Video/CCTV control signals along with all data transmissions. Fiber optic lines have been used by a number of leading LAN manufacturers as the medium of choice. The following LAN products all have the option of fiber optic lines as the network medium: NET/ONE by Ungermann-Bass, ProNET-10, ProNET-80 by Proteon, PRIMENET by Prime, and Token-

Ring-Network by IBM, to name a few.

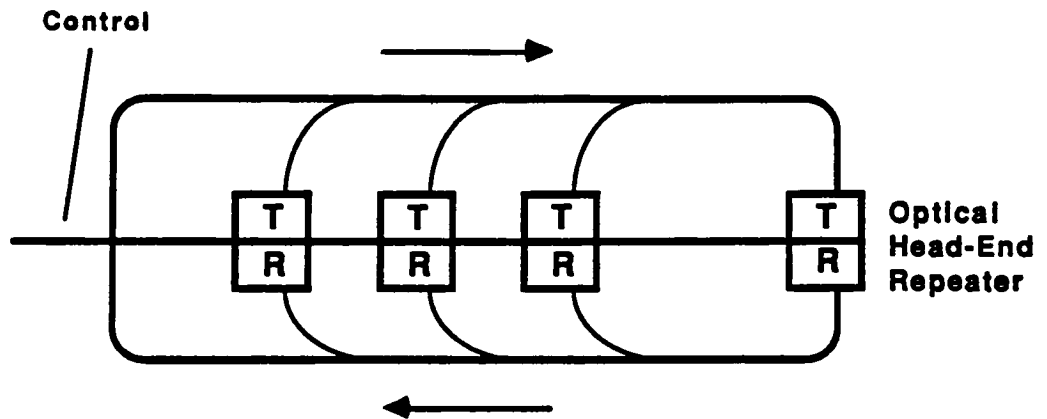


DIAGRAM 5.2.3.1
RING TOPOLOGY FOR OPTICAL LAN

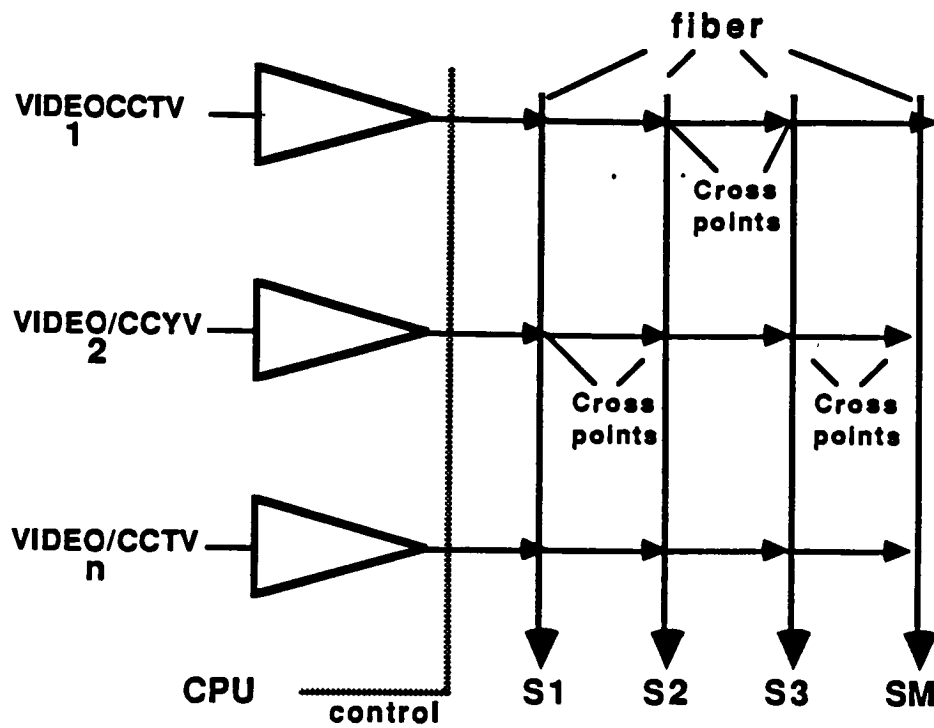


DIAGRAM 5.2.3.2
BROADBAND OPTICAL MATRIX

The major weakness in any optical based LAN topology is the power loss inherent with optical taps. The limited power budget associated with optical LANs can not offset the drains on the system because of tapping and launching optical signals. This has typically limited the number of nodes attached to the system without repeaters. See Fiber Optic Local Area Topologies [SUHY86] for a complete discussion of the power losses associated with various optical LAN topologies. The best topology for an optical LAN is a ring with a head-end optical repeater as shown in diagram 5.2.3.1. This topology will give us the additional nodes in our LED driven system.

The other two switches used in the system are optical matrix switches. Unlike the LAN, the matrix switch sets up dedicated connections, point-to point, between the stations. (Diagram 5.2.3.2 shows the structure of a typically optical matrix switch.) One feature of a matrix switch is it does not have to necessarily be symmetrical. In the case of video/CCTV, a broadband matrix switch is the more appropriate choice. The nice feature of a broadband matrix switch is its ability to accommodate a wide range of data rates. These signals could range from 30 MHz for high definition TV to 55MHz for digital TV. The control signals for the Video/CCTV switch will be carried over the LAN and transported to the controlling CPU as discussed earlier. Further reading on the status of optical video matrix can be found in Finally Optical Switching Looms For Video [GALL86].

A second optical matrix switch will be used for voice traffic. The idea of a optical matrix switch for frequencies less than 100 kHz is discussed in A Fiber Optic Broadband LAN/OCS Using A PBX [HARA83].

Such a switching system could act as a PBX for voice communications in our system. If an optical matrix switch for these frequencies is possible, it would be transparent to both 4 Hz analog and 64 k bps digital signals. Both signals could be routed through the same matrix. Attached to the PBX matrix switch would be an optical disk memory. This disk would act as a data base for both station and system features that are typically associated with a PBX. All control is handled by a separate CPU dedicated to this circuit switch.

5.2.4. Advantages Of An Active/Passive Optical Switching System

The major weaknesses of fourth generation PBXs discussed at the end of Chapter Four are easily corrected by an optical communication system. The broadband capabilities of a silicon based communication system would allow for the transmission of multiple signal formats across a common media. One technique for this is WDM mentioned earlier and discussed in detail in the previous section. In a user environment, the enormous bandwidth capabilities of fiber cable will allow a single cable to transmit all voice, data, and video demands. There will still be plenty of room for expansion without laying additional cable.

A second advantage of an active/passive optical switching system is its immunity from electromagnetic interference (EMI) and radio frequency interference (RFI) and durability in harsh physical climates. Harsh or noisy environments do not affect signal quality. Optical fiber can be virtually run anywhere. It is not susceptible to lightning strikes, does not spark/nonflammable, and can withstand temperatures in excess of 1000 degrees Centigrade. This can lead to cost savings since

routing considerations used for standard copper cable need not be considered [KEIS83].

Using optical fiber as the transmission medium has some major advantages over copper based technologies. Optical fiber does not emit any form of detectable radiation. This leads to very high security since it is virtually impossible to tap a line without detection. Any signal loss can be immediately detected. Also, any break can be detected very easily for maintenance purposes. Because of all these immunity factors, a bit error rate of one in one billion is standard. This error rate will improve even further as the technology is refined. Unfortunately, this has also been a problem in developing optical components. Cost efficient optical couplers splicers are still in the developmental stages because of the difficulty in tapping and splicing fiber cables. Optical media is also immune to such common annoyances as crosstalk, echoing and ringing [MIDW81].

Fiber cable also stacks up favorable over copper wire in many other ways. Optical fiber is made from sands or silica, the most abundant material on earth. One pound of glass can make over a mile of fiber cable. Copper, on the other hand, is a diminishing resource. Fibers are only 6/1000s of an inch thick yet have the tensile strength of steel of the same diameter. Optical fiber is so flexible that it can be tied into knots and will not snap yet it is so small and lightweight that it will fit anywhere. Fiber also conserves energy. An optic system uses about one third the electrical requirement of a comparable copper system. Also the number of repeaters is reduced by a factor of six [KEIS83].

The only thing that is holding back the development of total optical systems is the lack of component technology for other parts of the switching system. No one in the field really needs to be convinced that the ultimate solutions to total communication systems are optically based. The question is when will the technology be in place to economically take advantage of optical solutions.

5.2.5. Conclusions

State of the art WDM technology and its application to a office communication system have been reviewed. WD applications in the local loop are presently under active study. The impacts of this new technology should enhance system design and eventually system costs. WDM systems are particularly promising in the area of multiservice subscriber systems. Although steady progress has been made in multiplexers, demultiplexers, couplers, sources, and detectors for such systems, they are really not cost effective at this point in time. After examining the factor associated with using the 0.6 to 1.7 μm window in single mode fiber for a local communication system, such a system seems quite practical. For transmission systems of less than a few kilometers, the design constraints are quite different than those of a long haul system. The design problem primarily focuses on power budgets, and losses associated with passive components. More work needs to be directed at refining such systems. It remains to be seen whether the economies of wave division systems will materialize, but we feel from all our research they will.

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