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On the proposed integrated services digital network

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ON THE PROPOSED

INTEGRATED SERVICES DIGITAL NETWORK

A THESIS

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An introduction to the concept of ISDN and to the evolutionary forces leading to its inception.

1.1 Thesis Goals

The ultimate aim of this dissertation is neither to increase the state of the art in networking technology nor to predict the future structure of telecommunications networks. It is an attempt to raise the awareness of both the author and the reader as to the evolutionary forces driving vast changes in the telecommunications field. The impact of these changes will significantly alter the way we live and conduct business in the Information Age. For those involved with the communications field, the ability to make sound business decisions will require an in-depth knowledge of the technology and services that compose ISDN. Therefore this paper will explore the motivating forces, the potential services, and the technical components in the emerging Integrated Services Digital Network (ISDN).

My interest in data communications was kindled at RIT and has continued in my work, both as a systems programmer in the telecommunications division at the Travelers Insurance Company and as a printing systems analyst for Xerox Corporation. This thesis has helped me answer both personal and professional questions about the future of telecommunications and to share this information with others.

1.2 ISDN Concepts

1.2.1 Overview.

Commonly referred to as a digital superhighway, the Integrated Services Digital Network is a proposed world wide public telecommunications network capable of providing a variety of services to meet the diverse communications needs of today's users. Some of these services include: circuit switched voice, circuit switched data, information retrieval, electronic mail, electronic funds transfer, facsimile, teletex and telemetry. The next chapter will examine these services in detail. It will also look at how changes in one industry, banking, are helping to provide the impetus for an ISDN. Ultimately it is the customer who will benefit from increased services and a reduced cost per service. Also the technological impact will be substantial as sophisticated digital switching and multiplexing schemes are introduced. The ISDN will be defined by the standardization of user interfaces and will be implemented as a set of switches and paths supporting a broad range of traffic types.{43} While in actuality there will be many ISDN's, this will be transparent to the user.

The first set of standards for the ISDN was adopted by CCITT in October, 1984. Recommendation G.705 suggests that ISDN will begin by integrating data and digital telephone and will progressively incorporate additional functions and network features. It defines an ISDN this way: a public end-to-end digital telecommunications network with signalling, switching and transport capabilities supporting a wide range of services accessed through standardized interfaces with integrated user control.{18} The end-to-end digital connectivity results in higher speed and better performance. These

same digital paths and switches are used by all data being transported across the ISDN. The CCITT recommendation also suggests that the same interface should allow for user choice of bit rate, switching mode, and coding method on a call-by-call basis. Existing terminals will be connected in accordance with X.21 and X.25.

1.2.2 Motivation for ISDN

A look at some of the factors involving the evolution of existing voice telecommunications networks and value added data communications networks to an Integrated Services Digital Network.

This evolution towards end-to-end digital switching stems from telephony and combines two technologies: digital transmission and digital switching.{43} The first large digital carrier system was introduced into commercial service by AT&T in 1962 and the first large scale time-division digital switch, the Western Electric 4ESS, was introduced in 1976. These technologies are well established. The key lies in uniting digital transmission with digital switching. The combination of these two technologies in the public circuit switched telecommunications network is called the Integrated Digital Network (IDN). The IDN forms the basis for the ISDN. Figure 1.1 shows examples of non-integrated and integrated transmission and switching.{43} Traditionally there has been a separation of transmission and switching systems within the telephone system. In the analog network incoming voice lines are modulated and multiplexed at the end office and sent out over a frequency division multiplexed (FDM) line. These signals must pass through one or more switching centers before reaching their destination. At each

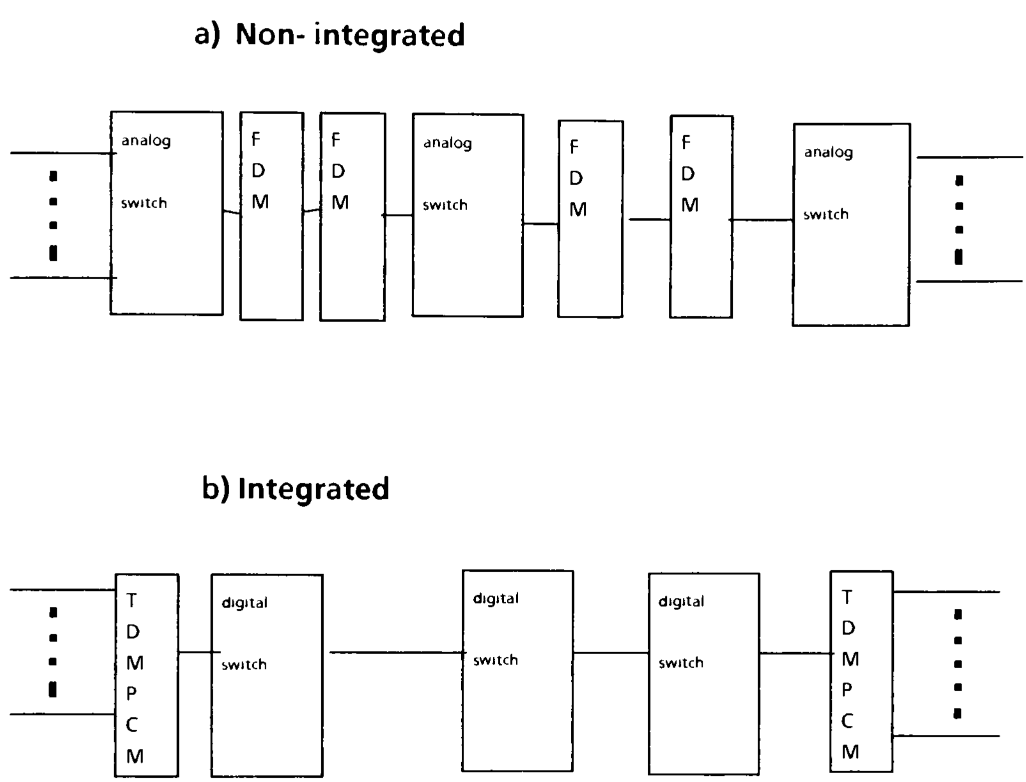


Figure 1.1 Integrated and Non-integrated Transmission

switching center the incoming FDM carrier has to be de-multiplexed and demodulated before being switched by a space division switch (a space division switch is defined in section 3.2). After switching, the signals have to be multiplexed and modulated again to be transmitted. This repeated process results in accumulation of noise as well as cost.

Integration can be achieved where both voice and data are digital. Incoming voice signals are digitized using pulse code modulation (PCM) and

multiplexed using Time Division Multiplexing (TDM). Time division digital switches along the way can switch the signals without decoding them and separate multiplex/demultiplex stations are not necessary since the function is incorporated into the switching system. This conversion of voice telecommunication networks to digital transmission and digital switching, the IDN, is well under way and is providing the basis for ISDN. The IDN will combine the extensive geographical coverage of the telephone network with the data carrying capacity of digital data networks. The end result will be the Integrated Services Digital Network.

1.2.3 ISDN Architecture

The architecture of an ISDN is comprised of three elements: integrated access links, integrated switches, and network termination equipment. Accessing the ISDN is done using a combination of B-channels and D-channels which will be dicussed at length in chapter 5. Briefly, a B-channel allows for 64 Kbps and a D-channel supports 16 Kbps. The D-channel is message oriented and carries the signalling information that controls circuit switched handling of B-channels through the ISDN. Telemetry and low-speed interactive data share, by statistical multiplexing, the D-channel together with signalling messages.{15} Functions of the B-channel include the transport of PCM speech at 64 Kbps high speed circuit and packet switched data, low bit rate voice (LBRV) combined with data information, wide band digital speech, facsimile, and slow-scan video.

This combination of B and D channels is intended primarily for small user terminal installations. Large terminal clusters such as PABX's (Private Area

Branch eXchange) and LAN's (Local Area Networks) can be served by a combination of 64 Kbps B-channels and a D-channel (nB + D). Aggregate B-channels will be used to support bit rates in the area of 1.54 to 2.048 Mbps. Broadband bulk data transfers will be supported by statistically multiplexed channels via satellite.

Network Termination devices (NT1 and NT2) will work like communication controllers. They will route incoming traffic and collect outbound traffic from attached voice and data devices and will provide the network signal synchronization and related functions. They will also serve as a diagnostic checkpoint and implement certain protocols needed to move information between the user's site and the network. A network terminating device can also perform code translation, protocol conversion, terminal emulation, data speed conversion, statistical multiplexing, and conversion of analog signals to digital signals. A codec, which does the actual conversion, resides in all links and analog trunk port circuits of digital exchanges.{23} Figure 1.2 shows basic ISDN access.

There are three types of network terminating devices: NT1, NT2, and NT12. NT12 is a combination of NT1 and NT2. The NT1 device handles the OSI physical layer functions. It also controls line transmission termination, timing, power feeding, maintenance and test loops. The NT1 defines the boundary of the carrier's ISDN and may be carrier controlled.

The NT2 device performs some physical layer functions as well as all level 2 link control and level 3 network control for the D-channel protocol. The NT2 connects user terminals to the NT1. Its services include switching, data

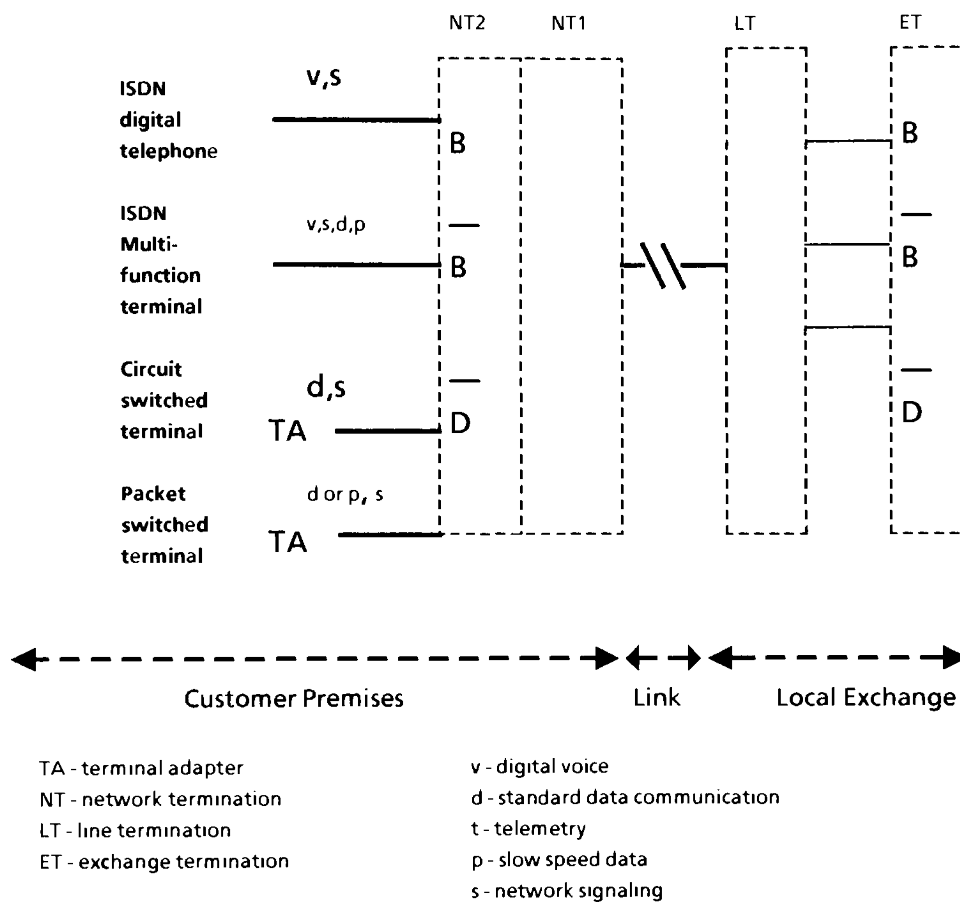


Figure 1.2 Basic ISDN Access

concentration, maintenance, multiplexing, and protocol handling for layers 2 and 3. It can be compared to a PABX or LAN. Each terminal-to-NT1 and NT1-to-NT2 connection is simply a point-to-point connection with the NT2 performing the controller or arbitration function.{13}

The ISDN relies on high speed integrated digital switches to move information quickly and efficiently. Presently, because of their inherent characteristics, most voice networks use circuit switching and most public data networks use packet switching. The integrated switches in an ISDN must be able to select the appropriate switching method not only for the many classes of voice and data transmissions but for the full range of services provided by the ISDN. For this reason it is worthwhile to investigate current switching technologies including circuit switching, message switching, packet switching, and hybrid combinations of circuit and packet switching.

Circuit switching involves setting all the switches before a communications session begins and maintains their position until the session ends. Message switching is similar to packet switching in that both use the store and forward method of transporting data. In message switching the entire message is passed through the network one node at a time. Each node receives the message, stores it and passes it to the next node. The switches do not maintain their positions for the duration of the session so no permanent circuit is established. But because the entire message must be stored until an outgoing line is clear it is relatively slow. Message switching is intended primarily for non real-time people-to-people traffic.

Packet switching is intended primarily for fast machine-to-machine traffic including terminal-to-computer connections. To be more efficient the message is split into pieces or packets and each packet is routed to its destination independently of the other. Once at its destination the message is reassembled from the individual packets and delivered to the user. Packet switching has

proven to be a fast and efficient mode of transporting data. There are several Packet Switched Data Networks in the United States. User access to these networks usually follows the 1980 CCITT X.25 protocol.

A detailed look at the services which will utilize ISDN. An examination of digitized voice, data, telemetry, facsimile, teletex, videotex, teleconferencing and others to see what services an ISDN must provide. An example from the banking industry is studied to see how ISDN can help solve a user's communication needs.

2.1 ISDN Services

The information media that the ISDN will serve come in many forms and will almost certainly grow and change. To facilitate this analysis these services are divided into four fundamental components and studied individually and as they relate to each other. This grouping is quite arbitrary and is to be used as an expedient for analysis rather than a final classification. The fundamental components include voice, image, text and data. These are defined as:

Voice - a medium for personal contact; informal, urgent or personal communication or discussion.

Image - a medium for visual display; pictures, pictorial overview or synthesis.

Text - a medium for the preservation of thought; qualitative, contextual or verbal communication or analysis; writing, printing, publications.

Data - a medium for the preservation of measurement; quantitative, numeric, precise information. {41}

Currently most voice and image transmissions tend to be analog. Text and data are usually created and stored digitally but transmitted as an analog signal. Under ISDN all would be transmitted digitally. Voice transmission usually means the existing telephone system with its widely dispersed network of telephone links connecting nearly all spots on the globe. These links include 2-wire and 4-wire twisted pair, fiber optics, high bandwidth cable, microwave and satellite connections. The telephone network is evolving into a digital network as new equipment is added. It is the telephone network which forms the basis of ISDN.

With the exception of business and engineering graphics image transmission is usually restricted to high bandwidth, dedicated channels such as microwave and satellite links. Because of the extremely large bandwidth requirements for full image transmission, such as Broadcast Video (96 Mbps), this form of digital transmission will probably be one of the last to be implemented by ISDN.

Data and text, however, are prime candidates for digital transmission. Usually stored in digital form, they must be modulated and transmitted as analog signals over most existing long-haul data networks. Data networks historically represent the union of data processing and telecommunications, often called teleprocessing. Most teleprocessing falls into two general categories: batch and interactive. Batch transmissions are usually non-real-time and continuous in nature This type of transmisson is served well by the conventional circuit switched telephone network. A good example of a batch transmission is sending print data from tapes out to remote printers via phone lines. Batch transmissions are inherently interruptible and are readily

implemented by a store and forward technique. This is true even of large file transfers although the file may need to be broken down at the sender and reassembled at the receiving end. Interactive transmissions, however, are generally more real-time and bursty in nature. More and more data traffic falls into this category. The airline reservation system is the classic example of a sophisticated interactive data base update and inquiry system involving instant transaction processing and bursty traffic.

Text transmission, in the form of documents, is usually short and requires no immediate response. This is often implemented using a message switched store and forward system. Figure 2.1 gives examples of different services and their response requirements.

These four media; voice, image, text and data, are often alternatives to one another. Data information can be displayed graphically. A text message could be sent as a voice message. A document or image can be digitized and transmitted via a facsimile device. A choice is made of which medium to use for a particular application. For example:

Voice: telephone, audio conferencing, audio processing.

Image: facsimile, business and engineering graphics, video.

Text: word processing, electronic mail, text processing.

Data: data processing, database, computation.

As computers and telecommunications combine, we get: {41}

	Real-Time	Store and Forward	Interactive
Voice	Telephone Audio Confer.	Voice Message	Voice I / O
Image	Tele- conferencing	Facsimile	Graphics Video
Text		Text Messaging	Word Processing Computer Confer.
Data	Process Control	File Transfer	Transaction Processing

Figure 2.1 Service Response Requirements

Media	Service
Voice / image	video teleconferencing, picture phone.
Image / text	videotex, photo composition, electronic blackboard.
Text / data	computer conferencing, word processing and data processing combined, teletext.
Voice / data	voice recognition, voice ordering, voice I/O.
Voice / text	dictation, voice messaging, voice annotation, display phone, text-to-speech.
Image / data	computer aided drafting, optical character recognition.

As computers merge with telephones and television via telecommunications, the possibilities in regard to information products are endless. Each product

will have associated with it certain properties which will make it more expedient to use one form of channel or switching technique over another. So the design of a transparent digital super highway is no small matter. Telecommunications involves many facets, including {28} :

-communications speeds and protocols;

-interconnect topology and connectivity;

-inter / intra office, local / long-haul communications;

-transmission media, switching equipment, wiring;

-voice, image, text and data communications traffic.

While voice and data are terms most people are familiar with, some services such as videotex and teletex may not be so familiar. Here is a brief description of each: {42}

1. **Telephone**: Telephone communications provided by an assembly of telephone stations, lines, channels, and switching arrangements for their inter connections, and accessories.

2. **Telex**: A telegraph type service involving the transmission of telegraph signals in a given direction between two terminal sets for the transfer and reproduction of alphanumeric text documents.

3. **Videotex**: A data bank enquiry and retrieval service allowing for transactional facilities. Videotex is bi-directional and eventually will provide a broad range of services limited only by the imagination. Some current videotex services in Europe where videotex is very popular include: {9}

a) General information such as news, sports, leisure and entertainment;

b) Professional information such as technical literature, calculations and circulars;

c) Message capability like image handling, electronic mail and newspaper reprinting;

d) Classified ads for employment, real-estate, and new products;

e) Shopping aids like mail order goods or special prices;

f) Business applications like sales, financial, production, inventory, accounting and personnel data;

g) Reservations for hotels, cars and airlines;

h) Education, such as courses at home and tutoring services.

4. Teletext : technically similar to videotex in its high bandwith requirements however it is sent via a broadcast television station and is unidirectional rather than bidirectional. It is sent to a TV set, not to a terminal.

5. Teletex : A service which provides communication between terminals which are used for the preparation, editing, and printing of correspondence. Information is transferred on a memory-to-memory basis. The CCITT Recommendations S.60 and S.61 provide for final form text documents to be transmitted between special workstations at 2400 bps. These documents include an extensive character set, are currently non-

editable and control the placement of text on a printed page. {17} There are four levels to this document level protocol:

a) **negotiation** - determines whether a receiving station is able to handle the document.

b) **initiation** - identifies the document in question.

c) **transfer**- the actual transmission of the document.

d) **termination**- acknowledgement of successful transfer. Provides for error recovery to page boundary.

6. **Facsimile**: A system which allows the transmission and hard copy reproduction of fixed images (photographic or otherwise) using a scanning technique. The reproduction may be in two significant states only (ie- black and white), it may contain intermediate states, or it may be colored. Facsimile data may be stored at the sender or receivers station, possibly a PC, and printed or transmitted at any time. The facimile device can store dozens of telephone numbers and hundreds of pages of documents in memory. As these devices grow in sophistication they will act like scanners and allow editing and merging of graphic input. CCITT standards define four classes of facsimile:

Group 1: Six minute transfer time.

Group 2: Three minute transfer time.

Group 3: Sub one minute transfer time

Group 4: (future) - Networked facsimile stations offering resolutions of 200,240,300, and 400 dots per inch.

Facsimile devices will also be able to restrict access to documents and to store and foward a document sent from a remote device to another remote device. Facsimile, which is a technology that has existed for a relatively long period of time, will play a key role in the office environment of the future and will benefit from the flexible, high speed end-to-end digital transmission of ISDN.

7. **Telemetry**: (remote metering service). Measurement with the aid of intermediate means that permit the measurement to be interpreted at a distance from the primary detector.

8. **Alarm Service** : A telegraph type service whereby an assembly of equipment and devices is arranged to signal the presence of a hazard requiring urgent attention and permits that signal to be interpreted at a distance from the primary detector.

9. **Teleprocessing Service**: A service whereby information is transmitted by data transmission means, to be processed by a computer at a point distant from the primary service.

10. Teleconferencing: Teleconferencing can take many shapes, from full motion audio -video to freeze frame images and voice. Images are the most resource consuming when it comes to transmission, requiring up to 9.6 Gbps. Today most existing teleconferencing sites (ie. Travelers Insurance Company) have attempted to minimize the image requirements and use some sort of freeze frame technique.

These forms of communication require a wide range of bandwidth and switching techniques. ISDN is designed to meet these requirements. Figure 2.2 displays these attributes as will as indicating the probable ISDN channel to be used for transmission. {2}

2.3 ISDN and the Banking Industry

It should be clear that we, as consumers, will be faced with a myriad of choices involving new home services such as voice mail, facsimile, videotex, and teletex. Much has been said about our entering the information age where knowledge workers replace industrial laborers. Nowhere is this more evident than in the banking industry whose product is information. Through the marriage of telecommunications and data processing banks, like other financial institutions (insurance, investments, etc.) have become masters at capturing, transmitting, processing, storing, printing, and re-transmitting data. Along with this extensive high tech investment and expertise comes the ability to offer many new banking services to customers. Indeed banks are being forced to provide new services if they wish to prosper and grow in an

Service	Bandwidth	ISDN Channel		Facilities Switching		
		B	D	Circuit	Packet	Channel
Telephone	8,16,32,64 Kbps	X		X		
Alarms smoke,fire, police	10-100 bps		X		X	
Utility Metering	0.1-1.0 Kbps		X	X	X	
Energy Management	0.1-1.0 Kbps		X	X	X	
Interactive Services Electronic banking Yellow pages Opinion polling	4.8 - 64 Kbps	X			X	
Electronic Mail	4.8 - 64 Kpbs	X			X	
Interactive Video	9.6 Mbps	X		X		X
Broadcast Video	9.6 Mbps	X				X
Bulkdata Facsimile	4.8 - 64 Kbps	X		X		

Figure 2.2 Bandwidth and Switching Techniques

increasingly deregulated and competitive environment. These new computer based products and services rely heavily on telecommunications. This section

examines some of the services and products and looks at how ISDN will facilitate their development. While this is only a glimpse at one industry, it may be representative of future trends and well worth exploring.

Like all users, banks hope ISDN will help overcome multi-vendor incompatibility by providing standardized digital interfaces and protocols. This may or may not happen. Even now companies like Motorola and Northern Telecom are coming out with their own implementation of NT1 devices on integrated circuit boards. While it attests to the inevitability of ISDN it may also be indicative of vendors trying to set an early standard.{17}

The push for ISDN by the banking industry comes from two sides: individual customer services and wholesale (corporate) services.{38}

Customer services include:

1. Teller services- rapid access to customer accounts is a must. Input is entered from a terminal at the branch office . It is sent as an interactive data stream in about 100 byte increments and must arrive at the host, be processed immediately, and arrive back at the terminal (this response is about 1000 - 1500 bytes in length) error - free. Most banks have from 500 1000 terminals. A nice feature for ISDN as automatic teller services become more popular will be a voice/data call so that the operator would have immediate access to the same data the customer was viewing. Another nice feature would be optical scanning of input and signature verification.

2. Credit authorization and point-of-sale services- This includes credit card and check cashing applications. For ISDN it means fast data connect to replace

current slow dial up services in the voice network. The traffic would be small (50 100) bytes and extremely bursty. Also point of sale applications such as inventory managment and credit card verification may involve storing data and sending it to different hosts.

3. Card operations 'Smart card' operations which will debit and credit accounts. ISDN must provide extremely reliable and secure transmission of interactive data.

4. Electronic banking Here there is a need for automatic identification of the calling party. Again ISDN must be able to simultaneously carry voice and data. In addition voice and data must be synchronized so that there is no voice transmission while data is waiting in a queue to be processed.

Here is a look at the wholesale side.

1. Cash management - Here the major requirement is fast, reliable global data and voice communications.

2. Funds transfer - This currently involves teletype and telex service. ISDN must provide efficient message handling capabilities.

3. Collections - ISDN will be expected to provide security and message services.

4. Letter of credit ISDN will provide for verification of signatures at the receiving end. This will help eliminate the 'float' problem where interest rates change while documents are in the mail.

5. Payment process Banks may even call upon ISDN to keep track of transactions and periodically execute a net settlement between banks as a clearing house function.

It is likely that banking in the future (like many industries) will be far different than it is today. Users will expect fast, flexible, efficient, secure, robust, and integrated digital networks to help them provide new and diverse services. The financial industry, like all industries, will eye telecommunications as an area to reduce costs and enhance product lines. Heavy demands will be placed upon ISDN to meet those needs in as transparent a manner as possible.

The basic structure of ISDN is explored beginning with the telephone network and the nature of digital signals and ending with the fundamental switching and signalling design of ISDN.

3.1 Overview

Now that the various services utilizing ISDN and their requirements have been discussed it is time to examine the physical structure of this proposed digital super highway. The existing telephone network will provide the basis for ISDN. This is still primarily an analog network. Indeed most naturally occurring messages are analog in nature.{28} The first part of this chapter examines the existing telephone system and discusses both how and why these analog signals will be transmitted digitally. The examination of digital transmission will be in three parts:

1. Conversion of an analog signal to a digital signal.
2. Multiplexing digital signals.
3. Transmitting digital signals.

The second part of this chapter will look at the evolving design of ISDN and the hardware and software necessary to implement its features.

3.2 The Telephone System

It was in 1876 that Alexander Graham Bell achieved the first successful transmission of intelligible speech. From this humble beginning has evolved a

worldwide telecommunications network of over 350 million telephones. This network has evolved with technological advances but still lags behind due to the huge capital investment in existing equipment. A prime example of this is the space division switch. Most existing switches are based on an electromechanical device called a Strowger Switch or uniselector. This device responds to the pulses provided by the dial on the telephone set. Each pulse rotates the dial by one posistion, where a new set of contacts exists. By arranging the Strowger Switches in a particular order a transmission path is routed through the exchange. While these original switches have been greatly improved they cannot compete with the latest development in electronic switches: Pulse Code Modulated (PCM) Time-Division Multiplexed (TDM) digital switches. This type of device can only switch digital data, therefore analog signals must first be converted to digital signals.

3.2.1 Structure of the Telephone Network

The telephone network has been designed to carry a frequency range of 300 to 3300 hertz (Hz) with a harmonic distortion better than 26 decibels (dB) and a signal to noise ratio better than 30 dB. While this is acceptable for voice transmission, consider the needs of high quality audio which ranges from 20 to 20,000 Hz. ISDN designers must meet the needs of all users to achieve true integration and transparency.

The structure of today's FDM based telephone networks is based on three types of links: local, tollcollecting trunk, and intertoll trunk, described below.

1. **Local** A local connection is one that joins a telephone subscriber to the local exchange.The physical distance is short.

2. **Tollcollecting trunk** - A tollcollecting trunk is one that connects a local exchange to a switching center.

3. **Intertoll trunk** An intertoll trunk connects two switching centers, both nationally and internationally, over long distances.

The local exchange switches all calls within its local network and passes all long distance calls to the nearest switching center. The switching center either passes the call to another local exchange within its area or passes it to a distant switching center.

Each telephone in the local network is connected to the exchange by a single pair of wires which are used for ringing the telephone bell and conveying the transmitted and received voice signals. This single pair of wires is usually the cheapest method of transmitting a voice frequency message between two terminals provided the distance is short (<8 km.).

When many telephone channels are to be transmitted via a single physical link they are multiplexed using FDM. To illustrate, suppose that ***n*** telephone channels, each having a bandwidth f_b of value **4 kHz** (including **2n** guardbands) are to be multiplexed by the technique of FDM. The multiplexer is arranged to modulate the telephone signals taken consecutively with

carriers of frequencies f_1, f_2, ..., f_n. The modulated signals will contain both positive and negative guardbands (neutral areas above and below the carrier frequencies), one of which must be removed by filtration otherwise they use up valuable bandwidth. If the values of the carrier frequencies are chosen such that each carrier is separated from the next by an amount at least as large as f_b, and the resultant signals are mixed together, then a continuum stretching from f_0 to f_n will be obtained.

3.2.2 Digitized Voice

The first step in utilizing high speed digital switches is to convert the analog signals to digital signals. The most commonly employed method for doing this is called Pulse Code Modulation (PCM). PCM was patented in 1939 by Sir Alec Reeves of ITT and involves sampling the analog signal at regular intervals and coding the measured amplitude value into a sequence of pulses {28}. Sampling is defined here as the instantaneous measure of the amplitude value of a signal.

PCM consists of three separate operations: sampling, quantizing and coding. Most equipment will sample a voice grade line with a range of 300 Hz to 3.4 kHz at 8 kHz. Nyquist's theorem says that if an arbitrary signal has been run through a low pass filter of bandwidth H, the filtered signal can be completely reconstructed by making only 2H samples per second. Sampling the line faster than 2H times per second is pointless because the higher frequency components that such sampling could recover have already been filtered out.

{28} So if the signal consists of *V* discrete levels, Nyquist's theorem states:

$$\textbf{maximum data rate} = 2H \log_2 V \text{ bits/sec.}$$

Therefore a noiseless 3 kHz voice channel, with 2 discrete voltage levels, cannot transmit binary signals at a rate exceeding 6 Kbps.

It must be remembered that these are noiseless channels. When noise is introduced the maximum bit rate decreases rapidly. Noise is measured by the ratio of signal power to the noise power, called the signal to noise ratio. If **S** is the signal power and **N** is the noise power then the signal to noise ratio is **S/N**. Usually it is not this ratio that is quoted but rather :

$$10 \log_{10} (S/N).$$

These units are called decibels (dB). So a signal to noise ratio of 10 is 10 dB, a ratio of 100 is 20 dB, a ratio of 1000 is 30 dB and so on. For noisy channels we refer to Shannon's limit which says the maximum number of bits/sec =

$$H \log_2 (1 + S/N)$$

where H is the bandwidth. In a 3 kHz channel with a signal to noise ratio of 30 dB, as is typical in a telephone system, no more than 30,000 bps may be transmitted no matter how many signal levels are used or how frequently the samples are taken.

A PCM sampling system actually requires the following components:

- *Input filter* this ensures there will be no frequencies in the signal above the maximum for a voice grade line, about 3.3 kHz.

- *Sampling circuit* interrogates the message signal at regular intervals (usually 8000 Hz, encoding this in 8 bits).
- *Holding circuit* This circuit holds the amplitude value of each impulse for the duration of the sampling interval. It is used at the transmitter to give a PCM encoder sufficient time to perform a series of operations that result in the coded pulse pattern. At the receiver the circuit is used to 'stretch' the narrow impulses that are provided by demultiplexing the composite time division multiplexed signal. The stretched signal is a rectangular wave which approximates the message waveshape.

The second step in PCM is quantization. This is the name given to the process of approximating the individual message signal samples to the nearest permitted voltage reference level. The error introduced by this approximation is called quantizing noise and can be a major impediment to PCM transmission. In sound transmission this error will cause a continuous background noise while in television and image transmission it causes the number of grey tones that exist between black and white to be limited. The closer the approximation of the analog signal the less noise that will be present. Figure 3.1 shows an approximation of a continuous message signal using a constant sampling rate and a discrete step size delta ***V***.

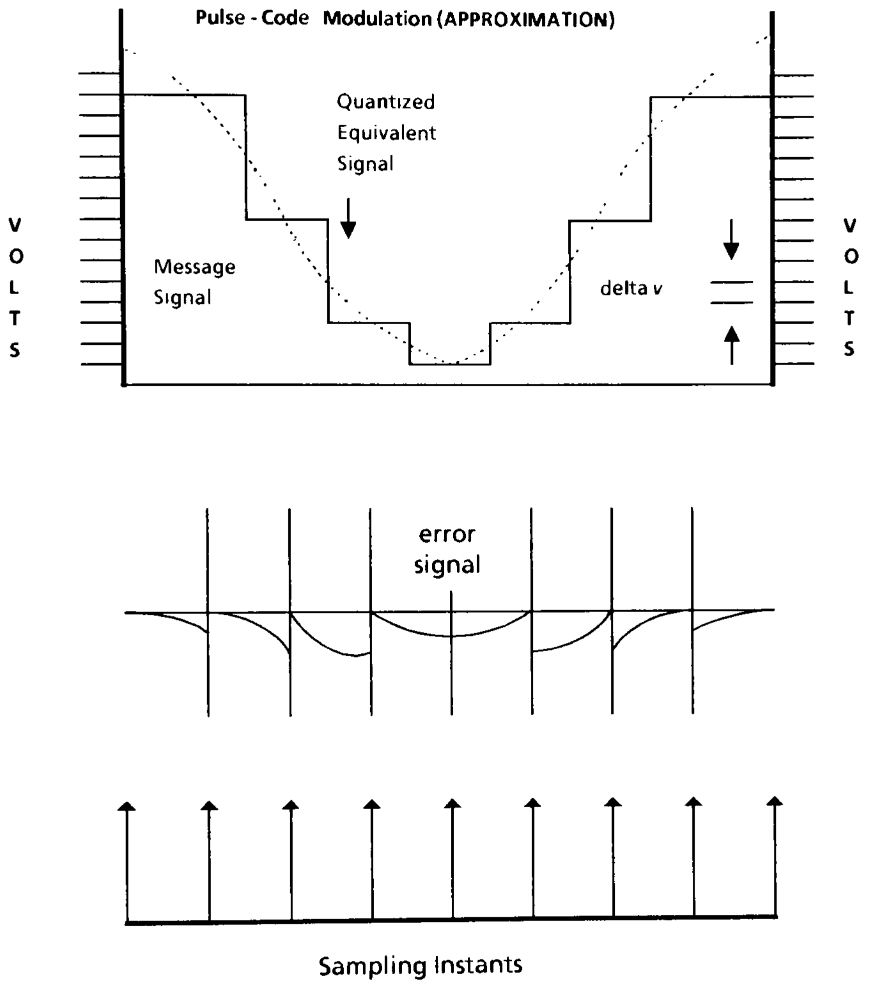


Figure 3.1. Uniform quantization of a sine wave.

By increasing the number of steps and correspondingly decreasing delta *V* we can more closely approximate the signal and reduce quantizing noise. We mentioned earlier that standard PCM samples at a rate of 8000 Hz and uses 8 bits to represent each sample. This means there are 256 ($\log_2 256 = 8$) possible voltage levels we can represent in each sample. It also means we need

a bandwidth of 64 Kbps to transmit our signal (8 bits x 8000 Hz), hence the planned ISDN B channel rate of 64 Kbps. Therefore the tradeoff is in bandwidth against accuracy in the reproduced signal.

There are many different methods of quantizing and many studies have been made to improve its performance. Classical PCM requires each discrete value to map to some fixed voltage. Other techniques involve communicating only the changes from the previous level rather than the level itself. This is called differential PCM. A particular form of differential PCM, called delta modulation, limits the amount of change to one discrete level. Thus in delta modulation, the shape of the message signal is communicated by informing that either a positive or negative change has occurred since the last sample. This can cause a problem if the slope of the signal is rising or falling very quickly. To overcome this one must increase the sampling rate and increase the quantum level (delta V). In cases where the sampling rate is not a factor delta modulation may offer improved performance.

Another method, called predictive quantization, attempts to predict, by using previous knowledge of the signal, the most likely value of the next sample and communicate only the difference between the guessed value and the sample value. For media where the basic characteristics of the message signal are known predictive quantization can reduce the needed bandwidth compared to conventional PCM. In the case of speech messages a form of predictive quantization called linear predictive coding (LPC) offers the ability to have spoken words stored in semiconductor memories (speech synthesis) along with lower bit rates. {28}

The last step in PCM entails analog to digital conversion: coders. Here is an example where 16 discrete levels from a sampled message signal are to be represented by 4 bits ($2^4 = 16$). A bit defines two logical states, 1 and 0. These can be specified electronically very easily say a voltage level of 5 represents a 1 and ground potential represents a 0. To save these values in memory a device called a flip flop is used. This can maintain equilibrium in either of the two states.

This coder would include a series of flip flops connected so that they formed a binary counter. Such counters range from straightforward asynchronous ripple types to very complex synchronous ones. This simple counter would consist of 4 flip flops which would initially be set to 0000 before each sampling instant. After sampling, the correct number of clock pulses would be supplied to the counter to produce the binary value of the sample. With 16 possible sample values and 4 flip flops any one of the 16 values can now be represented as a digital number from 0000 to 1111. Just as there are many types of counters there are many types of coding schemes. Of special interest are symmetrical codes because many message signals, like speech, extend about equally above and below the quiesent level. {28} A symmetrical code designates the first bit as being positive or negative and the remaining bits measure the magnitude of the signal above or below the quiesent level. Thus fewer bits are needed to encode the signal. Most coders are serial coders but in cases where extremely fast encoding is required as in the coding of television signals groups of coders may be arranged in parallel. Once the analog signals have been coded they must be multiplexed together using a TDM scheme.

3.3 Time Division Multiplexing

3.3.1 Introduction

The fundamental circuit element in any time division multiplexor is called the *serializer*. This circuit accepts the parallel channel inputs and allows each an output time slot. Depending on the length of the time slot the output signal is termed word interleaved, character interleaved or bit interleaved. Timing for the serial output is provided by a quartz crystal oscillator which is the master clock and specifies all timing functions within the multiplexor; see Figure 3.2. If all channel signals are derived from the same master clock they are called synchronous, otherwise they are called asynchronous.

3.3.2 Synchronous Time Division Multiplexing.

In order for the demultiplexor to synchronize or align with the multiplexor a predetermined sequence of bits must be interleaved with the information time slots. Such synchronizing bits lead to the idea of a frame which is defined as a set of consecutive digit time slots in which the position of each time slot can be identified by reference to a frame alignment signal. {28} The output bit rate of the multiplexor must be slightly higher than *n* times the channel bit rate where *n* stands for the number of channels. The frames, called frame alignment words (FAW), take up a finite number of time slots, usually n + 1 or n + 2. The extra 1 or 2 bits are for framing. If this is difficult to follow an example of an actual frame format, the Bell T1, will be displayed shortly. Thus the multiplexed output bit rate f_0 for a frame structure based on n + 2 time slots, where the

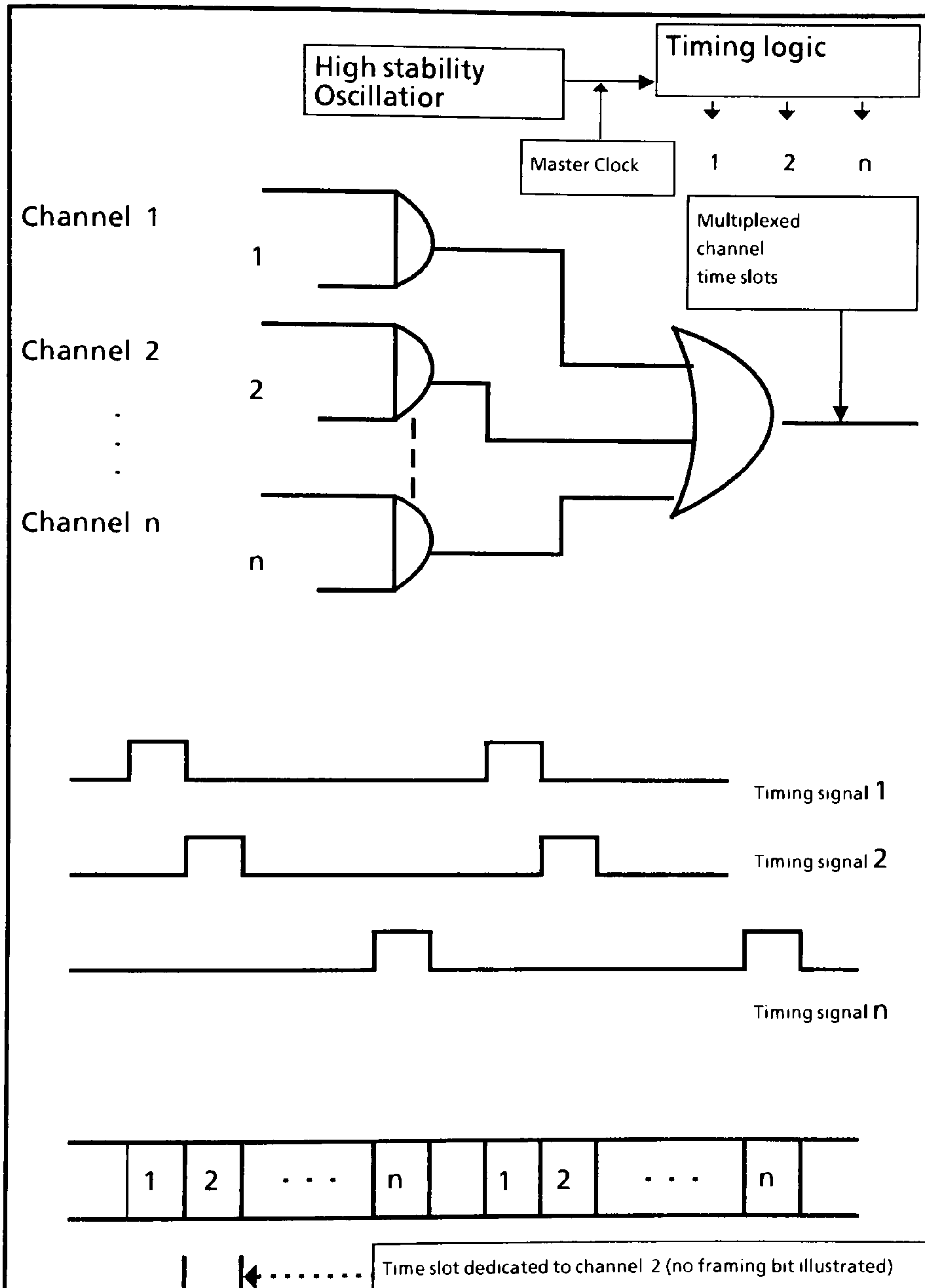


Figure 3.2 Time Division Multiplexing

analog channel messages are sampled at a frequency f_s and are encoded in w digits is given by:

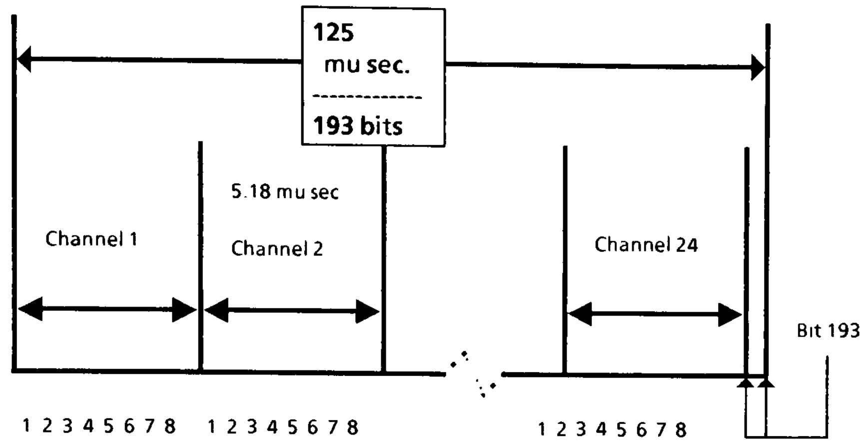
$$f_0 = wf_s(n + 2)$$

The two time slots allocated for frame alignment may be located anywhere in the frame.

The choice of the frame alignment word can have a considerable effect on the performance of the system. The FAW detection circuitry usually includes a shift register. When the sync pattern, which is a bit pattern that is not easily imitated by data, is shifted into the register the pattern is detected and the multiplexors are in sync. If this pattern is easily imitated false synchronization may occur. Also the size of the FAW should be kept fairly small. Large frames should be avoided because they are more likely to be corrupted by transmission errors, the synchronization time is longer and they require a larger buffer.

When problems do occur the CCITT recommends that alarms be included in all multiplexing equipment. The recommended alarms are Outgoing Signal Loss (OGSL), Incoming Signal Loss (ICSL), Frame Alignment Loss (FAL), Remote Frame Alignment Loss (RFAL), High Error Rate (HER) Encoder Fault (ENCOD), and Alarm Inhibit Signal (AIS).

Figure 3.3 shows the Bell T1 (DS1 digital signal) TDM and its frame format. The T1 includes 24 voice channels (CCITT recommendation G733). Each of the 24 channels are sampled at 8 kHz and encoded as an 8 bit word. The resultant channel messages are word interleaved forming a sequence of 192 bits. A



1. Bit 193 is a framing bit, used for synchronization.
Voice channels: 8- bit PCM used on 5 of 6 frames.
7- bit PCM used on every sixth frame, bit 8 is a signalling bit.
Data channels: Channel 24 used for signalling only in some schemes.
Bit 8 is a control bit.
Bits 1 - 7 used for 56 kbps service.
Bits 2 - 7 used for 9.6 kbps, 4.8 kbps, and 2.4 kbps service.

Figure 3.3 DS-1 Transmission Format

single framing bit is inserted at the start of each sequence giving a total frame length of 193 bits. Thus our data rate equals:

8000 samples per sec. x 193 bits per sec. = 1.544 Mbps.

For five of every six frames 8 bit PCM is used. Every sixth frame contains a 7 bit PCM word plus 1 signaling bit. This signal bit, when combined with other

signaling bits into a bit stream, contains network control and routing information used to establish a call or terminate a connection, for example.

This same DS-1 format is used to provide digital data service, as opposed to voice service. In this case 23 channels of data are provided. The twenty fourth channel position is reserved for a sync byte which allows faster and more reliable reframing following a framing error. {43} Each channel contains seven bits for data with the eighth bit used to indicate whether the channel, in that frame, contains user data or system control data. With 7 data bits sampled at 8000 Hz a data rate of 56 Kbps is provided per channel.

Lower data rates are provided by a process called subrate multiplexing. An additional bit is used to indicate which multiplexing speed is to be provided. If the bit is set then one of three slower rates will be selected, depending upon which rate had been previously assigned to that channel. If one of the data bits from the example above is used we are left with 6 x 8000 = 48 Kbps. This capacity is used to multiplex five 9.6 Kbps channels, ten 4.8 Kbps channels or twenty 2.4 Kbps channels. As an example, if channel 4 provides 4.8 Kbps service, then up to ten data sub-channels share this channel. The data for each appears as six bits in channel four in every tenth frame.

To achieve data rates higher than the DS-1 speed of 1.544 Mbps bits are interleaved from DS-1 inputs. The DS-2 transmission combines four DS-1 inputs into a 6.312 Mbps stream. Data from the four sources are interleaved 12 bits at a time. A TDM hierarchy exists and is defined up to DS-4 which can carry 4,032

voice channels (North American) at data rate of 274.176 Mbps. Figure 3.4 is a detailed look at the ATT TDM hierarchy {45}.

It is interesting to note the variety of signals entering level DS-1, shown by the wide horizontal box. These include synchronous and asynchronous data and analog and digital voice. These signals can be further multiplexed and combined with high bandwidth transmissions like digital television up to the DS-4 level. The physical transmission capacities of some carriers, both analog and digital, are summarized in table 3.1.

The DS-1 format is used extensively in North America today for both voice and data service. The data service is known as Digital Data Service (DDS). The DDS provides digital Data Service at data rates of from 2.4 to 56 Kbps. The service is provided at customer premises over two twisted pair lines. It should be pointed out here that the channels entering the multiplexor do not always contain data. To overcome this inefficiency time division multiple access (TDMA) devices are used. Such devices, often referred to as statistical multiplexors, rely on statistical methods to reduce the data rate of the output channel to less than the sum of the input channels. To further increase the number of voice channels accommodated in a system a technique called time-assignment speech interpolation (TASI) is used. TASI devices use speech detectors and high speed switches to connect other users when a given route contains no speech signals.

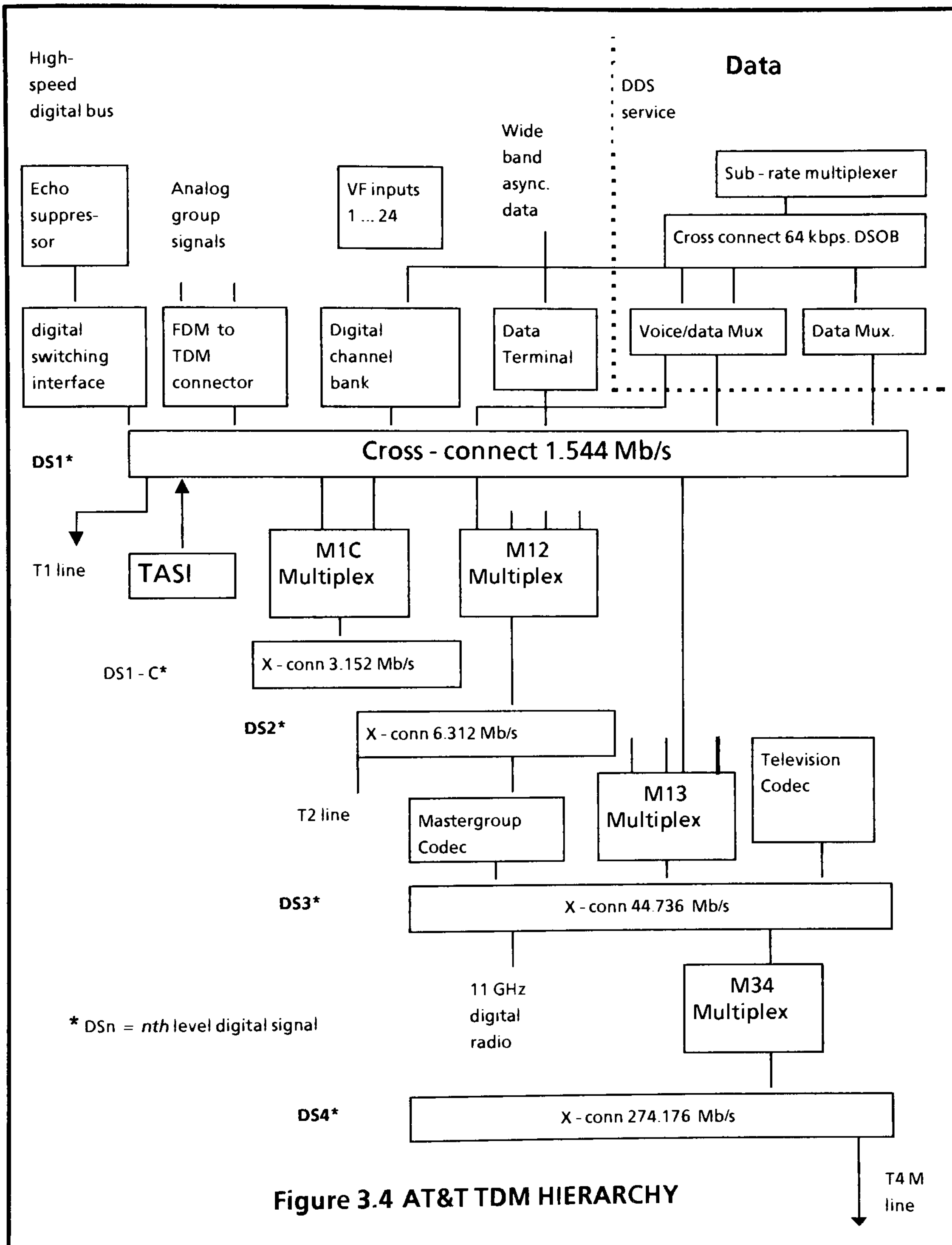


Figure 3.4 AT&T TDM HIERARCHY

Transmission Medium	Designation	Transmission	Number of Voice channels	Operating Frequency	Data Rate (Mbps)
Twisted pair	N3	Analog	24	0.172-0.268	
	T1	Digital	24		1.544
	T2	Digital	96		6.312
Coaxial cable	L1	Analog	600	0.006-2.79	
	L4	Analog	3600	0.564-17.55	
	L5	Analog	10.800	3.12-60.5	
	T4	Digital	4032		274.176
Optical fiber	FT3	Digital	672		44.736
Microwave	TD3	Analog	1200	3700-4200	
	TH1	Analog	1800	5925-6425	
	TN1	Analog	1800	10.700-11.700	
	11-GHz	Digital	672		44.736
	18-GHz	Digital	4032		274.176
Satellite	Intelsat V	Analog	24.000	6/4-GHz band	

Table 3.1

Capacity of Some Communication Carriers

3.3.3 Asynchronous Time Division Multiplexing

Up to now the discussion has dealt with synchronous multiplexing. But what happens when the incoming information streams have been derived from different clock sources? A technique known as justification or pulse stuffing must be used to synchronize each of the channel signals to a common clock reference. Once this has been performed the channels may be synchronously multiplexed.

The first step is to define a frame format such that

$$n = {}^{t}/_{m-t}$$

where n is the ratio of information to control digits , t = input bit rate, and m = multiplexed bit rate. Thus the proper number of control bits are inserted. A problem occurs however because each clock being used is capable of drifting one way or the other, thus not maintaining the same bit rate at all times. To get around this justification control bits are added to the frame. They signal that a clock has drifted past a predetermined threshold and how much corrective action is required. Until all data sources are synchronous such speed adapting techniques will be required when multiplexing asynchronous data over synchronous digital networks.

3.3.4 Transmission of Digital Signals

It is worth a short digression to explore how digital signals are transmitted over long distances. A digital transmission system can be broken down into three component parts: line terminal apparatus, repeaters, and transmission media. The transmission medium, be it twisted pair, coax, optical fiber or microwave, will exhibit certain characteristics which the other components in the system must attempt to overcome such as attenuation, phase shift and crosstalk.

Binary signals must first be coded in a form which allows a constant dc level, a suitable energy spectrum and an adequate timing information content. {28} One such code is Alternate Mark Inversion or bipolar code. Each logical 1 pulse (a mark is the name given to the *on* state when a pulse is present) is

interpreted as a positive or negative mark, taken alternately, while the binary logical zero condition continues to be transmitted as such. Figure 3.5 shows AMI and its energy spectrum relative to a straight binary signal. The chosen line code should have negligible energy at low frequencies to limit the complexity of the circuitry and to reduce the attenuation introduced by the cable. {28}

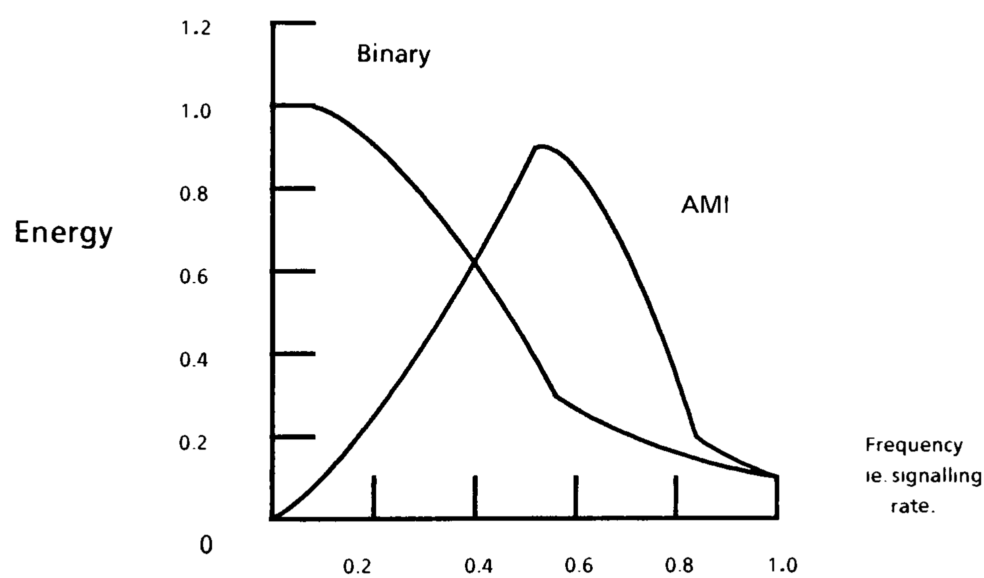
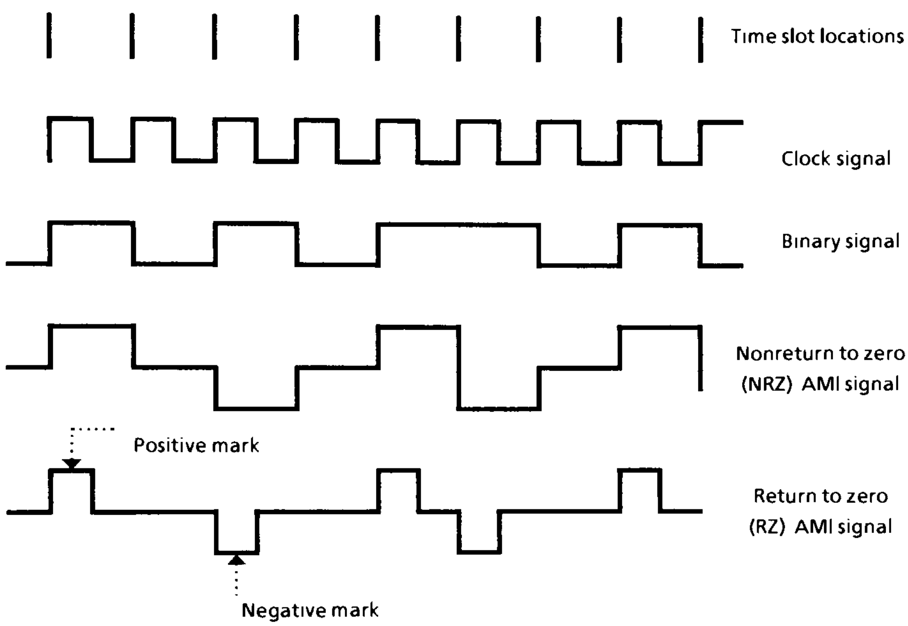


Figure 3. 5 Relative Energy Spectra of Binary and AMI Signals

If the transmitted marks occupy a full time slot they are called non return to zero (NRZ). Usually, however, a scheme called return to zero (RZ) is used. This

is because the energy spectrum exhibited by the code often makes it difficult for the receiving end to extract timing information from the transmitted line signal. In an RZ format (still utilizing AMI) the signal must first return to zero before exhibiting another mark, making it easier, because of less sophisticated circuitry, to extract the signal. The more times the signal crosses zero the easier it is for the clocking circuit. Repeaters and line termination apparatus usually reconstruct the original signal from the received line signal. When a square shaped pulse is transmitted over a cable it will be attenuated, dispersed and effected by random noise. Also it can be effected by interference from other cables, called crosstalk. Therefore limits must be placed on the amount of signal degradation allowed before it is regenerated. This distance will depend on several factors including transmission medium, transmission code, and load impedance. Figure 3.6 shows examples of the transmission of digital signals.



Alternate mark inversion (AMI) coding of a binary plus clock signal.

Figure 3.6 Transmission of Digital Signals

3.3.4 Digital Switching

Up to this point the discussion of transmission structure has covered the basic telephone network, conversion of analog signals to digital signals (PCM), and the multiplexing, coding, and transmission of these signals. It will now look at how these digital signals are switched. Switching is a key element in any telecommunications network. Without switching the network would require, assuming **'n'** stations, **(n * (n-1)) / 2** direct connections. This of course becomes cost prohibitive very quickly as more stations are added to the network. The next chapter discusses in detail the different switching techniques such as circuit, packet, and message switching as they relate to different information media.

Digital switching became economical with the advent of high speed microprocessors. Control of the switching operation is handled by a subsystem called SPC (stored program control) which is simply a set of instructions stored at each switch. The SPC maintains a permanent list of all subscriber destinations called a routing table. This list is updated as needed and can be done so dynamically as in the case of failure in a line or switch. Such dynamic routing schemes make the network more robust and make the transport system even more transparent to the user.

In our PCM/TDM model digital frames would arrive at the switch containing word interleaved pieces of channel messages. These would be stored, moved to the proper outgoing line, reformatted and sent to their destination. This is also referred to as space and time division switching. Such switching has

proven to be very economical as well as attractive in its ability to offer enhanced services such as speed calling, call waiting, call forwarding and don`t- disturb. Combined with common channel signalling, SPC will offer a large variety of features and services in ISDN.

3.3.5 Common Channel Signalling

The last topic in this chapter concerns common channel signalling. Look at figure 3.7. This shows three types of transmission services available under the initial implementation of ISDN. These are circuit switched, packet switched, and common channel signalling. CCITT has recommended Signalling System #7 as its channel signalling protocol. While circuit and packet switching technologies are familiar to most users of data networks, channel signalling has its roots in the telephone system. SS#7 is, however, a protocol and its bottom three layers correspond closely to X.25 and the OSI model. SS#7 will play a key role in ISDN due to the real time functions that a communications signalling system must provide: call establishment and billing, financial services, and supervision of the connection. {47} Signalling systems must provide the following functions:

* **Speed**- routes must be found, tested and established while the caller waits on-line. Significant resources are devoted to this setup but no billing can be started.
* **Accuracy**- errors can result in misconnected calls or improper billing.
* **Reliability**- if the signalling system fails the entire communications facility is useless. The design must accommodate redundancy and alternate routing capabilties.

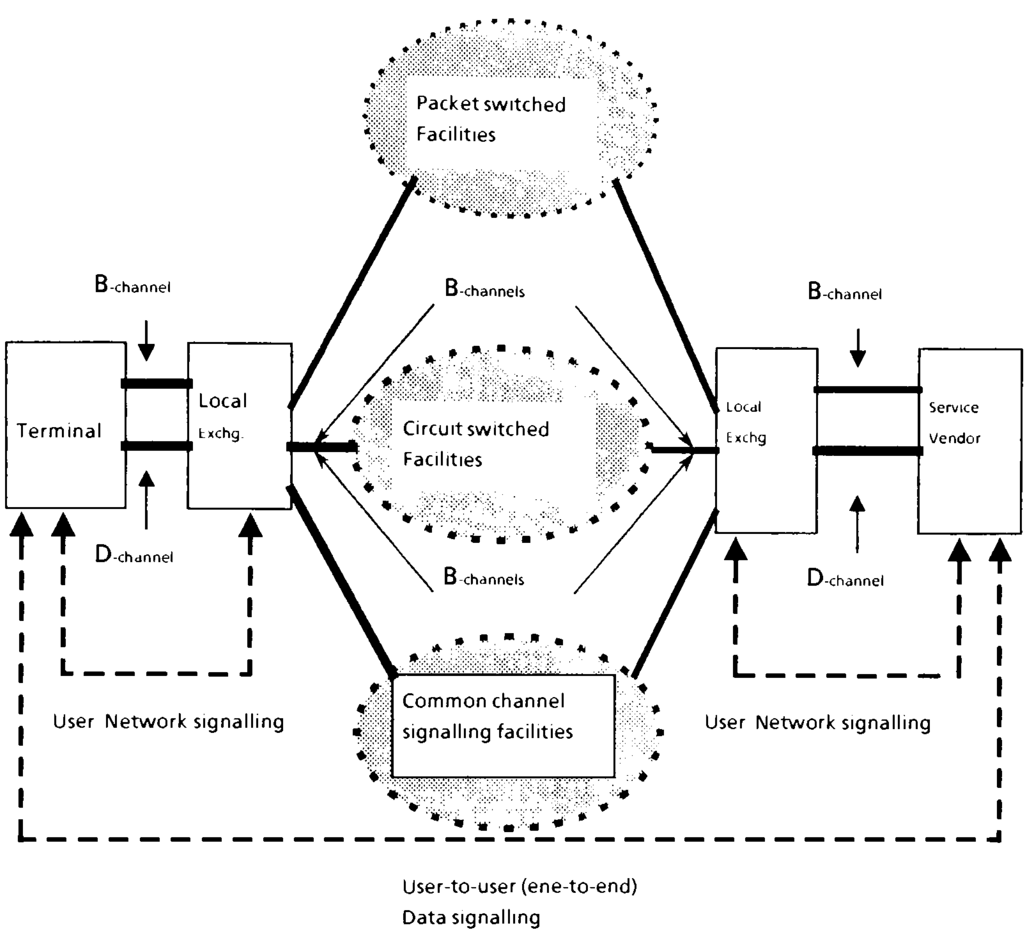


Figure 3.7. The three cloud network

* **Transaction Orientation**- establishing a virtual circuit for every signal would be impractical because of the time delay. Therefore a datagram or connectionless form of signalling is used.

Signalling has always played an integral role in telephone systems. Tone signalling in subscriber instruments and trunks allows for an efficient transfer of information. These tone signalling systems are channel-associated, that is

each speaking path carries its own signalling messages. Common channel signalling becomes very attractive when a central processor provides all the routing functions for a number of paths. A data link carries the signalling information for a number of speaking channels. The data link is the common channel, thus the term common channel signalling.

SS#7 is a special signalling system optimized for digital traffic with low error rates and speeds of up to 64 Kbps. SS#7 is a layered architecture and its lower 2 layers are in full conformance with the OSI physical and data link layers. The three lower layers are collectively called the Message Transfer Part (MTP). The MTP provides a service very similar to X.25. It accepts packets of data and reliably delivers them to their destination {47}. Layer 3, called Common Transfer Function, has been supplemented with an additional sublayer called the Signalling Connection Control Point (SCCP) to functionally resemble the OSI network level. Most of the higher levels have been combined into something called the 'user part', except for an operations and maintenance layer responsible for signalling message route creation, route verification, measurement collection, event reporting, clock initialization, real-time control and testing.

The physical layer of SS#7, as mentioned, is designed and optimized for high speed digital links. Because ISDN will be implemented piecemeal, lower speeds, switched connections and analog links are allowed. The links are full duplex and may be routed by satellite.

The data link layer is bit oriented like HDLC, SDLC, and LAPB but it has a unique frame format. A standard flag (01111110), bit stuffing techniques and

checksum (CRC-16) are used but there is no address checking at this layer. Because messages are usually short no NAKS (negative acknowledgements) are sent, only ACKS. Messages are retransmitted when the ACKS are out of sequence. Flow control is provided by the use of a status signal unit sent by the receiver when congestion occurs. High error rates caused by noisy physical links are detected by up-down counters. High error rates cause an alarm to be dispatched and the link to be taken out of service.

Layer 3, the Common Transfer Function, provides for routing (where the message must be sent), discrimination (whether or not the message has reached its destination), and distribution (deciding which User Part should receive the message). A 32 bit label is used for addressing. It is composed of two 14 bit origin and destination addresses and a 4 bit field used to distribute traffic among alternate routes. The SCCP provides additional addressing. Speed and reliability are achieved through this routing and flow control scheme. Signals may take a different path from the call in order to ascertain they will reach the destination.

The remaining User Part has not been fully defined. There is a separate layer which will be devoted solely to ISDN but it remains to be seen what role this level will play. At any rate the importance of Signalling System Number 7 cannot be understated. It will be the central nervous system of ISDN.

Basis of comparison	CCITT X.25	CCITT NO. 7	Comments
1) Function	Procedure for connecting data equipment to packet network	Procedure for Common Channel Signalling for establishing circuit switched telephone and data calls. Includes: Call Control, Management and Maintenance Signalling	
2) Functional Division	One: Data Communications	Two: (1) Message Transfer Part (Specifies the data communication function and its performance) (2) Specific User Parts	
3) Protocal Structure	Three Levels	Four Levels	The three lower levels of No. 7 are equivalent to the three levels of X.25
4) Modes of Operation	Switched Virtual Circuit (SVC) or Autoconnect (AC)	Pre-established path equivalent to X.25 Autoconnect	
5) Level 1	Bit Rate Independent Network Links (2.4 to 56 KBPS)	Optimized for 64 KBPS. Applicable down to 4.8 KBPS	
6) Level 2	HDLC	No 7 Level 2	Comparison follows.
7) Flags	01111110	01111110	
8) Zero Insertions Deletion	Yes	Yes	
9) Block Formats	a. Information Frame b. Supervisory Frame c, Unnumbered Frame	a Message Signal Unit (MSU) b. Link Status Signal Unit (LSSU) c. Fill in Signal Unit (FISU)	
10) Block Sequence Number	4 or 8 bit field	Not required in No. 7	
11) Outstanding Blocks	B	12B	
12) Address Field	B bits	Not required	
13) Error Control	CRC	CRC	
14) Error Correction	Timeouts and/or NACKS	ACKS,NACKS,length threshold	
15)Connection Establishment	Packet interchange or Autoconnect	Route pre-established	
16) Routing	By packet header	By routing label	

Table 3.2 **Comparison of X.25 and CCITT Signalling System No. 7**

To achieve true integration and efficiency of operation high speed digital switching technology will be required. This chapter examines circuit, message, packet, and hybrid (combined circuit and packet) switching technologies.

4.1 Introduction

Switching is an essential element in an economical and flexible communications network. The various characteristics of users of such a network have led to the development of several different switching techniques. This chapter will compare these techniques particularly in terms of their satisfaction of user needs.

Circuit Switching Every time a phone call is made an actual physical connection is established between each pair of callers for the entire duration of the call. A continuous two-way path is provided between each pair of users. A circuit switch must have two fundamental characteristics:(1) it must maintain a high degree of transparency across the switch matrix. The minimum requirement is to approximate a pair of wires with a bandwidth of 4000 Hz, compatible with the transmission characteristics of the world-wide telephone networks; (2) New switches must be "backward compatible" with the technology already in place because the system is so standardized and well used.

With voice communication, where the information content is high, use of such a circuit is fairly efficient. However, when we use a circuit to pass information between a keyboard terminal and computer, with various amounts of information bursts, maintaining that physical connection is very inefficient.

Contrary to circuit switching, packet switching involves moving information from place to place on an as-needed basis, where the amount of information, routes and the endpoints change with time. The calls, messages or transactions are divided into pieces called packets. The maximum message length is severely restricted and is based upon queuing principles and blocked message delay. Depending on the form and implementation of the information there may be more than one level of subdivision, ie - segments and segments further divided into packets.

Each switching center after receiving a packet "holds" a copy of the packet in temporary storage until the switch is sure that it has been received properly by the next switch or by the end user. This is called hold-and- forward because the copies of the packets are written over in memory (destroyed) when the switch is sure the packet has been successfully relayed.

This form of operation permits the network to achieve low overhead for short messages and eliminates the set up time for calls going through circuit switched networks. By moving packets through the network in almost real time, the switches can adapt their operation quickly in response to changing traffic patterns or the failure of some network facility.

In a message switching network each switch stores the message in its entirety, giving very reliable service. Traffic is delayed (stored) until capacity is available to deliver currently stored traffic. Message switching is mainly used for writer to reader communications such as telegrams. Message switching is often referred to as a record communications because the final output of the

communications process is a tangible product in a readable format, providing a permanent record of the information transfer. {34}

Another point to note is that in message switching the sender and receiver do not have to be simultaneously available to the network. This is not true of most circuit and packet networks. For example, one cannot place a telephone call unless a complete circuit can be established. In a packet network, unless there is some kind of storing service, the receiver must be available when the packets arrive. The transactional nature of the message switching process makes it difficult to conduct communications between two users on a real-time basis.

Message switching systems normally record all communications, either permanently to a disk or tape file or in a temporary buffer. The message storage feature is a direct consequence of the processing technique employed by the switches. Switch processing is on a store-and-forward basis, ie. each message is stored in its entirety at each switch before it is forwarded to the next.

It is unlikely that message switching will be in great demand under ISDN. Some of its features, however, such as not requiring both users to be active on the network and permanent records of messages, may well be built into ISDN. An example would be a central device where users could "logon" to check for mail, facsimile documents, etc. Here entire messages or documents could be stored until a user had the chance to examine them.

More likely ISDN will offer a choice between circuit switching and packet switching along with common channel signaling. Circuit switching has been in

user for a better part of a century. Packet switching can be traced back to the mid 1960's. Table 4.1 contrasts the basic technical aspects of circuit switching and packet switching.

Circuit	*Feature*	*Packet*
Call establishment and disestablishment	PROTOCOL	Interface and data flow
Fully transparent	TRANSPARENCY	Transaction oriented
Inherently full duplex	DUPLEX OPERATION	Independently bidirectional
Voice call statistics	HARDWARE	Data call oriented
Line dominated/ scan oriented	SYSTEM LIMITATIONS	Origination dominated/ interrupt driven

Table 4.1 Circuit vs Packet Switching

Protocol Circuit switching requires a call establishment and disestablishment protocol. Packet switching is protocol dependent throughout its operation. Well defined protocols interface each of the users to the network and guide the data flow through the network.

Transparency Circuit switching is fully transparent. Packet switching is transactional. Transactional vs transparent will be discussed a little later.

Duplex Operation Circuit switched connections are inherently full duplex, meaning that both sides can simultaneously send and receive information. Packet switching moves a packet or transaction at a time through the network. Using separate circuits in each direction is independently bidirectional meaning that the transactions can be simultaneously flowing in each circuit, but at any time the information going in one direction is independent of the information flowing in the other direction.

Hardware Circuit switched networks' hardware is keyed to voice call statistics. This means the hardware can look at the line every second or two to see if the user wants any service from the switch. Packet switching hardware is data call oriented. Interswitch lines are set up for quick response from the switches.

System Limitations Circuit switching is line dominated; limited in the number of lines that can be handled at any one time. It is also scan oriented meaning the switch call processor scans line by line looking for new requests. Packet switching is origination dominated; the information flow is dependent upon the number of transactions originated in the network. The processor is also interrupt driven. The initiation of a new transaction is permitted to interrupt the processor from one task to initiate action on the new request.

4.2 Comparison of Switching Techniques

To briefly summarize the advantages and disadvantages of the three switching types we have:

Circuit Switching

Advantages - Compatible with voice, commonality of calling procedures.

Disadvantages Subject to blocking, requires terminal compatibility, large processing and signalling burden.

Message Switching

Advantages - Permit code and speed conversion in switches, appears non-blocking, high efficiency and channel utilization.

Disadvantages Large variance in delay, poor responsiveness for interactive traffic Information accessibility and privacy. Requires powerful processors and large storage.

Packet Switching

Advantages Rapid exchanges of short messages, Provides features of message switching network, flexible and adaptable.

Disadvantages Uses many small processors, employs complex routing and control.

4.3 Transactional and Transparent Networks

In a transparent network the switches act as a bridge such that the end- to-end connections formed within the network functions as a pair of wires between the users. {34} Information that goes into the connection at one end comes out at the other end in real-time, without intentional change. Of course, there are slight unintentional malfunctions, such as delays due to the physical speed of propagation through the medium, amplitude, phase or frequency distortions of transmitted information, and interference. Other than the processing of the call set up information needed to establish the connection, no processing is done within the network. The best example of a transparent network is the voice telephone service.

In a transactional network information is introduced into the network in the form of a complete entity. The network guarantees acceptance of the information and somewhat guarantees delivery of the information. Much like a letter dropped in a mailbox, information that comes into the network is

physically disconnected from the input that originated it. The information will come out of the system at the destination address; you just are not sure exactly when and in what condition. When the information does come out of the system it represents a complete transaction. In a communications transaction, the contents of the information package arriving through the network have usefulness at the destination end, without the need for any additional information. The best examples of transactional networks are message networks, such as Telex and telegrams. Packet switching is also a transactional network.

The distinction between transparent and transactional switching is less identifiable as other forms of communications processing are introduced to a network. For example, with terminal polling in a computer network a number of computer terminals are linked to a common connection. As each terminal takes a turn in sending information over the shared circuit, the network looks like a transparent connection. Since the terminal may have to wait its turn before each transmission, the network is not truly transparent. Even packet switching can be implemented in a "virtual circuit" mode, which attempts to approximate a transparent connection.

There are also some very important technical distinctions between transactional and transparent - See Table 4.2 and the ensuing discussion. {34}

User Interface In transparent, calls are presumed to last for a few seconds or minutes. The time interval is long enough that the switch can scan each line looking for a request. In a transactional system,

Transparent	*System Attribute*	*Transactional*
Scanning	User Interface	Interrupt
Synchronous	User Timing	Asynchronous
Fixed	Bandwidth/Bit Rate	Adaptable
Dedicated	Resource Allocation	Protocol
Fixed	Routing / Overhead	Variable
Compatible	Terminal Design	Separable

Table 4.2 Attributes of Two Switching Approaches

transactions arrive more frequently. When the transaction has arrived at the switch, the processor is interrupted to accept and begin processing this new transaction.

User Timing Refers to the continuity of the communications flow through the switched network. Transparent systems attempt to preserve the time relationships between users. This is called synchronous. The time spacing and temporal relationships are not preserved in a transactional network. This is called asynchronous.

Connection Bandwidth or Bit Rate - This refers to the amount of capacity made available to a user when a connection is established through the

network. In a transparent network this capacity is fixed, which maintains the transparency. Network capacity is adaptable and assigned on an as needed basis in a transactional network.

Resource Allocation - In a transparent network the resource allocation is done through a dedicated line to a user pair over the duration of the information transfer. Protocol is the rule or procedure by which the information is transported through the network. This is the transactional approach by which resources are allocated.

Routing/Overhead In a transparent network the routing of information through the network is fixed at the beginning of the information flow and remains the same for the duration of the transparent connection. The overhead information required to establish communications is also fixed and is the same whether the information exchange is of long or short duration. In a transactional network routing is easily made variable in order to achieve more uniform and efficient utilization of all available resources. Since each transaction must employ a set of protocols, the overhead is also variable and increases as the length of the information transfer increases.

Terminal Design Transparent switching requires that the end terminals be compatible ie: common formats, codes, languages, signaling, and data transfer rates. Different user terminal characteristics can be used at either end in a transactional approach. The network processing elements have time to do conversion between one set of terminal characteristics and another. Therefore, the transactional network is considered separable.

4.4 Combined Packet and Circuit Switched Networks

4.4.1 Overview

Now it's time to show that it is possible to combine packet switching and circuit switching within one network to take advantage of both techniques. For short, bursty communications packet switching offers advantages in terms of delay, processing and system overhead. {34} For longer messages, however, circuit switching is more efficient.

Since communications characteristics are quite diverse there is a need for networks that can efficiently mix traffic with a wide variety of data rates and traffic statistics. By separating communications traffic into three general groups: continuous, bursty and interruptible, the characteristics of each can be studied. {34} Continuous traffic is characterized by a continuous flow of information over a fixed communications path with real-time connectivity between similar type terminals. Examples of continuous traffic are voice, video, and facsimile transmission. Bursty traffic is composed of short, fixed-length messages or transactions that are near real-time and may operate between dissimilar terminals, such as automatic teller transactions. Interruptible traffic is the "batch" type data transmission that can tolerate long delays, is generally lengthy, and need not be transmitted as a continuous stream, such as message transfers.

Classifying communications traffic helps in matching traffic with the different network techniques. Interruptible traffic does not match the capabilities of either packet or circuit switching. In a circuit switched network it is difficult to capitalize on its interruptibility. In a packet switched network,

the high overhead associated with the long message lengths tend to waste capacity. For most traffic, however, integrating circuit and packet switching characteristics into a single switch will handle the traffic the most efficient way. Transmission efficiency could be maximized by pooling total capacity and making it available to whichever service had the highest current demand.

4.4.2 Hardware Solutions

There are a number of different ways circuit and packet switching can be combined within a single network. Integration through shared transmission capacity will be the first approach reviewed.

Common trunking is where both circuit and packet switches have equal access to common transmission facilities via a multiplexor; see figure 4.1. {34}

Integration at this level does not provide much improvement in transmission efficiency, transmission utilization or the exchange of traffic between different user communities or terminal types. Another, more useful form of integration might be an imbedded network as shown in figure 4.2. {34}

In an imbedded network the packet switched network is imbedded in a circuit switched network with each switch having its own community of users. When the traffic between any pair of packet switches increases, the packet switch could request that the circuit switch provide additional connectivity or capacity by the addition of a physical circuit through the circuit switched network. The packet switches could complete the delivery of traffic to terminals connected to the circuit switches by establishing temporary connections to such users through the circuit switches.

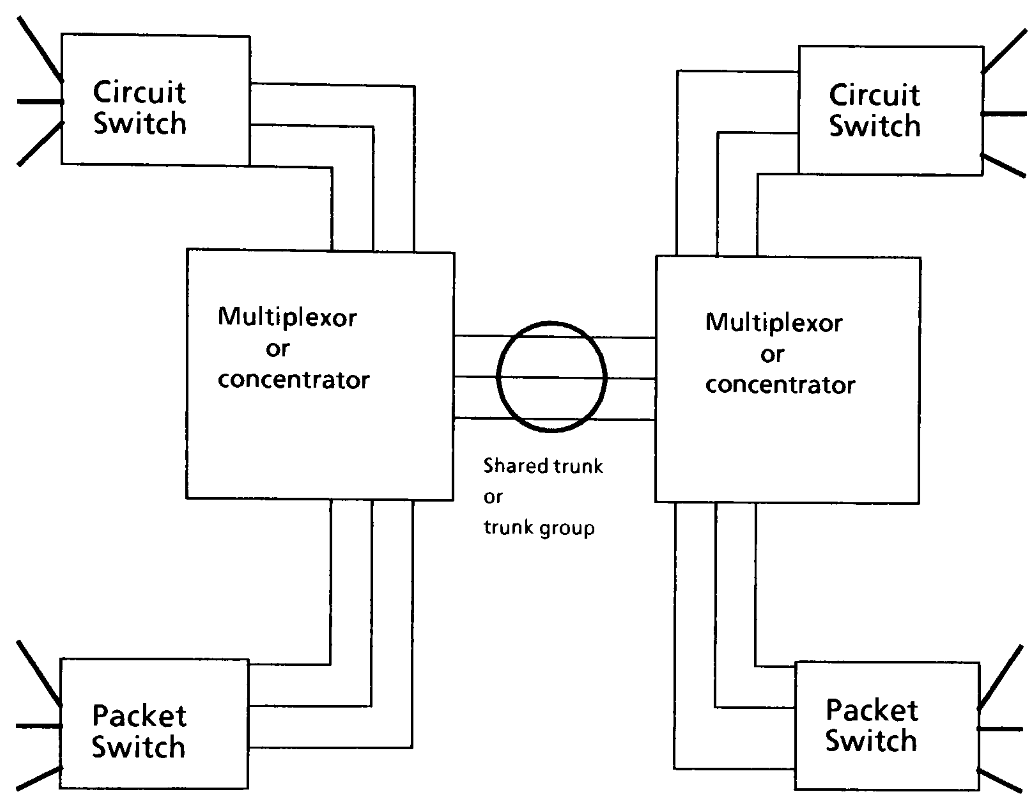
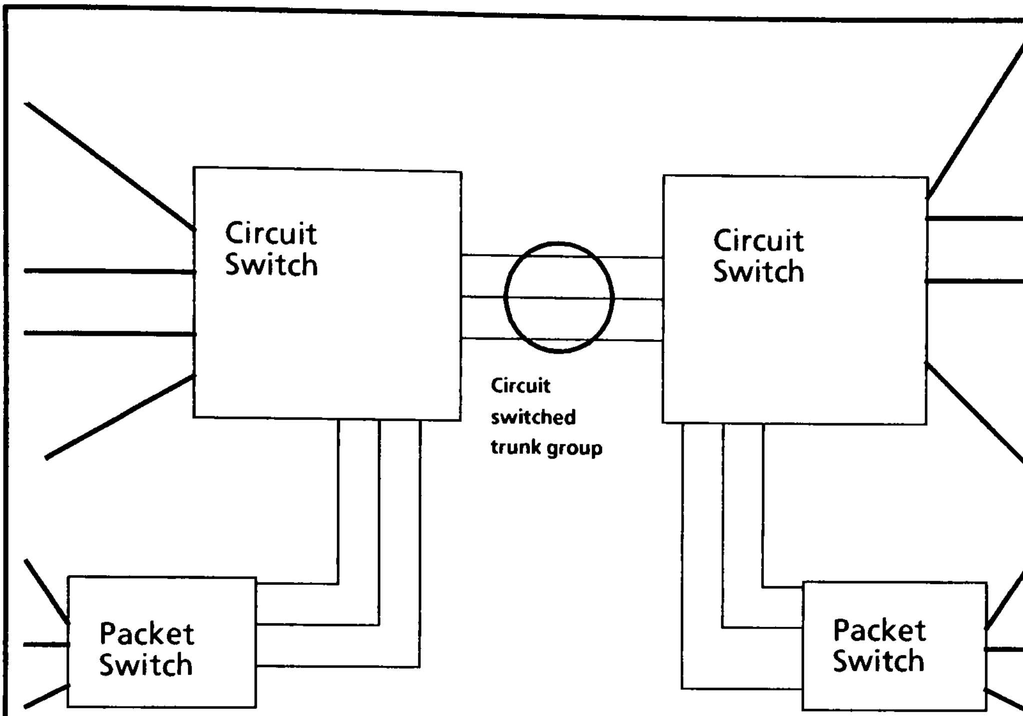


Figure 4.1 . Network integration achieved through common trunking.

Another approach is to combine two switching matrices under the control of a single processor. One matrix would be a circuit switching matrix, the other would be a system of buffers used to hold the information that is processed as packet switched data. The single processor would decide which matrix would



Network integration achieved by imbedding packet switches within a circuit switched network.

Figure 4.2 Network Integration Through Circuit Switching

meet the service demand and route the traffic accordingly. If the traffic exhibited characteristics best serviced by circuit switching it would be routed through the circuit switch matrix. If, however, it could best be serviced by packet switching it would be routed through the packet matrix. This approach is dependent upon a signalling technique between the user and the network that could properly make the matrix choice. Figure 4.3 illustrates integrated switching by combined packet and circuit matrices.{34}

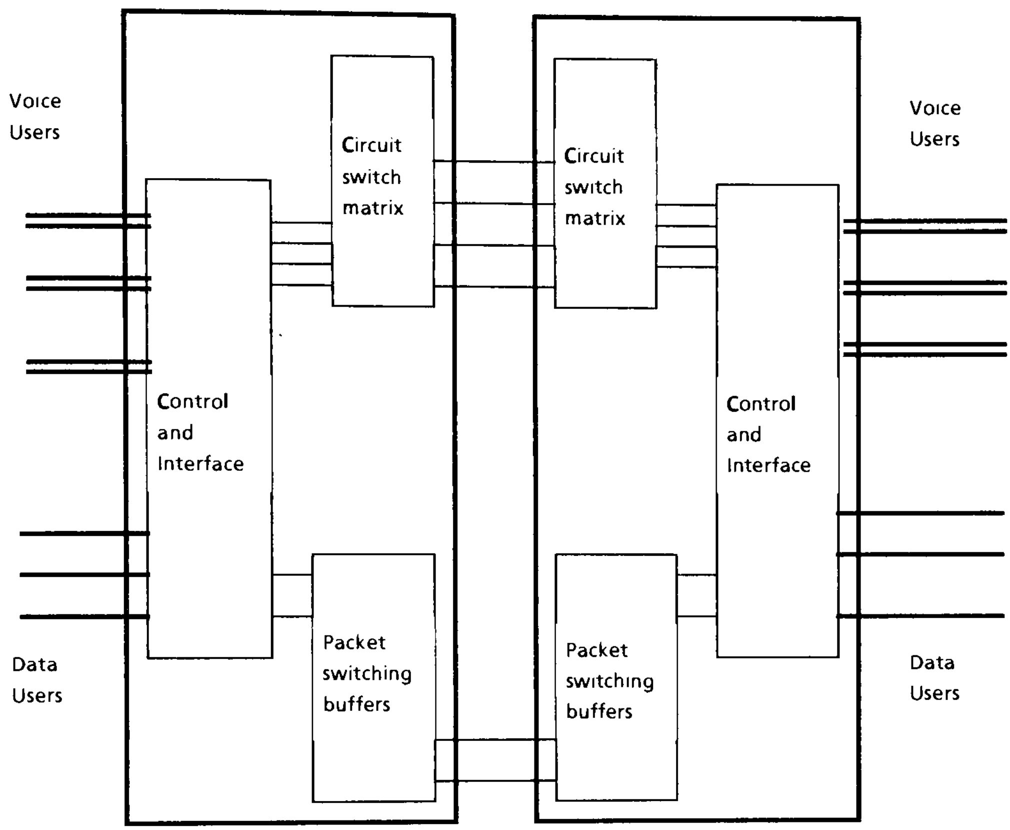


Figure 4.3 Integrated Circuit /Packet Switch

4.4.3 Software Solutions

The proceeding integration techniques involved hardware solutions. Here are some software integration techniques. These will fall into three categories. The first uses a "basically circuit switched" technique but adds some aspects of the multiplexing achieved by packet switching. Time Assignment Speech

Interpolation (TASI) or Time Assignment Data Interpolation (TADI) are two examples of this approach.

Another approach uses a "basically packet switched" technique which limits some of the dynamic routing features and uses some additional memory for reference tables. The result is a packetized virtual circuit.

The third technique uses a dynamically processed master frame with advanced interswitch signalling. Unlike the other two methods which in effect emulate one technique in a network based on the other, master framing is the only technique that actually achieves true integration of packet switching and circuit switching. The master frame approach is based on the time division multiplex structure discussed in the previous chapter. This acts as a fixed channel for circuit switched traffic but uses any excess capacity to send packets to facilitate bursty and interruptible traffic.

Critical to the dynamic master frame multiplexing function is high speed processing with the network switches and high capacity trunks. The initial structure of the frame approximates that of a standard digital time division multiplexor with user allocated time slots within the individual frames. The key difference between dynamic TDM and fixed TDM is that unused capacity in the form of empty time slots exists in fixed TDM. In dynamic TDM, however, a large buffer in each switch is used to assemble frames so that vacant time slots can be recognized and filled with packet traffic. Figure 4.4 is an example of the master frame technique in a switch network concentrating on the trunk between two network switches. We will assume the trunk has a capacity of 1.54 megabits/sec (corresponding to the T1 digital multiplex rate from last

chapter) and a master frame time period of 10 milliseconds. Thus each frame would consist of 15,440 bits during the 10-millisecond time period.

In the master frame approach the interval between two frames is always fixed. Thus each frame will contain the same number of bits. Had our trunk only carried 64,000 bits per second each frame in a 10 millisecond interval would contain 640 bits.

In this technique the portion of each frame assigned to circuit switched channels need not be the same in each frame. This is a key concept. In the following example there are several subframes within the master frame. If 24 channels were active at the same time, each carrying 64,000 bps digital voice signals, there would be a total of 15,360 bits of circuit switched information leaving only 80 bits of capacity in the master frame which is needed for timing and overhead. But if some of these channels were inactive or operating at bit rates of less than 64,000 bps the frame would be under utilized. This frame is built in the buffers associated with each switch during the time that the preceding frame is being transmitted. As the frame is assembled, the subframes associated with circuit switched connections are placed in the front. Any excess capacity then becomes readily apparent which is from the end of the circuit switched subframe to the end of the master frame.

So, during the 10 millisecond interval when the frame is being assembled one or more packets may arrive at the switch for transmission further through the network. Assuming excess capacity in the master frame, these packets would be transmitted at the end of the frame. If, after these packets had been

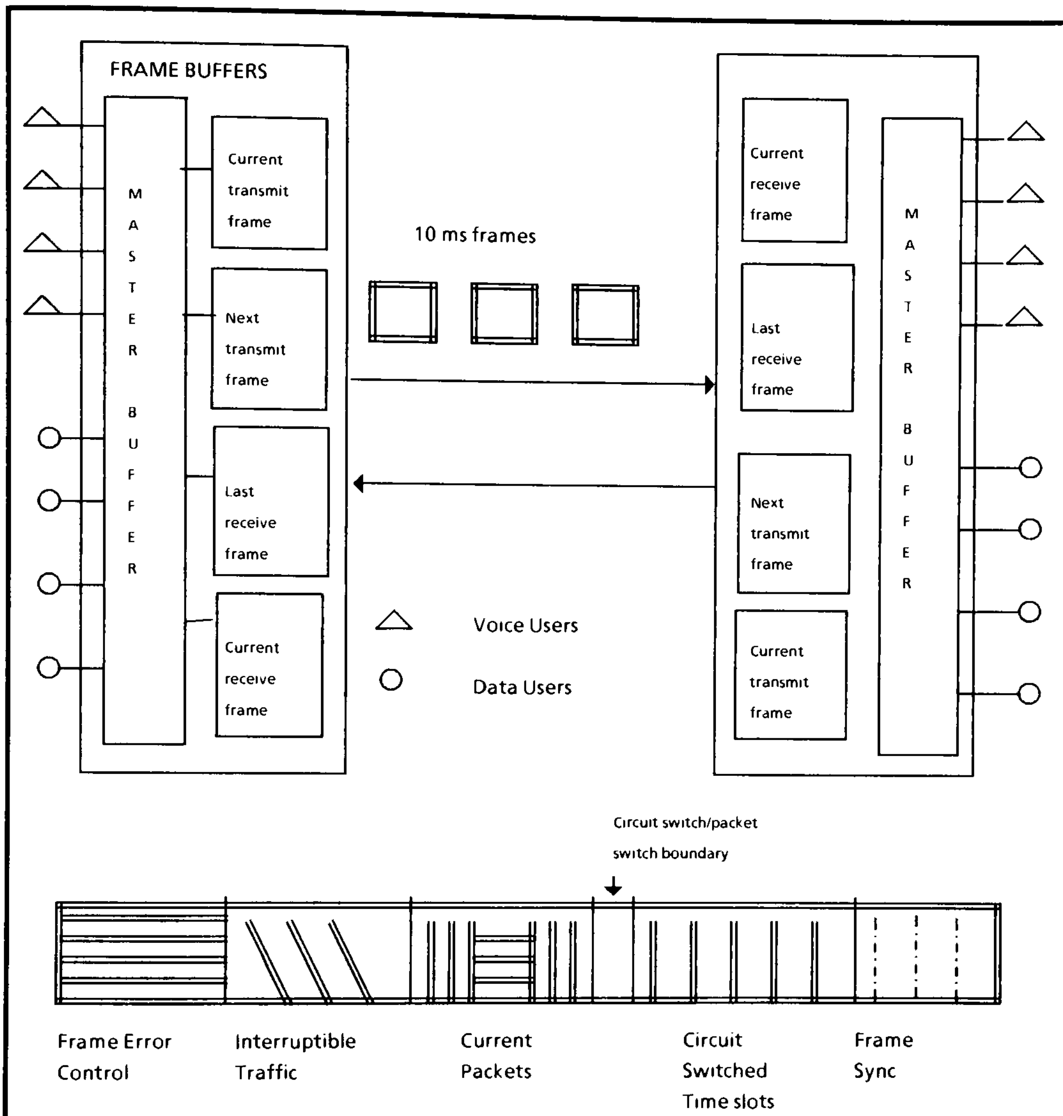


Figure 4.4 Master Frame Technique

transmitted, there existed in the frame still more excess capacity, then some long, interruptible batch-type traffic could be used to fill the frames.

Figure 4.5 shows a master frame link supporting twelve 64 kbps voice channels, twelve 9600 bps data channels and twenty four 2400 bps channels for a total of 9408 bits of circuit switched traffic per frame. The remaining 6000 bits per frame are available to carry bursty and interruptible packetized traffic, to the total frame size of 15,440 bits.

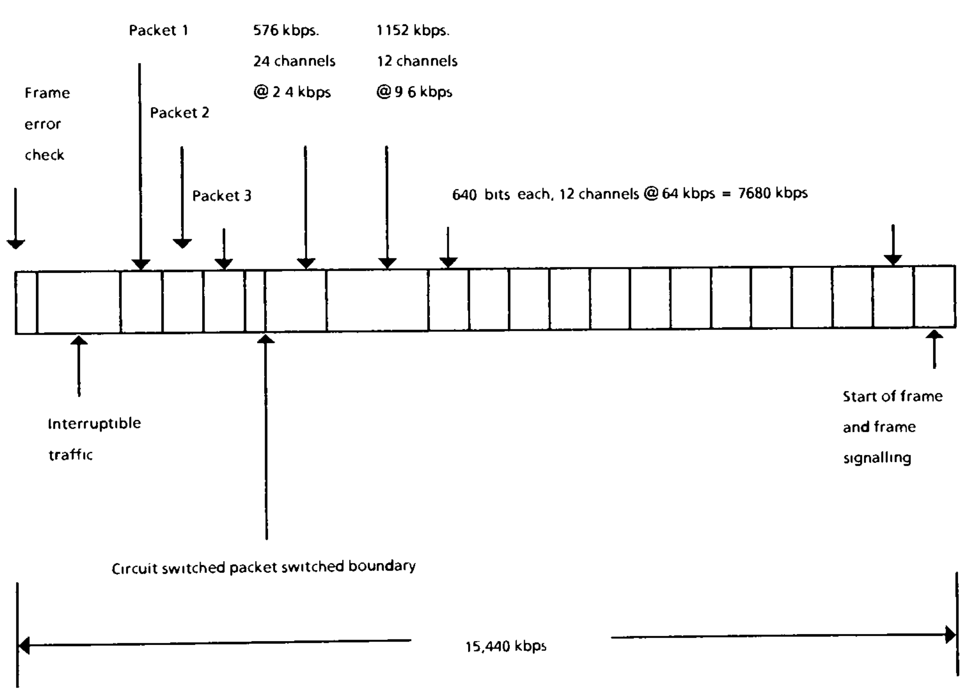


Figure 4.5 Master Frame Link

4.4.4 Example of Integrated Switches

The following are some ways of implementing a master frame approach to integrating circuit and packet switching. The first called SENET (for Slotted ENvelope NETwork) has been studied by analysis and simulation. The results show that link efficiencies of close to 100 % can be achieved for bursty, interruptible and continuous traffic {34}. By reserving some trunk capacity in each frame for high-priorty bursty traffic acceptable delays for real-time traffic can be achieved as well as maintaining efficient use of the trunk capacity. While such a switch has not yet been implemented a possible configuration may be seen in Figure 4.6. {34}

An example of an integrated switch which has been implemented in a commercial product line is a technique known as PACUIT (PACket and cirCUIT) switching. Typical configurations of these switches involve 9600 bps trunks with each frame lasting 0.1 second. The frame consists of three parts. One part is dedicated to circuit switched traffic between pairs of network terminals. Any available capacity not assigned to this part is used to handle bursty traffic.

This traffic is not contained in packets but in PACUITS. PACUITS are groups of bits or characters going between different user endpoints located at the same switches. Bits or characters transiting the network between the same pair of switches are grouped into a single large packet before the transaction is completed. This single packet contains data from many different users. It is assembled at the originating switch and disassembled at the terminating switch. Thus, between those two switches they are actually circuit switched.

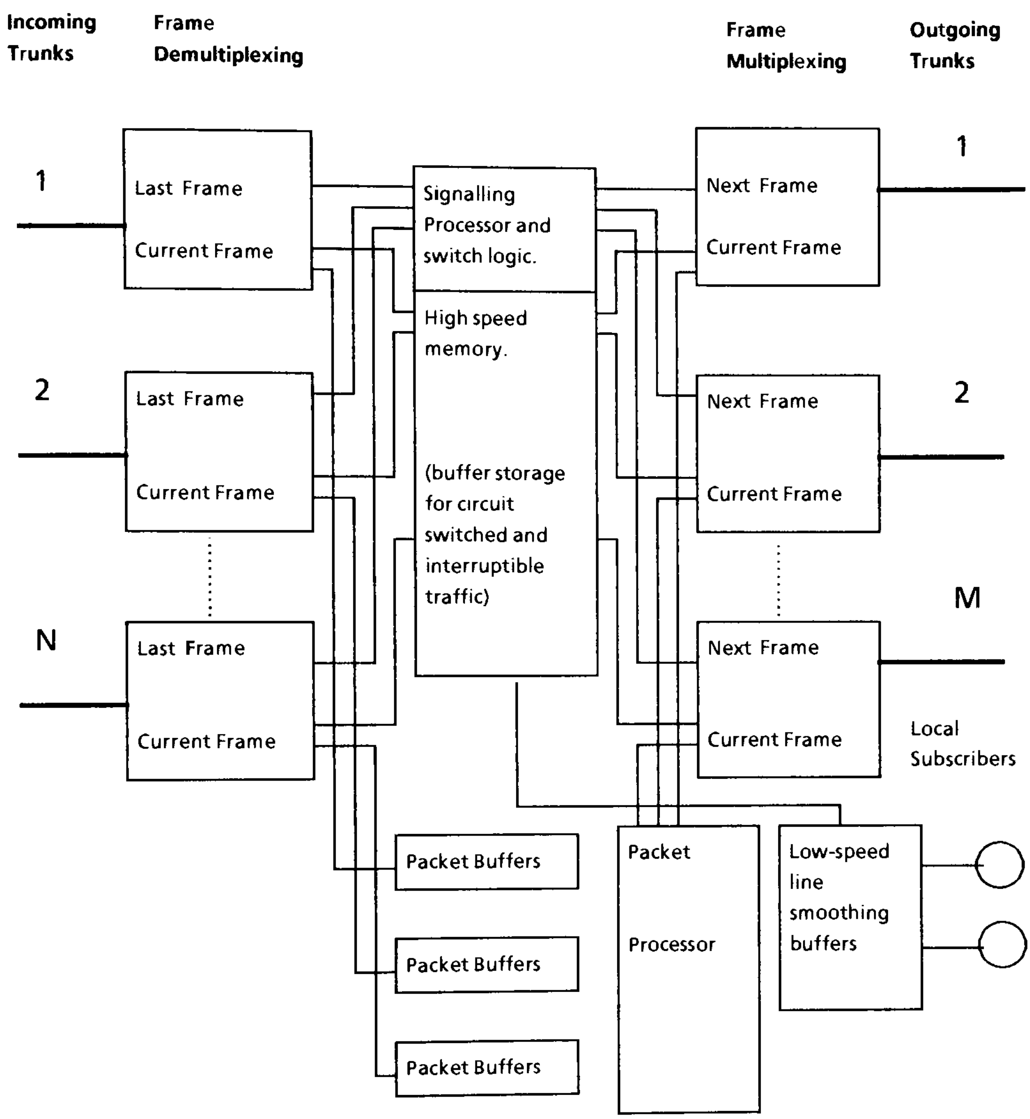


Figure 4.6 SENET Switching

As each succeeding PACUIT arrives at a switch it is assembled in a buffer during the transmission interval for the previous frame. At the beginning of

the next frame the PACUIT is then synchronously transmitted through any necessary intermediate switches to the destination switch where it is disassembled and the characters and bits within are delivered to their proper destination line. Overhead entails associating the bits in the frame to the proper destination line. Any capacity remaining in the line after the circuit and PACUIT traffic can be used for dedicated packets. Figure 4.7 is an example of PACUIT network. {34}

The integrated hybrid switch is a practical, viable solution to concurrent voice and packet traffic in ISDN. Hybrid switching offers many advantages. Bulk data, facsimile, and video signals can be transmitted with greater transmission efficiency and less delay in the circuit switching portion of a hybrid switch than a packet switch. {10} After call setup is completed, voice communication suffers less end to end delay in a hybrid switch than in a packet switch. Also, the ease of transition from today's circuit switch to the hybrid switch makes it even more attractive.

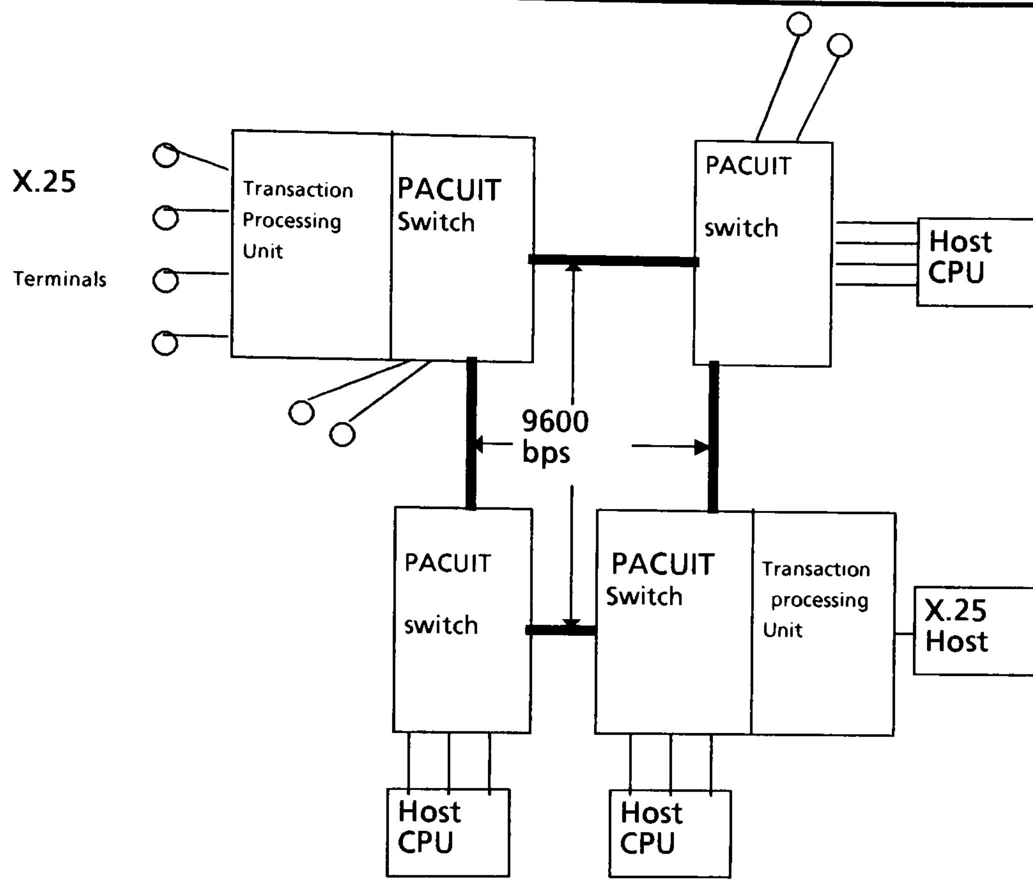


Figure 4.7 PACUIT Switching

This chapter demonstrates how the devices and services of chapter 1 will access the ISDN of chapter 3. We will examine the physical, link, and network level interface.

5.1 Introduction

Thus far this dissertation has examined the fundamental struture of ISDN and the nature of the media which will cross the network. All that remains is the user access to ISDN. The "user" now has an ever more sophisticated and diverse group of terminals and equipment and the question of designing a standard interface has received a lot of attention. This interface must be fair, reliable, secure, efficient and easily adopted. To get a better perspective of its importance here is a quote from Draft Recommendation I.411 of CCITT,

"The ISDN is completely described by the attributes that can be observed at an ISDN user/network interface, including physical, electromagnetic, protocol, service, capability, maintenance, operation and performance characteristics. The key to defining, and even recognizing an ISDN is the specification of these characteristics."

Some of the possible ISDN user interface configurations include:

1) Access by a single ISDN terminal

2) Access by a multi-terminal installation

3) Access by multi-service PBXs, LANs, or private networks

4) Access by a specialized storage and information processing center

5) Access by dedicated service networks, other multipleservice networks, or possibly another ISDN. {5}

Design interface objectives include:

1) Disparate terminals and applications to use the same interface

2) Portability of terminals

3) Allow for separate evolution of terminal and network equipment, technologies and configuration.

4) Efficient connection with specialized storage and processing centers .{5}

Some additional features of ISDN may be:

1) multidrop terminal arrangements

2) user choice of bit rate, switching mode and coding method

3) terminal to terminal compatibility check. {5}

Figure 5.1 is the basic ISDN Interface Reference Model

NT1 is Network Termination 1. It includes functions equivalent to Layer 1 of the OSI Reference Model which is known as the physical layer. These functions deal with the physical and electromagnetic termination of the network. They include terminating the transmission line, power transfer, converting signals, conditioning signals to the appropriate physical and electrical interface, Layer 1 multiplexing, specifications, and providing the appropriate maintenance and testing as required. {15}

NT2 is Network Termination 2. It includes functions roughly equivalent to layers 1, 2, and 3 of the OSI model (physical, data link, and network). PABXs, LANs, and terminal controllers are examples of equipment which provide NT2 support. NT2 functions include:

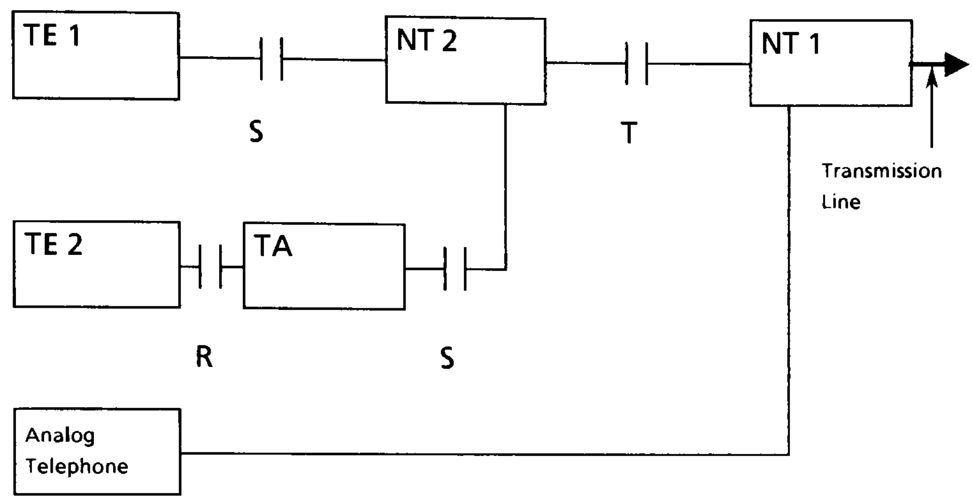


Figure 5.1 ISDN REFERENCE MODEL

1) Layers 1, 2, and 3 protocol handling

2) Layers 2 and 3 multiplexing

3) Switching

4) Concentration

5) Maintenance functions

6) Interface termination

A PABX could provide support for all three layers where as a terminal (cluster) controller could only provide layers 1 and 2. A time division multiplexor could provide only layer 1. In Europe the NT1 and NT2 may be combined into a single piece of equipment called the NT12.

TE or Terminal Equipment includes devices such as digital telephones, data terminal equipment and integrated work stations. TE is generic for a user terminal device.

TE1, Terminal Equipment Type 1, functions correspond to individual converting information to a form appropriate to be transmitted through the ISDN and will include a variety of applications such as facsimile, voice, data and video. {15} Thus a TE1 is any device capable of interfacing directly to ISDN.

TE2, Terminal Equipment Type 2, has interfaces other than the ISDN user/network interface. Thus these devices will require some form of adapter to interface with ISDN.

TA, Terminal Adapter, provides the requisite conversions to allow TE2 equipment to communicate with ISDN.

Note in the preceeding figure the points R,S, and T. These are references indicating the type of user/network interface available at that location. Reference point T segregates functions between NT1 and NT2 with only NT1 type devices connecting to the network. Reference point S defines the line of demarcation between terminal functions to network functions to the right. Points S and T are the same in terms of ISDN user/network interface procedural, electrical, and mechanical specifications. They both define appropriate locations for standardized user/network interfaces. {15} Point R is any user/network interface type supported by a TA and specified in standards other than the ISDN recommendation.

At reference points T and S the channel makeup consists of two 64 Kbit/s transmission channels plus a 16 Kbits/s signalling channel. The 64 Kbits/s channels provide transparent transmission paths that are not restricted by coding or bit density. These are labeled `B`channels. The `D` channel or signalling channel transmits signalling information along with packet and telemetry data. The level 2 protocol for the `D` channel is Link Access Procedure D which is very similar to the Recommendation X.25 data link protocol LAP-B.

The B and D channels form the fundamental interface to ISDN. They will normally be implemented as two B channels and one D channel, providing two 64 Kbits/s voice or data channels together with one 16 Kbits/s signalling channel. These are supported by a TDM assembly which allows for any subset of the basic B + B + D format (ie. B + D or D). Thus the loop transmission can operate at 144, 80, or 16 Kbits/s and all utilize the same interface running at 144 Kbits/s.

Another possible channel type under consideration is the C channel. This is similar to the D channel but uses an analog voice channel at 8 or 16 Kbits/s. The reason for this is to allow suppliers to make a simple change to existing switching equipment.

A third possibility, sometimes referred to as primary rate access, consists of some number (n) of B channels and a 64 Kbits/s D channel. Such a wideband configuration like 23 B + D (1.544 Mbits/s) or 30 B + D (2.048 Mbits/s) would

be required to support a large PABX or videoconferencing. Table 5.1 summarizes the 3 access types:

Basic Access
* B + B + D , Interface Net Rate: 144 Kbit/s
B + D ,Loop Net Rate 80 Kbit/s
D , Loop Net Rate 16Kbit/s
B = 64 Kbit/s; D = 16 Kbit/s

Combined Access
* A + C
A = Analog; C = 8 or 16 Kbit/s

Primary Rate **Access**
* nB + D, Interface Rates: 1544 or 2048 Kbit/s
B = 64 Kbit/s; D = 64 Kbit/s

Table **5.1 ISDN User Acess** Types

The basic physical interface that has been recommended is the familiar telephone jack (RJ plug and socket). This will use a four wire scheme and includes a transformer coupling between network and user. There is also a nominal power feed from the network to the user provided by an optional four wires. {5}

5.2 ISDN Layers.

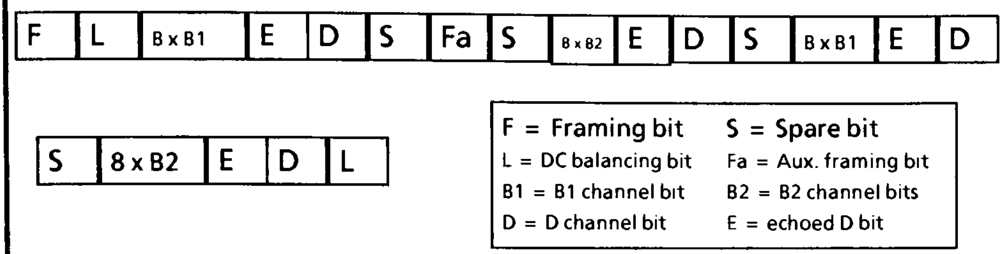
This section examines the physical, data-link and network layers of ISDN. The higher OSI layers, transport, session, presentation and application, are not

addressed. ISDN refers only to the three lower layers for basic switching and transmission services. Higher level services like end to end flow control, media presentation, and maintenance are done as an open end system although these services may actually co-exist within the network.

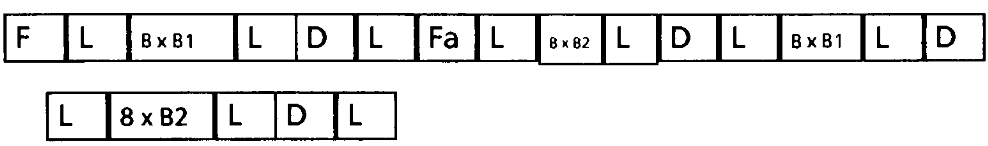
5.2.1 The Physical Layer.

This layer assumes responsibility for activating, deactivating, and maintaining the physical circuit. It may also provide physical layer multiplexing, demultiplexing and in the case of multipoint configurations, collision detection. Control information is included along with the upper level bit stream. The ISDN physical level data is transmitted at 192 k bits/s. This means there are 48kbps overhead (192 (64 + 64 + 16) = 48) which represents the physical protocol central information. The transmission is always full duplex. When a physical entity (NT2, TA or TE1) leaves the idle state and synchronization is achieved with a peer physical level entity then a connection is activated. The proposed physical frames are shown in Figure 5.2.

There is a separate format for transmission from NT to TE (called an NT frame) and for transmission from TE to NT (called a TE frame). Both frames start with a framing bit for recognition. Additional synchronization is provided by an auxillary framing bit. These framing bits allow the physical level to distinguish which bits are intended for a particular data link entity. In this manner the physical entity separates the frame into individual bit streams, one for the B1 data link entity, one for the B2 data link entity, and one for the D data link entity. {5} These physical layer frames are 48 bits long. They are



a) NT Frame (NT to TE - 48 bits in 250 microsec = 192 kbits.



a) TE Frame. TE to NT - Same length but delayed by 2 bit times.

Figure 5.2 ISDN PHYSICAL LAYER FRAME

really not frames in the OSI sense as the physical layer uses them only as a term of reference.

The TEs derive their synchronization (clocking) from the NT frames sent by the NT1 or NT2. The TE frames are transmitted from the TEs to the NT1 or 2 and are delayed two bit times from the start of the NT frame. Therefore the first E, or echo bit, seen in the NT frame is actually the echo of the last D bit of the previous TE frame. This echoing is done for contention resolution on the D channel. When a TE observes some predefined number (X) of E bits with a value of one it then begins to transmit. As the TE transmits it checks the E bit to ensure it has the same value as the last D bit transmitted. But if this value is

not the same the TE stops transmitting and returns to checking E-bits for the (X) all-ones condition. When a TE completes a successful D channel transmission it increments (X) by one and continues checking the E-bits for all ones, so that it may send again. This provides a method of equally sharing the D-channel. This is due to the HDLC rule that only 7 consecutive "1" bits can occur before the channel is free. If a terminal does not wish to send it puts a "1" bit in the slot. By waiting for a certain number of "1" bits a priority can be established. Note that the S, or spare, bits in the NT frame correspond to the L bits in the TE frame. The L bit is used for DC balancing and is used for the complete frame in the NT frame and for groups of bits (B1,B2,D) in the TE frame.

It should be noted that it will be quite some time before DTEs supporting current physical level interfaces like RS232, RS449, and X.21 are converted to ISDN interfaces. Thus, in the short run at least, most devices will require some sort of terminal adapter.

5.2.2 DATA LINK LAYER

The data link layer actually consists of three separate protocols to govern voice, data and signalling. ISDN really provides 64 kbps channels (and one 16 kbps channel). These 64 kbps (and 16 bps) bit streams may be used for whatever service the user desires. Thus the data link protocol could be a pulse code modulation protocol as a voice driver or a high level data link protocol like HDLC as a driver for data. The D channel data link protocol is an HDLC-like protocol called LAP-D (Link Access Protocol-D-channel). It is very similar to X.25's data link protocol LAP-B (Link Access Protocol-Balanced). LAP-D uses

frame structures comprised of frame delimiters (flags), zero insertion/deletion, and cyclic redundancy checking. This provides secure alignment, transparency, and error detection. The error recovery and flow control procedures should be the same as LAP-B. This is the preferred solution over that of the level 2 procedures for SS No. 7 which we mentioned earlier.

The address byte of LAP-D is used to support multiple logical links. This will provide independent multiple LAP'S between exchange termination and logical end points at user premises. {15} Logical links may be used to either access different level 3 (network layer) protocols for S-type, P-type, and T-type information or to address different terminal end points. This is done by the use of the Data Link Connection Identifier (DLCI). The DLCI is made up of a Service Access Point Identifier (SAPI) and Terminal Equipment Identifier (TEI). The SAPI indicates the Service (S,P,orT) and the TEI indicates the logical terminal. There can be up to 128 P-type network entities and 128 T-type Network entities. There is one S-type Network entity instance per terminal device but there may be multiple P or T type entities. {5} The support of different level three procedures for S-type and P-type is justified by taking into account that signalling does not require certain features needed by data (such as multiple virtual circuits, packet sequencing, and flow control). Support of multipoint capability can be justified by looking at the configuration in the figure 5.3. {15}

This configuration was considered in early ISDN protocol discussions and is characterized by the allocation of simple level 2 functions at NT2. By using the address field to differentiate terminal end points, the NT2 functions can be limited to queuing for D-channel access at T and distribution towards end

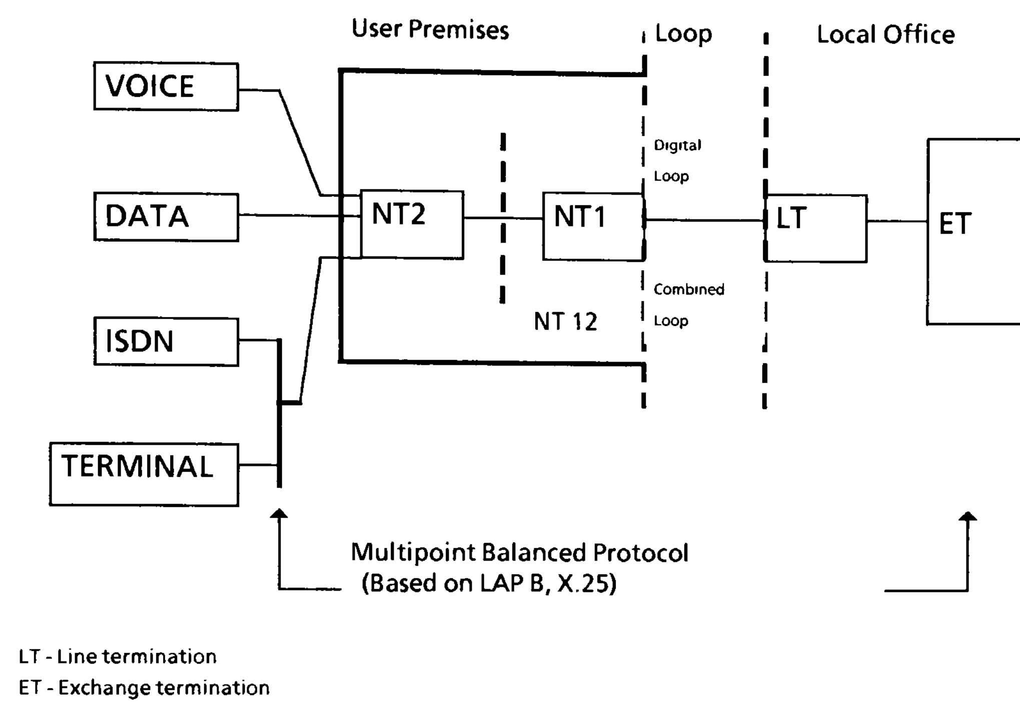


Figure 5.3. **Star distribution at user premises with simple NT2 functions**

points, without handling error recovery or flow control procedures. {15} Thus, the same level 2 protocol appears at S and T interfaces (transparent NT2). When the physical implementation NT12 (NT1 + NT2) is permitted this configuration becomes particularly interesting. Multipoint capability is also required in the case of multidrop distribution with little or no NT2 funtions. In the case of LANS or PABXS or complex NT2 functions, endpoint addresses may not be required, thus leaving the use of logical links simply to discriminate information types.

LAP-D should also properly handle the framed message traffic composed by various information systems. Signalling and data delay requirements over the D channel should dictate adoption of adequate delay control mechanisms such as limitation of the p-type frame size or adoption of a priority scheme for s-type information.

5.3 3 Level 3 - Network Layer.

The network layer in a basic configuration (B + B + D) consists of at least three separate entities. These include B1 and B2 entities and at least one s-type (signalling) entity. While p-type (packet) and t-type (telemetry) capabilities must be provided they need not be separate network entities in operation. {5} Above the network layer there is no difference between ISDN and any other open system as illustrated by the OSI model.

In the case of signalling a datagram oriented protocol like the one we discussed for SS No. 7 would be appropriate. The s-type information will require capabilities to support common telephony features such as conferencing, call forwarding, call hold, and call transfer. In addition, the s-type procedures should be defined to both the circuit switched data (according to X.21) and telephony terminal signalling, and an easy interworking with the future ISDN user part of SS No. 7.

For example, if a user chose to select a B-channel to access a packet switching service (at 64 Kbits/s), that B-network entity would be for the X.25 packet level protocol. If that same user wanted to access the packet switching network but

at a lower data rate the user could select the p-type entity in the D-channel. This would include the in-band call setup and clearing procedures of X.25.

In addition there may also be a need for other classes of p-information. This aspect is critical since it would involve the definition of new procedures for packet switched data. {15} At least two possibilities are being considered. The first deals with the handling of frame mode data. In this case, s-type information would be used to set up logical link connections on the D-channel. Data would then be transmitted over these links with no level 3 sequencing or flow control mechanisms. This is intended to provide simple packet switched communications. The second consideration supports the multiplexing of data at level 3. In this case data would be transferred over virtual channels in a format compatible with level 3 of X.25. However, the procedures for virtual channel set up and clearing would be different since the s-type information would be used to set up and clear these virtual channels. Both of the alternatives mentioned above make it possible to create calls consisting of both circuit-switched and packet-switched information using a single, integrated signalling mechanism (multimedia calls). {15}

5.4 Status of ISDN Standards

The following is a table of the CCITT Recommendations Governing the User-Network Interface to ISDN {48}.

I.410 General Aspects and Principles relating to recommendations on ISDN user network interfaces.

I.411 ISDN user network interfaces - reference configurations.

I.412 ISDN user network interfaces channel structures and access capabilities.

I.420 Basic user-network interface.

I.421 Primary rate user network interface.

I.430 Basic user-network interface - Layer 1 specification.

I.43x Higher rate user-network interfaces. (For further study see question L1XV11).

I.440(Q.920) ISDN user-network interface data line layer general aspects.

I.441(Q.921) ISDN user-network interface data link layer specifications.

I.450(Q.930) ISDN user-network interface layer 3 general aspects.

I.451(Q.931) ISDN user-network interface layer 3 specifications.

Draft recommendations Q.920, Q.921, Q.930 and Q.931 were approved by CCITT Study Group XI at their final meeting in May 1984. The level of maturity of these recommendations is still quite low and, although several companies have developed their own implementation (Northern Telecom, Motorola, Seimens) of the user/ISDN user interfaces, further evolution is expected.

In the case of Recommendation Q.921 (LAP-D), evolution is expected to occur as problems are resolved during design and implementation. Because it is so similar to LAB-B used in X.25 packet networks few problems are anticipated. Recommendation Q.931, which covers ISDN call control status, is not likely to be as stable. At least three technical issues remain outstanding and could precipitate some future changes in Q.931. These are: symmetry of signalling procedures; alignment with the ISDN User Part of CCITT No. 7 (the intranetwork common channel signalling standard favored for ISDN application see in particular Draft Recommendations Q.761-764 and Q.766)

application see in particular Draft Recommendations Q.761-764 and Q.766) and resolution of the more general problem of signalling protocol layering and partitioning.{48}

Much of the strength of these user/ISDN interfaces lies in the signalling techniques. Some particular advantages of the ISDN signalling protocols include:

1) Separation of the signalling (D) channel from the voice/data (B) channels.

2) The power and flexibility associated with the wide array of signalling messages and associated information elements.

3) The very fast response time and low signalling delays attainable with a 16 Kbits/s (Basic Access) or 64 Kbits/s (Primary Rate Access) signalling channel.

The following discussion examines the two types of user access: Basic Access (2B + D) and Primary Rate Access (23B or 30B + D) as specified in CCITT 1.420 and 1.421 respectively.

5.4.1 Basic Access Interface Standard

Basic Access is primarily intended for terminal access and is capable of multiple terminals in a star or bus configuration on the same access system. In a star configuration each terminal would have direct access to the D channel via a concentrator. When operating in a bus configuration, multiple terminals compete for a shared D channel to send signalling or data messages to the

network. To control access to the D channel between competing terminals, a contention resolution mechanism, based on collision detection and back off, is provided in layer 1.

In the case of 'dumb' terminals accessing the network a layer 3 stimulus signalling protocol has been designed.{48} It allows communication between the human user of the dumb terminal and the ISDN exchange, with the dumb terminal acting like an intermediary. These dumb terminals could be used to offer inexpensive access to the advanced features of ISDN. Much of the functionality needed to support such terminals would be provided by the ISDN exchange to which the terminal is attached. This may be a cost effective means of providing basic telephone services within ISDN.

Intelligent terminals like microcomputers and small PBX systems would use a form of signalling called functional signalling. Functional signalling requires that both ends (terminal and ISDN) keep track of call state information. These terminals may support powerful integrated voice/data devices in an interactive combination of local and remote intelligence. Such services will undoubtably include the complex combinations of voice, text, data and graphics discussed in chapter 2.

5.4.2 Primary Rate Access Interface Standard

Primary Rate Access is intended for large PABX or LAN applications which require a very high bandwidth interface. As office automation marches on the importance of this access method increases greatly. Potential applications can be grouped into two types:

1. Interconnection of switches

2. Computer interfaces to circuit switches

These applications are illustrated in figure 5.4. Both involve the interconnection of networks (networks of computers or switches). Because of this the signalling procedures need to be symmetric. {48}

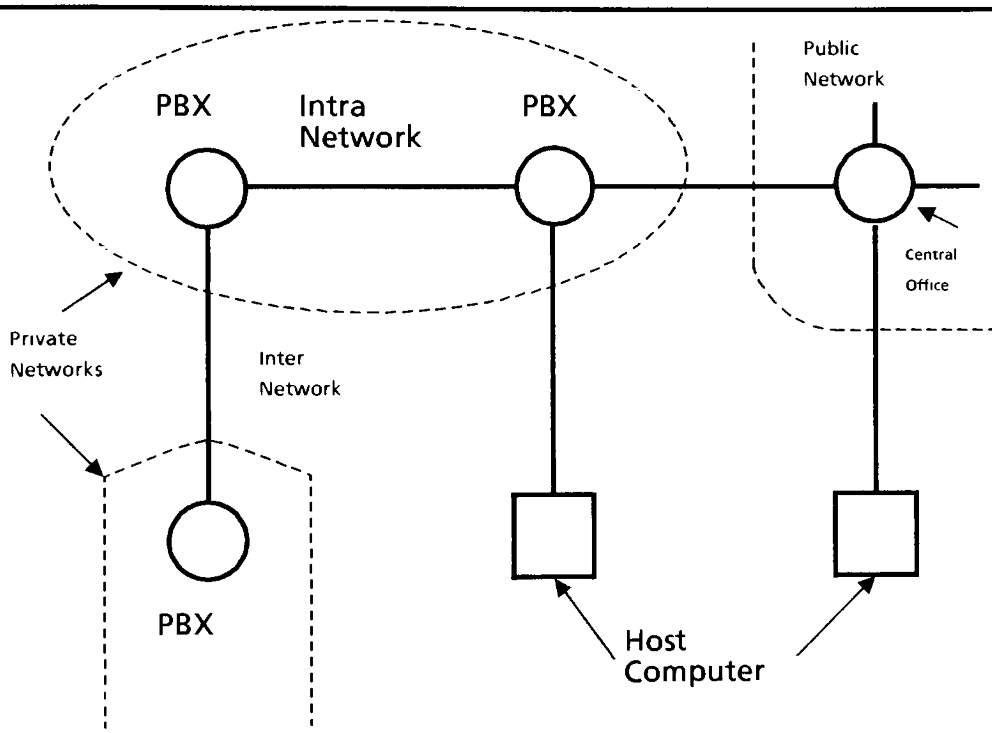


Figure 5.4. Potential Applications of ISDN Primary Rate Access Protocols.

Interconnection of switches - The main application for primary rate access, considered within the CCITT, is the PBX to ISDN interface. This will involve two PBXs speaking to each other across ISDN. More work is required to define the complex signalling procedures required for this interconnection.

Computer interfaces to circuit switches This involves computer systems interfacing either to ISDN or to a PBX and is considered key because it facilitates open systems interconnection (OSI) of computer systems to public and private circuit switched networks. Such interfaces are important in the utilization of PBX technology for local area networks (LANS).

Such computer interfaces are grouped into two categories:

1. To enable the use of a circuit switched network as part of a manufacturer's open system's architecture.
2. To enable a computer to provide services to the users of a circuit switched network.

If a computer is to provide an information service such as voice/ text messaging to the network users, then it may usefully exploit switch features which are accessible via the interface. As an example a messaging system could transfer a call from a user who has accessed a message to the original message sender or it could store the message and send a message waiting signal to a user's terminal. Again additional signalling will be required.

5.5 AT&T's Digital Multiplexed Interface

Such a PBX-to-computer interface has already been designed and proposed as a standard by AT&T. Called the Digital Multiplexed Interface (DMI), it provides T1 transmission between the two devices and supplies the ISDN required 23 user channels. Control is via ISDN common channel signalling{39}.

Some of DMI's key benefits are that it:

* allows 23 terminals in North America and Japan (30 in Europe) to simultaneously connect to a host computer over twisted pair cabling.
* greatly reduces the cost (due to the multiplexed implementation) of interfacing a PBX to a host.
* provides an interface between computers and PBXs that is consistent with the evolving architecture of the ISDN interfaces being defined by CCITT.
* allows economical, high speed (64 Kbits/s) access to hosts from terminals distributed over a digital PBX network.

Clearly, the important thing to note here is that AT&T developed DMI with ISDN in mind (common channel signalling, 23 user channels, 64 Kbits/s transmission). So when ISDN standards are completely established, both PBX and host computers can gracefully evolve to meet them. The common channel signalling of DMI uses a separate channel which corresponds to ISDN's D channel.

DMI provides an efficient means of interfacing EIA interface standard equipped data terminals (ie. RS-232, RS-449) to a host. Several configurations are possible as illustrated in figure 5.5. {39}

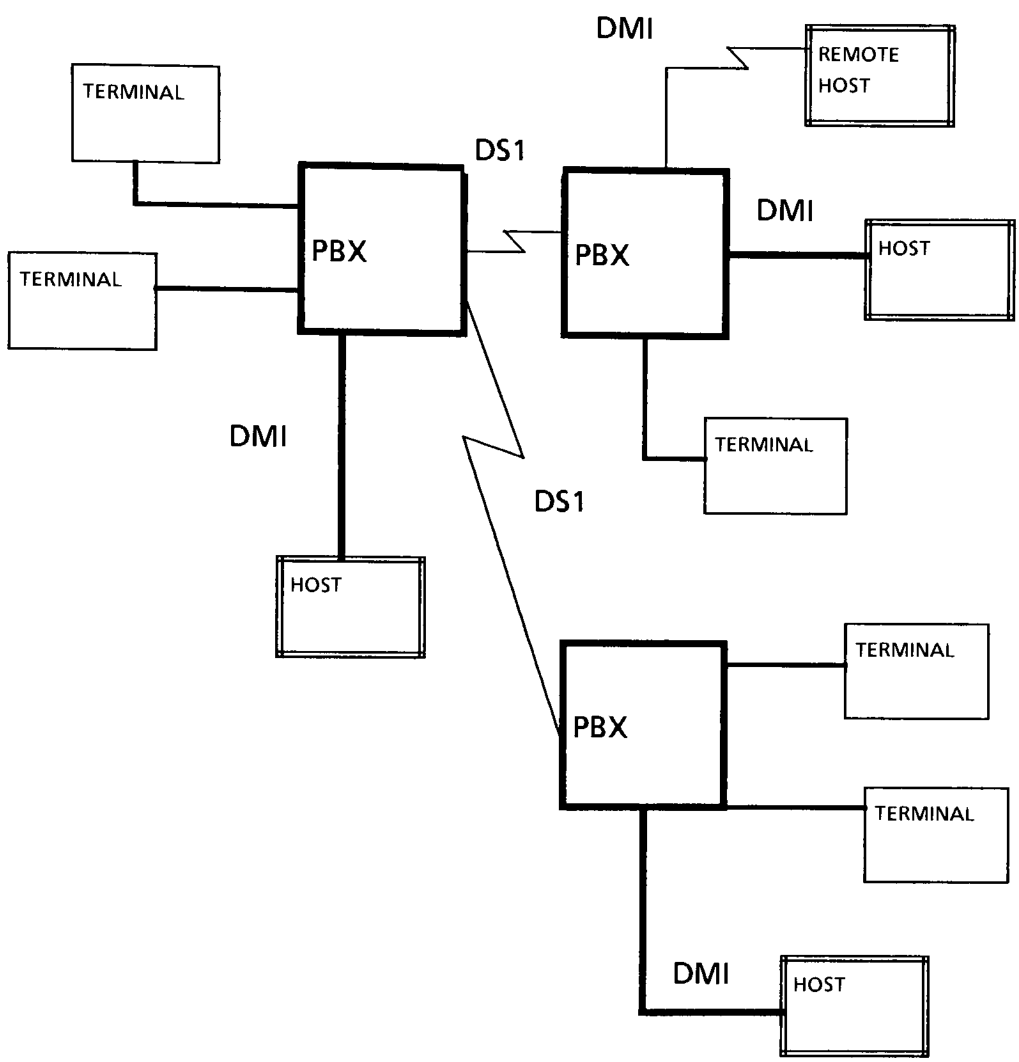


Figure 5.5 Digital Multiplexed Interface Configuration

Terminal -to -host and host -to -host communications are supported. Terminals and hosts can be located locally or remotely. A user may wish to establish communications with a host that is connected to the same PBX as his terminal or, through the use of digital trunks (T1) between PBXs, with a host that terminates on another PBX.

The physical layer is based on the use of the T1 carrier (1.54 Mbits/s). It conforms to national and international standards for transmission at DS1 (the signal designation on T1 carriers) rates. The frame format consists of 24 eight bit words and one frame bit for a total of 193 bits per frame (as was discussed in chapter 2). Channels 1 through 23 carry data and channel 24 is used for signalling.

DMI uses a two step approach to signaling. In step 1 a bit oriented method is used which is for initial implementation and is based on the widely used on-hook / off-hook tie trunk signaling. {39} Step 2 is a message oriented scheme and is a subset of the common channel signaling procedures defined thus far by the CCITT committees on ISDN. This two step approach is necessary because of the time frame involved before ISDN recommendations are finalized.

DMI provides three data formats, called modes 0, 1, and 2, to convey user data. These formats provide for synchronous and asynchronous data transmission. The mode of each data channel is individually negotiated. Mode 0 provides full duplex, synchronous transmission of user data. The data in mode 0 is inverted prior to transmission on the 64 Kbits/s channel. This permits the use of an HDLC based protocol to satisfy the ones-density requirements for

operation on those T1 facilities requiring enough pulses (1's) to maintain timing and synchronization.

Mode 1 provides 56 Kbits/s full duples synchronous transmission of user data. Mode 1 is compatible with 56 Kbits/s Dataphone Digital Service. Framing information divides the data stream into individual bytes for each of the 64 Kbits/s channels. Each byte is then subdivided into two fields. The first field consists of the first seven bits which contain either user data or control codes, depending on the status of the second field which is bit 8.

Mode 2 provides general data transmission at rates up to 19.2 Kbits/s. It supports either synchronous or asynchronous data, full or half duplex, and existing terminal interfaces. It allows existing HDLC framing devices to be used in the hardware design. User information is divided into blocks of variable lengths delimited by a flag. Bit stuffing is used so that the flag is duplicated by the user data.

5.6 Digital Satellite Network

DMI addresses the need to connect computers to a PBX. It must be remembered that the ISDN will, in reality, be comprised of several ISDNs (ie. North America, European, Japanese, etc.). In what manner will they interface? The most probable method suggested is via a digital satellite network. Unfortunately digital satellite networks have evolved independently of terrestrial networks. The following are some of the characteristics of a Digital Satellite Network (DSN). {27}

1) Power and bandwidth are at a premium. Therefore a DSN will employ voice encoding, digital speech interpolation, facsimile coding, bandwidth efficient modulation, and/or forward error correction coding techniques to improve bandwidth utilization.

2) There exists a multiple access capability in the up-link and a broadcast capability in the down-link.

3) Multi-transponder operation is usually employed. A transponder listens to some portion of the spectrum, amplifies the incoming signal, and then rebroadcasts it (at another frequency to avoid interference with the incomming signal).

4) Differing performance requirements for the different services covered in chapter 1, such as voice, data, facsimile, and video. These are generally met by applying appropriate error control techniques to each variety.

5) Due to the motion of the satellite and the resulting Doppler effect, the clock derived from DSN has some inherent frequency uncertainty which must be removed before it can be interfaced with the terrestrial network.

6) In the DSN, typical propagation delay is about 270 ms which is quite long compared to the delay experienced in a terrestrial network.

With these six characteristics in mind, the following is a discussion of some of the technical issues facing interconnection of ISDN and DSN.

5.6.1 Processing of Voice and Non-voice Traffic at an Earth Station.

The question here is whether the traffic from different services (ie. voice, interactive data, file transfer, facsimile, etc.) should be processed differently before being transmitted via a satellite or should it be processed in an integrated manner using common equipment. An argument can be made that it would be advantageous to divide the traffic into groups (at least two groups, voice and non-voice) and process them separately. Here are some reasons to support this view:

1) It would be possible to use multiple access techniques for different groups of services such as synchronous and asynchronous data.

2) The large variance in traffic source characteristics (message length, call generating rate, holding time, delivery delay requirements, etc.) could be better managed at the earth station.

3) Various baseband coding or interpolation techniques could be applied and tailored to various services for more efficient use of satellite bandwidth. Examples are delta modulation, non-instantaneous companding, DSI and facsimile encoding.

4) The differing performance requirements for various services could be met specially encoding each service for optimum satellite power/bandwidth utilization.

5) It would enable DSN to adopt different charging policies for different classes of service.

Such separation of the various services, say only voice and non-voice, would not require segmenting these two types within ISDN. {27} It would only necessitate each type being identified (say, by a message ID sent through the signalling channel) to the earth station as it is transmitted.

5.6.2 Handling the Synchronous and Asynchronous Traffic in a DSN.

1) Synchronous traffic - If the DSN is to connect two national ISDNs, CCITT Recommendation G.811 states that the interconnection be accomplished in a 'plesiochronous' (plesio = Gk -near, chronous = Gk- time) time. This means the digital bit streams have to be processed in a plesiochronous aligner which performs primary level frame slips. The bit streams must then go through a Doppler buffer which removes the effect of the clock uncertainty present in the satellite system. {27} Voice traffic is then interpolated while data traffic is not.

2) Asynchronous traffic The first method of handling asynchronous data involves the method describe in chapter 3 using dynamic TDMA and statistical multiplexing. Another method involves handling the traffic as packets capable of accessing the satellite transponders using random access techniques.

5.6.3 Improvement of ISDN Through a Digital Processing Satellite.

This becomes useful where a satellite is connecting two nodes using packet switching techniques. With a conventional satellite using a passive repeater the packet data concentration is done at the packet nodes. However, it is possible to deploy a digital processing satellite with on board

demodulation/modulation and storage capabilities. Now the concentration can be done on board the satellite. This has been shown to increase the throughput for a packet network. {27}

This concludes the discussion of user interfaces for ISDN. We covered user access arrangements, user equipment configurations, protocols, an example of how ISDN standards are shaping the development of new user interfaces - DMI, and satellite interfaces for terrestrial ISDNs. It must be remembered that those people responsible for the design of ISDN stress that it is an evolutionary concept. Its design will go on evolving and its implementation will come about over the next two decades. DMI, with its two step signalling procedures and three modes of operation, is a perfect example of how the industry is gradually bridging the gap between existing interfaces and ISDN.

A recent ISDN study by the management consulting firm of Booz Allen & Hamilton was published in the 12/15/86 issue of Communications Week and is appropriately entitled " ISDN - Putting The Vision In Focus ".It provides us with a good snapshot of where ISDN is today, both real and perceived, and offers some noteworthy observations about the state of ISDN which make for a fitting summation to this study. For it is the marketplace which offers the final judgement on any new technology regardless of its apparent merit or lack thereof.

Users don't want a risky new technology which may be able to solve all their communication problems. They want increased reliability, improved diagnostics, more bandwidth, and more user control. ISDN offers all these and more. While most users mentioned a need for integrating voice and data, few thought of this in the ISDN sense of sending separate media simultaneously over the same line from the same terminal. Most users refered to integrating existing voice and data networks. Clearly, ISDN must demonstrate its ability to meet current needs in a way that reduces overall costs and risks.

Some of the forces driving ISDN, as indicated in the above study, are:

- User dissatisfaction with existing data communications networks and requirements for better services.
- Intensifying competition for large user accounts among vendors.
- Continuing complexity of private networks and increased usage of T1 type links.
- Increased transmission of images along with voice and data as electronic mail, facsimile, and teleconferencing proliferate.

The key to the future of ISDN may very well be evolution. One scenario shows ISDN evolving over the next decade in three stages:

Stage 1:

Digital switching
Pre-ISDN services (ie. virtual private network network services)
Local optical fiber
Signalling System No. 7 deployment
ISDN trials
Private facilities integration

Stage 2

Standard interfaces
Integrated network access
Dynamic network control
Public ISDN services in major cities

Stage 3

High speed data/video
Integrated packet and circuit switching
New services and applications
Widespread deployment

In any event, whatever its final form, ISDN is coming. It appears no ribbon will ever be cut announcing its "official" arrival. Instead there continues to be the piece-by-piece implementation/ transformation / digitization of new / newer / newest transmission facilities as part of an accelerating process rather than a discrete event. Prototype ISDN's exist in the United States and Europe. Most forms of information can now be digitized, including voice, graphics and video. Once put into digital form this information can be stored,

processed(altered), retrieved, and transmitted with far greater reliability and at a lower cost than ever before, which is what ISDN is all about.

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