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A

# Intelligent ATM Networks

by

**Bouchra Ejjaki**

**A Dissertation submitted to the Graduate Faculty in Computer Science  
in partial fulfillment of the requirements for the degree of Doctor of  
Philosophy, The City University of New York.**

1999

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This manuscript has been read and accepted for the Graduate Faculty in Computer Science in satisfaction of the dissertation requirements for the degree of Doctor of Philosophy.

January 27, 1999

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# Abstract

## INTELLIGENT ATM Networks

by

**BOUCHRA EJJAKI**

**Adviser: Professor Syed V. Ahamed**

Intelligent Networking involves the interconnection of intelligent nodes such as switches and real time data bases via extended signaling (in Common Channel Signaling 7) to provide value added services and features. In the context of Intelligent Network Conceptual Model (INCM), these value added services have become a standard for the modern telecommunications industries. A wide variety of networking technologies are also now available to address the rapidly emerging class of bandwidth intensive applications. These broadband networking technologies include the fast-packet schemes, such as frame relay (SONET/SDH) and cell relay (ATM).

In this research, the intelligent networking will take one step further supporting any kind of communication networks intelligent services next to the current IN services based on circuit switched networks. The intelligent ATM networking is also able to support any type of service. By developing an open intelligent ATM platform including universal signaling system and management system, we aim to develop a truly virtual community where most of the broadband real time information can be accessible anytime and anywhere provided the ATM backbone can be deployed.

The basis of Intelligent ATM Networks is to provide intelligent services by accessing multiplicity of SCP's (service control points). These SCP's will be a network connecting different Files Servers. These services may include voice-based information services as well as other special applications (such as educational, medical, financial, etc.). Unlike the current Intelligent Networking

where an SSP (Service Switching Point) is triggered for IN services related to telephony, the futuristic intelligent ATM networking the file servers will be triggered for intelligent services related to any type of communication service. The access to the file servers will be invoked by a trigger from a local or tandem switching access point. Although the signaling network which will be based on X.25 protocol at the beginning, it is expected that the signaling network will also be based on the ATM protocol using a quasi-associated mode with the current X.25 protocol.

In both these cases, new modules are to be defined. It is suggested that the signaling network will be based on a cell command (as in computer instruction) and a fiber for transmission (as in case of a bus input and output in the case of computer system). A generic FireWall type of separation from the traditional voice and data network is suggested. In this network, a bank of super computers will have the ability to process any kind of query, data, information or knowledge or even an intelligent service request, create a cell based command, and then access to a large number of file servers through a localized ATM network for more instructions relating to the query. The ATM cell should be modified to distinguish between the type of query (call). In case of an intelligent service request, the call control processing element (with its own database) will change the structure of the cell header by making the reserved bit a 0 instead of the default 1. The ATM fabric will recognize the cells requiring special attention and forward them to the right IP (intelligent peripheral) germane to the type of service.

This study suggests a global architecture for Intelligent ATM Networks, signaling and transport, and some new elements still need to be defined. The ATM cell header is modified to facilitate the ATM intelligent services. The current electronic switching system need only minor modifications to include the effect of distributed processing. A national Intelligent ATM Platform also needs to be evolved for extended simulation results.

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# Intelligent ATM Networks

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# Chapter 1

## **BASIC COMMUNICATIONS ENVIRONMENT**

Once limited to voice and low-speed data transport, the global communications network is adjusting to the accelerated introduction of data. In the past decade, data has supported voice as the predominant source of public network traffic. Hospitals transmit X-ray images to specialists hundreds of miles away, students tap into distant research libraries, and executives log onto office networks from home. However, these are still the exception rather than the rule.

Data processing became, although not necessary, a part of communication systems. It helps in solving business or scientific problems. Communication systems technology is evolving rapidly, with computing capabilities, becoming widely distributed in the home, office, and factory. Information is being exchanged between humans and machines. This information exchange is a necessity between a wide range of locations, including the home, businesses, and

factories, both domestically and internationally. These factors have initiated the formation of an Information Movement and Management (IM&M) industry.

Communication networks are a collection of entities that function in a collaborative role to convey information from its source to its destination following an accepted code of standards. These standards set guidelines for monitoring the functions of networks in their macroscopic forms. To maintain the integrity of the networks and an accountability of their operations, large functions are broken down into more manageable modular levels. Such modular functions can be assembled, disassembled, and repackaged much like computer functions in hardware modules, software packages, and microprograms. However, additional element of complexity functions have to respond in real time. Computer systems and networks do not always have a human recipient of information. To this extend, the communication networks must satisfy a more stringent requirement at the human interface.

Scientists have been working on making any information available to anyone authorized to receive it, at anytime and anyplace. The quick access and dependability of typical computer systems are incorporated in these networks. These networks are programmed to be adaptive, algorithmic, resourceful, responsive, and intelligent.

## 1.1

---

# TRANSMISSION ENVIRONMENT

Connectivity of voice and non voice terminal equipment at user premises ends on various transmission facilities such as wire pairs, coaxial cable, microwave radio, fiber optics, and geostationary satellites. Telephone offices are connected to these media to provide customer services. Cables consisting of hundreds of wires provide connection in a metropolitan area. Coaxial cables, radios, satellites, or combination of both make the connection over long distances.

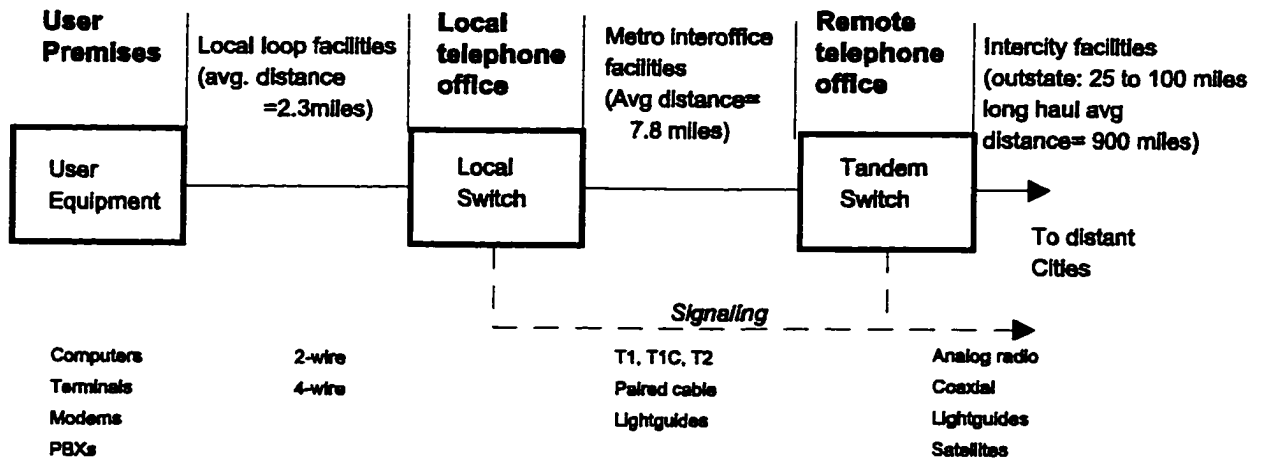
The basic elements of a telecommunication system connects a source to the receiver (Figure 1.1) through the various transmission media. At the user's premise, the equipment can consist of computers, terminals, modems, and PBXs. The modems and PBXs may be maintained by the user or telephone company at the customer's discretion. Typically, customer equipment is connected to the local telephone office switch via the local loop (loop subscriber) which usually consists of two wires. Other equipment can be digital multiplex that can combine up to 96 voice channels for transmission to user locations. This digital connectivity uses the same voice digitalization techniques (PCM) as the T1-carrier system.

Digital technology permits half-duplex operations on wire pairs that looks like full-duplex to the user. High speed data bursts are sent in alternate directions between the user and local office in a "Ping-Pong" fashion. This is called time compression multiplexing.

The metro/interoffice facilities between the local telephone office and another telephone office within a metropolitan area consists of many wire pairs in a cable

used for transmission of the T1 carrier (24 channels), T1C carrier (48 channels), T2 carrier (96 channels).

The intercity facilities may consist of analog microwave radio, coaxial cable, light-guides and satellites. All are capable of extremely high-capacity message transmission.



**Figure 1.1: Telecommunication network elements and facilities;** T1 carrier (24 Channels), T2 carrier (48 channels), T1C carrier (96 channels).

The transmission of information needs intelligence. The network can choose between a large number of paths and circuits. The network conditions strongly influence the path selected. These paths may be quite physical such as pair of wires, or they may be one or more channel multiplexed over one physical media, such as wire pair, a coaxial cable, an optical fiber, a digital radio circuit, or any viable medium for data transmission. Each transmission medium has a number of advantages and disadvantages. The primary differences between media are their cost and ease of installation, the bandwidth of the cable which may or may

not permit several transmission sessions to occur simultaneously, the maximum speed of communications permitted, and the geographic area scope of the network that the medium support.

### **1.1.1**

---

## **Twisted Pair**

Are the earliest and currently the most commonly used data transmission medium. Much of the terminology and technology regarding this communication medium derive from telephony and telegraphy because, in setting up its own data communications networks, the computer industry used the existing network of telephone and telegraph lines.

The twisted pair consists of two insulated copper wires arranged in a regular spiral pattern. A wire pair acts as single communication link. Typically, a number of these pairs are bundled together into a cable by wrapping them in a tough protective sheath. Over long distances, cables may contain hundreds of pairs. The twisting of the individual pairs minimizes electromagnetic interference between the pairs.

Twisted pair is the backbone of the telephone system as well as the workhouse of intrabuilding communication. In the telephone system, individual telephone sets are connected to the local telephone exchange or "end office" by twisted-pair wire. These are referred to as "local loops". Within an office building, telephone service is often provided by means of a private branch exchange (PBX).

Digital data traffic can also be carried over moderate distances. For modern PBX, data rates of about 64 kbps are achievable using digital signaling. Local loop connections typically require a modem, with a maximum data rate of 9600 bps. However, twisted pair is used for long distance trunking applications and data rates of 4 Mbps or more may be achievable.

The advantages of wire include their availability and relatively low cost. Their disadvantages include susceptibility to signal distortion or error and relatively low transmission rates they provide for long-distance links. For Analog transmission, amplifiers are required every 5 to 6 km. For digital transmission, repeaters are required every 2 or 3 km.

### **1.1.2**

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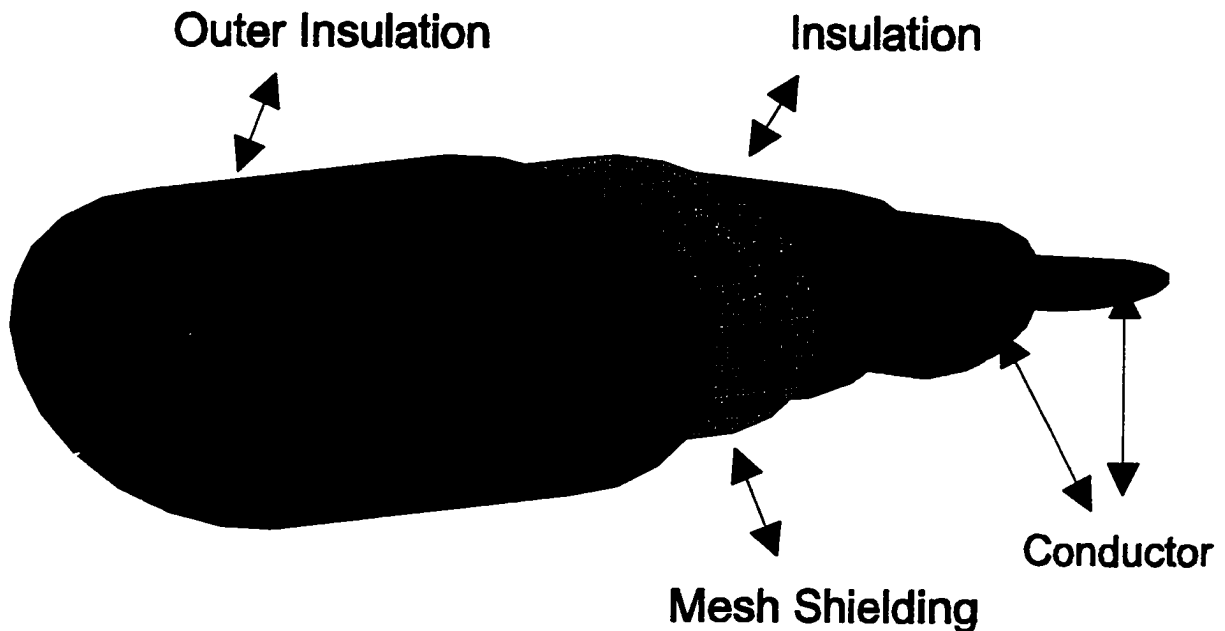
## **Coaxial Cable**

Coaxial cable comes packaged in a variety of ways, but essentially it consists of one or more two central data transmission wires surrounded by an insulating layer, and outer jacket (Figure 1.2).

Coaxial cable is primary used in LANs or over relatively short distances, generally fewer than 10 miles (except for use by common carriers) to carry analog and digital data effectively. Most LANs are privately owned and are restricted to relatively small geographical area.

Coaxial cables are also used for distributing analog cable TV signals. The societal demand for carrying data has been discouragingly low. However, the potential

of these CATV systems for carrying high capacity digital information exists and needs serious exploitation. Interfaced to intelligent networks, they can distribute highly customized data to a large number of subscribers bringing in new services generally not available in the cable TV industry. The FM hierarchy of the CATV network that permit the transfer of small quantities of upstream data to control large streams of downstream data makes these networks potential contenders for very personalized and individual intelligent video network services.



## A Single Conductor Cable

**Figure 1.2: a single conductor cable**

Coaxial cable has been an important part of the long-distance telephone network, although it is rapidly supplanted by optical fiber, microwave, and satellite. Using frequency-division multiplexing (FDM), a coaxial cable can carry over 10,000 voice channels simultaneously.

Coaxial cable is also used to connect terminals with terminal controller units. Cable networks servicing a large sector of customers prevail in some nations. Cable networks between Central Offices to carry high density interoffice traffic exist in most nations. Data transmission is up to 100 Mbps (100BT) are becoming more and more common, and theoretical bit rate is more than 400 Mbps. However, their capabilities and costs are continuously challenged by the emerging fiber optic networks. Bringing the fiber to home has become a serious concern for many telephone operating companies, and to desk for most designers of premises distribution systems.

Because of its shielded, concentric construction, coaxial cable is much less susceptible to interference and crosstalk than twisted pair. The principal constraints on performance are attenuation, thermal noise, and intermodulation noise. The latter is presented only when several channels (FDM) or frequency bandwidths are in use on the cable. To achieve a proper signal quality, a certain signal-to-noise (S/N) ratio must be maintained on the cable. Two variables are then to consider: the signal power and amplifier spacing. The S/N can be raised by spacing amplifiers closely to boost the signal frequently. However, it is

desirable to maximize amplifier spacing to reduce cost and because amplifiers introduce nonlinearities.

For long distances transmission of analog signals, amplifiers are needed every few kilometers, with closer spacing required if higher frequencies are used. For digital signaling, repeaters are needed every kilometer or so, with closer spacing needed for higher data rates.

In Broadband transmission the data is carried on high-frequency carrier waves; thus, several channels may be transmitted over a single cable. Frequency separation, referred to as guardband, helps keep one signal from interfering with another. Broadband technology allows one medium to be used for a variety of transmission needs, so that the voice, video, and multiple data channels of varying transmission speeds could all exist in one cable. A subchannel with frequencies between 200 and 250 million Hertz (MHz) might be used to carry video data, a subchannel operating between 175 and 200 MHz could be carried on.. make sure that values are correct.

Baseband transmission, on the other hand, does not use a carrier wave but sends the data along the channel by voltage fluctuations. Baseband technology cannot transmit multiple channels on one cable, but is less expensive than broadband because it can use less expensive cable and connectors. Some coaxial cables can be used for either broadband and Baseband.

### 1.1.3

---

## Fiber Optic Cable

The bandwidth capacity of fiber is now being exploited for digital transmission. Thinner than a human hair yet stronger than steel, fiber has become the basic medium for long-haul and interoffice telecommunications networks. The special characteristic of optical fiber is its low impedance to lightwaves and its tremendous modulation capacity, supporting higher bandwidths over longer distances—enough to accommodate tens of thousands of voice channels. Researchers have transported information at rates exceeding 350 Gb/s. At that rate, the entire content of 1.2 million books could be transported around the globe in a minute!. Unquestionably, one of the most dramatic changes in telecommunications over the last decade is the vast amount of fiber embedded in the public network, which provides almost unlimited bandwidth potential.

The fastest commercial fiber-based system today launches pulses down a fiber cable at roughly 2.5 billion pulses per second at much lower error rates (10<sup>-12</sup>) than the error rates (10<sup>-8</sup>) for the copper that it replaces. This permits high speed protocols that would not have worked over the error-prone copper lines. It also makes possible the simultaneous transmission of high-resolution video, digital audio, and data possible in the LAN and WAN environments.

Its use in LANs and WANs has created a demand for two types of fiber optic cable – single-mode and multi-mode.

Single-mode fiber cable is used in the public WAN for several reasons. Information travels greater distances without requiring repeaters, and it removes the bandwidth limitations of multi-mode fiber (which operates below 155 Mb/s). On the other hand, multi-mode fiber cable is more likely to be found in buildings because it is simpler to install, is lower in cost, and is supported by major computer companies. Consequently, multi-mode fiber cable has become the favorite for fiber LANs such as FDDI.

Comparison of Single-Mode and Multi-Mode Optical Fiber		
Source	50	125
	62.5	125
	85	140
	100	140
Step-Index Multi-Mode		
Source	50	125
	62.5	125
	85	140
	100	140
Gradual-Index Multi-Mode		
Source	7-9	125
Single Mode		
Note: 7 to 9 microns is equivalent to 1/30000th of an inch	Core Diameter (Microns)	Outside Diameter (Microns)

**Figure 1.3 : Comparison of Single-Mode and Multi-Mode Optical Fiber.**

## 1.1.4

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### **Microwave**

Microwave refers to the high end of the radio-frequency spectrum, from about 1 to 30 GHz. Microwave antennas can produce either highly directional or broader beams to cover large areas. Microwave transmission presents three challenges to the network designers.

- *Frequency Allocation.* High speed data communications is relatively recent application that must make use of frequency bands that preexisting applications are not already using.
- *Interference.* One important characteristics of microwave signals is that they can penetrate walls. This has the advantage that a microwave based LAN need not to be confined to a single room. However, if several wireless LANs are operating in the same building, there is the potential for interference. Interference can be avoided by prior agreements about frequency bands or by using spread spectrum techniques.
- *Security.* Because microwave signals penetrate walls, security becomes a consideration. One technique to increase security is to encrypt all transmission and carefully control the distribution of the encryption keys. Also, spread spectrum transmission can provide security if properly implemented.

### **1.1.4.1. TERRESTRIAL MICROWAVE**

The most common type of microwave antennas is the parabolic "dish". A typical size is about 10 ft on diameter. The antenna is fixed rigidly and focus a narrow beam to achieve line-to-sight transmission to the receiving antenna. Microwave antennas are usually located at substantial heights above the ground level in order to extend the range between antennas and to be able to transmit over intervening obstacles.

The primary use of terrestrial microwave systems is in long haul telecommunications service, as an alternative to coaxial cable or transmitting television and voice. Like coaxial, microwave can support high data rate over long distances. The microwave facility requires far fewer amplifiers or repeaters than coaxial cable for the same distance, but require line-of-sight transmission. Microwave are used for both voice and television. Finally, a potential use of microwave is for providing digital data transmission in small regions (radius < 10 Km). This concept has been termed "local data distribution" and would provide an alternative to phone lines digital networking.

Common frequencies for microwave are in the range 2 to 40 GHz. The higher the frequency used, the higher the potential bandwidth and therefore the higher the potential data rate (e.g., (2 GHz Band, 7 MHz bandwidth, 12 Mbps Data rate), and (18 GHz Band, 220 MHz bandwidth, 274 Mbps Data rate)].

**The main resource of loss is attenuation:**

$$d = 10 \log(4\pi d/\lambda)^2 \text{ dB}$$

where  $\lambda$  is the wavelength and  $d$  is the distance.

Note that the loss in microwave varies with the square of the distance as in case of coaxial or twisted it varies logarithmically with the distance. Attenuation is increased with rainfall.

Another source of impairment for microwave is interference. With the growing popularity of microwave, transmission area overlap and interference is always a danger. Thus the assignment of frequency bands is strictly regulated.

#### **1.1.4.2. SATELLITE MICROWAVE**

For a communication satellite to function effectively, it is generally required that it remains stationary with respect to its position over the earth, then must have a period of rotation equal to the earth's period of rotation.

If using the same frequency band, two satellite located close enough together will interfere with each other. To avoid this, current standards require a 4° spacing (angular displacement as measured from the earth) in the 4/6-GHz band, and a 3° spacing at 12/14 GHz. Thus, the number of possible satellites is quite limited.

The satellites are used for television distribution, long distance telephone transmission, and private business networks.

The optimum frequency range for satellite transmission is in the range 1 to 10 GHz. Below 1 GHz, there is significant noise from natural sources, including galactic, solar, and atmospheric noise, and the human-made interference from various electronic devices. Above 10 GHz, the signal is severely attenuated by atmospheric absorption and precipitation.

Most satellite providing point to point service use a frequency bandwidth in the range 5.925 to 6.425 GHz uplink transmission, and 3.7 to 4.2 GHz for downlink transmission. The combination is referred as C band.

The C zone (1 to 10 GHz) has become saturated. Therefore, the 12/14 GHz band, or Ku band, has been developed overcoming attenuation problems, but requiring higher uplink and downlink radiated power and greater transponder receiver sensitivity.

For the use of satellite in mobile communications (L band), small, low cost transmitter/receiver devices are used, and transmission is full-duplex directly between the users and the satellite. The service offered is similar to cellular radio, except that the overage is nationwide.

Delay from one earth station to the other, and the fact that many stations can transmit to the satellite or a transmission from one satellite to many stations remain a main inconvenient of satellite.

### **1.1.4.3. RADIO WAVES**

Unlike satellite, radio is omnidirectional and doesn't require dish shaped antennas. It is used to cover CHF and part of UHF band (30 MHz to 1 GHz).

Transmission is limited to the line of sight and distant transmitters will not interfere with each other due to reflection from the atmosphere. Radio waves are less sensitive to attenuation from rainfall. In digital communication, low bit rates have been achieved. Radio waves suffer relatively low attenuation because of the longer wavelength. A prime source of impairment for radio waves is multipath interference (reflection from land, water, and natural or human-made objects can create multiple paths between antennas).

### **1.1.5.**

---

#### **1.1.5. Role of Silicon Integrated Circuits in Transmission System**

High technology markets are fueled by the economies afforded by integrated circuits. Silicon architectures that duplicate simple atomic elements to form massively parallel systems are the basis for modern broadband technologies. With simple solutions in silicon, larger and more complex systems are economically possible. One result is the rapid evolution of the microprocessor, which is the engine of modern computers. These silicon chips continue to increase in function, power, and speed. The microprocessor has grown from the four-bit bus of the 1970s to the eight- and 16-bit bus of the 1980s, to the 32-bit bus of the 1990s. Today microprocessors using 64-bit buses are on the near horizon, creating integrated circuits that maintain the processing power limited to the mainframe computer only a few years ago.

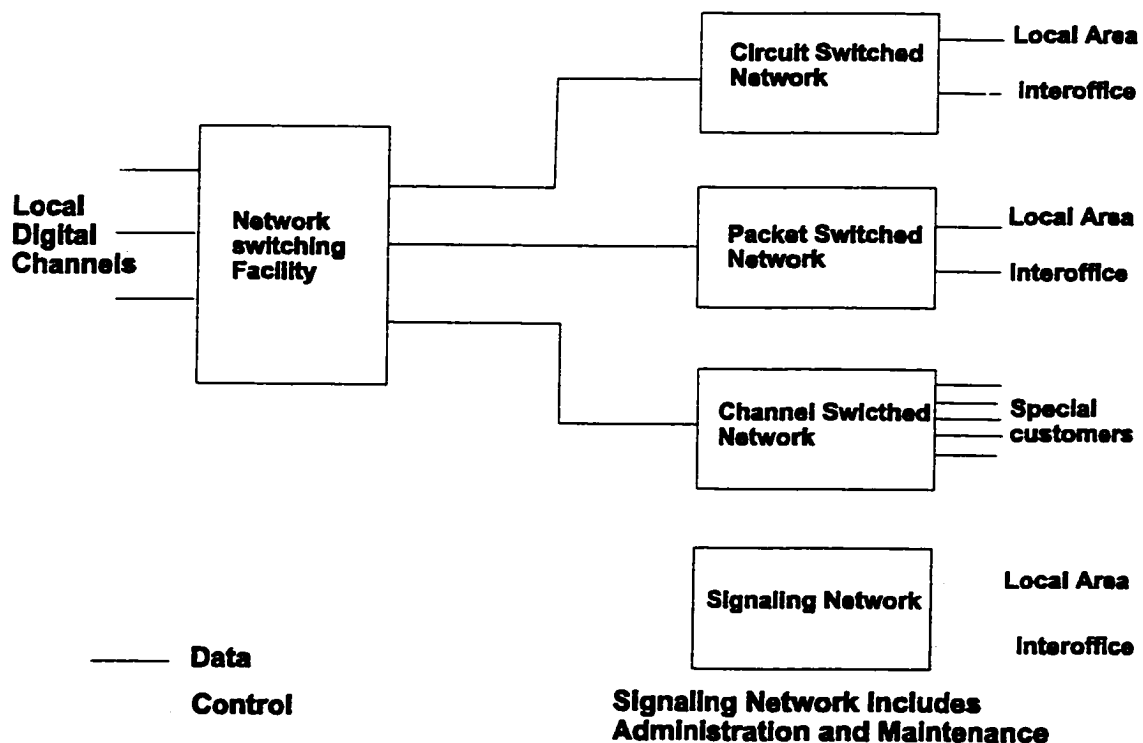
Power is not the only factor; the price of computer components has dropped considerably. If these trends continue, it will be possible to embed massive intelligence into everyday appliances. A future interactive television costing only a few hundred dollars could incorporate a 32 bit microprocessor and 16 megabytes of memory. With a connection to a broadband network based upon SONET/SDH and ATM chip sets, the television could present a friendly graphical interface with full motion video and voice response. It could transparently search for programs. A student in New York City could learn Japanese with text from a data bank in Tokyo overlapped with pictures from an image bank in London -- all in real-time.

The LAN revolution became reality when Xerox's Ethernet became a de facto standard with the backing of DEC and Intel. DEC supplied the customer base and Intel supplied the integrated circuits that resulted in affordable connection devices. The result was a form of broadband network -- the LAN. For some, the term broadband network evokes memories of the early LAN implementations that used multiple frequencies to transmit data over coaxial cable. Even today one of the largest application of broadband networking is in the cable TV industry where multiple channels are broadcasted to millions of households. A similar scenario is occurring with WAN communications where SONET/SDH and ATM chip sets are powering a variety of advanced broadband network elements. Communication circuit silicon suppliers are delivering the engines for the next wave in networking.

## 1.2.

## SWITCHING ENVIRONMENT

In a communication environment, the switching of the information bearing channels can be performed in three different ways. First, the channel can be switched by the network and allocated a certain user for as long as the user may need it and the channel is said to be circuit switched. Second, the channel can be packet switched which means the information is to be packetized and dispatched to the appropriate destination, either as an individual packet (if the message is short) or as series of packet if the message in an appropriate sequence to complete an informational transaction. Third, the switching can be performed by the channel itself and the information is said to be channel switched.

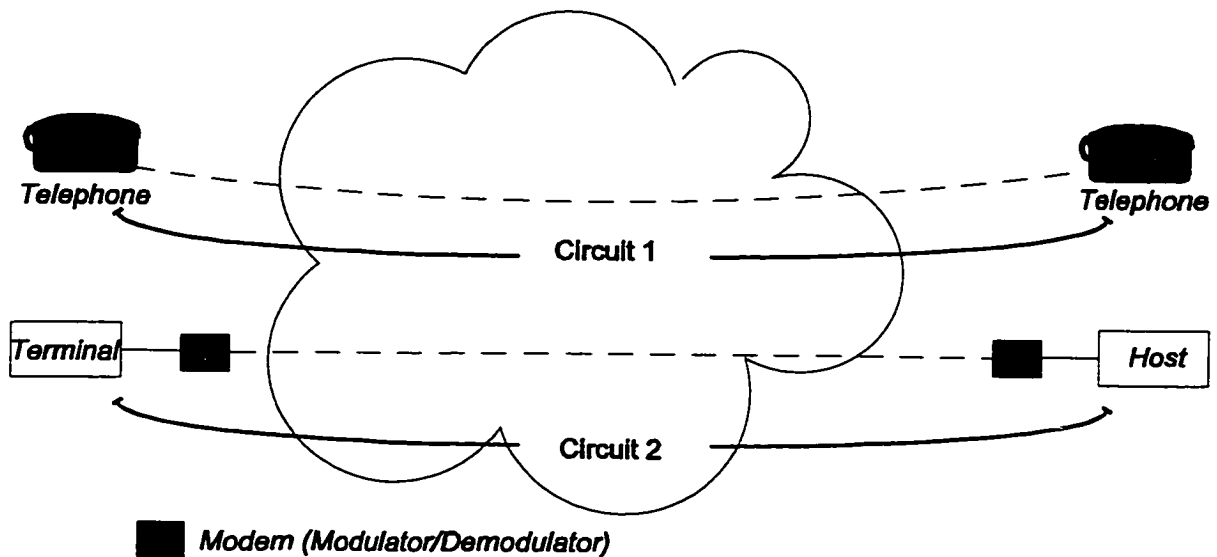


**Figure 1.4. Integration of various networks for the transport of data**

## 1.2.1

### Circuit Switching Networking

Circuit networks began with the standard telephone network (private lines and circuit switched). A circuit network (e.g., a telephone network) provides a **CIRCUIT** to its users by offering an end-to-end connection with constant delay where information is delivered in sequence. In this type of connections, the network is transparent to user's information, and there is an inefficient use of network connection.



**Figure 1.5. Evolution; Circuit network**

In these architectures (connection oriented), there are *five* stages for data transfer through a switched channel. The sequence may be summarized as follows: an idle channel is identified and tagged, a connection is established, the data is transferred, the channel is released, the channel resumes to be idle again. This



sequence of stages parallels the well established network steps in a typical voice call which follows the network functions: call setup, alert, connect, disconnect and release. In reality, the detailed functions within the network far exceed the five major steps listed earlier. For example, at the outset, the line scanners detect the off-hook condition. The scanning rate is enhanced to collect the dialed digits in the older exchanges or the tone detectors are invoked to identify the touch-tone frequencies. The call setup procedure is initiated after the digit collection is complete. Such a sequence of well identified responses (network macros) are “assembled” until the call is completed as a successful call or is terminated by user or the network. Such responses (macros) are amenable to computer environments rather than to mechanized systems, and this is the basic reason for the “call processing” environments in most of the digital central offices.

In context to the ISDN, the information necessary to accomplish these individual steps are incorporated in the ITU-T (formerly CCITT) Q.931/931 protocol at the network layer used over the D (Delta) channel to set up the circuit switched B (Bearer) channel.

### **1.2.2.**

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## **Packet Switching Networking**

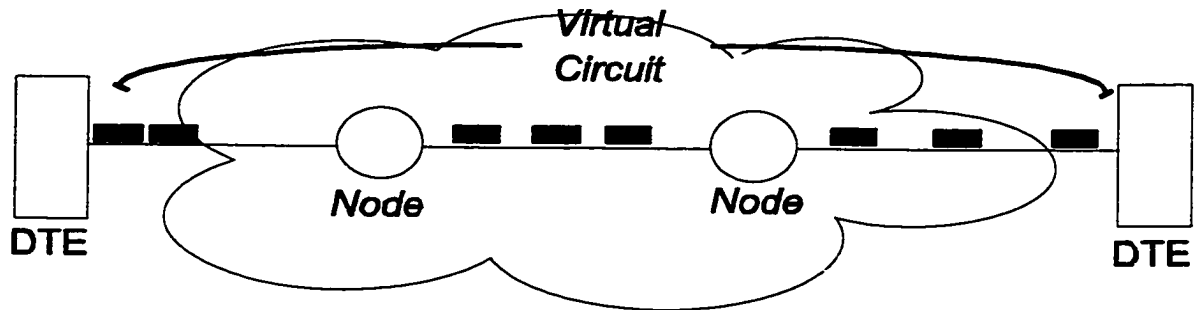
Based on a connectionless oriented architecture where individual packets of information are assembled with their own header block including its sender and destination addresses and the sequence number of the individual packet. The

packets are delivered in sequence through an end to end connection, called virtual circuits (i.e., almost a circuit), in a packet switched network.

After, the packetizing operation, comes the store and forward operation where the DTE transmits packet to first node where examine packet header is examined to find the addressing information. From the routing algorithm, the node determines on which output line (link) to send packet. If output line is available, the packet will be send to the next node on the path. If the line is NOT available then the packet is temporarily stored at the node (i.e., queued) and is transmitted when the output link which has been selected by the routing algorithm is available. This algorithm is repeated at every node in the path and at each node, a packet may be temporarily STORED and then FORWARDED (variable delay). For example, e-mail uses store and forward operation at application. The network is no longer transparent to what the user is sending i.e., format of packet and contents of header must be agreed on. CCITT recommendation X.25 provides the functional guidelines for implementing packet switching. There are many X.25 packet-switching networks in existence within US and abroad.

Packet switching is more efficient for certain applications than circuit switching. Any one of the packets may transit through any variable number of nodes until it reaches its destination. The exact physical routing depends on the dynamic network conditions. The innate functioning of every participating node through the network facilitates the correct delivery of the packet. In context to ISDN

packet-switching, ITU-T (formerly CCITT) has specified the widely accepted X.25, X.75, X.28, and X.29 protocol at the network layer.

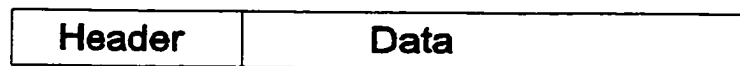


**DTE= Data Terminal Equipment**

*(any User Information Processing Device)*

**Node= The Network Packet Switch**

**■ = Data Packet= Header + Data**



*Note: Typical length of a Packet is 128 Octets*

*Header of Packet contains addressing information to indicate destination*

**Figure 1.6. Packet-Switched Network (ex. WAN)**

At the network level, the two (circuit and packet) architectures function cooperatively. At the media level, the data appears as encoded signals. At the switch level, the routing and channeling functions occur in the (synchronous or the asynchronous) time division multiplex (TDM) framework. At the signaling level, the ITU-T (formerly CCITT) standards for both modes of data transfer are well accepted throughout the world. In fact the modern networks strive to be massively distributed, but well modularized computer systems. These networks

are designed with the objectives of being as broadband as optical systems, of being as accurate as a computer system and of being as evolved as the OSI model with the ITU-T standards. This transition can only be a slow and insidious process because of the expanse of most national networks.

At the media level, wired, coaxial, wireless (cellular and microwave), satellite and optical facilities are most common. Frequency division (for the cable, wireless, satellite, etc.) and time division multiplexing exist for the synchronous (such as frame relaying in fiber-oriented SONET) and asynchronous (such as cell relaying also in fiber-oriented ATM) networks. Differences in concept and methodology do exist but the emerging all-digital networks tend to be broadband, architecturally well planned and designed to cope with congestion, well oriented to the OSI model, and operate with the worldwide ITU-T standards and protocol.

The network switch designers strive to successfully switch all customer (circuit-switched, packet-switched, message-switched, private-virtual, frame-relayed and cell-relayed) channels over all (optical, wired, cellular, wireless, cable) media with appropriate interfacing and signaling. In a sense, this assures the compatibility and cooperation between any of the major networks' ideologies. The integrated network can thus be all-digital and all-services oriented rather than be only POTS oriented as it used to be.

## **1.2.4.**

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### **Message Switching Networking**

Message-switched networks stores entire messages at switch location for transmission on circuits when message channels and the called terminal become available. Each switch location has a memory disc for message storage and subsequent forwarding (store and forward) to the next nodes. Telegram service is an example of message switching as the complete message is stored at each successive node while awaiting transmission for the next node. Message switching is used where tradition and procedures require that a hard copy be provided for record purposes such as in military communications. Messages are usually sent and received on teleprinters at each user location.

## **1.2.3.**

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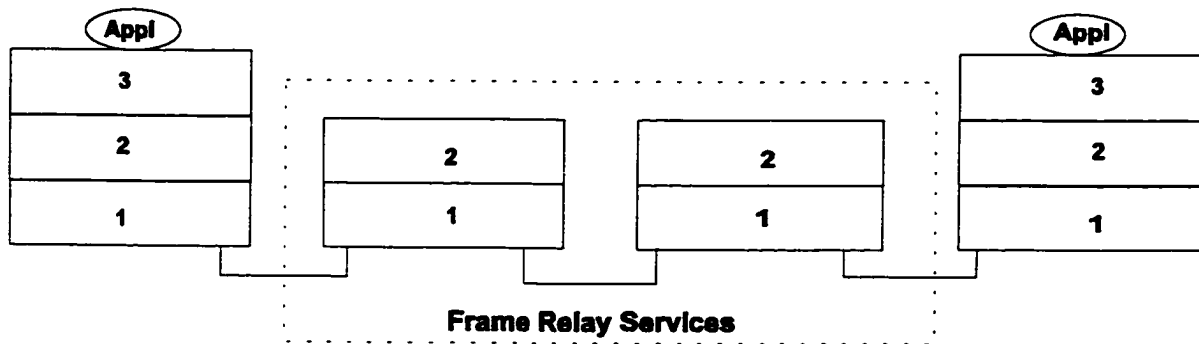
### **Frame relay and Cell Relay**

Frame and cell relay are used in high rate operating networks. The main difference between the two is that frame relay uses variable length frames to transport the data, whereas cell relay uses fixed length frames called cells.

#### **1.2.3.1. FRAME RELAY**

Frame relay is a protocol, not a service, for WAN. It provides high speed/low delay packet transport on fiber allowing interconnectivity of LANs . The rates are at 1.5 Mbps (DS1) and above. Frame relay was designed for digital technology covering layer 1 and 2 of the OSI scheme (Figure 1.7), then the

capability of detecting but not correcting errors. The corrections are left to the higher levels within the application itself.



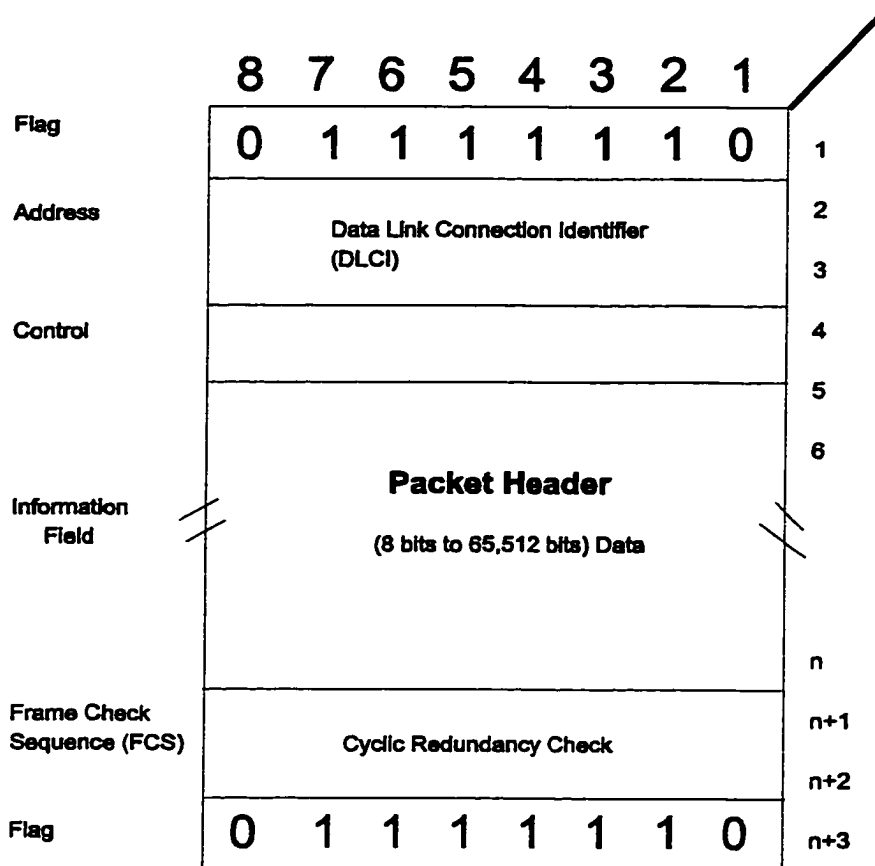
**Figure 1.7. Frame Relay covering OSI 1 & 2**

Figure 1.8 shows the frame format. Frames should always follow each other along the same route because this connection types of protocols provides no sequencing information. There is only destination routing and congestion control information in the address. Since each frame is addressed individually, data from several applications can be multiplexed onto the same link. Bandwidth is not dedicated to any one application which means that the full bandwidth of the circuit could be taken by one application for a short period of time.

The protocols relating to frame relay are the CCITT Recommendations I.122 and Q.922. They add relay and routing functions to the data link layer. Frame relay is a form of multiplexing that transports frames through the network as quickly as possible.

Frame relay approach the performance of private lines while maintaining advantages of a switched environment: 1) Approaches propagation delay of

equivalent private line, and 2) offer a statistical reuse of network resources. Other advantages are: Reduced switch processing and switch delay, supports a large numbers of users, higher bandwidth, and higher speed transmissions.



DLCI Field provide routing ; identifies virtual circuit at each End. Control filed identifies frame type. Information field ma be a packet. FCS provides error detection; frames with wrong CRC are discarded at each node. Flags provide drame delimitation.

**Figure 1.8: Frame format**

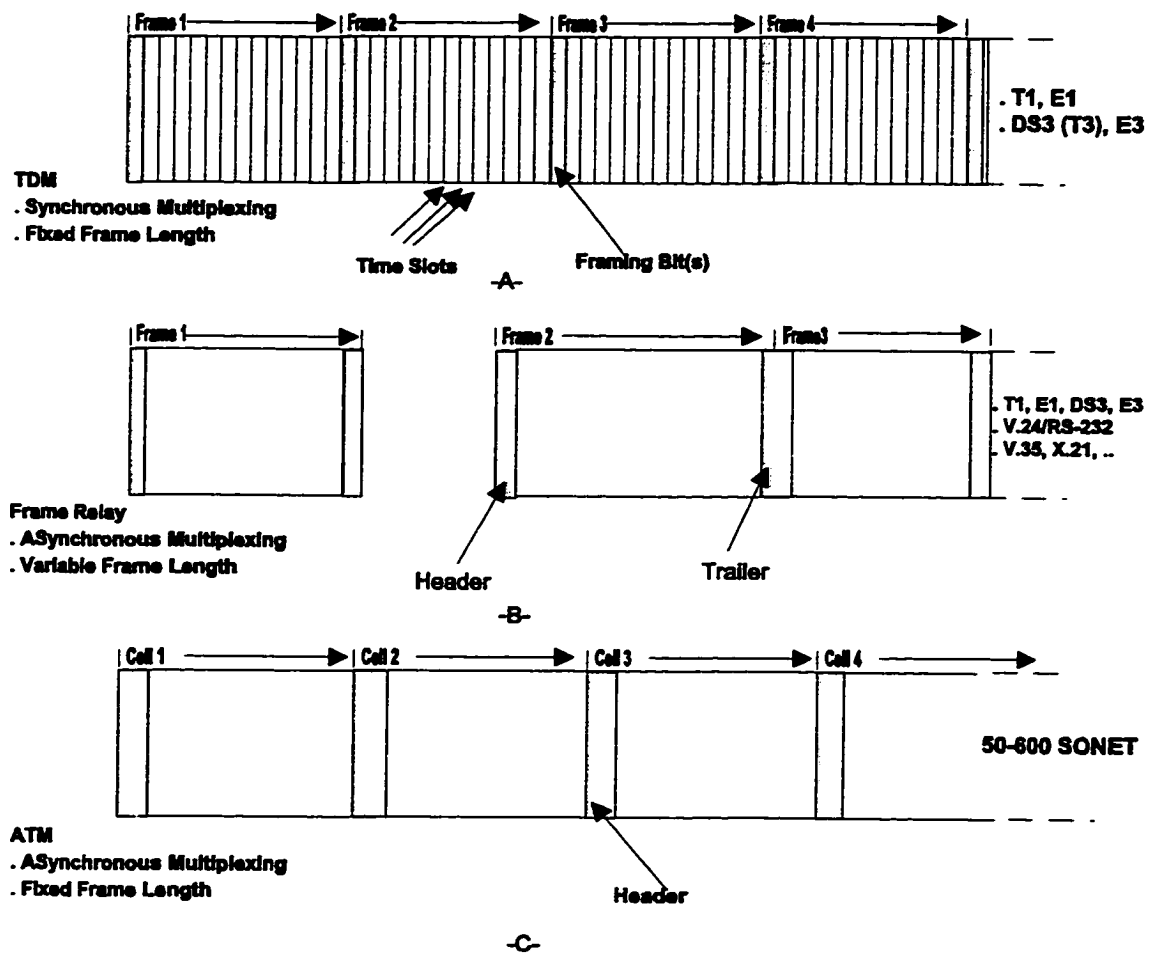
For a point of view of the process of a frame relay data transfer, Figure 1.9.-A-, which shows data and signaling paths through the protocol architecture. For a file to be transferred from A to B, the user (application A) initially sends a request to establish a session via the presentation and session layers to the

transport layer. Call control information is forwarded through the ISDN D channel by the transport layer using CCITT Rec. Q.931. As is also indicated on the same figure, the signaling is used to define a virtual path and call control necessary to set up the data transfer. When the call has been set up, data is transferred from A to B using DLCI in the frame header. The amount of frame processing by the network is reduced to a minimum for frame relay, which facilitates very fast data transfer. One problem with this technique is that in some circumstances reliability can be compromised for speed. In other words, some data can be lost. This is because frame relay has no error correction. This is in comparison with fast packet switching, which guarantees virtually 100% data transfer but takes relatively a long data transfer time [48]. Frame relay is suitable for high volume data transfer such as imaging and visualization. It is less suitable for services such as voice and video that are sensitive to time delay.

### **1.2.3.2. CELL RELAY**

Communication technologies that uses fixed-sized blocks of data to transport information through a network are known as cell relay technologies. The ATM and IEEE 802.6 WAN are examples of cell relay technologies.

ATM is designed to operate over optical fiber (CCITT Rec. I.121). The transmission bandwidth is organized into cells which are periodic sequences of undedicated, fixed-sized blocks of data. This scheme indicated in figure 1.9 -c- where each cell has a 5 byte header followed by a 48-byte block of information data.



**Figure 1.9: Multiplexing methods**

Figure 1.10 shows the signaling and data paths, as a comparison to frame relay

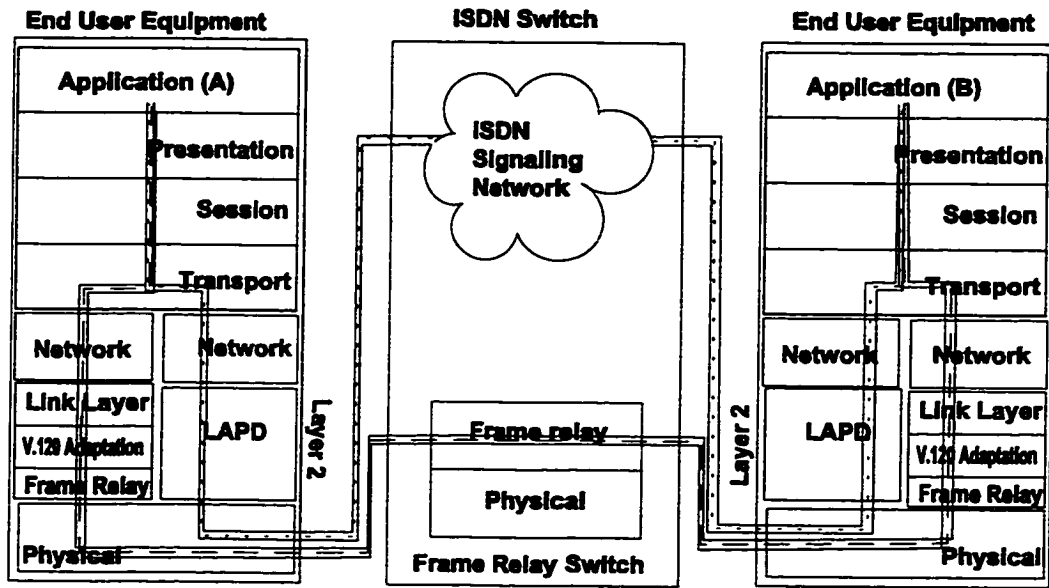
The ATM operates over the broadband ISDN.

The ATM takes care of cell routing, cell multiplexing and demultiplexing, and header error control. The ATM adaptation layer AAL transforms the upper-layer message format and the fixed cell format of the ATM layer. The AAL also performs error detection, correction, and flow control. ATM switches that will

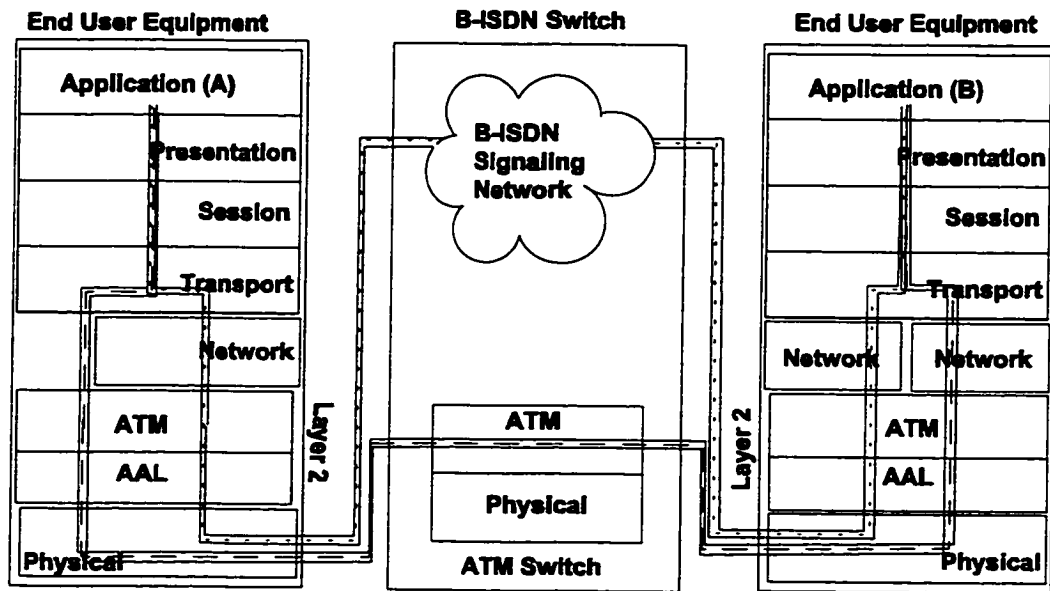
packetize data at the proposed 600 Mbps have not yet been developed. Once occurred, data transfer will be initialized, for example, an application A requesting session establishment to the transport layer, etc., as before. Call control information is carried through the network on a separate virtual circuit. A signaling message is sent by ATM layers convert the signaling virtual channel. The AAL and ATM layers convert the signaling message into cells, after which they are sent to the ISDN signaling network. During cell transfer, cells are routed through the ATM switches by a cell VPI and routing information organized during the call setup procedure. Very fast information transfer is established by minimizing the amount of processing performed by the network.

Cell relay has some advantages over frame relay. First, the switching functions can be performed more efficiently, which translates to a lower transmission cost per bit. Also, delay-sensitive applications such as voice and video have a more acceptable quality. This is because cell relay has no long delays bursts from one user monopolizing the circuit, as can occur with frame relay.

ATM has got some disadvantages one of them is the Loss of bytes if information is no large enough to fit one cell. Is it possible to find a way to multiplex information from other application if cell is still not completely used.



-A- ISDN (Frame Relay)



-----  
Data Path
-----  
Signaling Path
=====  
Signaling and Data Path

-B- B-ISDN (ATM)

**Figure 1.10: ISDN frame relay protocol architecture Vs B-ISDN ATM protocol architecture**

### 1.3.

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## **SIGNALING ENVIRONMENT**

Signaling consist of what operation is to execute (opcode within the computers).

It conveys to the switches the commands and data to process and monitor calls much the same way as the opcode conveys to the control circuits of the CPU what to do with the data in the data field of an instruction.

The signaling network makes the nodes within the network to communicate with each other executing the function necessary for call processing:

- capture an available channel,
- establish voice path, hold during call progress,
- release channel after call completion, and
- reallocate channel for future use.

Control signaling needs to be considered in two context: 1) signaling between a subscriber and the network and 2)signaling within network. Typically, signaling operates differently within these two contexts.

The signaling between a telephone or other subscriber device and the switching office to which it attaches it, to a large extent, determined by the characteristics of the subscriber device and the needs of the human user. Signals within the network are entirely computer-to-computer. The internal signaling is considered not only with the management of subscriber calls but with the management of

the network itself. Thus, the internal signaling is more complex, and a set of parameters is needed.

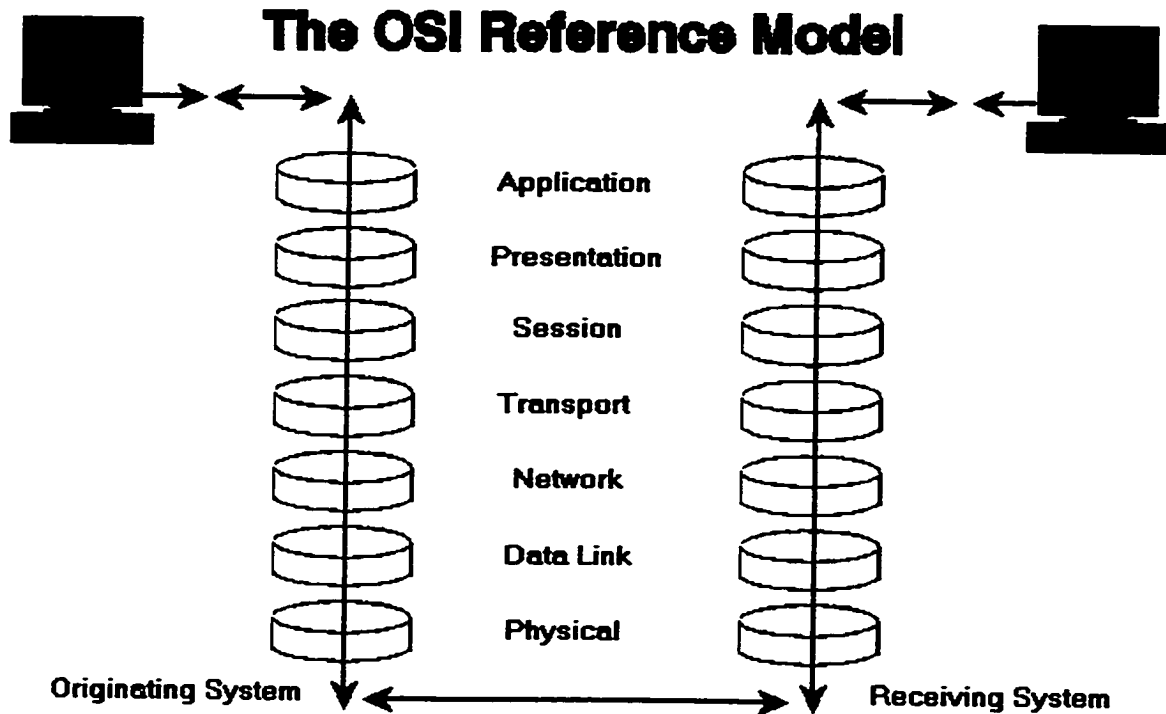
### **1.3.1.**

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## **OSI Model and Communication Systems**

The first step in implementing any communication network is to adhere to a common model for networking. Earlier proprietary networks such as IBM's SNA required equipment from a single vendor. While the network worked well, it did not allow the customer to use the latest technologies and equipment from other vendors. Nonetheless, IBM's SNA was used as the basis for modeling an open system, that is to say, a network formed by building blocks that adhere to standard protocols.

The foundation for open systems is the open systems interconnect (OSI) reference model. This model of a communication system, developed by the International Standards Organization (ISO), is most valuable as a way to characterize networks and as a goal for uniform network implementation. Today, all major networking players attempt to define their equipment using the OSI model. The model describes the seven steps, or layers, required for end-to-end communications.



**Figure 1.11: The OSI Reference Model**

The *Physical Layer* used to transmit bits between data stations over physical media. The physical layer may provide multiplexing and may be point-to-point or multipoint (e.g., EIA 232C, V.35, X.21). The *Data Link Layer* provides a relatively error-free path between adjacent stations. It delimits data in frames, provides error detection, and controls data link (establish/terminate) (e.g., ITU-T LAPB, LAPD, IEEE 802.2 for LAN). The *Network Layer* is used to transfer data host-to-host across subnetworks. It routes and relays packets, may provide

network connection and/or flow control (e.g., ITU-T X.25 PLP, ISDN Q.931, ISO CLNP). The *Transport Layer* is used to provide end-to-end data transfer. Some of its functions are establishment and release of transport connection, end-to-end error recovery and providing quality of service, and multiplexing transport connections onto network connections (ISO 8072 and 8073, ITU-T X.214 and X.224). The *Session Layer* is used to provide for organized and synchronized data transfer such as session connection to transport connection mapping, session connection and release, manage data exchange and enforce dialogue disciplines, synchronization points, and exception reporting (ISO 8326 and 8327, ITU-T X.215 X.225). The *Presentation Layer* provides for representation of data (independence of local encoding) such as Syntax negotiation, code conversion and encryption, and data compression (ISO 8822 and 8823, ITU-T X.216 and X.226). The *Applications Layer* used to provide common application services such as establishment of association between applications, naming and addressing, electronic mail, and file transfer (ISO 8571, ITU-T X.217 and X.227, X.400).

Each layer of the OSI reference model communicates with the layer directly above and below it, and has its own communications mechanism. This structure allows error checking and recovery at the lowest possible layer, thereby improving throughput. Most communications systems do not use all of the layers; in fact, the network portion of end-to-end communications is found only in layers four and below. Higher layers such as application, presentation, and session are normally associated with the sending or receiving computer. Also

associated with the layers is equipment that allows networks to be interconnected – gateways, routers, bridges, and repeaters. Throughout this discussion, the OSI reference model will be used to describe networks and the equipment that forms them.

### **1.3.2.**

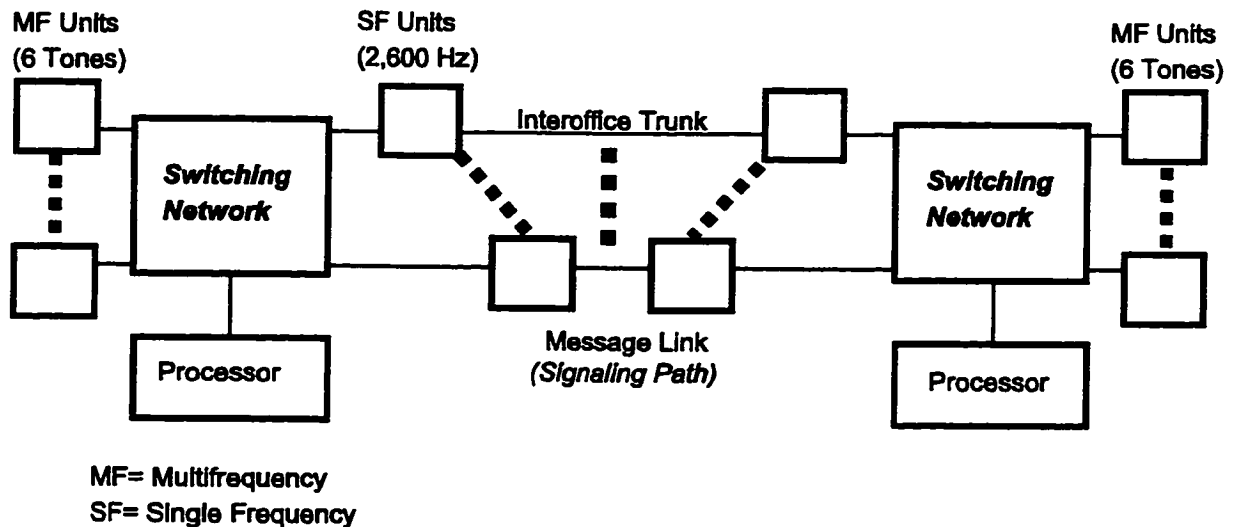
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## **Signaling in a conventional communication environment**

There are three types of signaling functions in telephony: supervisory, addressing, and call progress. These functions can be categorized as station or subscriber-line signaling and interoffice signaling. Transmission of these signals could be analog ac, analog dc, or digital. Further subdivision of signaling can be in-channel or common channel. In-channel signaling can be divided as in-band or out-of-band. In-band methods transmit signaling information in the same band of frequencies used by the voice signals (e.g., single frequency, multiple frequency). Out-of-band signals use the same facilities as the voice signal but a different part of the frequency band (e.g., direct current on customer loops to recognize on- or off-hook conditions). Current usage sometimes refers to common channel signaling as being out-of-band.

Supervisory signaling involves the recognition of busy or idle states on subscriber lines and interoffice trunks, and then transmitting that information to the caller and switching system. Single-frequency (SF) tone signaling (2.600 Hz) is used in conventional in-band signaling to indicate the busy/idle state of trunk

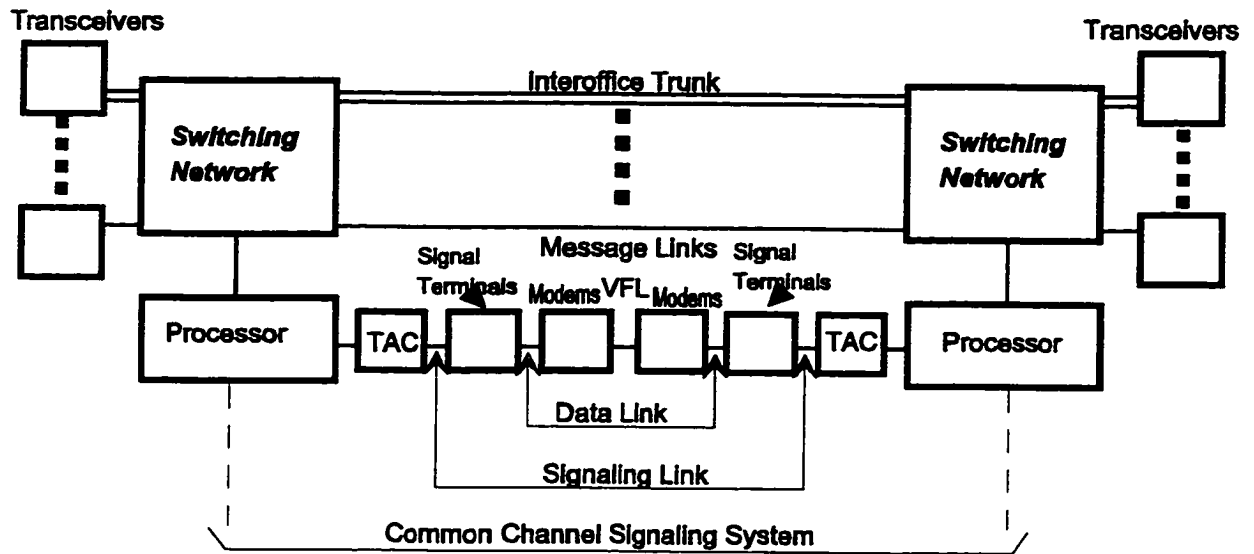
circuits between switches. Address signaling involves the transmission of digits of a called telephone number to a switching system or by one switching system to another. Dial pulse and dual-tone multifrequency (DTMF) signaling are used on subscriber lines while multifrequency (MF) signaling was the primary in-band system for passing address information on interoffice trunks (Figure 1.12)



**Figure 1.12: Single Frequency/Multi-Frequency (SF/MF) in-band Signaling**

Prior to 1976, most signaling between switching systems was performed by the MF/SF method. Call Progress signals are transmitted to a caller to provide information to callers and operators relative to the establishment of a connect through a telephone network. Call progress signals are a variety of audible tone signals that indicate dialing, line busy, and ringing on the circuits.

CCS is a method that is employed in the public telephone networks to exchange signaling data between processor equipped switching systems over a network of signaling links. The links are dedicated to control of signaling functions that are common to a number of channels.



TAC= Terminal Access Circuit  
VFL= Voice frequency link

**Figure 1.13: In-band and Common channel signaling. Basic functional diagram of the CCIS System. It represents the typical configuration of an STP for signal transfer between switching systems.**

When AT&T introduced CCS in 1976 it was known as Common Channel Interoffice Signaling. The message format was similar to CCITT Signaling System No. 6 with the signaling information transmitted across 2.4 kbps analog data links. The use of CCIS improved trunk signaling between 4ESS toll switches (introduced at the same time) operation under stored program control (SPC).

The introduction of CCIS increased the speed and lowered the cost of call setup and disconnect on the intertoll network.

CCS has three modes of operation: associated, quasi-associated, and disassociated.

### **ASSOCIATED MODE**

In associated mode, the CCS link closely tracks along the entire length, the interswitch trunk groups that are served between end points.

### **QUASI-ASSOCIATED MODE**

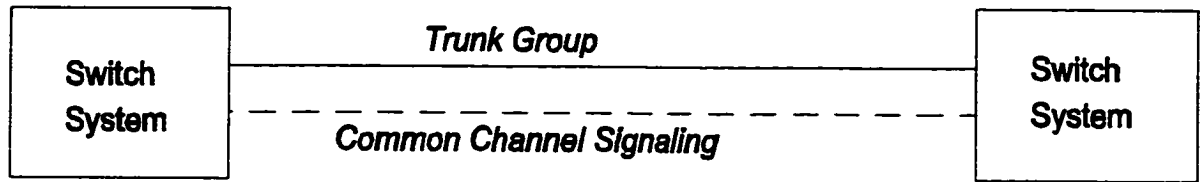
In the quasi-associated mode, the CCS links may not be closely associated with the trunk groups that are served.

### **DISASSOCIATED MODE**

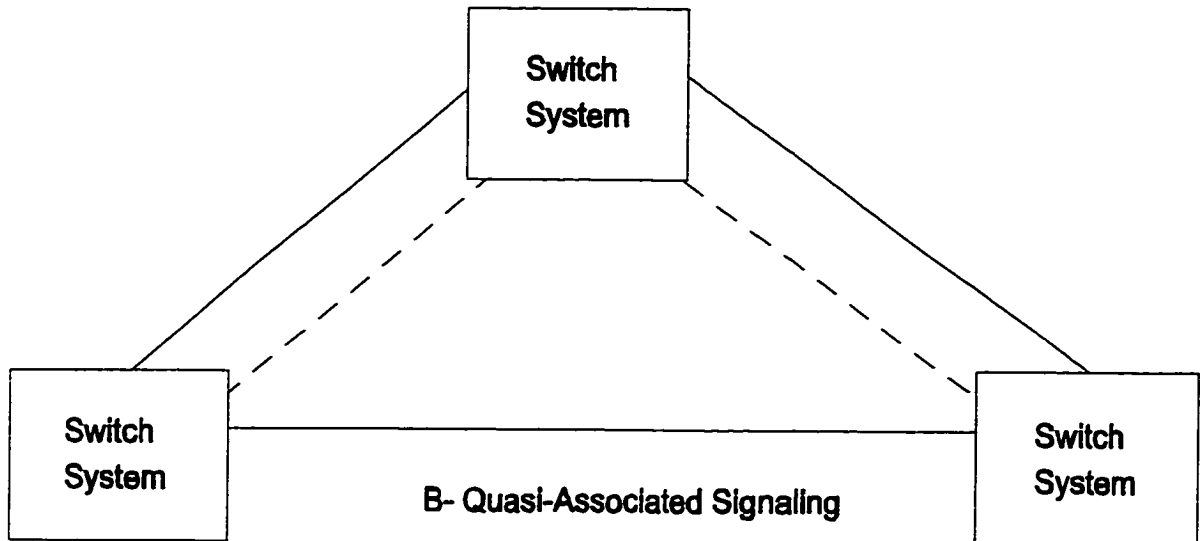
In the disassociated mode, there is no close or simple association between the CCS links and the trunk group being served. The disassociated mode permits nodes to communicate via signaling links when there are no functioning connecting trunks.

The implementation of common channel interoffice signaling (CCIS) resulted in the Open System Interconnection (OSI) seven -layered reference model (Table 1.1) with the higher layers geared toward applications and the lower layers geared toward network and transport of data.

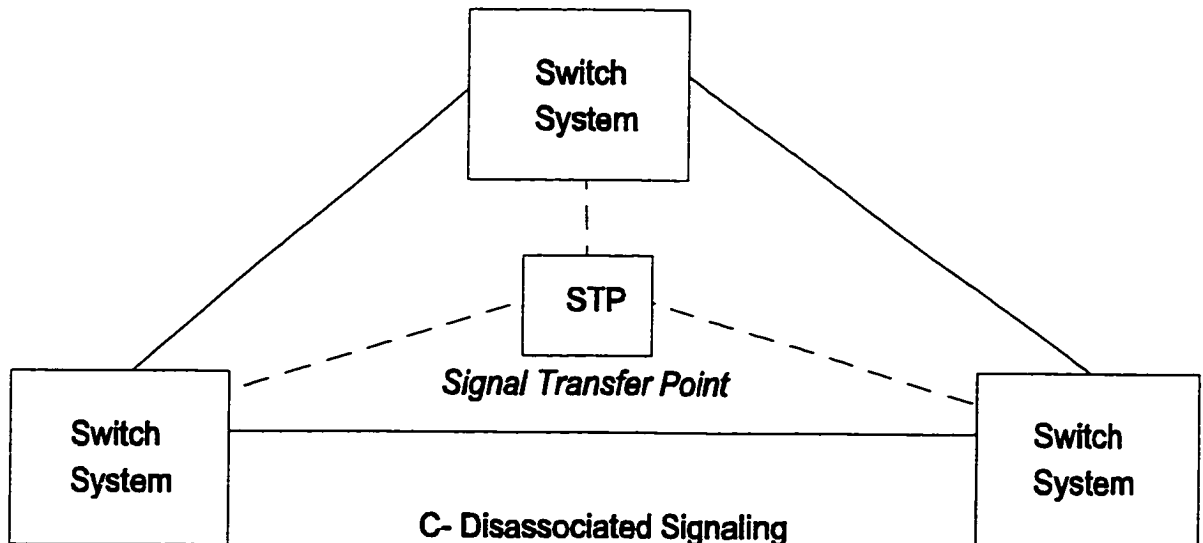
In a sense, The Signaling System 7 (SS7) has an evolutionary architecture of its own. The exchange of signaling information also follows of the OSI model.



A- Associated Signaling



B- Quasi-Associated Signaling



C- Disassociated Signaling

**Figure 1.14: CCS modes of operation**

<b>Application</b>	AL: Facility to serve the end user. Provision of the distributed information service. Communication management between the AL and PRL. Service to the user, the application (SASE), and the group of applications (CASE) served. Authentication of user IDs, destination IDs, authority to exchange information. Determination of service quality from the lower layers. Data integrity, error recovery, and file transfers are also assured. (ISO DIS 8649, protocol ISO 8640, CCITT X.400 message handling)
<b>Presentation</b>	PRL: Assures the delivery of information to the end users in a form that is usable and understood. The information content (semantic) can be altered to suit the source(s) and destination(s) of the information. (ISO 8824 for abstract syntax notation ASN.1, ISO 8825 for encoding, CCITT X.409)
<b>Session</b>	SL: Provision of transfer data to transfer control in an organized and synchronized manner. User may define the degree of control and synchronization that the session layer will provide. (CCITT X.25, ISO 8326, X225, ISO X.8327, T.62)
<b>Transport</b>	TL: Selection of network service. Evaluation of the need for multiplexing. Selection of the function from the lower layers. Optimal data size decision. Mapping the transport addresses to network addresses or the end-point users and negotiated. Data flow regulation between the end recovery. (ISO 8072, CCITT X.214, ISO 8073, ISO DIS 8602-connectionless, CCITT X.224)
<b>Network</b>	NL: Establish, maintain, and terminate switched connections. Addressing and routing functions. Service TL, independent of DL and PHL, (a) Connection: network connection data transfer, optional expedited data and receipt transfers, reset and connection release, (b) connectionless: UNITDATA, (ISO 8378, CCITT X.213; Protocols CCITT X.25, 1984, packet, ISO DP 8878 with X.25 for connection oriented network service: ISO 8473 for connectionless internetworking; CCITT Q930/Q931 ISDN)
<b>Data Link</b>	DL: Synchronization/framing, error detection and recovery, and flow control for information transmitted over the physical link. (ISO DIS 8886, 1745, 2111, 2628, 2629: CCITT X.212, X.21 for basic mode: ISO 3309, 4335, 6159, 6256; CCITT X.25, X.75, X.71 for HDLC; CCITT Q.920/Q921 for ISDN.)
<b>Physical</b>	PL: Activation, maintenance, and deactivation of the physical connection. The electrical and mechanical characteristics for physical interface and the transmission media. (ISO 2110, CCITT V.24, V.28 or EIA RS-232-C, also EIA RS-449, CCITT X.21, V.35; EIA RS-422, CCITT V.11, X.27 for balanced voltage; EIA RS-423, CCITT V.10, X.26 for unbalanced voltage; CCITT I430 for (2B+D) ISDN and CCITT I.431 for 24 B or 30 B channels.)

**Table 1.1: Open System Interconnection (OSI) seven -layered reference model.**

## 1.4.

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### **BROADBAND ENVIRONMENT**

If computing needs were to remain static, the narrowband WAN infrastructure and, consequently, the disparity between LANs and WANs, would probably continue for years to come. But change is inevitable with the emergence of bandwidth-hungry applications in everyday business activity. Consider the growing popularity of the intelligent building. In 1995, in Europe alone, investment in intelligent buildings was \$15 billion because of the ability to save up to 30% of building operating costs. This environment optimally integrates people, property, and technology by managing a variety of computer services on a local network. These islands of computer power then connect into the worldwide public telephone network. For the user, the manner in which information is accessed and the burden it places on the network remains transparent.

#### 1.4.1.

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### **Explosion in Graphic information Systems and Multimedia**

With the proliferation of applications such as the intelligent building, there is a parallel increase in bandwidth demands upon the public network. Regardless of the size of the communication links, more connections will increasingly tax network capacity. Even LANs, with their relatively large megabit buses, are not immune to bandwidth overload. Ethernet, token ring and FDDI all share

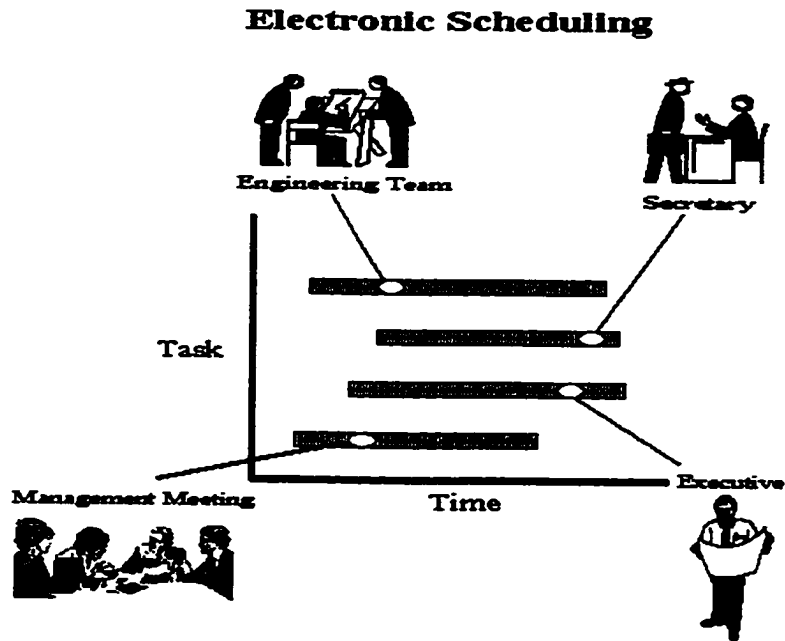
media—a situation that is always potentially prone to creating bottlenecks. A shared LAN can bog down under the burden of large numbers of users and of information-intensive transfers. With so much information being shipped over a single route, the ability to manage and dynamically allocate it on the fly is critical. Individual users can be better served by having exclusive use of the bandwidth, rather than by sharing it, and that is the basis for some of the new switching technologies and the driving force for new broadband transport technologies that can manage huge amounts of bandwidth.

#### **1.4.1.1. WORKGROUP SOFTWARE**

Workgroup software, sometimes referred to as cooperative software or groupware, allows a single action to generate a host of other actions automatically. It takes advantage of new, more powerful operating systems to make the physical topology of the network transparent to the user. Users no longer need a sense of where resources are on the network, or how they communicate. With this structure, all applications become workgroup applications, overcoming the constraints of time and distance.

Workgroup software boosts productivity by letting an organization leverage highly skilled people, relaying advice and analysis from specialists to the group. For example, Lotus Notes is based on an object-storage mechanism that uses replication technology to update remote copies of E-mail databases housed on a Notes server. It is also used for forms and images. Everyone on the network has an electronic bulletin board over which they can automatically send and receive

information in whatever form they desire, to and from as many people as they want. Coworkers can share information.

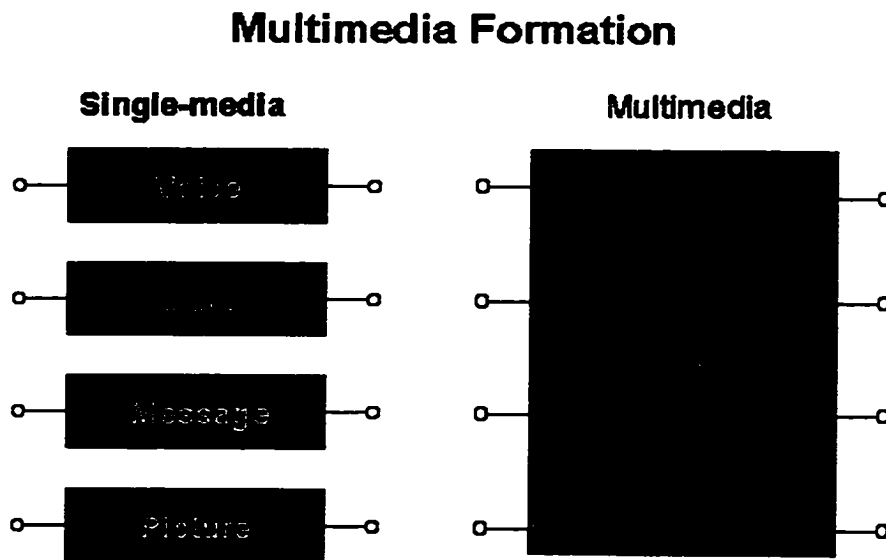


**Figure 1.15: Electronic Scheduling**

Marketing and sales groups can share a common database that tracks and monitors all contacts with a single customer. Accountants can share financial and tax information. Secretaries can exchange letters. Engineers can exchange design files. A single voucher on a computer can be simultaneously forwarded to a supervisor for approval and to a secretary for filing. When signed by the supervisor, it can automatically be sent to accounting where a voucher can be generated. In another instance, a team project can be coordinated by an electronic schedule that automatically updates each member. This results in

better and faster decision-making and an increase in the total volume of routine business transactions that traverse the network.

There are some caveats, however. While increasing productivity, such software does consume large amounts of bandwidth between end-stations because bandwidth-intensive multimedia applications are generated without user awareness.



**Figure 1.16: Multimedia Formation**

Additionally, workgroup programs can improve a company's productivity, some of its benefits can also raise concerns .

### 1.4.1.2. MULTIMEDIA AND HYPERTEXT

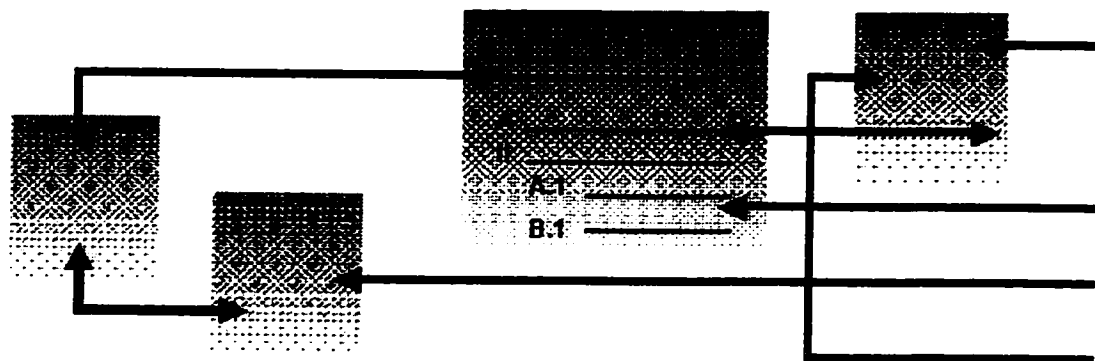
In the future, people will interact with computers as easily as they do with each other. Information will flow freely, formatted for the rapid assimilation of ideas.

#### Non-linear Information

Paper documents are linear structures



The content is often non-linear

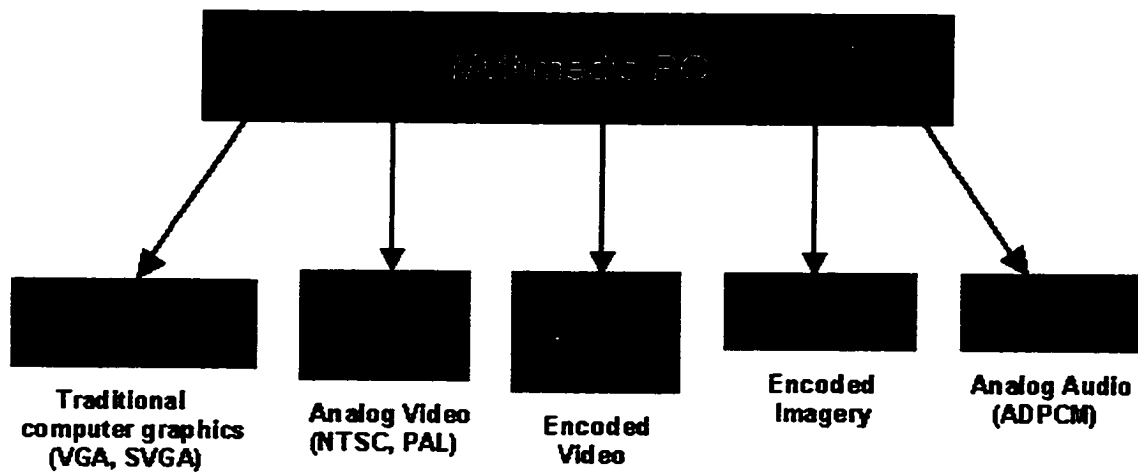


**Figure 1.17: Non-linear Information.**

The interface will be composed of today's voice, image and text all melded into multimedia presentations. The decision to employ multimedia is no different from other business decisions. Any new technology must address issues of purpose, cost, and resources. The real value becomes apparent when the technology fulfills a purpose in ways that are clearly superior to other means.

Used appropriately, multimedia can introduce new avenues for workgroup communication and collaboration (e.g., sound can greatly benefit sight impaired users).

## Multimedia Standards



**Figure 1.18: Multimedia Standards**

Multimedia is based upon the same digital technology as computing. The information that resides in an analog format – video, audio and text – is translated into ones and zeros, a language that is easier to send, store and manipulate for computers. Multimedia relies upon the electronic integration of different media to enhance the ability of users to interact with the information in

a format that suits their particular needs, via the use of specialized peripherals such as CD-ROM or videodisk players, or VCRs. Multimedia standards are still being worked out, and many of today's microcomputers will not support the ones that are finally established.

Network managers want to implement applications that support high-quality, full-motion video that require broadband networks. Even with compression, VHS-quality video takes 1.2 Mb/s of bandwidth, and advanced digital video can use 30 to 130 Mb/s. LANs like Ethernet, which were designed to carry data, have problems transporting full-motion video, which requires the delivery of packets in a particular order with small, consistent delays. Packet video on LANs needs faster, isochronous protocols to deliver the continuous data streams of voice, video, and data with little delay. Such isochronous protocols establish a virtual channel that behaves like a dedicated point-to-point circuit between devices. Among the technologies competing for multimedia traffic are ATM, fast Ethernet and FDDI-II.

Hypertext is a term for a retrieval system in which individual words are indexed and threads created to have one word lead to another. By merging text and graphics and not requiring specialized computers, hypertext constitutes a logical migration point to multimedia. This level of integration would be highly useful in developing computerized training modules, for example. Hypertext is especially useful in conjunction with massive databases because it takes information retrieval to a new level by making it simpler to use and find. To

accomplish this, many database vendors rely exclusively on Boolean searches to create indexes automatically that list the occurrence of every designated word or related sets of words in the database. While this capability is valuable, by itself it is not true hypertext. To create true hypertext, the contents of the database must be coded to link one subject with another.

Just as it took the installation of millions of LANs to create sufficient LAN traffic to impact the WAN, multimedia applications are now gathering the same kind of critical mass. Hastening this process is the introduction of multimedia files that increase by 4 to 10 times in size each year due to the amount of charts, pictures and even voice that are being incorporated into the text. Today, most of the personal computers and workstations sold in the United States will have multimedia capability. Coupled with hypertext, use of that capability will increase network traffic. Every time such traffic enters the network, the demand for bandwidth increase -- as does the average network traffic load. To date, the limited amount of bandwidth available in the public telephone network has tended to keep multimedia applications on the premise. Yet, the goal of multimedia remains to allow networked computers to handle sound, image, and video as readily as they handle numbers and text. Consequently, a major factor for success will be the availability of sufficient bandwidth to manage large numbers of multimedia applications. Although the full impact of these applications is still several years away, the network infrastructure must be in place well in advance.

### **1.4.1.3. VIDEOCONFERENCING**

Businesses spend hundreds of billions of dollars on travel. Videoconferencing could substantially reduce business travel costs by transporting conversations and images instead of people. Video signals can be transmitted over private links, dedicated T1 and T3 links, and over the public telephone network. The difference is the degree to which the transmission can approximate full-motion video. One way to transmit video at lower bandwidths is with signal compression such as that specified by the motion pictures expert group (MPEG). Some compressed video uses image prediction algorithms that have trouble tracking rapid scene changes. Rather than the stop and start of slow-scan video, consumers favor transmissions that resemble what they can see on a television screen. With a broadband network-based videoconferencing system, broadband switches can be employed that use multiple fixed cameras. Viewers can select which person they want on screen and the system will select the best angle and the right camera for the picture. The increased consumer acceptance of 'motion' video transported over broadband networks could jump-start the videoconferencing market, and outweigh the increased cost for the necessarily larger amounts of bandwidth.

### **1.4.1.4. IMAGE PROCESSING**

Researchers estimate that 95% of the world's information resides on paper or microfilm. The task of moving it from these dated technologies to electronic form is accomplished by image processing systems. Image processing converts existing documents of all kinds to digital representations which can be quickly

viewed on a computer monitor, printed, and distributed in a variety of ways, either as maps, fingerprints, CAT scans, drawings, or anything else that can be represented in two dimensions. The components of a basic image processing system include scanners resembling fax machines that make a digital record of every small sector (called dots) of an existing document, a controlling computer (microcomputer, minicomputer or mainframe), a monitor, and storage. In most networks, the efficient use of storage media such as optical disks and jukeboxes is pivotal to a successful imaging system. The high gigabyte content of most CD-ROM platters is essential to archiving libraries of documents because a single, high-resolution image can consume tens of megabytes of storage.

Imaging is often viewed simply as a storage and retrieval vehicle, but document imaging systems can mix data, photographs, and full-motion video and can completely transform the way an organization operates. Evidence of this is the federal government where image processing has been used widely since the mid-1980s, replacing microfilm and microfiche, which employ photographic technology. Since the management of financial records and contracts is the top application for federal agencies, imaging technology has reduced the storage space needed for documents by a factor of twelve and allowed quicker retrieval. For the IRS alone, the potential exists to save \$41 million a year in the storage and retrieval of paper documents.

Given today's technology, it remains a sizable challenge to shuttle hundreds of megabytes of images across the enterprise network. The digital storage

requirements of a single full-color image can exceed 16 megabytes, providing the detail for the rich variations of color and texture in fine artwork. Experts in the US could study great works of art in Russia without leaving their terminals. The transfer of such large files over a public or private network requires bandwidths that provide a tolerable amount of delay or transmission time – that is, the hours or seconds required to complete the file transfer. Higher bandwidths cost more, but allow information to be transported with less delay. In the case of specialists such as art assessors or medical radiologists, their time alone may be worth the increased network costs. For other businesses, the increased cost of bandwidth is justified by considerable overall savings in other operations. For worldwide catalog sales, instead of producing costly monthly paper catalog updates in several languages, central image storehouses can be accessed by users in different countries across high-speed, real-time links.

One of the world's largest pizza restaurant chains has managed to cut response time to suppliers from five days to ten minutes, while receiving a 16% after-tax return on investment. A network of imaging equipment tracks the company's fixed assets and handles the more than 10,000 faxed invoices it receives from worldwide suppliers each month. As invoices come into the accounts payable department, they are scanned into the imaging system. Over forty workstations, connected by fiber optic cabling to servers interconnected by means of an Ethernet LAN, access two mainframes. On the mainframe computers reside

databases that store the locations of restaurants and can identify more than 90,000 vendors.

#### **1.4.1.5. OPTICAL COMPUTERS**

Introduced in the 1950's, silicon integrated circuits have transported much of the world from the Industrial Revolution into the Information Age. Many believe that the next great leap will come from the marriage of light and electricity in the optical computer, a technology already under investigation. A prototype developed at the University of Colorado consists of lasers, electronic switches and optical fibers arranged in layers and tightly packed into an area about the size of a desk. As in electronic computers, information is represented by binary ones and zeros. The optical computer uses the presence or absence of light pulses to represent the binary pattern. The prototype optical computer operates at a clock rate of 50 Mb/s, controlling a 16-bit microprocessor. Data-encoded light pulses are stored in some 3 miles of spooled fiber cable. Each bit of information is carried in a 12-foot long light pulse, which traverses the memory spool every 20 millionth of a second. The pulses are synchronized by the unvarying speed of light. From such prototypes a number of information age applications may emerge.

#### **1.4.2.6. MOBILE COMPUTING**

The marriage of the communications, computer, and consumer electronics industries is rapidly creating products that allow people to transfer data without wires. Wireless communications are made possible by the emerging electronic superhighway, and by the services, products, and information that will travel

along it. The growth in wireless data transport is driven by the proliferation of portable computers, LANs, client/server computing, spread spectrum and cellular telephony, and by the falling cost of transporting data via broadband networks.

Wireless data communication is freeing workers from traditional constraints. Mobile applications such as the officeless desktop, one- and two-way messaging, long-distance file uploads, and the ubiquitous E-mail allow them to reach anyone no matter how far away. Consider the benefits of wireless E-mail: it can always find individuals; business deals can be negotiated while in transit; people can stay in touch with the home office and with clients by messages that can be initiated and responded to whenever and wherever needed. Wireless E-mail may be the most significant tool for the professional on the move, replacing pagers and cellular telephones.

#### **1.4.1.7. HOME VIDEO**

Although local cable companies offer limited versions of home video, only the more comprehensive switched telephone network can offer universal video access. Despite the desirability of this service, fiber optic connections to the home are not being made as fast as many experts originally predicted because of the traditional time for planning and study incurred by local exchange carriers (LECs). The stakes are high. There are potentially 140 million access lines in the US alone, but they could cost \$1,000 each to convert.

However, this market is expected to start expanding rapidly. Government regulators have allowed the LECs to enter the home video market and, in return, the local loop has been opened to competition from alternate access providers and cable companies. A steady stream of projects is under way with the goal of merging PC and TV technologies to create a super information appliance. Fiber optic vendors, cable TV operators and telephone companies are teaming up. All want to be among the first to install fiber optic cable to carry a variety of services to the home market. Digital TV broadcasts will enable intelligent video dial tone systems. Such systems go beyond today's pay-per-view that requires a user to call the cable company to sign up for a movie that the cable company has scheduled for broadcast. With video dial tone a viewer uses a remote control to program the TV to capture and download a movie from a menu of movies. When the movie ends, the viewer can download other information such as from a shopping network or stock market quotations source.

The race is on to build an electronic superhighway capable of delivering hundreds of new TV channels, as well as data and phone services. Once the loop is opened, and low-capacity systems are upgraded to support broadband transmission, the path will be cleared for electronic encyclopedias, shopping catalogs, travel, banking services and more, all brought to the home on a real-time basis. Full-range audio and high-resolution video-on-demand will make home entertainment an entirely new experience. At the same time, video to the home will also open the way to truly effective telecommuting, favorably

impacting the performances of the more than 30 million home-based businesses in the United States.

### **1.4.2.**

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## **Bandwidth Intensive networks requiring special attention.**

Keeping in good health, having the right education, and being always in touch with the family are basically any human's dreams. The creation of a global computer is the dream of any scientist, businessman, etc. Imagine if a molecular biologist who works on a cancer subject has the ability of getting worldwide statistics related with the subject in one organized file with all necessary information (such information is a mixture of data provided from a different hospital servers, pharmaceutical companies, current and previous research on the subject, etc.). Of course this will activate the process of maybe finding a cure for cancer. Medical students in NY, while in the classroom, can assist to an ongoing surgery in hospital in CA and of course before all can get copy the patient file before the surgery. The file should include X-rays, CadScan, blood work, hand written report, and other necessary information. The medical network being connected to the education network, students should be able to access to information about previous cases, for review before the actual surgery. There has to be some kind of knowledge that connects different networks, so we can take better advantage to the communication networks.

The following are networks requiring special attention. They all require a high bandwidth, and intelligence. The Intelligent ATM Network will offer to this of networks and others, the ability to access any kind of information in an intelligent way.

## **MEDICAL NETWORK**

Providing good medical service is the physician and the nurse responsibility. To provide better service to the patient, communication between hospital workers with each other and with data centers in different areas around the globe should be customized.

Medical networks have been improving since the introduction of computing and networking. Many databases are available to keep track of patients' medical records, medical providers, consultants, administration, medical lab, accounting, etc. But still, until today there is no real exchange of information between different hospitals (and even sometimes within the same hospital). The patient history for example is embedded in the hospital's server he goes to, and even sometimes on different servers in different hospitals (small pieces). At the current stage some data is even available at international users as well domestic users. But the type of data and service is still limited. The idea of making the medical network intelligent have been proposed. The published architectures of intelligent medical networks are mostly based on the IN functions, where the use of CCS7 is a necessity, and the services provided are of low bandwidth. The knowledge based medical network will be the ideal network, but still a lot of

work to be done to get to that point. The first step is the Intelligent ATM Network, and all digital network, providing a high bandwidth and intelligent services. In this kind of networks, high bandwidth requiring information can be also stored in different servers and commuted within the network when needed. The transmission will be based on fiber using SONET frames and ATM switching [3, 4, 5, 6, 7].

Once the ATM is implemented, and the intelligent services are provided through ATM network, with a good strategy, medical networks will enter a new era. Then information will be exchanged between different hospitals in a secure and even economical way. Insurance companies will save dollars by connecting their servers to the medical network and pharmaceutical network through the Intelligent ATM Network. A patient information, can be accessed at anytime, anywhere but only by authorized people if a finger print, being the most secure way, is to be done to access the patients information.

Once the medical network is all intelligent, it will serve some other domains such as insurance fraud elimination who may use finger prints for each customer, and cutting down crimes by giving police departments of FBI access to all DNA databases. The idea doesn't then touch only the medical network but also some other domains who's network will be connected to the medical network through the intelligent ATM network.

### **1.4.2.2. EDUCATIONAL NETWORK**

Before college, in most countries, student will spend more time at school than with his parents getting educated. During college, after college, people spend more time searching for the information than using it.

Educational institutions have spend al lot of money on providing classrooms with microcomputers, and even connecting them to the Internet (a time consuming network) through which we get redundant information, sometimes valuable.

When modern computing systems, communication networks, and mass media are to be taken advantage in all educational facilities and disciplines, then a certain attribute of the educational environment will become strongly influenced by the technological revolution.

If a global file server is to be installed and connected to different file servers in different schools and universities using an intelligent network that provide Broadband services, then students as well as educators will be able to exchange information between each other and with the network. If a provided with a logical connection with other servers in different other domains, then the throughput will be higher. Each local and global network and information we get should not be redundant which will not just save the time bandwidth, but also provide us with better service and more possibilities to educate our children. To minimize the crimes toward the children, the use or access to each

server should be authorized either by the network local network manager to the specific user or by a different servers to the local network.

Intelligent ATM based educational network will offer the possibility of assisting to conference while in the classroom, or to a lecture while home. Some more other services are to be defined.

#### **1.4.2.10. KNOWLEDGE PROCESSING SYSTEMS**

Year 2000 is almost here, and some workers are still searching on the Internet to get information about the Y2K plan trying to find the right information to solve the problem and sometimes just to find companies that will help solving the problem. Surfing the Internet is a hustle, and takes a lot of time especially when you don't have the time to waist. The company will still pay the employee a salary. Traveling to another country to see a product demonstration is a waist of money especially if you are not sure you'll buy it.

Knowledge machine is a in sense a global computer that processes object as a query. It's architecture is based on the electronic switching system architecture. Four building blocks are defined: 1) communication module to control and guide the communication germane to the various processors, 2) switching modules to select and switch large quantities of information from knowledge bases. 3) administrative module to take care of housekeeping functions. 4) knowledge module(s) [3, 4, 5, 6, 7].

**1.5.**

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**CONCLUSION**

There is no single reason why the demand for services that only broadband networks can provide has accelerated. Certainly a global competitive environment where businesses must be well informed and nimble to survive has fueled the flame. The growth of personal computers and sophisticated software applications that make businesses more efficient and effective is another contributor because these pockets of information must be interconnected. Cooperative software has gone beyond development systems and E-mail, allowing users to share knowledge in the form of images and video as well as text. Workgroups and entire companies automatically spawn messages into the network that track customers, create conferences for collaborative problem solving, and access a library of policies, documentation or news. Never before has the networking of computers or the ability to message anywhere and anytime been so important to the success or failure of a business. And never before have the torrent of bandwidth demands deluged public and private networks, pressing for intelligent and concurrently managed information superhighways. This has created an explosive market for broadband networks, equipment and carrier services.

It's time to provide the end user, provider, and the network itself with a service to support Broadband services in intelligent manner. The Intelligent ATM

networks, SSP-STP-SCP capabilities, will enhance the network capabilities, and provide Broadband hungry applications in intelligent way. The question is How are we going to access the ATM networks? Is the current signaling will support Broadband ISDN? This, and some other will be discussed in chapter 6 (High speed networks).

## Chapter 2

### **ELECTRONIC SWITCHING SYSTEMS**

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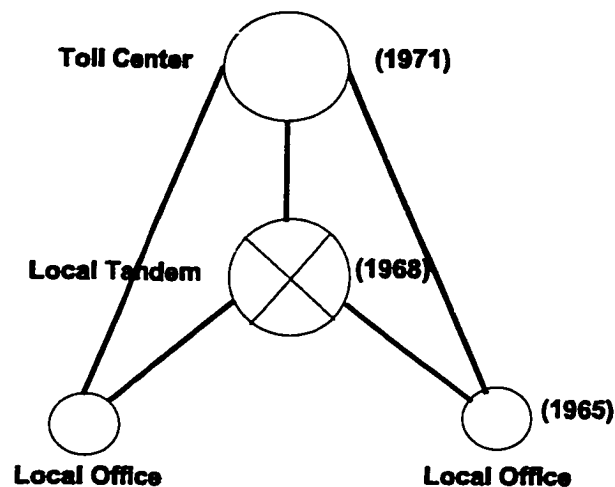
The switch represents an enormous capital investment for service providers. It is essential to the functioning of an intelligent and dispersed network. The same physical media has to perform a large variety of functions such as carrying voice, data, and video information from any other location bidirectionally. It must also be able to monitor the network response by program or by user interaction and seek and respond with the appropriate responses to system and user queries. Packets of information have to be switched within the network to the appropriate modules that can respond to the appropriate user or network commands. Hence, the architecture of a dispersed intelligent network is not complete without an intricate fabric of remotely controlled switching elements.

#### **2.1.**

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### **HISTORICAL OVERVIEW**

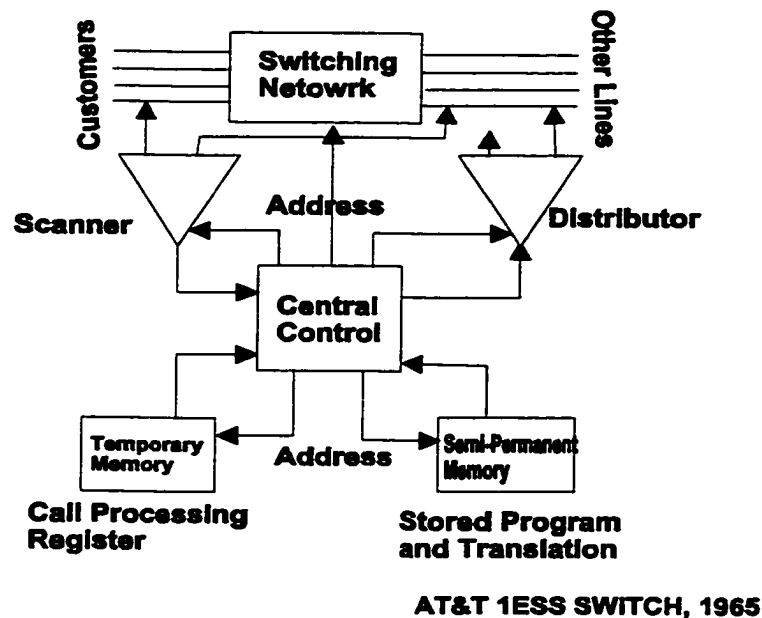
The cutover of the 1 ESS Switch in Succasunna, NJ in 1965. Introduced software controlled intelligence into bell-system's Public Switched Network (PSN). This was the first of the technological advances which laid the foundation for intelligent networks. The flexibility and compactness of software as opposed to hardware control meant that central office switches could offer new and more sophisticated features. Switches became islands of intelligence within the PSN. The 1 ESS switch was first used for local tandem purposes at Nashville, Tennessee in 1968 and its toll center use was in 1971 [56].



**Figure 2.1: 1ESS used for local tandem purposes**

Numerous subsystems (line-scanner, Signal distributor, and Central pulse distributor) are combined into basic hardware unit to interface with the line and trunk circuits and perform local functions. They are monitored by their own program control. The switching network interconnects different segments of the communication path depending on the need and availability. This switch was primarily electromechanical in nature. Because of software considerations, the 1

ESS switch-based Central Office was still designed specifically to handle the communications functions in real time with the dependability in about 2 hours downtime and 4 years life span for a typical central office.



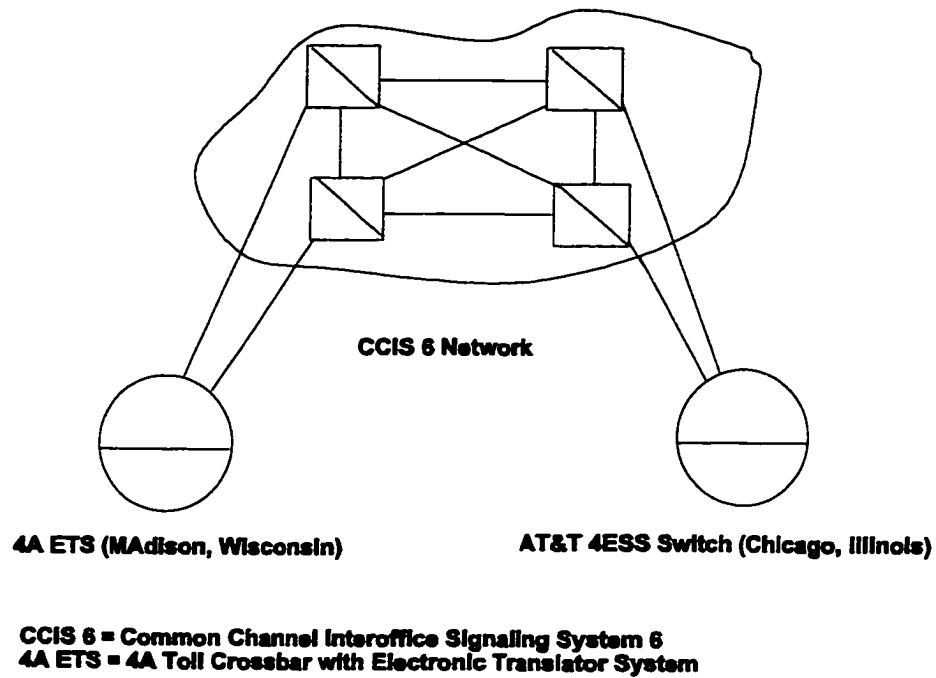
**Figure 2.2: 1ESS Architecture after introduction of Stored program Controlled**

Rapidly followed the introduction of the major 1 ESS architecture to the switching systems was the No. 1A ESS switch. Integrated circuits and faster clock rates offered greater call handling capacity. The processor evolution was the 1A processor, a high -capacity, custom designed processor that has proven itself in both 1A ESS and 4 ESS switch product line. The 1A ESS initiated the impact of computerized control onto local switching. Numerous Central Offices (No 2 ESS, No 2B ESS, and No 3 ESS) were introduced during the late 1960s and early 1970s, but the impact of the switches was not as dramatic since the systems addressed the growth and speed aspects of switching systems [55].

Stored Program Controlled (SPC) switching was introduced commercially with Common Channel Signaling was introduced into long-distance portion of the Bell System network in 1976. The signaling protocol used was Common Channel Interoffice Signaling (CCIS) based on CCITT (now ITU-T) Signaling System 6.

In 1976, the Bell System introduce common channel signaling capability (CCIS 6 based on CCITT Signaling System 6) into the long distance of its switched network. The first use of CCIS 6 was for purpose of inter-switch trunk set-up - between a 4A ETS Switch in Madison, Wisconsin and a 4 ESS Switch in Chicago, Illinois. CCIS 6 provided speed, security and flexibility advantages over the previous multi-frequency signaling protocol employed in the intertoll network. This was technological foundation for intelligent networking - allowing more extensive messaging among existing islands of intelligence.

On January 17, 1976, the first switch was placed into service, culminating the single largest switch development ever undertaken in the Bell System until that time. When it was introduced, the 4 ESS switch was the largest high-capacity digital switch in existence, capable of terminating 107,520 trunks and completing over 500,000 calls per hour. The hallmark reliability and capacity of the 4 ESS switch set the standard for digital switching. Since then, neither technology nor service needs have stood still. Although the classic digital switch architecture pioneered by the 4 ESS switch has passed the test of time, every major subsystems within the switch has gone through one or more technology upgrades to accept new features and reduce costs.



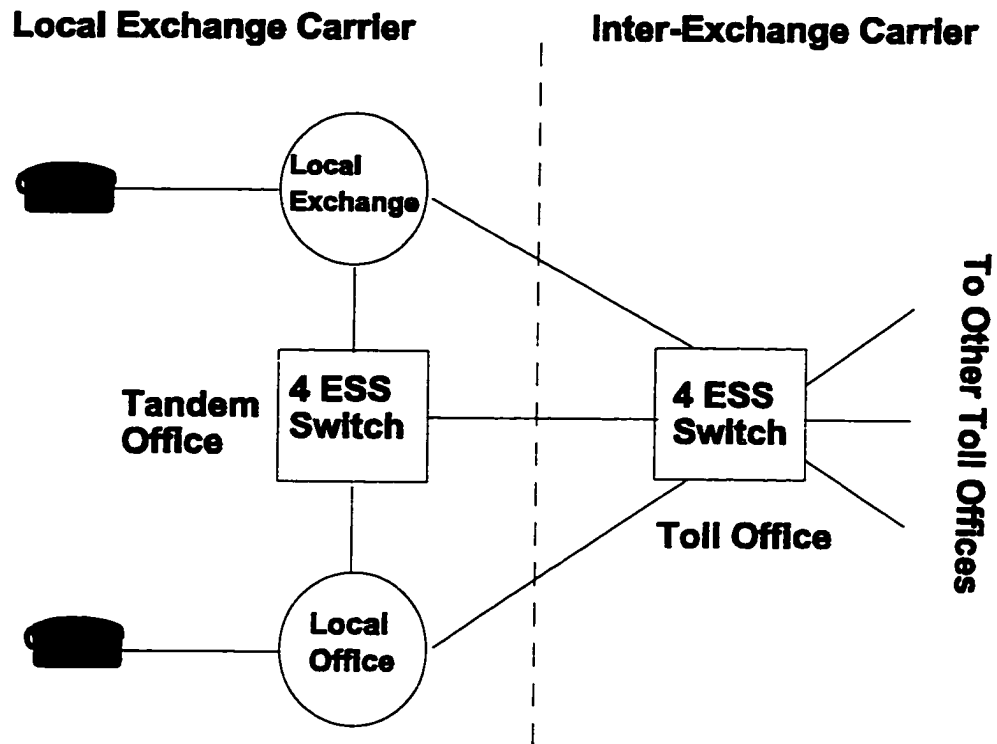
**Figure 2.3: Introduction of Common Channel Interoffice Signaling**

## **2.2.**

### **THE 4ESS SWITCH**

The 4ESS switch is a feature-rich, high capacity toll and tandem service node used by both interexchange and local exchange carriers. (figure 2.4).

The classical architecture of 4 ESS includes three main building blocks: The switching processor platform, the switching fabrics platform, and the customer interface platform [43].



**Figure 2.4: Typical Architecture using the 4 ESS**

### 2.2.1.

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## Switching Processor Platform

The switching processor platform requires three basic resources:

- Real-time functioning for call processing and billing data processing
- Memory for call-related data and service databases; and
- Input/output ports and bandwidth to support signaling links and OAM&P links to Operation Support Systems.

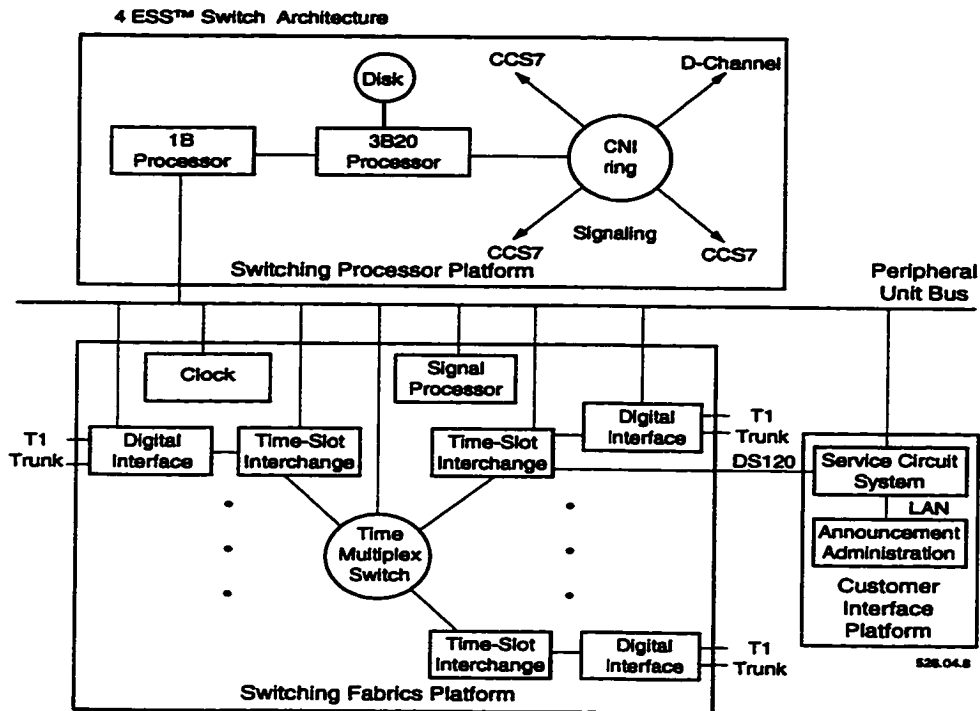
Total switch downtime for all causes cannot exceed three minutes per year per switch, so all processing resources must be extremely fault tolerant. In the early

years, these processing needs could not be met cost-effectively through standard commercial processing components.

As such, the starting point for the processor evolution was the 1A processor, a high-capacity, custom-designed uniprocessor that has proven itself in both the 4 ESS switch and the 1A ESS switch product lines. Although the 4 ESS with 1A processor has an advertised capacity of 570,000 completed calls per busy hour, its actual performance often exceeds its advertised capacity. The number of calls completed has exceeded 800,000 per hour with an attempt rate approaching 1,200,000 per hour in times of natural disasters.

The customers' increasing real time needs required a replacement for the 1A processor to 1B processor, placed in existing 4 ESS early in 1994, met needs by increasing throughput by 2.4 time more, and significantly increasing the memory spectrum. The switching processor platform with the 1B processor has demonstrated support for well over 1,000,000 busy-hour calls, even when a high percentage of the calls are complex.

The switching processor platform contains a custom uniprocessor central controller (the 1B processor) supported by a number of special-purpose adjunct processors (figure 2.6). The entire processing system is the central intelligence of the system providing overall call control, call signaling processing, overall operations, administration, maintenance and provisioning (OAM&P) in an overall system integrity.

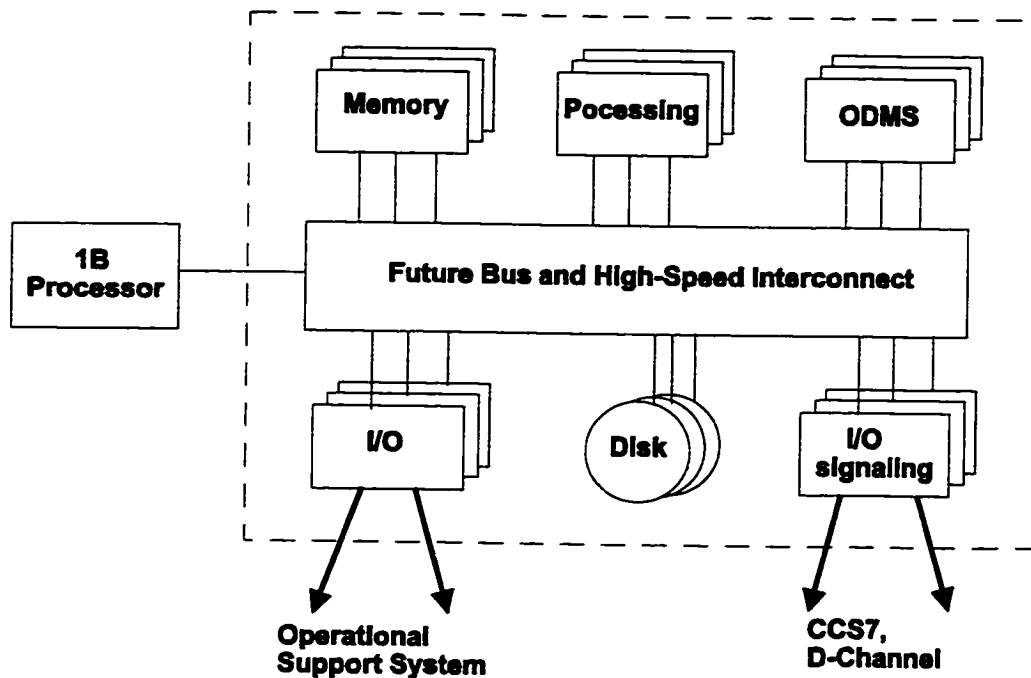


**Figure 2.5: Architecture of a Tandem/Toll Office Switch.(4 ESS); CNI: Common Network Interface, CCS7: Common Channel Signaling 7.**

Software routing in the 1B processor determine which type of service will be provided, and interface with both the customer interface platform and the switching fabrics platform. The 1B processor also handles overall maintenance for both the switching fabrics an the customer interface platform, as well as software audits and system integrity functionality [5, 43].

The *Common Network Interface (CNI) ring* is the interface for various common channel signaling links such as CCS7 an Q.931. Call related signaling received over these links is forwarded to the 1B central controller for processing. The 3B20

processor handles the CNI ring administration, call billing, formatting, disk backup, and general input/output functions for the switch.



**Figure 2.6: The Current architecture of the switching processor platform.**

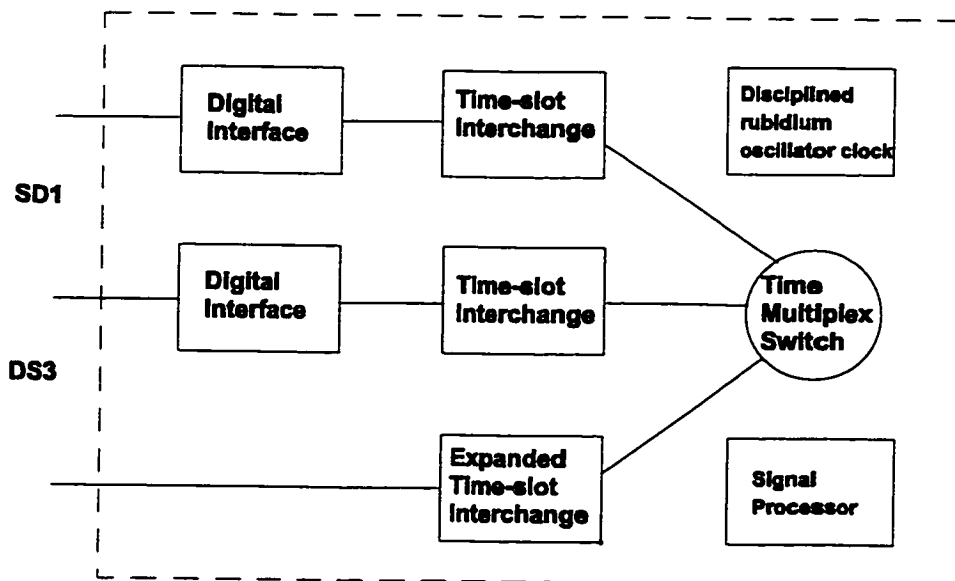
## **2.2.2.**

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### **Switching Fabric Platform**

The switching fabrics platform dominate the switch. Although there is only one central 1B processor, there can be as many as 129,000 transmission interfaces (trunks). The original 4 ESS switch interface was upgraded once in the 1980s. In the early 1990s, the development of a new expanded time-slot interchange offered an opportunity for other significant cost reduction.

The expanded time-slot interchange is a new fabric component that replaces both the digital interface and time-slot interchange, providing a new, high speed transmission interface and a time-division switch function for the fabric. The expanded time-slot interchange initially supports a DS3 interface for trunks. Later, it will support other synchronous optical network (SONET)-based transmission interfaces. To maintain the high reliability required for a switching fabric, the main control and the fabric of the expanded time-slot interchange are fully duplicated. The DS3 interface are protection switched against failures. A DS3 transmission format provides 672 DS0 channels. Each expanded time slot can support as many as six DS3 interfaces, comprising 4032 DS0 channels. As many as 32 expanded time-slot interchange can be configured within a 4 ESS switch, increasing the switch's termination capability from 107,520 to 129,024 channels.



**Figure 2.7: The current architecture of the switching fabrics platform**

The highly integrated design of the expanded time-slot interchange reduces costs, power, and floor space. In addition, the high-speed transmission interface and capability for integrated echo cancelers can eliminate significant capital investment in associated transmission.

The switching fabric interface (Figure 2.7) contains the space-division *time multiplex switch*, which interconnects many time slot interchange units. The primary digital trunk interface to the switch is through T1 (24-channel) trunks. *Digital interface* units demultiplex T1 trunks to digital signal level 0 (DS0) format and provide basic bit-oriented and in-band signaling functions. They also translate the 64 kbps DS0 information (voice or data) into the format required by the ore switch fabric. The switching fabrics platform includes a *network clock*, to keep various signal processor that enable in-band signaling and system maintenance and control.

### **2.2.3.**

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## **Customer Interface Platform**

During a call, the customer interface platform enables the switching processor platform to interact with end customers. It does this using capabilities, such as touch-tone reception and announcements, and newer capabilities, such as automatic speech recognition. The customer interface platform contains a high-capacity, fault tolerant, fully-integrated Service Circuit System, and an

Announcement Administration Processor that remotely records, verifies, and deletes announcements on the Service Circuit System.

### **2.3.**

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## **DISTRIBUTED SWITCHING**

A telecommunication network can be viewed as a collection of interconnected terminal equipment that provides end users with different forms of telecommunication services, ranging from voice and/or data to image and video.

A network comprises one or more network nodes connected by some transmission media. Each node, such as a wire center, contains one or more network elements, such as switching, operations, or transmission systems. The architecture of a network element is evaluated based on its control, fabric, and interface domains. The switching systems incorporate all three domains.

- A fabric domain includes the functions that provide the communication paths that users request. It also controls transport - the encoding, bandwidth, and similar characteristics of the information passing through the switch -and topology- the grouping of ports and users into sets that share and exchange information. Fabrics may be analog or digital, electronic or optical, narrow- or broadband; topology may be the circuit- or packet- switched, synchronous or asynchronous, single- or multistage, blocking or non-blocking, etc. New network initiatives - such as packet switching, data, and broadband- have added diversity to fabric architecture. However, adding new ATM fabrics to

existing switching systems to complement their narrowband fabrics will, initially, increase the degree of fabrics.

- The control domain includes the functions that interpret the switch user's requirements and issue commands to the fabric and interface domains to carry them out. Control functions include coordination of state. Fault recovery, status supporting, traffic measurement, billing, and similar administrative activities.
- The interface domain is the bridge between the outside of the switch and the inside. Outside the switch, not under its control, are a variety of written and the facto standards; inside a simpler, generally more proprietary protocol set. The interface domain provide translation between these environment, including voltage, line and interoffice signaling, craft and operations system interfaces, and similar functions.

## **2.4.**

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### **THE 5ESS SWITCH**

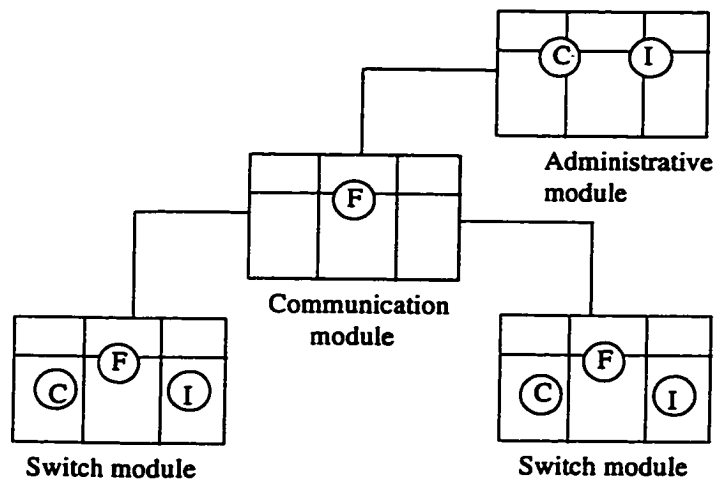
The introduction of the first single-module 5ESS switch in 1982, and then the introduction of the multimodule 5ESS switch in 1983, started another major trend in Central Office capabilities. In essence, the integrated architecture and time division digital switching that is inherent in this machine provides for most of the distributed processing and multifunction capabilities for the expanding telecommunication market.

## 2.4.1.

### 5ESS architecture

Figure 2.8 shows the general architecture of a 5ESS switch. It has a modular hardware and software that can be arranged to suit a customer's needs. 5ESS uses distributed processing to enable growth of capacity and capability. It has been designed to serve a range of applications and markets. In some instances the 5ESS could replace older switches (1ESS for example) or work in a coprocessing environment with another switch such as 1A ESS.

Built from a single administrative module (AM) and a variable number of switching modules (SM), interconnected by a communications module (CM), the switch can carry both transport and interprocessor messaging.



**Figure 2.8: Overview of the 5ESS switch architecture.**

- The AM provides control and some interface functions. It handles the administration, operation, and maintenance functions for the global switch. The processing is handled by active/standby dual 3B20D processing facility. These two processing systems function in perfect time and functional synchronism to be able to monitor and act as duplicates to prevent data loss under fault conditions. Call processing and time slot allocation takes place in this module.

The AM performs call-processing functions such as line and trunk group routing determination for the switch modules, and the allocation of time multiplex-switch time slots. Call processing support functions including system with craft maintenance, diagnostics control, software recovery and initialization, and certain fault recovery detection.

A disk memory and input/output processor (IOP) are subunits of the AM. The disk memory provides mass storage for programs and data while the IOP interfaces with support systems, video display units, printers, tape drives, and a master control center which provide the interfaces for manual control of system operations and a system status display.

- The CM contains fabric and in some cases control. It acts as an intelligent communications intermediary between the AM and SM(s) of the global switch. Information forwarded consists of call processing and administrative data messages. The format for the exchange conforms the CCITT X.25 recommendation for the data link layer packet switch protocol.

The CM has a time-space-time (TST) architecture that is implemented by a time multiplex switch to perform time-shared space-division multiplexing, and a time-slot interchange unit for time-division switching. The fiber optic data links, which provide communication paths between SMs, are switched through by the centralized time multiplex switch.

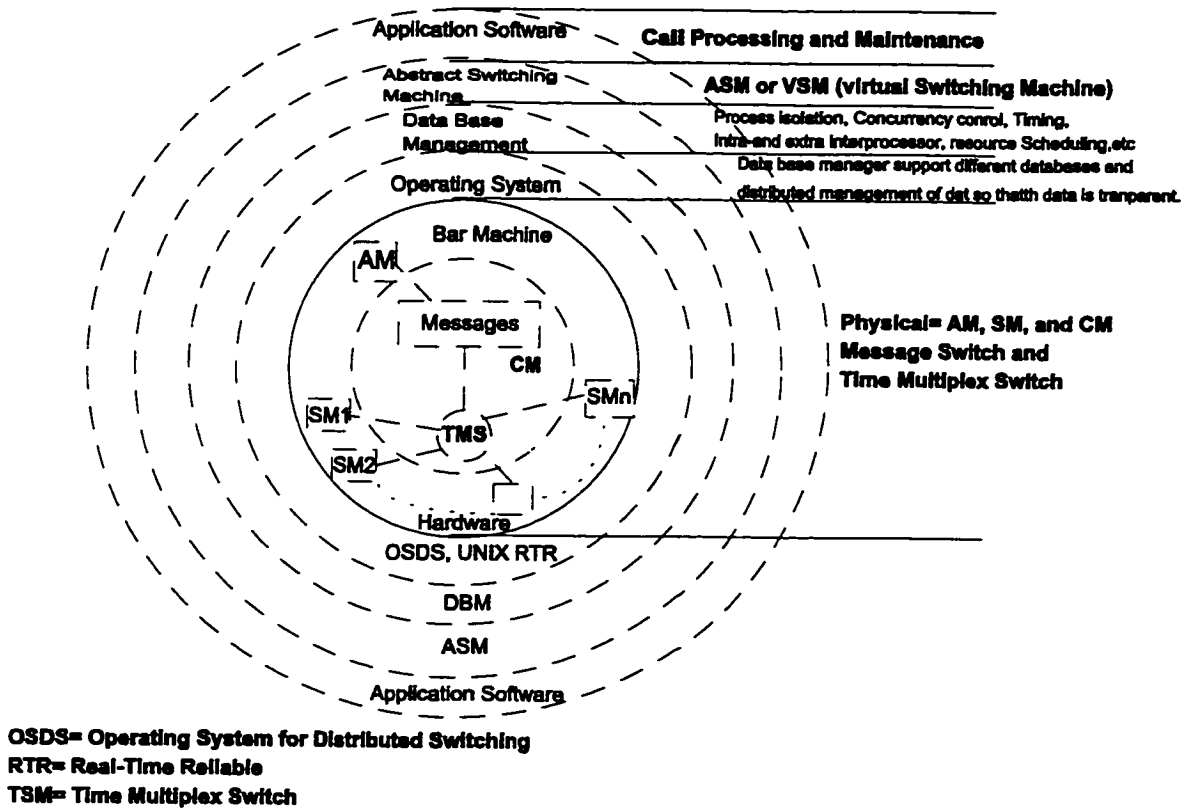
- The SM contain control, fabric, and interface functions. The SM performs as the first stage switching device and provides the trunk and line termination for various lines and trunks. In addition, it provides for initial call processing. Numerous analog and digital trunks may terminate at the SM, at different rates (up to 10 of T1 at 1.544 Mb/sec, up to 16 at 2.048 Mb/s), and in addition subscriber loop carrier systems may also terminate at the SM. Each information bearing channel time slot is identified and mapped in an appropriate TST slot of the CM for completing the circuit switched connection.

Equipment that is common on the SMs includes dual link interface (NCTs), duplicated module processors and slot-interchange units, and a digital services unit. The redundant processors control call-processing, call distribution, and call maintenance. The slot interchange unit switches 256 time slots on each NCT link (512 total). It can connect these time slots to peripheral devices or to the command module. The digital services unit provide time generation and decoding.

**2.4.2.**

**Software**

5 ESS software can be described as conceptual and operational. In conceptual terms, the 5 ESS software architecture consists of a hierarchy of vested virtual machines where software is structured as a set of sequential abstract layers figure 2.9.



**Figure 2.9: Concept of 5ESS software architecture**

These layers apply to all processors located within the hardware of the administrative, communications, and switching modules. The hierarchy is that

any machine, at any layer, uses the services of the lower layer machines and provides services for higher-layer machines.

The functioning of all layers has been given on figure 2.9 except for the abstract switching machine, also known as Virtual Switching Machine (VSM), and represents the fourth layer. An analogy of the virtual channel hierarchy can be made. A conventional operating system provides a virtual machine environment on application level programs by implementing an abstraction of a bare machine. Similarly, the ASM/VSM provides high level abstraction of the switching periphery to the switching application software.

Within the VSM, there are basic resources available to the application software—the logical ports (LPs), network paths, and connectors. Also, there is a terminal process in the VSM controlling the LPs, paths, and connectors it has acquired.

### **2.4.3.**

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## **The 5ESS Switch ISDN Architecture**

The evolution of ISDN involved a number of enhancements, which gracefully evolved to significantly larger capacities and improved reliability:

- Introduction of remote switching modules, including remote clusters with stand-alone capabilities;
- Movement of line-routing functions from the administrative module into a set of designated switching modules;

- Introduction of intelligent peripherals, such as the Integrated Services Line Unit and Protocol handler;
- Introduction of packet switching unit provisioned in switching module that terminate ISDN lines;
- Introduction of Signaling System 7 (SS7) interfaces through the packet switching unit in international versions of the switch, including an optional capability to support an integrated signal transfer point; and
- Implementation of software that supports the International Wireless Standard, Groupe Speciale Mobile.

## **2.5.**

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### **THE 5ESS-2000 SWITCH ARCHITECTURE**

Initially introduced in 1993, the ESS-2000 switch architecture couples switching and transmission functions to cost-effectively support synchronous optical network. Synchronous digital hierarchy (SONET/SDH) interfaces, reduce the cost of wire center operations, and provide a platform for future technologies and services. Significant evolutionary changes in the 5ESS-2000 switch include [5, 24]:

- A new, expandable, advanced technology switching module, featuring a more powerful processor, expandable fabric and interface unit, and SONET/SDH interfaces;

- Development of switch software to facilitate intelligent network control; and
- An expandable, optional digital networking unit, which provides an integrated SONET and, later, SDH interface.

ISDN, intelligent network, wireless access, and SONET/SDH interfaces have been incorporated into the 5ESS switch architecture. Next, the 5ESS switch based-ATM switching entities will be offering broadband services.

Initially, broadband services will be part of an overlay ATM network. Architecturally, the system contains an ATM fabric and a service management module, which provides operations, administration, maintenance and provisioning. The speed of the ATM fabric can be set at 2.4Gbit/s or 20 Gb/s, and has a duplex fault-tolerant architecture. It terminates and supports ATM, frame relay, Switched Megabit Digital Service, and STM clear channel.

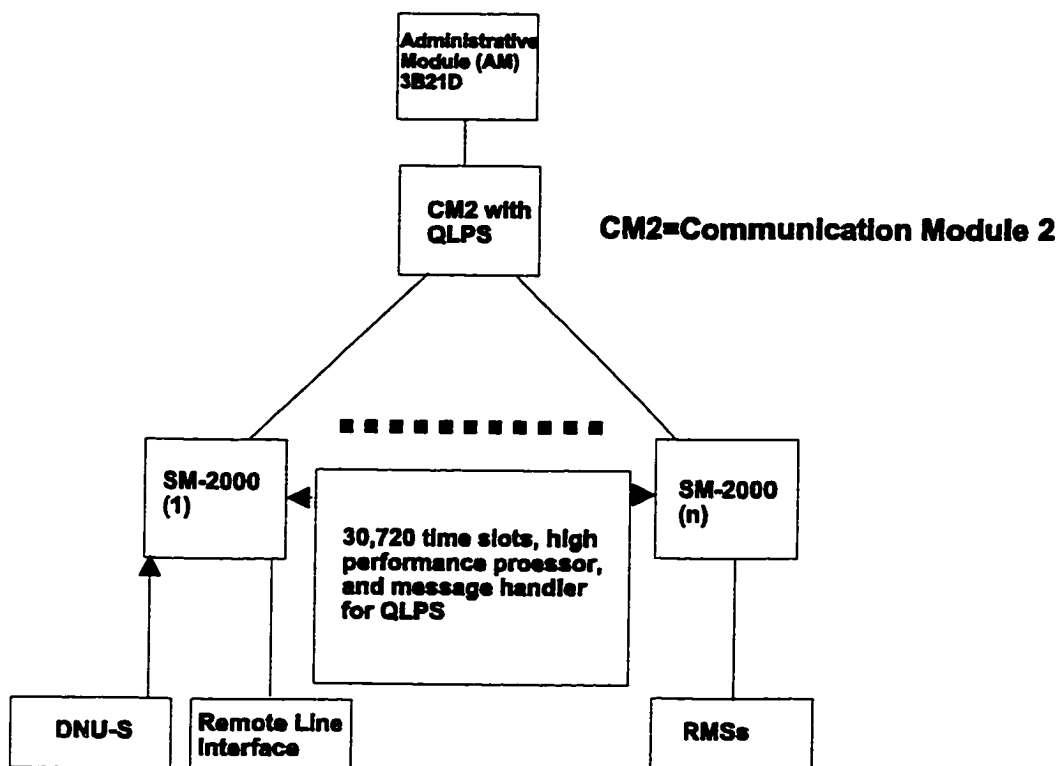
The future network can also be configured for switched broadband services by adding a switched service module, which provides both switched services and signaling.

General Market needs	Services and Applications	Distributed and Small Switches	Emerging Services
Low Cost Wide capacity range Reliability Flexibility Single system Solution	Local/toll Wireless Intelligent Network ISDN Gateway Operator services	Varying size needs Service transparency	Multimedia Hybrid fiber/coax Integrated ATM capability

**Table II.1: Switching System Marketplace Requirements**

The switch units and modules, shown in figure 2.10, are generally independent, allowing the internetworking of all old and new components. Although much

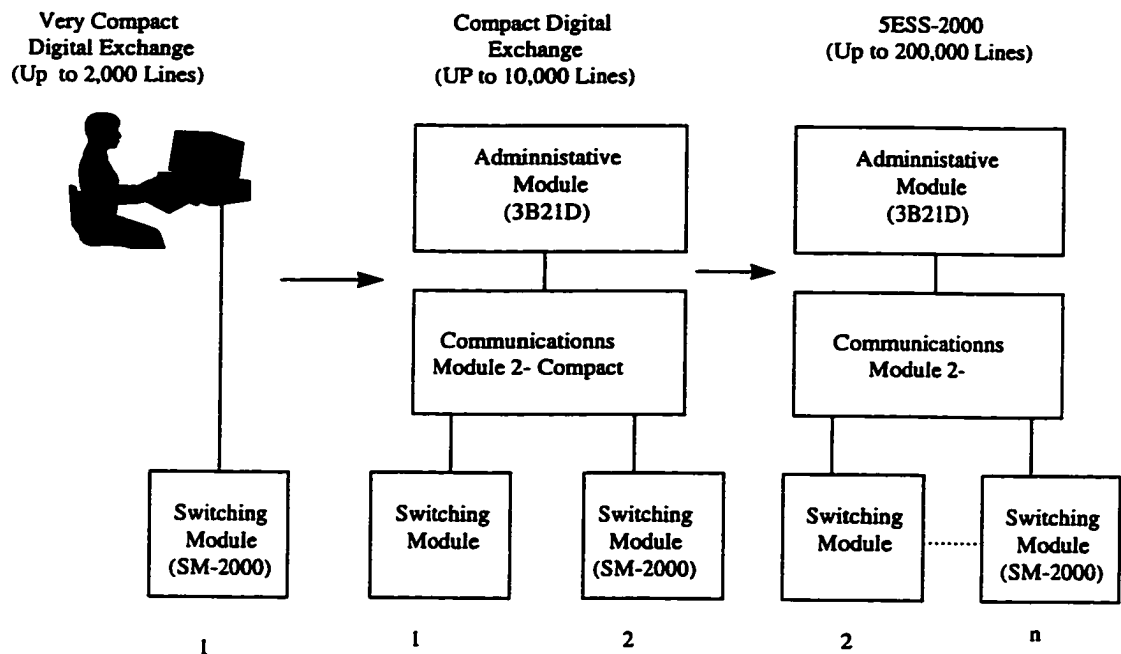
has changed, the basic architecture elements remain the same for both hardware and software. No matter which configuration is selected, the operations, administration, maintenance, and provisioning (OAM&P) interfaces and the subscriber features-set remain the same. Together, these enhancements offer an economical total switching solution having the widest practical range of configurations, capacities, applications, and interfaces.



DNU-S= Digital Networking Unit -SONET/SDH  
 QLPS= QuadLink Packet Switch  
 RSM= Remote Switching Module

**Figure 2.10: Illustration of new components of 5ESS-2000 which are generally independent, allowing them to internetwork with those that were previously available.**

*A new Switching Module: SMs are 5ESS-2000's fundamental distributed elements. The number of 64 kbps time slots in each SM has been increased from 512 to 30,720 for the switching module 2000 (SM-2000). A more powerful, duplex processor complex has been added to handle the increased call processing capacity. A set of message handlers has also been added to facilitate the increasing number of messages within the 5ESS switch architecture.*



**Figure 2.11: Three configurations of the 5ESS switch.** All the 3 configurations can serve ISDN, local, and business service needs. The smaller configurations may also be interfaced into remote switching modules, PBX line groups, and remote integrated service line groups.

The 5ESS-200 switch (figure 2.11 ), in the American environment, contends with both ISDN and IN functionality. In the addition, this newer 5ESS platform permits enhanced processing capabilities, as they exist in IN environments, and servicing capabilities with simpler operations, administration, maintenance and provisioning (OAM&P). Servicing capabilities include handling as many as 200,000 lines, newer remote-switching modules with line capacities of 20,000 lines, etc. The provisioning aspect is unique to most of the recently designed ESS platforms for the United States and Europe. The reliability of the 5ESS platform is better than of most computer facilities.

The switching module (SM-2000) of this machine permits its interfacing with the fiber optic nets using the synchronous digital hierarchy (SDH), with OC-1 and OC-12 carrier capabilities. At the other extreme, the (mB+nD) servicing features of ISDN make provisioning of digital services quite easy and manageable with the 5ESS-2000 platform.

## Chapter 3

### **EVOLUTION TOWARD ATM ENVIRONMENT**

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Voice, data, video. Virtually every new network service places demands on the networks for more and more intelligence.

Intelligence first came to telephone exchange networks in the form of switchboard operators. Over the years, these operators were, in general, replaced with electromechanical switches. Today the electromechanical systems are being rapidly replaced by electronic switches, which due to their stored-program control software have greater service capabilities and flexibility than their electromechanical predecessors.

Intelligent networks are defined to as the carriers of information with distinct algorithmic adaptation. In the context of hardware, intelligence resides in the customized integrated circuit chips, their sophisticated layout, and their interconnections. In the context of software, intelligence is coded as programs,

utilities, or modules. It may reside in the active memories of computers during the execution phase. In the context of firmware, intelligence is microcoded into the control memories. Thus, the basic computer (hardware, software, and firmware), which control, monitor, and process the information, become a part of intelligent networks.

### **3.1.**

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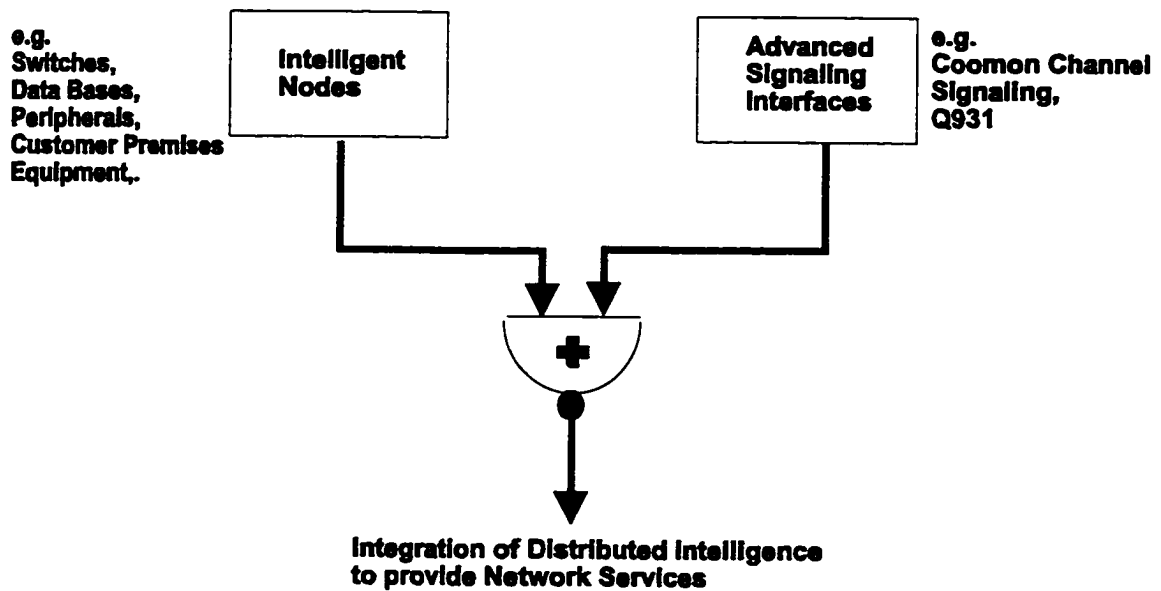
## **INTELLIGENT NETWORKS**

Current INs allow human/machine interaction to add value to the processing and completion of a telephone call. We can tailor this programmed intelligence to satisfy subscriber's needs as they occur. Potentially, the network's added intelligence can generate substantial telecommunications administration (International Exchange Carriers (LECs)) if callers and subscribers to the services definition of an IN encompass not just voice, but other types of public and private networks and information, to add network-wide intelligence to the completion of information transfers. Because an IN reuses network capabilities in a service in a service-independent way, new network services can be introduced.

The Intelligent Network (IN) has been under development for many years, with the goal of allowing all types of information to pass through the telephone Network without special circuits or long installation cycles.

Intelligent networking-in the context of telecommunications industry- involves the interconnection of intelligent nodes (such as switches and real-time data base

systems) via advanced signaling systems (such as common channel signaling system 7 (CCS7)) to provide value-added services and features figure 3.1.



**Figure 3.1: Basic of Intelligent Networking**

The concept of being able to access all kinds of information is not new. The network to support this type of service is new. The IN provides the backbone to support and define these services. As the need for new features and services becomes more important to customers, the need to deliver those services and features in an economical way becomes equally important.

Typically, functions are allocated to network nodes to produce new services and features as well as to provide a more economical way to furnish existing services. Integrating the functions performed by the various nodes allows application of the network's existing intelligence on a network-wide basis. Thus, rather than

having islands of operating independently within the network, the network itself functions as a processing system from an external point of view.

### 3.1.1.

## IN Services

IN services can be grouped into three categories according to the Network capabilities that are required for their implementations:

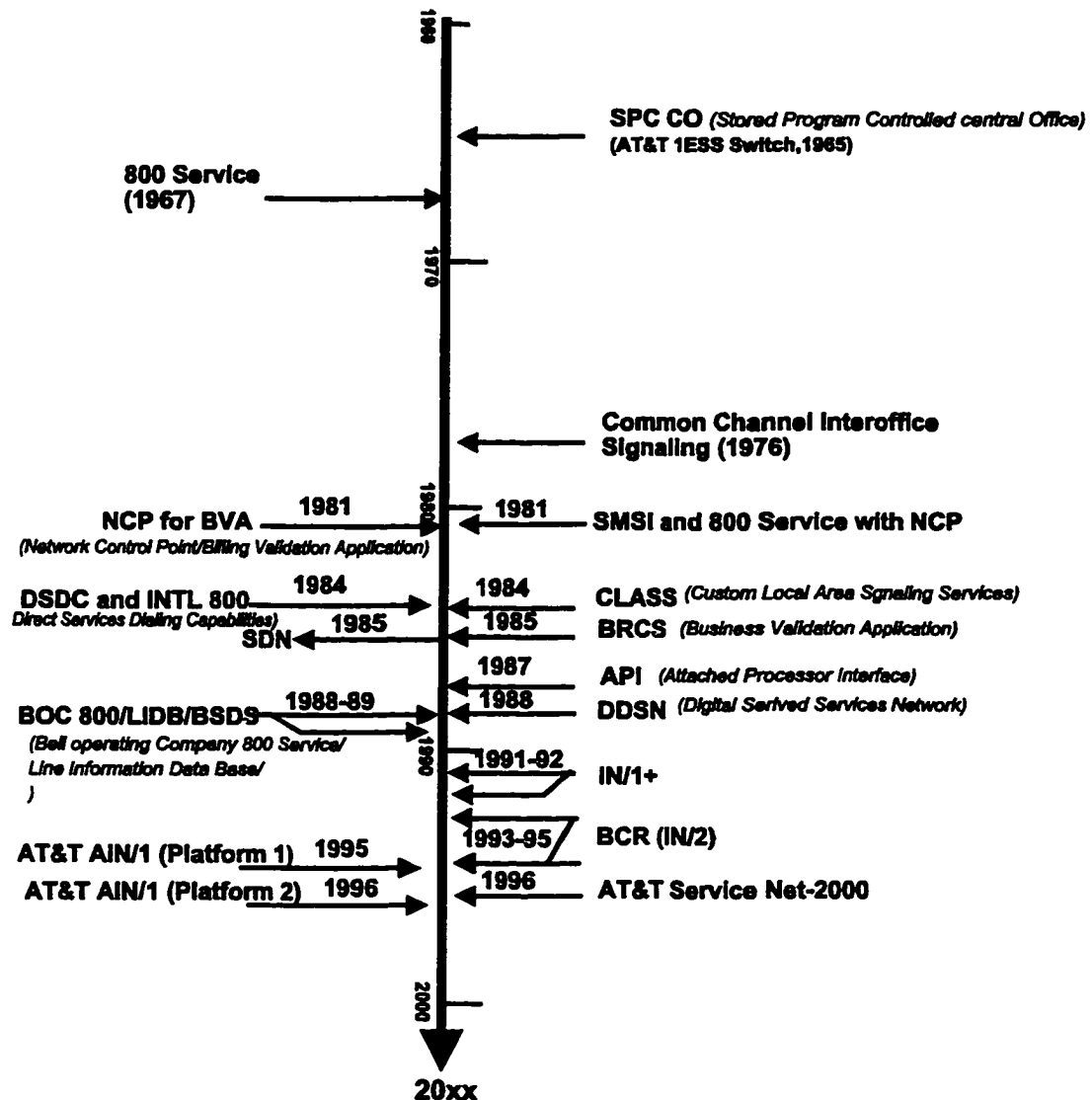
Category	Service	Description
B (called party number services)	Basic Green number service (free phone)	Toll-free service paid for by called party. A special access code e.g. 800 service as trigger
A+B(called and called party) number services	Alternate billing services Enhanced emergency response service Virtual private network Area wide centrex	Allows the user to bill the call to a number other than his or her own number.  Dialing of a special country wide emergency number service as a trigger. The service control number determines the appropriate emergency response control number for call completion.  Provides the functionality of a private or dedicated network using the shared facilities of the public network.  Offers dynamic response and dynamic allocation and uniform numbering plan over geographically dispersed locations. the centrex service utilizes a public network local exchange to provide PBX like features to a group of business customers. Area wide centrex interconnects multiple customer locations as if they were connected to a single switch.
Interactive services	Interactive green number service. Voice messaging Call completion	Allows the user to select one of a set of offered alternatives associated with green number.  Requires user dialogue for entering control commands, for example.  comprises various features to assist the calling party in completing the call to the called party.

**Table 3.1: IN Services**

### 3.1.2.

## Evolution toward Intelligent Networks.

Three events are of particular importance in the evolution of INs because they are firsts with respect to three important foundations of today's intelligent networks - stored program control switching, introduction of advanced signaling protocols and service independence [52].



**Figure 3.2: Intelligent Networks Evolution**

Stored Program Controlled (SPC) switching was introduced commercially with the cut-over of the 1 ESS Switch in Succasunna, NJ in 1965.

Common Channel Signaling was introduced into long-distance portion of the Bell System network in 1976. The signaling protocol used was Common Channel Interoffice Signaling (CCIS) based on CCITT (now ITU-T) Signaling System 6.

Service independent capabilities were first developed under the name of Direct Services Dialing Capabilities (DSDC) in 1984. These DSD capabilities form the foundation of numerous AT&T service today.

### **3.1.2.1. INTRODUCTION OF STORED PROGRAM CONTROLLED CENTRAL OFFICE (AT&T 1ESS SWITCH, 1965)**

The cutover of the 1ESS Switch in Succasunna, New Jersey in 1965 introduced software controlled intelligence into the Bell System's Public Network (PSN).

This was the first of the technological advances which laid the foundation for intelligent networks. The flexibility and compactness of software as opposed to hardware control meant that central office switches could offer new and more sophisticated features. Switches became islands of intelligence within the PSN.

The 1 ESS switch was first used for local tandem purposes at Nashville, Tennessee in 1968 and its toll center use was in 1971.

### **3.1.2.2. INTRODUCTION OF THE COMMON CHANNEL INTEROFFICE SIGNALING**

In 1976, the Bell Systems introduced common channel signaling capabilities (CCIS 6 based on CCITT Signaling System 6) into the long distance portion of its switched network.

The first use of CCIS 6 was for the purpose of interswitch trunk set-up -between a 4A ETS Switch in Madison, Wisconsin and 4 ESS Switch in Chicago, Illinois. CCIS 6 provided speed, security and flexibility advantages over the previous multi-frequency signaling protocol employed in the intertoll network. This was the second technological foundation for intelligent networking -allowing more extensive messaging among existing island of intelligence.

### **3.1.2.3. INTRODUCTION OF THE FIRST NCP- BASED SERVICE (BILLING VALIDATION APPLICATION, 1981)**

The Billing Validation Application (BVA) was the first use of intelligent networking to provide a value-added service -calling card service.

This service uses the archetypal intelligent network distribution of functions. A switching system (in this case, an operator system) provides most call processing functions with a query being directed to a remote data base system to obtain a specific piece of data which will affect subsequent call control.

This application of intelligent networking reduced operator work time for Calling Card calls by allowing the caller to enter the calling card number directly, rather than having to read it to the operator. The BVA at remote data base system verified the calling card number whether entered directly by the caller or by an operator. The another of calling card service - a distributed need for access to global data - made the switch-data base architecture a natural choice.

BVA was the first use of signaling message within the CCIS 6 network.

### **3.1.2.4. INTRODUCTION OF THE NCP BASED 800 SERVICE**

In 1981, 800 Service in the United States was enhanced by using a centralized intelligent networking architecture to provide 800 call routing information.

The distributed need for global data (the translation from 10-digit 800 number to 10-digit routing number) once again made the switch -data base architecture a natural choice. The data base approach eliminated the need for storing data at the many switches functioning as Screen Offices. The messaging abilities of CCIS 6 enabled more information about the caller to be made available (such as the caller's geographic location) and the ten digit translation capabilities of the network control point's INward Wide Area Telecommunications Service (INWATS) database extended the six-digit capabilities previously used. Together these advances allow new 800 features to be offered, for instance, one number service (same 800 number for intra and interstate calls) and area code routing.

### **3.1.2.5. INTRODUCTION OF ADVANCED 800 SERVICE USING DIRECT SERVICES DIALING CAPABILITIES**

Direct Services Dialing Capabilities (DSDC) is an explicit attempt made to create service independent capabilities. A set of events/circumstances at a switch would result in a query to a remote data base system. The data base then executed at a switch would result in a query to a remote data base and responded to the Action Point (ACP). The switch to data base interface was specified in terms of commands sent from the Network Control Point (NCP) to the ACP. The commands are independent of the service being provided. The

sequence of commands and parameters within commands determine pre-existing software will happen. Thus, new services can be constructed by manipulating capabilities to operate in slightly different contexts.

Routing control, announcement, digit collection, recording (billing) and network management commands were defined to allow invocation of specific functions at the ACP or Network Services Complex (NSCX).

The intent of DSDC was to provide a set of capabilities which could be used to provide a multiplicity of new services and features.

Advanced 800 is the first member of the set of services provided by this second generation of data base-based intelligent network services.

With the introduction of DSDC, 800 service was improved with interactive announcement and digit collection capabilities provided by NSCX as well as with other new features, such as command routing, and the introduction of direct customer control.

The service was provided using an architecture whereby all 800 queries were still routed to the INWATS Data base (IDB) and then, only if the customer subscribed to advanced features, to the DSD NCP. This allowed a customer-by-customer migration of service logic as dictated by customer feature needs.

### **3.1.2.6. SIMPLIFIED MESSAGE SERVICE INTERFACE (1981)**

This publicly available interface allowed third-party as well as AT&T equipment to interface with some of the central office's capabilities. As the name indicates,

this interface was designed to allow provision of a message service via a combination of central office and adjunct intelligence.

### **3.1.2.7. CUSTOM LOCAL AREA SIGNALING SERVICES**

In 1984, trialing of a new set of intelligent network services began with the Harrisburg, Pennsylvania installation of Custom Local Area Signaling Services (CLASS) capabilities in 1A ESS switches. CLASS represents the first common channel signaling use of intelligent networks in the local exchange network in the United States. These new services, such as selective Call Rejection and Auto Recall/Auto Call Back, were provided by making use of the extended messaging capabilities of CCIS 6 to transport new call-related information (e.g., Automatic Number Identification (ANI)) between switches.

Since information required for feature operation reside uniquely at either originating or destination switch, a switch-to-switch intelligent network architecture was chosen. Unlike the services previously described, CLASS required deployment of CCIS among a community of end offices.

CLASS represented the first extension of some historically single-switch services, such as Auto Call Back, into the interswitch arena.

Other events such as the creation of IN/1, IN/1+, IN/2 and so on will be discussed while discussing each of the these subjects.

### **3.1.3.**

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## **IN Infrastructure**

Intelligent networks (INs) function like massive and widely dispersed computer systems. Both carry out three distinct functions:

### **3.1.3.1. CONTROL**

The levels of control occur via the different layers of the network. Each layer provides appropriate information for the lower layer to perform the specific function expected from that layer. The structure imposed upon the network functions via the network layers is a definite form of control. One widely recognized model for network functions and its hierarchical structure is the Open System Interconnection (OSI) model. In most computer communication, the OSI layers permit detailed development of protocol.

### **3.1.3.2. COMMUNICATION**

Within the computer system is handled via bus structure in dedicated or shared mode. The outside communication is generally handled via dedicated or addressable ports and standard interfaces. In the network environment, the levels for handling communication of information are not easily categorized because of the wide variety of media and channels, and their interconnecting capabilities. Networks accomplish two basic communication functions (Communications within the network, and Communications with the user).

Computation: Networks are mainly used for exchange of information and not for computation.

### 3.1.3.2. SWITCHING

In the network environments, the switching function is essential and dominant. Real-time, concurrent control of many thousands of communication channels is essential at most of the nodes within the network. For this reason, the design and architecture of switching systems is a major area of specialty and can become as intricate as the design and architecture of any major mainframe computer system.

Switching functions, and thus the intelligence associated with them, form a major part of the network operations. In general most of the common public networks have to switch and allocate channels to permit the flow of information from node to node, customer to customer, from workstation to workstation. The same physical resources are used to carry different channels or even blocks or packets of information. The channels may be dedicated to a certain user for the duration of time of usage.

Intelligent networks carry and communicate information with distinct algorithmic adaptation. Adaptability occurs at a local or global level. Local adaptability is administered by the programs resident at a local or a global level. Local adaptability is administered by the coherent functioning of the node to process requests from another node and, thus global and extraneous functions.

The functioning of any generic IN depends upon four essential components: *Interfaces* to provide the flow of information into and out of the network, *Monitoring Computers* to supervise and control the flow of information via

various channels according to software modules in the computers, *Switching Systems* to switch the channels and complete the physical (electrical, microwave, optical, satellite, etc.) or logical channels between the user interfacing with the network, and *associated transmission* facilities: to carry the electrical, optical, or microwave signal in the appropriate medium to convey and communicate the information.

### **3.1.4.**

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## **Intelligent Networks Basic Elements**

In the specific and implemented versions of INs in the public domain, five basic building blocks have been identified:

### **3.1.4.1. SERVICE SWITCHING POINT (SSP)**

The major function of the SSP is to detect events -called *triggers*- that indicates an IN call. After this triggering, the service switching point suspends call processing and starts a series of transactions with the *service control point* to determine the final handling of the call.

The software in switches separates basic call control from the service control of an IN. The service switching point requires a call model and trigger table. In addition, a platform capabilities to communicate with external entities (e.g., service logic control point, service circuit node, or adjunct) or internal entities (service logic that resides within the service switching point). The service switching point must also communicate with other IN elements, using standard

IN protocol, as they become available. The service switching point typically is provided in local/ or transit exchanges.

#### **3.1.4.2. SIGNAL TRANSFER POINT (STP)**

The request is provided by the CCISS signals to appropriate service/information providers. These signals calling for such services/information have to reach the appropriate data bases distributed within the network. The STP is a switch to forward the signals in real time and generate the response from the data bases in SCPs.

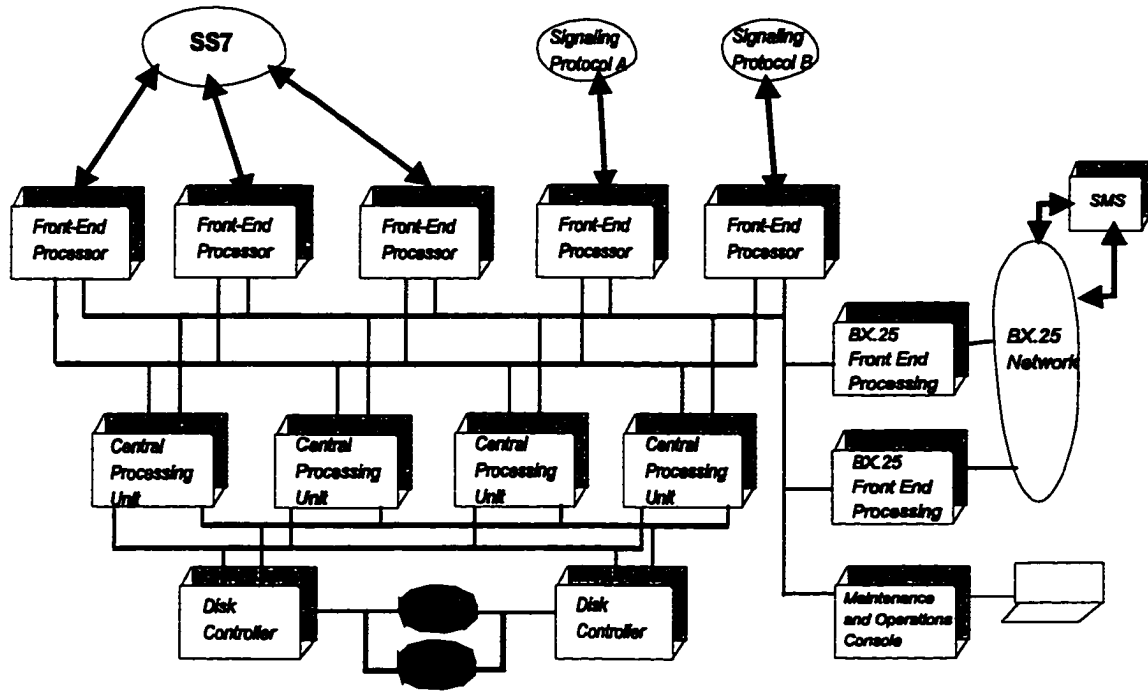
Three type of links spans these STPs. Type A links connect the switching offices with the STPs. Type B links connect The STPs of neighboring (adjoining) regions. Type C links are specially balanced, mated pairs that interconnect STPs with the regions.

#### **3.1.4.3. SERVICE CONTROL POINT (SCP)**

The SCP is a real-time database system that, based on a query from the SSP, performs subscriber- or application- specific service logic, and then sends back instructions to the SSP on how to continue the call processing. It also contains a group of processors that can provide switches with customized instructions for handling as many as a million calls in an hour.

The SCP is located at the heart of the regional intelligent- network architecture. It receive queries from and sends instructions to service switching points (SSPs) via the common-channel signaling network and packet switches called signaling

transfer points (STPs). Customer records and call-processing programs are entered into the SCP from the Service Management System. The SCP



**Figure 3.3: Hardware Architecture OF the SCP**

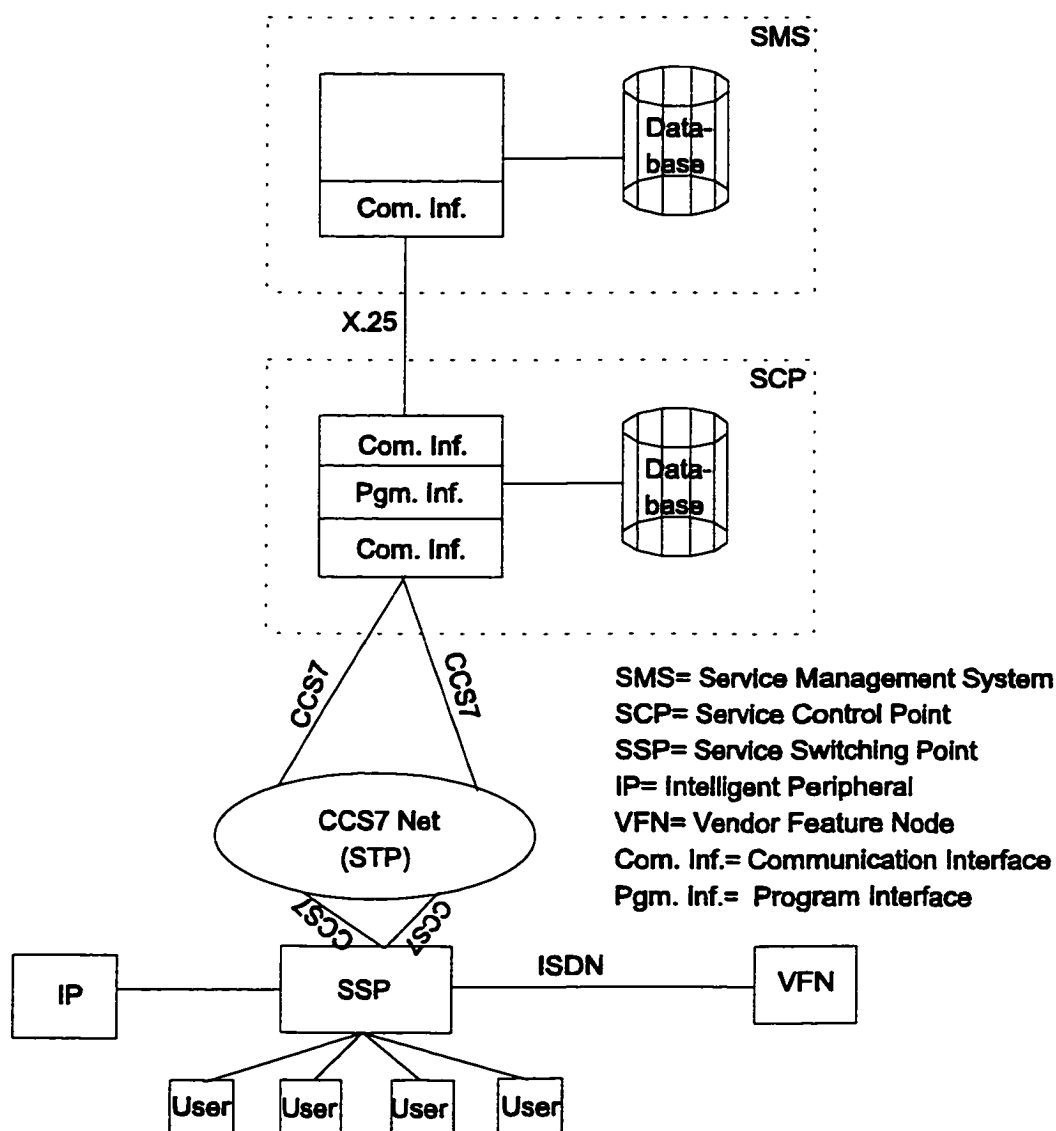
A new service can be introduced in a geographic region by loading new software into the service control point, rather than into hundreds of switches.

The IN core network connect the SSPs, STPs, and SCPs by common channel signaling number 7 (CSS7) links. SSPs detect special services requirements of the particular call. These SSPs have the CCS7 facilities to query a data base located at the SCP via the transaction capabilities part (TCAP) of the CCS7 protocol.

#### **3.1.4.4. SERVICE MANAGEMENT SYSTEM (SMS)**

The SMS function is intricately tied with the functions of the SCP. In fact the SMS is an off-line support facility used to enter customer data into the SCP's data

bases within an intelligent network. It is also used to enter the call-processing software into the SCP. The SMS communicates with the SCP through the interface cards that processes the BX.25 protocol used for communication between the SMS and the SCP. The SMS also needs a series of front end processors to link with the dual buses, which provide data paths between the SCP's front end processors and the SCP's bank of CPUs.



**Figure 3.4: IN Components**

### **3.1.4.5. INTELLIGENT PERIPHERAL (IP)**

The intelligent peripherals are generally installed to perform both voice and data services. voice services include storage and retrieval of messages, recognition of voice and its synthesis, and code conversion between the various coding techniques. Data services include protocol conversion for narrow band and wideband, data rate conversion, video format, etc.

Figure 3.4 shows the component of an intelligent network. The TCAP is probably the most versatile of all SS7 protocols. TCAP is used for two purposes: Accessing remote databases and invoking features in remote entities.

The signal transfer point facilitates the communication between SSPs and SCPs. The transfer takes place to dispatch the query to the right SCP for quick response. This communication takes place via the CSS7 links. The functions of the STP may be controlled by resident signaling engineering and administration system.

The SCP acts primarily as an end of the CCS network to find the answers to queries from the various SSPs. These answers are generally in the form of data crucial for the completion of the service requested. Data are typically retrieved from highly organized operational data base resident in the SCPs. The main hardware component of the SCP is a powerful specialized computer that handles the numerous SCP functions in parallel and in real time. The SCPs are also generally interconnected through the IN core packet switching capability.

Depending upon the requirements, it is also feasible to interconnect the packet switch of the various SCPs to the packet switch public data network.

Typical of the intelligent networks features are network resource sharing, error detection, and self heading after network and hardware malfunction. Fault tolerance, global interconnect capability, local area networking and wide area networking, interfacing, dynamic expansion, high-speed machine-to-machine linkage, and management of dynamic and distributed switching capability of the network are programmable, if not embedded.

### **3.1.5.**

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## **Standards**

Service standards is based on country requirements for service functions.

The CCS 7 (TCAP, SCCP, and MTP) must be based on CCITT standards or Bellcore requirements depending on the country of deployment.

The following X.25 requirements are for the US and may only be different from county to country. The signaling protocol between:

- The SMS and the SCP is based on the X.25 according to CCITT
- The SCP and the SMS is based on the BX.25 protocol
- The SCP and Operation Support Center is based on the BX.25 protocol
- The PSPDN and the SCP is based on X.25, according to CCITT.

The X.25 and BX.25 protocols are very similar on the lower three levels of the OSI model.

Trunk signaling between the network exchanges may vary from country to country, and ISDN access is based on country-specific requirements

### **3.1.6.**

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## **Capacity and Performance**

The traditional methods of characterizing the capacity of any system relate to the measured of perceived delay, which the system adds to any action that requires a response. Since virtually all of the network components being used in IN are engineered according to the expected load and would typically have excess capacity to handle the excess traffic with delays, queuing models are applied to all analyses of the effects of IN. In order to meet a particular delay requirement (normally expressed as maximum average delay and 95% less than a specific value), it would be necessary to provide more facilities where possible (e.g., signaling links) or to support fewer subscribers with fewer transactions per network entity (e.g., the SSP).

### **3.1.6.1. SERVICE SWITCHING POINT**

SSP capacities are typically quoted in terms of Busy Hour Call Attempts (BHCA) where a call is considered to a normal and simple call. Where distributed processing is used in SSP, an attempted but failed call requires virtually the same processing time as a completed call, and the time is largely unaffected by the

application of any features. Only multi-way calls tend to use significantly more resources and these are typically counted as multiple calls when quoting BHCAs.

An SSP does not have an absolute maximum BHCA capacity since the limit is not the result of a physical limit. Therefore, This BHCA rating is quoted at a particular performance value based on delay (i.e., based on the SSP meeting the performance requirements for time to respond, such as return of dial tone, through connection, disconnection, etc.).

### **3.1.6.2. SIGNALING LINKS**

Signaling Links are engineered to provide for the rapid transfer of messages between network entities. As with the SSP, this is based on requirements to meet a maximum delay, which is usually quoted as round trip. This can also be stated in terms of an average and of 95% limits at particular traffic loads, and must consider the random nature of the SS7 network, with the conventional network architecture, is closely related to the volume of offered call attempts, with five messages per typical call. In addition, the traffic tends to be distributed throughout the network from/to all SSPs and, therefore, through all STPs. Due to natural disasters, calling patterns of the subscribers may be anything but random for short periods of time, which can cause delays and failed calls, but the attempted call/signaling load remains relatively constant.

With the IN architecture the offered call/signaling link load ratio change drastically if the calling patterns or service usage shifts significantly. At this time,

there could be a large number of SCP accesses to activate services followed by a much larger volume of calls being forwarded.

Number of Messages For Typical non-IN Call at Each SSP		Number of Messages Added for IN Call at Originating SSP	
Successful call set-up	3	800 Successful call set-up	
		Basic	2
		Improved	4
Unsuccessful set-up (busy)	2	Call Forwarding (SLI in SSP)	
		activating	2
		forwarding	2
Call release		Call Forwarding (SLI in SCP)	
ANSI T1.113-1987	3	activation	6
CCITT Rec. Q.764	2	forwarding (Simple Scenario)	8

Not Including Messages for new call leg

### 3.1.6.3. SERVICE CONTROL POINT

Due to the widely varying tasks of the SCP for IN, its capacity can be expressed in several ways. One simple criteria of capacity is the number of received and sent SS7 messages, regardless of their content or purpose. However, this approach would treat the SCP simply as a packet switch, as if did not process the messages. Instead, the capacity must be stated as the number of various type of transactions or functions that it can perform. Here again, the variety of functions does not allow a simple statement, as is done for the SSP (BHCA).

Nodal delays and their management need sophistication from network utilities. For example, the electromechanical stowger capability operates at about a hundred milliseconds per bit, whereas the experimental optical switches perform channel switching at a few gigabits per second (Gbps). Widely varying channel capacities are also encountered in established networks, thus demanding another dimension of sophistication from the network utilities. For example, the newer optical transmission capacity ranges from 2.4 to even 30 Gbps over 30 Km to 170

Km. At the other extreme, the sub-rate capability ranges from 300 bits per second to 1200bps over the older unconditioned twisted wire pairs. The newer, fast modems operate at 28.8 kbps or even 32kbps. In 1995, the very high speed modems operate at 2.048 and even 6.3 Mbps under special conditions. In the context of the emerging high-speed transmission technology, the convergence of silicon processing and switching with fiber optic transmission in the same macroscopic environments becomes a desirable intermediate goal.

The control of the flow of information through the network assumes a more complex form. However, the ESSs play a very definite role since the B and D channels are separated in the data packs and demultiplexed accurately. The control information is processed and governs the flow of B and D channel data. At large central offices where ISDN switching is incorporated (5ESS), the network control assumes the role.

Newer fiber communication links open new frontiers for new high rates, the standards of the synchronous optical network (SONET), start out with the OC-1. Higher rates offer multiple SONET synchronous transport system (STS) rates and become attractive for massive mainframe networks. To integrate the optical capability of the fiber and the current limited usage of the digital transport by the present day services, the ATM standards has been proposed and accepted throughout the global community.

In a social sense, it remains to be established that a multiple society despite the knowledge revolution [1, 2, 3, 4, 5].

## **3.2.**

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### **SIGNALING IN IN ENVIRONMENT**

The signaling network is the backbone of IN-based services. The signaling network's ability to perform message switching between network elements enables it to use network resources more efficiently. The signaling network, which uses a separate signaling link, not connected with voice traffic, was first introduced in the U.S network in 1976.

The signaling network perform functions essential to network operations and provides many benefits for the telecommunications administration. For example, the signaling network allows the exchange of information messages between network elements without tying up network trunks and service facilities. A signaling network also can prevent unnecessary tie-ups of network resources when service subscribers are busy or paths to subscribers unavailable.

#### **3.2.1.**

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### **Signaling System 7**

Signaling System 7 (SS7), a specific type of signaling, carries types of information elements, such as calling-party number, which are useful for intelligent service management. For example, if the service logic receives the caller's origination information, it can route automotive repair calls to the nearest auto mechanic shop. In addition, access to some network resources (such as service control point) is only possible with a signaling network.

The Signaling System 7 (SS7) has an evolutionary architecture of its own. The exchange of signaling information also follows of the OSI model (see chapter 1).

In the composite functions of signaling and communication, the signaling flows through the SS7 network and the user information flows through the communication network (in all circuit-switched configurations).

The Signaling System 7 (SS7) architecture consists of four levels (or layers):

### **3.2.1.1. MESSAGE TRANSFER PART (MTP)**

The lowest level, corresponds to the lower three layers of the OSI Model ( Network, Data Link, and The physical layer). It forces any two signaling points in the CCS7 network to route information accurately, dependably, and quickly. The MTP provides a reliable but connectionless service for routing message through the network. The lowest level, *signaling data link*, corresponds to the physical layer of the OSI reference model and is concerned with the physical and electrical characteristics of the signaling links. These include links between STPs, between an STP and an SSP, and control links between SSPs. The *signaling link level* is a data link control protocol that provides for the reliable sequenced delivery of data across a signaling link; It corresponds to layer 2 of the OSI model. The top level of the MTP, referred to as the *signaling network* level or function, provides for routing data across multiple STPs from control source to control destination. These three levels together do not provide the complete set of functions and services specified in the OSI layers 1-3, most notably in the areas of addressing and connection-oriented service.

### **3.2.1.2. SIGNALING CONNECTION CONTROL PART (SCCP)**

The SCCP corresponds to the transport layer of the OSI reference model. It finds a means for the transportation of the messages between the signaling points. The SCCP and MTP together are referred as Network Service Part (NSP) which is simply a message delivery system. The remaining levels of the SS7 model deals with the actual content of the messages.

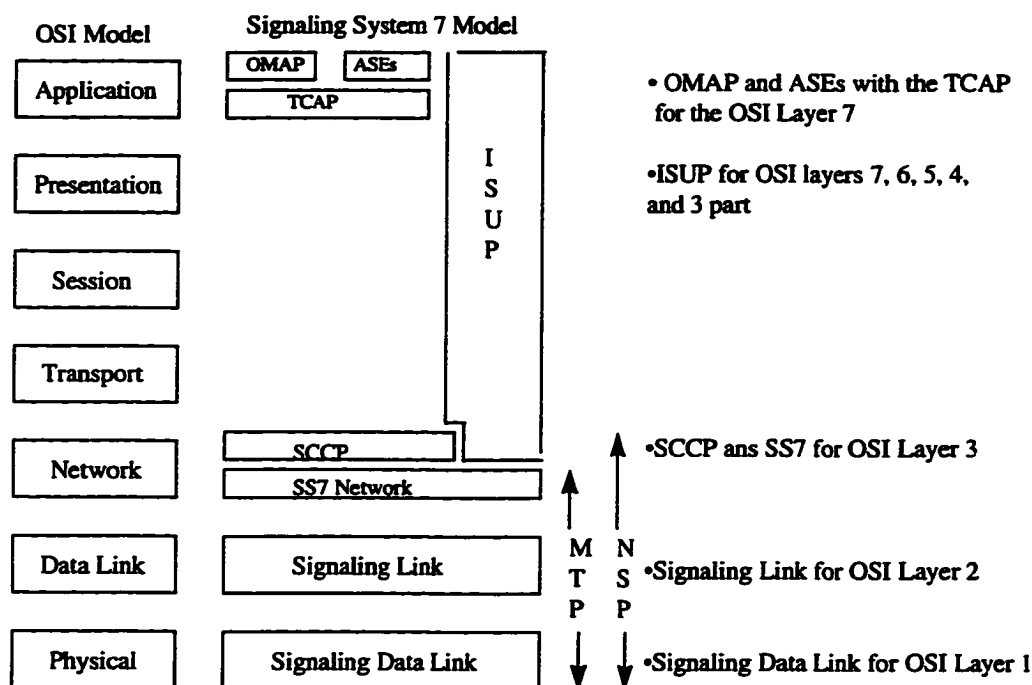
SCCP enhances the connectionless sequenced transmission service provided by the MTP, to meet the needs of those user parts requiring enriched connectionless or connection-oriented service to transfer signaling information between nodes. Classic circuit switched telephone call-related signaling does not use SCCP.

### **3.2.1.3. TELEPHONY USER PART (TUP)**

The TUP is essential for call set up and call disconnect in the regular telephone networks.

### **3.2.1.4. TRANSACTION CAPABILITIES APPLICATIONS PART (TCAP)**

The TCAP, first introduced in 1988, is used to find the applications with data bases like 800 service, alternate billing service, or credit call service. ISUP deals with ISDN *Users Part*, and OMAP deals with the *Operations, Maintenance, and Administration Part*.



**Figure 3.5: SS7 Protocol**

Transmission, protocols, signaling, switching, and the applicability of the OSI reference model prevail in both data networking and communication networking.

A certain amount of network intelligence has been placed in the network elements to substantiate the newer services, such as, call forwarding, call waiting, automated redialing, and the 800 and 900 services. This embedded intelligence, in the form of the system utility programs for the ESSs is finding its way in the networks. These simultaneous demands of ISDN and INs on the modern networks have had a profound impact on the functionality of the switching systems and their controlling software. These new switching systems blend their three basic

functions (Switching, Control, and administration) with service provisioning in the most innovation and unique compositions.

The IN consists of a series of intelligent nodes, each capable of processing at various levels, and each capable of communicating with one another over data links.

The IN relies on the SS7 network, which forms its backbone. SS7 provides the basic infrastructure needed for the service switching point (SS7), which provides the local access as well as an ISDN interface for all signaling Transfer Point (STP), which provides packet switching of message-based signaling protocols for use in the IN and for the Service control Point (SCP), which provides access to the IN database.

The SCP is connected to a Service Management System (SMS), which provides a human interface to the database, as well as the capability to update the database when needed.

When a call is placed in the IN, a request for call-handling instructions is sent to the SCP using the Transaction Capabilities Application Part (TCAP) protocol.

The database provides the instructions for handling the call based upon the customized service instructions the subscriber has programmed, and sends them to the office switch. The end office switch then communicates to the IP using the ISDN protocol to attain the use of resources such as recording and other devices.

The call setup and teardown is handled using conventional SS7 protocols.

### 3.2.2.

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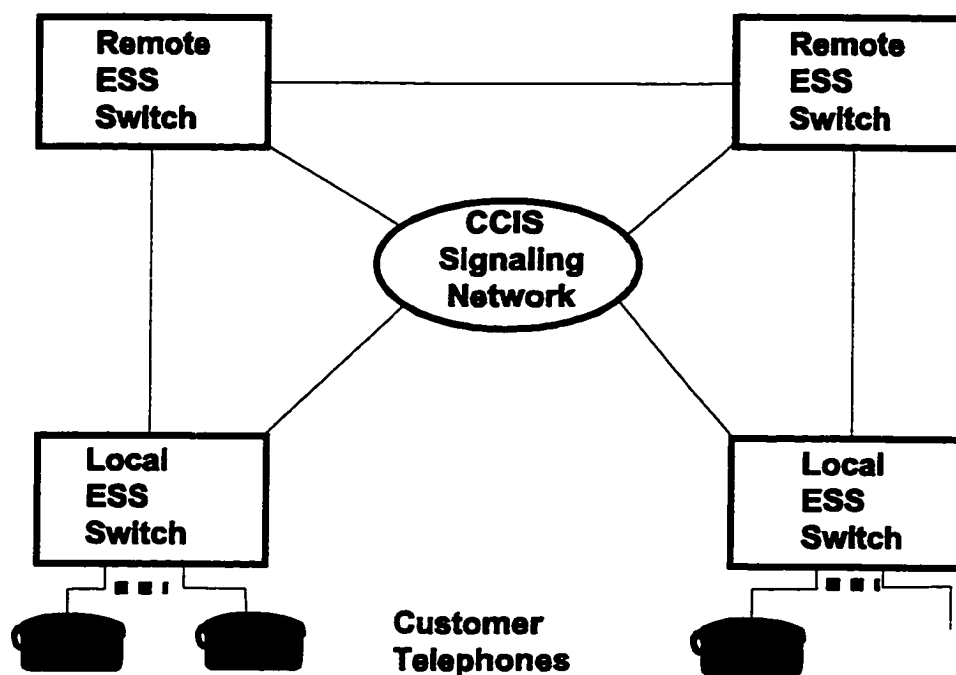
## Common Channel Interoffice Signaling network

The signaling network is an essential pathway to carry standard signals. It's highly dependable and almost totally duplicated to ensure that its functionality is disaster proof. The signaling network are vital to all network functions. Until today, the same network is used to carry signaling and voice or data between customers. The signaling networks that are used to control and monitor the flow of data are called Common Channel Signaling networks. Signaling is essential to the functionality of IN.

When AT&T introduced CCS in 1976, it was known as Common Channel Interoffice Signaling. The message format was similar to CCITT Signaling System No. 6 with the signaling information transmitted across 2.4 kbps analog data links. The use of CCIS-improved trunk signaling between 4 ESS toll switches (introduced at the same time) operation under stored program control (SPC). The introduction of CCIS increased the speed and lowered the cost of call setup and disconnect on the intertoll network.

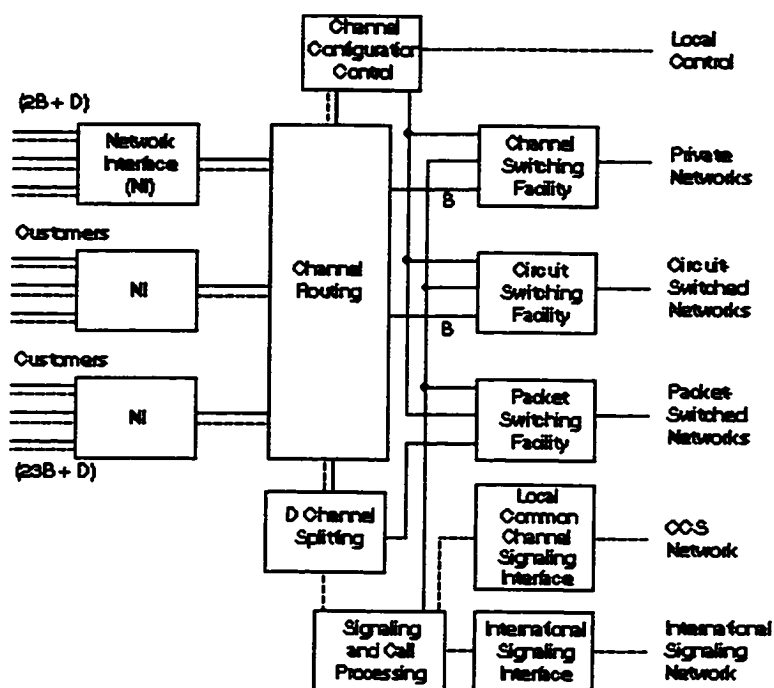
The CCIS sends the information over its own separate data network for control of many voice and data circuits (See Figure 3.6). This is called *common channel signaling*. Essentially, a complete transparent subnetwork controls the switches by transferring control information between SPC processors at locations that are called *Signal Transfer Points (STPs)*. With CCIS, phone calls and data message

services are connected faster and more efficiently, thus saving time and operating expense, and offering more services to the network user.



**Figure 3.6: A separate network to control of telephone system**

In the context of ISDN signaling is carried out by the standard CCS7 network adopted in USA and Europe. The network uses out-of-band (see Chapter 1) information (i.e., information on the D channel). To control signal and the various switches to complete and monitor the B channels (see figure 3.7). This network becomes essential in the circuit switched context because channels provide transparent end-to-end digital connectivity for the network users. If and when a transaction to the packet mode to occur, the CCS information is used to provide the transition and vice versa.



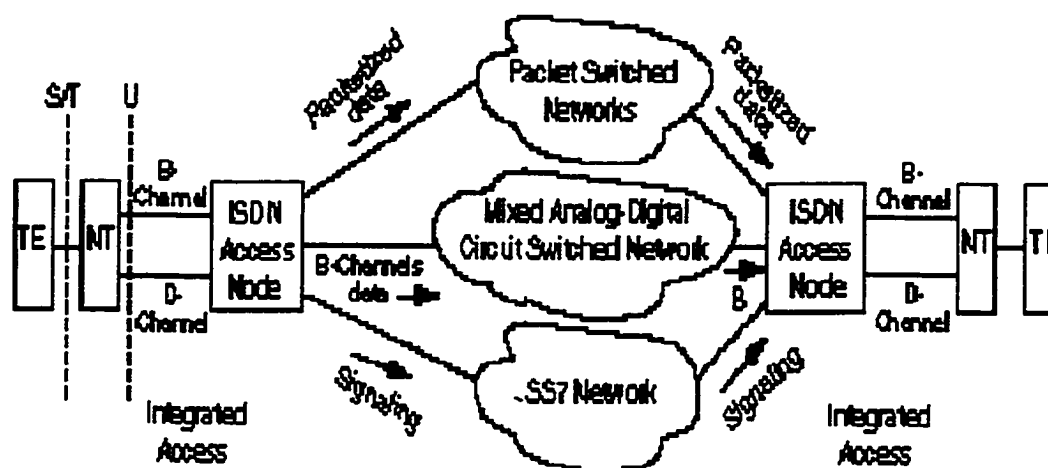
**Figure 3.7: The need for Common Channel Signaling Network to control and monitor the B channels via the D Channel in reference to the ISDN environment is depicted.**

### 3.2.3.

## IN versus ISDN

Digital telephone environments allow for transmission, switching, routing and storage to be accomplished in the digital mode. The channel control of voice and speech channels is also digital. The encoding and decoding of speech are both done at the telephone set. Even low bit rate speech (such as stored messages and announcements) is reconverted to analog form at the telephone set. All the inherent advantages of digital computing and processing techniques can be imported in the digital telephone networks. In a sense, the ISDN framework blends digital telephone network features with other digital services such as

packet-switching, message storage, forwarding, automated message selection, etc.



**Figure 3.8: ISDN Architecture Showing the Separation of Signaling Packet and Circuit Switched Data.** ISDN - Integrated Services Digital Network; NT - Network Termination; SS7 - Signaling System 7; TE - Terminal Equipment

ISDN as a universal all-purpose Digital Services Network: Signaling is accomplished over the SS7 network fairly well evolved and adopted in the United States and Europe. This network uses the out-of-band information (i.e., the information on the D channel), to control and signal the various switches to complete and monitor the B channels. Figure 3.8 depicts the role of the signaling and control of the information bearing B channels.

This network becomes essential in the circuit-switched context because the B channels provide transparent end-to-end digital connectivity for the network users. If and when a transition to the packet mode on the 'B2' channel of the BRISDN is to occur, then the SS7 signaling information is provided for the

transition and vice versa. BISDN and N-ISDN functions also depend upon the separation of the customer data in the 'B' channel from the control and packet data in the 'D' channel. Figure 3.8 depicts the functional details of an ISDN node including customer interfacing, channel routing, and D channel splitting to separate control signals from packetized/localized control and telemetry information (if any).

### **3.3.**

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## **INTELLIGENT NETWORKS: IN/1, IN/1+, AND IN/2**

Intelligent network has been undergoing through a series of changes to answer the user's as well as the provider's and mainly the researcher needs. The major development of the IN are IN/1, IN/1+, and IN/2.

### **3.3.1.**

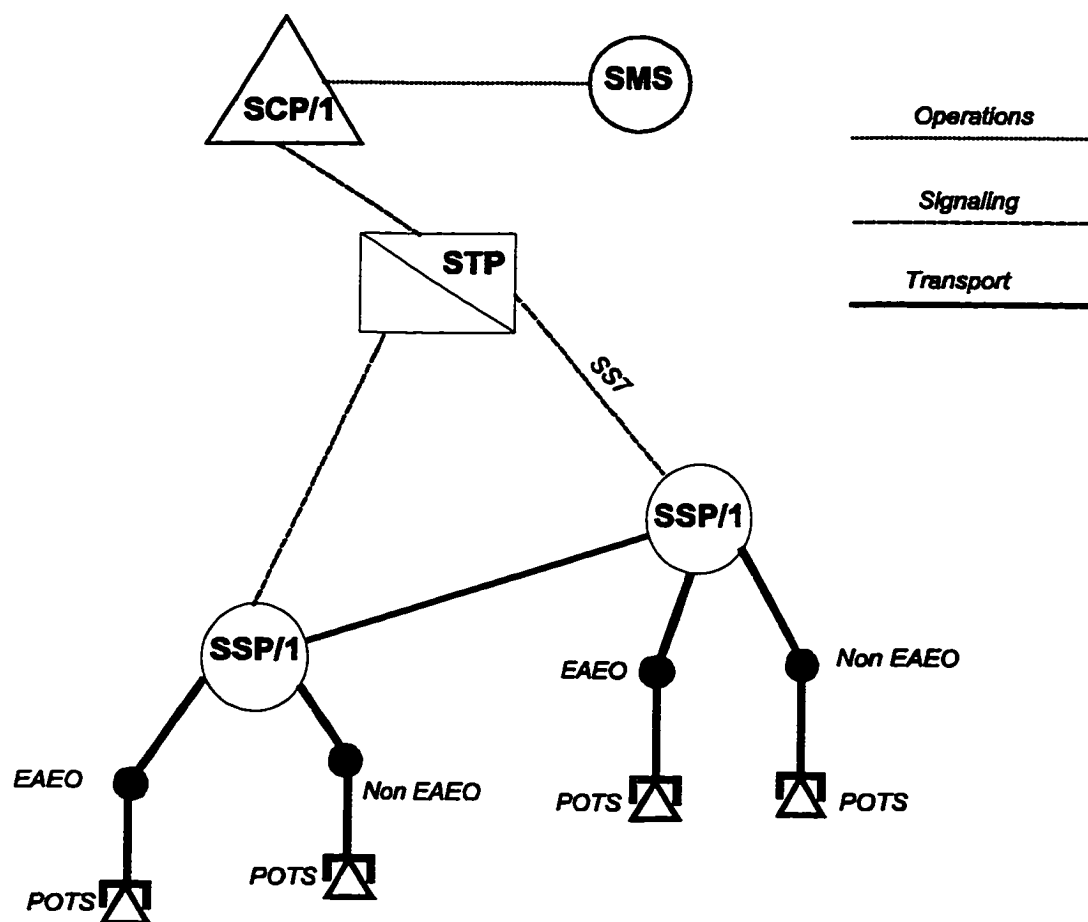
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## **Intelligent Network 1**

In the 1988-1989 time-frame, the Regional Holding Companies (RHCs) began deployment of three intelligent network applications based on use of a centralized data base. The three applications are Line Information Data Base, BOC 800 and business Data Base.

The IN/1 architecture was target toward services that can be provided by centralized switching vehicles, for example access tandems, serving a large customer base. The switching entities contain call processing software (SSP), that reacts to specify service triggers and initiates queries to a centralized SCP over a

common channel signaling network. The SCP contains service specific applications software and customer records. The SCP acts upon the query from the SSP and sends back message containing the data and instructions needed to complete the service. The final element in the architecture is the SMS that supports the administration of the customer records within the SCP.



**Figure 3.9: IN/1 architecture**

The IN/1 meets its principal objective, which is to introduce new service capabilities at a high level in the network. It has, however, inherent limitations and weaknesses. The major software elements, SSP-SCP-SMS, are service specific

and must be defined and developed for each new service. This is a long and expensive process. Furthermore, the services that can be provided at a tandem level in the network are limited. Experience with the deployment of IN/1 applications has clearly demonstrated that SSP is the most critical, most costly and longest lead time element in the architecture. The next step evolution of intelligent networks (IN/1+, IN/2) alleviate some of these problems.

### **3.3.2.**

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## **Intelligent Network/1+**

While there is a certain degree of flexibility and functional reusability of earlier service capabilities, most of the earlier services start at originating (dialed digits or line class) and end shortly afterwards through successful routing. Some caller prompting, screening, and route selection functions are also performed. The earlier functions are typically designed from a single call leg<sup>1</sup> perspective. IN/1+ services encompass the manipulation of multiple call legs and go beyond originating type features. To name a few, these services include network Automatic Call Distributor (ACD), Network 911(N911), Area-wide Centrex, Dialed Number Services, Custom Local Area Signaling Services (CLASS), and others. And also, because of the growing interest in advanced, driven by the promise of newer hardware and software technologies and dependent upon the use of more granular service primitives.

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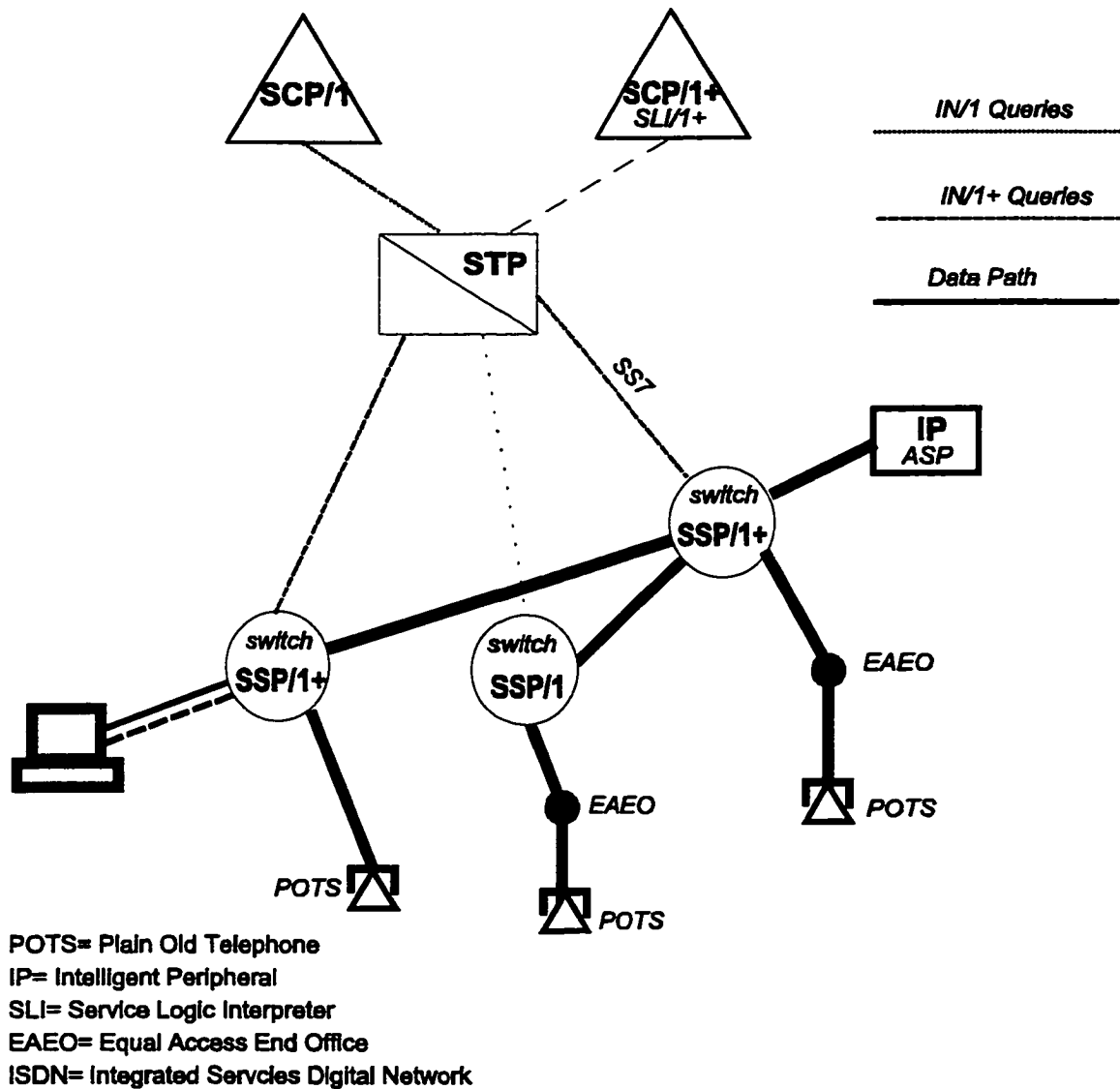
<sup>1</sup> A leg is a connection from a specific node to an addressable entity. The addressable entity may be another network node, such as switching system, but will ordinarily be a termination to a user of the network.

IN/1+ provides the three specific enhancements to call processing functionality mentioned above: manipulation of multiple call legs, non-originating-type features, and low-level primitives [37]. More broadly, however, IN/1+ is a combination of the following:

- An evolution of the existing network.
- Multi-vendor network elements that include SSP, STP, SCP, SMS, and Intelligent Peripheral (IP).
- Defined Standard interfaces across all network nodes.
- Distributed service intelligence and control.
- Standard service-independent capabilities.

The control hierarchy for the service-independent IN/1+ is as follow: a trigger table mechanism at the SSP determines when IN/1+ control is required for a call. Information in the trigger table is used to formulate a query to the service logic in the Service Logic Interpreter (SLI), which is a logical module in the SCP.

Normal call processing is suspended and the SSP awaits instructions from the SLI. The SLI will process the query and will instruct the SSP to execute a set of FCs for the call. The design of IN/1+ service independence centers around the trigger table and the FCs [37].



**Fig 3.10: IN/1+ Architecture**

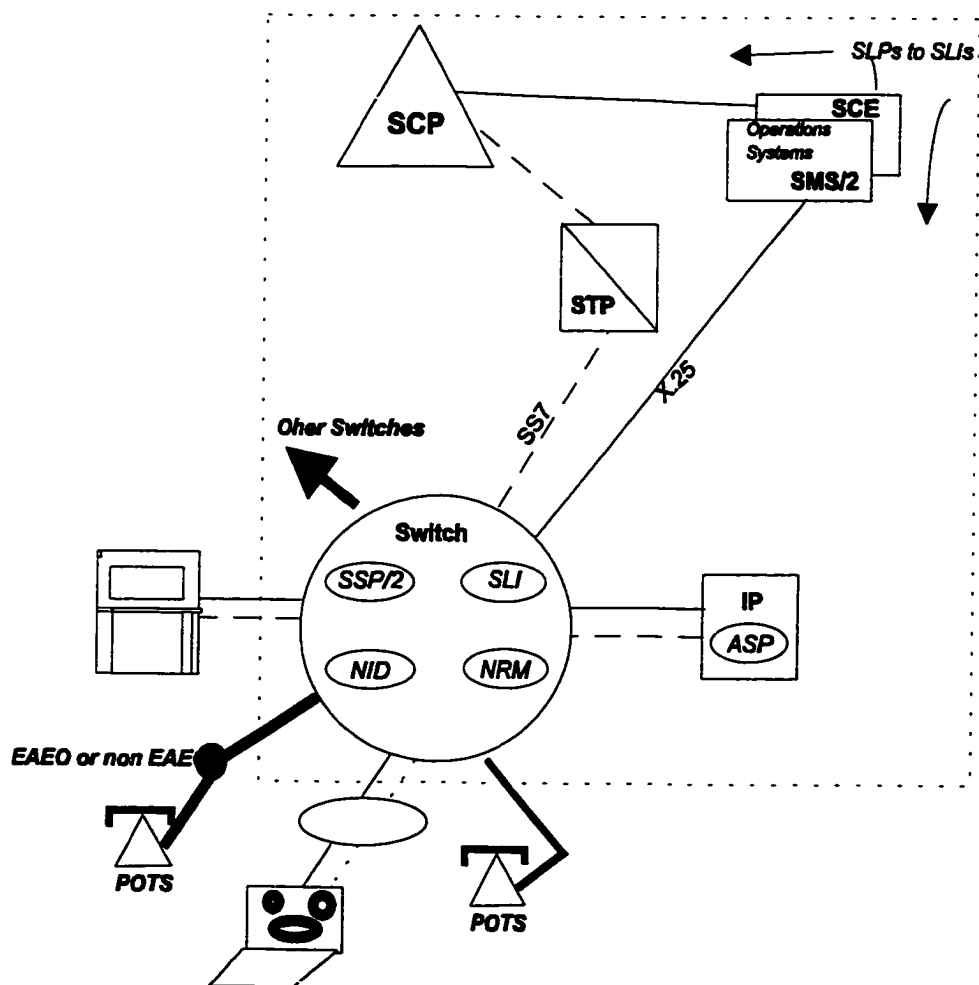
### 3.3.3.

## Intelligent Network/2

IN/2 extends the distributed Intelligent Network/1 architecture to accommodate a concept called service independence. The IN/1 architecture provides new service introduction from a centralized point, allowing users

throughout a wide geographic area to simultaneously have access to the service. The IN/2 concept extends the IN/1 approach and sharply reduces the new service introduction interval; IN/2 applies the premise that multiple network services or service features can be constructed from a set of reusable network capabilities. Service independence result from the reusable property of these capabilities; the capabilities apply to many services, and are therefore independent of a particular service application.

### 3.3.3.1. IN/2 CONCEPT



**Figure 3.11: IN/2 architecture**

In previous architectures, service logic has been localized at individual switching systems. The IN/2 architecture provides flexibility in the placement of service logic, requiring the use of advanced techniques to manage the distribution of both network and data service logic across IN/2 modules [5, 56, 57, 58].

An IN/2 service creation process provides the ability to create new features using reusable network capabilities. A Service Creation Environment (SCE) includes a set of tools that help mechanize the service creation process. Service creation and its associated SCE promise to allow the local exchange carrier to prototype, test, and introduce services more rapidly.

### **3.3.3.2. ELEMENTS OF IN/2 ARCHITECTURE**

#### **SERVICE SWITCHING POINT /2**

The SSP/2 resides at either the local end-office or a tandem-based switching system. The SSP/2 recognizes, through its "trigger table", IN/2 call handling requirements. Information from either a calling or called line may be used by the SSP/2 to trigger an IN/2 call processing routine. An SSP/2 launches queries to other IN/2 modules requesting call processing instructions and acts on those instructions.

#### **SERVICE CONTROL POINT /2**

The SCP/2 consists of a transaction processor-based service logic system, databases, and a signaling network interface.

### **SERVICE LOGIC PROGRAMS**

(SLPs) define IN/2 services in terms of Functional Components (FCs), which are elemental network call processing actions that direct internal network call resources to perform specific actions (e.g., collect dialed digits). *FCs are unique to IN/2 architecture.*

### **SERVICE LOGIC INTERPRETER**

The SLI executes SLPs and handles requests and responses exchanged between IN/2 modules.

### **NETWORK INFORMATION DATABASE**

The NID contains information about customer access lines and the network. The NID is queried by other IN/2 modules for information; it does not interpret service logic.

### **NETWORK RESOURCE MANAGER**

The NRM provides a location function that determines the IN/2 module to provide network resources required to continue call processing; the NRM may respond to the SLI with information required to establish a connection to an IN/2 module.

The flexibility of the IN/2 architecture allows the SLI, NID, and NRM to reside at the switching system or the SCP/2; Location of these modules is dedicated by traffic and performance considerations.

The Common Channel Signaling network transports communications between IN/2 modules. The Signal Transfer Point (STP) is a fast packet that provides the routing function for common Channel Signaling messages.

### **INTELLIGENT PERIPHERAL**

The IP provides specialized telecommunications capabilities (e.g., announcements, prompts, voice recognition) required by IN/2 SLPs.

### **ADJUNCT SERVICE POINT**

The ASP typically resides at the IP and responds to SLI requests processing.

### **SERVICE MANAGEMENT SYSTEM/2**

The SMS/2 supports service provisioning and database administration.

### **SERVICE CREATION ENVIRONMENT**

The SCE supports the introduction of network services features, from definition through verification and deployment.

Interface between the local exchange customers and the network is accomplished through a small set of transaction-oriented, standard user-to-network interfaces; both ISDN and traditional (e.g., POTS) access is envisioned.

### **FUNCTIONAL COMPONENT**

An FC is an element network call processing action unique to the IN/2 architecture that directs internal (e.g., collecting dialed digits). FCs can be combined in different ways to create different network services or features (e.g.,

Transfer of Control, Connection Control, Network Participant Interaction, Network Information Management, Processing, and Information Collection).

The definition of service independent FCs ensures that the IN/2 architecture will not be limited by the characteristics of a particular service, or by a particular service implementation. The ability to use the same FC in the construction of multiple network service or service features results in the service independence of the IN/2 architecture.

### **3.3.3.3. IN/2 CAPABILITIES**

The IN/2 evolution begun with an initial step, IN/1+, that established the initial platform required for full realization of the service independent concepts embodied by IN/2. IN/1+ adds service-independent capabilities for circuit-switched applications, primarily voiced based, to the IN/1 architecture. IN/1+ FCs and capabilities represent a subset of those defined for IN/2. The IN/1+ architecture has been extended to attain full IN/2 flexibility by adding capabilities for packet-switched data and multimedia applications.

Since IN/2 architecture is built upon standard network capabilities, rather than specific services, it provides a similar platform for control in emerging technologies such as Broadband Integrated Service Digital Network (BISDN) and Next Generation Switch.

### 3.4.

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## ADVANCED INTELLIGENT NETWORKS

The PSTN providers (e.g., local exchange carriers (LEC in US)) are deploying or planning their own IN-based services. Dialed number services (e.g., 800) are characterized by their ubiquitous access and use of a dial number to signify a special service. Virtual business group services (e.g., private virtual networks (PVN) and multi-location business group (MBG) services) are based on establishing a virtual business group within the PSTN. They aim to offer business customer feature-transparent operation for their private network features. Billing and account services (for example, alternate billing services based on line information database (LIBD) applications and calling card services) establish alternate billing for calls. Personal mobility services (e.g., personal number service or follow me service) enable a caller to reach subscribers even if they are traveling. Televoting use IN to count the calls made to a specific number. Terminating service (e.g., Local Area Signaling Service (LASS)) are based on incoming call information, and are designed to provide call-related information, improved screening, and call routing to the users. Originating-based services (e.g., area wide calling service) use the calling party's number to decide the routing of the call.

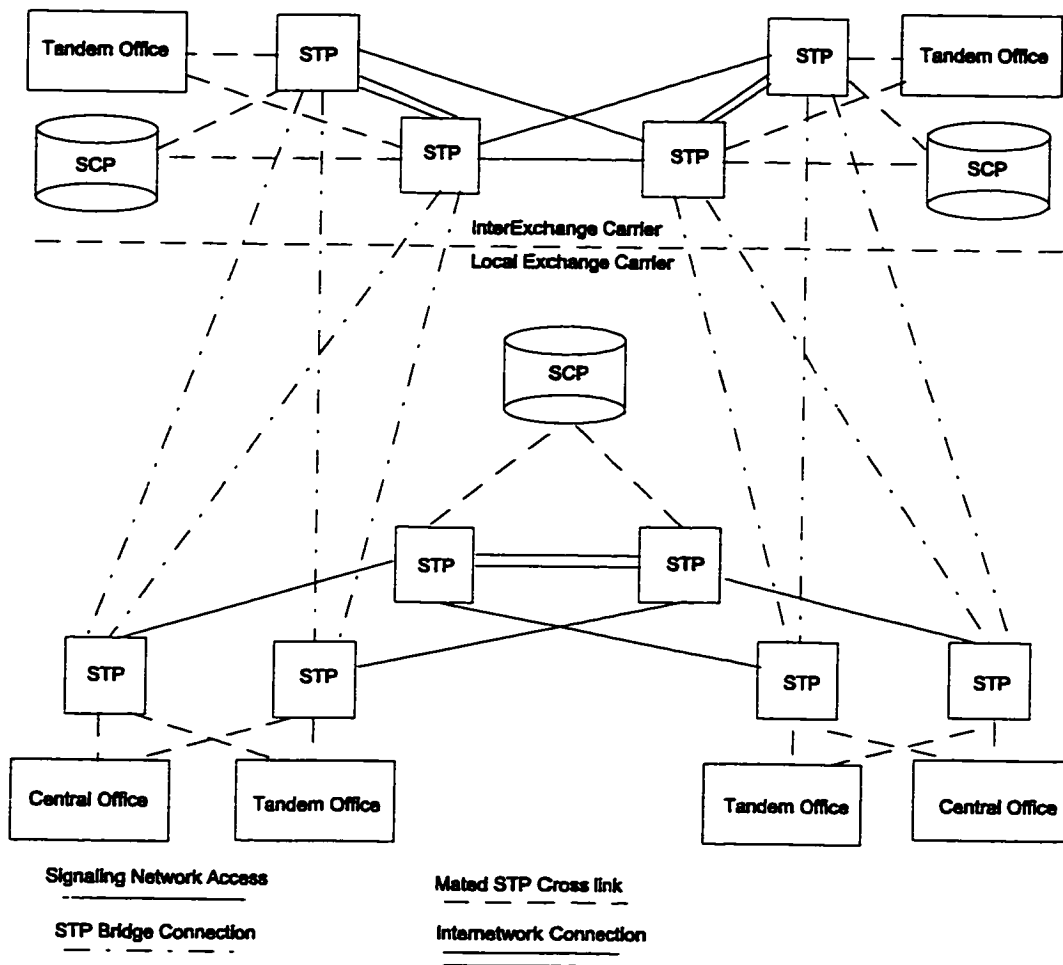
To realize this kind of services, sophisticated management capability is necessary. So is the movement to the universal information services provided by

UISN integrated into an Integral part of the Advanced Intelligent Network (AIN) services[32].

### 3.4.1.

## Architecture of AIN

From the operations point of view, the major components of an American IN include the SSP, Network Access Point (NAP), SS7 network, Service Logic Execution Environment (SLEE), SMS, and Operating Systems [51].



**Figure 3.12: The SS7 Signaling network architecture is a duplicated network with multiple STPs and SSPs**

The SS7 Network is critical to the SCP/SSP IN architecture. It transports signaling information as message packets between network elements. These packets are switched through STP that form the backbone of the SS7 network [47, 5]. The most salient SS7 network function for IN is the transport of queries and responses between SCPs and SSPs. The call control-specific data in the query and response are encoded in the transaction capabilities application part (TCAP) of either message.

The data that the SS7 work needs to route each message are encoded in the signaling connection control part (SCCP) and message transfer part (MTP) areas of the messages. A service switching point, to pick the correct routing, relies on the customer-specific "global title" in the query's signaling connection control part. Thus, the SSP sends the query to an STP that translates the global title to the SS7 network address ("code Point") of the right SCP, and to a "subsystem number" that corresponds to the right service application process within the SCP. The STP then forward the query to the SCP. When it receives the query, the SCP routes it to the process shown by the encoded subsystem number, creates a response, addresses it to the SSP's point code (obtained from the query), and sends the response to the SSP via one or more STPs. These signal transfer points route the response directly (i.e., without global title translation), based entirely on the SSP code encoded in the message transfer part of the message.

The SMS system performs service management functions in A-I-Net release 0, including service creation, network management, support for call processing in the SCP, and support of OAM&P functions.

The characteristics of IN are: 1) distributed intelligence in multiple network elements, 2) tight coupling between network elements, 3) the ability to introduce new software on existing network elements, and 4) standard interfaces.

### **3.4.2.**

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## **Intelligent Network Platforms in the USA**

The LECs see intelligent-networking products as a strategic need for building new services and becoming more responsive to their subscribers. Such products support a distributed- processing environment, which gives the LEC more flexibility in providing services. at the same time, these products provide open interfaces between network elements, which supports the LECs objective for a multivendor environment.

To respond to these needs, AT&T has introduced its AI-Net advanced-service-platform products as Release 0 of Bellcore's AIN concept. AIN Release 1 is being considered to make available more advanced AINs service-to-service, business, and mobile wireless customers.

### **3.4.2.1. AIN RELEASE 0**

Is the initial offering for the A-I-Net advanced services platform. It consists of the service switching point (SSP) and Network -Access-Point (NAP) feature

packages for the 5ESS switch and the NAP feature package for the 1A ESS switch. These products support open, standard interfaces -such as Signaling System 7 (SS7)- that other vendors can interface to competitive database products. The advanced service platform supports the intelligent-networks concept and has been integrated into the existing infrastructure of switch-based features and switch-based operations, administration, maintenance, and provisioning<sup>2</sup> (OAM&P) functions. Because of this internetworking, the LECs can minimize the use of their previous investment in network-switching hardware and software, while offering new services through the centralized database interface. The advanced services platform can evolve to include more capabilities -such as service-circuit nodes (SCNs), intelligent peripherals, and new switch-based features that expand the service opportunities .

The intelligent-network architecture of AIN release 0 (Figure 3.13) consists of:

- The SSP feature package for the 5 ESS switch (i.e., AT&T's A-I-Net advanced services platform, or ASP).
- A centralized database (the SCP)
- A signaling interface between the ASP and the SCP. This interface uses the transaction-capabilities application part (TCAP) of SS7 messages.
- The A-I-Net service-management system (SMS) that, along with the A-I-Net service-creation environment (SCE) provides a user interface for the ASP SCP.

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<sup>2</sup> Setup process for a switch in which parameters are initialized to operate the switch and its features.

Services can be built from ASP components, SCP command scripts, and existing 5 ESS switch-based features. Subscribers can be located on either an SSP or a NAP.

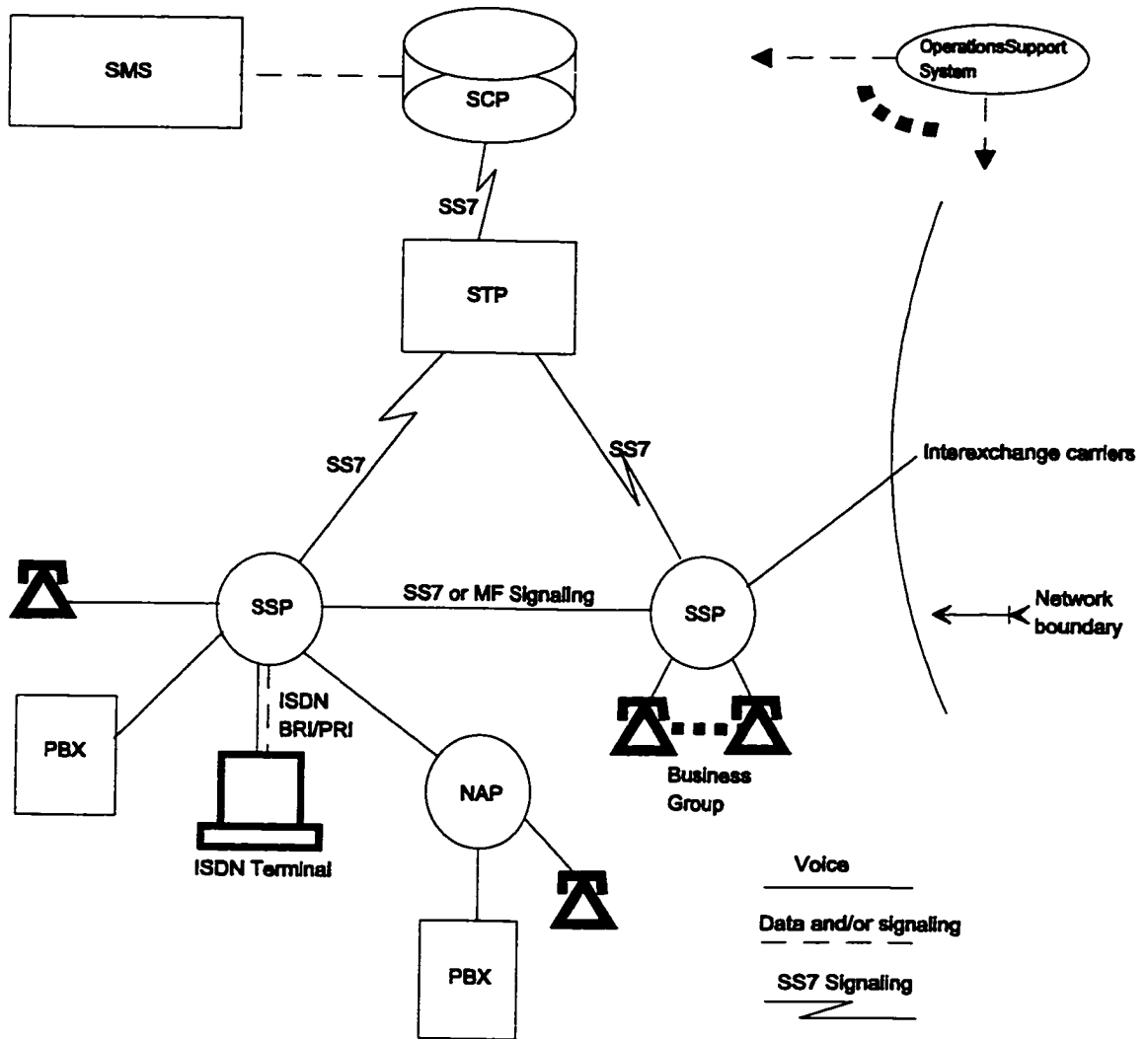
In a typical call scenario through the ASP architecture, a subscriber initiates a call to the network and, thus, activates the ASP feature. When the SSP detects the line or trunk seizure (depending on the type of arrangement), it suspends call processing and launches a query through the SS7 network to the SCP. The network's signal-transfer points (STPs) will route the query to the proper SCP, according to the service to be provided. For example, the SCPs that control routing for 800-number calls provide different functionality than the ASP does.

At the SCP, the service application will locate the proper service logic node determine proper call handling. The SCP will then formulate a reply to the SSP, sending it commands to route or terminate the call, ask for additional information, play an announcement to the caller, etc. The SCP will also send billing information to the SSP.

The NAP feature package, which can be on either a 5 ESS or 1A ESS switch, provides access to ASP features in switches that may not support an SS7 interface to the SCP. Thus, the NAP feature package allows greater geographic coverage without the need for direct SS7 connections to an SCP.

When the NAP detects a call that requires ASP services, it forward the call to the SSP, which can interface with the SCP for further call handling [46]. The SCP's

typical responses are similar to those for calls that originate on the SSP. That is, it can play an announcement, collect more caller input, route the call, block the call, etc.



**Figure 3.13: Architecture of AT&T's A-I-Net advanced services platform (ASP) (Release 0).**

Figure 3.13 shows the ASP, an enhanced set of capabilities, provides either SSP or NAP functionality on the ESS switch. These capabilities are generic, service-

independent building blocks that allow the switch to respond to commands sent from the SCP, enabling the LECs to provide a wide variety of new services

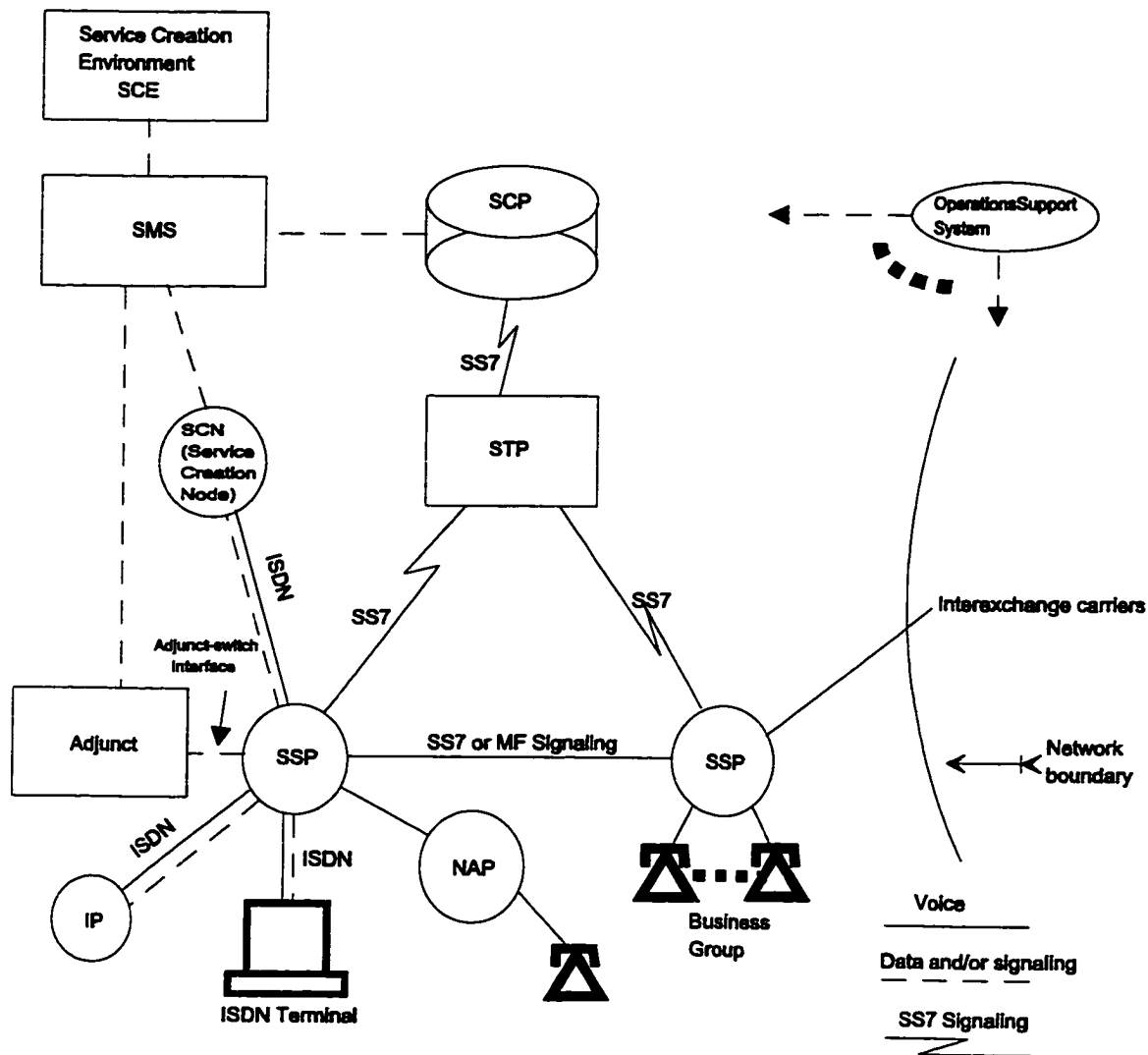
### **3.4.2.2. AIN RELEASE 1**

The AIN Release 1 concept is the next-generation, intelligent-network architecture proposed by the Regional Holding Companies and Bellcore. Because AIN Release 1 expands the interface between switch call processing and an adjunct processor, it will provide opportunities for an even wider variety of services. The expanded interface permits this remote processor to control call processing at many more points in the call than allowed with the ASP feature; thus, customer-specific, line-based features are feasible.

The AIN Release 1 is considered as to be the next generation of intelligent-network products. It is based on Bellcore's AIN architecture proposals and US West's adjunct-call-processing architecture. The planned architecture for AIN Release 1 is shown on figure 3.14. The intelligent-network architecture presented in release 0 is extended by new switch capabilities (which replaces the SSP) and the inclusion of an adjunct, the A-I-Net service-circuit node, and intelligent peripheral elements. As it does in the ASP, the NAP detects certain triggers and routes the call to the AIN-switch capabilities for further processing [46]..

The term AIN-switch capabilities replaces the term SSP used in the A-I-Net advanced services platform, reflecting the additional capabilities and functions that the release 1 switch provides. These include a more extensive call model, additional triggers and subsequent events, resources-status checking, and

switch-data management. As in the ASP, the NAP continues to detect certain triggers and can route the call to the AIN switch for further intelligent-network call processing.



**Figure 3.14: Release 1 network.**

Three network elements that can influence call processing at the AIN switch complement the ASP SCP:

- ⇒ *Adjunct*. The adjunct's function is similar to the SCP's. Both contain the same functional entity, the service-logic execution environment (SLEE); and both use the same application-level messages to communicate with the AIN switch. However, the adjunct is connected directly to the AIN switch, while the SCP connects via a signal transfer point. Thus the adjunct is better suited to services that require an extensive exchange of messages with the AIN switch, and it better addresses services that are geographically localized. The SCP is better suited for services that require centralized data.
- ⇒ A-I-Net service-circuit node. The SCN contain a switch fabric<sup>3</sup> that allows for the termination of bearer channels and direct support of service circuits. The A-I-Net service-circuit node can support such specialized resources as announcements, voice recognition, voice synthesis, and store and forward transmission. In addition, it supports a service-node execution environment that manages and controls its resources.
- ⇒ Intelligent peripherals. The IP provides additional resources that can be used by the SCP or the adjunct. The IP's resource-control execution environment controls and manages its resources. Like the SCN, the intelligent peripheral is connected to the AIN switch via ISDN. Nut unlike the SCN, the intelligent peripheral doesn't support service-creation capabilities.

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<sup>3</sup> Switch fabric refers to the hardware and circuits that allow voice calls to be connected to the SCN. It provides "talking" path for the caller to receive announcements from the SCN.

The SCE allows the development of new services that take advantage of the capabilities offered by the AIN-switch architecture. The programs that the SCE develops and tests can be sent to the SMS for distribution to the appropriate network elements, or they can be sent directly to the SCN. As in the A\_I\_Net advanced-services-platform architecture, the A-I-Net service-management system is also responsible for service administration and coordination.

Finally, updated operations-support systems will support the new AIN-switch functionality. A service that involves multiple network elements requires sophisticated testing, maintenance, and network functioning and allow troubleshooting to address customer complaint quickly.

### **3.4.3.**

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## **Evolution for International Applications**

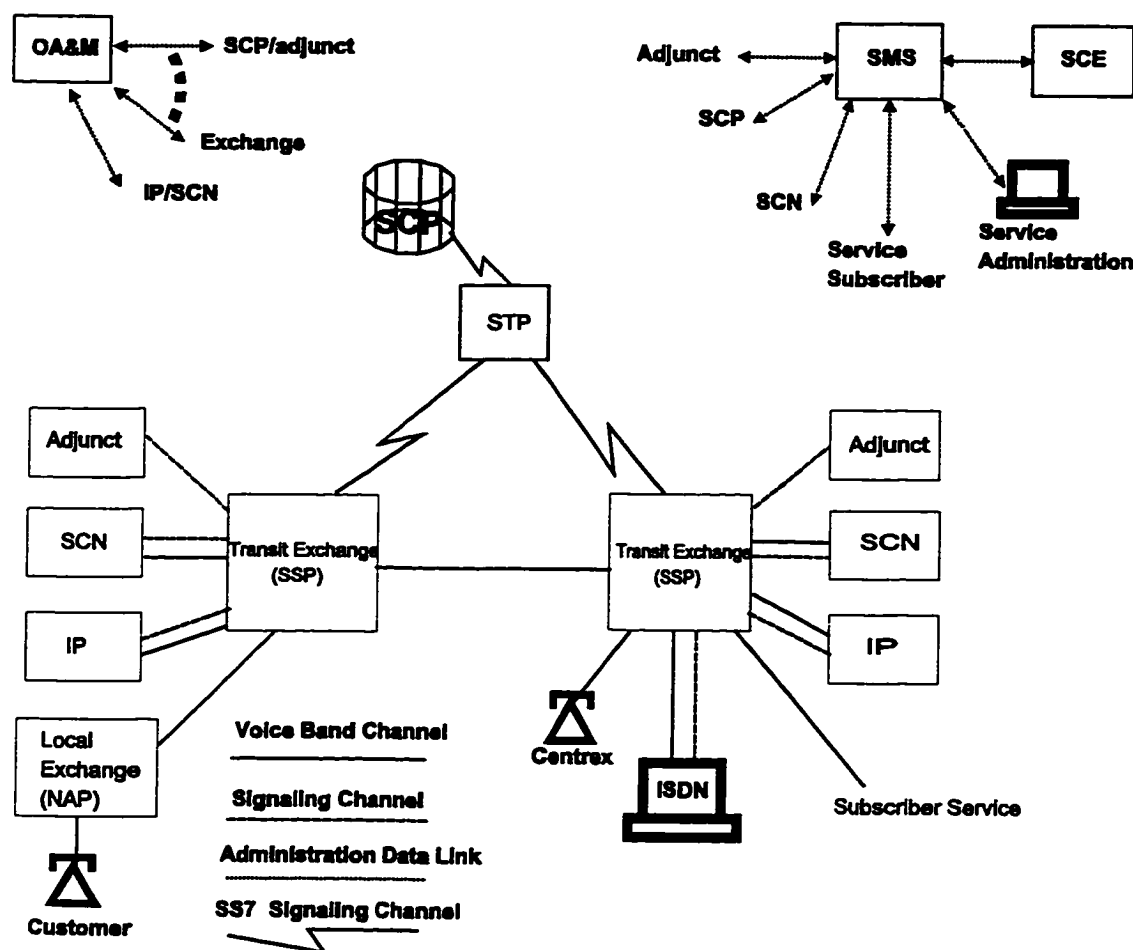
IN is a major investment for telephone administrations, Therefore, flexibility in the evolution of IN capabilities is a principal concern. The IN architecture should allow new hardware and software technologies to be gradually introduced. But the new technologies must add value to IN elements via lower cost, improved performance, and increased throughput. The architecture should be flexible enough to allow the telephone administration to expand the hardware and software with relative ease.

New network elements are being designed for a global network (see figure 3.15).

With enhancement for SSPs, SCPs, and IPs, these new components should give administrations additional capabilities tailored to the customers' needs. These include:

- *Service Circuit Node (SCN)*: provides service logic processing and call termination for service assistance functions such as digit collection and announcements, and for complex services such as FAX store-and-forward. Service Circuit Nodes typically are used to distribute intelligence through the network. Some of the service control and specialized resource functions are combined to provide the service circuit calls. Because SCNs can terminate calls, they connect to the serving switching points using ISDN interfaces.
- *Adjuncts*: provide local service logic functions within the local exchange area. Adjuncts, which contain service logic and local customer data, have capabilities similar to service control points, but at a local level. Adjuncts are accessed from the service switching points through the same trigger points. Using a high-speed data-link, the SSP requests instructions from the adjuncts in message form. The adjuncts would most likely support local, transaction-intensive services (e.g., Centrex services) as opposed to network-routing services, which are typically supported by the service control points.
- *Service Creation Environments (SCE)*: provide flexible service creation and modification capabilities for service providers and subscribers. It enables network and service providers to create new revenue-generating services that are independent of equipment vendors' deployment schedules. Many

administrations are asking vendors of IN equipment to provide them with service creation environment capabilities.



**Figure 3.15: Global IN architecture**

The OA&M capabilities are essential to keep IN elements running smoothly so service quality can be maintained and controlled. OA&M also coordinates the functioning of the IN elements among themselves and with the public network. Managing an IN telecommunications network introduces a greater level of OA&M complexity.

Besides the standard OA&M procedures applied to the PSTN, the IN OA&M capability must provide additional capabilities for network-wide management, traffic measurements, and diagnostic functions. These functions are unique to IN because they must be closely coordinated among several IN elements to permit management and control. Different controls are applied to the PSTN because they could involve only one or two network elements.

IN's OA&M functions should also provide tools to identify high-demand services, services that cause overload from inadequate facilities, and services not in great demand. The administration can use these tools to relieve potential bottlenecks that could degrade call-completion rates. The capability can also be used to market and tariff new services.

#### **3.4.4.**

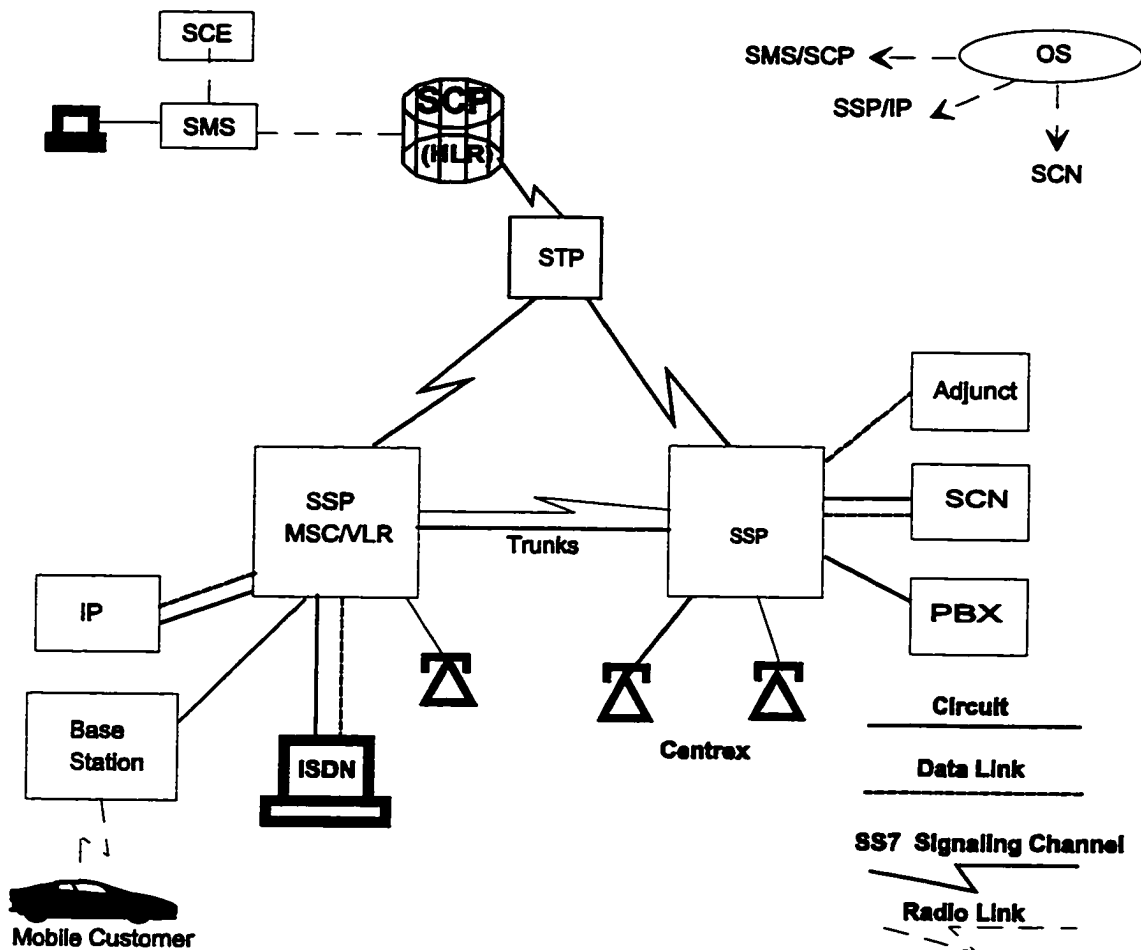
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### **AIN and PCS**

The Personal Communications network, which provides users with wireless access, has been defined by Group Special Mobile (GSM) in ETSI and by TIA/IS41 in EIA/TIA in the United States.

The GSM architecture consists of a home location register (HLR), the equipment identifier register (EIR), and the authentication center (AUC), all of which can be maintained at a central node in the network. The central node's master file contains the wireless customers' records. Local records for wireless customers

are maintained at the visitor location register (VLR). The VLR information can be located in the mobile switching center (MSC), or in the adjunct.



**Figure 3.16: PCS application supported by an IN.** The service control point maintains the home location register, while the local exchange, serving as a service switching point, maintains the mobile switching center and visitor location register. Using wireless or wire-line subscribers have access to the IN and personal communications services.

Figure 3.16 shows, mobility service, also called personal communications services, have strong synergies with the evolving IN architecture. An IN control structure offers a robust platform to support PCS applications. Many of the conceptual model requirements for the IN apply to PCS. Integrating PCS services

on an IN platform potentially reduces an administration's operations, maintenance, and training costs.

From an architectural perspective, we can view wireless access as a technology (such as ISDN or BISDN) that service subscribers can use to access the network. This network can be fully integrated with local exchanges or provided as an overlay architecture.

The IN can flexibly separate call and connection control point from the underlying access infrastructures. As such, an IN platform also can support PCS applications. From a network entity viewpoint, the network access function is conceptual similar to a base station system in the mobile communications world. The service switching functionality could be implemented in a mobile switching center, allowing that center to interact with service control residing in other network elements.

### **3.5.**

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## **WIRELESS COMMUNICATION**

Wireless subscribers are becoming increasingly accustomed to using wireless phones, more mobile and requiring more services as well as the same functionality as what they are using at landline phones.

In, the fast-growing world of wireless networks, IN platforms designed according to these principals are now being applied to two fundamentals needs:

1) The independence from physical network configuration and geography

inherent in the IN has made it a natural platform to support the basic mobility functions required in wireless mobile networks; and 2) Just as in its earlier application to wireline networks, the IN's ability to support the rapid development and deployment of differentiating services makes it a necessary weapon in the arsenal of wireless carriers operating in highly competitive environments.

Wireless Multimedia Intelligent Network, may be offered by using ATM backbone networks.

### **3.6.1.**

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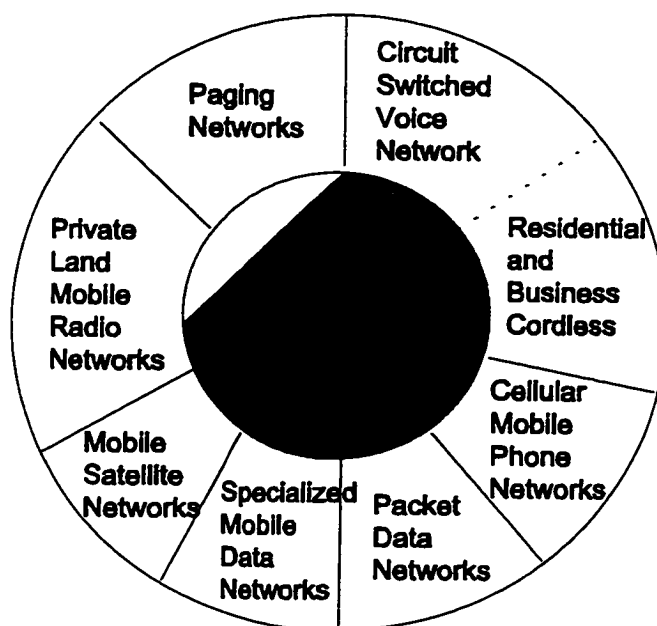
## **Overview of Wireless Communications**

Wireless networks customers are portable in-premises users, mobile on-the-street users, fixed location indoor/outdoor users, commuters, and globe trotters. The devices, interfaces, and characteristics of networks are optimized to suit the particular application.

Three generations of wireless systems evolution. First, cellular phones with a roaming capability, a residential cordless phones covering a limited area, and the analog FM voice technology. Second, the cellular phones bringing an automatic roaming, cordless phones expanding the coverage and access capabilities, and digital voice technology. Third, cellular, cordless & paging converged to person-to-person communication, integrated digital wireless access to services provided

by an intelligent telecommunications network, and personalized telecommunications number to identify a user.

costs and generate revenue.



**Restricted environment for users**  
**Variety of different user interfaces**  
**User service flexibility constrained by the local service access distribution network**

**Figure 3.17: Overview of Communications Environment**

With this evolution, both the subscriber and the provider become more and more demanding. The subscribers expects a Mobile service that's available while in motion and in multiple environments (cars, within buildings, in cities), telephone-type services such as voice, data, fax transmission, custom-calling features, telephony quality service with a clear voice quality, calls free of interference, unblocked access, reasonable cost, and a secure transmission, and an accessible service supporting a large numbers of people available anytime

and anywhere. The service providers must administer a subscriber base such as sign-on new subscribers and maintain subscribers records, administer a facilities network by interconnecting radio and land networks, managing radio/air interface, and providing continuous service, anticipate and plan for

- 1) maintenance (self-diagnostic, fault tolerant, easily reconfigured networks),
- 2) reliability (offering downtime, repairable without disrupting service),
- 3) growth (flexible equipment that upgrades easily and accommodates new technology),
- 4) ways to reduce operating

### **3.6.2.**

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## **Basic Cellular Principals**

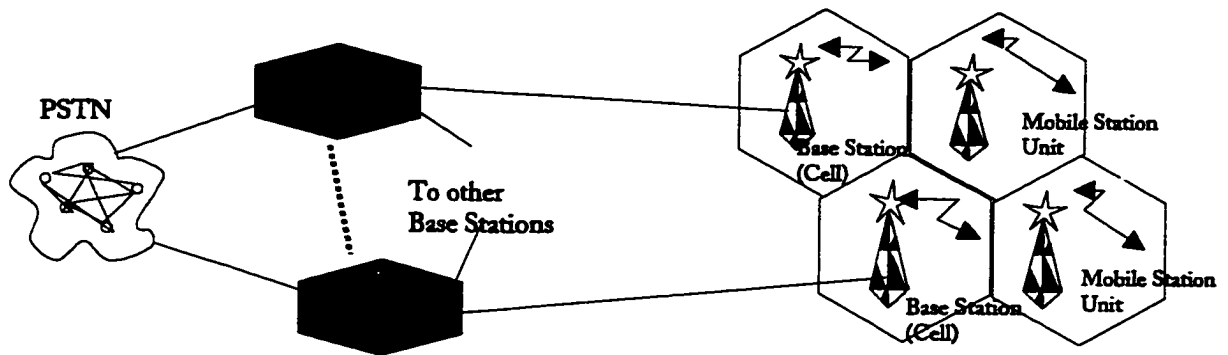
The mobile telephone system uses hexagonal “honeycomb” structure of cells, with a base station at the center of each cell which gives radio coverage to that cell and connects into the public telephone network (figure 3.18).

The hexagonal cell pattern arises from the best method of covering a given area, remembering that radio coverage is ideally radial in nature. three possible methods of covering a particular area: regular hexagon zoning, regular square zoning, and regular triangle zoning.

The area coverage of a hexagon, for example, can be calculated as :

$$\begin{aligned} \text{area} &= 6 * (\text{area of a regular triangle of base } r) \\ &= 6 * (2 * r / 2 * r / 2 * \cos 60) \end{aligned}$$

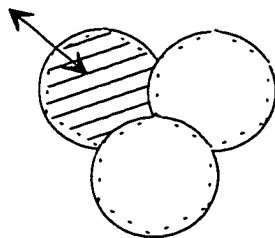
$$=3 \cdot r^2 \cdot \cos 60 = 2.6r^2$$



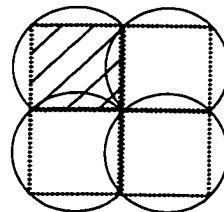
- Mobile-service Switching Center (MSC)**
- Connecting BS to PSTN
  - Performs typical call related functions of an end office
  - Performs handoff
  - Performs a paging function in support of call delivery
- Base Station (BS)**
- The location of radio equipment
  - Manages the radio channel resources
- Mobile Station (MS)**
- Two-way radio

**Figure 3.18: Architecture of Cellular Network**

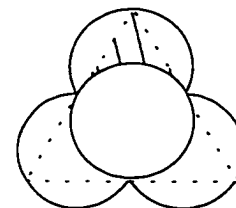
**Unit Zone/Cell Coverage**



a. Regular hexagon



b. Regular square zoning



c. Regular square zoning

**Figure 3.19: Cell Structure Possibilities**

By using basic algebra, we can calculate the center-to-center distance, unit zone coverage, area of overlap, width of overlap, and minimum number of

frequencies for each of the cells type. The area of overlap should be calculated for a completely surrounded cell. Table 3.2 contains the results.

Cell type	Center-center distance	Unit zone coverage	Area of overlap	Width of overlap	Min. number of frequencies
Triangle	$r$	$\cong 1.3 r^2$	$\cong 3.7 r^2$	$r$	6
Square	$r\sqrt{2}$	$2 r^2$	$\cong 2.3 r^2$	$0.59 r$	4
Hexagon	$r\sqrt{3}$	$\cong 2.6 r^2$	$\cong 1.1 r^2$	$0.27 r$	3

**Table 3.2: Cell characteristics for the three main cell types**

We can observe that the area of coverage of a hexagon is twice that of a triangle, with a square midway between the two. The hexagon has a small overlap compared to the triangle. To cover an area of three hexagonal cells, or  $7.8 r^2$ , would require six triangle or four square cells.

It become obvious that the regular hexagon is the most widely used structure, with the triangle suitable in only difficult propagation areas which require deep overlapping of radio zones.

As the subscriber moves (roams) between cells, the communication with the base station of the departing cell ceases and communication with the base station of the entering cell commences. This process is known as handoff (handover in Europe). each adjacent base station transmits a frequency that is different from its neighbor. the handoff is accomplished when the received signal from the base station is low enough to exceed a predetermined threshold. At the border between two cells the subscriber is under the influence of two or even three base stations, and the link could pass back and forth between base stations as the

moving subscriber receiver experiences a fluctuating field strength depending upon the immediate environment, such as being surrounded by tall buildings.

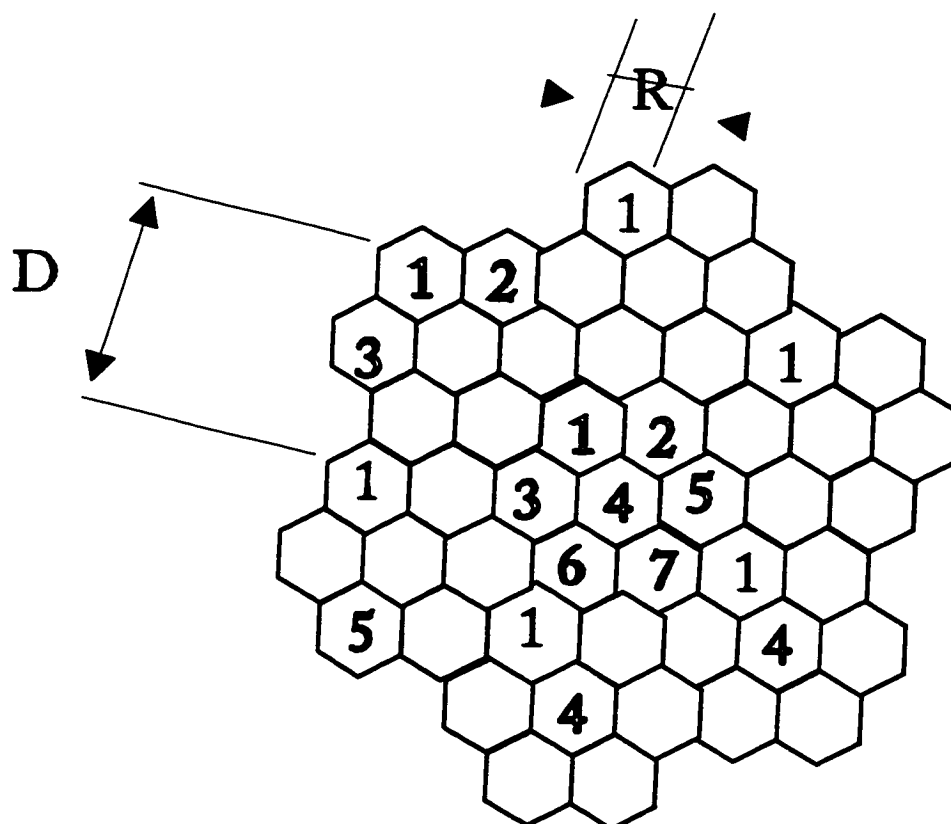
The nominal cell diameter also varies depending on the traffic density. Typically, the center of a city is the most populated, with the suburbs gradually decreasing in population. This leads to the cells in the center having a small diameter with a gradual increase in diameter when moving outward.

To enable the available bandwidth to be used efficiently and thereby increase the number of users, a frequency reuse mechanism is built into the cellular structure. In the diagram in figure 3.20, the cells are clustered into groups of seven, each group having the same pattern of seven base station frequencies. The distance between different base stations using the same frequency is  $D (=4.6 \cdot \text{cell Radius})$ . A considerable effort is made to check that signal strength contours for each cell within the hole area of the system are not distorted to the point which would allow two calls to interfere with each other. Each cell can use only one-seventh of the channels available within the system.

number of channels per cell =  $\frac{\text{total number of channels per repeat pattern}}{\text{number of cells (base stations) within the repeat pattern}}$ .

A cellular structure with smaller and smaller cells is evolving for two main reasons: first, the increasingly limited available power which can be transmitted by the smaller and smaller mobile subscriber telephone sets, and second, increased capacity. can be enhanced even further by cell sectorization. As cell sizes

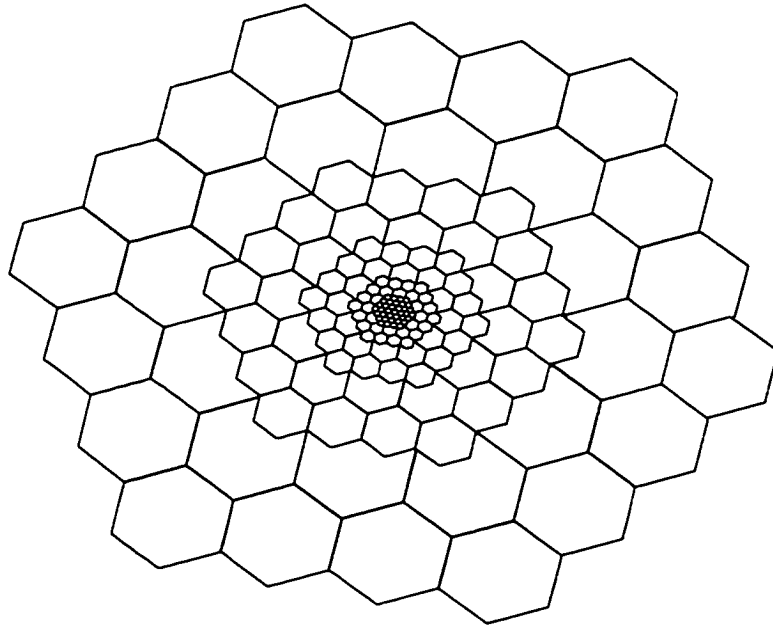
decrease, the distance between base stations of identical frequency also decrease. This can offset by the careful control of the power radiated from either the base station or the mobile station.



**Figure 3.20: Cell Structure using Hexagonal Zoning with frequency mechanism.**

For a mobile subscriber to receive a call, the subscriber's precise location must obviously be known. There are several possible ways to track the movement of a mobile station. A convenient method is to split up the whole cellular network into a number of location areas, each having its own ID number. Each base station within a particular area periodically transmits its area ID number as part

of its system control information. As the mobile station moves from the of a base station to another, eventually it will move to a new ID number region, and the network is updated with the new area in which it can be found. This is all done using signaling.



**Figure 3.21: Reduction of Cell Sizes to serve increasingly congested Area. The allocated bandwidth is used repeatedly in the cellular structures that overlay the serving area. Thus the capacity increases greatly especially if the cells are also designed as Pico cells.**

The basic cellular principles :

- Fixed spectral allocation limits the number of channels that can be used, but users continue to increase.
- Channels must be reused through a service area, but users should not interfere with each other.
- Signals attenuate with distance so channel can reused.

- Low transmitter power allows nearby channel reuse so more subscribers can be serviced in a given area.
- Lots of small, low-power cells can be used to support many users within a given service area.

Network reconfiguration occurs dynamically during calls (handoff).

### **3.6.3.**

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## **Wireless intelligent networks**

Services providers need to come up with quality service and wide area coverage at fair price, then they'll need to control their cost while at the same time introducing new services. Wireless intelligent network (WIN) enhanced service applications can provide the help providers desire to resolve these problems. Equipment vendors must develop products capable of handling the fast-paced, ever-changing requirements their customers, the service providers, need. To do so, researchers must come up with a programmable switch in the network configuration that can compensate for the lack of call models and triggers in many wireless networks, and a platform that provides multiple applications.

The WIN solution offers service providers an advantage because it:

- Supports a wide set of service applications,
- Supplies customized management of mobility and enhanced services,
- Simplifies database and service administration,
- Provides ubiquitous services,

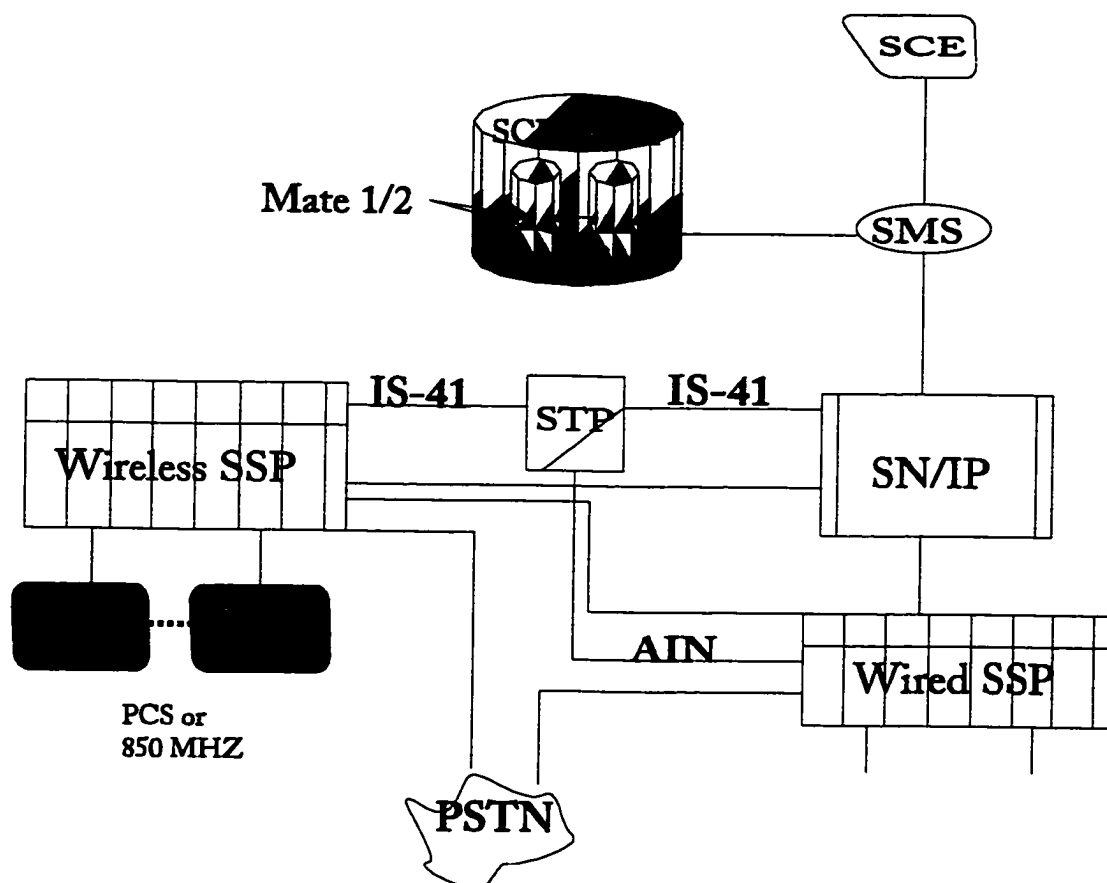
- Helps get new services to market more quickly,
- Presents an efficient test-bed for new services and features, and
- Leverages utilization of subscriber data.

### **3.6.3.1. WIN ARCHITECTURE**

The WIN is based on an architecture that separates call processing from enhanced features functionality (see Figure 2.22). The mobile switching center (MSC) suspends control of call processing when it encounters a trigger, passing control to the WIN network element. This element manages the intelligence to provide the enhanced service.

Lucent's WIN vision includes a single logical SLHR on a mated-pair of service control point (SCPs), other SCP's providing enhanced database services, and a service node (SN), all maintained, provisioned, and administered via a single logical service management system (SMS). Services for the SCP and SN are created and deployed using one service creation environment (SCE). SCP and SN are modular with high level of programmability to ensure the rapid and economical introduction of new services.

The home location register (HLR) is a specialized database storing the profile of wireless subscribers. The HLR is found in one of two places in the wireless network -either collocated in the MSC or as independent database element. If separated from the MSC, it's named SHLR



AIN- Advanced Intelligent Network  
 IS- Interim standard  
 PCS- Personal Communication services  
 PSTN- Public Switched Telephone Network  
 SCE- Service Creation Environment  
 SCP- Service Control Point

SHLR- Standalone Home Location Register  
 SMS- Service Management System  
 SN/IP- Service Node/Intelligent Peripheral  
 SSP- Service Switching Point  
 STP- Signal Transfer Point

**Figure 3.21: WIN architecture**

### 3.6.3.2. STANDARDS

The Telecommunications Industry Association (TIA) is developing standards for wireless networks. The TIA subcommittee TR-45, the Wireless Intelligent Network task group, is building on the American National Standard Institute (ANSI)-41 standard protocol with ANSI-41-WIN to address standards issues specific to the WIN. A WIN deployment should be built using network elements

that are open, published interfaces, relying heavily on industry standards (service triggers should be launched from the MSC to the WIN element).

### **3.6.3.3. MOBILITY MANAGEMENT AND TRIGGERS**

Mobility is a concept born from wireless telephony. In the landline networks, the subscriber's phone is always available and is always located in the same geographic area. With wireless networks, a mobile may travel between different serving switches during the course of one phone call. Moreover, a mobile may be turned off or out of range, making the unit unavailable for calls. Mobility was conceived to handle these kinds of unique situations.

Mobility Management includes capabilities related to the HLR used to route calls and forward subscriber profile data. Usually, when a mobile is powered on, the first action it takes is to register with the local MSC. The serving MSC queries the HLR database for that subscriber to learn about the features it will offer to the mobile. If the HLR record is stored on another MSC or on an HLR, the query occurs over the SS7 network using ANSI-41 protocol. The HLR "remembers" where the mobile registered in order to assist in call delivery to the mobile. Because the HLR is used during call delivery and other mobile services, the HLR is more than just a query/register database. It is critical part of the logic in wireless call processing.

Today, enhanced services are provided using service triggers on the MSC. the triggers are used to direct call control to query for data from the WIN network elements on call origination or dialed number from the mobile and call

termination to the mobile. A similar analogy can be drawn in a landline network. In the wireless networks, because of mobility management, new triggers based on the uniqueness of mobility, known as mobility triggers, will be defined.

Upon registration of the mobile or a change in mobile status (that is, location roaming), a mobility trigger is invoked. Mobility triggers are used for registration and location update functions and are independent.

### **3.6.4.**

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## **WIN Enhanced Services**

### **3.6.4.1. BASIC ARCHITECTURE**

The projected architecture of WIN is basically the same as the one for IN with the capacity of handling mobility. The Basic elements are:

#### **SERVICE CREATION ENVIRONMENT:**

The SCE is the development environment for creating new services and programming the platform software for the SCP and SN. One of the inherent advantages is the ability for the service provider and third parties to develop services independently from the platform Supplier.

#### **SERVICE CONTROL POINT**

Contains a network database that centralizes service logic software and data. It serves as the host platform for the SHLR, authentication center, and over-the-air service provisioning applications.

### **SERVICE MANAGEMENT SYSTEM**

The main task of the SMS is the provisioning of WIN services, where the SMS distributes specific service logic and data into the network elements, for a specific. The SMS administers network elements on which certain service logic or subscriber data is located and can retrieve or update existing logic and data.

### **SERVICE NODE/INTELLIGENT PERIPHERALS**

SN/IP, provides all the functions of an IP and service logic to control an advanced service circuits such as voice announcement, automatic speech recognition, text-to-speech, and fax store-and-forward. These capabilities are used to provide services such as short message service center and voice dialing.

### **SERVICE SWITCHING POINT**

SSP detects events or triggers that signal when an IN call is being placed by separating basic call control from IN-based service control. Upon detecting an IN trigger, the SSP queries the SCP for information needed to route the call.

### **SERVICE TRANSFER POINT**

STP nodes in the SS7 network that route signaling messages between switches and between SSPs and SCPs in the IN.

In order for MSCs to launch intelligent network (IN) services, the MSCs must provide call processing triggers. The wireless standards identify three types of triggers:

- originating triggers: occurs each time a wireless subscriber places a call,
- terminating trigger: occurs just before a call for a subscriber is terminated to the mobile, and
- dialed number trigger: is a from of the originating trigger, and occurs only when the subscriber dials a specific digit string, such as an extension dialing feature.

These triggers are points in the MSC call processing operation when call processing is suspended until a network database or call processing element, such as HLR or another SCP, provides data or instructions to the MSC. The MSC uses information to continue call processing.

The triggers used during call processing are determined on a per subscriber basis. The user profile information in the HLR contains information on the type of triggers provided to the user. When the serving MSC requests the HLR profile to create a record in the visitor location register (VLR), the MSC is informed of the type of triggers to which a user subscribes.

At the current stage, using a service query capability or mobility trigger on an SHLR can help speed up time to market until the trigger service find its way to MSCs provided by multiple vendors.

#### **3.6.4.2. WIN SERVICE EXAMPLE**

WIN services should be provided with high capacity, reliability, and with efficiency that can coexist with already existent standards.

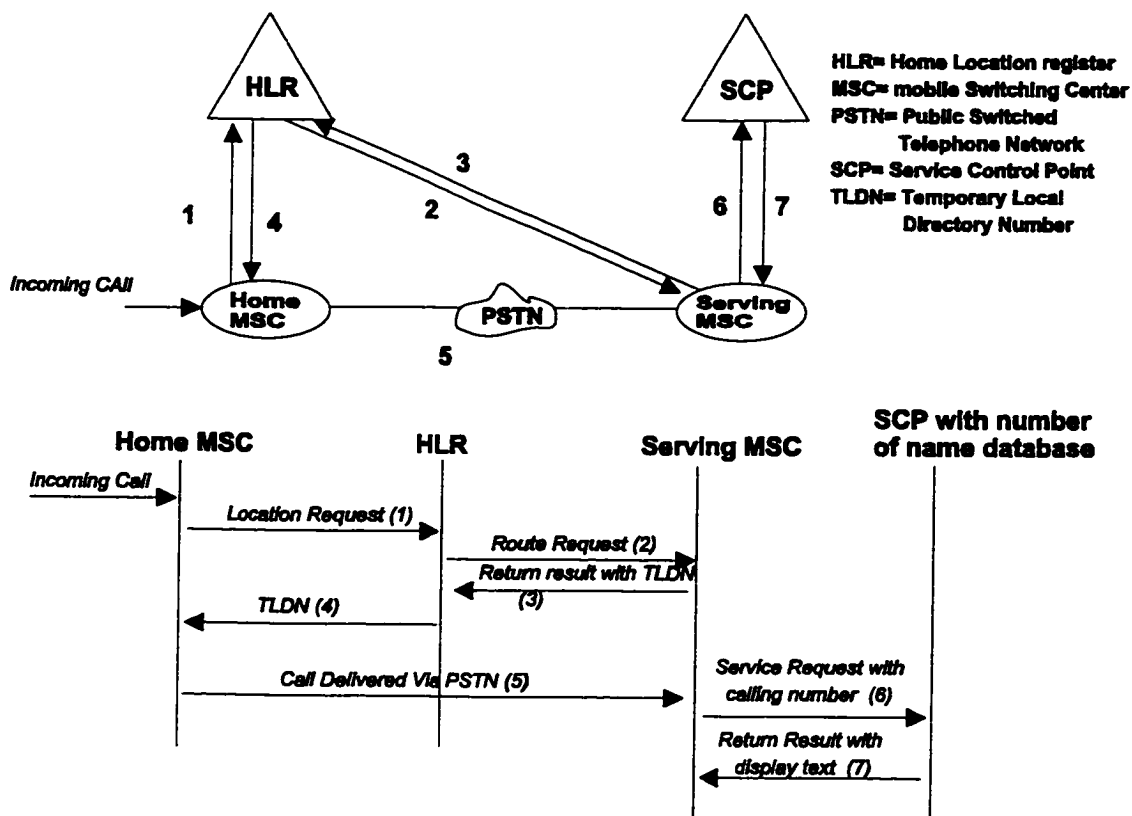
For multi-MSC networks to provide ubiquitous services, it's recommended to use one SHLR to access data from other SCPs.

**CALLING NAME DISPLAY:**

(CNAM) can either be an MSC or SHLR database query. The ANSI-41 standards committee is defining two methods for delivering the calling name of the calling party to the mobile. CNAM has been offered in landline networks for many years. Typically, to deliver the name of the calling party, the switch queries a number-to-name database on an SCP using the calling party's number, which was delivered using signaling interface such as interface such as ISDN user part (ISUP). The standard committee suggests that in a wireless network, the subscriber should have a terminating trigger defined in his or her profile. The MSC will know that the subscriber has the trigger armed, indicating a query of the number-to-name database before completing the call to the mobile. Figure 3.22 shows how the terminating trigger would be used to capture the calling name of the caller.

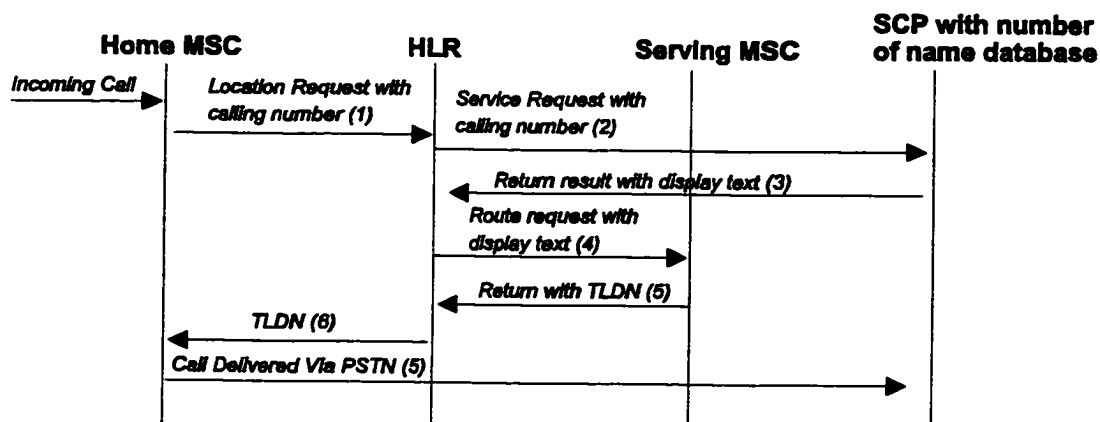
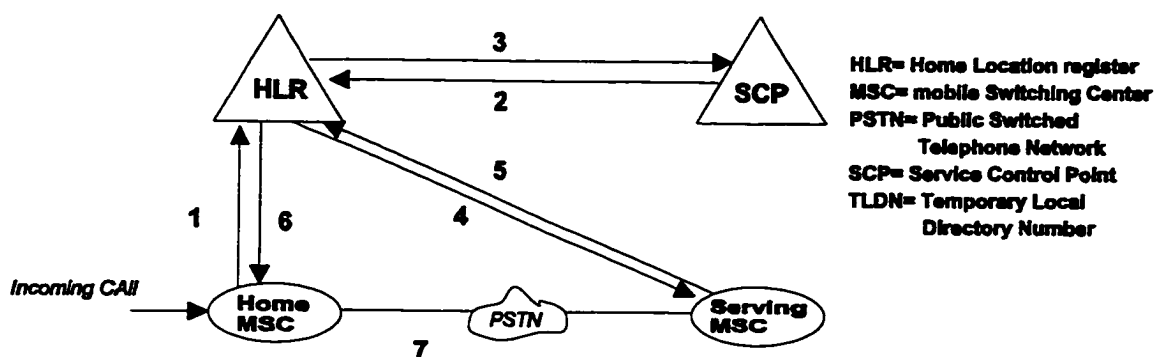
When a call comes in for a wireless subscriber, the subscriber's home MSC receives the call. The home MSC queries the HLR to determine where the mobile is located. The HLR sends a route request to the MSC serving the subscriber and receives from the serving MSC a temporary local directory number (TLDN), which is routed back to the home MSC. The home MSC uses the TLDN to route the call over the public switched telephone network (PSTN) to the serving MSC.

When the call is delivered to the serving MSC, the caller's number is also delivered.



**Figure 3.22: Service Mobile Switching center accessing number to name database**

Noticing that the subscriber has calling name delivery service, the service MSC queries a number-to-name database. The text that is returned to the serving MSC is sent to the mobile for display. This scenario works well if the serving MSC supports terminating triggers and knows how to communicate with the CNAM database. But what if the MSC does not have these capabilities? Then the service would not be available to subscriber.



**Figure 3.23: Standalone home location register providing number-to-name database .**

An alternative way to provide the CNAM service is to make it available on all MSCs by using the query capabilities of the HLR when the MSCs do not provide terminating triggers. This method is also being defined by ANSI-41-WIN standards committee. (Figure 3.23) shows this process. When a call arrives at the home MSC for a subscriber, the home MSC queries the HLR just in the last scenario. This query also includes the number of the calling party. Before requesting the TLDN from the serving MSC, the HLR queries the number-to-name database. Once it has the display text, the HLR sends this text to the serving MSC in the route request message. The serving MSC stores the display

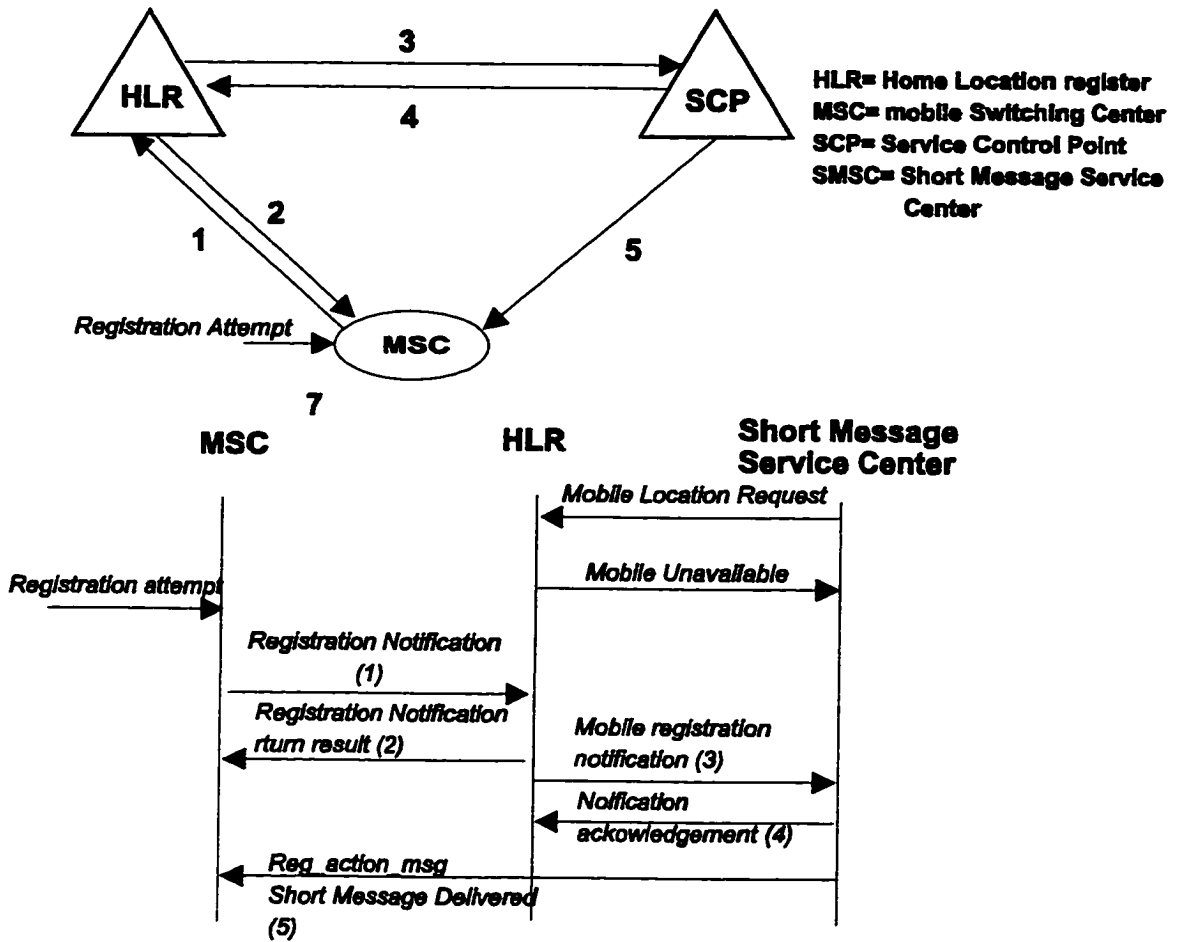
text until the call comes in from the PSTN. When delivering the call to the mobile, the display text is included in the call setup information.

The network operates more efficiently when the serving MSC provides the number-to-name database query because the transaction does not have to take place if the mobile is busy or does not can not receive the call. However, the method of providing an IN service using the HLR to perform the query is a means of providing that service throughout the network with the same look and feel. Then, when new enhancements are desired on the service, the changes need to be made in only one place--the SHLR.

### **INCOMING CALL SCREENING**

Another example that uses the SHLR to perform the service query is the *ICS application*. Figure 3.24 is a diagram of this service.

The purpose of this service is to deny the delivery of a call to the mobile if the number of the calling party is not on an approved list. When a call arrives at the home MSC, the home MSC queries the HLR for the location of the mobile, as described earlier. The query to the HLR contains the calling party's number. The HLR, noting that this subscriber has the ICS feature, queries the SCP database containing the approved list. The SCP replies back to the HLR if the number is on the approved list. The HLR then continues processing the call by sending a route request to the serving MSC as described earlier or halting the processing of the call.



**Figure 3.24: Notifying a short message service center of a registration**

**3.6.5.**

**Wireless And ATM Networks**

In 1992, the future generation of wireless technology was discussed, and the Wireless Asynchronous Transfer Mode (WTAM) network's concept was considered ubiquitously. WATM is motivated by the increasing importance of portable computing applications in business, consumer and industrial

applications. As multimedia applications migrate to portable devices, wireless extensions to existing broadband networks will be required to support the seamless delivery of integrated voice, video, and data with very high quality. In this context WATM is intended to be a direct extension of the wired ATM network with uniformity of end-to-end QoS guarantees.

While wireless communication is experiencing fast evolution, the fixed network has been going toward B-ISDN with ATM concept. ATM will offer data rates that are considerably higher than current fixed network services. Internetworking with ATM will set extremely hard requirements on the wireless air interface, but continued development in technology will enable the industry to manufacture smaller and less power consuming terminals with increased performance and functionality.

The prediction of future is always uncertain, but it can be assumed that frequencies under 2 GHz remain mainly for mobile communications where only low bit rate services are offered (both data and speech). In this case, connections requiring close to 2 Mbps or more will need to be moved on to the higher frequencies. The possible choices at the moment seem to be around 5.2 GHz and.

The successful introduction of wireless ATM is strongly related to the success of ATM/B-ISDN in wired networks. If ATM/B-ISDN networks are to be a commercial success, wireless ATM could be seen as not today's technology but as inevitable of near future.

There are three standard bodies that have defined the physical layer in support of ATM: ANSI, CCITT/ITU-T, and the ATM forum. None of these forums have considered the wireless ATM air interface. ETSI RES10 sub-technical committee is the first standardization body to start

### **3.6.**

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## **CONCLUSION**

Innovations in Signaling, Accessing, Switching, Routing, etc. have emerged to take the role of leadership. Wired and wireless, IN tries to deploy the ATM concept in defining the high-speed wireless mobile access system, while developing the excellent Quality of Service (QoS) guarantee solutions. By developing an open intelligent ATM platform (wired and wireless) including generic signaling system, management system, we aim to provide a true virtual community -where any broadband real-time information can be reachable making the words "Anytime anywhere" more real.

## Chapter 4

### **BROADBAND TECHNOLOGIES**

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The widespread use of digital technology in the WAN occurred because bandwidth was and still is expensive. Digital technology provides the same economies in communications as it does in computers, making better use of available bandwidth and allowing more complex services and equipment to be offered at lower cost. Encouraged by favorable tariffs, users have upgraded to digital services. This growth is accelerating because of the benefits offered by advanced digital data communications protocols. These protocols, first deployed in the computing environment, are being modified to create synchronous overlays to the global asynchronous telephone network.

The Telco/PTT use of asynchronous protocols derives from worldwide networking standards implemented over the past forty years. These protocols operate at the physical level (OSI layer 1) and are transparent to the computer communication protocols they transport. In North America, the standard for

high-speed WAN interconnection is the T1, with a fixed bit rate of 1.544 Mb/s. This rate supports 24 digital channels of 64 kb/s each, plus eight kb/s for signaling. At the time of the introduction of T1 service, 64 kb/s was the bandwidth needed to transport a single voice conversation, using a method of modulation called pulse code modulation (PCM). Today, toll-quality voice can be compressed even more, allowing a 64 kb/s channel to carry multiple voice conversations. Nonetheless, the fundamental rate for voice transmission and switching in the public telephone network remains 64 kb/s.

Within the public telephone network there is a scale of rates that goes beyond T1, referred to as the asynchronous standard hierarchy. While consistent at each level, it has little overlap, so asynchronous DS1 frames have little resemblance to DS3 frames. As a result, the public network is synchronous only on a piecemeal basis, and therefore lacks the management capability and bandwidth flexibility for many newly demanded services. In contrast, LANs provide inexpensive connectivity, at least in a local area. But LANs, too, have come under pressure from the increasing bandwidth demands of new applications.

One result is a tenfold increase in bandwidth and use of fiber media. FDDI, a dual counter-rotating token ring topology LAN operating over fiber, is being promoted as a solution for broadband applications. But shared-media LANs of any variety including FDDI are intrinsically limited. Video, as well as voice traffic, tolerates relatively small delays and requires synchronization signals that shared-media LANs lack. In campus networks, for example, video cannot be

effectively transported, even with compression as low as 128 kb/s. If there were only a few workstations on an FDDI LAN, such compression might be enough, but that is not the case in a shared-media environment.

#### 4.1.

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## NETWORK CHANGES

The situation is improving. Rather than jury-rigging the asynchronous standards for improved management capability, a more general approach has been taken by the worldwide standards bodies. Networks have begun the migration from asynchronous to synchronous. With the almost inexhaustible bandwidth availability of fiber cable, the silicon integrated circuit revolution, and the improved synchrony of network clocks, both packet-switched and isochronous traffic can be transported synchronously over large distances. Today, a variety of networking technologies is available to address the rapidly emerging class of bandwidth ravenous applications. These broadband networking technologies include the fast-packet schemes, such as frame and cell relay (ATM) (Chapter 1), and SONET/SDH. SONET/SDH represents a transmission format that includes in-band network management. It overlays the existing asynchronous broadband network and will eventually replace it.

#### 4.2.

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## BROADBAND INTEGRATED SERVICES DIGITAL NETWORK

A broadband service is defined as a service which requires an information transfer rate in excess of the capacity of a primary rate interface that is 2.048 Mbps. The BISDN is a virtual circuit switched network that can be used for services requiring high-speed packet switching. The basic aim of B-ISDN is to use ATM cells on the SONET specifications as physical layer.

BISDN is based on the ATM architecture which associates calls with cells by a label in the header. In a labeled channel, data is put into cells prefixed with a header, which contains at least one VCI (Virtual Circuit Indicator) and an error detection field. The ATM provides great flexibility in supporting both continuous and bursty traffic streams by integrating services in common switching and transmission facilities.

While SMDS is a network service, BISDN is a service platform which is supported by ATM (see BISDN ATM Connection.) Using a limited set of network interfaces and network equipment configurations, a single BISDN network can support a wide range of customer data, voice, and video applications. BISDN takes advantage of the availability of high capacity synchronous optical fiber networks (SONET/SDH) to reliably transport enormous quantities of information. Access to BISDN is at 155 Mb/s and 622 Mb/s using high-speed, fixed-length ATM cells. With that much capacity, BISDN is expected to carry a multitude of services.

With BISDN, private networks require only a small number of physical interfaces. Users needing 45 Mb/s for high-quality videoconferencing have only

to dial up a 45 Mb/s connection. Virtually any bandwidth between network locations can be established on demand. The list of services that could be provided with BISDN is just beginning to unfold. They fall into two basic categories: interactive, and distributed. Interactive services include videoconferencing, video telephony, high-speed data, electronic mail with images, and interactive database services with high-resolution imaging and audio enhancements. Distributed services include cable TV (existing quality), high-definition cable TV, pay-per-view TV, and compound document distribution.

### **4.3.**

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## **ASYNCHRONOUS: T3**

T3, an asynchronous transmission service, represents the equivalent of 28 T1 lines at a rate of 44.736 Mb/s. Although offered by Telcos/PTTs, there are issues to consider before deploying T3 in corporate networks: First, asynchronous signal types lack a precise relationship to each other. Within a 45 Mb/s DS3 signal, the 1.5 Mb/s DS1 signal has no visibility. The entire DS3 has to be taken apart to reach the DS1. Then the signal has to be reframed, taking time and processing power. Second, DS3 is often transmitted over fiber, which requires an interface for electrical-to-optical signal conversion. Unfortunately, asynchronous transmission only provides a standard electrical interface. The lack of an optical standard for asynchronous signals has led to many proprietary interfaces. These proprietary optical transmission rates include 45 Mb/s (typically 48 to 50 Mb/s),

90 Mb/s, 135 Mb/s, 405 Mb/s, 560 Mb/s, 565 Mb/s, and 1.1 Gb/s – all of which are unique to individual terminal equipment vendors.

#### 4.4.

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### **ASYNCHRONOUS: E3**

The T3 rate of 44.736 Mb/s represents the North American standard. The Japanese T3 equivalent is 32 Mb/s, while the European equivalent is E3, which operates at 34 Mb/s and supports E1 signals at 2.048 Mb/s transmission rates. Both E3 and T3 are insufficient for intelligent networks. Although there is some capability for maintenance signaling, T3/E3 actually have less maintenance visibility than lower-rate T1/E1.

#### 4.5.

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### **SWITCHED MULTIMEGABIT DATA SERVICE (SMDS)**

SMDS is a specification for a metropolitan area, high-speed data service distributed by the Regional Bell Operating Companies (RBOCs), some Interexchange Carriers (IXCs), and several European public network providers (Table: SMDS vs. Private Lines). SMDS supports a variety of high bandwidth applications, providing the economies of shared and switched access within metropolitan areas. It allows the same type of connectivity for data that the telephone offers for voice and fax. Information is transported over counter-rotating rings that extend over a municipality.

Because SMDS is a metropolitan-area network specification, it has been limited to use within local access and transport areas (LATAs). The inter-area connections needed for a national network have been harder to come by. Although coast-to-coast communication among local SMDS networks is possible through standard direct lines, local networks have not employed consistent implementations of SMDS. Consequently it has been a harder sell than the providers anticipated. Despite this, SMDS now has hundreds of customers and is emerging as a niche service that meets the communications needs of many companies.

#### **4.5.1.**

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### **Technology**

SMDS is a connectionless, cell switched data transport service that combines a shared medium LAN with ATM, and offers switched access. It was conceived as a high-speed public packet service, mainly to interconnect local area networks. The SMDS standard defines a tiered network architecture. As with a LAN, SMDS requires an access technique to prevent overlapping transmissions. It is also like a LAN in that it is connectionless—it does not set up the sequence that has become known as a virtual circuit. Each packet or datagram is addressed and switched independently, with no prior network connection. SMDS differs from LANs that use either CSMA/CD or token passing techniques because it uses a distributed reservation scheme in which each node keeps count of the access requests made by the nodes ahead of it.

SMDS utilizes the IEEE 802.6 protocol, which transmits ATM cells over a shared bus. The 802.6 protocol supports a complete mix of services: constant bit rate, variable bit rate, and available bit rate. Data packets are transferred across an SMDS network, employing a distributed queue dual bus (DQDB). DQDB is the IEEE 802.6 cell relay network standard for switching and transmitting 53-byte cells. With DQDB there are two buses that pass data in opposite directions (Figure: Switched Multimegabit Data Service (SMDS)). Stations transmit on both buses in opposite directions, after queuing their data and waiting for the head end station of the bus to grant permission to send. This station continuously transmits empty cells that travel onto and down the bus and may be used by the other stations to transmit information. When the empty cells reach the end of the bus they are discarded. The last station or tail station monitors the bus and will disable devices that are monopolizing the bus. The buses are usually configured in a loop for greater reliability. When a failure such as a cable break occurs, the two adjacent stations temporarily serve as upstream and downstream masters.

Each SMDS packet has the capacity for as many as 9188 bytes of data, which accommodates entire packets from most legacy LANs—Ethernet and token ring - - except for 16 Mb/s token ring that can have a maximum frame size of 16,000 bytes. The SMDS packets are divided into cells or slots of fixed length—53 bytes per cell, of which 48 contain data and 5 are for control. The lower layers of SMDS interface protocol (SIP) transmit the 53-byte cells in conformance with the ATM structure. Because 802.6 was developed early in the ATM standardization

process, it uses AAL 3/4 cell format instead of the AAL 5; and therefore, it requires an 8-bit access control field rather than the 4 bit generic flow control field of conventional ATM. Otherwise the cells, by design, are ATM cells. Thus, DQDB delivers a compatible format to carry SMDS data and interface with ATM networks that are accessed across a dedicated link, typically from a customer's network. According to current specifications, access is by means of a DS1 or DS3 line, with service classes defined for a sustained transfer rate of 4, 10, 16, 25, and 34 Mb/s. A SONET/SDH interface at STS-3/STM-1 is expected in the future.

#### **4.5.2.**

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### **SMDS Internetworks**

The LAN-like features of SMDS make it a natural backbone network for seamlessly interconnecting Ethernet, Token Ring, FDDI and ATM LANs over extended geographical areas. In a MAN configuration, SMDS can interconnect users for distances of up to 50 kilometers. Beyond that, interconnection by means of a SONET-WAN is desirable, with DS3-based networking as an alternative. To connect a LAN to an SMDS network requires only a router and an SMDS compatible DSU/CSU or SMDS host adapter card. For users to communicate over the WAN from one SMDS/MAN to another requires a gateway or bridging function. Interface guidelines, developed by the SMDS Interest Group, support the networking protocols found in LAN environments – TCP/IP, Novell's IPX, AppleTalk, DECnet, SNA and OSI. The SMDS Interest Group, a consortium of internetworking and wide area network vendors, also designed an SMDS

interface (see Proposed SMDS Interface). The interface allows LAN routers to connect to SMDS networks via data service unit/channel service units, or DSU/CSUs, and to partition SMDS functions between the router and the DSU/CSU. The interface is based on a high-level data link control (HLDC) protocol that is widely used for communications between LAN and DSU/CSU equipment. The bridge/router may map the DS1/DS3 signal to SONET VT1.5/STS-1 prior to a SONET NE interface. The SONET NE would provide DS1/DS3/OC-1/OC-3 extensions as required.

SMDS does not require carrier switches to establish a call path between two points of data transmission. Instead, SMDS access devices pass 53-byte cells to a carrier switch. The switch reads addresses and forwards cells over any available path to the desired end point. SMDS addresses ensure that the cells arrive in the right order. One benefit of this connectionless service is that it puts an end to the need for precise traffic flow predictions and connections between fixed locations. With no requirement for a predefined path between devices, data can travel over the least congested routes in an SMDS network. In fact, SMDS cells are released into the public network delivery system as letters are to a post office. By knowing an SMDS address, subscribers can call up and send/receive data.

Because SMDS coexists with dedicated facilities, customers can create hybrid public/private networks. Existing networks can be easily expanded and new sites quickly added to an SMDS network without totally reconfiguring it. Additions to an SMDS network only require an update to a screening database

on the SMDS switch. Since SMDS is connectionless, its users may construct mesh networks within which each site is connected to all other sites.

SMDS data services, at speeds from 56 Kb/s to 45 Mb/s, are offered in the United States and Europe. Initially conceived as a switched access service at T1 and T3 rates, SMDS's popularity increased when the service was offered at lower access rates. Slower speed service became available when several of the companies that make the channel service unit/data service unit (CSU/DSU) equipment (which connects customers' equipment to the local exchange) collaborated to provide a lower rate interface. This digital exchange interface encapsulates cell based SMDS datagrams into variable length frame packets that can be carried over 56 Kb/s lines and converted back into SMDS 53 byte cells at the other end. CSU/DSUs for 56 Kb/s connections are significantly less expensive than those for higher speed T1 or T3 services. The routers and other equipment that support the digital exchange interface are compatible, which is not always the case with specialized equipment that uses the higher speed SMDS interface protocol.

The effect of the digital exchange interface standard has been to make SMDS competitive with frame relay. The cost of subscribing to the service is about the same as for leasing several frame relay ports.<sup>2</sup> Nonetheless, in order to become a viable service, SMDS will have to offer higher bandwidth than frame relay does and be able to seamlessly interconnect with other carrier services, neither of which are yet possible. Interconnection services are offered by independent

telephone companies and IXCs. Co-Net Communications Inc. of Orlando, Florida, an SMDS network operator that is aimed at the graphics industry, and MCI, among others, offer inter-LATA connections. The pricing of these services depends on customer equipment, software, store-and-forward capabilities, and inter-LATA connections that are based on the total transmitted number of megabytes.

The needs for MAN services, and for SMDS in particular, are quite varied. Any data transfer requiring large amounts of bandwidth, such as the interconnection of large numbers of subscribers to a wide area public data network, imaging, computer-aided design, publishing, and financial applications, can benefit from SMDS.

#### 4.6.

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### **SONET/SDH**

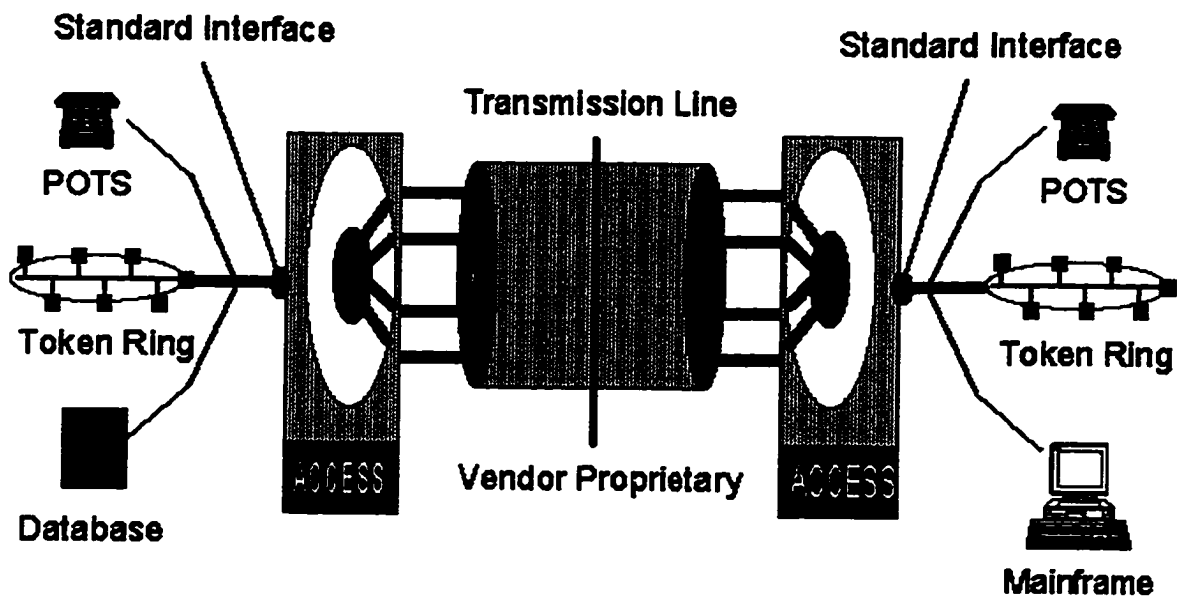
The *Synchronous Optical Network* (SONET) was first proposed as a standard by Bellcore for a family of interfaces to be used in optical networks. The lack of standards for high-speed digital transport to be used for single-mode fiber led to a number of propriety standards. SONET is now an international standard accepted by ANSI.

The SONET standard provides a point-to-point link over fiber-optic media. It provides the physical layer connection for B-ISDN and is defined in terms of a hierarchy of data rates. This hierarchy is defined in terms of the *Optical Carrier*

(OC) speeds and the corresponding electrical signals (*Synchronous Transport Signals*) used to interface with electrical components.

The asynchronous standards specify the termination and aspects of a limited amount of network management. Access to the network is specified as a standard electrical interface for connection compatibility. For DS3, the rate of 44.736 Mb/s is defined, together with a signal level, impedance, and other characteristics. This is not adequate to achieve interoperability among equipment vendors

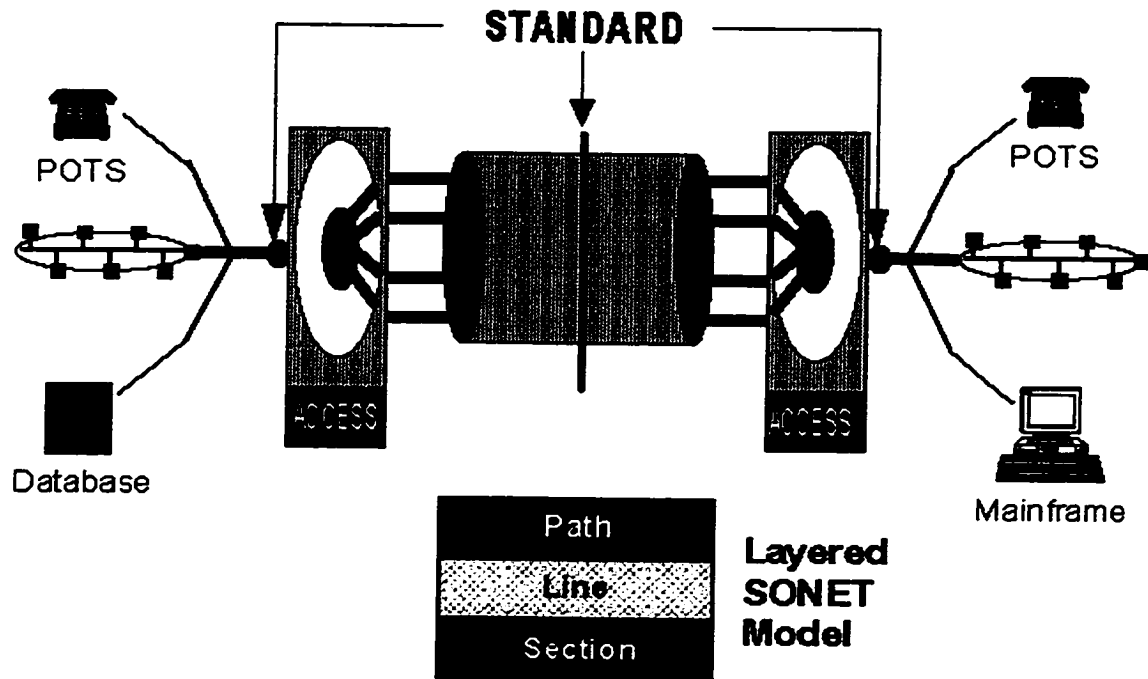
## ASYNCHRONOUS TRANSMISSION



**Figure 4.1: Asynchronous Transmission.**

Both SONET and SDH standards specify the electrical and optical interfaces as well as protocols at different points on the communication line, permitting the use of equipment from different vendors anywhere along the fiber span

## Synchronous SONET Transmission



**Figure 4.2: Synchronous SONET Transmission.**

This capability is referred to as mid-span meet. Customers get better bandwidth control as well as reduced Telco/PTT costs because SONET/SDH compliant equipment is interoperable. Furthermore, its ability to concatenate signals consistently at any speed makes multiplexing sufficiently low-cost to be embedded in fiber optic equipment. Given this, real-time switchable

multiplexers, add/drop systems, and ring networks can now be built for improved network reliability, performance, and cost. As a result, SONET (in North America) and SDH (in Europe) deployment is gaining great momentum.

#### **4.6.1.**

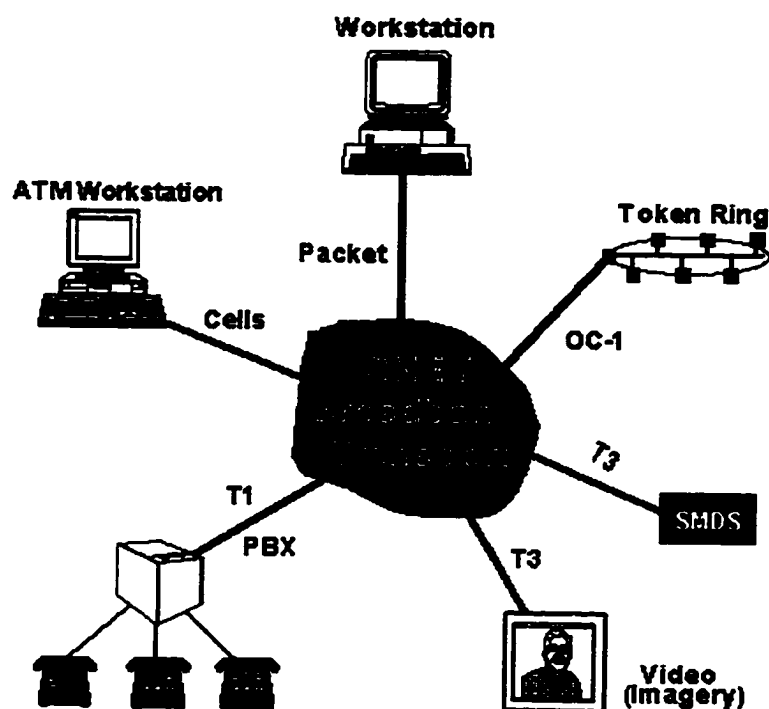
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### **Network Architecture**

SONET/SDH technology combines the historical elements of LANs and WANs with much borrowed from advanced computer technology. The SONET in-band network management ASN.1 protocol derives from the OSI standards on object language definition that was developed for computers. This characterization by a communications standards body reflects the coalescing of LAN and WAN technology. Due to its powerful architecture, SONET integrates existing point-to-point fiber links into true networks, enabling them to route single-voice grade channels digital service level (DS0) without the need for multiple stages of multiplexing and demultiplexing. The eventual role of SONET and fiber systems will not be simply to replace the dedicated copper lines used in transmitting multiplexed digital signals. SONET networks will simultaneously offer greater functionality, standardized fiber interfaces, higher transmission speeds and reduced maintenance. SONET provides a flexible, controllable network with centralized network management. Because of its capacity to manage large amounts of bandwidth, as well as its simplicity and cost-effectiveness, it will eventually halt purchases of non standard fiber and asynchronous communications equipment. It already is the preferred delivery vehicle for a

number of emerging broadband services, including frame relay, SMDS, BISDN, and ATM

### SONET Delivers Services



**Figure 4.3: SONET Delivers Services.**

For this vision of broadband networking to be fully realized, the Telcos/PTTs will have to upgrade their infrastructure to support SONET signals that operate at higher data rates than their asynchronous counterparts. Although the asynchronous network is unable to support SONET signals, SONET will transport asynchronous signals such as DS1, DS1C, DS2, and DS3. Therefore, existing asynchronous equipment such as multiplexers, DACs and switches will

not have to be immediately replaced as SONET equipment migrates into the public network.

## 4.6.2.

### Transmission rates

The OC and STS rates are multiples of 51,84 Mbps. The basic building blocks is the 51,84 Mbps line rate. Thus OC-48 is  $48 \times 51,84 \text{ Mbps} = 2488.32 \text{ Mbps}$ . The standard defines up to OC-240; that is 12.4416 Gbps rate. STS-1 rates are roughly equivalent to the T3 (45 Mbps) data rates available today.

Design goals of SONET included providing a way to accommodate T3 data rates and resolving the incompatibility between the North American and European digital hierarchies, as expressed in the T1, T2, and T3 signals. The North American standard is based on a T1 rate of 1.544 Mbps, whereas the European standard is based on 2.048 Mbps.

OC	STS	STM	RATES (Mb/s)		
			Line	Payload	Overhead
OC-1	STS-1		51.840	50.112	1.728
OC-3	STS-3	STM-1	155.520	150.336	5.184
OC-12	STS-12	STM-4	622.080	601.344	20.736
OC-24	STS-24		1244.	1202.688	41.472
OC-36	STS-36		1866.240	1804.032	62.208
OC-48	STS-48		2488.320	2405.376	82.944
OC-96*	STS-96		4976.640	4810.752	165.888
OC-192*	STS-192		9953.280	9621.504	331.776

\* Indicates Rates Under Study

NOTE: Theoretical Maximum N=255 or 13.2192 Gb/s Line Rate

**Table 4.1: SONET/SDH Rates and Bandwidths.**

Using a basic building block called the synchronous transport signal level-1 (STS-1), with a line rate of 51.840 Mb/s for SONET and 155.52 Mb/s for synchronous transport mode level-1 (STM-1) for SDH, SONET/SDH reaches 2.488 gigabits per second in exact multiples of STS-1.

Table 4.1 shows the relationship of line rates, payload, overhead bandwidths, and the corresponding STS levels. STS electrical signals, when transmitted over fiber, are converted to a corresponding optical signal called optical carrier (OC). For example, higher SONET transmission rates are established by concatenating "N" STS-1s to form an STS-N. Currently, "N" is defined as 1, 3, 12, 24, 36 and 48, but rates up to OC-255 or 13.2192 Gb/s are possible.

### **4.6.3.**

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## **Synchronous Clocks**

Unlike asynchronous signals, the synchronous signal is not demultiplexed and then remultiplexed at every central office through which it passes. SONET allows simple add/drop multiplexing. That is, information as fine as a single DS0 can be taken from one broadband data stream and inserted into another. This process requires clock synchrony within the public telephone network. When the asynchronous public network was designed, such clocks did not exist; Delays due to the vacuum tubes and transistors used then were too great. SONET network clocks may be distributed across thousands of kilometers,

making synchronous transmission difficult to maintain. The issue occurs because of the way that the clocks are formed.

The clocks in the synchronization network are classified on the basis of performance. There are four levels, called strata (Table: Public Telephone Network Clocks). Stratum 1 (ST1) is the highest level and its free running clocks are used as primary reference clocks (PRC), compared with ST3 clocks that are normally locked to an incoming reference, which is traceable to an ST1.

#### **4.6.4.**

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### **Layered Structure**

The overhead is divided into layers:

#### **4.6.4.1. PHYSICAL**

The physical layer transports bits as optical pulses through the fiber medium. No overhead is associated with this layer. The main function of the physical layer is to receive the STS-N signal bit stream, convert each bit into an optical pulse, and transmit the optical pulse across the fiber medium toward the far-end terminal. Key concerns are pulse shape, power levels, and the line code for error detection and recovery.

#### **4.6.4.2. PATH**

The transport of network services between two SONET multiplexing nodes is the function of the path layer. Examples of such services are the provisioning of DS1, DS2, DS3, DS-4NA, FDDI, ATM, video, etc. The path layer maps the services into the SONET payload format as required by the line layer, and communicates end-

to-end via the path overhead. The path terminating equipment (PTE) interprets or creates the overhead defined for this layer.

#### 4.6.4.3. LINE

The line layer is responsible for transporting the payload and the line overhead to its peer at the far end. All lower layers provide transport. This layer maps the payload and the line overhead into STS-N frames. The payloads and line overheads are synchronized and multiplexed within the STS-N, before the STS-N signal is passed to the section layer. The overhead associated with this layer is interpreted or created by line terminating equipment (LTE).

### SONET LAYER STRUCTURE

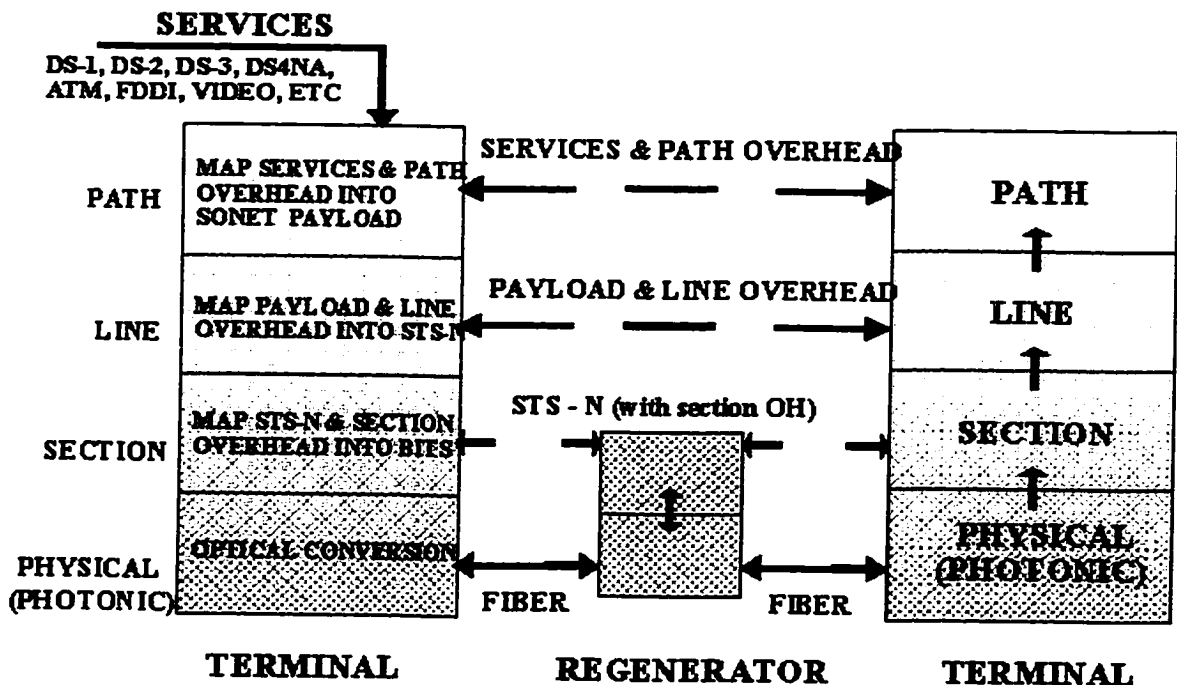


Figure 4.4: SONET Layer Structure.

section: The section layer transports the STS-N frame across the physical medium. It uses the physical layer for transport. Section layer functions include framing, scrambling, section overhead processing, and error monitoring. The overhead defined for this layer is interpreted or created by section terminating equipment (STE).

This layers govern the transport of the payload across the network.

The relationship between SONET layers plays a very important role with regard to the various network services, DS1, DS3, etc., that it supports.

The SONET/SDH path and line section manages heterogeneous environments, providing bearer channel end-to-end control. This is done with frame overhead bytes.

The actual overhead byte patterns depend upon how the equipment will be used. For instance, the K1 and K2 bytes transport messages between the network elements and the network management system. With a ring topology, a more sophisticated K1/K2 protocol would be required to switch traffic direction, since the traditional alarming protocol of the telephone network would be incompatible with a ring architecture.

## SECTION, LINE PATH OVERHEADS

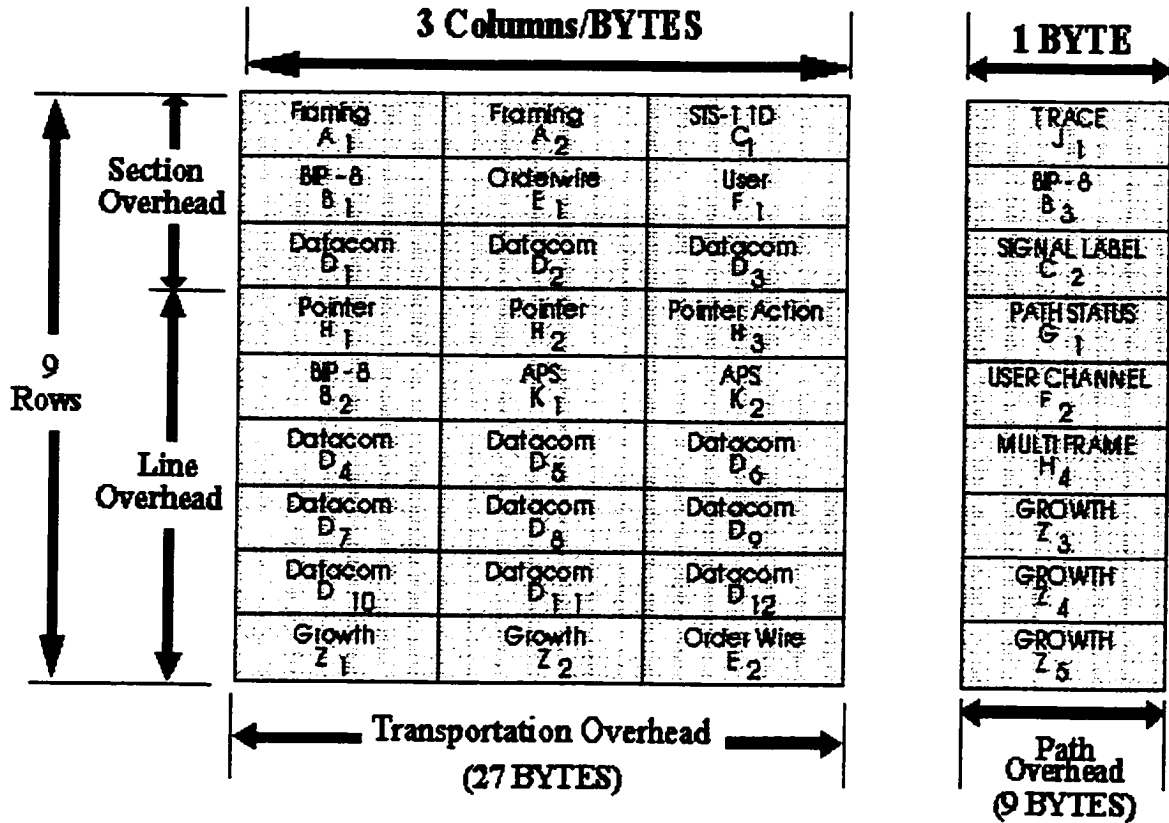


Figure 4.5: Section, Line Path Overheads.

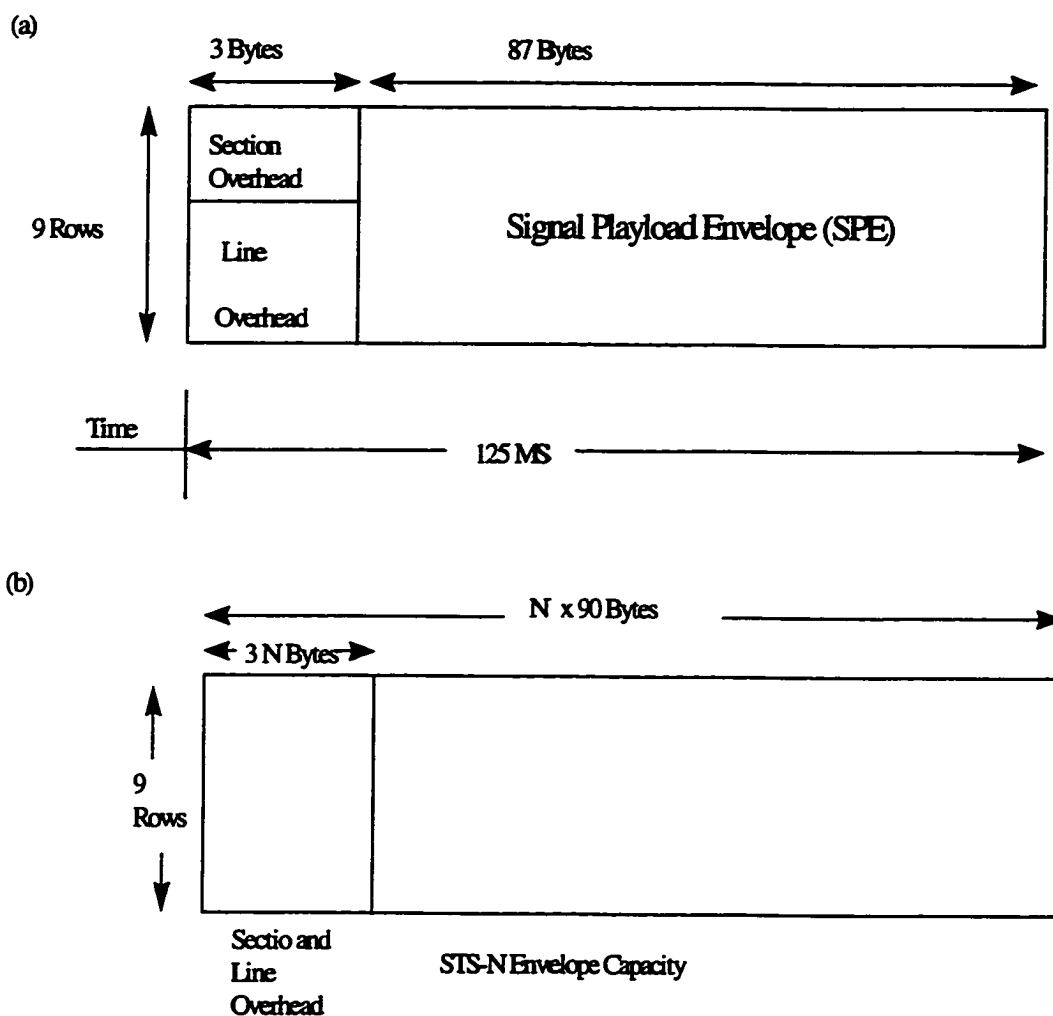
### 4.6.5.

## SONET Frames

STS data streams can be combined to yield higher STS rates. Three STS-1 rates.

Three STS-1 rates multiplexed together, therefore, can yield a data of STS-3.

The STS-1 frame is 810 bytes and is made up of a 90 column-9-row matrix with each cell of this cell of this matrix being one byte long. This data matrix is transmitted in one STS-1 frame, as seen in figure 4.6.



**Figure 4.6: STS-1 and STS-n frame Format**

The first three bytes of each row contain overhead, and the remaining six rows contain 18 bytes of line overhead. The combined section and line overhead of each frame is 27 bytes. The remaining 87 columns contain the payload or data.

This frame payload is called the *Synchronous Payload Envelop (SPE)* and works out to be 783 bytes (9x87). Not all the SPE contains data; The first nine bytes contain path information.

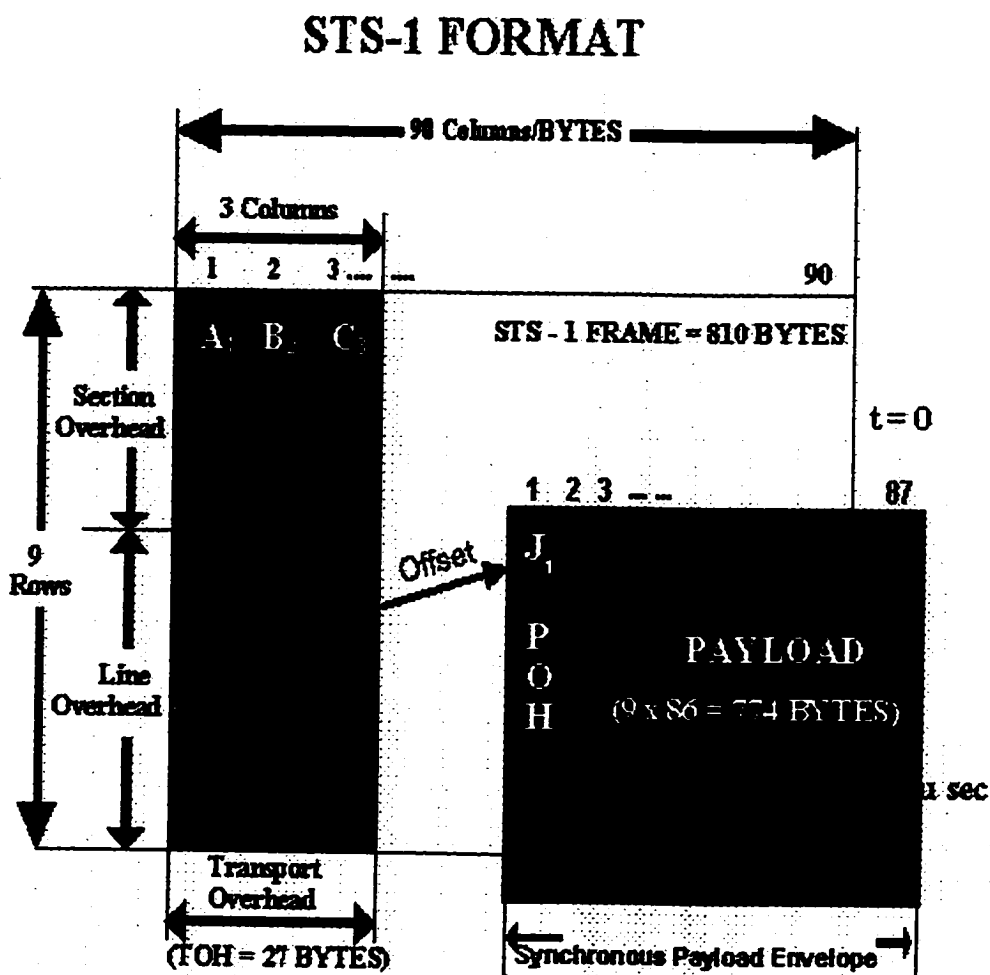
The STS-1 frame is sent once every 125 microseconds or at the rate of 8,000 frames every second. Because each byte in the frame is eight bits, the data for each byte of data in a frame corresponds to an  $8,000 \times 8 = 64,000$  bps channel. We can use this data rate to transport digitized speech or data.

Of the 783 bytes that make up the SPE, nine bytes are used for transport overhead for every row. This transport overhead contains a pointer to the location in the SPE that the actual data begins, thus allowing data to float.

The lowest data rate supported by SONET/SDH is the 51.840 Mb/s STS-1 frame. An STS-1 frame is a 9 row by 90 column byte matrix structure totaling 810 bytes, or 6480 bits.

Figure 4.7 shows a detailed STS-1 frame format. The byte from row 1, column 1 is transmitted first, then is followed by row 1, column 2, and so on. The transmission is from the left column to the right column and from the top row to the bottom row. The STS-1 frame transmission duration is 125 microseconds, or 8000 frames per second, which maintains compatibility with the existing telephone network. The first three columns, or 27 bytes, are assigned to the transport overhead, which is subdivided into section overhead (9 bytes) and line overhead (18 bytes). The remaining 9 rows by 87 columns constitute the STS-1

synchronous payload envelope (SPE) that has a total of 783 bytes. Of these, 9 bytes in the first column are designated as STS path overhead, or POH. The actual payload is 774 bytes, which results in a total of 49.536 Mb/s of payload capacity.



**Figure 4.7: Detailed STS-1 Format.**

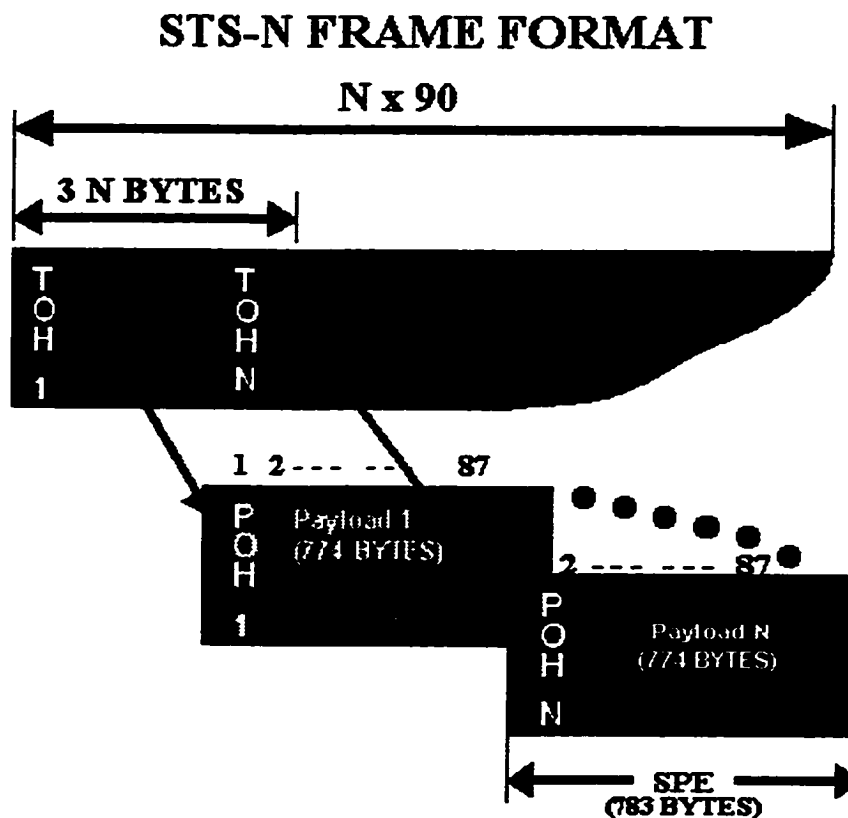
This payload capacity is used to carry a DS3 signal at 44.736 Mb/s, or 28 DS1s each, at 1.544 Mb/s. All this is at the STS-1 level. For the transport of payloads

with bandwidth requirements greater than STS-1, several STS-1s can be combined or concatenated (byte interleaved) and transported as a single entity. For example, the ITU has defined the SDH lowest-rate STM-1 as the equivalent to SONET STS-3.

Before byte interleaving, the STS-1s are frame-aligned so the transport overhead of each STS-1 can be combined to form an STS-N transport overhead. Thus, all transport overheads are frame-aligned, while the individual payloads float within the envelope as indicated by the respective payload pointers within each STS-1 transport overhead. The byte interleaved STS-N signal, when transmitted, results in an optical signal called Optical Carrier OC-N, with values of N. The overhead byte C1 in the section overhead identifies the STS-1 within an STS-N frame format.

The transport overhead (TOH), together with section and line overheads, performs the functions needed to transport the SPE over the fiber link. Also, the payload pointer, or offset, resides within the line overhead of the TOH. The payload pointer plays a very crucial role in the transmission process because it indicates the start of the STS-1 SPE. It also permits the payload to float within the envelope, and helps adjust frequency deviations between network elements. A 9 byte path overhead (POH) is allocated within the synchronous payload envelope to support transport of the payload from the point at which it is assembled to the point at which it is disassembled.

The size of an STS-N frame is N times the STS-1 frame size or N times 810 bytes.

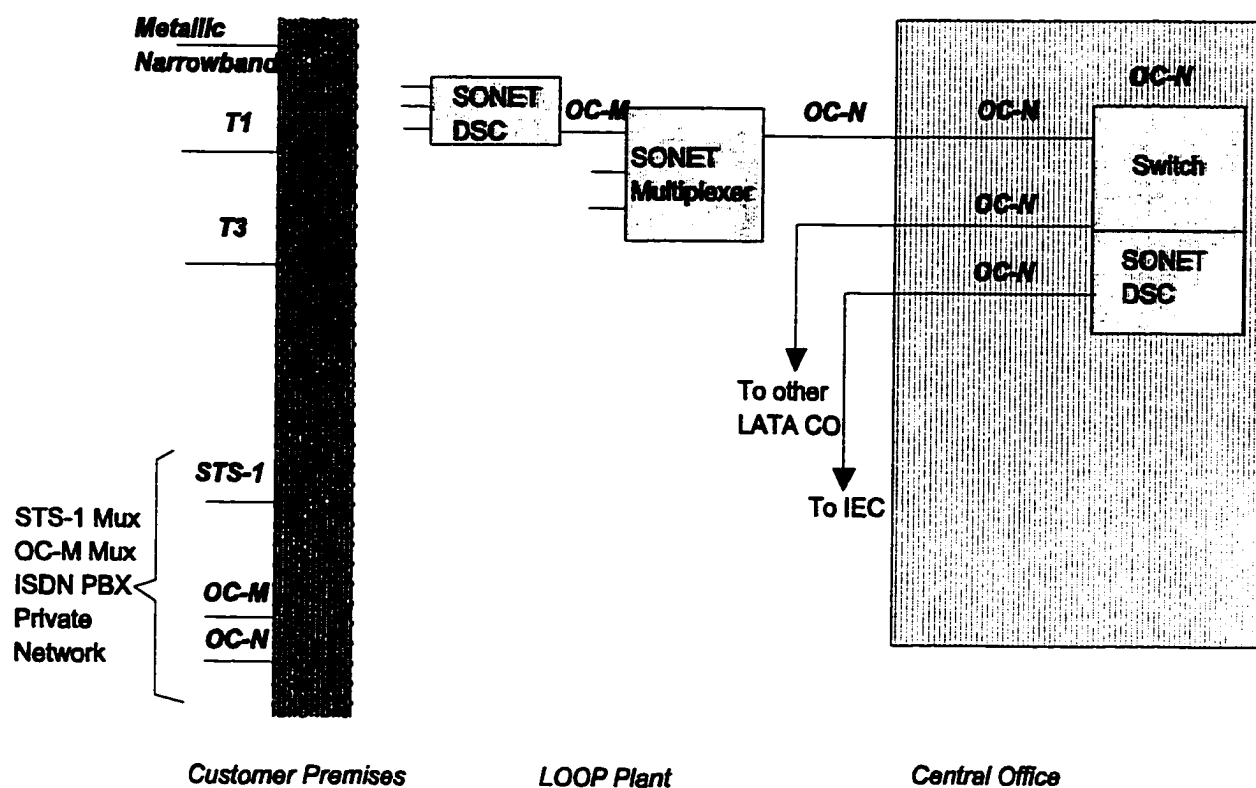


**Figure 4.8: Detailed STS-N Frame Format.**

Similarly,

- o Transport overhead = N times 27 bytes
- o Synchronous payload envelope = N times 783 bytes
- o Path overhead = N times 9 bytes
- o Payload size = N times 774 bytes

Multiplexing of various SONET frame and digital cross connects are handled by the SONET central offices (see figure 4.9). T1, T3, HDSL, VHSL, and other STS-N based services are reduced by their common basis of their synchronism (125  $\mu$ s), sent through the appropriate switch and then the information is reconstituted in the appropriate signaling format and sent to the customer.



**Figure 4.9: The typical configuration of a SONET Based Central Office.**

Both ATM and SONET will built up the future of optical information, and then all the systems will assume one standard format which will be the ATM format using the ATM switch.

## 4.6.6.

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### Virtual Tributaries

SONET framing accommodates lower-rate signals, providing transport for existing North American and international formats. The STS-1 payload may be subdivided into smaller virtual tributaries (VTs) that transport signals at less than DS3. Each VT functions as a separate container within the STS-1 signal with its own overhead bits. Because SONET/SDH is an international standard, the most common North American (DS1 at 1.544 Mb/s) and international (E1 at 2.048 Mb/s) tributaries have defined VT mapping. Less common tributaries, such as DS1C and DS2, are also represented. There is a special mapping for higher-rate DS3 signals within a SONET payload—and eventually 10 Mb/s for Ethernet and 16 Mb/s for token ring LANs.

There are both 'locked' and 'floating' channelized and unchannelized VT modes. The unchannelized floating mode is for wideband cross-connection down to the DS1 level. In unchannelized operation, the lowest addressable level is a VT. Two unchannelized mappings have been defined: asynchronous and bit synchronous. The former requires minimum timing consistency between the tributaries and the SONET clock, whereas the latter establishes a common clock frequency, but assigns an arbitrary phase. The locked mode fixes the VT location within the SPE that supports DS1 cross-connects. DS0 channels are uniquely addressable within a SONET payload by means of channelization. This is also known as byte-synchronous operation, in which both a fixed clock frequency and a fixed phase

are established based upon the DS1 frame. This enables individual DS0s (8 bit bytes) to be easily identified and cross-connected. The channelized locked mode provides for DS0 level cross-connection and bridging into an existing DS3 network. The floating mode employs pointers that define the location of the VT. An advantage of VT pointers is that they remove the need for slip buffers. In SONET equipment, the synchronizer, desynchronized, and slip buffer are replaced by a pointer processor.

A VT group architecture accommodates the mixes of various VT types within an SPE. The VT group size is a constant 9 rows by 12 columns, or 108 bytes. Mixing VT types within the same group is not allowed, but the number of VTs within a VT group depends on the VT size. For example, four VT-1.5s, three VT-2s, two VT-3s, or one VT-6 may be packaged into a single VT group, and seven VT groups may be byte interleaved with the overhead to form the SPE.

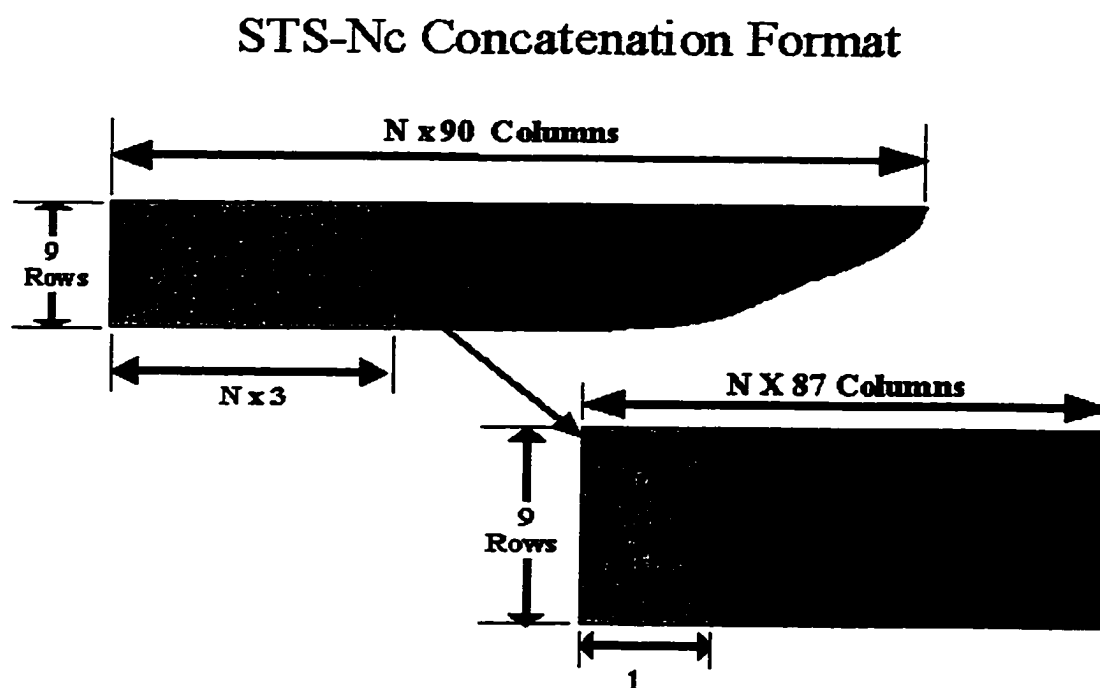
#### **4.6.7.**

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### **STS-Nc Concatenation Format**

The structure for STS-Nc concatenation is illustrated in Figure 4.10. SONET accommodates higher transmission rates by synchronously byte interleaving 'N' STS-1s to form an STS-N signal. Each STS-1 within the STS-N, however, remains a separate entity that is assembled at the source and disassembled at the sink point. The size of an STS-Nc is N times 810 bytes, while the transport overhead is N times 27 bytes. Both overhead and payload are treated as single entities. Only

one POH is shown. The remainder is allocated to the payload that, in this case, is  $(N \text{ times } 783 \text{ minus } 9)$  bytes. Moreover, in the STS-Nc frame format, the first STS-1 carries a normal pointer in its transport overhead, while the pointers of other STS-1s, forming the STS-Nc, carry the concatenation indicator. This indicator helps in binding the constituent STS-1s together and multiplexing, switching, and transporting them as a single entity. A concatenation indicator in the transport overhead shows that the STS-1s of an STS-Nc are joined—and will remain—together until terminated. By byte interleaving 'N' STS-1s, SONET provides for super rate services, such as BISDN, which uses multiples of the STS-1 rate.



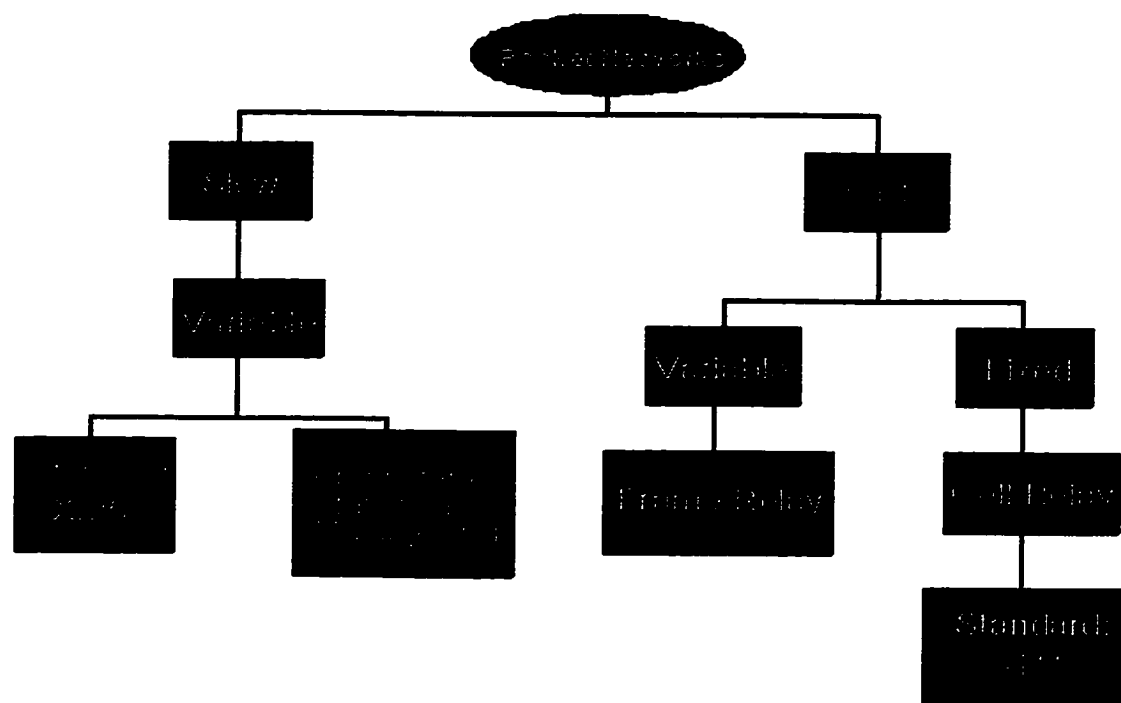
**Figure 4.10: STS-Nc Concatenation Format**

## 4.7.

## FAST PACKET: FRAME & CELL RELAY

Every application has an optimally sized data packet or frame for transporting information over a network. Real-time applications favor smaller packets since they are less likely to create bottlenecks.

### Packet Network Hierarchy



**Figure 4.11: Packet Network Hierarchy.**

Most existing data networks favor larger packets so that they can move bigger chunks of data. With the X.25 slow-packet standard, some public carriers settled on a 128-byte packet, because this allowed the best network throughput when

errors and retransmissions were considered. New packet transmission technologies are now being deployed that achieve orders of magnitude greater than the throughput rates of X.25.

One such technology – fast-packet – is already a success in private T1 networks. Fast-packet is a generic term that is applied to many different high-speed transmission technologies.

It achieves high performance by eliminating many of the overhead functions that are carried out at intervening nodes by X.25. Fast-packet networks exhibit low latency and very high-speed switching and thus are suitable for broadband communications. Frame relay is a fast-packet standard capable of transporting both small and large size packets at speeds up to T3.

#### **4.7.1.**

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### **Frame Relay**

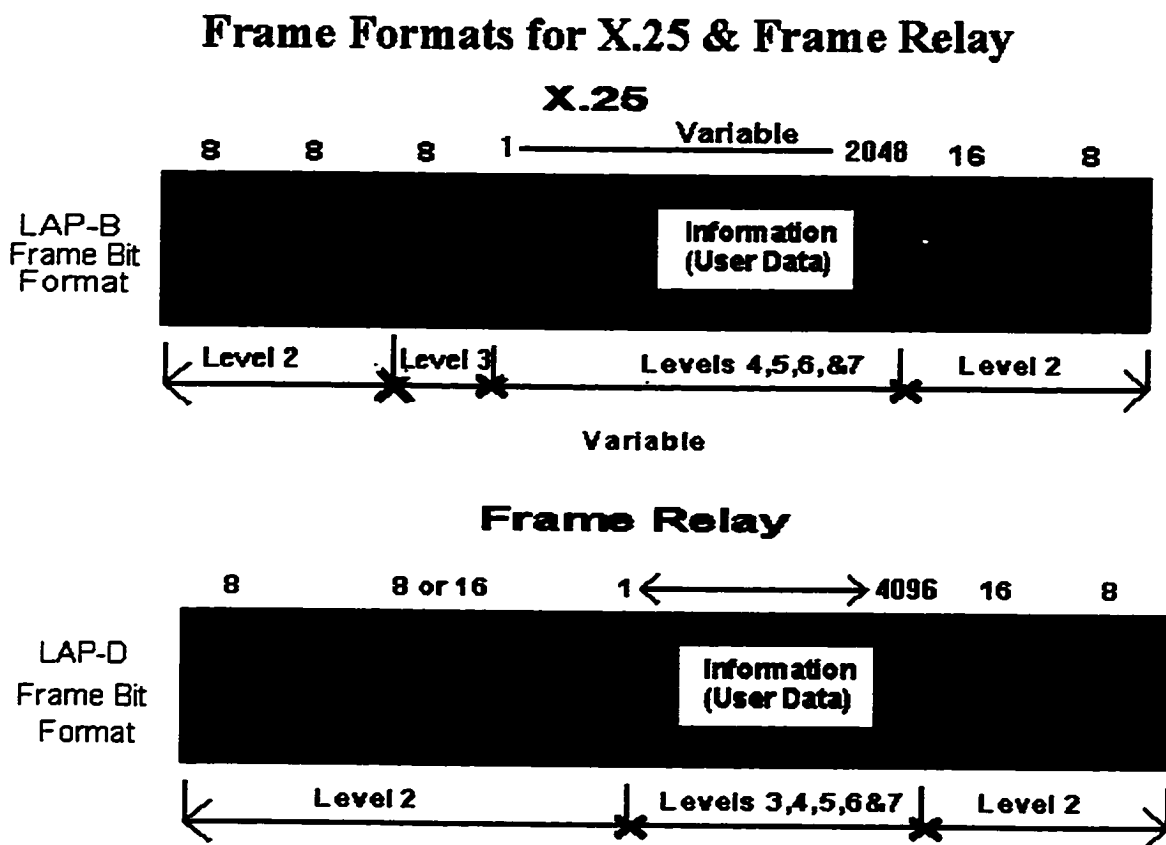
The frame relay protocol is a data transfer protocol defined by the ANSI and the International Telecommunications Union (ITU). It is similar to the X.25 and ISDN standards but it assumes a reliable transmission medium and therefore contains very little error recovery functionality. As a result, it is more straightforward and the data transfer is more efficient. At present it is used mainly in North America and Japan. A typical application is LAN-to-LAN interconnection.

Frame relay operates at OSI model layers 1 and 2 and resembles LAN networking protocols. It assumes that the customer is using error-correcting

protocol suites such as TCP/IP, IBM's SNA, Digital's DECnet, and Novell's IPX/SPX, among others, as well as relatively noiseless fiber and digital transmission. In contrast, X.25, developed for older analog leased lines, operates at OSI layers 1 through 3 and provides extensive error control. While frame relay resembles a stripped down X.25, it combines reduced overhead with high-speed interconnecting trunks and operates at 64 kb/s rates and above. Overhead is reduced by eliminating the error checking and correction done in the data link (layer 2 of the OSI reference model) and by moving the function of network layer 3 to increasingly intelligent end-stations. X.25 networks perform extensive error checking and correction because they were designed to compensate for poor quality analog communication lines and dumb end-stations. For a packet network to work in that climate, it had to verify the integrity of the information that it was transporting.

A primary difference among packet technologies is the way in which the address and control field bits are used.

By extending its address field, frame relay performs basic frame routing and control within the data link layer. An abbreviated destination address is processed at each node by a frame handler that has a list of routes between end-stations. There are three key elements in the address field; the data link connection identifier (DLCI), the forward and backward explicit congestion notification (FECN/BECN) bits, and the discard eligibility bit (DE).



**Figure 4.12: Frame Formats for X.25 and Frame Relay.**

## 4.7.2.

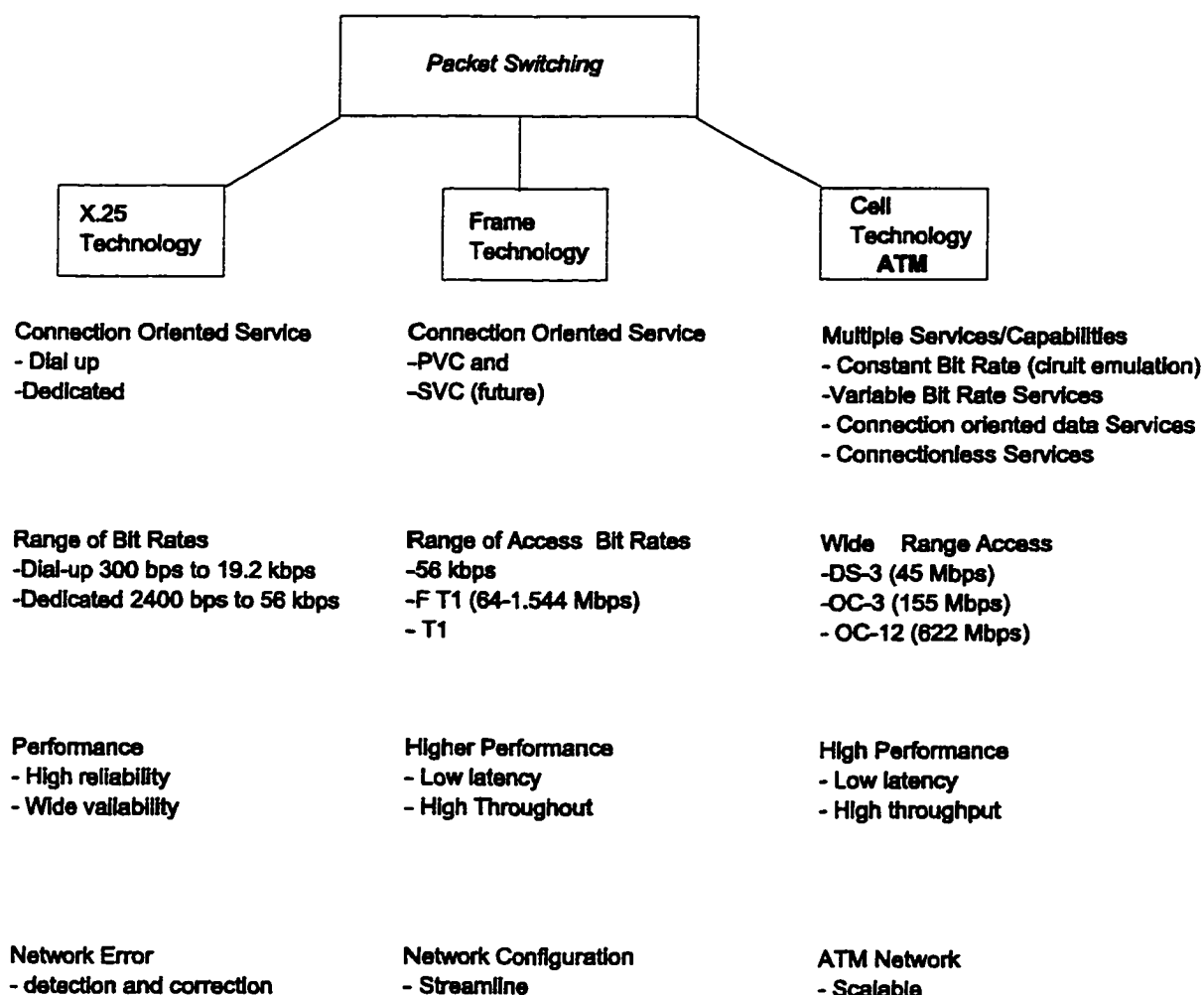
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### Cell Relay

Cell relay employs tiny frames to transport information at even higher rates than frame relay. At the time of transmission, the cells are filled by data packets that arrive at irregular intervals (asynchronously). The line of cells itself is synchronous, transporting a continuous stream of data. But packets enter the

stream only when they are available. The process resembles the loading of an endlessly circulating train of box cars. Each car in the train travels in synchrony with the others, but the rate at which information is loaded or removed is asynchronous. When there is a packet to be shipped, it is dropped into a car. At any one time, packets from different sources may contend for a specific box car, but only one packet can be loaded at a time. The rest must wait until a car is available, forming a queue. This process introduces a variable delay across the train. Sometimes there are many cars in a row, all filled with data packets. At other times only one car is filled, leaving a large gap between it and the next car, that contains a packet. Since an observer looking at the train in the middle of its run would not notice a reproducible pattern in the rate that the filled box cars pass, the cars or cells are deemed to be transferred asynchronously. The maximum rate that information travels is the rate at which the train is traveling.

When a user wants to send information to another user, the originating user signals the receiver with the cells. The first transmitted cell in the message contains the number of box cars of information. This cell goes to the destination. There it is removed and the destination sends a cell back to the source by a returning car. This cell defines the rate at which information may be transferred. For example, one box car full of information must be followed by five empty cars. This process, called signaling, is used to establish a connection and to negotiate throughput, grade of service, delays, and so on.



**Figure 4.13: Frame Relay versus Cell Relay**

Cell relay uses small packets for high-speed, low latency transport. Large amounts of information can be transmitted without monopolizing the network for long because the information is segmented. The cells are formed from the contents of much larger data words and are then rapidly switched and interleaved together. Software programming is too slow for controlling this process, so hardware is used with silicon integrated circuits as the enabling technology. Cell segmentation and reassembly equipment would be too

expensive without employing specialized silicon integrated circuits that are specifically designed for the task.

Several emerging broadband carrier services are based upon fixed and variable sized cells. Fixed sized cells have several advantages over the variable sized ones used in carrier services such as SMDS. Fixed sized cells reduce the queuing delay for high-priority calls because they are more rapidly processed at the user network interface. They are also switched more efficiently, which is the key to achieving the very high data rates achieved by the ATM form of cell relay.

#### **4.8.**

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### **ASYNCHRONOUS TRANSFER MODE (ATM)**

Telephone companies (Telcos) and their overseas equivalent private telephone and telegraph companies (PTTs) are migrating to standard network technologies to support the high bandwidth data communications needs of their business customers. Frame relay provides high-speed access, but lacks the capacity for a backbone Telco/PTT network. SONET/SDH networks have the bandwidth capacity, but lack the switching fabric. ATM fills this void. It was designed as a telecommunications technology, specifically as the transport for Broadband ISDN (BISDN). These standards define two ways to transfer blocks of information across a network—a synchronous transfer mode (STM) and an asynchronous transfer mode (ATM).

Despite the fact that it was developed for public telephone networks, it is in LANs that ATM found its first application. As long as LAN users continue to transfer ever larger data files and attempt to integrate isochronous voice traffic with data traffic, there is a great need for ATM. Nonetheless, its widespread use in LANs could still benefit the Telco/PTT by creating a demand for ATM Services. Carrying LAN traffic over WANs is a challenge because LANs employ different technologies and have different transport needs from WANs. LANs rely on communication standards that support high data rates over relatively short distances. WANs use communications standards that work over longer distances, but are relatively slow. Should WAN traffic become ATM based and the same technology be used in the LAN, carriers will face a considerably easier task of integrating WAN and LAN traffic.

### **4.8.1.**

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## **Broadband Switching**

The ATM switching fabric is data-rate independent and supports both public network and LAN switching at ultrahigh rates—exceeding 1 Gb/s. All ATM switching uses standard 53 byte cells. Each ATM cell has a 5 byte header that contains virtual circuit and virtual path identifiers.

The header is transmitted first and contains the addressing information. It does not carry any service-specific data. Rather, it defines the user/network interface by means of various fields:

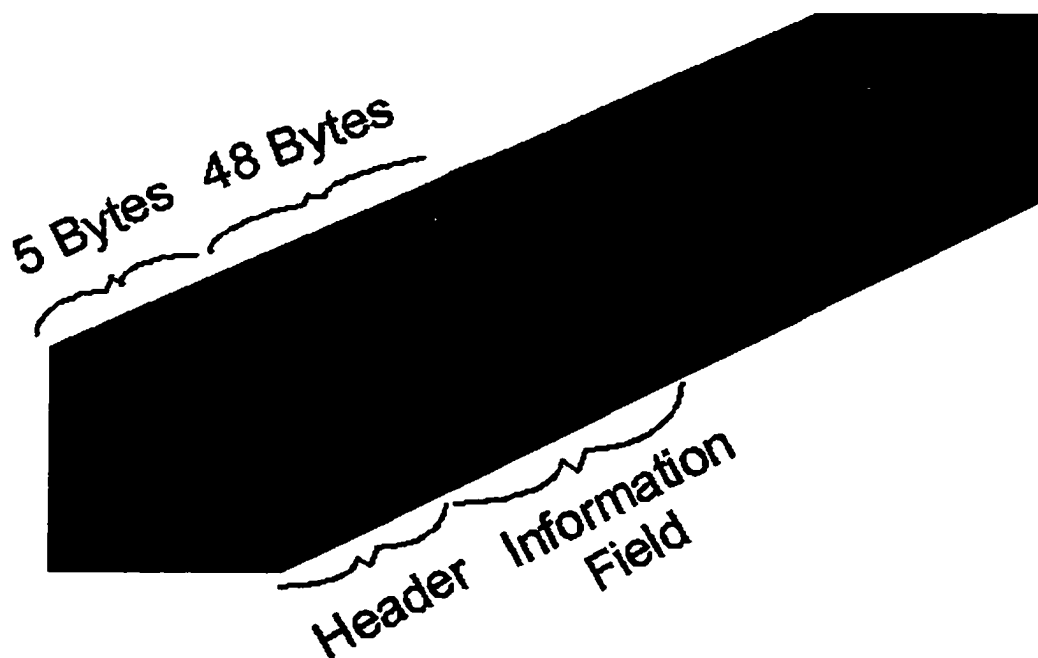
Generic flow control (GFC) – the four-bit GFC is used for end-to-end flow control.

Virtual channel identifier (VCI) – the VCI (sixteen bits) is similar to an X.25 virtual circuit; it defines a local logical connection between two ATM nodes.

Virtual path identifier (VPI) – the VPI (eight bits) is an aggregate of VCIs.

Payload type (PT) – the three-bit PT field indicates whether the cell payload contains user or network management information.

## The ATM Cell



**Figure 4.14: The ATM Cell.**

Cell loss priority (CLP) – the one-bit CLP field indicates whether a cell can be discarded if the network becomes congested. If the CLP is set to one, the cell is subject to discard.

Header error control (HEC) – the HEC uses an eight-bit error code to correct single-bit errors in the header, and to detect double-bit errors. Information field (48 bytes) carries the payload within the cell.

## **4.8.2.**

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### **ATM Standard Layers**

The ATM standard defines three layers:

#### **4.8.2.1. THE PHYSICAL LAYER**

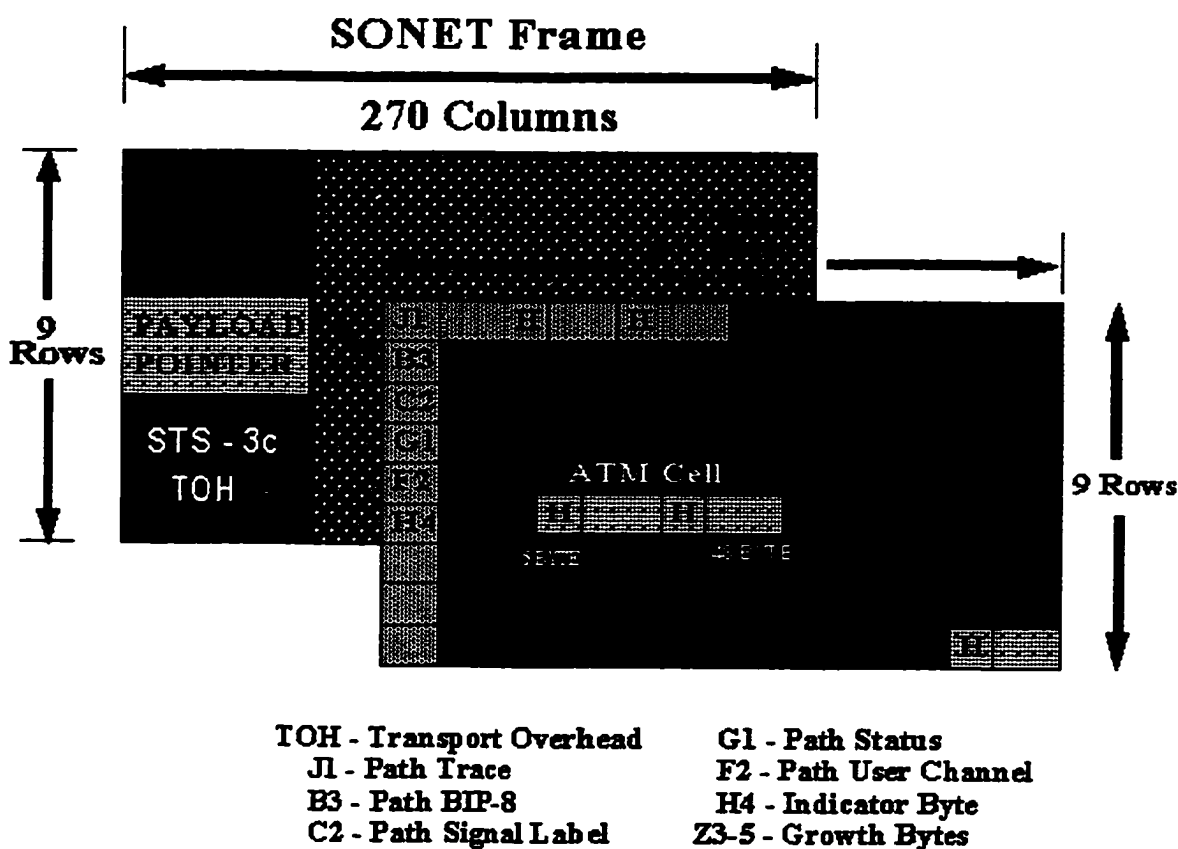
ATM cells may be transported over many different physical media and still maintain compatibility. The physical layer defines how ATM cell streams are transmitted over the physical media, as well as the interface to the ATM layer. Cells are transported within the ATM layer either asynchronously, as in packet switching, or synchronously, as payloads encapsulated in SONET envelopes (see figure 4.15).

#### **4.8.2.2. ATM LAYER**

The ATM layer provides the switching capability for the ATM cells by means of virtual connections. Two kinds of virtual connections have been standardized virtual channel connections and virtual path connections. By reading the VPI and VCI bits in the header of each cell, an ATM switch routes cells to their destination. Virtual channels are grouped together to form a virtual path. Many

virtual channels may share a single physical link at the same time. For example, all the virtual channels that belong to a customer may be bundled within a single virtual path, simplifying network management. A virtual channel may have other attributes such as quality of service, associated with it. Should congestion occur, the ATM switch selectively drops cells until the congestion clears. The selection of which cells to lose is based on the guaranteed quality of service.

## ATM MAPPING TO SONET/SDH PAYLOAD

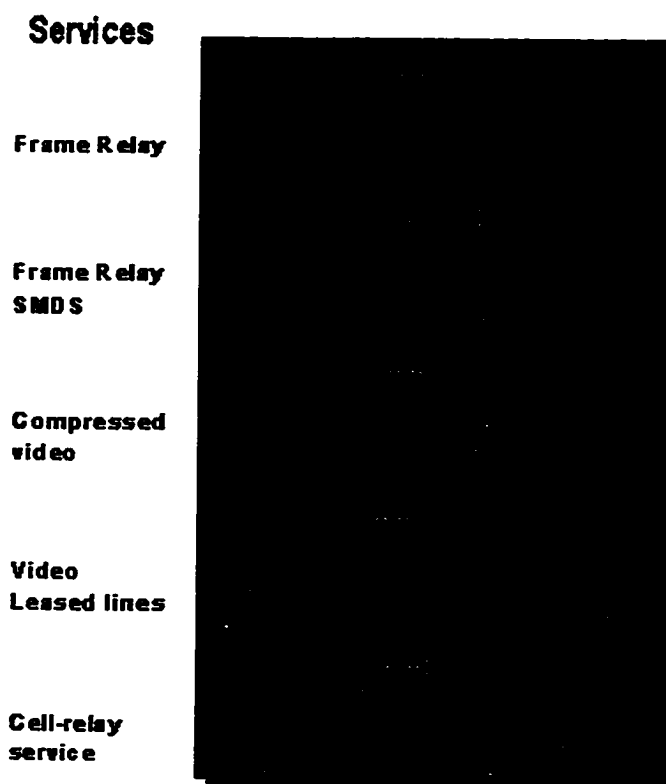


**Figure 4.15: ATM Mapping to SONET/SDH Payload**

### 4.8.2.3. ATM ADAPTATION LAYER (AAL)

Above the cell switching layer are ATM adaptation layers (AAL), which map various kinds of traffic into and out of the cells (Figure 4.16: ATM Adaptation Layers).

## ATM Adaptation Layers



CPCS = Common-part convergence sublayer

CS = Convergence sublayer

SAR = Segmentation and reassembly

SMD S = Switched Multimegabit Data Service

SSCS = Service-specific convergence sublayer

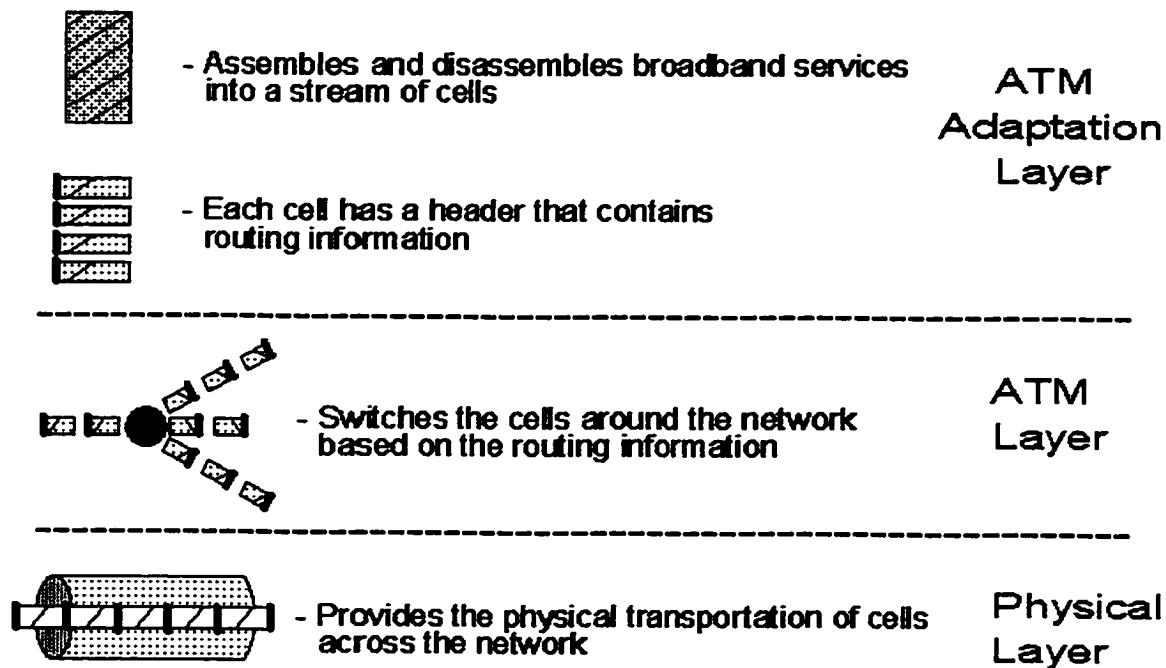
**Figure 4.16: ATM Adaptation Layers;** AAL1: A constant bit rate service for voice circuit emulation, AAL2: A variable bit rate service for audio and video, AAL3: Connection oriented services for data, AAL4: Connectionless services for data, AAL5: Each AAL handles a different type of traffic. AAL5, for example, could bring ATM to the desktop for applications such as distributed data base services and computer-aided design and manufacturing. In effect, high speed multimedia applications will be cheaper and easier to bring to the desktop computer.

The adaptation layers must differentiate between data, voice, and video traffic because of their very different transmission requirements. There are five types of AAL as defined in the ATM Standard that are specialized to types of traffic .

Feature	AAL1	AAL2	AAL3	AAL4	AAL5
Timing relation between source and destination	Yes	Yes	No	No	No
Bit Rate	Constant	Variable	Variable	Variable	Variable
Connection oriented mode	Yes	Yes	Yes	No	Yes

**Table: ATM Adaptation Layer Types.**

## ATM - A Layered Standard



**Figure 4.17: ATM - A Layered Standard.**

There are five types of AAL defined in the ATM Standard. These are based on

- o Whether a timing relationship must be maintained between source and destination.
- o Whether the application requires a constant bit-rate.
- o Whether the transfer is connection oriented.

The ATM Adaptation Layer continues to evolve. The earlier AAL3/4 supports native BISDN, while the later, AAL5 was constructed to support the use of BISDN for the transport of existing protocol services at a higher performance level. AAL5 Services include:

- o Notification of corrupted received AAL protocol data units (PDUs)
- o Unverified data transfer (error recovery in higher layers)
- o PDU transfer from one AAL to another

AAL5 could bring ATM to the desktop for distributed data base services, computer-aided design, and for manufacturing, and could facilitate high speed multimedia applications.

To provide a particular service, ATM maps the service into the information field of a cell. When the cell is full, the correct VCI/VPI information is placed in the header field and the cell enters the ATM cell stream. The AALs have two logical sublayers known as the convergence sublayer (CS) and the segmentation and Reassembly sublayer (SAR) that support this process. The CS accepts data units from the ATM user interface and delivers the data units back after receiving them from the SAR. The SAR divides each data unit into cells on the

segmentation side, and reconstructs incoming cells into data units on the Reassembly side.

The CS ensures that the different types of traffic receive the right level of service at the user-to-network interface (UNI) and at the network-to-network interface (NNI). The CS passes PDUs on to the network from the UNI and delivers PDUs back after reception. The SAR sublayer divides each PDU into cells during segmentation and reconstructs incoming cells into PDU during reassembly. When transmitting into the network, the CS appends a PDU trailer and pads the PDU plus trailer in multiples of 48 bytes. The trailer contains a cyclic redundancy check field (CRC-32), a PDU length field, and a control field [53]. Cells formed during the AAL layer segmentation process will all be 48 bytes. The SAR reassembles incoming cells into complete PDUs before passing them to the CS. Information for the reassembly is obtained from a combination of the VPI/VCI fields and the end-of-message indicator encoded in the PT field. After the CRC-32 is verified, the PDU is delivered to the ATM user.

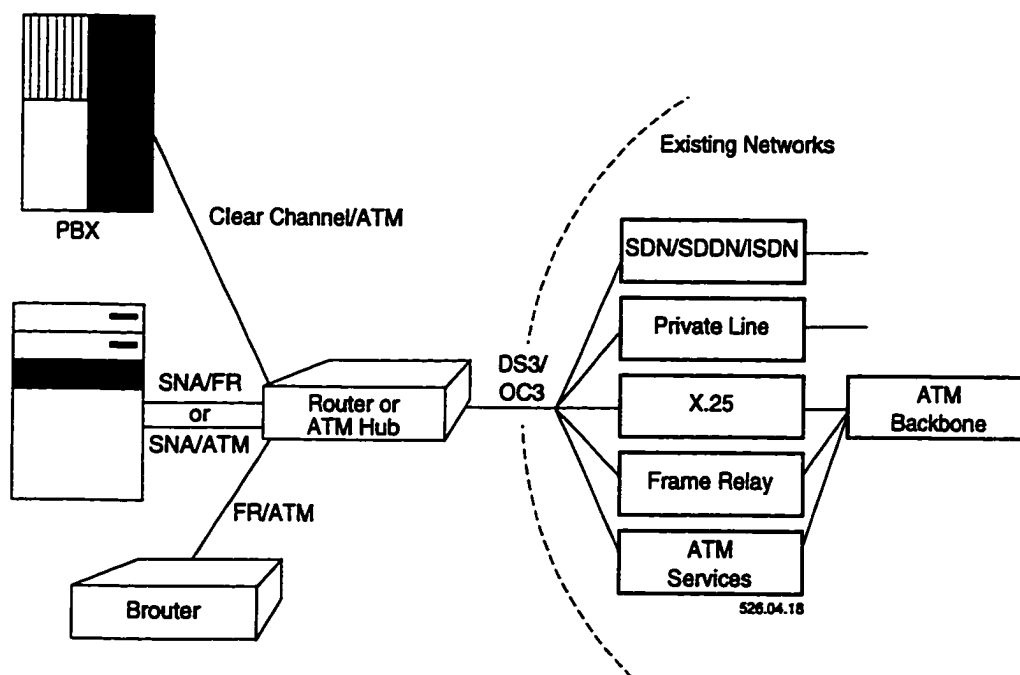
#### **4.9.**

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## **CONCLUSION**

With hundreds of millions of desktop computers connected by worldwide LANs, it was natural for the LANs themselves to be interconnected, first in geographically close areas, later in metropolises, finally in nations and around the globe. Not just LAN computer traffic was infused into the WAN, but computer technology as well. The LAN topologies, high bandwidths, stripped

down protocols, open architectures, and management were adopted by the WAN. New broadband technologies such as frame and cell relay, SONET/SDH, and ATM have left the standard bodies and entered the Telco/PTT networks. ATM, for example, allows the Telco/PTT to tailor a variety of services to the specific needs of business and residential customers. Now the same technologies, honed in the WAN, are returning to the LAN mainstream to route information at unprecedented speeds.



**Figure 4.18 : Evolving Broadband Network; ATM and ISDN**

SNA:	SYSTEMS NETWORK ARCHITECTURE
FR:	FRAME RELAY
SDN:	SYNCHRONOUS DIGITAL HIERARCHY
SDDN:	SOFTWARE DEFINED DATA NETWORK
ISDN:	INTEGRATED SERVICES DIGITAL NETWORK

Many approaches to networking are hot now. Most are limited by their inability to integrate voice, data, and video. High bandwidth add-drop multiplexing SONET pipes for transport with built-in network management and self-describing features with superfast ATM routers hold the greatest promise for the future. X.25, SNA, Frame relay, and TCP/IP protocols will still be used for access to applications at the edge of the network. They will be replaced by ATM backbone with SONET transport and smart network management systems with distributed functionality will replace today's relatively simple operations support systems. These will be replaced in the very far future by photonic networks, the SONET/ATM networks offer the greatest hope of embedding network management and handling the diverse needs of an "on screen" society.

BISDN is the most advanced technology for voice video, and data transfer. Fiber optic networks allow for high performance, reliable, and secure data transfer. The Broadband interface should be based on ATM and to be built one SONET framed interfaces. ATM provides the technology to integrate a wide variety of information into multimedia services by supporting voice, video, and data transfer services separately. It's time to give to ATM the predominant role in B-ISDN.

## Chapter 5

### **PROTOCOLS AND INTERFACES**

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Network environments are expanding at an alarming rate. The impact of the new technologies and their implementation is likely to influence everyone in the society. These networks have started to bring about an immense wealth of information to every household. The revolution within our intellectual and personal lives can be dramatic for those who can exploit these technologies. But the impact on telecommunication service providers can only mean an intense competition to survive in a complex technological environment of an all digital and mostly programmed communication systems. The rate of progress is accelerating in the 1990s. Providing access to these all digital networks of the future offer the most candid area of technical challenge. Realizing the innovations as world wide standards for the new devices, networks is the next step.

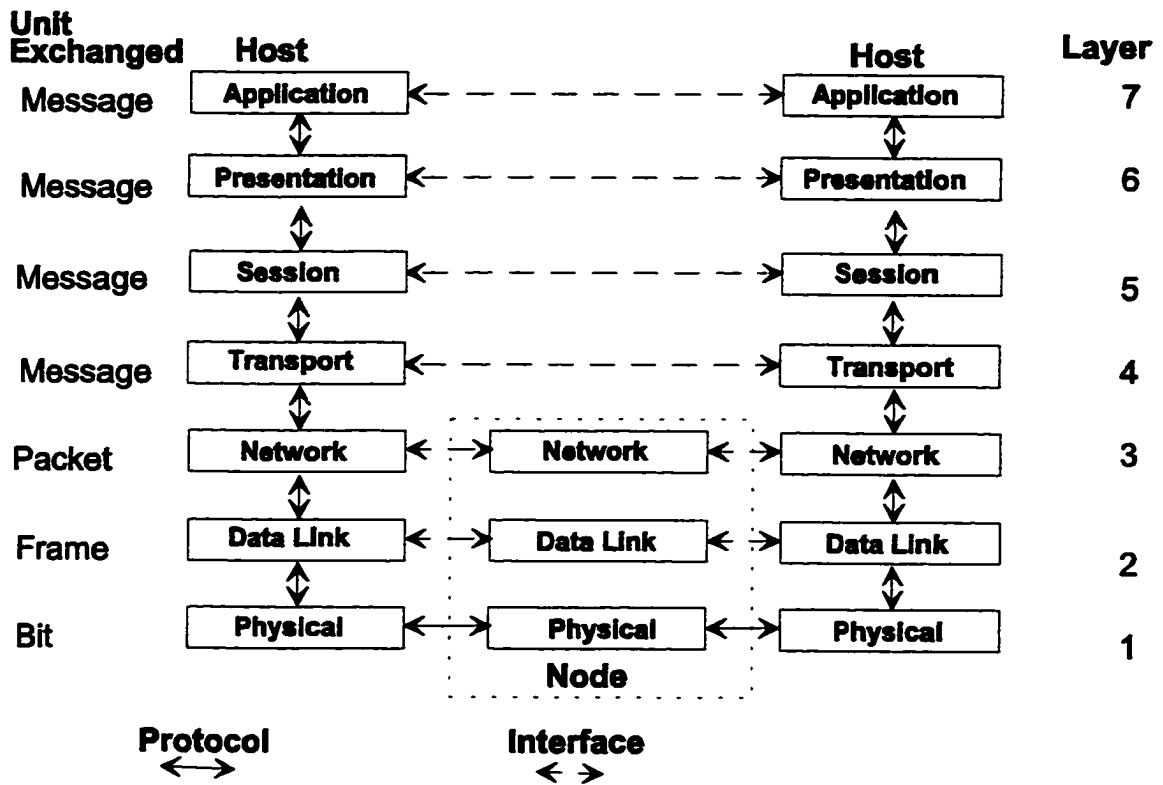
## 5.1.

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# PROTOCOLS AND INTERFACES IN COMPUTER NETWORKS

A computer network is a collection of host computers that are interconnected, autonomous, and mutually suspicious (controlled by itself). In any network, there are hosts which run application programs. These hosts communicate with each other through the communications subnet. The communication subnet may consist of a set of point-to-point channels connecting switching elements that are known as nodes, or a single broadcast channel (bus, ring, loop, star, or satellite/radio) [6, 40].

Most networks are designed as a set of layers. Each offers a class of transparent services to the layer above. Each layer on a given host establishes a virtual communication with the associated layer on another host. An individual layer, within a host, receives a message from the next highest level and envelops the message with required information for the corresponding layer in the next host. The expanded message is subsequently passed to the next lower layer. Finally, at the lowest layer, the message is physically transmitted. In the targeted host, each layer strips off the information transmitted by its counterpart and sends the remainder to the next layer. At the lowest layer, physical communication is established with the corresponding host, whereas the virtual communication is used by the higher layers.



**Figure 5.1: Open System Interconnection Reference model**

A protocol is the set of rules which facilitate (i.e., allow) communications to occur. Protocols are needed when communication between two modems, a DTE and a node, or Customer's Premises equipment and a Local Exchange in ISDN.

Many organizations are active in defining standards such as the International Telegraph and Telephone Consultative Committee (ITU-T), the International Organization for Standards (ISO), the American National Standards Institute (ANSI). the IEEE (Professional Society) 802 committee is active in defining LAN standards, and the EIA (group of manufacturers and users in defining interface standards. Government Organization such as FIPS, FED-STD, MIL-STD, and

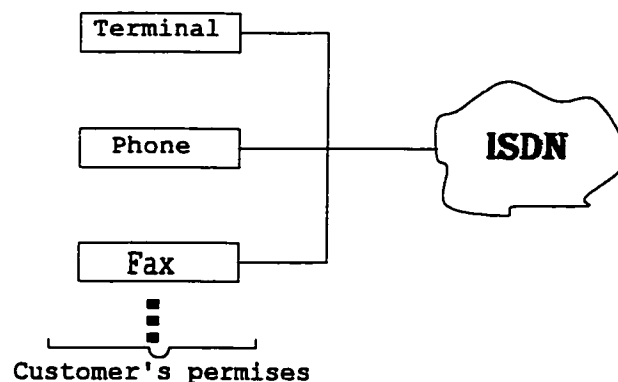
GOSIP are also active in defining protocol and interface standards. (Corporation for Open Systems (COS)).

## 5.2.

### INTEGRATED SERVICES DIGITAL NETWORKS

Digital switching and transmission offer the advantage of no degradation in signal quality over long distances, equipment reliability, ease of maintenance, the accommodation in one network of both voice and data traffic, and improved interoffice signaling. The idea of extending digital access to the end user became an obvious one, although difficult to achieve in practice, that has become imbedded in the concept of the integrated services digital network (ISDN), now series of ITU-T recommendations.

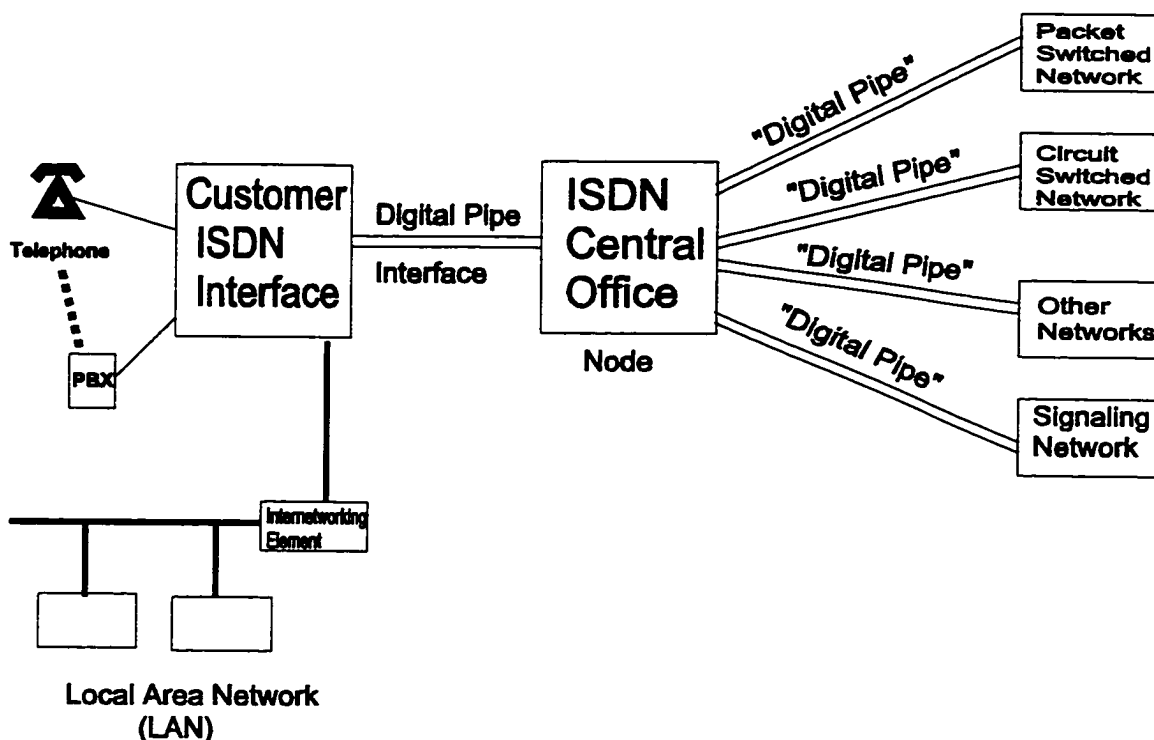
ISDN is an evolving set of international standards for connecting voice, data, and video equipment giving the end user direct digital access to the public telecommunications network, at high digital transmission rates at reasonable cost.



**Figure 5.2: ISDN Customers**

An end user can take advantage of the integrated services and obtain local, long distance, or global packet-switched or circuit-switched connection.

ISDN have been used for major telecommunications conferences. Today the public has access to using ISDN lines, but still is not taking advantage of its real features. However, the world of telecommunication is working on introducing ISDN in a wide network.



**Figure 5.3: Conceptual View of ISDN Connection Features**

### 5.2.1.

## Goals and Principle of ISDN

Above all ISDN is an enduring solution for International Telecommunication.

The goal is to ensure that networks, terminal equipment and service providing

vehicles, function compatibility and uniformly to provide a wide range of voice, data, and video services. Standards for ISDN are thus a key ingredient for success. The ISDN solution has been evolving from the existing voice and data networks. Using wiring as a starting point while providing for future capabilities such as optical transmission which is in use at current stage in many regions. All ISDN stations within a class of service (e.g., Primary or Basic Rate bearer services) can be satisfied by a single connector. Terminal Adapters must be available to adapt existing terminal equipment to operate on the ISDN.

Some of the principals of the ISDN are:

- End-to-end digital connectivity.
- Support voice and non voice applications.
- Well defined set of services.
- Limited set of user-network interfaces.
- Limited set of network-network interfaces.
- Layered Interfaces definitions per OSI.
- Switched and non-switched services.
- Circuit and packet mode switching.

### **5.2.2.**

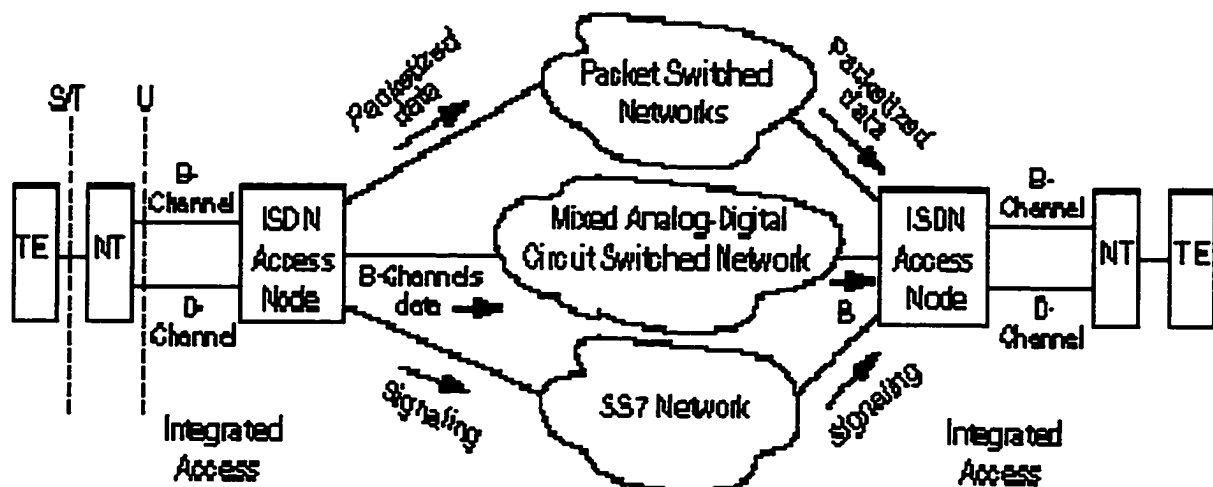
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## **ISDN and Digital telephone Environments**

The fundamental architectural postulates of ISDN are:

- separation of information traffic from signaling traffic,

- integrated digital access from subscriber to ISDN switches (the different media are not necessary carried as integrated traffic within the interoffice network),
- well defined addressing and signaling protocols, giving end users the option of “functional” signaling in which descriptive messages convey instructions, rather than the “simulus” signaling that subscribers use in the telephone network, in which on hook-off hook and a limited number of push button actions have different meanings on the status of a call.



**Figure 5.4: ISDN Architecture Showing the Separation of Signaling Packet and Circuit Switched Data.** NT = Network Termination. TE= Terminal Equipment.

Digital telephone environments allow for transmission, switching, routing and storage to be accomplished in the digital mode. The channel control of voice and speech channels is also digital. The encoding and decoding of speech are both

done at the telephone set. Even low bit rate speech (such as stored speech messages and announcements) is converted to analog form at the telephone set. All the inherent advantages of digital telephone computing and processing techniques can be imported in the digital telephone networks. In a sense, the ISDN framework blends digital telephone network features with other digital services such as packet-switching, message storage, forwarding, automated message selection, etc.

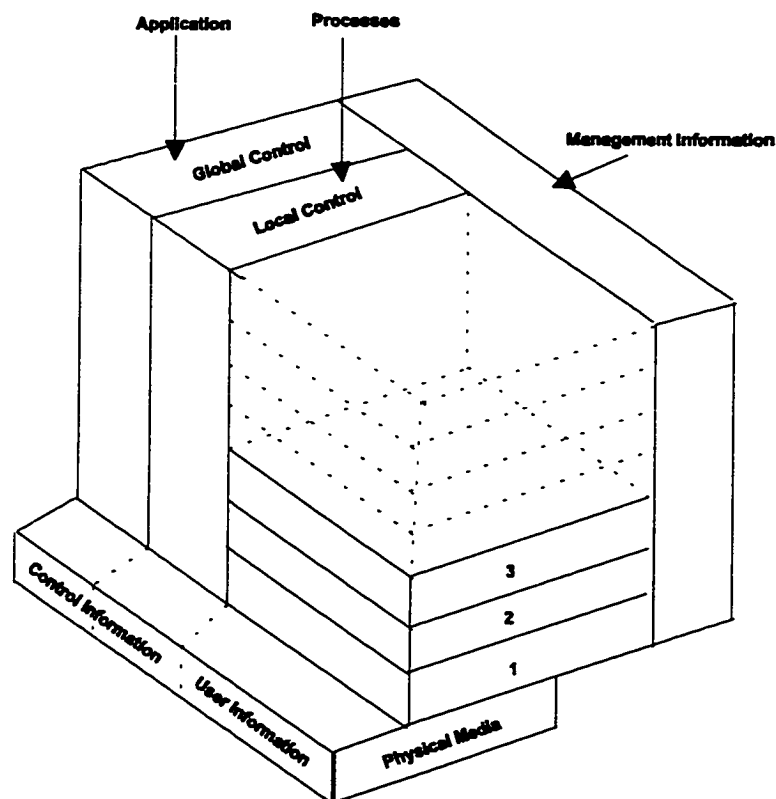
ISDN as a Universal All-purpose Digital Services Network: Signaling is accomplished over the SS7 network fairly well and adopted in the United States and Europe. This network uses the out-of-band information (i.e., the D channel), to control and signal the various switches to complete and monitor the B channel. Figure 5.4 depicts the role of signaling and control of the information bearing B channels.

This network becomes essential in the circuit-switched context because the B-channels provide transparent end-to-end digital connectivity for the channel of the BRISDN. If the transition from the SS7 signaling information is provided for the transition and vice versa. BISDN and N-ISDN functions also depend upon the separation of the customer data in the 'B' channel from the control and packet data in the 'D' channel.

### 5.2.3.

## ISDN Protocol Reference Model (connection control)

The overall ISDN Generic Protocol Block architecture is shown in figure 5.5, representing three types of information with parallel protocol layering. The three types of information are control, user, and management, each of which is communicated independently of the others. The logical separation of the ISDN control information from user information through all layers stems from this ISDN protocol reference model. In contrast, the OSI model allows the separation of control from user information, but the OSI does not require such separation.



**Figure 5.5: Generic Protocol Block**

ISDN is designed to serve multiple applications, including voice, video, and data. Therefore, it is not practical for all layers to always provide functionality. While roughly modeled after OSI, the layers are explicitly not named. Adjacent layers within a type of information communicate using service primitives. However, if a layer is empty, the primitives are mapped directly onto primitives for the next nonempty layer.

In the *blue book* version of the ISDN protocol reference model (recommendation I.320), the concept of local control and global control are introduced as refinements of the ISDN information. Local control (L.C.) is control used between (logically) adjacent peer entities (with the same layer of protocol). Global control (GC) is used between two arbitrary peer entities where these entities are expected to be “remote” from each other. Coordination among all information types is accomplished using the management information type.

### **5.2.3.1. PHYSICAL LAYER**

The logical separation of the bearer or B channels from each other and from the Delta or D channel make ISDN feasible. The B channels carry customer data throughout the network, and the D channel carry more than just the signaling bits to control and route the B channels. The ISDN line (physical layer) in its simplest configuration has two interfaces which are the basic rate interface and the primary rate interface. In both interfaces, layer primitives are defined whose primary function is to activate and deactivate the physical layer, providing service to the link layer.

## **BASIC RATE INTERFACE**

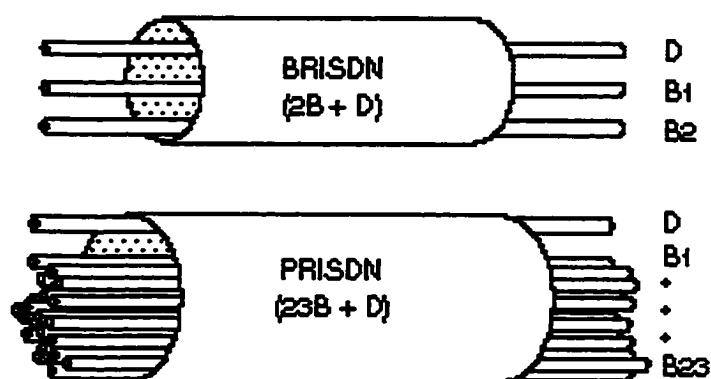
The BRI is a single access point into ISDN. It's also known as DSL (Digital Subscriber Line) and represent the customer premises interface (CPI). It is defined in Recommendation I.430 as 192 kbps with well defined framing. The most common structure on BRISDN operates at two bearer channels operating at 64 kbps each, and a single delta channel operating at 16 Kbps. The B channels are used to carry information with no restrictions on the format or type of the data that passes through (so the name clear channel). The D channel is used for signal and control information. At least two features in the basic rate interface are worthy of note with respect to higher layer protocols and services:

- 1) Customer premises multipoint configurations are supported ("Passive Bus") allowing contention for the use of the local (intrapremises) transmission facility.
- 2) D-channel signaling messages are given priority over other D-channel messages as a matter of policy governing the interactions between layer 2 and layer 1.

## **PRIMARY RATE INTERFACE**

PRI is used to connect multiple users to ISDN. It is Known as EDSL (Extended Digital Subscriber Lines)and used to connect PBX, LAN, or other multi-user switching device to an ISDN network. The primary rate interface physical layer is defined in recommendation I.431 as either 1544 kbps or 2048 kbps, each of which has well-defined framing. The North America standard, followed by the

United States, Canada, Mexico, Japan, and south Korea, consists of 23 B channels of 64Kbps each, and one D channel of 64 Kbps. The aggregate capacity is 1.5544 Mbps, or equivalent bandwidth of a T-1 facility. The European standard for PRI consists of 30 B channels and 1 D channel, for an aggregate capacity of 2.048 Mbps.



**Figure 5.6: Two of the Major ISDN Access in the United States.** The Primary Rate ISDN in Europe has (30B+D) channel configuration. Bit rate (in Europe) : 2.048 Mbps; Composition: 30B+D (B at 64kbps, and D at 64kbps). The Basic Rate ISDN has (2B+D) channel configuration; bit rate: 144 kbps; composition: B+B+D (+Synchronization and Framing)

Because of the capacity available under PRI, it supports an additional type of channels, H. Three types of H channels are specified:

H0 = 384 Kbps. User-to-User information transfer.

H11 = 1.536 Mbps. User-to-User information transfer.

H12 = 1.920 Mbps. User-to-User information transfer.

In North America standard incorporates H0 and H11 channels, while the European standard incorporates H0 and H12.

On the primary rate interface, only point-to-point configuration of terminating equipment are supported, and no priority schemes exists on D-channel messaging.

### **5.2.3.2. LINK LAYER**

The purpose of ISDN Link Access Procedure on the D channel (LAPD) is to provide the OSI layer 2 functionality of guaranteeing data integrity of information passed by layer 2 over the ISDN user-network interface. The general aspects of ISDN user-network interface link layer are described in Recommendation I.440 (duplicated in Q.920), including the layer services expected from the physical layer and provided to the network layer with the associated primitives. The link layer service is based on OSI's Data Link Layer Service, providing efficient and timely data transfer, link synchronization, error detection and correction, flow control, frame sequencing, and addressing for multiplexing.

Link layer procedures for LAPD are given in recommendation I.441 (duplicated in Q.921). LAPD evolved from a basis in the recommendation X.25 layer 2 protocol called LAPB. In particular, the ISDN LAPD protocol supports multiple layer 3 entities and multiple terminal equipments on one interface using octets of addressing via Data Link Connection Identifiers (DLCI's). DLCI's have two octets, the Service Access Point Identifier (SAPI) in the high-order octet, and the

Terminal Endpoint Identifier (TEI) in the low-order octet. LAPD supports multiple frame operation with modulo 128 for point-to-point information transfer associated with most call control. Unacknowledgement information procedures in LAPD handle broadcast information transfer for initial call setup to a passive bus from the network.

### **5.2.3.3. LAYER 3**

Recommendation I.450 (duplicated in Q.930) describes general aspects of the ISDN layer 3 protocol. This recommendation explains the services expected from the data link layer by layer 3, and lists the peer functions of the layer 3, for example, network connection control and conveying user-to-network and network-to-user information.

The layer 3 protocol, specified in Recommendation I.451 (duplicated in Q.931), is the key to ISDN call control. While the procedures contained in these procedures have been designed to allow extensions to provide a wide variety of more complex calls, supplementary services and network specific services (Rec.932).

## **5.2.4.**

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### **ISDN Equipments**

The CCITT has defined several different types of ISDN equipment (figure 5.7):

*TE-1* equipment is ISDN-compatible and can be connected directly to the network.

*TE-2* equipment isn't ISDN-compatible and requires an interface device known as *terminal adapter (TA)*.

A TA can convert signals from one international standard, such as RS-232C, to the ISDN standard.

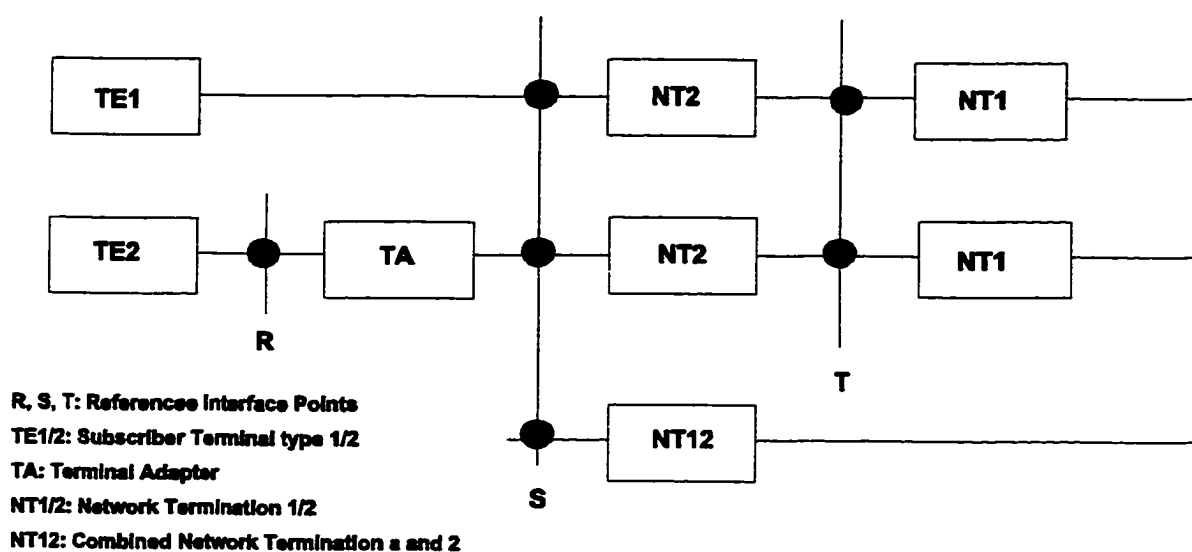
Network termination equipment can take two forms under ISDN. *NT1* describes public switched network demarcation devices, such as a termination block or a registered jack. This equipment will have some built-in intelligence under ISDN because of the functions it must perform. *NT2* is the designation for customer-owned switching equipment, such as a PBX or a LAN. *NT2* can provide additional capabilities beyond *NT1*, such as call switching or concentration.

*Line-termination equipment (LT)* is located within the local exchange company's or common carrier's network in situations where lines must be extended beyond the normal range of the central office.

*Exchange termination equipment (ET)* terminates the digital subscriber line or extended digital subscriber line in the local exchange. *ET* can be characterized as central office equipment, important to the phone company but nothing you need to worry about.

The definition of functional grouping also define, by implication, the reference points (R, S, T, and U) at the customer premises (Figure 5.7):

*Reference point S (system)*, and corresponds to the interface of individual ISDN terminals.



**Figure 5.7: ISDN reference points and functional groupings.**

*Reference point T* (terminal) corresponds to a minimal ISDN network termination at the customer's premises. It separates the network provider's equipment from the user's equipment.

*Reference point R* (rate) provides a non ISDN interface between user equipment that is not ISDN-compatible and adapter equipment. Typically this interface will comply with an older interface standard, such as EIA-232-D.

The transmission line ends at the *U* reference points.

Reference Points	Basic Access	Primary Access
R	Existing interfaces (e.g., RS-232)	Same as Basic
S	4-wire, 144kbps/(2B+D), 192kbps/(2b+D+overhead)	Same as basic
T	Same as S for basic	4-wire, 1.54 Mbps (23B+D+overhead)
U	2-wire, echo canceler	4-wire transmission system (e.g., T1 line)

**Table 5.1: ISDN reference points**

## 5.2.5.

### Functions of an ISDN-exchange

The Functions of an ISDN Exchange can be subdivided into those performed at the subscriber's premises and those performed at the site of the exchange itself (see figure 5.8).

#### 5.2.5.1. FUNCTIONS AT THE SUBSCRIBER'S PREMISES NETWORK TERMINATION

The ISDN subscriber network is terminated in the network termination (NT). It provides the transmission interface to either the ISDN terminals or to the terminal adapters.

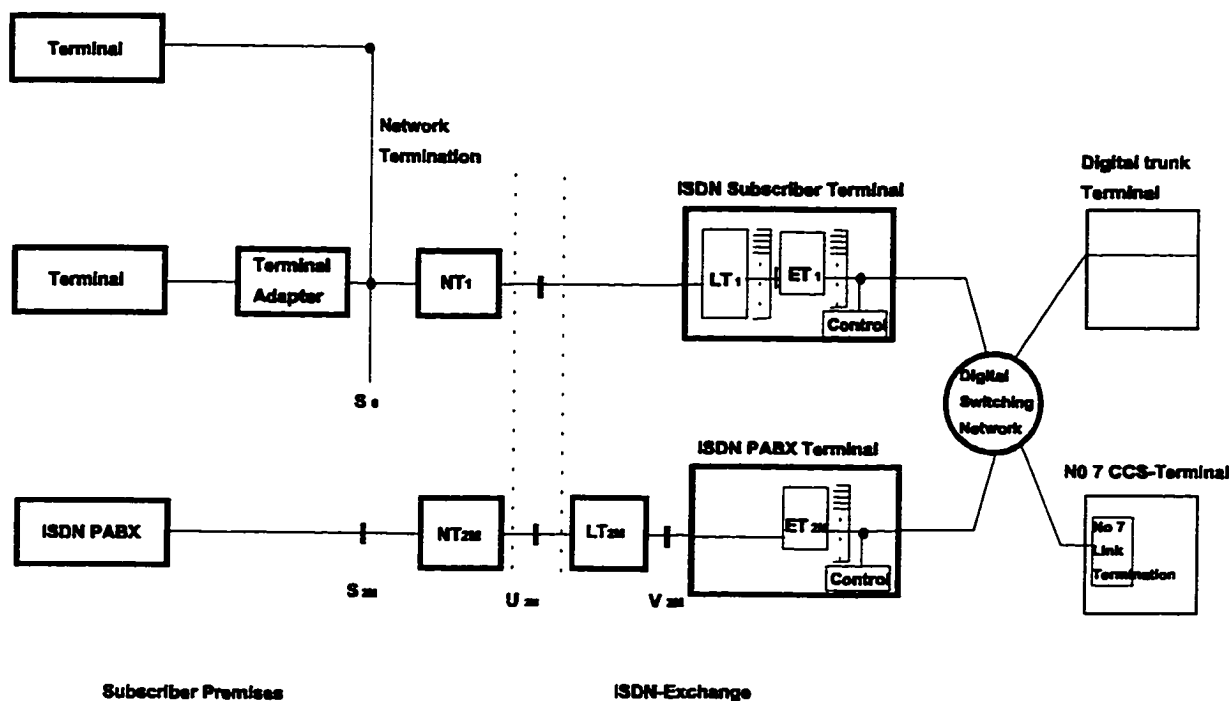


Figure 5.8: ISDN Exchange with Basic Access and Primary Rate Access.

## **THE S-INTERFACE**

Also called *Reference point S (system)*, and corresponds to the interface of individual ISDN terminals. It separates user terminal equipment from network-related communications functions.

In the case of a basic access, i.e. the ISDN subscriber has access to two B and one D channel, the NT supplies the standardized S0 Interface to the terminal side. Thus S0 Interface is normally realized as a 4-wire passive bus to which up to eighth ISDN devices can be connected.

The bit rate of the user information (B and D channels) crossing the S0 interface is, however, 192 kbps in order to provide further control information for:

- Frame alignment. All channels are transmitted in a frame-multiplexed structure with 48 bits every 250 microseconds.
- Activation and deactivation. In order to reduce power consumption, the NT goes to a power down mode, where only a limited set of functions, e.g. recognition of activation information, are executed. The NT may be activated both from terminals and from the exchanges.
- Balancing. The interface uses a modified AMI code, i.e. logic one is represented by an electrical Zero signal, logic zero is represented by alternating positive and negative signals. Balancing channels ensure that an equal number of positive and negative signals cross the S0 interface to make it DC-free. Since a terminal receives the complete frame, from NT to terminal

the entire frame is balanced. In the direction of the terminal to the NT each B-channel is also balanced, since one terminal only can send information on one B channel simultaneously.

- Echo channel. Used for resolution of D-channel access contentions.

### THE U-INTERFACE

Also called *Reference point U*. In the case of basic access, the transmission interface to the subscriber line side -U-interface- is based in most European countries on a ternary code, where four bits of the binary data stream are coded to three ternary line signals (4B/3T-coding). Unused ternary combinations are used for transmission error monitoring.

Another standard for coding the line signals is to encode two bits one quaternary signal. This method is preferred by the USA telecommunications providers.

In the case of 4B/3T-coding, the transmission is realized by introducing a frame with 120 ternary signals each with a duration of 1 ms. Of these, 108 ternary signals per frame are required for the two B and one D channel, 1 ternary signal is used for synchronization, and one ternary signal is used to constitute a service and maintenance channel (M-channel).

The transmission of the ternary signals is carried out over a 2-wire subscriber line. Using hybrid in the exchange and the NT. Due to mismatches in the line-balancing circuit from the hybrid and possible mismatches of the line impedance,

echo are generated. To make sure that a secure transmission via existing copper wires is possible, the echo cancellation principle is applied.

### **FUNCTION OF THE NETWORK TERMINATION (NT)**

The main tasks of the NT are to translate the data flow from/to the S0 interface and to respond to the additional signals crossing the S0 and Uk0 interfaces.

Besides the following functions:

- Power feeding in a normal condition. Up to four telephonic terminals are fed by the NT, which takes the power from the main supply.
- Power feeding in a restricted condition, i.e. mains supply failed. One telephonic terminal is fed via Uk0 interface, i.e. by the exchange batteries.
- Resolution of the D channel access of the D channel to all terminals connected with the S bus.
- regeneration of the line clock (Uk0) and derivation of the clock for the S0 interface.
- Switching of a test loop. This is done under the control of the exchange using the M-channel of the Uk0 interface.

### **5.2.5.2. FUNCTION WITHIN AN ISDN EXCHANGE**

The functions required to establish an ISDN call can be allocated to different levels of abstraction, so-called layers according to the OSI Reference Model. Consider the call setup of a call, using the D-channel protocol (signaling messages between terminal and exchange).

Layer 1, the physical layer, has to transport the data stream between terminals and exchange. This layer is basically realised by the S0 and Sk0 interfaces. Layer 1 functions in the exchange are performed by the Line Termination (LT). In general, the transported information is not recognized by the layer 1 function. However, there are exception commands and information sent over the M channel.

Layer 2, the data link layer, is related to the D channel and has the task of the D-channel messages; it does not, however, process them. It also uses layer 1 as a support function. The layer 2 functions are performed in the terminal and within the exchange.

Layer 3, the network functions, supervises the correct sequencing of D-channel messages and triggers the processing of the messages to set up and release the calls.

### **5.3.**

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## **BROADBAND INTEGRATED DIGITAL NETWORK**

The idea behind ISDN is to digitize voice and data transmission signals right at the subscriber's home office, combining them into a single stream of bits that could be switched by telephone network just as digital phone signals are now.

### **5.3.1.**

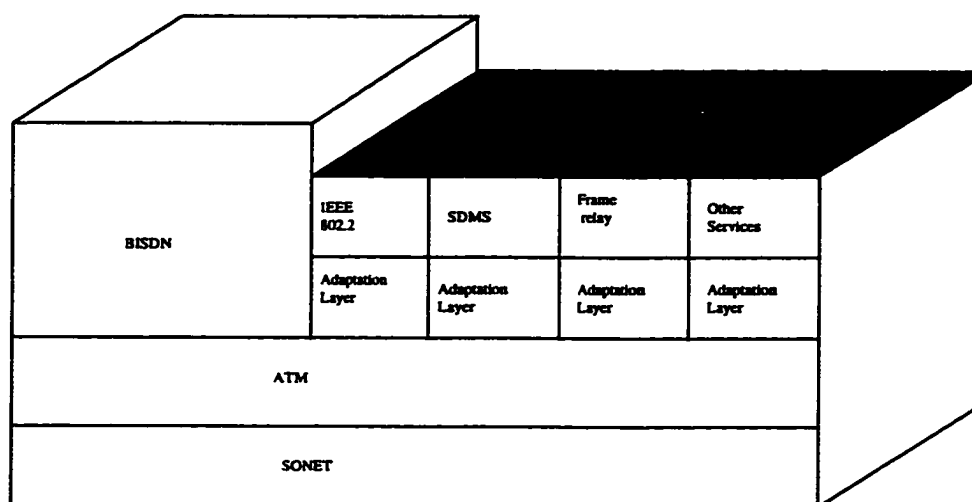
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## **Concept behind B-ISDN**

BISDN is a service or System requiring transmission channels capable of supporting rates greater than the primary rate .

D channel is used for signaling (control and maintenance). The D channel protocol is defined by CCITT Q.921 specification or LAPD ( Local Access Protocol for D channel), This protocol is used for signaling and information Transfer. It is similar to the protocol defined for Packet data in X.25 (LAPB). However, it allows for more than one logical link between end points because of the difference in nature of the B channel information, which is targeted to one end point. In instances where the D channel carries X.25 data together with signaling, the LAPD protocol is used protocol is used (at the data link layer, i.e., layer 2 of 7-layer ISO model) for both logical channels - one reserved for signaling and one for X.25 data.

Initial B-ISDN interfaces provide data-rate transmission of 51,155, or 622 Mbps over fiber-optic media. The following figure shows that the physical layer support for B-ISDN is provided by ATM and SONET.



**Figure 5.9: B-ISDN support infrastructure.**

In the definition of ISDN, CCITT has placed certain requirements on the functionality. ISDN is a digital network utilizing the existing telephone networks around the world to provide end-to-end digital connectivity. A wide range of services including voice and non-voice services are to be provided by using a limited set of standard multipurpose user-network interfaces.

Basic rate interface is accepted at 144 kbps with (2B+D) format. This interface provides a worldwide standard to the users accessing the two B or less channels at 64 kbps each and one D channel or Delta channel at 16 kbps. The 144 kbps information bit rate needs 192 gross transmission rate. At H0 interface, the information rate is at 384 kbps, and is generally expected to have (5B+D) format.

At the primary rate interface, two channels are accepted to provide the network access: H11 [with an information rate of 1540 kbps, a gross bit rate of 1.544 Mbps with (23B+D) format], or H12 [with an information rate of 1920 kbps, a gross rate 2048 kbps with (30B+D) format]. The D channel for the primary rate is at 64 kbps. The H11 channel may also be recomposed with (5H0+D) format. A certain flexibility exists in the hierarchical of the B and D channels.

These rates, with multiple B and a single channel, suffice for the conventional applications and over metallic interfaces where the transmission generally takes over conditioned or unconditioned twisted wire pairs. However, much higher bit rates are feasible over fiber and are desirable for multimedia services that ISDN is required to offer to the customers. Hence, the integration of the broadband services within ISDN also becomes a necessity. For this reason, BISDN is a potential scenario for the future, and the evolving networks have to interface and content with all the rates for ISDN (basic, Primary, and Broadband). The ISDN access to the network is enabled by a single optical fiber to the subscriber in a star network topology. Restricted access is feasible with the bus or ring configuration for distributing broadband services, such as video and text retrieval. The societal push to move into multimedia services, video-conferencing, and (still and dynamic) graphic communication requires higher data rates than those supported by PRISDN services. For this reason, the need to interface with optical networks is projected. The BISDN capability will tie the

users with the backbone transport network at optical carrier (OC) rates from OC-1 to OC-192.

In the circuit-switched mode, high rates of several megabits for each customer are feasible at this time. In the packet-switched mode, new high-speed packet transfer techniques (such as ATM), with appropriate high-speed digital switching techniques, can also provide higher bit rates. High-rate data transmission and distribution, via circuit-, packet- or channel networks, provide for a large variety of interactive and/or video services such as:

- Broadband video telephony / video conference
- Video surveillance
- High-speed unrestricted data communication
- High-speed file transfer, telefax, tele-action
- Video document retrieval
- TV distribution with existing TV, extended, or HDTV

The asynchronous time division multiplexing framework provides a foundation for high-speed ATM network. Typically oriented toward the evolving synchronous optical networks, the ATM approach takes advantage of the ultra high-speed of fibers already in use. These networks depend upon the extremely low bit-error rates and the high switching rates of the optical channels.

For ATM to support Intelligent Network needs, switching and interface systems require special control and interface functionality. An investigation of the

switching systems, and the possibility to add ATM into them will be a first step in this thesis.

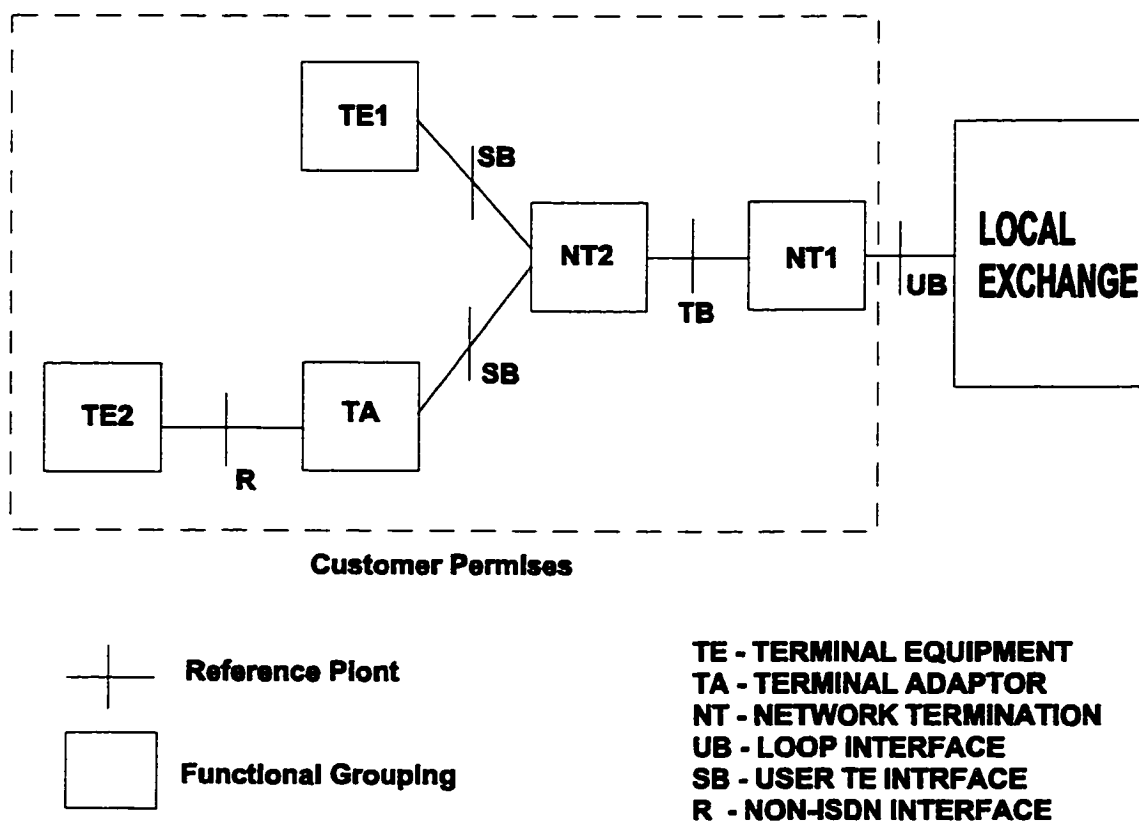
### **5.3.2.**

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## **Broadband ISDN Configuration**

Figure 5.10 provides a conceptual model for BISDN reference configuration which consists of functional grouping partitioned by reference point R,  $S_b$ , and  $T_b$ . The reference points can also be viewed as physical units. Each functional grouping has a different set of functions to perform. The TE functions, which include the lowest three layers of the OSI reference model, are usually performed by digital telephone, data terminals, and integrated workstations. The TE1 functions follow ISDN recommendations whereas the TE2 functions do not follow ISDN recommendations and use the TA converter to connect to an ISDN interface. The R reference point is for non-ISDN physical interfaces. User equipment complying with ISDN standards can be connected as the  $S_b$  and  $T_b$  reference points. The NT<sub>2</sub> functions are performed by, and terminal controllers. They contain local switching between the  $S_b$  interfaces with the  $T_b$  interface, multiplexing of several  $S_b$  interfaces at layer 2 and 3, and concentration functions. Note that the NT2 functional grouping can be removed from figure 5.10. For a specific configuration. In this case, a single interface would bring the  $S_b$  and  $T_b$  reference points together (hybrid interface) and the network would then perform all switching and routing functions. NT1 functions terminate the transmission line from the central office. The function of this grouping is to provide the

physical layer internetworking of the  $T_b$  and  $U_b$  interfaces. Although only one broadband  $T_b$  input is supported, multiple  $U_b$  interfaces can be supported to increase reliability. The  $U_b$  interface is the designation for the physical interface between the NT and the local exchange [41].



**Figure 5.10: B-ISDN Reference Points**

## 5.4.

### ASYNCHRONOUS TRANSFER MODE

Asynchronous Transfer Mode (ATM) is a connection oriented switching and asynchronous multiplexing methodology transporting fixed-length cells. The word asynchronous refers, in part, to the untimed relationship between the

sender and the receiver. Being a packet oriented technology, the cell (packet) has an information field for data and a header for identifying and routing the cell. Transport speeds are possible at 2.4 Gigabit rates and higher.

### **5.4.1.**

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## **ATM traffic Categories**

Circuit based services include ordinary voice telephone calls to high-speed video transfers. This constant Bit Rate (CBR) traffic is to be carried as an evenly spaced stream of cells.

Variable Bit Rate (VBR) traffic is generally generated by variable compressed or silence-suppressed circuit services. The more familiar packet and frame based data services also generates VBR traffic. These VBR services only generate cells at the rate necessary to carry the current traffic load.

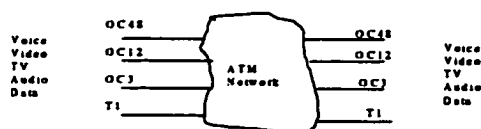
These are ways the ATM network can carry traffic from existing and planned user services, in which subscribers don't necessarily have a cell interface. But subscribers don't direct access to the basic cell layer of ATM networks. They then are able to send multiplexed cells to distinct destinations. This is similar to frame relay capability to send multiplexed frames to distinct destinations.

### **5.4.2.**

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## **ATM DEPLOYMENT**

As a public network telephone technology, ATM was suggested (in the proposal defense) to be the switching fabric that will bring greater transmission throughput, better transmission utilization, and faster switches speeds. ATM is now being enthusiastically pursued as a potential private network technology for both LAN and WAN use. At the same token, there is a large body of people who are eager to deploy ATM technology in private networks, even within a single site. Public network leased line facilities may interconnect these local ATM environments initially, until there is enough critical mass to make the deployment of public ATM services viable.



**Figure 5.11: ATM Network concept**

### 5.4.3.

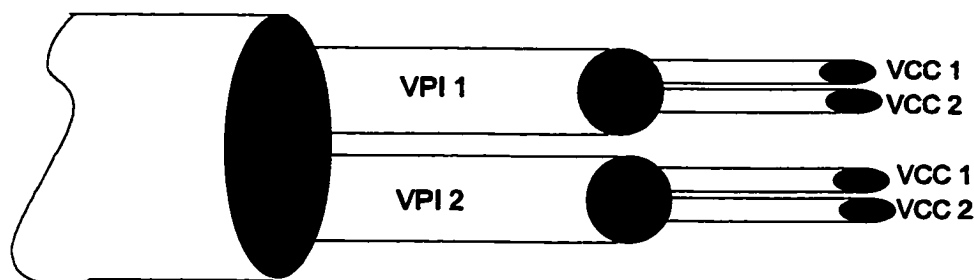
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## ATM Technology

Like frame relay, ATM is a connection-oriented technology that can be used to offer connection-oriented or connectionless services. All cells in a connection

follow the same path through the network. Thus cells arrive in the same order as which they were sent.

ATM defines two major types of virtual connections, *Virtual Path connections* (VPC) and *Virtual Channel Connections* (VCC). There are fields to identify both of these in the ATM header, called *Virtual Path Identifier* (VPI), and *Virtual Channel Identifier* (VCI).

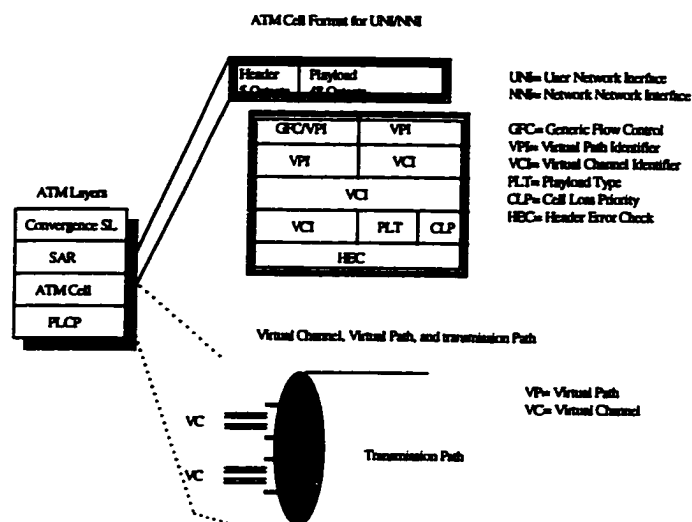


**Figure 5.11: Virtual Path Connections & Virtual Channel Connections**

As shown in figure 5.11 VCCs are subsets of VPCs. This allowing the ATM network to use VPCs as virtual trunks between ATM switches. The virtual path connection virtual trunks may be routed amongst facilities by using ATM virtual path connection cross-connect.

At the lowest level, an ATM network uses fixed-size frames called CELLS to carry data. ATM requires all cells to be the same size because doing so makes it possible to build faster switching hardware, with the capability of carrying multiple types of traffic. These includes traffic types such as high-speed Constant Bit Rate (CBR) that requires very little variation in network delay.

ATM standard specify a 53 octet cell. The first five octet are a cell header used for ATM functions such as virtual connection identification. The remaining 48 octets are cell payload available for information transfer and other higher layer functions.



**Figure 5.12: ATM Cell format Structure and its transport on the Virtual channel, Virtual Path and transmission Path.**

The cell header format is slightly different at the ATM UNI and NNI interfaces. At the UNI, there is provision for some flow control functionality. This might allow several devices to equitably share a single ATM access line to the network.

For example, an ATM access line to a residence may need an efficient simple way to handle several phones, televisions, and computer workstations without the need for switching. The Generic Flow Control (GFC) field of the ATM header in UNI is available to support this type of functionality.

The Network to Network Interface (NNI) by contrast is assumed to be point-to-point. For this reason, the GFC field in the UNI header has been eliminated from NNI header. This allows for a larger number of virtual paths to be used as virtual trunks between switches.

The remaining fields in the header include a Payload Type (PT) of 3 bits, one of Cell Loss Priority (CLP), and an 8 bit Header Error Check (HEC).

ATM is currently defined to run on T3, as well as any of the Synchronous Digital Hierarchy (SDH) or SONET rates. At these high rates, cell switching allows the low delay variation sought for circuit emulation, while still allowing the statistical multiplexing gains of packet-oriented traffic. At lower rates, cell switching has no technical advantages over frame switching.

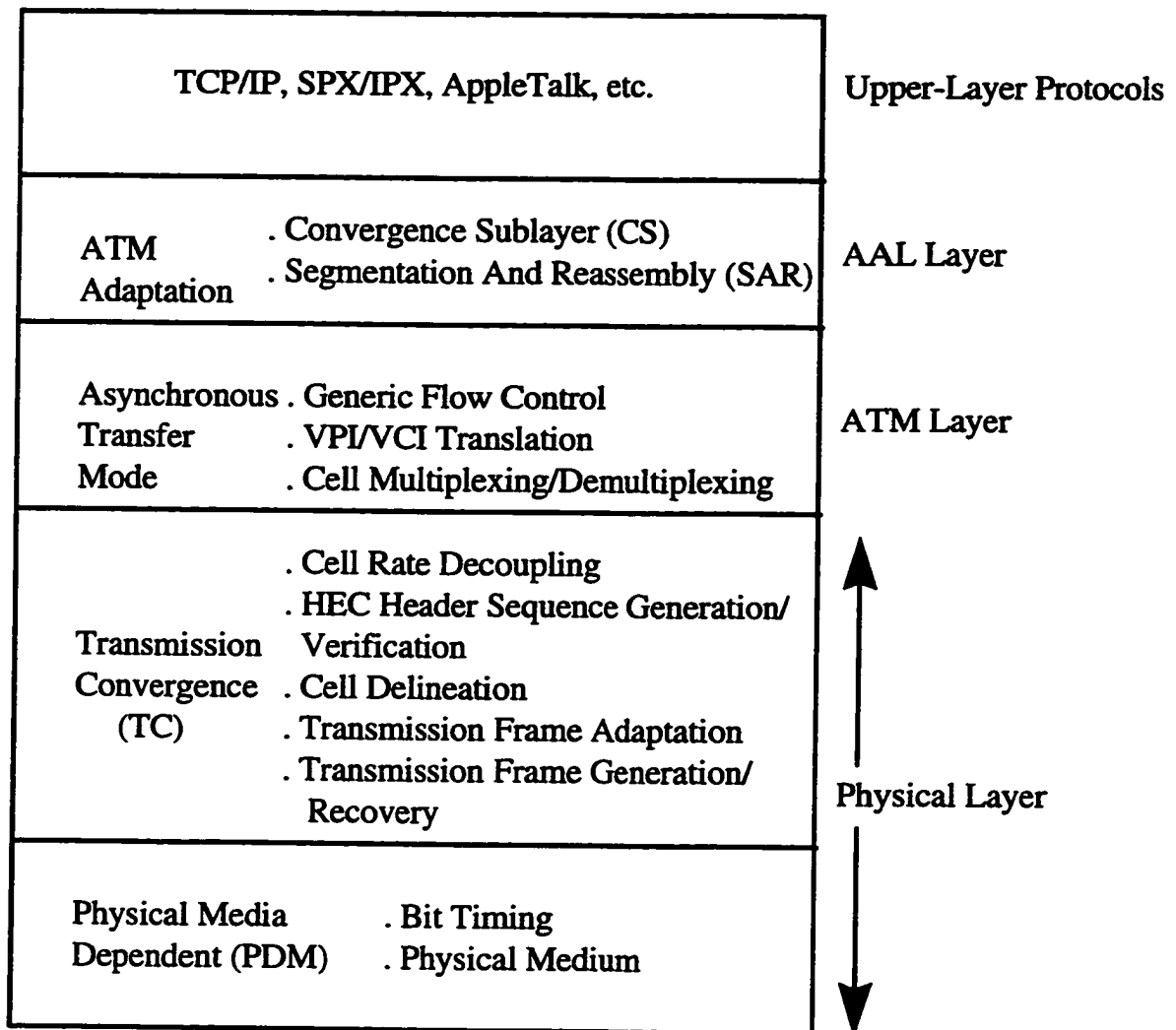
During the phase of this research, a special attention has been done to the payload type field, the three bits field. The PLT is used to distinguish between user traffic and various forms of operations, administration and management (OA&M) traffic.

To carry traffic across an ATM network, the traffic has to be divided into a sequence of cells. Conceptually, this is no different from the previous practices of

dividing traffic into packets (X.25) or frames (in frame relay) or even datagrams (in Internet Protocol (IP) or Connectionless Network Protocol (CLNP)). For circuit mode traffic, the analog is a little looser. In those existing time division multiplexed systems, the time slots into which the traffic is divided are not self-identified as belonging to a particular conversation, which is necessary in ATM to bring the benefits of statistical rather than time division multiplexing.

The *physical layer* for ATM consists of two sublayers: The *Physical Media dependent* (PDM) layer and the *Transmission Convergence Layer* (TC). The PDM deals with physical functions such as bit transmission, bit alignment, encoding, and electrical conversion. ATM can support diverse media types such as twisted-pair wiring, fiber optic, and coaxial. If fiber optic cable are used, the PDM layer performs the electrical conversion between light and electrical signals. The TC layer provides a conversion between ATM cells used by upper layers and bits used by PDM layer. It also performs transmission frame generation and recovery, transmission frame adaptation, cell delineation, header error control sequence and verification, and cell rate decoupling.

The *transmission frame adaptation* is used for formatting the data in the cells to be transmission frame structure to be used for transmission. In the United States, SONET is used as one of the transmission frame options for ATM. This means that data frame must be extracted and placed in the DS3 envelope format used by SONET. When data is received in a DS3 payload format, it must be extracted and placed in the ATM cell format.



**Figure 5.13: ATM communications protocol stack**

The *transmission frame generation and recovery* is responsible for the physical generation and reception of the transmission frame (such as SONET).

The *cell delineation* prepares the flow of cells so that the cell boundaries can be recovered by the receiver. The ATM cells are scrambled and placed in the cell stream to protect against malicious users. Either a polynomial-based,

self-synchronizing scrambled or a modulo addition of a pseudo-random sequence can be used. At the receiver, the cell boundaries must be identified by reversing the scrambling process and the cells recovered.

The *header error control sequence generation and verification* is used to compute the header checksum at the transmitter. The header checksum also is computed at the receiver to detect errors in the ATM cell header. The checksum can be used to perform single-bit error correction or multiple-bit error detection.

The *cell rate decoupling* is used to insert or suppress idle cells so that the rate of valid cells matches the payload capacity of the physical transmission system used.

The *ATM layer* provides cell multiplexing and demultiplexing, VPI and VCI translation, and cell header generation and extraction.

The *cell multiplexing and demultiplexing* function of the ATM layer performs cell multiplexing functions in the transmit direction and cell demultiplexing function in the receive direction. In the transmit direction, cells from individual virtual paths and virtual channels are placed in a cell flow. In the receive direction, the cells are sent to the appropriate VP or VC.

The *VPI and VCI translation* occur within ATM switches. The values of the VPI and VCI fields in the incoming ATM cells are mapped to a new VPI and VCI value based upon the lookup tables within the ATM switches.

The cell header generation and extraction are performed at the end-points of the ATM connection, such as desktops and servers that need to send the ATM cells for processing to a higher layer. At the transmitting ATM station, the cell header generation function receives the data from an upper layer and generates an ATM cell header except for the Header Error Control (HEC) field, which is computed by ATM physical layer. At the receiving station, the ATM cell header information is stripped, and the resulting 48-byte cell sent to the upper layer.

The *ATM Adaptation Layer* (AAL) maps the upper layer Protocol Data Units (PDUs) from protocol such as TCP/IP, SPX/IPX, and so on, to the ATM cells required by the ATM layer. The AAL functions can be divided into two sublayers. These sublayers are the *Convergence Sublayer* (CS) and the *Segmentation and Reassembly* (SAR) sublayer. The *Convergence Layer* (CL) is specific to each type of upper-layer protocol that is supported by and interfaces with the characteristics of the upper-layer PDUs into 48-byte cells. On the receiving side, it assembles the 48-byte cells received from the ATM layer to upper-layer PDUs.

### **5.4.3.**

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## **Synchronous Transfer Mode versus Asynchronous transfer Mode**

Packet-switching technologies such as ATM exploit communication channels more efficiently than *Synchronous Transfer mode* (STM) technologies. A classic example of STM technology is the T1 carrier. In T1, a period of time called the

*frame* is divided into a separate voice channel, making it possible to *time multiplex* 24 voice channels in a single frame. The time slots are combined with a frame bit from the T1 signal. Each time slot is synchronized with respect to the frame bit, hence the name Synchronous Transfer Mode. In STM, the data for a particular voice channel is identified by its position in the frame. This type of technique uses *position multiplexing*. In ATM, the data is identified by a label (address information) in the five-bytes header of the ATM cell. For this reason, ATM is sometimes referred to a *label multiplexing* techniques. One of the fundamental differences between the two is the method used to identify a data slot: STM uses position multiplexing, and ATM uses label multiplexing.

In STM, because a time slot is assigned per voice channel, each channel is guaranteed a certain bandwidth goes unutilized. Even if the voice signals obtain additional unutilized time slots, the voice communications do not improve significantly. If however, unutilized time slots become available for data traffic, data transfer rates improve dramatically. In ATM, access can be given to data traffic or voice traffic on demand. In general, data traffic tends to occur at random intervals and for random lengths of time. ATM is well-suited for transporting this type of traffic because additional data cells can be generated on demand. Voice, video, and other real-time traffic is time sensitive and needs immediate access to data cells, never has to wait for more than one 53-byte cell. For one 53-byte cell, this works out to be the following :

53 bytes/155 Mps=53 Bytes x 8bits/Byte/155000000=2.7 Micro seconds

By keeping the cells short, real-time traffic is guaranteed ready access.

## 5.5.

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### DIGITAL SUBSCRIBER LINE

ADSL is dedicated to exploiting the presence of the twisted wire pair access from the telephone central office to almost every household, every business and to every non-mobile location of any significance. Given the investment of the telephone companies and public switched telephone network owners, the ADSL offers *two* attractive propositions. *First*, it promises to deliver the maximum data rate to the customers from the central office and *second*, it provides ample data rate from the customer to the central office to control and facilitate what and how much information is being delivered. Having been initiated with this motive, the ADSL has taken a stride into the premises distribution systems where the data rates can approach the OC-3 rates at 155 Mbps.

There is a rich history to the ADSL. The most early efforts were in the 1970s with the investigation of the subscriber loop plants in view of deploying them for data traffic rather than for analog voice traffic. Digitized voice would serve the plain old telephone type of service and additional digital capacity would serve any new applications that may emerge. Early goals for the digital capacity of the twisted wire pairs were modest, if not trivial, from the 1990s perspective. A rate of 32 kbps was experimentally and computationally proved in a few months. A

rate of 64 kbps was proved in a shorter while during 1974. A rate of 80 kbps in duplex mode was proved in 1975. A rate of 192 kbps was proved in 1976. Complete bidirectionality (i.e. duplex mode of operation) was proved and demonstrated in 1979. The results were published in 1981 and 1982.

The apparent set back came due to misconception that TCM would be an alternative to the poorly developed echo cancellation technology of the late 1970s. Such TCM systems were indeed built and marketed as CSDC systems in the United States during the early 1980s. The business communities subscribed to this 56 kbps bi-directional line service through the mid eighties. A host of other new services did emerge, but their life span was short due to impending BRISDN services at 144 kbps promised for the late 1980s.

ADSL is a rapidly emerging technology that uses existing copper-telephone wiring to deliver a high-speed, downstream channel to subscribers and a lower-speed, upstream channel to the network while simultaneously providing POTS over a single, copper loop. Researchers believe that demanding bandwidth and speed applications (e.g., Interactive multimedia, telecommuting, video conferencing, video-on-demand, distance learning and remote training, high speed internet access, remote LAN access) can be served or enhanced with ASDL.

There are several versions of this technology (DSL), and to keep up with it, the generic XDSL has been coined to refer to the category.

**DSL- Digital Subscriber Line** which operates at a maximum of 144 Kbs for ISDN subscriber lines. ISDN is used for voice and data communications.

**ADSL- Asymmetric Digital Subscriber Line** which operates at 32Kbps to 8.192 Mbps downstream (to the customer) and 16 to 640 Kbps Upstream (to the network) over existing twisted-pair copper wire. ADSL is envisioned for use for Internet access, Video On Demand (VOD), Simplex Video, Remote LAN Access, and Interactive Multimedia.

**HDSL- High-bit rate Digital Subscriber Line** delivers data symmetrically at rates up to 1.544 Mbps Full-Duplex for equivalent T1/E1 service, or at 2.48 Mbps (require two pairs of wire) for subscriber lines, or at 2.49 Mbps Duplex (require three pairs of wire) for feeder plant, WAN services, LAN access, or server access.

**RADSL- Rate Adaptive ADSL** is a version of ADSL where the ADSL modems test the line at start up and adapts its data rate to within 32 Kbps of the maximum throughput the line is capable of supporting.

**SDSL- Single line DSL** offers two-way channels, but over a single copper wiring pair and operates at 1.544 Mbps or 2.048 Mbps for subscriber line duplex premises access. This is the same as HDSL plus ability to support symmetric services.

**VDSL- Very High Speed Digital Subscriber** is under development for twisted-pair access services at 12.9 Mbps to 52.8 Mbps Downstream and 1.5 to 2.3 Mbps Upstream. However, the maximum reach will be reduced to 4500 to 1000 feet

and it will need fiber-optics cable. Applications are the same as ADSL plus high-definition TV.

VADSL- Very High Speed Asymmetric Digital Subscriber Line which operate at a subset of speeds of VDSL when it supports symmetric operation.

## Chapter 6

### **INTELLIGENT ATM NETWORKS**

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Today many companies are introducing ATM for routing in existing telecommunication networks supporting voice telephony and constant bit rate video services. The use ATM in such networks has proven a better service and big savings. The ATM facilities will allow B-ISDN to support greatly enhanced variety of services. This will include the possibility to integrate a wide range of existing and future services such as voice, image, text and high speed data.

The idea behind Intelligent ATM Networks is to provide intelligent services by accessing to the SCP which in the future will be a network connecting various file servers. The aim of this networks is not only to provide intelligent services for voice based calls, but also to provide intelligence to different applications sources with constant or variable bit rates. In this chapter we will discuss:

- The architecture of switching systems and the possibility of designing a switching system based on ATM fabric.

- The compatibility of Common Channels Signaling N 7 with Intelligent ATM Networks.
- The possibility of modifying the ATM standard giving the capability of carrying information about the type of call made to the ATM cell.
- Providing a functional modules for the Intelligent ATM services.

The study has been elaborated for both the typical infrastructure, where all IN calls are detected at the SSP level by checking up the dialed digits, and on the new infrastructure where the signaling is to be based on ATM address. In each case, new functions (functional modules) have been defined to be implemented on ATM environment.

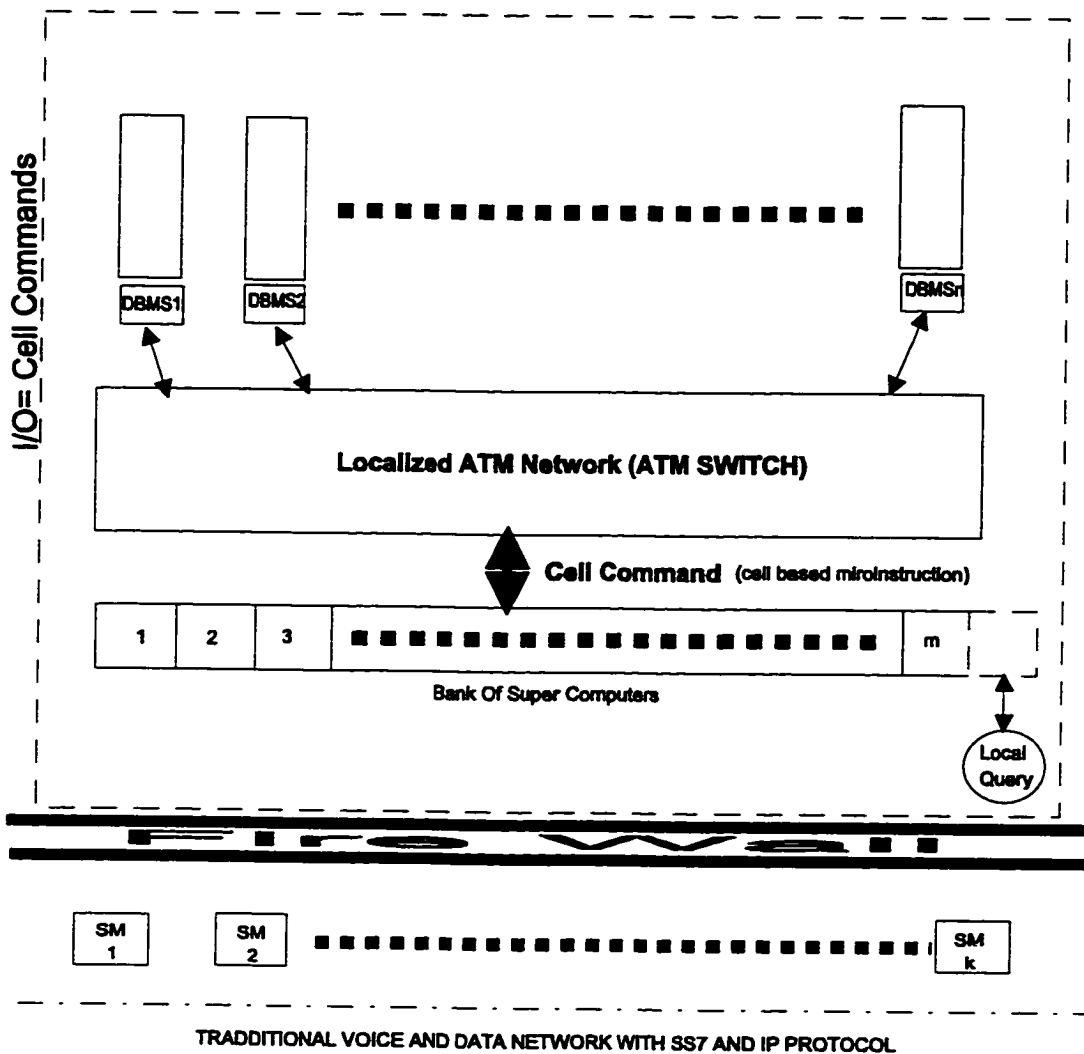
## 6.1.

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### **FUTURE NETWORKS**

Unlike the actual intelligent networking where the SSP is triggered for IN services related to telephony, in future intelligent networking the SSP will be triggered for IN services related to any type of communication schemes. The future Control Point will be present by a large number of File Servers that have the ability to communicate with each other. The access to this File Servers will be an answer by a trigger from a local or tandem switching point, and through the signaling network which will be based on X.25 protocol at the beginning. In the far future, it is expected that the signaling network will be based on the ATM protocol due to the possible congestion. The signaling network, then, will be based on a cell command (as in a computer instruction) and a fiber for

transmission (as in the case of a bus for input and output in the case of a computer system).



**Figure 6.1: Future Networks**

Figure 6.1 shows a possible architecture of future networks which will be separated from the traditional voice and data network by a fire wall. This networks need to be highly secured. The use of ATM protocol on a fiber transmission will ensure the transmission security.

In this networks, a bank of super computers will have the ability to process any kind of query, knowledge or only an intelligent service request, create a cell based command, and then access to a large number of file servers through a localized ATM network for more details about the query. To answer the query, the access to remote systems may be a possibility.

In case of queries related to knowledge, a new protocol is to be defined. In the case of a query related to intelligent services, the already existent SS7 can be used, or a new Signaling System based on ATM for transport of the signals can be defined. In any of this cases, the switching system should have the ability to trigger the query, decide if the access to the file servers is a must, and connect to other systems. The following deals with answering queries related to IN services in an IN environment.

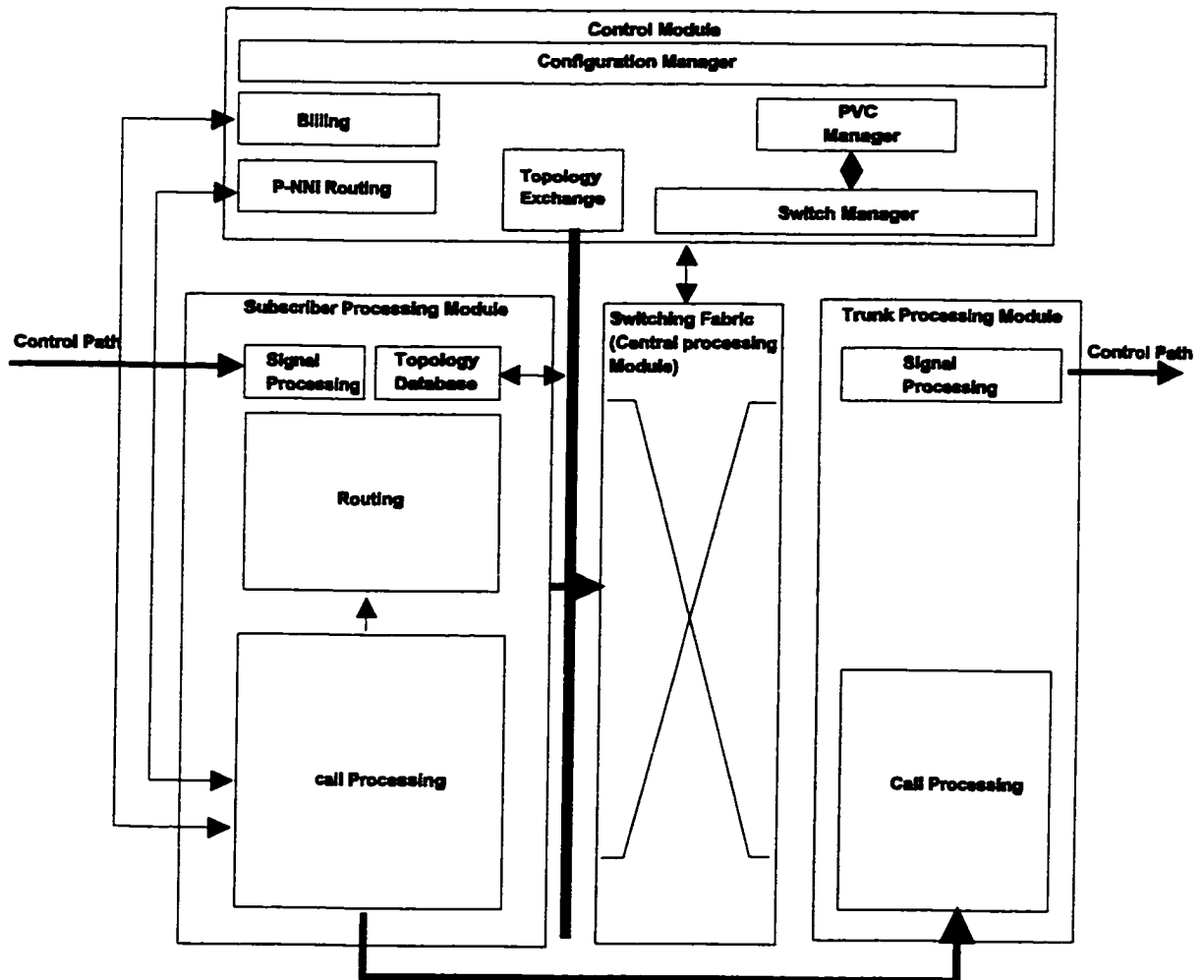
## **6.2.**

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### **ATM SWITCHING SYSTEMS**

The ATM network, is a connection-oriented and a cell directed communication network, a connection from the source to the destination has to be established before the user data information can be transmitted via the connection. Call processing in ATM switching environment has been attracting researchers interest, and still until today the ATM switch architecture has not been published.

The switch represents an enormous capital investment for service providers. To support IN needs, ATM switches require special control and interface functionality.



**Figure 6.2: Architecture of ATM switching system**

Signaling message processing is a major bottleneck in call processing in ATM environment. To speed up call processing we will need to reduce the time to process signaling.

While working on ATM simulations, it can be realized that the centralized call processing architecture is not scaleable. One of the more intensive processor-tasks is the parsing of signaling and needs to be as simple as possible.

The architecture on figure 6.2 shows a switching system configuration for call processing. Besides the fabric (central processing module), there is an ingress and egress interface modules (respectively subscriber processing module, and trunk processing module), and a control module each with its own processing and memory elements. The central processing module (CPM) links between the subscriber processing module and the trunk processing module. The CSM is a self routing switching network that switches all information in cells of uniform format (header and information).

The ingress interface module will accept UNI signaling, and the egress interface module will accept the NNI signaling. Signal processing takes place in each interface module. Each module should have its own dedicated processor, capable of running a multitask kernel. As its name implies, each interface module in this architecture will terminate signaling stacks and participate in the call control procedure. The control module doesn't operate the signaling stacks. The processing of signaling messages is the most time-consuming task in call processing. Making this task as a part of the interface modules (subscriber processing module and trunk processing module) enables the architecture to achieve higher performance and scalability because the concurrence of signaling processing usually improves the call processing latency. The *Call processing* is

used to coordinate connection management and implementation of local traffic management policy across the switching system. *Connection management* establishes and maintains call records, determines service categories, and coordinates connection states for signaling stacks on subscriber/trunk sides of connections. *Traffic management* set of actions taken by a switching system during the call establishment phase to determine whether the connection request should be accepted or rejected. *Routing* is to find path across the network between two or more major switching systems. The selected path is required to meet end-to-end service requirements of the connection. To achieve this, the routing system has three major functions (setup, connect, release). In the *topology exchange*, each switching system advertises to other switching systems information about NNI links attached to it, typically based on a link-state routing protocol. A switching system uses link-state information to build a topology database of advertised capabilities, attainability, and desirability of other switching systems to carry connections with different characteristics. *Topology database* provides a logical view of the network on which the route can be determined.

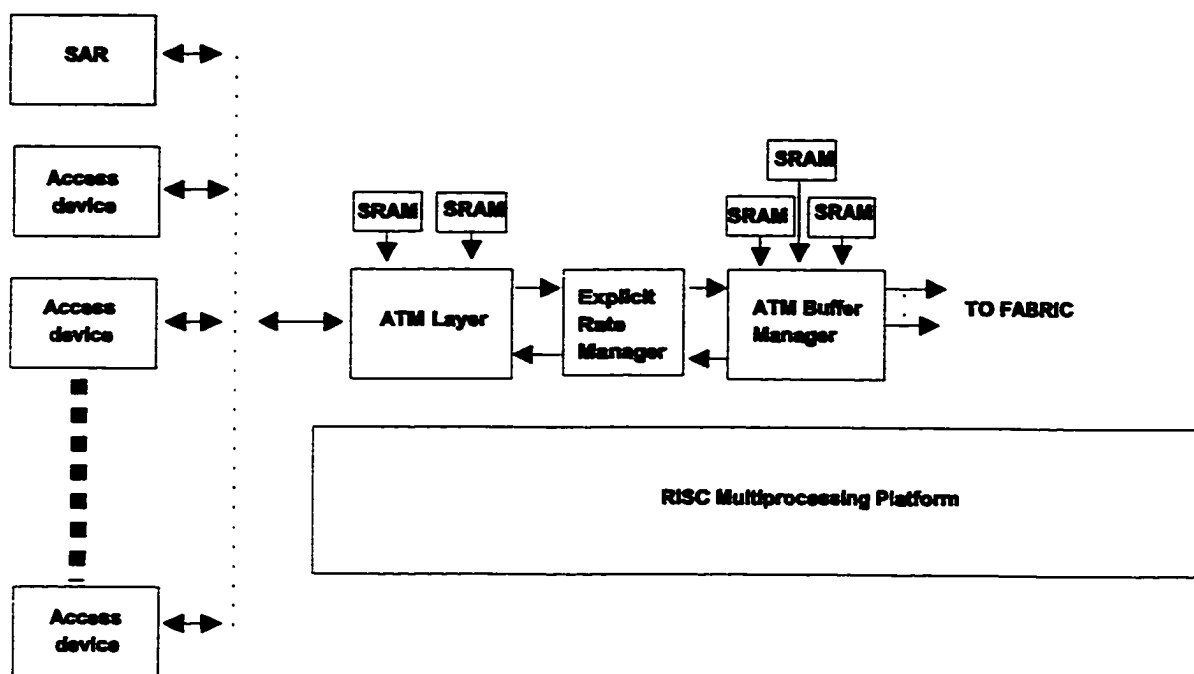
In this architecture, the ingress signaling agent is activated by the reception of a signaling message (requesting connection) from the incoming link (SETUP). An internal setup message will be then created (I\_SETUP). The I\_SETUP is passed to the call control sub-module which will initialize and create a call records for the new request. Then, the route determination sub-module, answering to the call control request, will find the appropriate path based on the topology information

residing in the topology database. If the call is to be accepted then the call control will arrange for the resource information and the programming for the switching hardware, notify the ingress signaling, and then forward the request to the egress signaling which after sending a message to the next hop of the selected path will clear the timer and get ready to get another connect request.

Call processing for an ATM intelligent service request has not been taken in consideration at this step.

Based on packet-switching technology, the signaling system is effectively able to provide reliable services and simplify the introduction of new features and services. The inter-module cell structure is most important in determining the architecture for this inter-module communication. To implement inter-module communication via central switching module, the cell header components must include a connection identifier so that the interface modules can distinguish between end-to-end information and inter-module data, a sending module number, and a receiving module number.

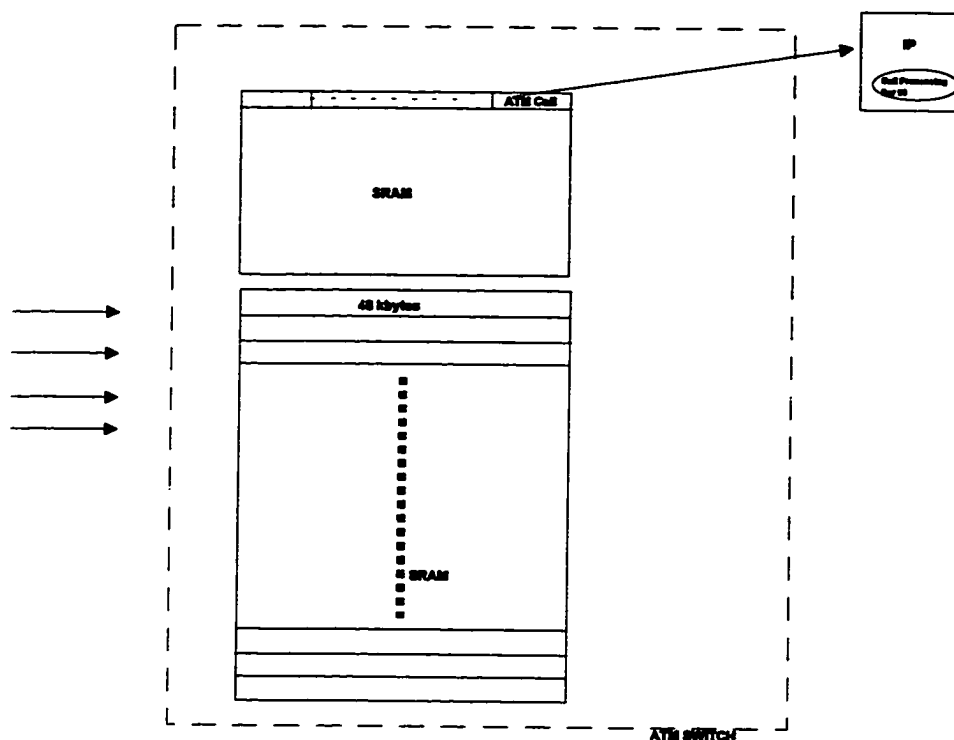
Software to take care of inter-module messages is to be implemented. This software will be able to distinguish between the following: Basic service processing, new service processing, and the SS7 processing. In this section, maintenance information (traffic data collection for example) and the fault detection and recovery will be addressed.



**Figure 6.3: An ATLANTA based Customer Interface Module**

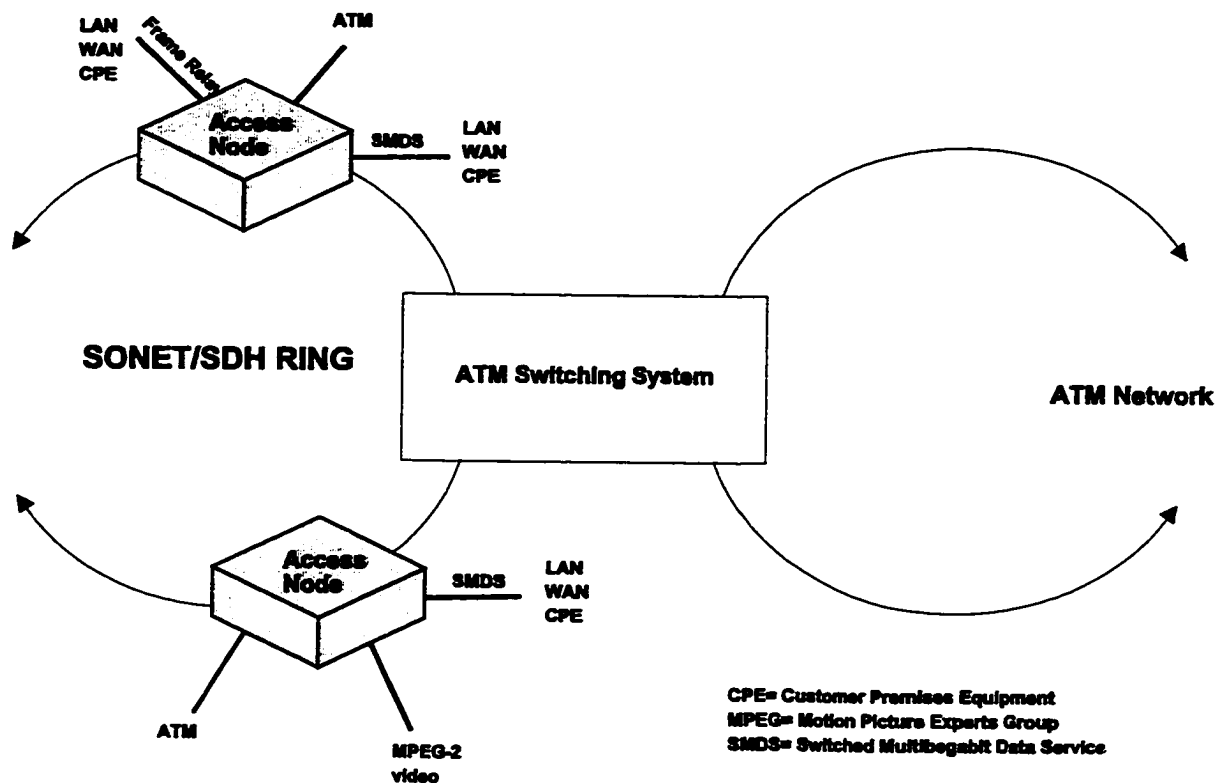
Figure 6.3 shows the customer interface module in an ATLANTA based ATM switch. The current switch does not have the SAR module for Segmentation and Reassembly which has been added to the current configuration to minimize the processing time. All modules should function in real time for better performance. In the trunk interface module, the explicit rate manager is absent.

All cells requesting connection will be placed in the stack random access memories of the ATM buffer manager before routing. In case of an intelligent service request, the call control processing element (with its own data base) will change the structure of the cell header by making the reserved bit a 0 instead of 1. The ATM fabric, will recognize the cells requiring special attention.



**Figure 6.4: ATM Stack Random Access Memory at the switch level. The Memory can be divided into two separate parts; one dealing with control and the other with Information itself.**

An access node is to be added to convert existing protocols into ATM cell based protocol and vice versa. The ultimate, is to have an all ATM network, but during migration, a software for protocol conversion is needed, and should be installed at the access node. Such packages are currently available. Then the information can flow from the access node to the service node which is suppose to support both service management module and switched service module.



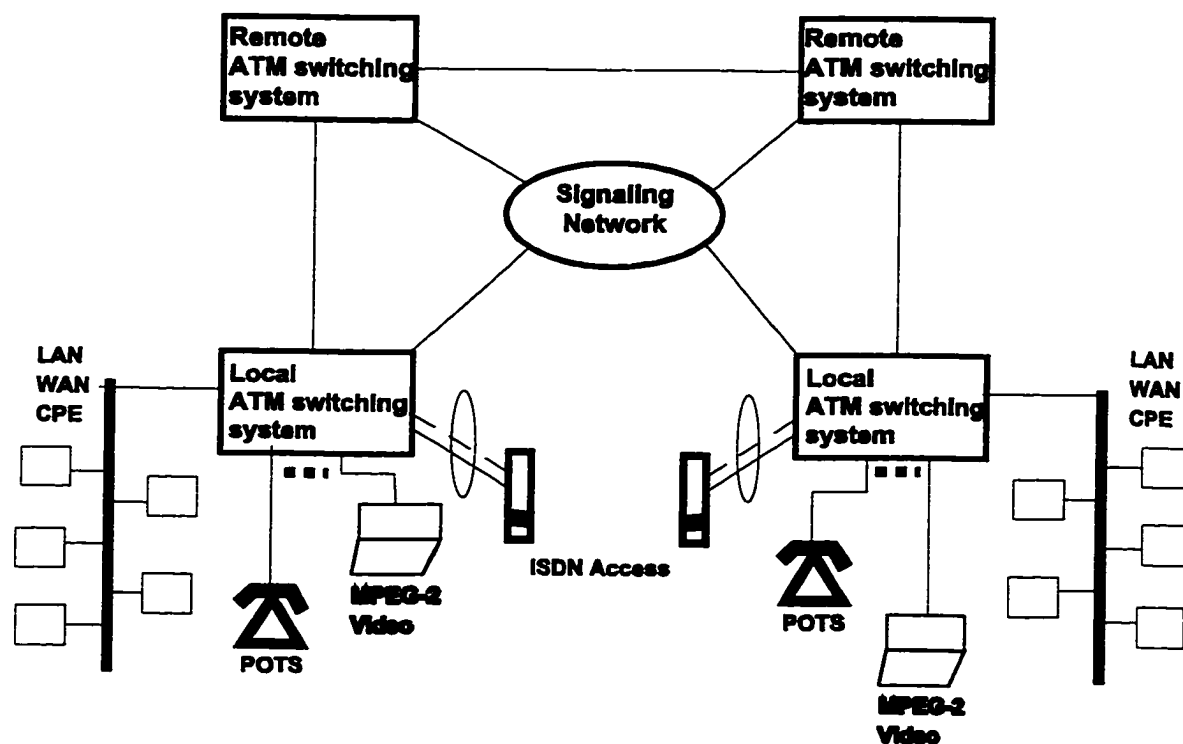
**Figure 6.4: A SONET ring connecting Access nodes the ATM switching System. (AT&T Technical Journal)**

The access node is synchronous with digital hierarchy (SONET/SDH) rings. It provides low cost, point to point ATM connectivity and protocol conversion of many types to and from ATM. Within ATM-only network (aim of this research), the access node multiplexes between the various SONET/SDH rates, which carry ATM at the transport layer.

Call and signal processing for various types of calls, will be provided through a highly reliable processor. These calls can be single, switched-virtual-circuit to multimedia calls comprised from of voice, data, and video segments.

The access node will serve to convert the various protocols of LANs and PBX voice systems, which are found through today's network, to the ATM protocol. When the PBX is based on ATM, LAN, WAN, and MAN are based on the ATM switching, then all broadband services will be provided.

The ATM switching system will reside at either the local end-office or a tandem-based switching system.



**Figure 6.10: Local and remote ATM Switching Systems sharing a signaling network.**

### 6.3.

## SIGNALING IN INTELLIGENT ATM NETWORKS

Currently, the SS7 is used for signaling in intelligent networks (mainly based on voice switching). Future networks will be based on ATM cell switching, and

providing intelligence services may become a necessity. Will the SS7 be able to keep up with this new fabric, or a new protocol is to be defined?.

The SS7 can still be the Signaling Protocol for an ATM network. However, some changes in the signaling protocol and architecture appears necessary. A new SS7 Broadband ISDN Signaling User Part (BISUP) is required to support Broadband Services, and the specification for it is already available (ITU Q.2010 draft Broadband ISDN Overview Signaling Capability, Bellcore GR-1417-CORE Broadband Switching System SS7 requirements Using BISUP, Bellcore GR-1431-CORE CCS network Interface Supporting Broadband Services).

Some Changes in the transport network architecture which is used to carry the BISUP signaling protocol are becoming a necessity. The actual Quasi-Associated Mode (end nodes communicate through a packet switch called STP) as opposed to Associated Mode (where end nodes communicate with direct signaling links). With ATM, this architecture is to change to the associated mode. For telephony, the Quasi associated will still be used, but for Broadband Services the associated will become a necessity.

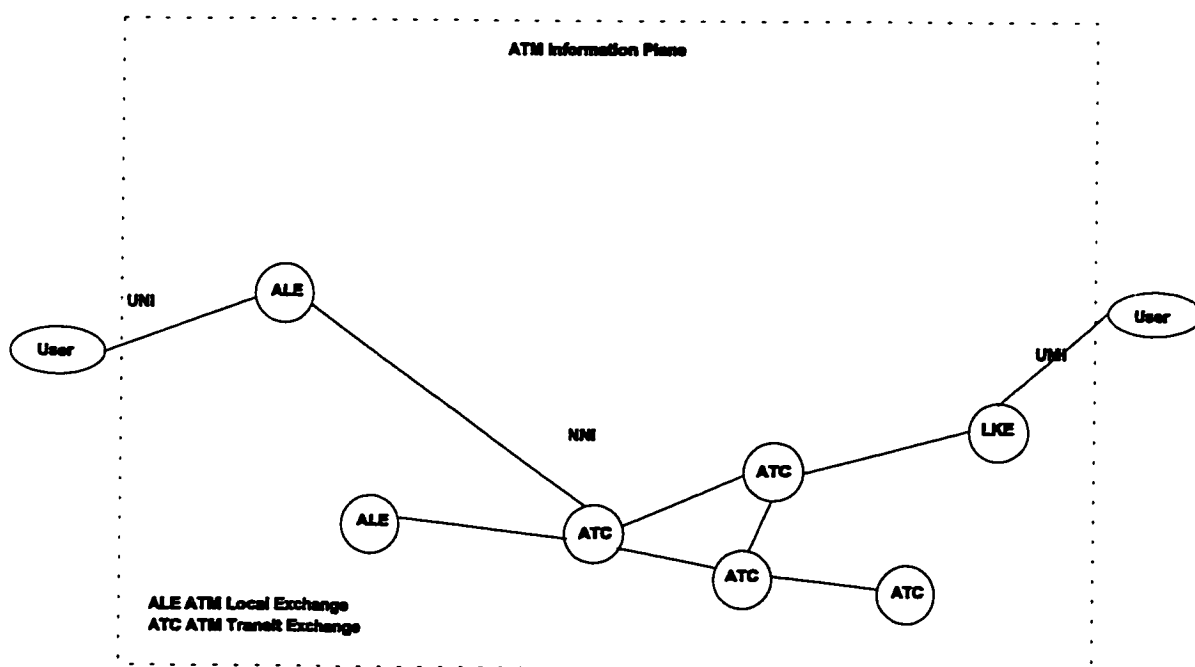
SS7 protocol can still be the signaling protocol in ATM networks. However, some new modules are needed to detect an IATMN service trigger, and establish to connection to complete the service.

### **6.3.1.**

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## **X.25 based Signaling System**

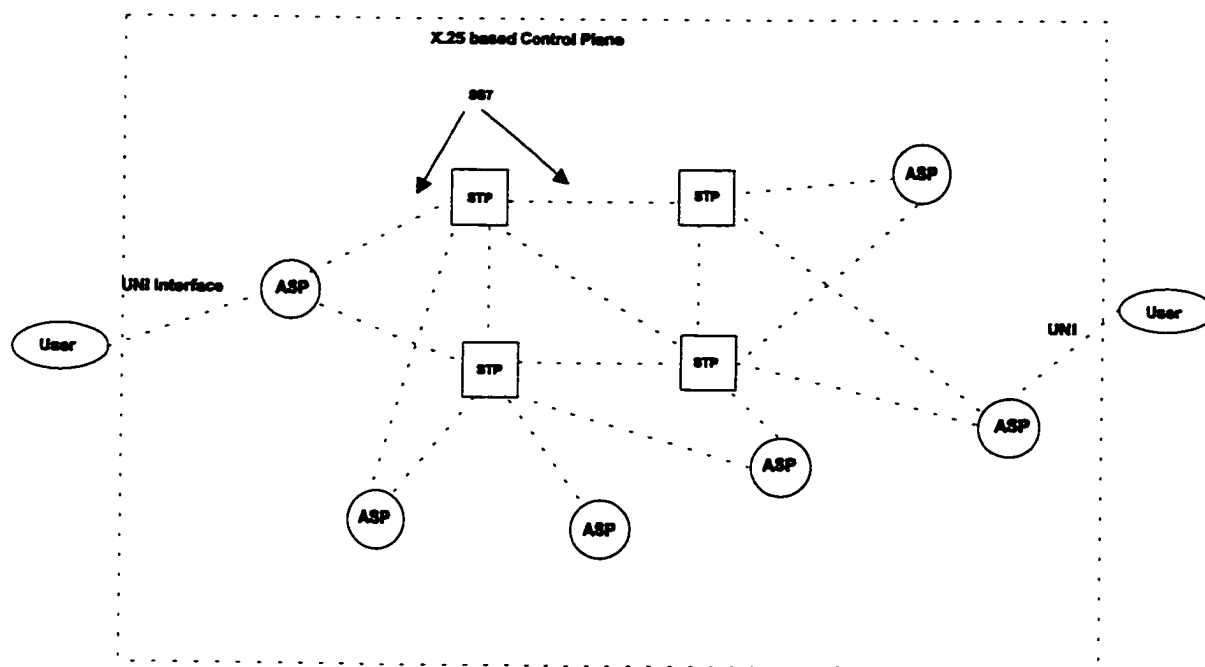
Control messages are routed through the network to perform call management (setup, maintenance, termination) and network management functions. These messages are short blocks or packets that must be routed through the network. Thus, the network to be controlled is ATM based, and the control signaling is also implemented using packet switched technology. SS7 defines the functions that are performed in the packet-switched network but does not dictate any hardware implementation.



**Figure 6.11: The ATM Information Plane**

Signaling Network will remain the same, and the transport network will be based on the ATM technology. Figure 6.11 shows an ATM information plane. User information will flow to the ATM Local Exchange (ALE) using the User network Information Protocol, and between different ALEs and ATCs (ATM transit exchange using Network to Network Protocol.

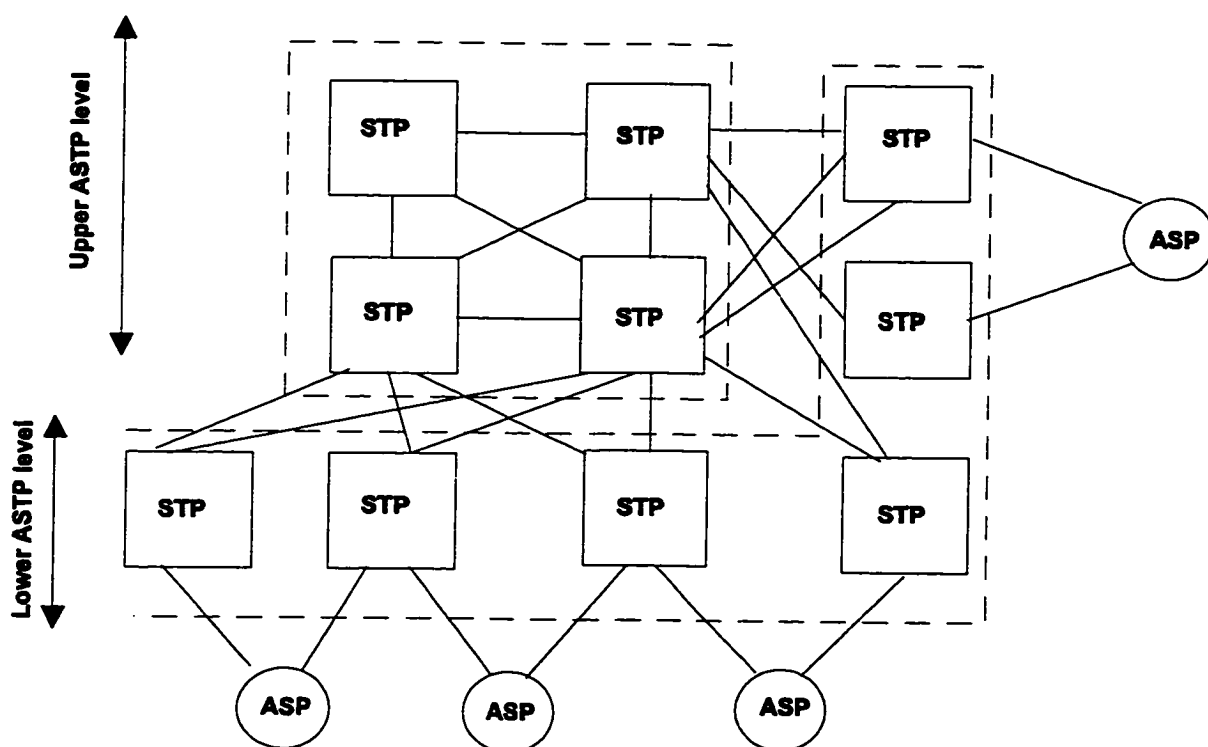
Figure 6.12 shows the Signaling Transfer network. The ASP (ATM Signaling Point) is any point in the ATM network that is capable of handling SS7 Control messages and routing them; that is received on one signaling link is transferred to another link. An ASP could be pure routing node or could also include the functions of an endpoint. A signaling link is used to connect endpoints. The non-associated mode signaling can still be used in this case.



**Figure 6.12: The Control (Signaling) Transfer Network.**

We can still consider two plans of operation. The control plane is responsible for establishing and managing connections. These connections are requested by the user while asking for intelligent services from an ATM local exchange. The ATM local exchange acts as signaling point, since it must convert between the dialogue with the user and the requested actions (to access to the signaling network).

Internal to the network, Signaling System is used to establish and maintain connection; this process may involve one or more ATM signaling points and signaling transfer points. Once a connection is set-up, information is transferred from one user to another using, end-to-end, in the ATM information plane. A circuit is setup (ATM is a connection oriented network) from the ATM local exchange of one user of that of the other (routed through one or more other cell switching nodes referred to as ATM transit centers).



**Figure 6.13: Hierarchical signaling Network with two STP level**

A complex network will have both ATM signaling points and transfer points, in a hierarchical structure (see figure 6.6 ) in where the ASPs constitute the lower level and the STPs the higher layer. The STP capacities, network performance,

and availability and reliability should be considered while designing this networks.

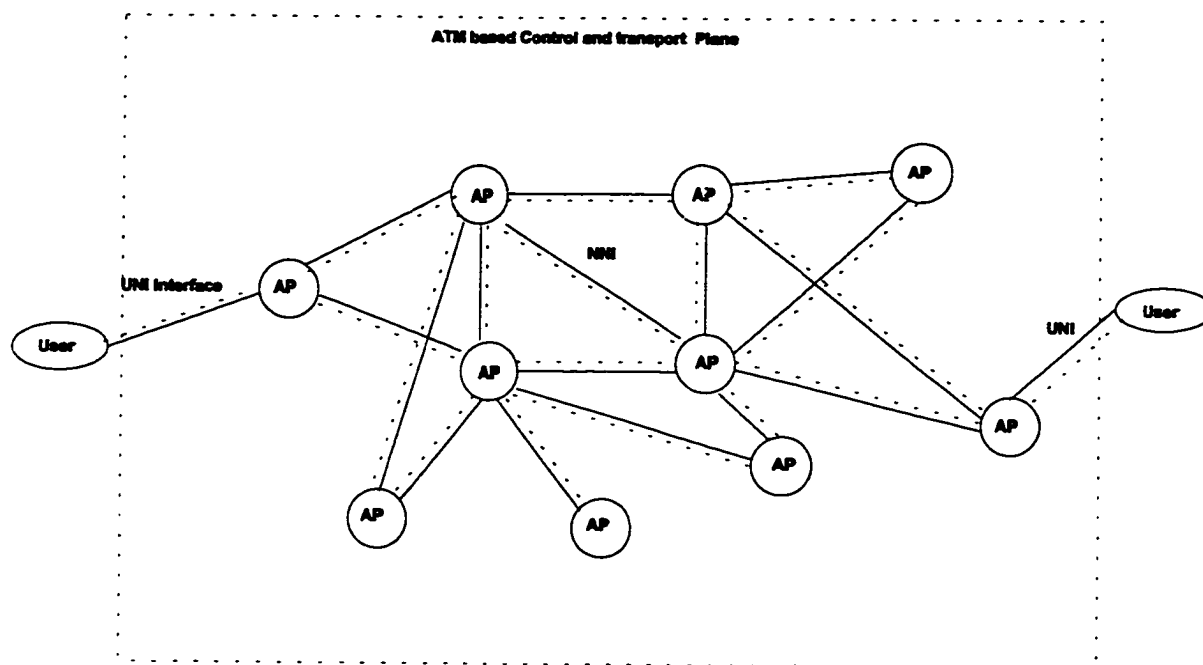
In a hierarchical network with two STP levels, each ASP that is not an STP is connected to at least two STPs of the lower level, each STP in the lower level is connected to at least two STPs of the upper level, and the STPs in the upper level are fully meshed. This type is typically designed such that the lower levels are dedicated to handle traffic in a particular geographic region of the network, and the higher level handles interregional traffic.

### **6.3.2. Cell based Signaling System**

When necessary, Signaling Network will based on ATM cells commands, and define two major entities: ATM Point (AP) which will be considered as both a service signaling point and a signal transfer point, and interconnection Links.

The AP is any point in the ATM network that is capable of handling ATM signaling commands and routing them; that is received on one signaling link is transferred to another link. An AP could be pure routing node or could also include the functions of an endpoint. A signaling link is used to connect endpoints. The associated mode signaling is the best architecture of ATM node.

Figure 6.5 highlights the distinction between the cell-switching command functions and the cell switching information transfer functions.



**Figure 6.14: ATM Signaling and Information Transfer Networks.**

We can consider that there are two plans of operation. The control plane is responsible for establishing and managing connections. These connections are requested by the user while asking for intelligent services from an ATM local exchange. The ATM local exchange acts as signaling point, since it must convert between the dialogue with the user and the requested actions (ATM signaling System).

Internal to the network, ATM based Signaling System is used to establish and maintain connection; this process will involve one or more ATM signaling points which will also act as ATM signaling transfer points. Once a connection is set-up, information is transferred from one user to another using, end-to-end, in the ATM information plane. A circuit is setup (ATM is a connection oriented network) from the ATM local exchange of one user of that of the other (routed

through one or more other cell switching nodes referred to as ATM transit centers).

A complex network may have both ATM signaling points and ATM transfer points, in a hierarchical structure (as seen in the previous case) in which the ASPs (ATM Signaling Points) constitute the lower level and the ASTPs (ATM based Signaling Transfer point) the higher layer. The ASTP capacities, network performance, and availability and reliability should be considered while designing this networks.

### **6.3.3.**

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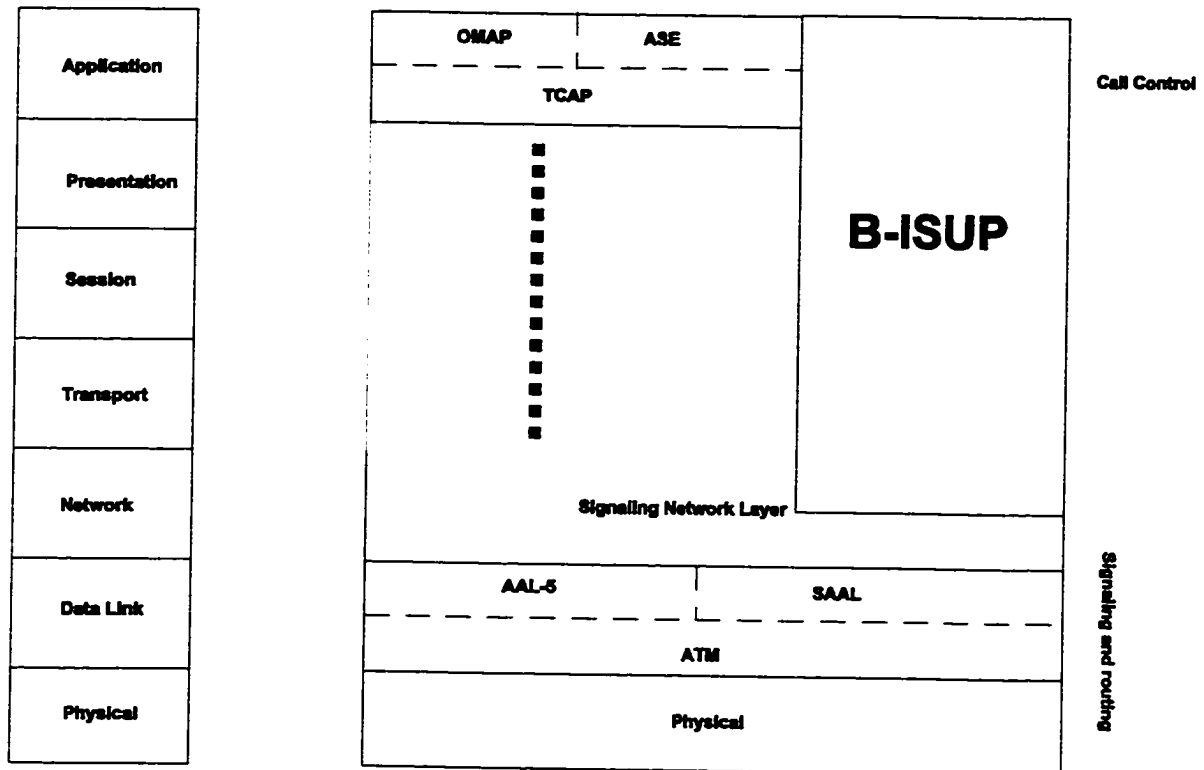
## **Protocol Architecture**

In the case of using a non-associated mode for the common channel signaling, the SS7 will still be used based on X.25 packet technology. When the an associated mode becomes necessary, the SS7 will be based on the ATM cell technology.

As with the Open System Interconnection (OSI) model, the ATM signaling System Standard is a layered architecture, but the lower layers are based on the ATM protocol for connection and routing.

The ATM signaling system architecture consists of 4 levels. The lowest three levels of the ATM-SS architecture, provide a reliable service for routing signaling messages through the ATM signaling network. The Physical layer of the ATM signaling system (as in OSI) is concerned with the physical and electrical

characteristics of the signaling links for message transfer. These include signaling (buses) between APs (or between ASTPs, and between a ASTP and an ASP in a more complex network), and control buses between APs. The second layer is subdivided into sublayers: ATM, AAL-5, and S-AAL.



Based on a network to network interface

**Figure 6.15: Signaling protocol based on ATM.**

The signaling network layer provides for routing data across the multiple SPs (or even ASTPs in more complex networks) from control source to control destination. The remainder of the protocol deals with application. The B-ISDN User Part (B-ISUP) provides for the control signaling needed in an B-ISDN to deal with ISDN subscriber request for knowledge and related functions. The

TCAP provides the mechanism for transaction-oriented applications and functions. The O&MAP specifies the network management functions and messages related to operations and maintenance. ASE will be used to support new applications based on ATM domain address.

## **6.2.**

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### **INTELLIGENT ATM NETWORKS ARCHITECTURE**

INs become the core networks that operate and support a large number of peripheral networks, such as POTS and other circuit-switched networks, packet networks, basic rate, and even broadband. The IN services are only provided to the calls based on a digit dialing. Both typical intelligent services, and a newly defined intelligent services which will be based the ATM domain (As in IP domain). Intelligent services can be provided for this type of addressing where the signaling to the control point database will need to be defined.

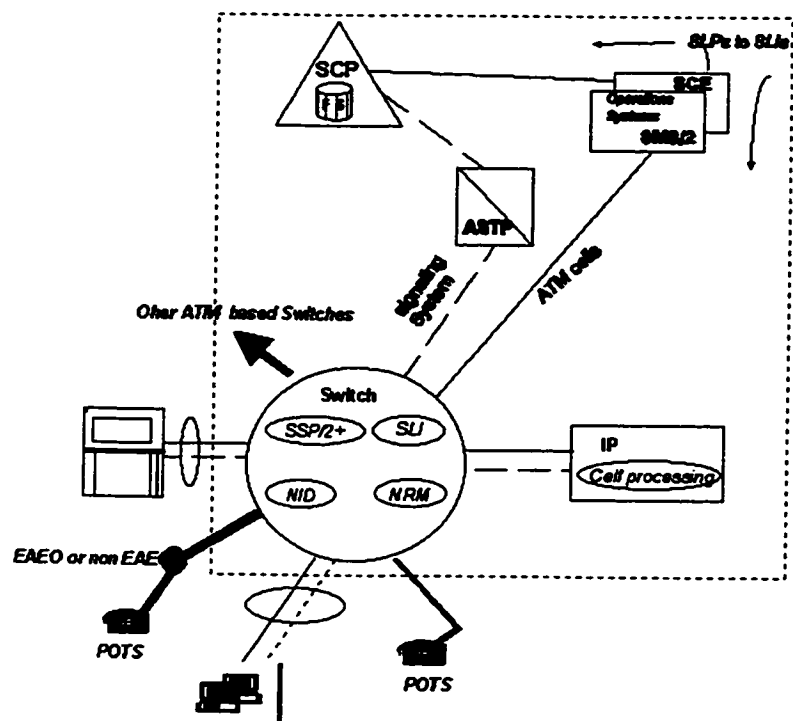
#### **6.2.1.**

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### **Intelligent ATM Network Architecture based on IN/2 Architecture**

The introduction of Intelligent ATM networks, will provide a high speed switching and transport, by providing service-independent capabilities for circuit-switched applications based. IN/2 have the capabilities to support packet-switched data and multimedia applications. The architecture of the Intelligent ATM networks, will be based on the architecture of IN/2 which

provides a platform for control in emerging technologies such as Broadband Integrated Service Digital Network (BISDN). The switching will be based on ATM switching system mentioned before, and the transport on SONET/SDH ring.

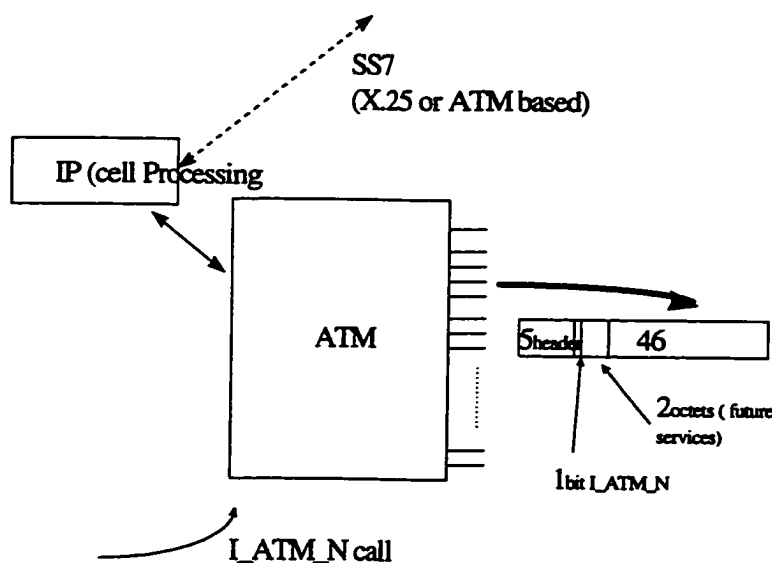


**Figure 6.16: Intelligent ATM Network architecture based IN/2 architecture. The Signaling can be either based on X.25 or ATM protocols.**

In case where the signaling is based on X.25, the actual common Channel Signaling can be kept the way it is. If, ATM signaling becomes a necessity, then the Common channel Signaling will switch to the associated mode by using ATM cells to transfer the control signals. The same nodes and links can still be

used for both signaling and information transfer. In case of more complex networks, we can still separate the signaling network from the transport network the same way as it is at the actual stage, the Signal transfer Points can be based on ATM (ASTP).

In any of the previous cases, the Intelligent Peripheral will need to be upgraded to process ATM cells in case of Intelligent Services. (More is coming up in the next paragraph).



**Figure 6.17: ATM cell definition upgraded to fit ATM intelligent services.**

One bit from the ATM cell header will recognize if the call is an intelligent call or not. The switch can transfer the cells to the intelligent peripheral module which can interact with the service logic interpreter resident in the service control point. The transactions capabilities applications part (TCAP) of IP are invoked in this

exchange. The signaling passes through (A)STP in the I\_ATM\_N. The ATM switching system is capable of communicating with IP via CCS network and may use ISDN capabilities. The software module resident within IP filters out calls not needing I\_ATM\_N functions. Once an I\_ATM\_N function is invoked, the service logic interpreter, which is resident in the SCP facilitate further call processing by responding to the type of connection to be made in completing the service.

The IP for I\_ATM\_N is a stand alone network compatible element, which can be connected to a switching system based ATM. The signaling to and from the IP is communicated via ATM switching system network to (A)STP in the I\_ATM\_N network. IPs also transfer voice and data to and from switching system, even though a switching system may have different IPs. The IP may contain resources to interact with call participants, or may be triggered by the customer inputs.

Since the destination node will need to get the information (from SCP) about the type of the call while receiving the packets, a few octets from the ATM cell payload will be borrowed for accomplish this task.

Information vendors play a critical role in supplying customer information via the network. There is a definite need for these Vendors to seek and supply information from their own data bases. The queries can be complex and also in a natural language. For this reason, The front end processors of the vendor data base systems need the recently induced natural language.

Because of the exponential growth of Intelligent services and ATM based services demand, the ATM based signaling system has been defined.

The three emerging technologies (transmission, switching, and data-base) unified into one composite science of network architecture are modifying the course of IN evolution.

## **6.2.2**

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### **Elements of Intelligent ATM Networks**

#### **SERVICE SWITCHING POINT**

The ATM service switching point is based on SSP/2 with the capability to recognizing new type of services and deciding when the access to IP is needed (for cell processing). An ATM exchange can interface with a database at a service control point (SCP) for the performance of I-ATM-N services through the IP. The SSP/2 can be upgraded to provide ATM switching function.

To provide intelligent services, protocols to exchange signaling are to be updated; In the case of calls based on circuit switching, the signaling can remain the same, but if packet switched then the protocols are to be defined. We may think about the CCS7 for digit dialing calls and a new signaling scheme for domain addressing calls (need to access to IP).

#### **SERVICE CONTROL POINT**

The SCP can still be the same as SCP/2 which consists of a transaction processor-based service logic system, databases, and a signaling network interface. The

databases are to be upgraded when creating new type of services (application processing, execution of network interface protocols to include the ATM Signaling Protocol). The SCP can interface with many networks simultaneously to process service-related, and operation and administrative messages.

In the future, when each large organization will need its own intelligent network, and the ability to access to other Intelligent corporations, then the use of large number of file servers will become a necessity. A number of file servers may be directly connected to the switching system since a separate Signaling Network will not be needed (Control signals and information can use the same path, and the signaling using ATM promise a faster service).

### **SERVICE LOGIC PROGRAMS**

(SLPs) define I-ATM-N services in terms of Functional Components (FCs), which are elemental network call processing actions that direct internal network call resources to perform specific actions (e.g., collect dialed digits).

### **SERVICE LOGIC INTERPRETER**

The SLI executes SLPs and handles requests and responses exchanged between I-ATM-N modules.

### **NETWORK INFORMATION DATABASE**

The NID contains information about customer access lines and the network. The NID is queried by other I-ATM-N modules for information; it does not interpret service logic.

## **NETWORK RESOURCE MANAGER**

The NRM provides a location function that determines the I-ATM-N module to provide network resources required to continue call processing; the NRM may respond to the SLI with information required to establish a connection to an I-ATM-N module.

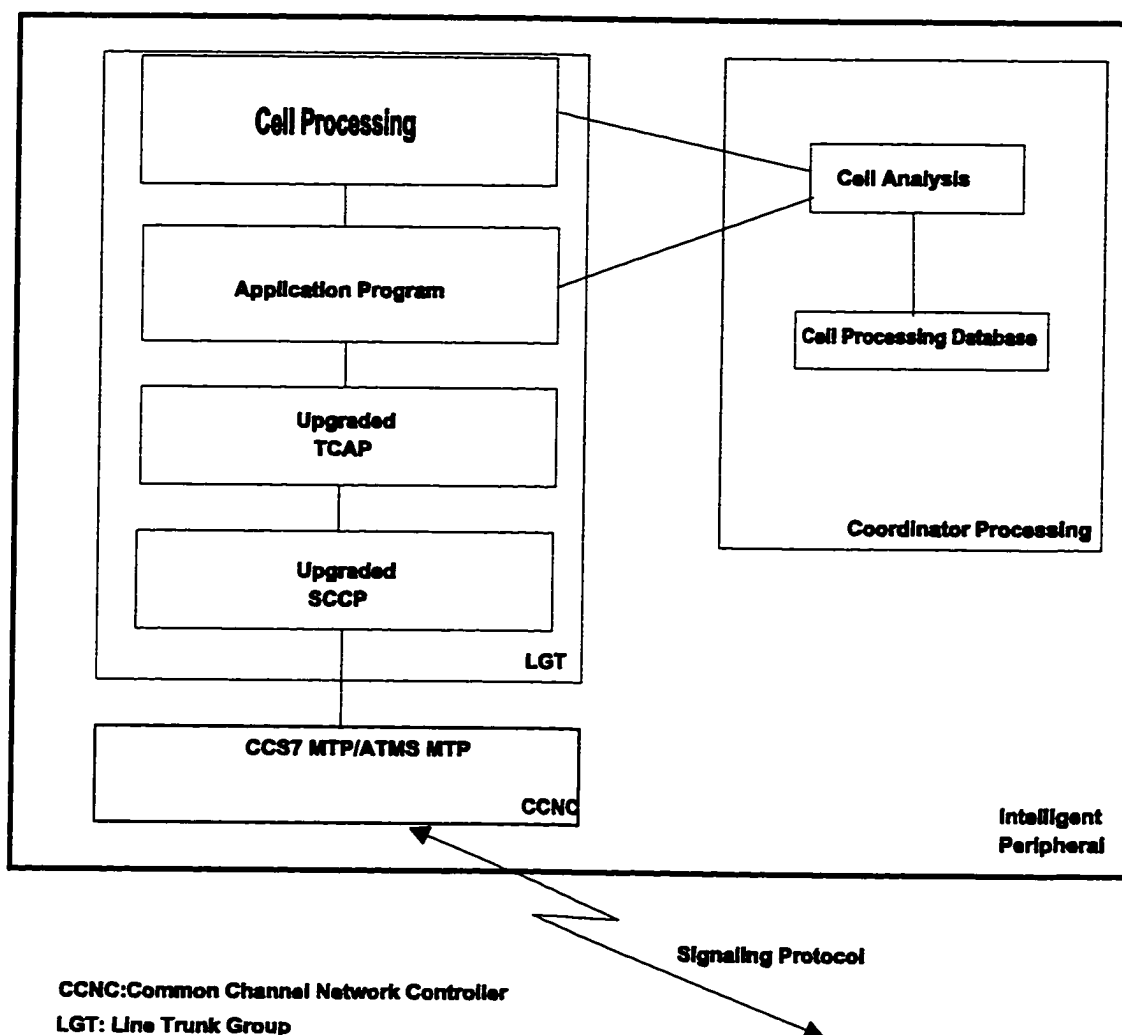
## **INTELLIGENT PERIPHERAL**

The IP provides specialized telecommunications capabilities (e.g., announcements, prompts, voice recognition) required by I-ATM-N's SLPs. It will also provide cell processing function for Intelligent ATM Services. When an ATM call is invoked, the switch will transfer the cell header with needed information to detect the type of the call to the IP. The IP recognizes, through its "trigger table", IN-ATM call handling requirements. Information from either a calling or called line may be used by the IP to trigger an ATM-IN call processing routine. An IP launches queries to other ATM-IN modules requesting call processing instructions and acts on those instructions.

The IP will need to have some new capabilities (see figure 6.17):

- Application parts which define the services that are supported
- A Transaction Capability Application Part (TCAP), which provides a part of the application layer protocol.
- A network layer, which defines the communication network for which the protocol is designed. The SCCP defined as the network layer can be based on

either the actual, but updated to fit the new services, CCS7 MTP or an ATM signaling MTP (ATMS MTP).



**Figure 6.17: Intelligent Peripheral new capabilities.**

### ADJUNCT SERVICE POINT

The ASP typically resides at the IP and responds to SLI requests processing.

### SERVICE MANAGEMENT SYSTEM

The SMS supports service provisioning and database administration.

## SERVICE CREATION ENVIRONMENT

The SCE supports the introduction of network services features, from definition through verification and deployment.

Interface between the local exchange customers and the network is accomplished through a small set of transaction-oriented, standard user-to-network interfaces; both ISDN and traditional (e.g., POTS) access is envisioned.

## FUNCTIONAL COMPONENT

An FC is an element network call processing action unique to the IN/2 architecture that directs internal (e.g., collecting dialed digits), and can be used for the I-ATM-N. FCs can be combined in different ways to create different network services or features (e.g., Transfer of Control, Connection Control, Network Participant Interaction, Network Information Management, Processing, and Information Collection).

The definition of service independent FCs ensures that the I-ATM-N architecture will not be limited by the characteristics of a particular service, or by a particular service implementation.

### **6.3.**

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## **CONCLUSION**

The three emerging technologies (transmission, switching, and data-base) unified into one composite science of network architecture are modifying the course of IN evolution.

Information vendors play a critical role in supplying customer information via the network. There is a definite need for these Vendors to seek and supply information from their own data bases. The queries can be complex and also in a natural language. For this reason, The front end processors of the vendor data base systems need the recently induced natural language.

The Intelligent ATM Network will start a new era of information technology.

## Chapter 7

### **SIMULATION ENVIRONMENT**

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In simulation analysis, the computer traces out in detail the implications and consequences of a proposed network or change. As result, simulation is more realistic and simulation results are easier to understand than with other forms of analysis. Level of detail become Important. This chapter is an overview of the simulation techniques in a communication network environment, as well as the simulation packages taken in consideration to study the Intelligent ATM networks, and a an introduction to the actual topology of a national networks. The results on next chapter will be based on this topology.

#### **7.1**

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### **SIMULATION TECHNIQUES**

The simulation procedures introduced in the late seventies assume the role of elaborated computer aided design (CAD) when optimization is needed. The consistency and interpolarity of the simulation programs play a critical role in the CAD procedure.

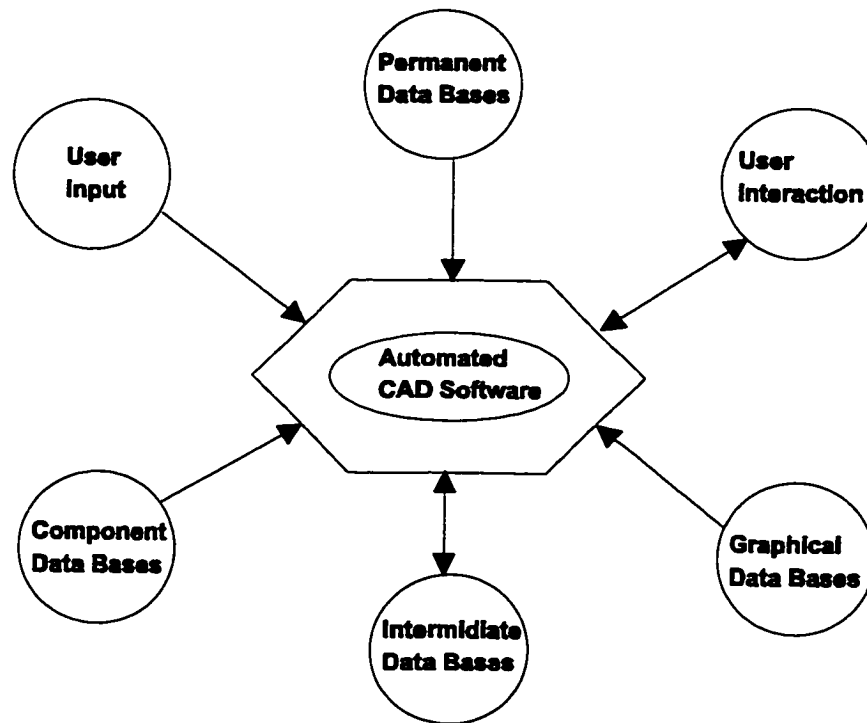
In the typical computer aided design (CAD) facilities, there are three phases of interaction with the designer. The first phase consists of defining the system configuration. The system components are assembled by a set of interactive graphic routing. These routines spell out to the designer all types of components available in the component library. The system configuration is then specified as the user selects the components from the library according to the network function. The system then presents the user with vendors in the data bases for each of the components selected. The vendor choice for each component is user defined input. This procedure is also carried out in an interactive graphics mode. The system permits the user to backtrack at any stage and change the components or the vendors selected. Once the system is configured and vendor selection is complete, the system offers the user the choice to generate a hard copy of the final system before Simulation starts.

During the second phase dealing with the simulation of the entire system, the user is actively informed of the status of the simulation procedure and any errors resulting from user inputs or posted data base access. Recoverable errors are tackled by the system and non-recoverable errors terminate in the CAD facility.

A large variety of analysis programs are activated in the second reentrant series of computer programs.

The final phase permits display (and generation of the hard copy) of the system performance. This phase also operates in an interactive mode. The system offers the designer to see the time domain results as wave shapes at any nodes in the studied system. The designer can track signals and their strength at different nodes in the system.

An overview of the general purpose CAD software is depicted in figure 7.1. Four major data bases (permanent, component, intermediate and graphical) are activated during the CAD function.



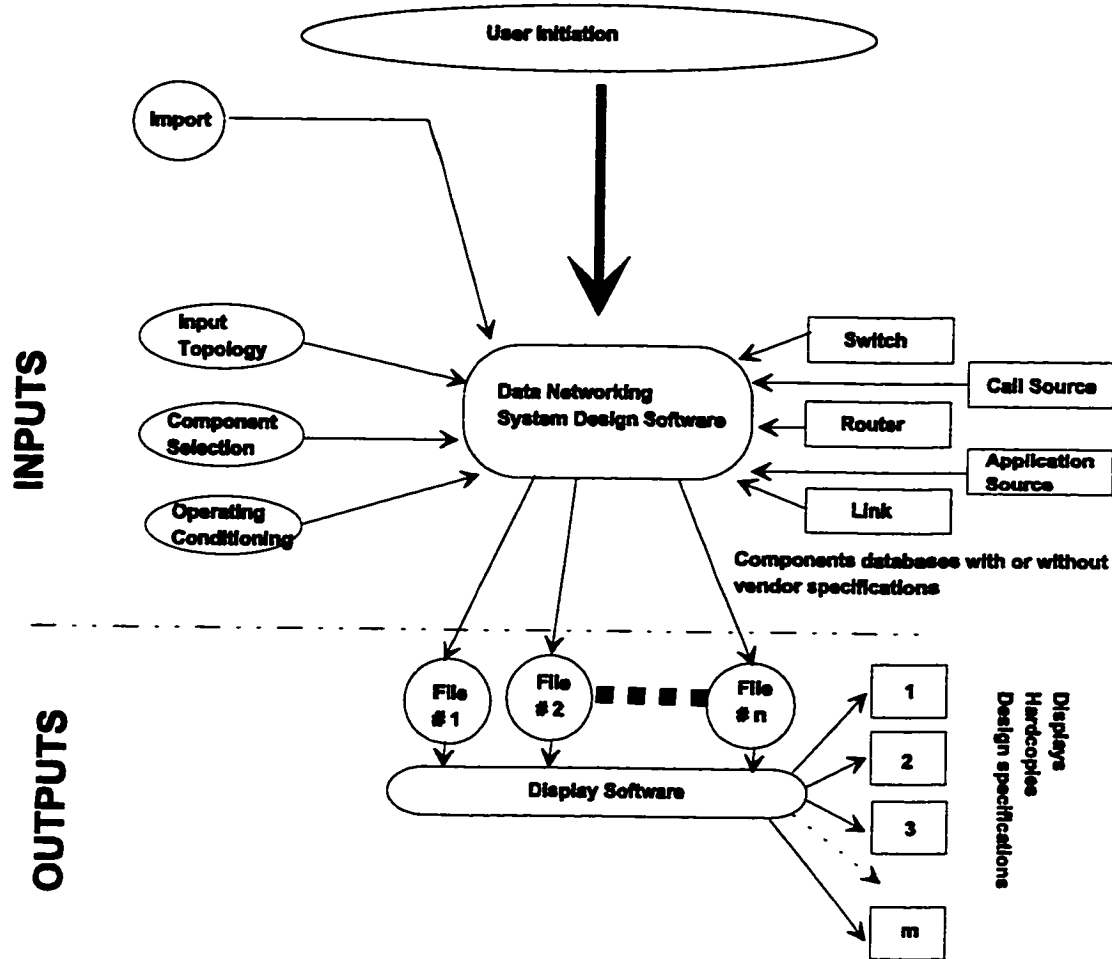
**Figure 7.1: Overall Software Organization. CAD -Computer Aided Design**

User input and interaction facilitate the fine tuning of software to the specific application being processed. In particular, this model has been used in most of the software.

Permanent data bases hold the characteristics of system components that retain specific characteristics.

As communication networks expand in scope and diversity, the planning challenge has become immense. Network monitoring tools that collect traffic data, discover and display network topologies, and display alarms are in wide use. There is a great need, however, for network design and planning tools that allow new networks or proposed challenges to existing networks to be quickly evaluated before committing to a course of action. This need for performance prediction is met through simulation analysis.

In simulation analysis, the computer traces out in detail the implications and consequences of a proposed network or change. As result, simulation is more realistic and simulation results are easier to understand than with other forms of analysis. Level of detail become Important. Generally the simulations are done over a long enough period to have some degree of Confidence in the Results. The Aim of this simulations is to relieve congestion, improve performance, reduce number of retries, increase revenues, enhance network utilization, design and develop the existing Network for the projected Growth, use and incorporate New Technology, select between Design alternatives, maximize the return on investments, meet the projected demand, Introduce New Services, etc.



**Figure 7.2: Software Organization. Network Simulation Packages**

In any communication network simulation package, the network topology, workload, and protocols are to be defined.

Network topology describes the layout and resources that model the desired aspect of the given physical network. Topology consists of nodes to represent hardware, links to represent transmission media between the nodes, and ports to associate nodes to links and to model the node's connection to the links. Nodes model computers, switches, routers, multiplexers, or any type of communication processing or computing device. Links model any type of transmission medium.

Nodes and links each have time to failure and time to repair distributions for modeling failures or down time. A port model a node's interface to a particular link.

## 7.2

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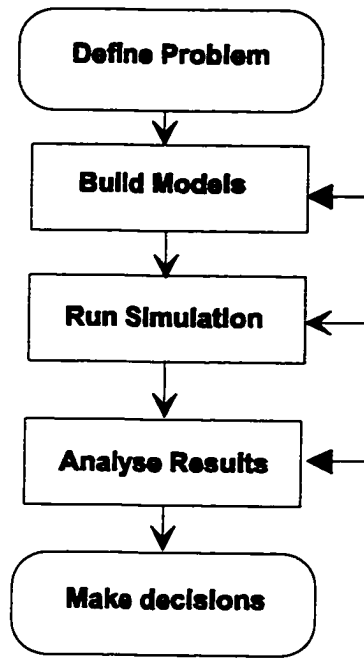
# DATA COMMUNICATION SIMULATION PACKAGES

The CAD process involves simulation. The man-machine feedback occurs when the designer defines the system, and studies the simulated results to evaluate system performance. System optimization is carried out by the iterative procedure of readjusting the input parameters till a satisfactory design is negotiated.

The simulation software consists of five major segments consisting of defining the configuration of the system ( see figure 7.3):

- Defining the configuration of the system
- System Final Drawing, and selection of the components from vendor databases
- Approving final configuration and component selection (Simulation)
- Display Generation (decide which parameter is to be displayed)
- Make a decision

Graphics is used extensively to facilitate the user interface.



**Figure 7.3: Simulation software major segments.**

To predict an end-to-end delays, throughputs, and utilization of links, buffers, and processors two simulation packages has been taken in consideration.

## 7.2.1

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### OPNET

OPNET can be utilized as a decision support tool to provide insight into the performance and behavior of existing or proposed networks, systems, and processes. To provide useful data, network models must combine accurate descriptions of topology, data flow, and control flow. Since no single paradigm of visual representation is ideally suited for all three of these model types, OPNET utilizes separate model format for each.

OPNET Network models define the position and interconnection of communicating entities, or nodes. Each node is described by a block structured data flow, or OPNET Node Model, which typically depicts the interrelation of processes, protocols, and subsystems. Each programmable block in a Node Model has its functionality defined by an OPNET Process Model which combines the graphical power of a state-transition diagram with the flexibility of standard programming language and broad library of pre-defined modeling functions.

The OPNET Network Editor graphically capture the physical topology of a communication network. Networks consist of node and link objects, which are graphically assembled and parameterized via pop-up dialog boxes. To create node objects, users select node types from a library of example and user-defined models. Each OPNET Node Model has a specific set of attributes that are used to configure it.

Network models can be constructed in a dimensioned workspace with a user-selected grid. Since users may define node abstractions, OPNET Network Models may represent LAN's, MAN's, WAN's, on-board vehicular networks, or any combination thereof. Subnetwork objects can be used to structure an unlimited topology hierarchy within any Network Model.

The OPNET Node Editor graphically captures node architectures, which are diagrams of data flow between modules typically representing hardware and software subsystems. Module types include processors, queues, and traffic

generators, receivers, and transmitters. Processors are general modules that provide complete flexibility in protocol and algorithm specification. The functionality of processors and queues objects is defined using OPNET Process Models.

The OPNET Process Editor uses a state-transition diagram approach to support specification of any type of protocol, resource, application, algorithm, or queuing policy. States and transitions graphically define the progression of a process in response to simulated events. Within each state, general logic can be specified using a library of over 350 pre-defined functions. The full flexibility of the C programming language is also accessible. As with other OPNET editors, users can construct entirely new process models, or modify those that are provided.

Once a set of OPNET Network, Node, and Process Models are fully defined, users can run simulation studies based via the OPNET Animation Viewer, and plot statistical performance measurements based on simulation studies of the OPNET Analysis Tool.

The OPNET Analysis Tool provides a graphical environment that allows users to view and manipulate data collected during simulation runs.

Standard and user-specified probes can be inserted at any point in a model to collect statistics. Simulation output collected by probes can be displayed graphically, viewed numerically, or exported to other software packages. First and second order statistics on each traces as well as confidence intervals can be

automatically calculated. OPNET supports the display of data traces as time-series plots, histograms, probability density and cumulative distribution functions, and scattergrams. Graphs may be output to a printer or saved as bitmap files to be included in reports or proposals.

OPNET is a comprehensive software environment for modeling, simulating, and analyzing the performance of communication networks, computer systems and applications, and distributed systems.

### **7.2.2.**

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## **COMNET**

COMNET is used to simulate the network's detailed operation producing dynamic animation and reports depicting network performance. It predicts end-to-end delays, throughput, and utilization of link, buffers, and processors reproducing random, bursty traffic patterns to view peaks and valleys of traffic and not just snapshots and average. It pinpoints sources of delays and bottlenecks.

COMNET has been chosen to simulate the intelligent ATM network because of its capability to simulate the circuit-switched traffic and the packet-switched traffic on the same platform (e.g. circuit switching is used to send voice over digital transmission.). Regardless of whether it is used with packet-switching or alone, circuit-switching brings with it a number of challenges that are distinct from packet-switching.

COMNET III object oriented framework offers the possibility of creation of a large number of scenarios. COMNET answers questions about link and node utilization, message delays, and congestion points in a large network. It can answer questions about the performance of the current network topology based on historical or projected traffic loads.

As in any other communication network simulation package, the network topology, workload, and protocols are to be defined.

### **7.2.2.1. TOPOLOGY**

The topology is defined by switches (nodes) that are interconnected by transmission facilities (links).

#### **NODES**

COMNET has three types of nodes: Processing nodes, Router nodes, and Switch nodes. Each node type can switch packets or calls.

*Processing\_nodes* model computer hosts as well as communication processing devices. Each processing node has an internal processor that executes software applications and processes packets. A processor node also has a disk storage device to simulate file reads and writes. A number of identical computers can be represented by a single Group Node. At simulation time, COMNET III expands the group node into the specified number of Processor nodes, each with its own independent traffic sources based of the initial group node. When the model schedules an application for execution at a processor node, the application waits in the node's pending queue until the internal processor is available. When it

becomes available, the processor runs the application by executing in sequence each command requested by the application. The commands are defined in the node's *Command Repertoire*. There are commands for processing data, sending and answering messages, reading and writing files, establishing sessions, and suspending and application until a required message arrives.

*Router Nodes* model routers, bridges, switches, hubs, and other devices that have shared internal backplane that moves packets between port buffers. The backplane is characterized by a bus and a bus count. In addition to the internal backplane, the router node has all of the capabilities of the processor node, so it can, for example, run applications that update routing table and distribute routing table information. Ports on a Router node can share a line card, which allows packets to move from an input port to an output port on the same line card without transiting a backplane bus. Nonblocking switch fabrics may also be modeled with Router node by specifying a bus count that is at least equal to the number of line cards on the node.

In the case of circuit switched networking, different routing algorithms are available for routing a call to its destination. Routing modeling capabilities include:

- ◇ The backbone and each of its top-level subnets have their own routing algorithm and routing tables.

- ◇ Routing is based on a shortest-path routing algorithm or user-defined routing tables.
- ◇ Shortest-path routing uses a flexible distance metric for each link that can vary by routing class and by link utilization.
- ◇ User-defined routing tables can also vary by routing class.
- ◇ User-defined table routing uses a route selection criterion when more than one route is listed; criteria include dynamic alternate routing, first available, maximum idle bandwidth, random list, and round robin.
- ◇ Calls traversing a node or a link that fails can be rerouted or simply disconnected.

*Switch Nodes* model switches as well as routers, hubs, and other communication devices that have an insignificant delay in moving packets from input port buffers to output port buffers. Packet in an input buffer always switch to the appropriate output buffer in zero time, provided buffer space is available. The switch node models head-of-line blocking for those switches that allow packets to wait at an input buffer for the required output buffer to become available.

### LINKS

Links are used to model a variety of different transmission media. The physical characteristics of a link are defined in terms of bandwidth and propagation delay. The link physical transmission unit in COMNET is a frame. Packets are generated or assembled into frames for transmission on a link. Each link has a

frame size (minimum and maximum), a frame overhead, and a frame error rate. There are different types of links in COMNET corresponding to the type of medium access protocol used when multiple nodes need to share the same medium: ALOHA, Priority FDDI, Point-to-Point, Token Passing, CSMA, Polling, Token Ring, CSMA/CD or Ethernet, FDDI, DAMA, CSMA/CA. The CSMA/CD library provides parameter sets based on IEEE 802.3 standard. The Token-Passing library provides parameter sets based on the IEEE 802.4 and 802.5 standards. The CSMA/CA library provides parameter sets based on IEEE 802.11 standard. The point to point links modeling ISDN, and SONET are a part of the COMNET library.

### **WAN CLOUDS**

Used to model frame relay, cell relay, and packet switching services. The WAN cloud contains an access link for each node connected to the cloud. An access link models the connection from a user site to a network service's point-of-presence. Each access link is a point-to-point link with some number of channels, a channel transmission rate, and a propagation delay. The cloud has a delay attribute that determine the time required for a frame to transit the cloud from one access link to another.

### **SUBNETS**

The subnet object is used for modeling a topology hierarchically so that separate subnets have independently a separate routing algorithms that are also independent from the backbone.

Connections between the internal topology of the subnet and the backbone topology are through the Access Points. There may be as many access point as required. However, if the traffic will be routed into the subnet based upon the routing algorithm at the next higher level. Typically the backbone routing algorithm is used.

### **7.2.2.2. WORKLOAD**

The workload in COMNET is defined by traffic sources representing voice and data communications over the network. In the case of circuit switching, for example, the workload is defined by call sources at each node that generates calls during the simulation. Each call originating at a particular node has a bandwidth requirement, a destination drawn from a probability-weighted list, a duration drawn from a call holding time distribution, a routing class, and other attributes.

Workload modeling capabilities include:

- ⇒ Arbitrary number of call sources per node.
- ⇒ Circuit Switched call sources and packet switched message sources can be combined in the same model
- ⇒ Call scheduling can be based on times drawn from probability distribution, an external file of call arrival events, or a triggering event, such as delivery of a message.
- ⇒ Combinations of randomly-generated and externally-scheduled calls are permitted.

- ⇒ Blocked and preempted calls can try again later based on a call-retry distribution.
- ⇒ High priority calls can preempt low priority calls.
- ⇒ Probability distributions available for modeling call interarrival times and holding times include the beta, erlang, exponential, gamma, hyper-exponential, log-normal, normal, pareto, triangular, uniform, and weibull.
- ⇒ User-defined table distributions are available for empirical data not well-modeled by standard probability distributions.
- ⇒ All probability distributions can be time-varying to model both time-of-day effects and networks spanning multiple time zones.
- ⇒ When utilization levels exceed user-defined thresholds, alarms are displayed, trace options can be activated, and traffic sources can be triggered.

### **7.2.2.3. TRAFFIC SOURCES**

#### **APPLICATION SOURCES**

Generate applications that execute on Processor or Router nodes. Each application source specifies a sequence of commands. When an application begins execution on the processor of the node connected to the application's source, the node runs each command in the sequence. If the command is in the node's command repertoire, the node runs the local command, otherwise, it runs the command in the model's global command.

### CALL SOURCES

Generate circuit-switched calls. When a call originates, the model attempts to establish a dedicated, end-to-end circuit of the specified bandwidth. In order for a call to use a path, each node and link along the path must have enough idle bandwidth to satisfy the call's bandwidth requirements.

### SCHEDULING

Traffic sources generate data using time-based scheduling or received-message scheduling. Call sources originate calls using time-based scheduling. With time-based scheduling, a source data (or calls) periodically, with the interval between applications given by an inter-arrival time distribution.

#### **7.2.2.4. PROTOCOLS**

Physical and data link functions of the OSI reference model, including medium access control are performed in COMNET III by links.

Network layer functions are performed in COMNET III at nodes, which make routing decisions for packets. Each packet has a routed protocol that can be used as one of the factors to determine both packet's port processing time and node processing time. The node's routing decision are based on a routing algorithm.

The backbone and each of the subnets have separate and independent routing algorithms. The routing algorithms in COMNET III are based on a shortest path computation that automatically initializes and updates routing tables. The various shortest path algorithms differ in the metrics used to compute shortest

path and in how the routing table updates are scheduled. Static algorithms compute the tables only at the start of the simulation. Dynamic algorithms periodically update the tables based on dynamic measures. Both static and dynamic algorithms update the tables whenever a node or link fails. Multipath routing, which load balances traffic among routes with metrics within some specified percentage of the best metric, can be modeled. In addition to a number of shortest path routing algorithms that automatically populate the routing tables, there is a possibility to define a new table. When selecting user-defined routing tables, route selection criteria becomes possible.

Routing decisions can also vary by a packet's routing class, which allows different types of traffic to be routed over different kinds of facilities. To model connection-oriented routing of sessions (so the traffic associated with a session follows the same path through the subnet), some of the following algorithms can be used:

- ◇ RIP Minimum Hop,
- ◇ Shortest Measured Delay,
- ◇ Link-State Shortest Path First/OSPF,
- ◇ Minimum Penalty
- ◇ User-Defined Routing Table

End-to-end or transport layer function of the OSI model are modeled in COMNET III by the transport protocol assigned to each message. Messages are

produced by Transport, Answer, and Setup commands. Different commands can reference different transport protocols.

The transport layer model includes parameters for specifying packet size and overhead bytes, which flow control mechanism to use, and whether or not to retransmit blocked packets. Each transport protocol has a routed protocol identifier that is one determinant of the packet processing time at multiprotocol routers and nodes.

Each message produced during a simulation is segmented into packets by the message's transport protocol. The packets are transported, subject to flow control, to the destination node, where the message is reassembled. By explicitly modeling transport layer segmentation and Reassembly of messages, COMNET III can provide measures of end-to-end delays for messages such as files, e-mail, inquiries, or responses, in addition to the end-to-end delays experienced by the packets formed from such messages. When workload is characterized simply by packet volumes, more meaningful measures of message-level delay are not available.

The Setup Session command models events that typically are included at the session layer of the OSI model.

The effects of presentation and application layer functions in the OSI model are accounted for in a COMNET III model by the specific commands executed by various applications at the nodal processors.

COMNET III includes traffic management features for improved ATM and frame relay modeling, including:

- ◇ Burst definitions in term of ATM cell rates or frame burst sizes
- ◇ Burst option for traffic policing
- ◇ Packets spaced by a fixed time between packets (CBR) or by burst limit (VBR, ABR).

### **BUFFERING POLICY ENHANCEMENTS**

COMNET III includes these buffering policies:

- ◇ Buffer sorting options by priority, DE, or FIFO
- ◇ Buffer may mark packets with congestion flags Based on buffer use relative to a user-defined threshold
- ◇ Option for rejecting the remaining frames of a higher level packet if a frame is already lost at the buffer
- ◇ For ATM, early packet discard (EPF) and partial packet discard (PPD)
- ◇ Virtual cut-through nodes which are useful for ATM switches and switched LANs

## **7.3**

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# **DESCRIPTION OF A NATIONAL NETWORK**

Global Description of the Telecommunication Network in Morocco.

### **7.3.1.**

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## **Current Topology**

The National Network is divided into height areas (area codes):

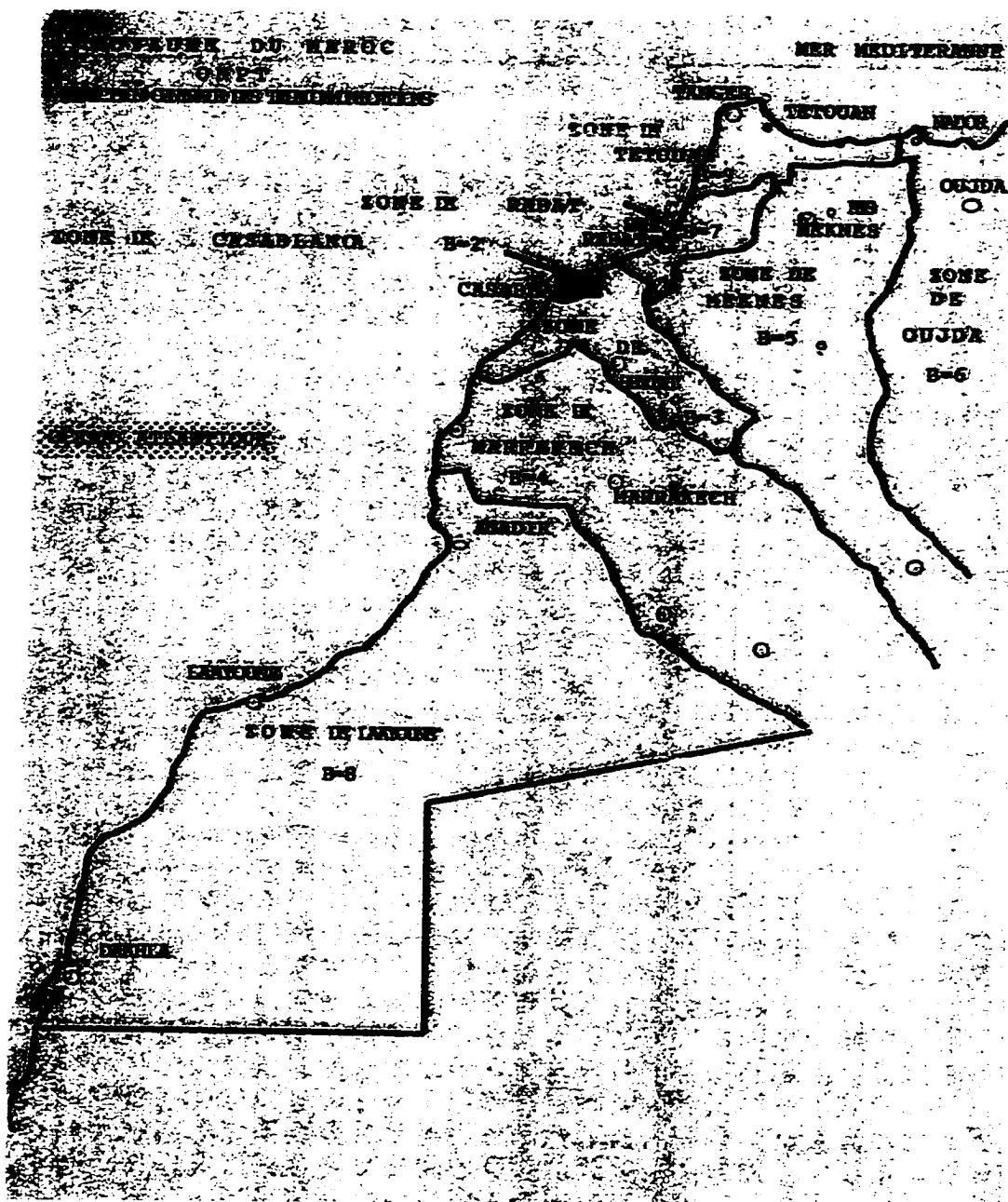
CasaBlanca Region (B=2), Settat Region (B=3), Marrakech Region (B=4),

Meknes Region (B=5), Oujda Region (B=6), Rabat Region (B=7), Laayoune Region (B=8), Tetouan region (B=9).

Concerning traffic transport, the National Network is subdivided into seven transit regions: CasaBlanca, Rabat, Marrakech, Agadir, Oujda, Meknes, and Tetouan. (CasaBlanca has got two area codes).

Each region has got a national center of transit (CTN). All CTNs are interconnected with each other.

To send telecommunication traffic to the international, the network is equipped with two international transit centers (CTI) both belong to the AXE 10 family and located in CasaBlanca and Rabat. This CTIs are interconnected with each other and with all other CTNs. CTI Rabat serve the CTNs of Meknes, Oujda, Rabat, and Tetouan as first choice, and then the CTNs of Agadir, CasaBlanca, and Marrakech. CTI CasaBlanca serve the CTNs of Agadir, CasaBlanca and Marrakech as a first choice, and then the CTNs of Meknes, Oujda, and Tetouan as a second choice.



**Figure 7.4: Map of Morocco indicating Major Regions in the National Telecommunication Network.**

	CASA			RABAT			MEKNES			OUJDA			MARRAKECH			AGA DIR			TETOUAN			FROM
	CTN	CTI	CTU	CTN	CTI	CTU	CTN	CTI	CTU	CTN	CTI	CTU	CTN	CTI	CTU	CTN	CTI	CTU	CTN	CTI	CTU	
CASA	150	119	435	30	90	45	301	135	120	135	45	45	301	135	120	120	135	120	120	135	1425	
	150	18	80				60						60								353	
	120	30																			150	
RABAT	435	40	55	150	225	135	120	90	120	90	135	90	120	90	120	150	80	150	150	1400	1400	
	150		95	120			60						60								595	
			0	0																	0	
MEKNES	90		225			60	30			60			30			30					435	
																					0	
																					0	
OUJDA	45	45	165	60	60	60	30			60			30	15	30	30					450	
																					0	
																					0	
MARRAK	255	120	120	60	28	30				30			45			30					688	
																					0	
																					0	
AGADIR	135		90				45			15					15						285	
																					15	
																					0	
TETOUAN	120	45	150	40	30	30	30			30			30	15							460	
																					0	
																					0	
TO	1500	430	137	1360	245	270	433	0	0	450	0	0	676	0	0	300	0	0	455	0	6256	

**Table 7.1: Moroccan Network; Links' Load capacity. (Links connecting between major nodes)**

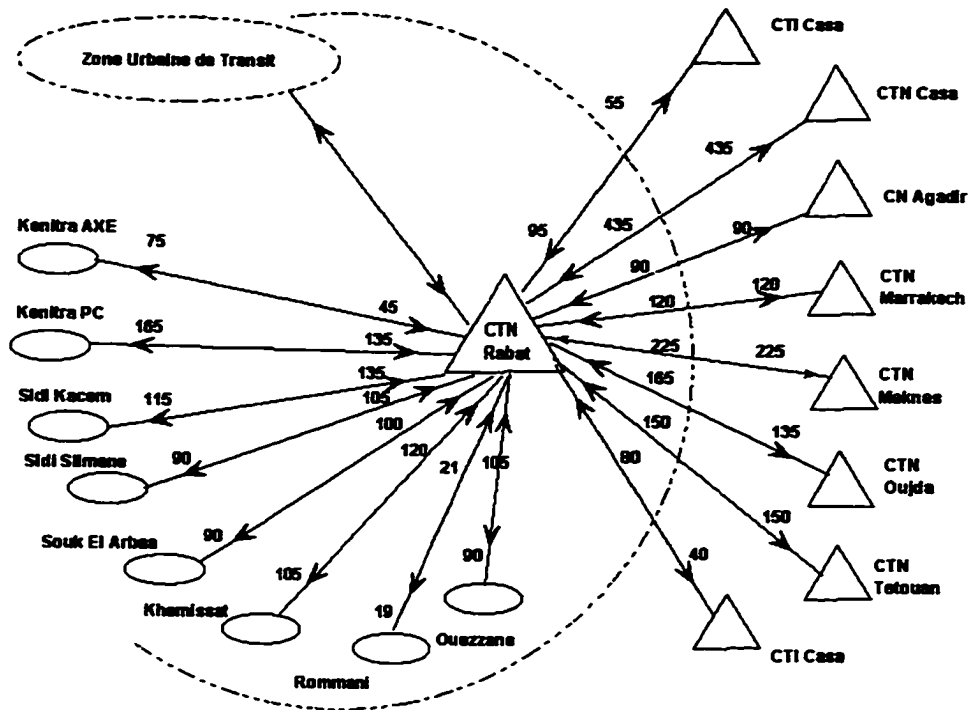


Figure 7.5: CTN Rabat; Topology

ZONE DE TRANSIT DE RABAT

	FROM CNT RABAT		TO CNT RABAT	
OTHER CTN AND CTI	1400	65%	1360	64%
KENITRA AXE	75	4%	45	2%
KENITRA PC	165	8%	135	6%
SIDI KACEM	115	5%	135	6%
SIDI SLIMANE	90	4%	105	5%
SOUK EL ARBAA	90	4%	100	5%
KHEMISSSET	105	5%	120	6%
ROMMANI	12	0%	21	0%
OUEZZANE	90	4%	105	5%
Total	2142	100%	2126	100%

Table 7.2: CTN Rabat; Link Load Capacity

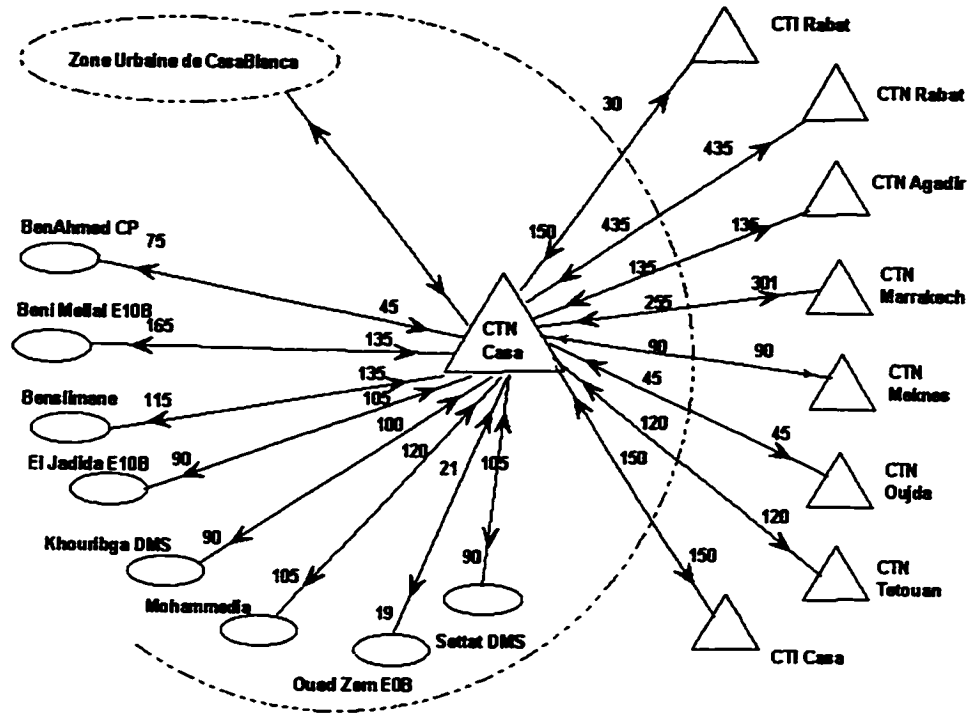
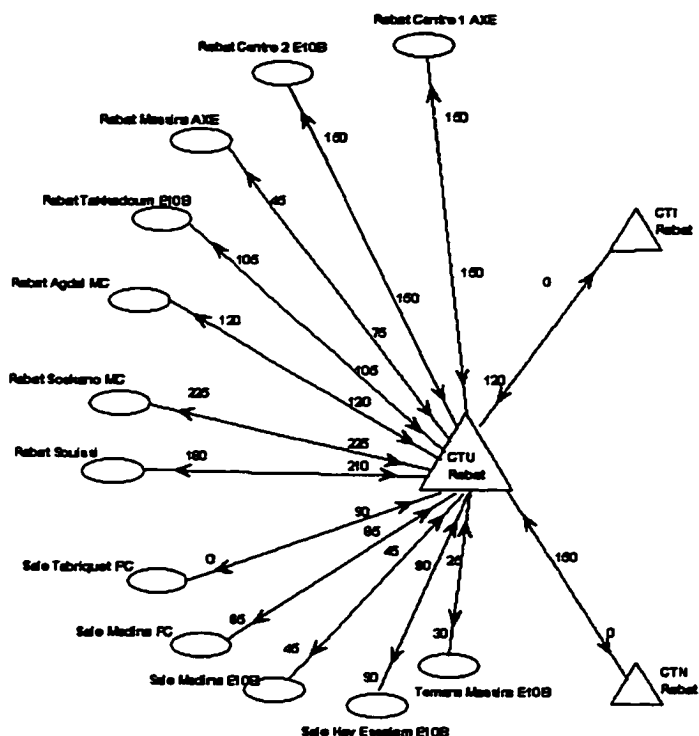


Figure 7.6: CTN Casa; Topology

ZONE DE TRANSIT DE CASABLANCA

	FROM CNT CASA	TO CNT CASA
OTHER CTN AND CTI	1400	1360
BENMHAMED CP	18	22
BENI MELLAL E10B	120	120
BENSLIMANE DMS	60	60
EL JADIDA E10B	150	210
KHOURIBGA E10B	90	120
MOHAMMEDIA DMS	150	150
OUED ZEM E10B	85	95
SETTAT DMS	180	209
<b>Total</b>	<b>2253</b>	<b>2346</b>

Table 7.3: CTN Casa; Link Load Capacity

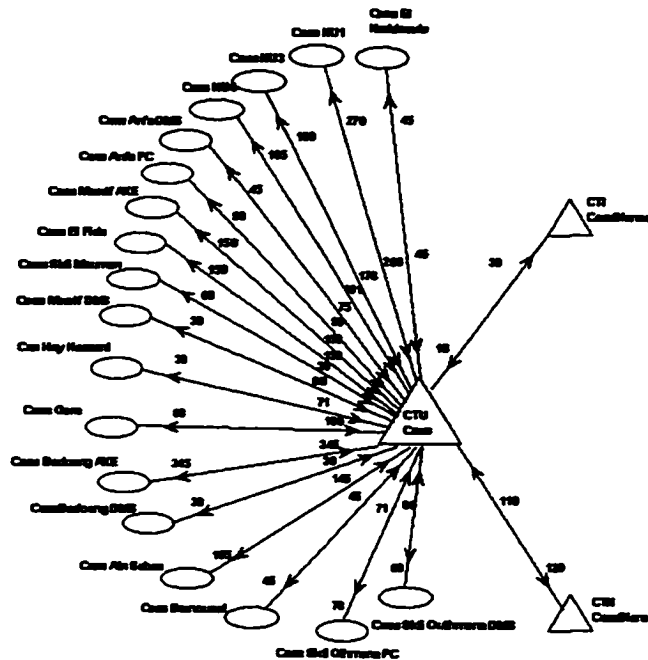


**Figure 7.7: CTU Rabat; Topology**

ZONE URBAINE DE RABAT

	FROM CTU RABAT	% FROM	TO CTU RABAT	% TO
OTHER CTN AND CTI	0	0%	270	10%
RABAT CENTRE 1 AXE	150	5%	150	5%
RABAT CENTRE 2 E10B	150	5%	150	5%
RABAT MASSIRA AXE	105	4%	105	4%
RABAT TAKKADOUM E10B	120	4%	120	4%
RABAT AGDAL MC	225	8%	225	8%
RABAT SOEKARNO MC	180	6%	210	8%
RABAT SUISSI PC	0	0%	90	3%
SALE TABRIQUET PC	65	2%	85	3%
SALE MEDINA PC	45	2%	45	2%
SALE MEDINA E10B	30	1%	30	1%
SALE HAY ESSALAM E10B	90	3%	90	3%
TAMARA MASSIRA E10B	30	1%	25	0%
	<b>1190</b>	<b>43%</b>	<b>1595</b>	<b>57%</b>
TOTAL CIRCUIT COMING INTO AND GOING FROM RABAT CTU :			<u>2785</u>	

**Table 7.4: CTU Rabat; Link Load Capacity**

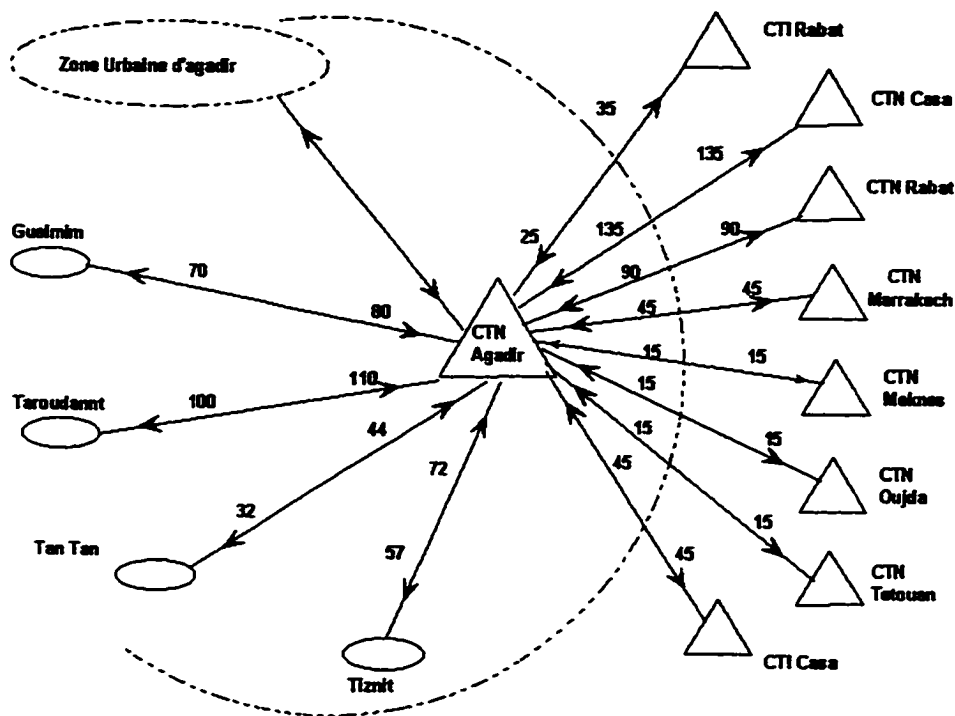


**Figure 7.8: CTU Casa; Topology**

ZONE URBAINE DE CASABLANCA

	FROM CTU CASA	% FROM	TO CTU CASA	% TO
OTHER CTN AND CTI	150	4%	137	3%
CASA EL HADDAOUIA	45	1%	45	1%
CASA NU1	270	6%	269	6%
CASA NU2	180	4%	175	4%
CASA NU3	105	2%	101	2%
CASA NU4	45	1%	75	2%
CASA ANFA PC	90	2%	90	2%
CASA MAARIF AXE	150	4%	150	4%
CASA EL FIDA	150	4%	150	4%
CASA SIDI MOUMEN	60	1%	60	1%
CASA MAARIF DMS	30	0%	30	0%
CASA HAY HASSANI	30	0%	71	2%
CASA GARE	90	2%	150	4%
CASA BADOENG AXE	345	8%	345	8%
CASA BADOENG DMS	80	2%	30	0%
CASA AIN SEBAA	105	2%	105	2%
CASA BERNOUSSI	45	1%	45	1%
CASA SIDI OTHMANE PC	78	2%	78	2%
CASA SIDI OTHMANE PC	60	1%	60	1%
	<b>2108</b>	<b>49%</b>	<b>2166</b>	<b>51%</b>
TOTAL CIRCUIT COMING INTO AND GOING FROM CASA CTU :			<b>4274</b>	

**Table 7.5: CTU Casa; Link Load Capacity.**



**Figure 7.9: CTN Agadir; Topology**

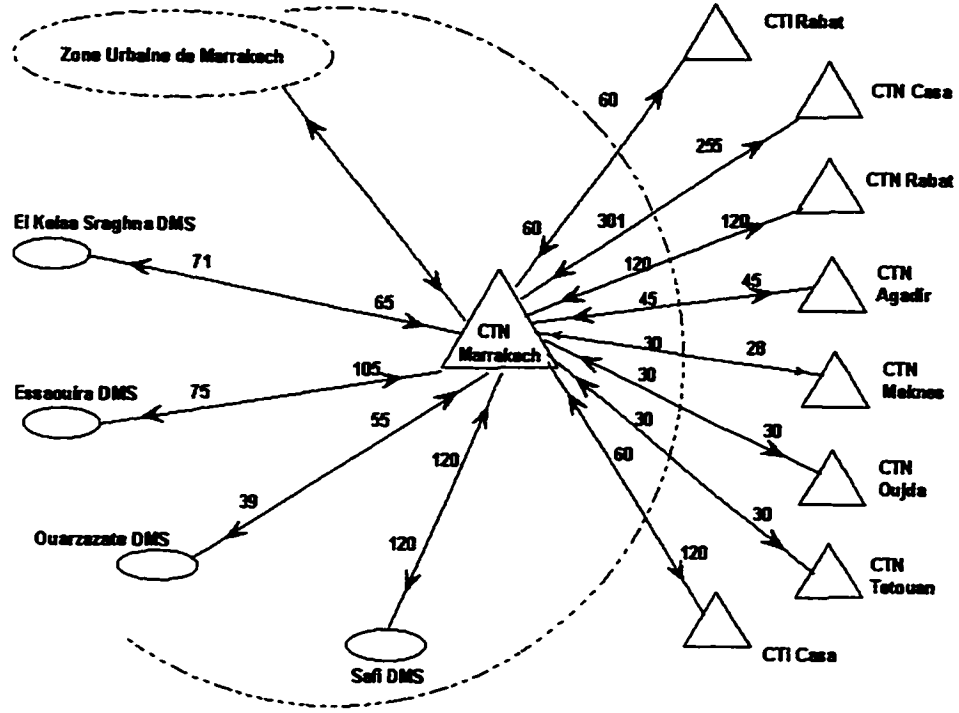
**ZONE DE TRANSIT DE AGADIR**

**CTN AGADIR :**

**TECHNOLOGY: AXE**

	FROM CNT AGADIR	TO CNT AGADIR
OTHER CTN AND CTI	285	300
GUELMIM	70	80
TAROU DANNT	100	110
TAN TAN	34	44
TIZNIT	57	72
Total	546	606

**Table 7.6: CTN Agadir; Link Load Capacity**

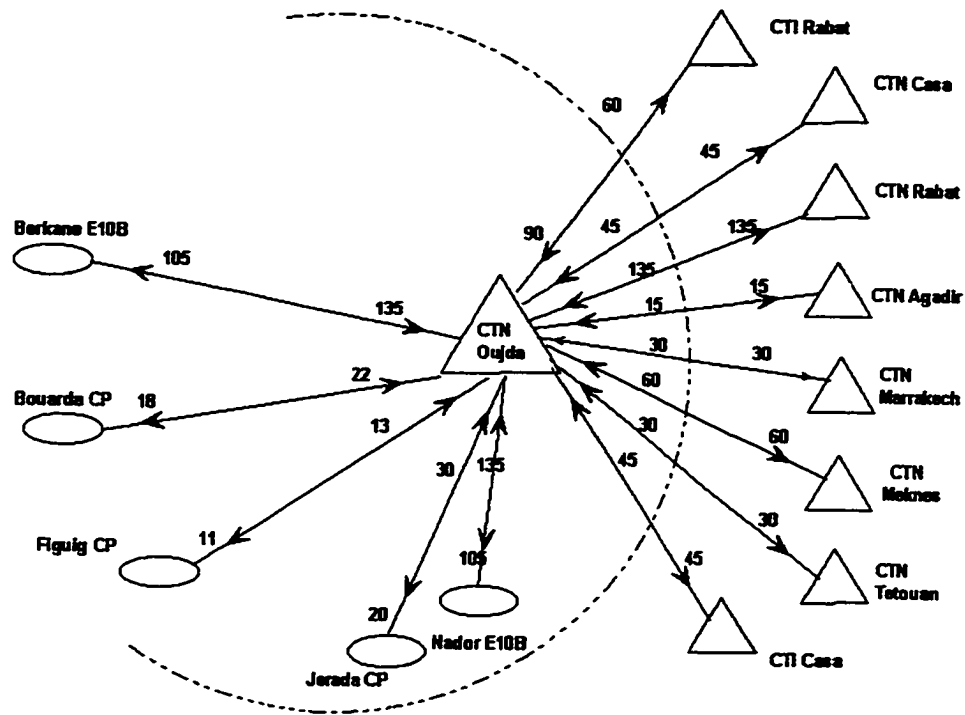


**Figure 7.10: CTN Marrakech; Topology**

**ZONE DE TRANSIT DE MARRAKECH**

	FROM CTN MARRAKECH	TO CTN MARRAKECH
OTHER CTN AND CTI	688	676
EL KELAA SRAGHNA DMS	71	55
ESSAOUIRA DMS	75	105
OUARZAZATE DMS	38	55
SAFI DMS	120	120
<b>Total</b>	<b>992</b>	<b>1011</b>

**Table 7.7: CTN Marrakech; Link Load Capacity**

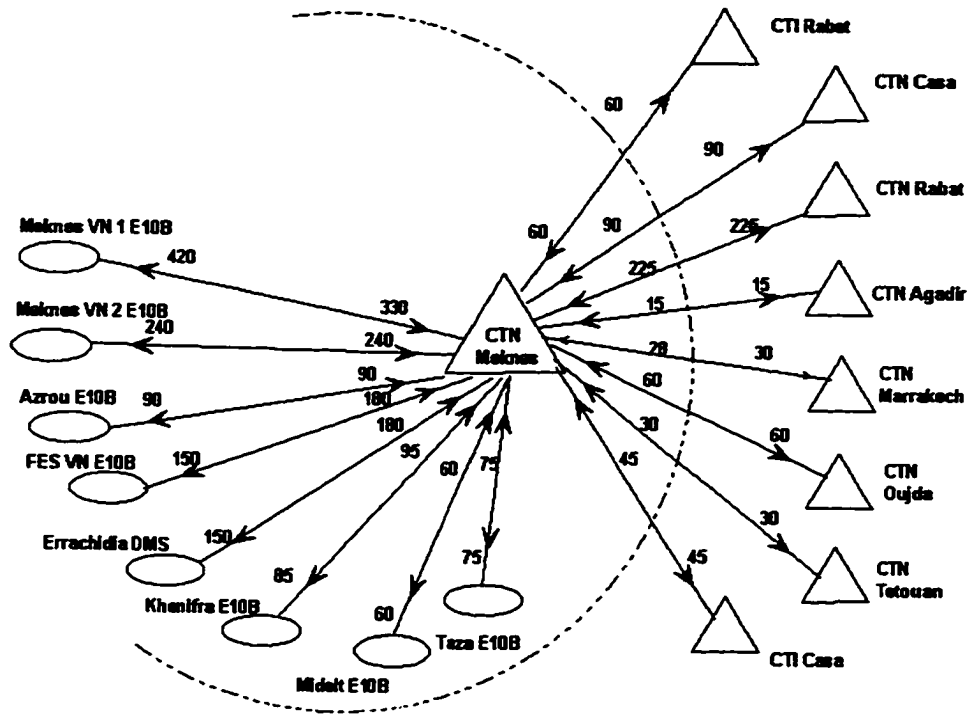


**Figure 7.11: CTN Oujda; Topology**

ZONE DE TRANSIT DE OUJDA

	FROM CTN OUJDA	TO CTN OUJDA
OTHER CTN AND CTI	688	450
BERKANE E10B	105	135
BOUARFA CP	18	22
FIGUIG CP	11	13
JARADA CP	20	30
NADOR E10B	105	135
<b>Total</b>	<b>947</b>	<b>785</b>

**Table 7.8: CTN Oujda; Link Load Capacity**

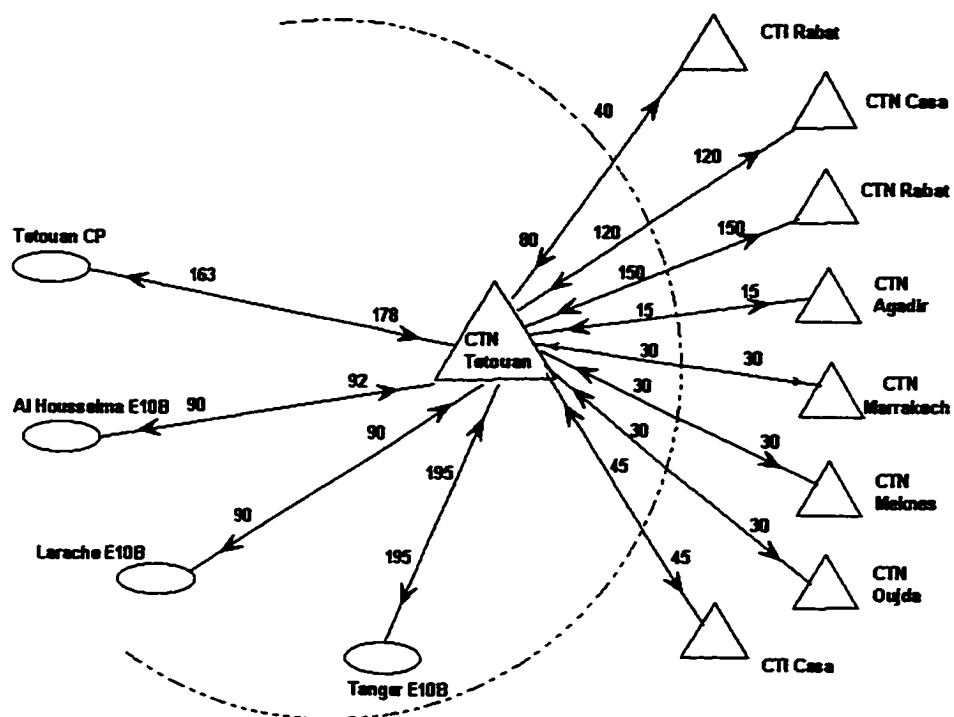


**Figure 7.12: CTN Meknes; Topology**

**ZONE DE TRANSIT DE MEKNES**

	<b>FROM CNT MEKNES</b>	<b>TO CNT MEKNES</b>
OTHER CTN AND CTI	435	433
MEKNES VN 1 E10B	420	330
MEKNES VN 2 E10B	240	240
AZROU E10B	90	90
FES VN E10B	150	180
ERRACHIDIA DMS	150	180
KHENIFRA E10B	85	95
MIDELDT E10B	60	60
TAZA E10B	75	75
<b>Total</b>	<b>1705</b>	<b>1683</b>

**Table 7.9: CTN Meknes; Link Load Capacity.**



**Figure 7.13: CTN Tetouan; Topology**

**ZONE DE TRANSIT DE TETOUAN**

	FROM CNT TETOUAN	TO CNT TETOUAN
OTHER CTN AND CTI	460	455
TETOUAN CP	163	178
AL HOCEIMA E10B	90	92
LARACHE E10B	90	90
TANGER E10B	195	195
<b>Total</b>	<b>998</b>	<b>1010</b>

**Table 7.10: CTN Tetouan; Link Load Capacity**

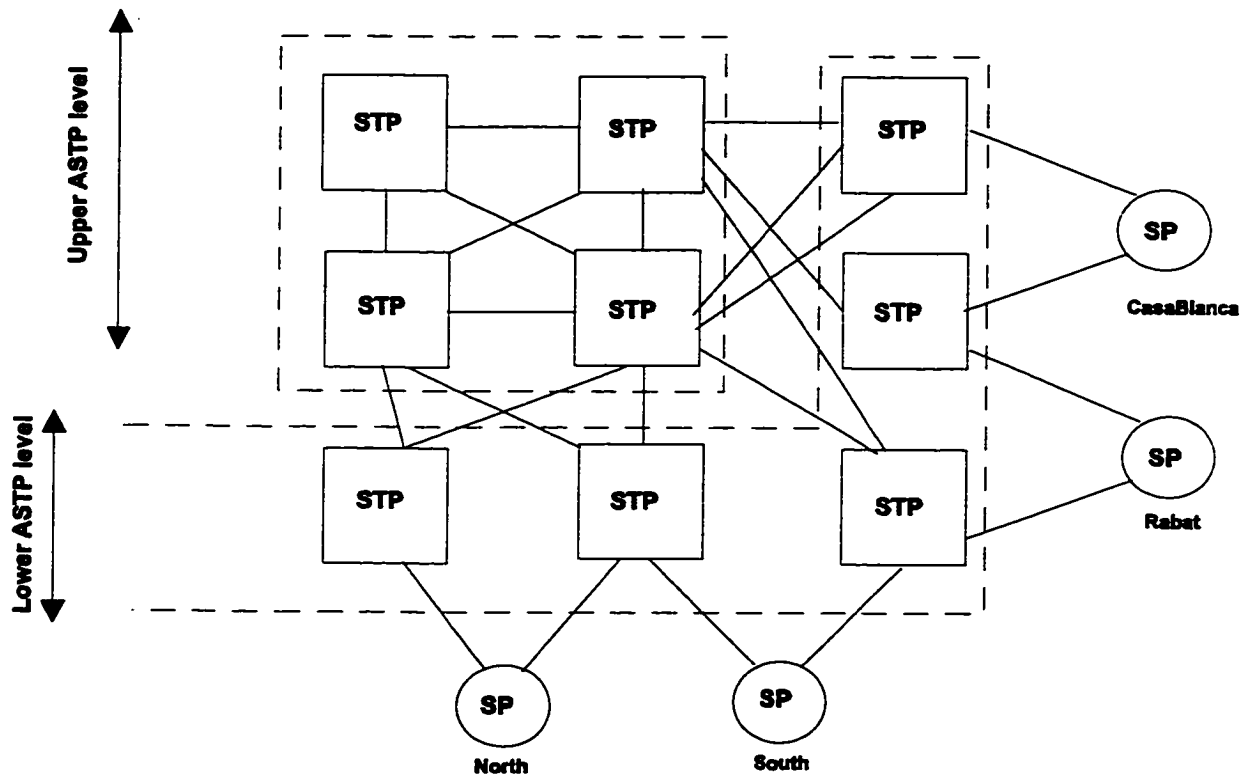
**Global network information**

	Total lines to and from	percent
CTN AGADIR	1152	4%
CTN CASA	4599	17%
CTU CASA	4274	15%
CTI CASA	783	3%
CTN MARRAKECH	2003	7%
CTN MEKNES	3388	12%
CTN OUJDA	1732	6%
CTN RABAT	4268	15%
CTU RABAT	2785	10%
CTI RABAT	840	3%
CTN TETOUAN	2008	7%
	<hr/> 27832	100%

**Table 7.12: Global network information****7.3.2.****Intelligent Networking in Morocco**

The Moroccan Network is not intelligent at the current stage. The Signaling System number 7 is not installed. Some intelligent services like the Green number have been provided using a non intelligent architecture. An example of how the Green number (recently created) functions is as follow:





**Figure 7.15: Common Channel Signaling Network to fit the Moroccan network topology.**

The Switching Points can be any tandem switch, connected to a signaling system.

A number of IPs can be connected to different switching points.

In the far future, when ATM is implemented for information transport, then the signaling for intelligent ATM networks services can be transferred through the transport network.

## Chapter 8

### **SIMULATION RESULTS**

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In recent years, many organization have started to integrate voice with data traffic, thereby combining circuit-switching with packet-switching to reduce operation and costs management.

The following results are based on an ATM switching systems supporting voice, data, and video transmission over optical networks. The study has been subdivided into 3 main parts where the transport networks as well as the signaling networks has been studied in an ATM environment. Firstly, the network supporting voice communications. Secondly, The network is supporting only data communication (as well as video) by using cell switching. And last, the combination of all is studied. The study is based on a national topology.

## 8.1.

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# **CIRCUIT SWITCHED TRAFFIC OVER ATM NETWORKS**

In this part, the circuit-switching is used separately to study the transmission of voice using ATM networks over digital transmission. First, the study will be performed on a two node network and then on a national network to see the extend and the nature of congestion that is likely to develop in the critical node and links on the backbone network. The introduction of ATM switching system and the expansion in the trunk capacity will be the subject of the second part of the simulation of the national network looking to get to the optimum solution for the studied national network (including a better quality of service).

## 8.1.1

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### **Circuit Switching Networks**

In circuit-switched communication, a dedicated end-to-end connection is established between the communicating parties. A signaling protocol is used to setup the circuit-switched connection. During the setup phase, a route is selected for the call and the required transmission capacity is reserved for the use during the call. Once call setup is completed, a call holds the reserved transmission capacity for the entire duration of the call and the capacity is unavailable to other users. The circuit established for a call typically passes through several switches and transmission systems. As a consequence, such an individual call

simultaneously consumes capacity on a sequence of different transmission devices.

Circuit-switching makes sense for network traffic where: 1) the holding time for a connection is relatively long compared to the setup time (making it worthwhile to incur the overhead of a setup delay), and 2) congestion-induced delays are unacceptable.

Voice traffic is well-switched to circuit switching. Voice call setup times are typically on the order of a few seconds compared to call holding times on the order of a few hundred seconds. In addition, voice signals must be transmitted without the introduction of intermittent pauses caused by delay variability. Public-switched Telephone Networks (PSTN) throughout the world use circuit-switching, as well as most private voice networks in commercial, government, and military organizations. Circuit-switched connections are also required by users of many mobile, wireless networks. Bulk data transfer that requires full-time use of a circuit for an extended period of time is also well-switched to circuit-switching.

Currently, many organizations are trying to integrate voice and data traffic on the same network resources in an attempt to reduce operation and management costs. Such integration can occur at different levels (transmission, switching) and in different forms (static, dynamic). A recent development in this area is the dynamic allocation of a bandwidth to circuit-switched and packet-switched traffic on the same digital transmission facilities.

In the circuit switching environment, the delays are to be minimized, circuits to be dedicated, quality of service to be kept in an appropriate level, and ultimately the call to go through.

Because of the large size of this reports, as well as the large number of diagrams that we get when simulating the national network (or even a region), only a flavor will be given.

### **8.1.2.**

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## **Simulating the Circuit Switching Networks**

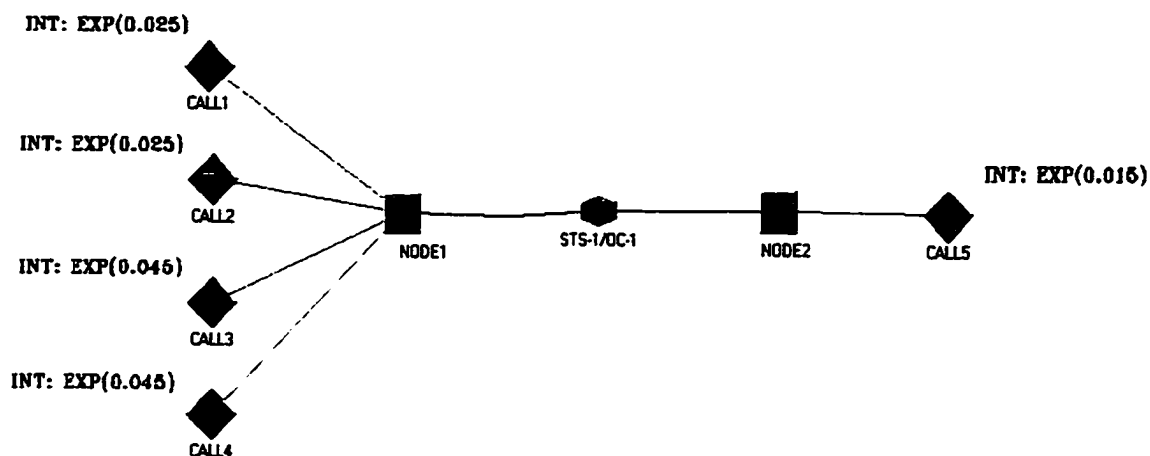
Using COMNET in a circuit switching environment, several reports can be generated :

- ◇ Blocked call Counts for Call Sources,
- ◇ Disconnected Call Counts for call sources,
- ◇ Preempted Call Counts for call Sources,
- ◇ Call Bandwidth Utilization for Call Sources,
- ◇ Call Counts for Nodes and Links,
- ◇ Call Bandwidth Utilization for Nodes and Links.

Because of the large size of this reports, as well as the large number of diagrams that we get when simulating the national network (or even a region), only a flavor will be given.

### 8.1.2.1. TWO NODE SIMULATION

#### Two NODE based netowrk



Links AT A=OC1 Rate; NODES Capacity 155520 kbps.

9720 call switching

**Figure 8.1: Two Node Simulation;**

The previous topology was simulated on the IBM ThinkPad 760E based on Pentium processor, with about 48MB of RAM and running windows 95. For the simulation runs, the execution time is 13 minutes (to get result relative to 360 seconds real time) and the saving time was 3 minutes. Using the same machine, the simulation time as well as the saving time can vary depending on the number of call sources and the interarrival time. Poisson distribution is the most used circuit switched simulation.

The simulation was made for both ATM switches and 5ESS switches, and the results is the same (considering that in both case that end-to-end connection established between the communicating parties will take 64Kbps channel).

While running the simulation, snapshots were taken, and revealed that after 68 seconds real time 100% of node capacity were used. Node1 got to the saturation first after 48 seconds real time, and then node 2. At the nodes saturation, the use of links started dropping down from 49% to 18% by the end of the simulation time.

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2Ntest1

NODES: CALL COUNTS

REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS

NODE NAME	CALLS ATTEMPTD	CALLS BLOCK AVAIL	CALLS BLOCK TRAFF	BLOCK PROB	CALLS CARRIED	CALLS DISCON- NECTED	CALLS PRE- EMPTED
NODE1	451085	0	277345	0.61	173740	0	0
NODE2	251572	0	140177	0.56	111395	0	0

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2Ntest1

NODES: CALL LEVEL

REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS

NODE NAME	% AVAIL	NODE FAILS	BANDWIDTH USED AVERAGE	(KBPS) STD DEV	MAXIMUM	NODE UTIL %
NODE1	100.00	0	154330	10981	155520	99.23
NODE2	100.00	0	154060	12031	155520	99.06

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2Ntest1

LINKS: CALL COUNTS

REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS

LINK NAME	CALLS ATTEMPTD	CALLS BLOCK AVAIL	CALLS BLOCK TRAFF	BLOCK PROB	CALLS CARRIED	CALLS DISCON- NECTED	CALLS PRE- EMPTED
LINK	7757	0	0	0.00	7757	0	0

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2Ntest1

LINKS: CALL LEVEL

REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS

LINK NAME	% AVAIL	LINK FAILS	BANDWIDTH USED AVERAGE	STD DEV	(KBPS) MAXIMUM	LINK UTIL %
LINK	100.00	0	7445	3733	25648	14.36

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2Ntest1

CALL SOURCES: BLOCKED CALL COUNTS

REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS

ORIGIN / CALL NAME: DESTINATION LIST	CALLS ATTEMPTD	CALLS RETRY	BLOCK PROB	HOPS AVG	MAX
NODE1 / call CALL1:					
NODE2	14517	0	0.794	1.0	1
NODE1	129795	0	0.530	0.0	0
SUBTOTAL	144312	0	0.557	0.0	1
NODE1 / call CALL2:					
NODE1	129814	0	0.528	0.0	0
NODE2	14317	0	0.789	1.0	1
SUBTOTAL	144131	0	0.554	0.0	1
NODE1 / call CALL3:					
NODE1	72109	0	0.768	0.0	0
NODE2	8028	0	0.945	1.0	1
SUBTOTAL	80137	0	0.785	0.0	1
NODE1 / call CALL4:					
NODE1	71939	0	0.768	0.0	0
NODE2	8036	0	0.941	1.0	1
SUBTOTAL	79975	0	0.785	0.0	1
NODE1 (TOTAL)	448555	0	0.637	0.0	1
NODE2 / call CALL5:					
NODE1	24027	0	0.965	1.0	1
NODE2	216482	0	0.565	0.0	0
SUBTOTAL	240509	0	0.605	0.0	1
NODE2 (TOTAL)	240509	0	0.605	0.0	1
** T O T A L S **	689064	0	0.626	0.0	1

2Ntest1

## CALL SOURCES: DISCONNECTED CALL COUNTS

REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS

ORIGIN / CALL NAME: DESTINATION LIST	PRI	CALLS ATTEMPTD	CALLS CARRIED	CALLS DISCON	CALLS REROUT
NODE1 / call CALL1:	1				
NODE2		14517	2984	0	0
NODE1		129795	61011	0	0
SUBTOTAL		144312	63995	0	0
NODE1 / call CALL2:	1				
NODE1		129814	61239	0	0
NODE2		14317	3019	0	0
SUBTOTAL		144131	64258	0	0
NODE1 / call CALL3:	1				
NODE1		72109	16753	0	0
NODE2		8028	443	0	0
SUBTOTAL		80137	17196	0	0
NODE1 / call CALL4:	1				
NODE1		71939	16685	0	0
NODE2		8036	471	0	0
SUBTOTAL		79975	17156	0	0
NODE1 (TOTAL)		448555	162605	0	0
NODE2 / call CALL5:	1				
NODE1		24027	840	0	0
NODE2		216482	94138	0	0
SUBTOTAL		240509	94978	0	0
NODE2 (TOTAL)		240509	94978	0	0
** T O T A L S **		689064	257583	0	0

2Ntest1

## CALL SOURCES: PREEMPTED CALL COUNTS

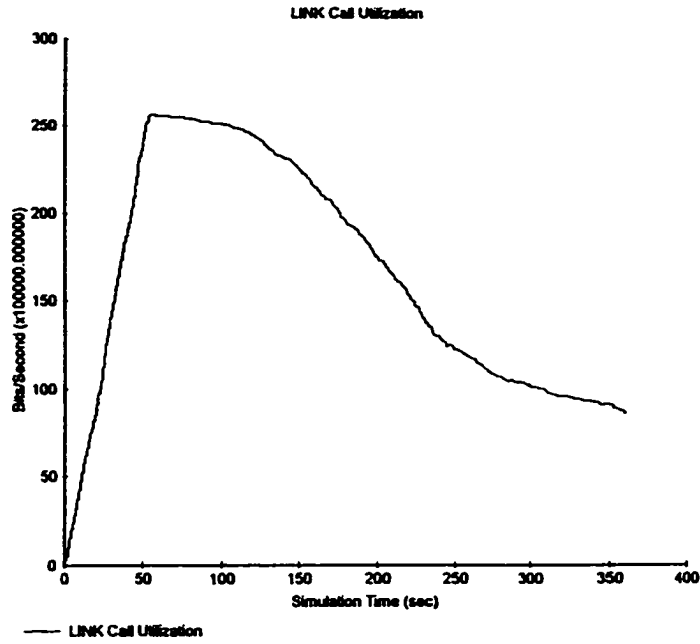
REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS

ORIGIN / CALL NAME: DESTINATION LIST	PRI	CALLS ATTEMPTED	CALLS CARRIED	CALLS PREEMPTED
NODE1 / call CALL1:	1			
NODE2		14517	2984	0
NODE1		129795	61011	0
SUBTOTAL		144312	63995	0
NODE1 / call CALL2:	1			
NODE1		129814	61239	0
NODE2		14317	3019	0
SUBTOTAL		144131	64258	0

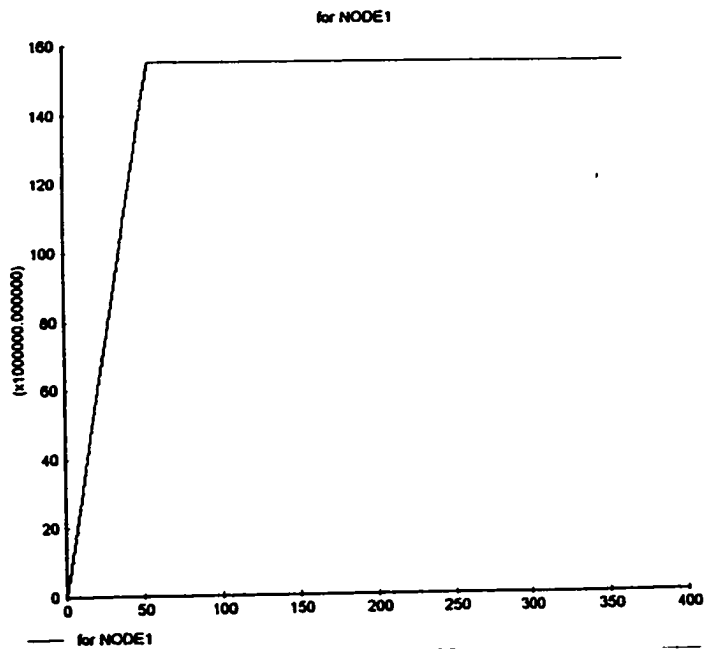
NODE1 / call CALL3:	1			
NODE1		72109	16753	0
NODE2		8028	443	0
SUBTOTAL		<hr/> 80137	<hr/> 17196	<hr/> 0
NODE1 / call CALL4:	1			
NODE1		71939	16685	0
NODE2		8036	471	0
SUBTOTAL		<hr/> 79975	<hr/> 17156	<hr/> 0
NODE1 (TOTAL)		<hr/> 448555	<hr/> 162605	<hr/> 0
NODE2 / call CALL5:	1			
NODE1		24027	840	0
NODE2		216482	94138	0
SUBTOTAL		<hr/> 240509	<hr/> 94978	<hr/> 0
NODE2 (TOTAL)		<hr/> 240509	<hr/> 94978	<hr/> 0
** T O T A L S **		<hr/> 689064	<hr/> 257583	<hr/> 0

**TABLE 8.1: Two node simulation generated statistics; Include link and nodes related statistics.**

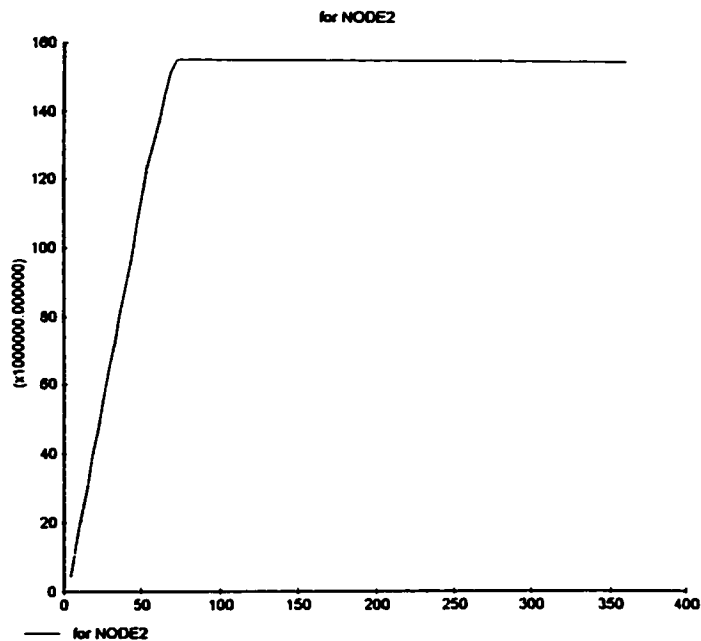
The simulation shows a blocking rate of 60% for the nodes, and 0% for the links. A large number of call sources are Blocked because of the small switch capacity which need to be increased to support the rate of call sources. This simulation is meant to give a show the type of problems that are needed to be solved while optimizing telecommunication networks The following figures show a Link call Utilization where the link is not optimally used (curve is going down after reaching a maximum, and the link is not well exploited), and we can also see the saturation of the nodes which reached its maximum after 50 MS. In large networks, The first problem can be solved by studying the topology, and the second one by applying more bandwidth.



**Figure 8.2: Link utilization in case of a two node simulation**



**Figure 8.3: Node 1 call Bandwidth in case of two node simulation**



**Figure 8.4: Node Call Bandwidth in case of a two node simulation**

**CONCLUSION:**

Many calls are blocked because of the saturation of the nodes, and the links are under used. Optimizing the network becomes a necessity before any further steadies. To cut it short, the results for the two node network using optimum values will not be given. But, it will preferable to take in consideration the load estimation of the network for the next 5 years, before providing the optimum values. After optimization, both the nodes and the links should get to relatively some maximum numbers that are in best case equal or less than the predefined capacity of the respective nodes and links. If the numbers keep going up and

down, then we can decide the nodes and links are underutilized and the topology of the corresponding network is reviewed (case of the Malaysian Celcom wireless network). Unfortunately, this is not the case of the Moroccan network where we have a complete mesh connection between major nodes, and a three connection on different subnetworks.

### **8.1.2.2. NATIONAL NETWORK**

The computer system on which the simulation is performed, limits the topological representation of the national network. It also influences the maximum topology and the duration for the execution of the simulation runs.

The simulations for the national network will be based on the topology of the Moroccan network. The chart of the national network in the previous chapter shows that there is a heavy load on the regions of CasaBlanca and Rabat where the population is very high comparing to other regions in Morocco. For this reason, it was decided to study the network as follows: 1) The region of CasaBlanca, 2) Both CasaBlanca and Rabat, and then 3) All the Moroccan Networks. This will simplify the task and offer better results as if we study all the regions one by one, and then combine all the national network. The results of network simulation for the region of Casablanca, the regions of Casablanca and Rabat will not be presented, only a flavor of the results of the global network will be given.

The call sources in the following simulations are based on the exponential distribution function which takes one positive, real number which is the mean, and returns a positive real number. The inter-arrival time is calculated by taking

in consideration the saturation of links that are directly connected to node to which the call source is connected. The destination is a weighted list destination. It is calculated by looking at the list of destination nodes, and the number of links connected to each of this nodes. A call source connected to a node X should have a probability to be send to node  $Y_i = \frac{\text{sum (links connected to } Y_i)}{\text{sum (links connected to all nodes with which the node x can communicate)}}$ . If a link is connected to two different nodes, then only the half of it will be taken in consideration.

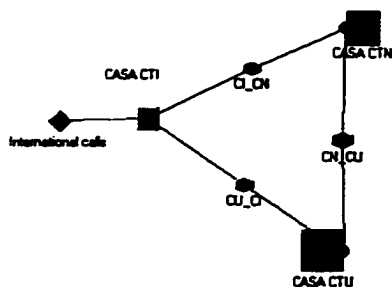
The statistics collected are related to the Blocked call Counts, Disconnected Call counts, and preempted call counts for the call source. For the nodes, the statistics that are collected are related with the Call counts, Call level, and Output and input Buffer. And for the links, Call Counts, and Call Level. The Call Bandwidth utilization is also collected for both Nodes and Links. The results are based on a backbone load equal to 25% taking in consideration that two customers use one even line, and that the probability of the used interarrival time as well as the weighted distribution can be true or false (probability =0.5).

### CASABLANCA REGION

The topology of Casablanca in the previous chapter as well as the interconnection to other regions is taken in consideration. The distribution is exponential distribution with weighted list destination (based on a probabilistic distributions).

**Figure 8.5: The region of CasaBlanca.** (a) subdivided into 2 subnetworks and connected to an international node, (b) the national transit subnetwork, (c) the urbane subnetwork.

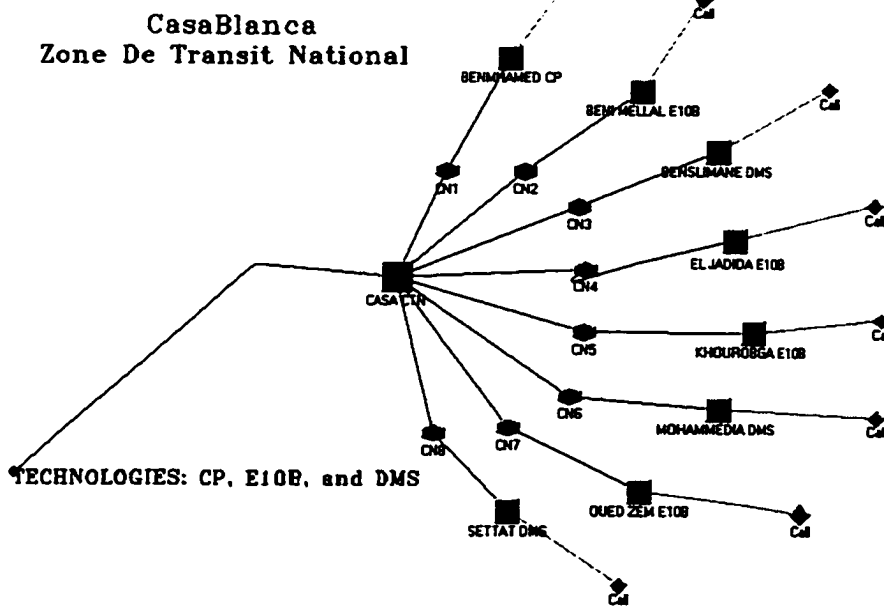
### CasaBlanca



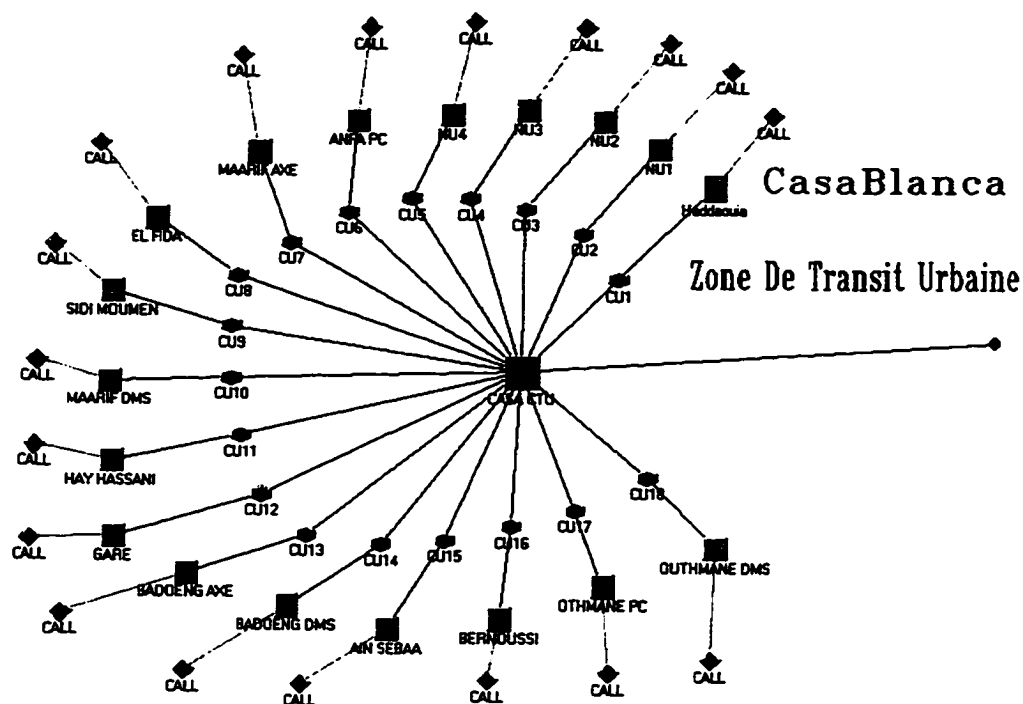
CTI: Zone De Transit International  
 CTU: Zone De Transit Urbaine  
 CTN: Zone De Transit National

RANDOM DESTINATION FOR ALL CALL SOURCES

8.5.a



8.5.b



8.5.c

A significant congestion at the link level shows in the current network of the region of Casablanca. A consistent mismatch between the size of the regional trunks in relation to the capacity of the regional nodes. In effect, if a majority of calls within any given region do generate enough traffic to approach the capacity of the switch, then there is no capacity in the regional trunks to carry this traffic to their destination. The switches have enough capacity to accomplish the switching of the communication channels, and in fact, they appear under utilized.

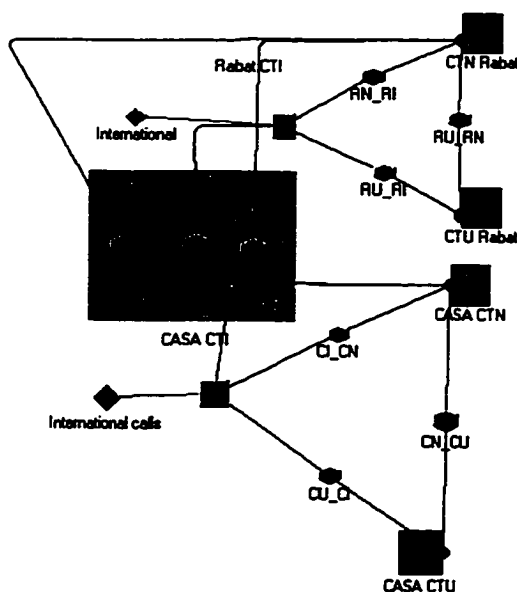
Some of the links are under utilized, and many get to the saturation points just a few seconds after running the simulations.

## BOTH CASABLANCA AND RABAT REGIONS INTERCONNECTED

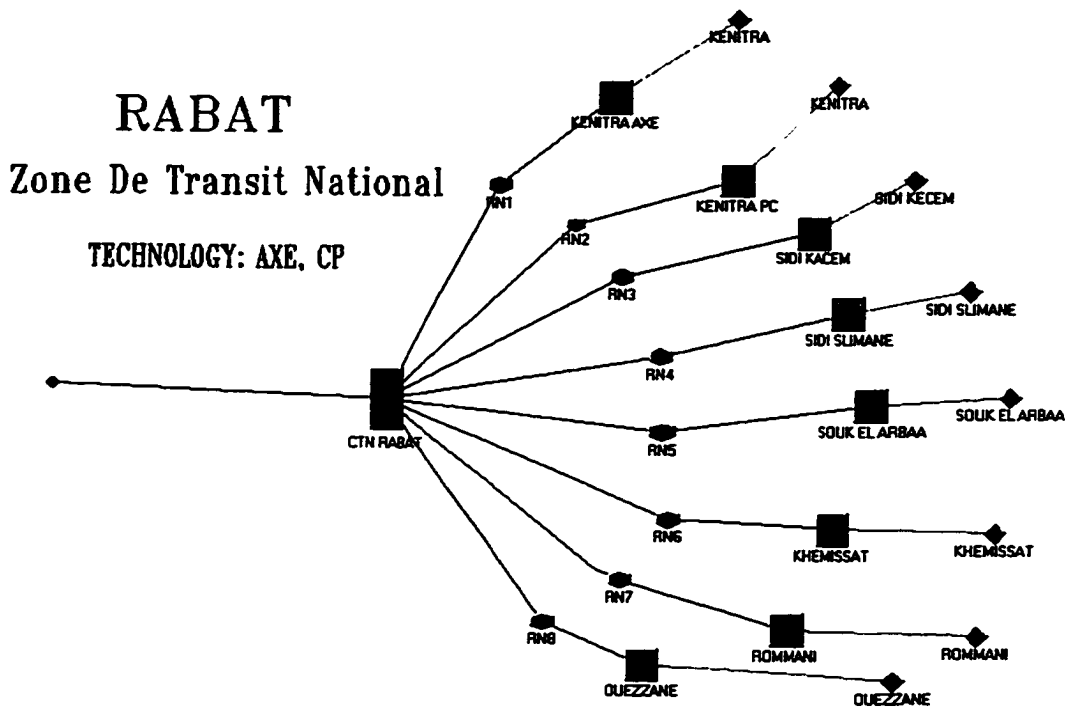
This previous network as well as detailed information about the network of the Rabat region are taken into considerations. Two new interconnected subnetworks as well as the Rabat international node are added. The Rabat\_CTN and Rabat\_CTU subnetworks internal topologies is shown in the following figures. The Topology the Casablanca\_CTN and Casablanca\_CTU topologies will practically remain the same.

**Figure 8.6: The region of CasaBlanca and Rabat Interconnected** (a) Both regions Interconnected(b) the national transit subnetwork for Rabat, (c ) the urbaine subnetwork for Rabat. The Other Subnetworks have been given in the previous simulation.

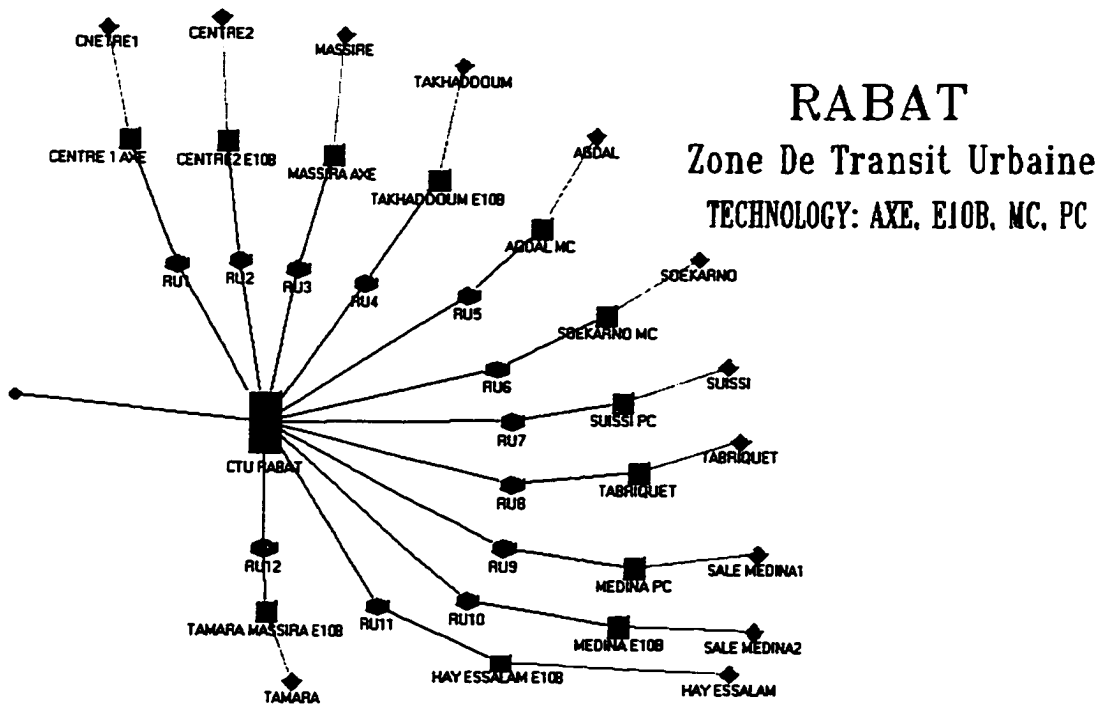
### CasaBlanca and Rabat connected



8.6.a



8.6.b



8.6.c

Same as for the region of Casablanca, simulating the network of the region of Rabat and Casablanca interconnected shows that some of the links are under utilized and many are congested, the nodes are under utilized. A large number of call get blocked because of the link congestion.

### **THE GLOBAL NETWORK**

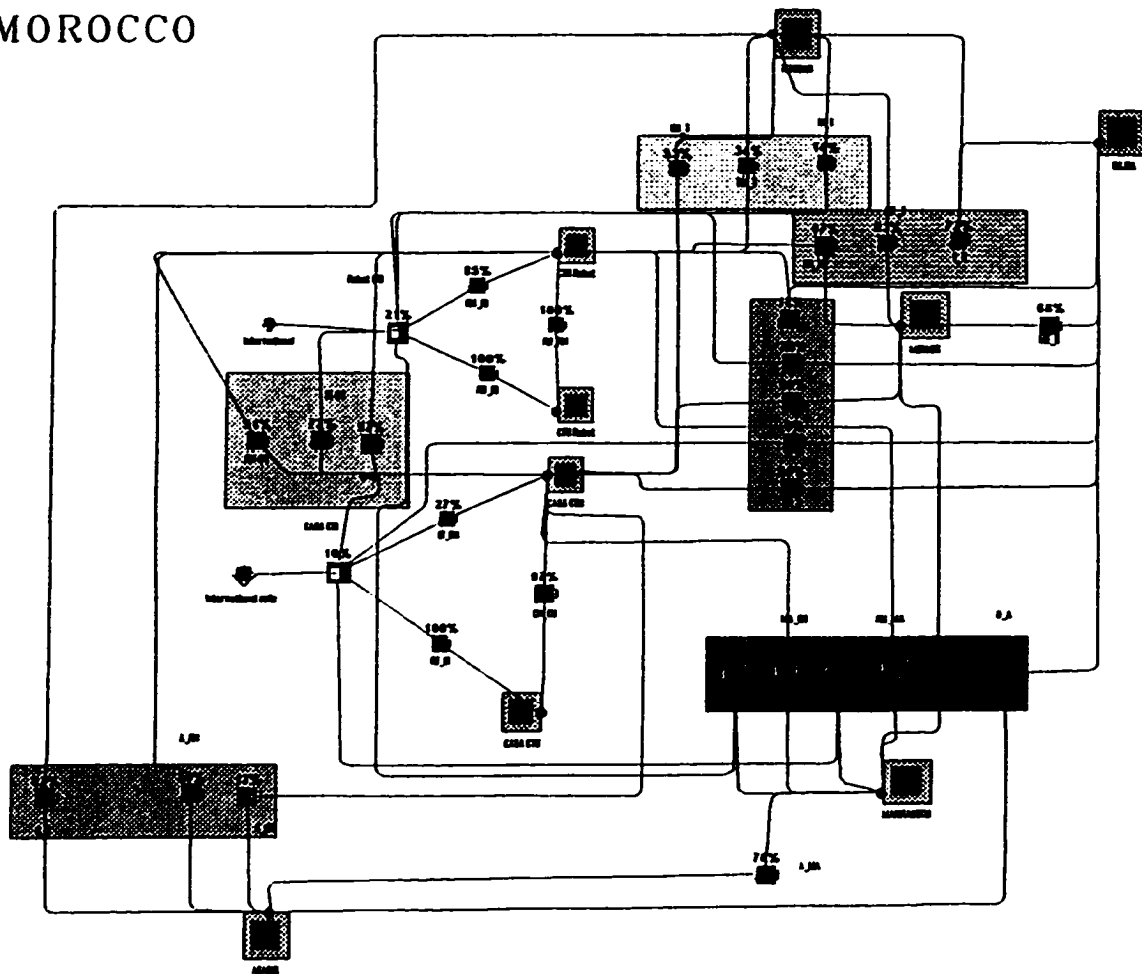
To implement the national network, we will use the topology of the regions of Casablanca and Rabat and add to it the remaining subnetworks needed to complete the global network.

The architecture of the remaining subnetworks looks like any other subnetwork architecture shown before, but of course by changing the link and node capacities, call sources Interarrival time, probabilistic rate of distribution, etc. to fit the specific region.

The following are the topologies of the global network and some national transit subnetworks used for the simulation. They are based on the current network architecture.



MOROCCO

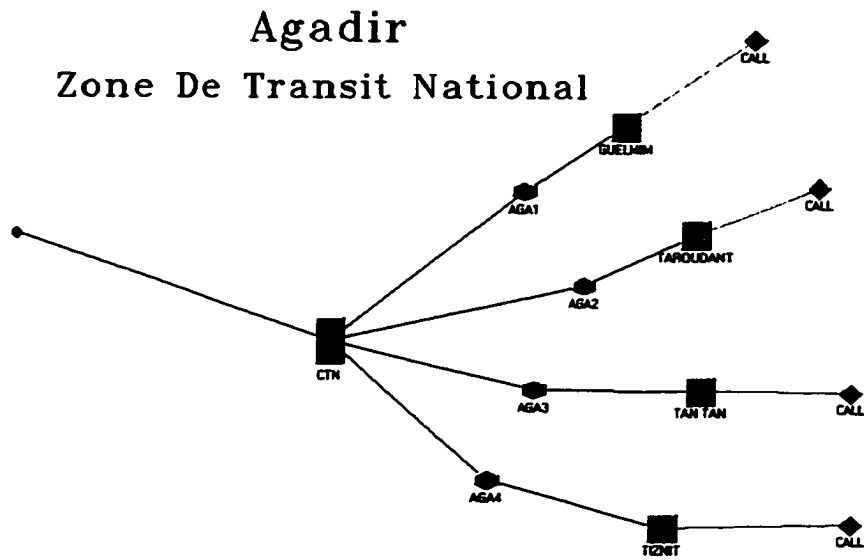


CTI: Zone De Transit International  
 CTU: Zone De Transit Urbaine  
 CTN: Zone De Transit National

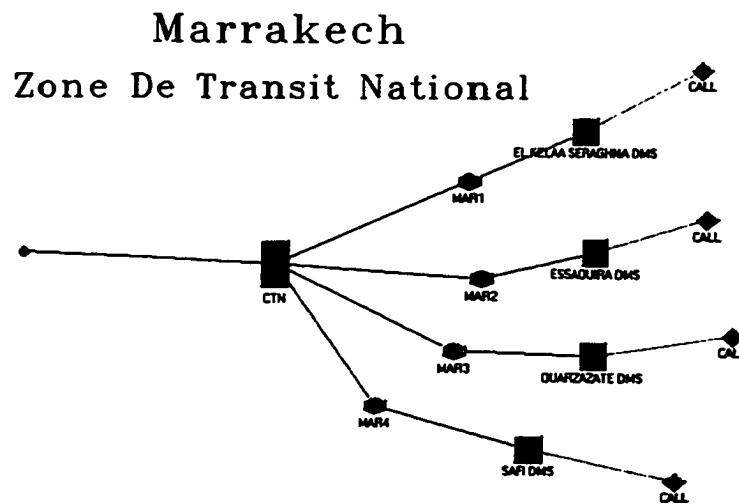
WEIGHTED DESTINATION FOR ALL CALL SOURCES

Figure 8.7: National Network.

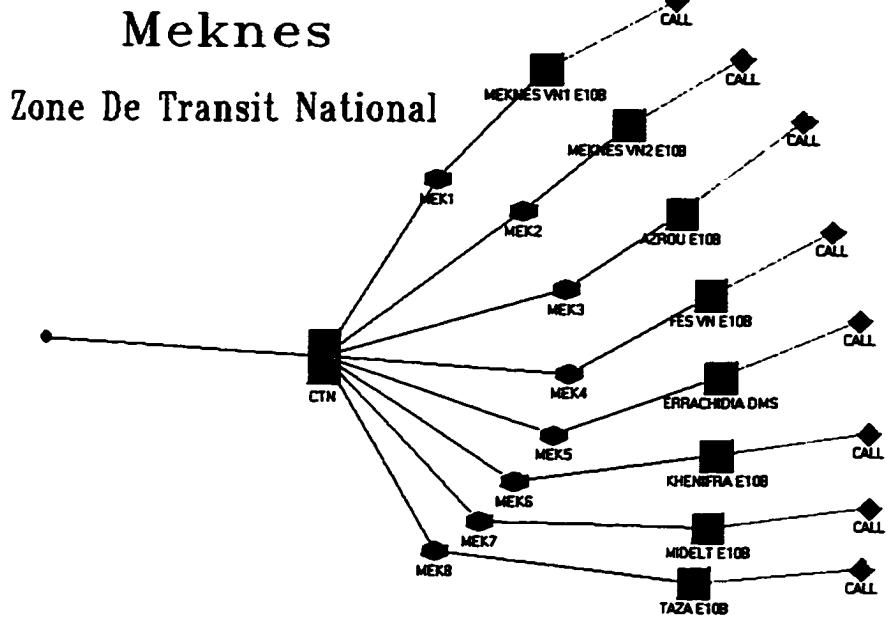
**Figure 8.8: Subnetworks in the National Network** (a) CTN Agadir (b) CTN Marrakesh (c) CTN Meknes, (d) CTN Oujda, (e) CTN Tetouan; The other subnetworks were previously defined.



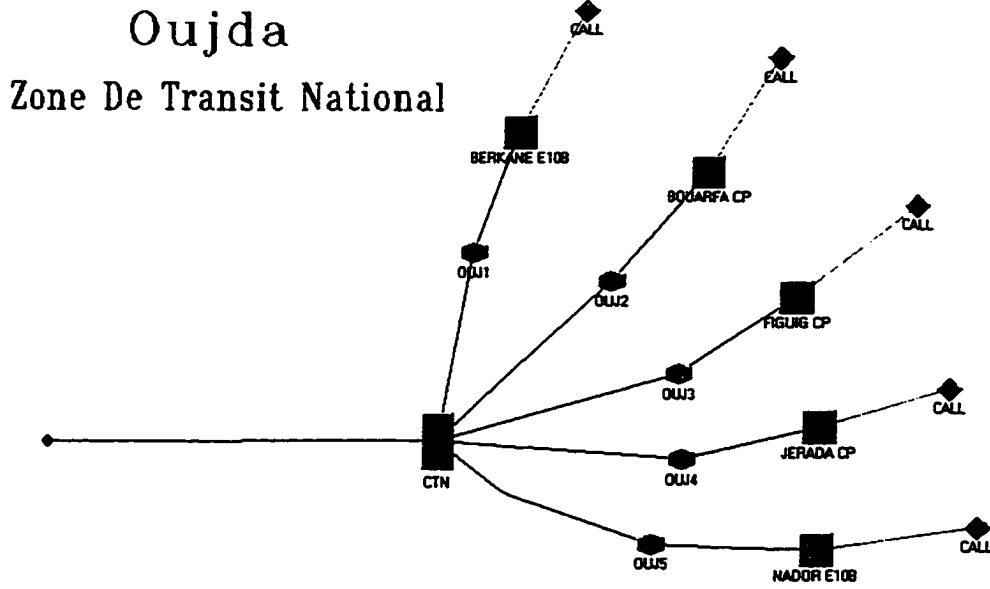
8.8.a



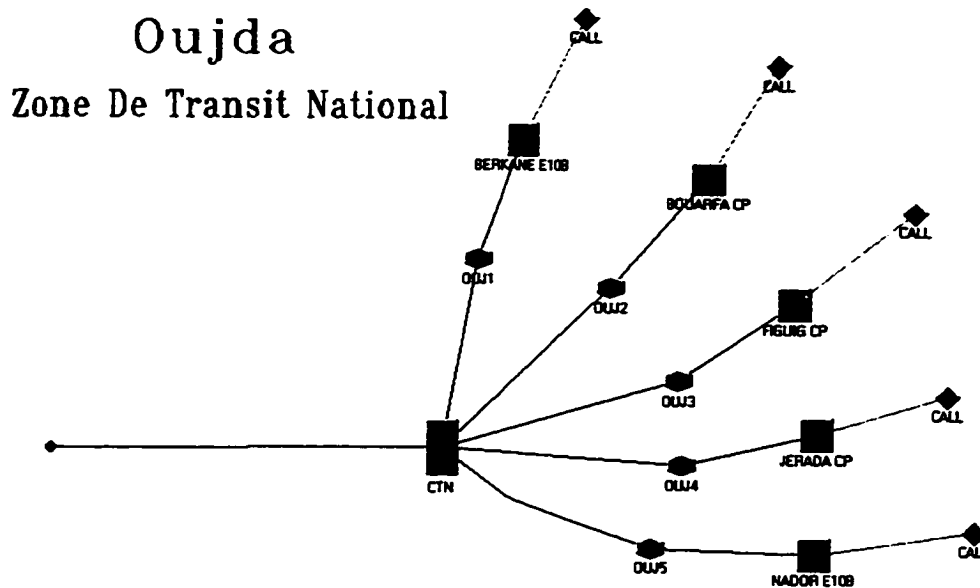
8.8.b



8.8.c



8.8.d



8.8.e

### RESULTS BEFORE OPTIMIZATION

A significant congestion at the link level shows in the current network of the regions of Casablanca, Rabat, and other regions. A consistent mismatch between the size of the regional trunks in relation to the capacity of the regional nodes. In effect, if a majority of calls within any given region do generate enough traffic to approach the capacity of the switch, then there is no capacity in the regional trunks to carry this traffic to their destination. The switches have enough capacity to accomplish the switching of the communication channels, and in fact, they appear under utilized.

Some of the links are under utilized, and many get to the saturation points just a few seconds after running the simulations.

**Table 8.1: Statistics before Optimization**

CACI COMNET III Release 1.3 (Build 1524) Sun Mar 15 21:42:53 1998  
Moroccan Network at the current stage  
CALL SOURCES: DISCONNECTED CALL COUNTS  
REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS

	CALLS ATTEMPTD	CALLS CARRIED	CALLS DISCON	CALLS REROUT	rate
CASA CTI (TOTAL)	3760	2271	0	0	60%
CASA CTU.Haddaoula (TOTAL)	831	343	0	0	41%
CASA CTU.NU1 (TOTAL)	5362	2257	0	0	42%
CASA CTU.NU2 (TOTAL)	3590	1435	0	0	40%
CASA CTU.NU4 (TOTAL)	1570	671	0	0	43%
CASA CTU.ANFA PC (TOTAL)	1851	795	0	0	43%
CASA CTU.MAARIF AXE (TOTAL)	2963	1207	0	0	41%
CASA CTU.EL FIDA (TOTAL)	2922	1260	0	0	43%
CASA CTU.SIDI MOUMEN (TOTAL)	1196	487	0	0	41%
CASA CTU.MAARIF DMS (TOTAL)	587	221	0	0	38%
CASA CTU.HAY HASSANI (TOTAL)	1442	616	0	0	43%
CASA CTU.GARE (TOTAL)	3001	1231	0	0	41%
CASA CTU.BADOENG AXE (TOTAL)	6889	2873	0	0	42%
CASA CTU.AIN SEBAA (TOTAL)	2126	880	0	0	41%
CASA CTU.BERNOUSSI (TOTAL)	907	365	0	0	40%
CASA CTU.OTHMANE PC (TOTAL)	1563	640	0	0	41%
CASA CTU.OUTHMANE DMS (TOTAL)	1208	500	0	0	41%
CASA CTU.BADOENG DMS (TOTAL)	580	235	0	0	41%
CASA CTN.BENMHAMED CP (TOTAL)	458	114	0	0	25%
CASA CTN.BENI MELLAL E10B (TOTAL)	2366	1409	0	0	60%
CASA CTN.BENSLIMANE DMS (TOTAL)	1214	612	0	0	50%
CASA CTN.EL JADIDA E10B (TOTAL)	4249	2544	0	0	60%
CASA CTN.KHOUROBGA E10B (TOTAL)	2436	1432	0	0	59%
CASA CTN.MOHAMMEDIA DMS (TOTAL)	2932	1792	0	0	61%
CASA CTN.OUED ZEM E10B (TOTAL)	1871	1095	0	0	59%
CASA CTN.SETTAT DMS (TOTAL)	4098	2475	0	0	60%
Rabat CTI (TOTAL)	3564	2136	0	0	60%
CTU Rabat.CENTRE2 E10B (TOTAL)	2936	1762	0	0	60%
CTU Rabat.CENTRE 1 AXE (TOTAL)	2912	1770	0	0	61%
CTU Rabat.MASSIRA AXE (TOTAL)	2088	1263	0	0	60%
CTU Rabat.TAKHADDOUN E10B (TOTAL)	2414	1475	0	0	61%
CTU Rabat.AGDAL MC (TOTAL)	4601	2676	0	0	58%
CTU Rabat.SOEKARNO MC (TOTAL)	4169	2527	0	0	61%
CTU Rabat.SUISSI PC (TOTAL)	1787	707	0	0	40%
CTU Rabat.TABRIQUET (TOTAL)	1695	947	0	0	56%
CTU Rabat.MEDINA PC (TOTAL)	888	362	0	0	41%
CTU Rabat.MEDINA E10B (TOTAL)	618	228	0	0	37%
CTU Rabat.HAY ESSALAM E10B (TOTAL)	1801	1103	0	0	61%
CTU Rabat.TAMARA MASSIRA E10 (TOTAL)	521	201	0	0	39%
CTN Rabat.KENITRA AXE (TOTAL)	906	485	0	0	54%

CTN Rabat.KENITRA PC (TOTAL)	2793	2793	0	0	100%
CTN Rabat.SIDI KACEM (TOTAL)	19	12	0	0	63%
CTN Rabat.SIDI SLIMANE (TOTAL)	2077	1233	0	0	59%
CTN Rabat.SOUK EL ARBAA (TOTAL)	2032	1212	0	0	60%
CTN Rabat.KHEMISSAT (TOTAL)	2410	1425	0	0	59%
CTN Rabat.ROMMANI (TOTAL)	384	97	0	0	25%
CTN Rabat.OUEZZANE (TOTAL)	2160	1251	0	0	58%
MEKNES.MEKNES VN1 E10B (TOTAL)	6653	4369	0	0	66%
MEKNES.MEKNES VN2 E10B (TOTAL)	4659	3054	0	0	66%
MEKNES.AZROU E10B (TOTAL)	1796	1164	0	0	65%
MEKNES.ERRACHIDIA DMS (TOTAL)	3663	2388	0	0	65%
MEKNES.KHENIFRA E10B (TOTAL)	1890	1270	0	0	67%
MEKNES.MIDELT E10B (TOTAL)	1176	772	0	0	66%
MEKNES.TAZA E10B (TOTAL)	1546	1025	0	0	66%
MEKNES.FES VN E10B (TOTAL)	3502	2288	0	0	65%
OIJDA.BERKANE E10B (TOTAL)	2661	1851	0	0	70%
OIJDA.BOUARFA CP (TOTAL)	470	320	0	0	68%
OIJDA.FIGUIG CP (TOTAL)	234	119	0	0	51%
OIJDA.NADOR E10B (TOTAL)	2729	1868	0	0	68%
OIJDA.JERADA CP (TOTAL)	577	377	0	0	65%
MARRAKECH.EL KELAA SERAGHNA (TOTAL)	1093	791	0	0	72%
MARRAKECH.OUARZAZATE DMS (TOTAL)	1118	846	0	0	76%
MARRAKECH.SAFI DMS (TOTAL)	2394	1753	0	0	73%
MARRAKECH.ESSAOUIRA DMS (TOTAL)	2157	1622	0	0	75%
AGADIR.GUELMIM (TOTAL)	1576	1190	0	0	76%
AGADIR.TAN TAN (TOTAL)	873	657	0	0	75%
AGADIR.TIZNIT (TOTAL)	1435	1069	0	0	74%
AGADIR.TAROUDANT (TOTAL)	2190	1615	0	0	74%
TETOUAN.TETOUAN CP (TOTAL)	3522	2529	0	0	72%
TETOUAN.AL HOCEIMA E10B (TOTAL)	1779	1313	0	0	74%
TETOUAN.LARACHE E10B (TOTAL)	1783	1233	0	0	69%
TETOUAN.TANGER E10 (TOTAL)	3979	2804	0	0	70%
<b>** TOTALS **</b>	162208	94532	0	0	58%

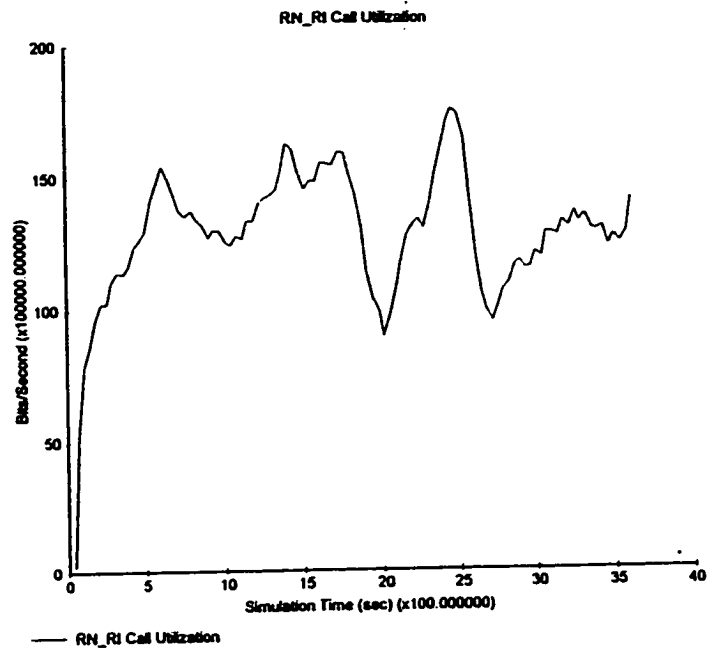
The previous table shows a high blocking probability which is due to the links capacity.

Some graphs related to the links Utilization as well as the bandwidth for different nodes were generated, and only a few of them are shown here.

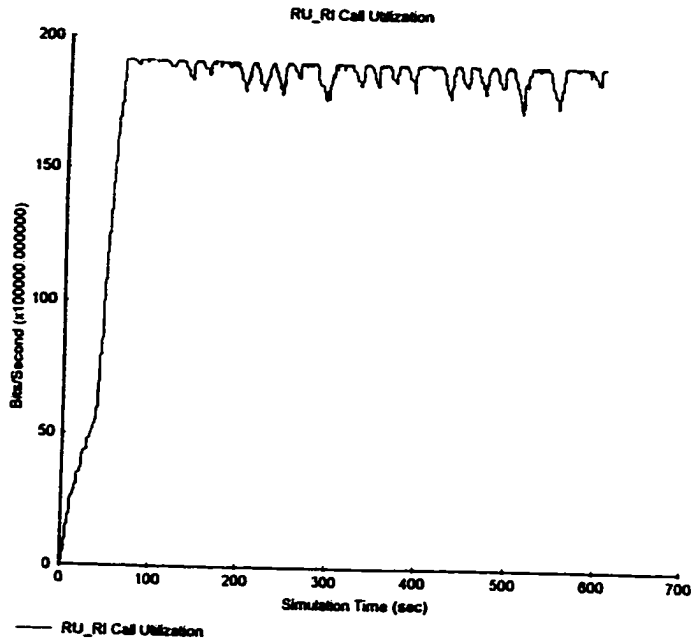
Link Utilization shown in the following graphs demonstrate some of the problems that the actual National network has. Link RN\_RI is under utilized, Link RU\_RI is saturated and many calls are blocked because of that.

The node Rabat CTI reached a maximum point which is less than the actual capacity, etc.

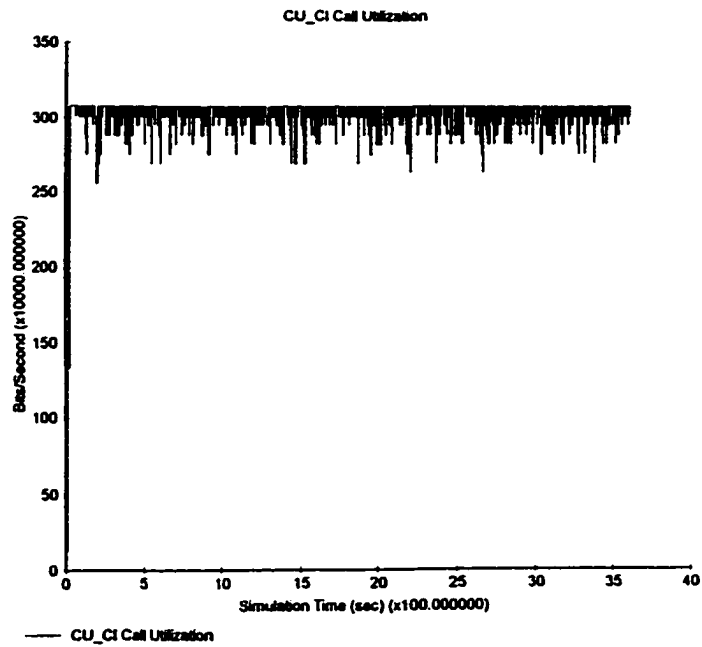
**Figure 8.9: Examples of Link Call Utilization;**



**8.9.a**

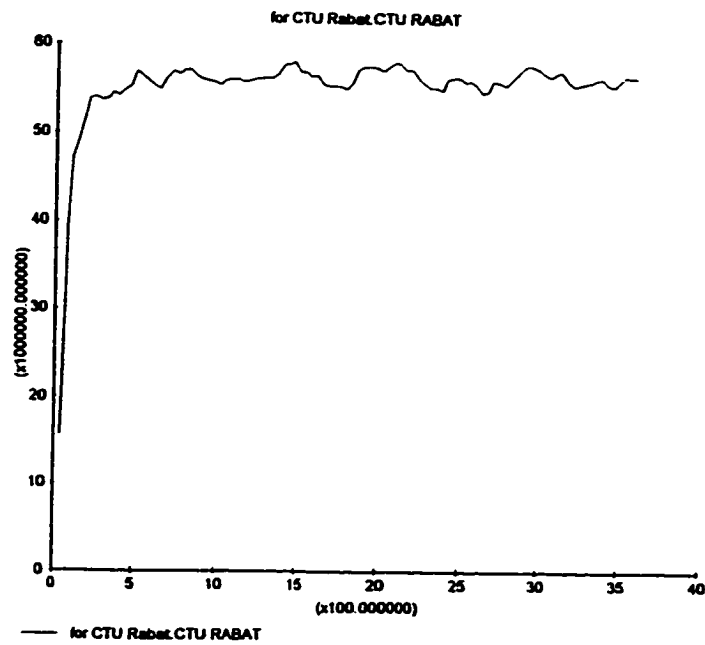


8.9.b

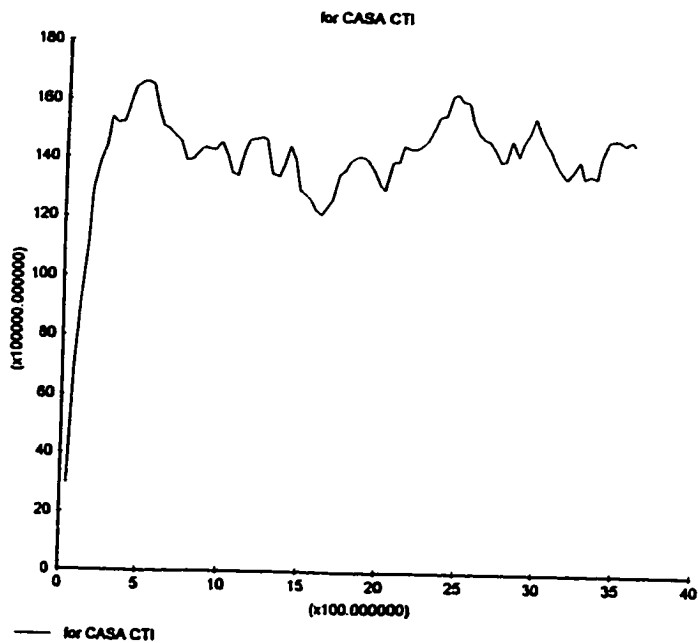


8.9.c

**Figure 8.10: Examples of nodes Bandwidth Utilization**



**8.10.a**



**8.10.b**

## OPTIMUM SOLUTION WITH INTRODUCTION OF A NEW NETWORK

### BASED ON FIBER DIGITAL TRANSMISSION

In this case the same topology was studied for an optimum resolution. Fiber is being added, and the call inter-arrival time modified to support the load of the network 5 years from last now. The use of ATM nodes or existent nodes didn't show much change on the results.

Table 8.2 shows the statistics of international calls from CASA CTI to other national nodes. The average Blocking Probability improved to be 0.3%, which is a very good rate.

**TABLE 8.2: International calls Statistics**

Moroccan Network after Optimization  
REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS  
Node Utilization by Specific Call Source

ORIGIN / CALL NAME:	CALLS ATTEMPTD	CALLS CARRIED	CALLS RETRY	BLOCK PROB	HOPS AVG		
CASA CTI / call International calls:						MAX	
CASA CTN.CASA CTN	152	147	0	3.3%	1	1	
CASA CTU.CASA CTU	152	152	0	0.0%	1	1	
CTN Rabat.CTN RABAT	145	145	0	0.0%	1	1	
CASA CTI	153	153	0	0.0%	0	0	
Rabat CTI	146	146	0	0.0%	2	2	
CASA CTU.Haddaouia	127	127	0	0.0%	2	2	
CASA CTU.NU1	144	144	0	0.0%	2	2	
CASA CTU.NU2	128	128	0	0.0%	2	2	
CASA CTU.NU3	139	139	0	0.0%	2	2	
CASA CTU.NU4	155	155	0	0.0%	2	2	
CASA CTU.ANFA PC	146	146	0	0.0%	2	2	
CASA CTU.MAARIF AXE	127	127	0	0.0%	2	2	
CASA CTU.EL FIDA	146	146	0	0.0%	2	2	
CASA CTU.SIDI MOUMEN	167	167	0	0.0%	2	2	
CASA CTU.MAARIF DMS	148	148	0	0.0%	2	2	
CASA CTU.HAY HASSANI	113	113	0	0.0%	2	2	
CASA CTU.GARE	135	135	0	0.0%	2	2	
CASA CTU.BADOENG AXE	139	139	0	0.0%	2	2	
CASA CTU.AIN SEBAA	133	133	0	0.0%	2	2	
CASA CTU.BERNOUSSI	127	127	0	0.0%	2	2	
CASA CTU.OTHMANE PC	150	150	0	0.0%	2	2	
CASA CTU.OUTHMANE DMS	167	167	0	0.0%	2	2	
CASA CTU.BADOENG DMS	145	145	0	0.0%	2	2	
CASA CTN.BENMHAMED CP	165	163	0	1.2%	2	2	
CASA CTN.BENI MELLAL E10B	163	161	0	1.2%	2	2	
CASA CTN.BENSLIMANE DMS	153	153	0	0.0%	2	2	
CASA CTN.EL JADIDA E10B	162	160	0	1.2%	2	2	

CASA CTN.KHOUROBGA E10B	143	142	0	0.7%	2	2
CASA CTN.MOHAMMEDIA DMS	130	127	0	2.3%	2	2
CASA CTN.OUED ZEM E10B	137	134	0	2.2%	2	2
CASA CTN.SETTAT DMS	157	154	0	1.9%	2	2
CTU Rabat.CTU RABAT	155	155	0	0.0%	2	2
CTU Rabat.CENTRE2 E10B	122	122	0	0.0%	3	3
CTU Rabat.CENTRE 1 AXE	138	138	0	0.0%	3	3
CTU Rabat.MASSIRA AXE	157	157	0	0.0%	3	3
CTU Rabat.TAKHADDOUM E10B	153	153	0	0.0%	3	3
CTU Rabat.AGDAL MC	152	152	0	0.0%	3	3
CTU Rabat.SOEKARNO MC	140	140	0	0.0%	3	3
CTU Rabat.SUISSI PC	162	162	0	0.0%	3	3
CTU Rabat.TABRIQUET	148	148	0	0.0%	3	3
CTU Rabat.MEDINA PC	141	141	0	0.0%	3	3
CTU Rabat.MEDINA E10B	139	139	0	0.0%	3	3
CTU Rabat.HAY ESSALAM E10B	158	158	0	0.0%	3	3
CTU Rabat.TAMARA MASSIRA E10B	168	168	0	0.0%	3	3
CTN Rabat.KENITRA AXE	154	154	0	0.0%	2	2
CTN Rabat.KENITRA PC	148	148	0	0.0%	2	2
CTN Rabat.SIDI KACEM	163	163	0	0.0%	2	2
CTN Rabat.SIDI SLIMANE	152	152	0	0.0%	2	2
CTN Rabat.SOUK EL ARBAA	158	158	0	0.0%	2	2
CTN Rabat.KHEMISSAT	170	170	0	0.0%	2	2
CTN Rabat.ROMMANI	157	157	0	0.0%	2	2
CTN Rabat.OUEZZANE	156	156	0	0.0%	2	2
<b>SUBTOTAL</b>	<b>7685</b>	<b>7664</b>	<b>0</b>	<b>0.3%</b>	<b>2.1</b>	<b>3</b>
<b>CASA CTI (TOTAL)</b>	<b>7685</b>	<b>7664</b>	<b>0</b>	<b>0.3%</b>	<b>2.1</b>	<b>3</b>

The following table shows the statistics of calls made to all major nodes in the national network. It shows the improvement in call blocking rate.

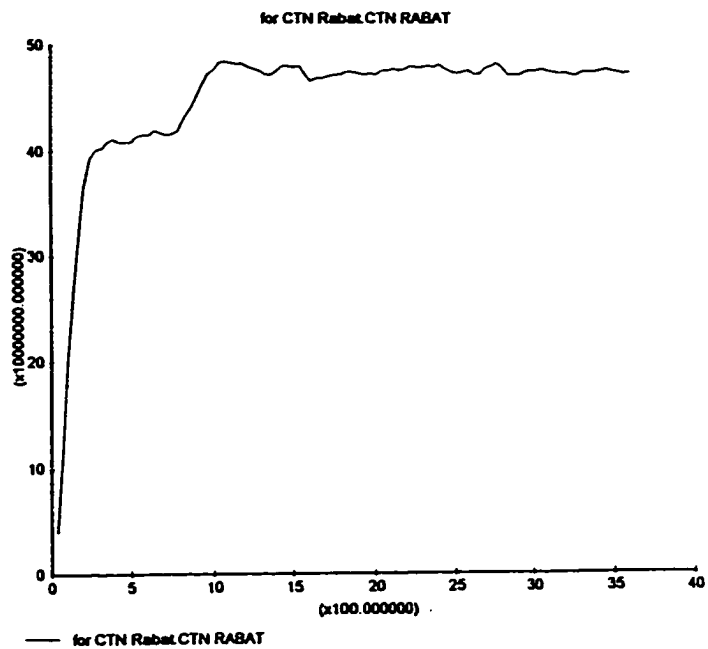
**Table 8.3: Major nodes call utilization (after Optimization)**

Moroccan Network after Optimization						
REPLICATION 1 FROM 0.0 TO 3600.0 SECONDS						
Node utilization						
ORIGIN / CALL NAME:	CALLS	CALLS	CALLS	BLOCK	HOPS	
ATTEMPTD	ATTEMPTD	CARRIED	RETRY	PROB	AVG	MAX
CASA CTU.Haddaouia (TOTAL)	1804	1802	0	0.1%	3.2	5
CASA CTU.NU2 (TOTAL)	7175	7163	0	0.2%	3.2	5
CASA CTU.NU3 (TOTAL)	4004	3994	0	0.2%	3.2	5
CASA CTU.NU4 (TOTAL)	2965	2962	0	0.1%	3.2	5
CASA CTU.MAARIF AXE (TOTAL)	6069	6054	0	0.2%	3.2	5
CASA CTU.EL FIDA (TOTAL)	6158	6143	0	0.2%	3.2	5
CASA CTU.GARE (TOTAL)	5965	5955	0	0.2%	3.2	5
CASA CTU.BADOENG AXE (TOTAL)	13745	13719	0	0.2%	3.2	5
CASA CTU.AIN SEBAA (TOTAL)	4260	4247	0	0.3%	3.2	5
CASA CTU.BERNOUSSI (TOTAL)	1804	1803	0	0.1%	3.2	5
CASA CTU.OTHMANE PC (TOTAL)	3098	3091	0	0.2%	3.2	5
CASA CTU.OUTHMANE DMS (TOTAL)	2426	2422	0	0.2%	3.2	5
CASA CTU.BADOENG DMS (TOTAL)	1249	1248	0	0.1%	3.1	5
CASA CTN.BENMHAMED CP (TOTAL)	770	762	0	1.0%	2.9	4
CASA CTN.BENI MELLAL E10B (TOTAL)	4751	4696	0	1.2%	2.9	4
CASA CTN.EL JADIDA E10B (TOTAL)	8421	8325	0	1.1%	2.9	4
CASA CTN.KHOUROBGA E10B (TOTAL)	4878	4815	0	1.3%	2.9	4
CASA CTN.MOHAMMEDIA DMS (TOTAL)	5813	5734	0	1.4%	2.9	4
CASA CTN.OUED ZEM E10B (TOTAL)	3930	3888	0	1.1%	2.9	4
CASA CTN.SETTAT DMS (TOTAL)	8497	8397	0	1.2%	2.9	4
Rabat CTI (TOTAL)	7199	7184	0	0.2%	2.3	4

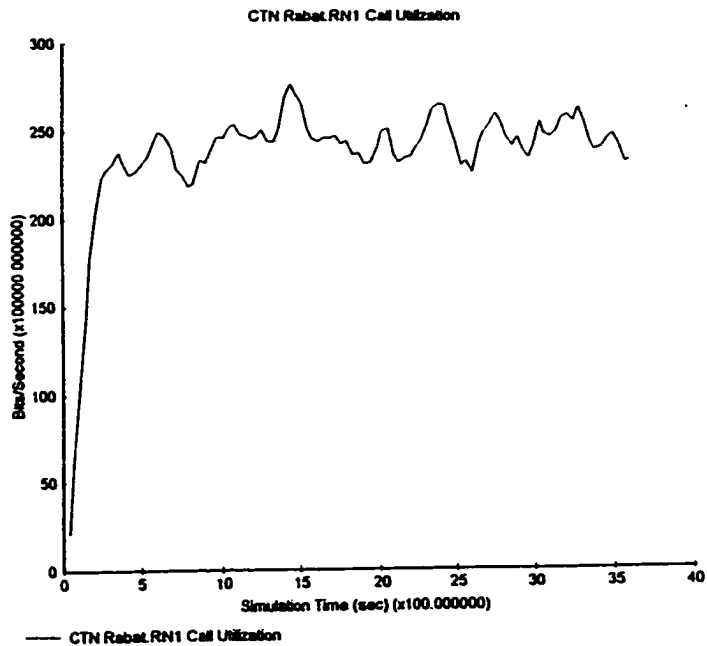
CTU Rabat.CENTRE2 E10B (TOTAL)	6066	6058	0	0.1%	3.5	6
CTU Rabat.CENTRE 1 AXE (TOTAL)	6029	6011	0	0.3%	3.5	6
CTU Rabat.MASSIRA AXE (TOTAL)	4254	4248	0	0.1%	3.5	6
CTU Rabat.TAKHADDOUN E10B (TOTAL)	4616	4609	0	0.2%	3.5	6
CTU Rabat.SOEKARNO MC (TOTAL)	8487	8469	0	0.2%	3.5	6
CTU Rabat.SUISSI PC (TOTAL)	3645	3636	0	0.2%	3.5	6
CTU Rabat.TABRIQUET (TOTAL)	3321	3308	0	0.4%	3.5	6
CTU Rabat.MEDINA PC (TOTAL)	1733	1730	0	0.2%	3.5	6
CTU Rabat.MEDINA E10B (TOTAL)	1152	1148	0	0.3%	3.6	6
CTU Rabat.HAY ESSALAM E10B (TOTAL)	3677	3672	0	0.1%	3.6	6
CTU Rabat.TAMARA MASSIRA E10B (TOTAL)	976	975	0	0.1%	3.5	6
CTN Rabat.KENITRA AXE (TOTAL)	1839	1835	0	0.2%	3	4
CTN Rabat.KENITRA PC (TOTAL)	5529	5529	0	0.0%	1	1
CTN Rabat.SIDI KACEM (TOTAL)	295	295	0	0.0%	2.9	4
CTN Rabat.SOUK EL ARBAA (TOTAL)	4020	4012	0	0.2%	3	4
CTN Rabat.KHEMISSAT (TOTAL)	4751	4746	0	0.1%	3	4
CTN Rabat.ROMMANI (TOTAL)	838	837	0	0.1%	3	4
CTN Rabat.OUEZZANE (TOTAL)	4209	4199	0	0.2%	3	4
MEKNES.MEKNES VN1 E10B (TOTAL)	13235	13205	0	0.2%	3.3	6
MEKNES.MEKNES VN2 E10B (TOTAL)	9526	9512	0	0.1%	3.3	6
MEKNES.AZROU E10B (TOTAL)	3640	3632	0	0.2%	3.3	6
MEKNES.ERRACHIDIA DMS (TOTAL)	7279	7262	0	0.2%	3.3	6
MEKNES.KHENIFRA E10B (TOTAL)	3889	3886	0	0.1%	3.3	6
MEKNES.MIDELT E10B (TOTAL)	2477	2468	0	0.4%	3.3	6
MEKNES.TAZA E10B (TOTAL)	2962	2953	0	0.3%	3.3	6
MEKNES.FES VN E10B (TOTAL)	7195	7180	0	0.2%	3.3	6
OUJDA.BERKANE E10B (TOTAL)	2686	2677	0	0.3%	3.3	4
OUJDA.BOUARFA CP (TOTAL)	439	438	0	0.2%	3.3	4
OUJDA.FIGUIG CP (TOTAL)	496	494	0	0.4%	3.2	4
OUJDA.NADOR E10B (TOTAL)	10692	10670	0	0.2%	3.2	4
OUJDA.JERADA CP (TOTAL)	2450	2447	0	0.1%	3.2	4
MARRAKECH.OUARZAZATE DMS (TOTAL)	2163	2152	0	0.5%	3.3	4
MARRAKECH.SAFI DMS (TOTAL)	4835	4827	0	0.2%	3.3	4
MARRAKECH.ESSAOUIRA DMS (TOTAL)	4162	4156	0	0.1%	3.3	4
AGADIR.GUELMIM (TOTAL)	3152	3149	0	0.1%	3.3	6
AGADIR.TAN TAN (TOTAL)	1735	1731	0	0.2%	3.3	6
AGADIR.TIZNIT (TOTAL)	2877	2874	0	0.1%	3.3	6
AGADIR.TAROUDANT (TOTAL)	4329	4323	0	0.1%	3.3	6
TETOUAN.TETOUAN CP (TOTAL)	7195	7187	0	0.1%	3.2	5
TETOUAN.AL HOCEIMA E10B (TOTAL)	3641	3641	0	0.0%	3.3	5
TETOUAN.LARACHE E10B (TOTAL)	3639	3630	0	0.2%	3.2	5
TETOUAN.TANGER E10 (TOTAL)	7927	7917	0	0.1%	3.2	5
<b>** TOTALS **</b>	<b>328980</b>	<b>327946</b>	<b>0</b>	<b>0.3%</b>	<b>3.1</b>	<b>6</b>

The nodes and links get to a maximum that is less or equal to their respective capacity, and vary around that maximum offering a lot of savings.

**Figure 8.11: a flavor of the node and link utilization in the optimized National network.**



**8.11.a**



**8.11.b**

## CONCLUSION

The National network taken in consideration has got nodes with capacity exceeding the actual need, but the links capacity is very low which cause the blocking of a large number of calls. The simulation was made by using an average call duration of 3 minutes, and the very slow increase of the use of the Internet Network (which initial introduction date to 1 year back ) was not taken in consideration. In more advanced countries where a large number of users can afford a computer, and then the possibility to sign online, an average of holding a connection can go over 1 hour which will influence the results of the simulations and then the requirements for the given network.

Fiber is a on its way to the national network infrastructure, and will provide higher capacities and digital transmission. This was the aim of the second part of simulating the national network.

## 8.2

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### **PACKET SWITCHED TRAFFIC OVER ATM BASED TRANSPORT NETWORK**

In this part we will first study the effect of processing in an ATM environment and its effect on messages (which are broken into cells for transmission) delay.

To speed up call processing we will need to reduce the time to process signaling messages. Then the idea of an egress interface module for trunk processing which will take the NNI signaling. (see chapter 6).

## 8.2.1.

---

### **Packet Switching Environment**

The simplest packet switching network is based on two nodes communicating messages between them (the simplest should be one node and one message just like the case of one client and one server). These messages are broken into packets (or cells in case of ATM nodes) and sent through a transmission link. In the following, the nodes are ATM nodes, the transmission is optical using SONET, and the protocol is based on ATM stacks. In this case, the messages are broken into cells, and then sent to an output buffer for transmission using a NNI. At the receiving side, the messages are reassembled. That was the simplest description of an ATM transport network (more details are provided in previous chapters). The ATM node has an internal processor node that executes software applications and processes packets. It also has a disk storage device to simulate file reads and writes. At any instant in time the node processor is either Idle or Busy. The processor is counted as busy when executing process commands, read and write commands are accessing files, or cells are being created by message, session or response generators. When a cell is created, the processor is made busy for the packetizing delay followed by the packet switching time. In addition, the processor is busy for the switching time when cells are routed across the node. The node is connected to the links via an interface port. Each port has an input buffer, and output buffer. If there is an insufficient buffer space, a packet may be blocked. The input buffer on the ATM node will first determine if there is a room

for a cell by checking that the specific input buffers has buffer space available to fit the incoming cell. If not, the cell can be preempted (preemption policy) or tresholded (treshold policy used in ATM, and frame relay).

The ATM cells are sent through an optical link using SONET frames. This Frames can be delayed (the delay is minimum). The transmission delay is the time between when the frame is created at the input to the link and when the frame is delivered at the end of the link. It includes transmission, contention-resolution (for LAN), and propagation time.

### **8.2.2.**

---

## **Simulating the Packet Switching Networks in an ATM Environment**

The message delay is the time between creating the first packet of the message on the originating and the time that the originating node is notified that the message has been assembled by the destination.

The message delivery delay is the time between creating the first packet of the message on the originating node and the time of receiving the last packet on the destination node. The message may still be worked on by the source after the message is delivered because of transmissions, waiting for acknowledgments, or for modeling the close-sequence of the connection, but these values are not factored into the message delivered delay.

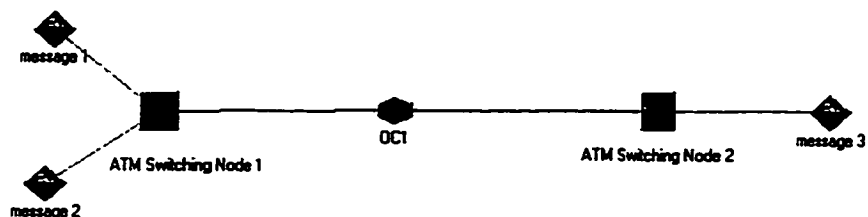
Packet delay is the time between creating a packet on the originating node and the time of receiving the packet at the destination node.

### **8.2.2.1. TWO NODE SIMULATION**

Figure 8.12 shows a two node simulation topology. The Message plus Response Sources create packet-switched workload. Response Sources execute in response to a received message, typically from a Message Source, and automatically direct their traffic to the originator of that message.

Messages 1, 2, and 3 are based on an exponential distribution. The message size is based on a Poisson distribution of mean equal to 400 and a stream equal to 20 for all messages. Only results related to message 1 will be given, as well as some results related to the switch and the link channel utilization.

The number of processing elements at node 2 is constant. The four experiments taken differ only in changing the number of processing elements in node 1 while all other data remains the same. The delay of message 1 at node 1 is to be collected (or even message 2 @ node1). The simulation was running for 360 microseconds (real time) with the first arrival is at time 0 microseconds and the last at 60 microseconds seconds. This will allow enough time to the last messages to be processed.



**Node 1:**  
 Number of processing elements = 1; 5; 10; 20  
**Node 2:**  
 Number of Processing elements fixed  
**Message 1@node 1**  
 IAT: Exp(0.1)  
 First Arrival at t0= 0  
 Last Arrival at t1=60MS (The Run Time is 360 MS)  
 Probability Distribution for Message Size calculation  
 Poi(400.0,20.0) (Mean=400; Stream=20)

**Figure 8.12: Processing Elements in an ATM Environments. Study is based on two node simulation and optical transmission.**

The reports to be collected, are:

- The message delay Report message delay which present the statistics as perceived by the sender. Message delay is the time between creating the first packet of the message on the originating node and the time that the originating node is notified that the message has been assembled by the destination.
- The message delivered report presents statistics on the delay before the message is reassembled by the destination; That is the time between creating

the first packet of the message on the originating node and the time of the receiving the last packet of the message on the originating node and the time of the receiving the last packet on the destination node. The message may still be worked on the source after the message is delivered because of the retransmission, waiting for acknowledgment, or for modeling the close-sequence of the connection, but these values are not factored into the message delivered delay.

- The Packet delay Report presents a summary of the number of packets created, delivered, resent, or dropped for each message and response source. It also provides the average and maximum packet delay. Packet Delay is the time between creating a packet on the originating node and the time of receiving the packet at the destination node.

Table 8.4 shows the results of the simulation related with message 1. As we can see the message delay of message1 from node1 to node1 as destination is the same for all the experiments taken (P=1; 5; 10; and 20), and is the same as the message delivered delay to the same destination. The packet delay is the same for all experiments (P=1; 5; 10; and 20) when the node destination is the same as the originating node.

When the destination is node2, the average and maximum of message delay (which remains the same as message delivered delay) have significant decrease while increasing the number of processors (and automatically the same results for packet delay), but the average delay and maximum delay remain important.

The transmission delay is a constant to each of the experiments and the more important is the value remain the same for all experiments.

		<i>Number of Processors At the originating Node</i>			
		<i>1</i>	<i>5</i>	<i>10</i>	<i>20</i>
<b>Message + Response Source at Node 1</b>					
<b>Message Delay/ Destination Node 1</b>					
	<i>Assembled</i>	302	302	302	302
	<i>Delay Average</i>	8.788	8.788	8.788	8.788
	<i>Delay STD Dev</i>	0.516	0.516	0.516	0.516
	<i>Delay Maximum</i>	10	10	10	10
<b>Message Delay/Destination Node 2</b>					
	<i>Assembled</i>	313	313	313	313
	<i>Delay Average</i>	29.237	10.866	10.185	9.865
	<i>Delay STD Dev</i>	55.329	6.58	2.424	0.47
	<i>Delay Maximum</i>	286	61	33.554	11.042
<b>Message Delivered/ Destination Node1</b>					
	<i>Assembled</i>	302	302	302	302
	<i>Delay Average</i>	8.788	8.788	8.788	8.788
	<i>Delay STD Dev</i>	0.516	0.516	0.516	0.516
	<i>Delay Maximum</i>	10	10	10	10
<b>Message Delivered/ destination Node2</b>					
	<i>Assembled</i>	313	313	313	313
	<i>Delay Average</i>	29.237	10.866	10.185	9.906
	<i>Delay STD Dev</i>	55.329	6.58	2.424	0.802
	<i>Delay Maximum</i>	286	61.554	33.554	19.554
<b>Packet Delay/ Destination Node 1</b>					
	<i>Created</i>	2654	2654	2654	2654
	<i>Delivered</i>	2654	2654	2654	2654
	<i>Average delay (MS)</i>	1	1	1	1
	<i>Maximum Delay (MS)</i>	1	1	1	1
<b>Packet Delay/ Destination Node 2</b>					
	<i>Created</i>	2756	2756	2756	2756
	<i>Delivered</i>	2756	2756	2756	2756
	<i>Average delay (MS)</i>	20.885	3.071	2.371	2.109
	<i>Maximum Delay (MS)</i>	279	57.042	29.042	15.042
<b>Link : Channel Utilization</b>					
<b>Result is same in all cases</b>					
<b>The Transmission delay depends only on quality and Load of Link</b>					
	<i>Average Delay (MS)</i>	0.42			
	<i>Maximum delay (MS)</i>	0.42			

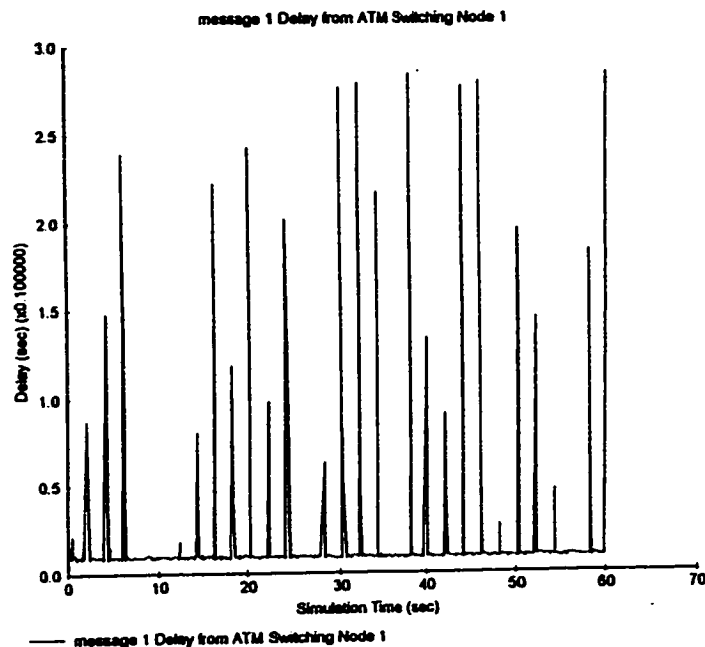
**Remark:**  
Destination Node has the number of processing elements

### **Table 8.4: Effect of number of distributed Processing on the ATM switching System**

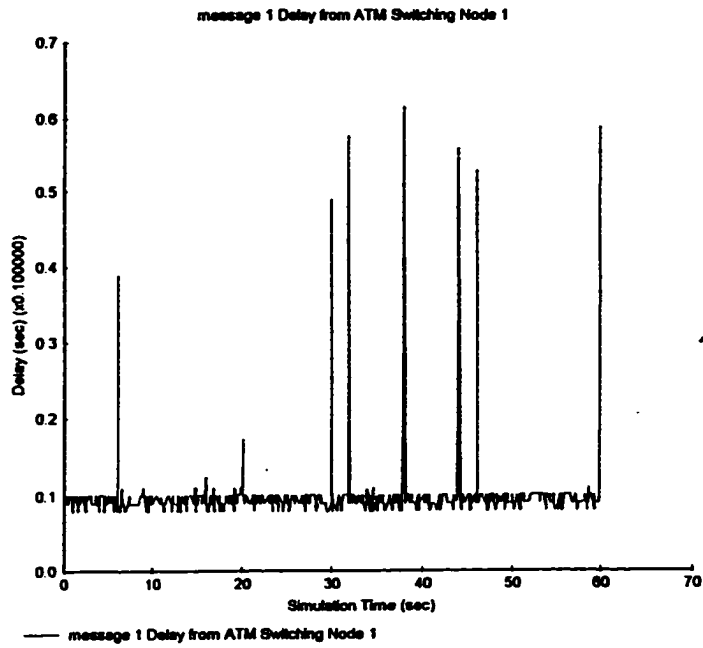
For a closer look to this phenomena, some real time diagram have been taken (see following figures). Figure 8.13 represents the message 1 delay from ATM switching node 1 which is the same as the message delivery delay from ATM

switching node 1 (No Acknowledgment need to be returned in case of ATM networks). All delays are fixed except the routing processin.

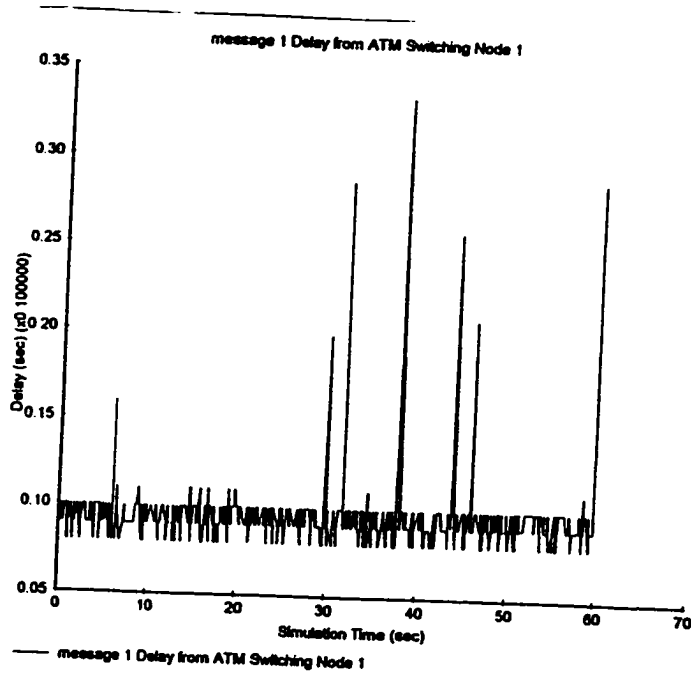
**Figure 8.13: Message 1 Delay from ATM Switching Node 1 which is the same as message 1 delivery delay from ATM Switching Node 1; (a) P=1, (b) P=5, and (c) P=10.**



### 8.13.a. P= 1



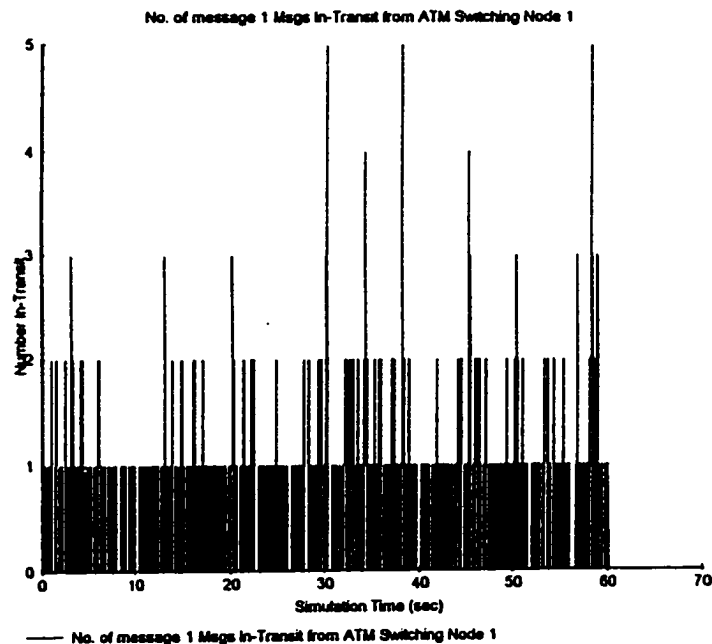
8.13.b. P = 5



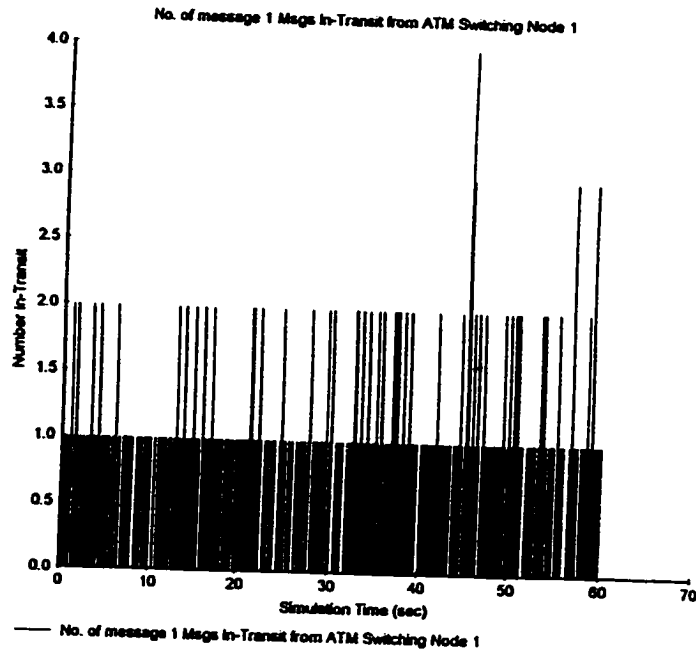
8.13.c. P = 10

Figure 8.14 shows the diagrams that represent the number of messages (message number 1) in transit from the ATM switching node 1. The same distribution function was applied in all cases, and we still find some difference in the number of messages in transit from the ATM switching node 1. This is due to delay caused by the call processing.

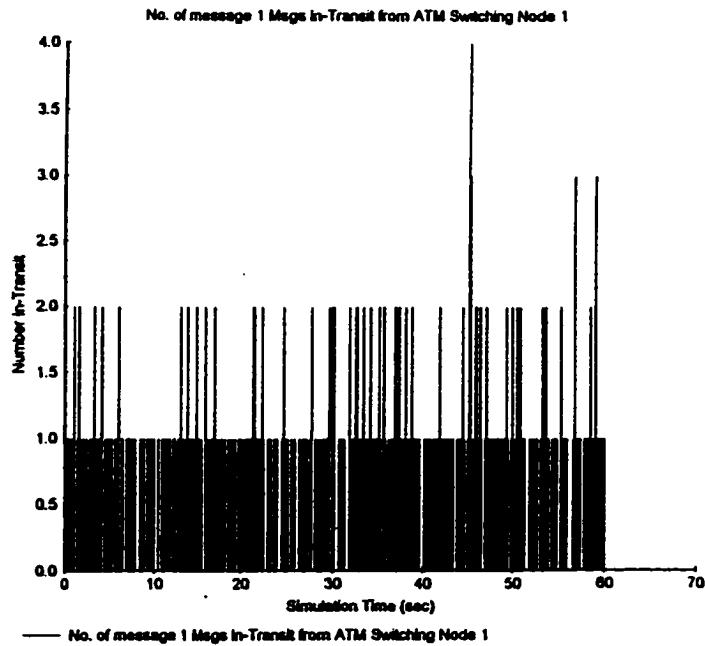
**Figure 14: Number of messages (1) in transit from ATM Switching node1; (a) P=1, (b) P=5, and (c) P=10.**



#### 8.14.a. P = 1



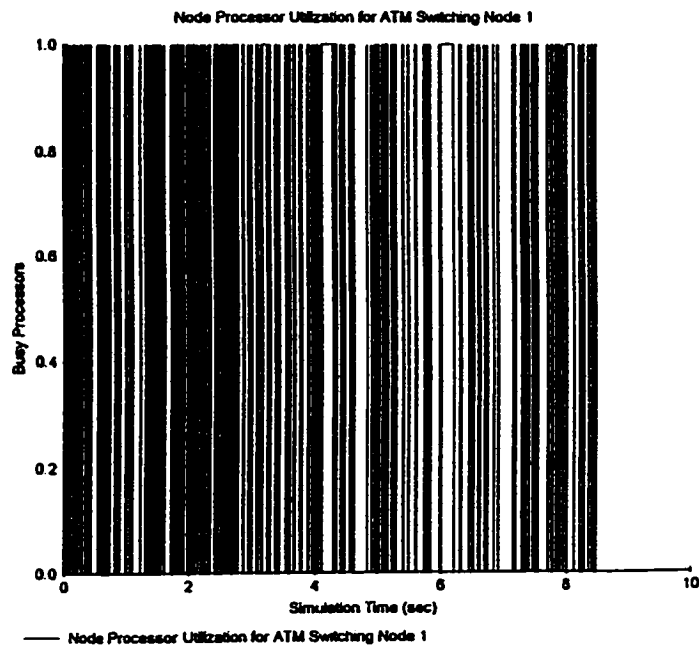
8.14.b. P = 5



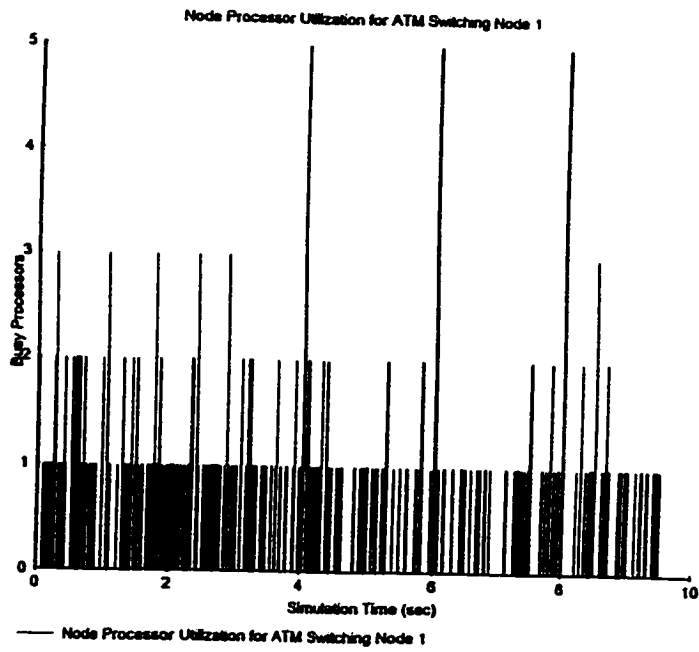
8.14.c. P = 10

Figure 8.15 shows the node processor utilization for ATM Switching node 1 which is the number of processor in use.

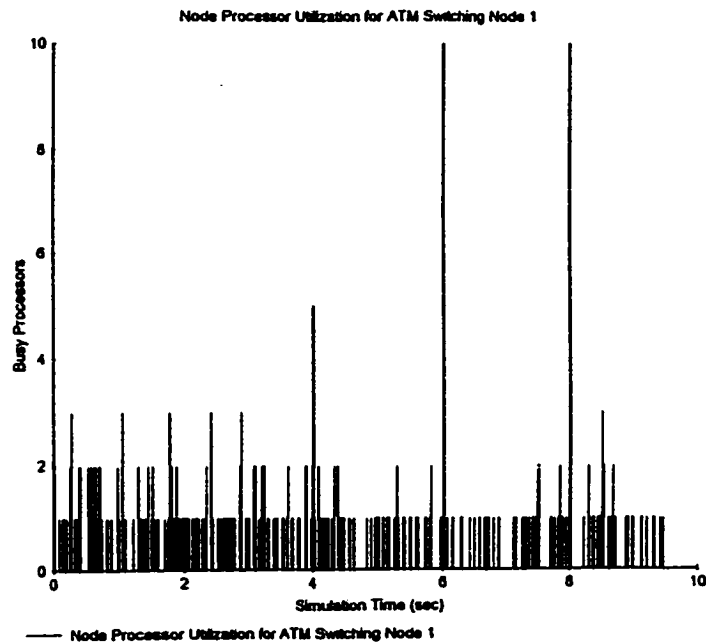
**Figure 8.15: node processor utilization for ATM Switching node 1 which the number of processor in use**



**8.15.a. P = 1**



8.15.b. P = 5

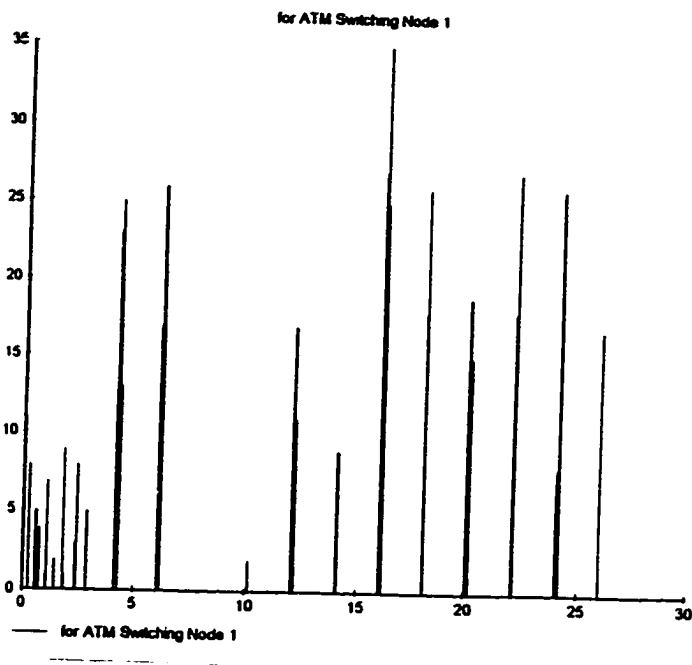


8.15.c. P = 10

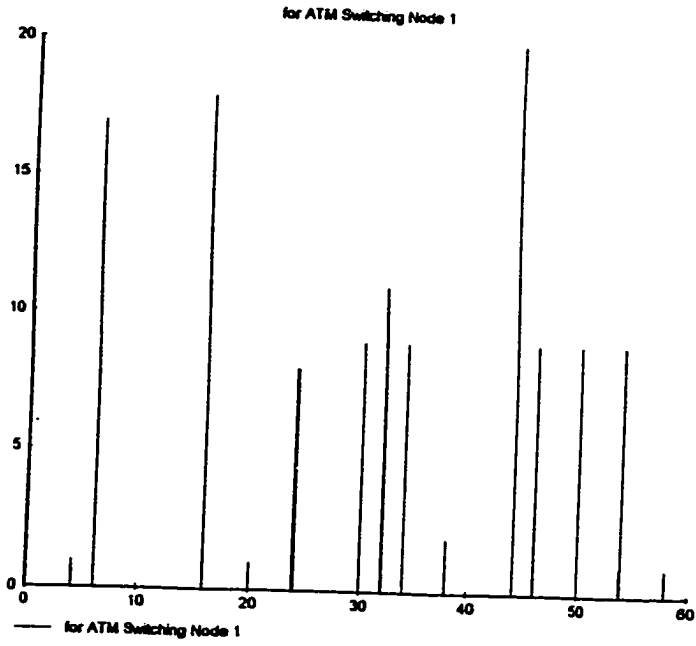
Figure 8.16 shows the input buffer for ATM switching Node 1. This figure will show the number of messages type 1 holding the buffer waiting to be processed.

The larger is the number of processor in use, the lower is the number of messages waiting in the buffer to be processed.

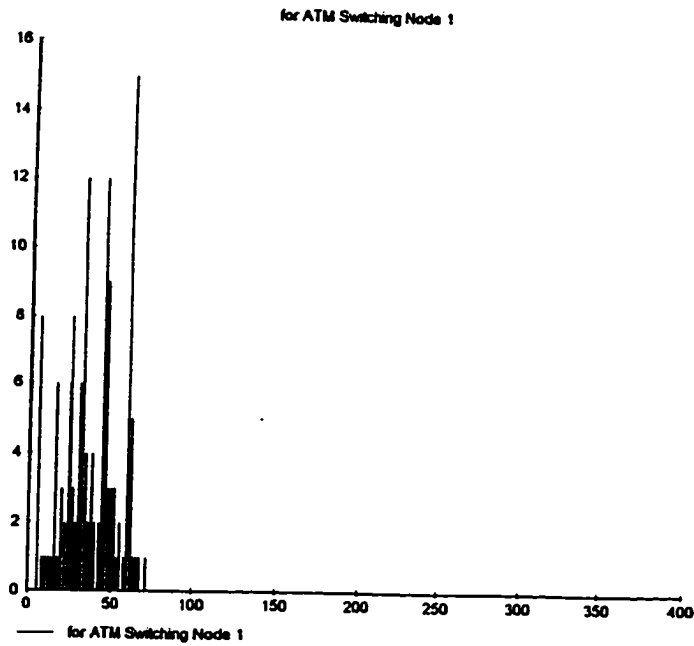
**Figure 8.16: ATM switching Node 1; Buffer use by message 1**



**8.16.a. P = 1**



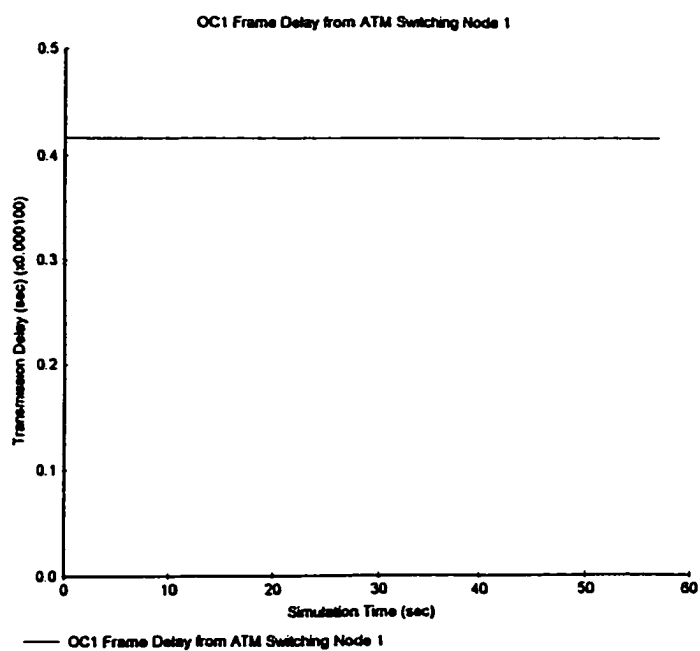
8.16.b. P = 5



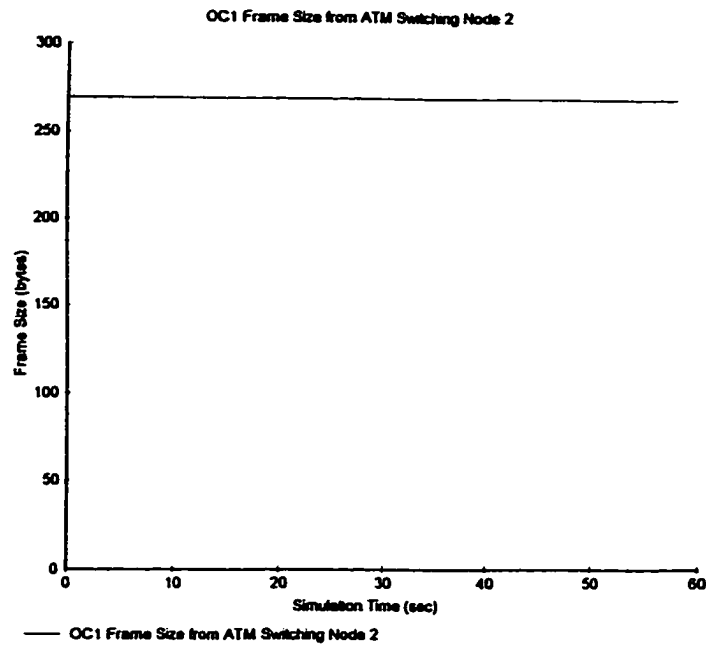
8.16.c. P = 10

Figure 8.17 shows the OC1 frame delay from ATM Switching Node 1 which is the same for different number of processors, and is the same as the OC1 frame delay from ATM Switching Node 2. It only depends on the link itself and the load on that specific link. The delay is a constant  $d = 0.4 * 10^{-3}$  Seconds.

Figure 8.18 will show the OC1 frame size which is defined to be 155Mbps for all experiments, and the OC1 Channel Utilization from switching node 1.



**Figure 8.17: OC 1 frame Delay from ATM Channel Node 1 ( $p=1, 5, 10, 20$ ); It is the same as OC 1 Frame Delay from ATM Channel Node 2.**

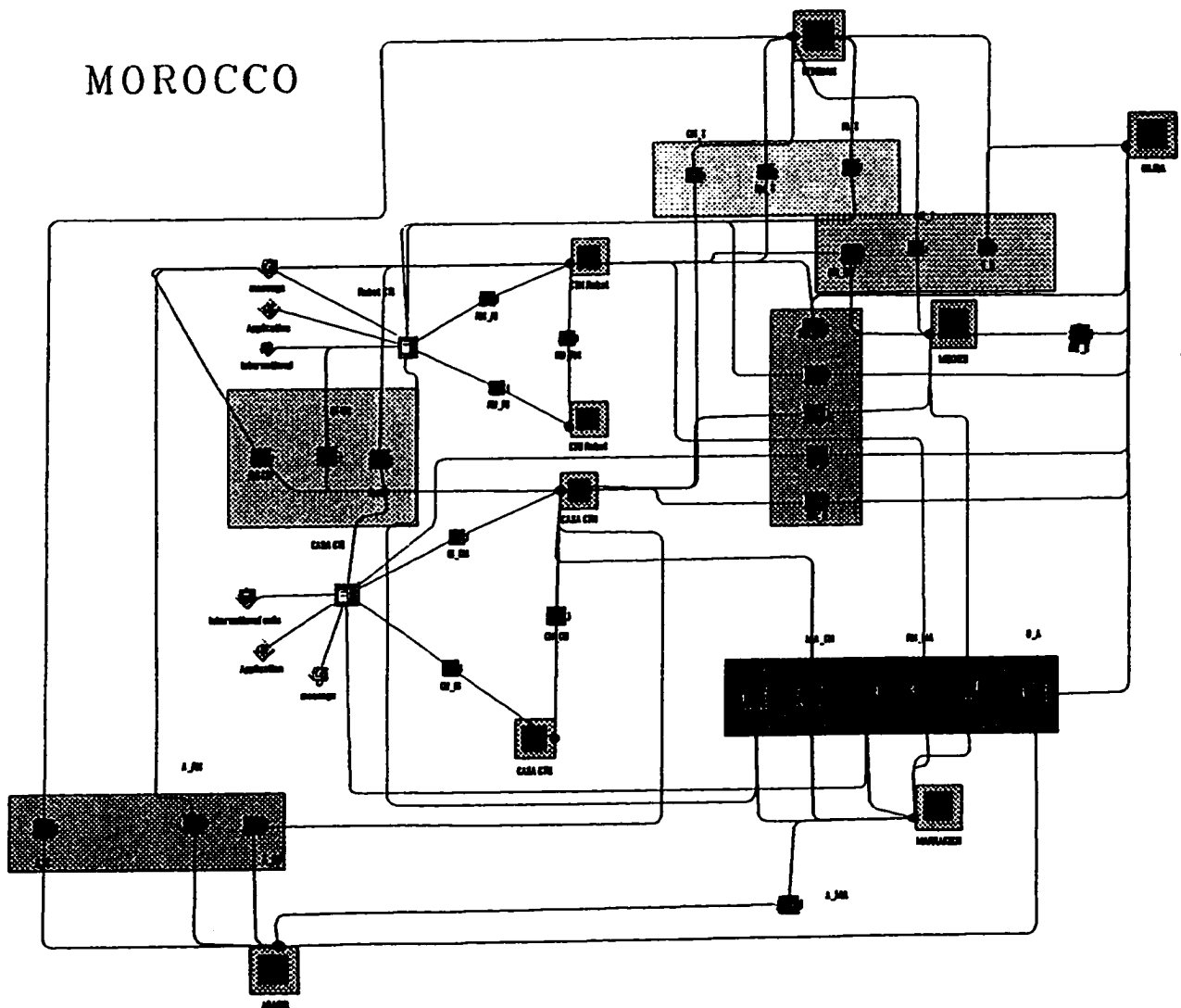


**Figure 8.18: OC1 Frame Size from ATM Switching Node 1/ 2**

The Link Utilization Delay is a constant, and the node delay were fixed to a constant except for NNI processing. The message delay is the same as the message delivered delay. It includes the ingress processing delay, the delay at the switch level, the delay at the egress interface, the transmission delay, and the packetize delay. All delay due the processing at the egress interface , and the transmission delay were fixed to constant values that are the same for all experiments ( $P=1,5, 10, 20$ ). The message delay is determined, and depend on the number of processing elements. The transmission delay is a constant. Then we can conclude that the delay due to the processing at the egress interface will decrease dramatically by increasing the number of processing elements. Function Egress delay looks like an exponential of number of processors.

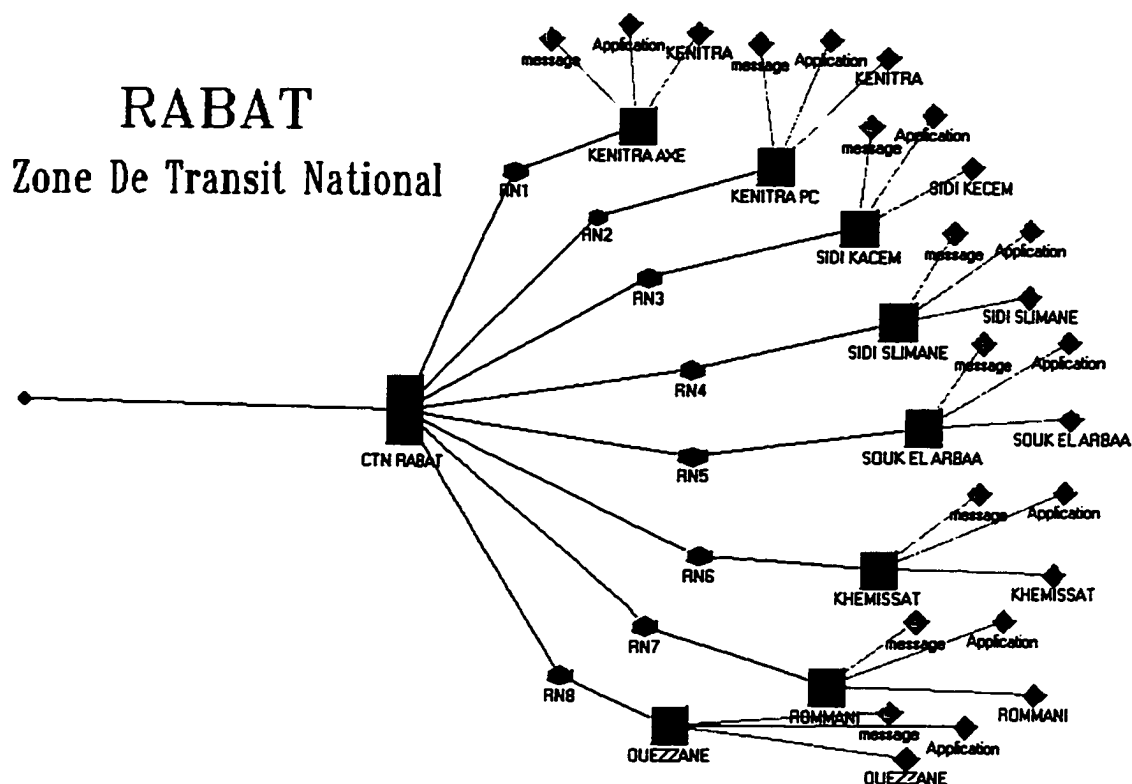
### 8.2.2.2. NATIONAL NETWORK SIMULATION

A few application like e-mails, file transfer were added to the national network. In case of video, we will use the same scheme, since it will be digitized, and compressed before it's actually send through an ATM network. The old nodes are kept the same providing voice communications, and a new ATM nodes were added to provide very fast packet transfer. This new services should be allowed to subscribers who pay for the service.



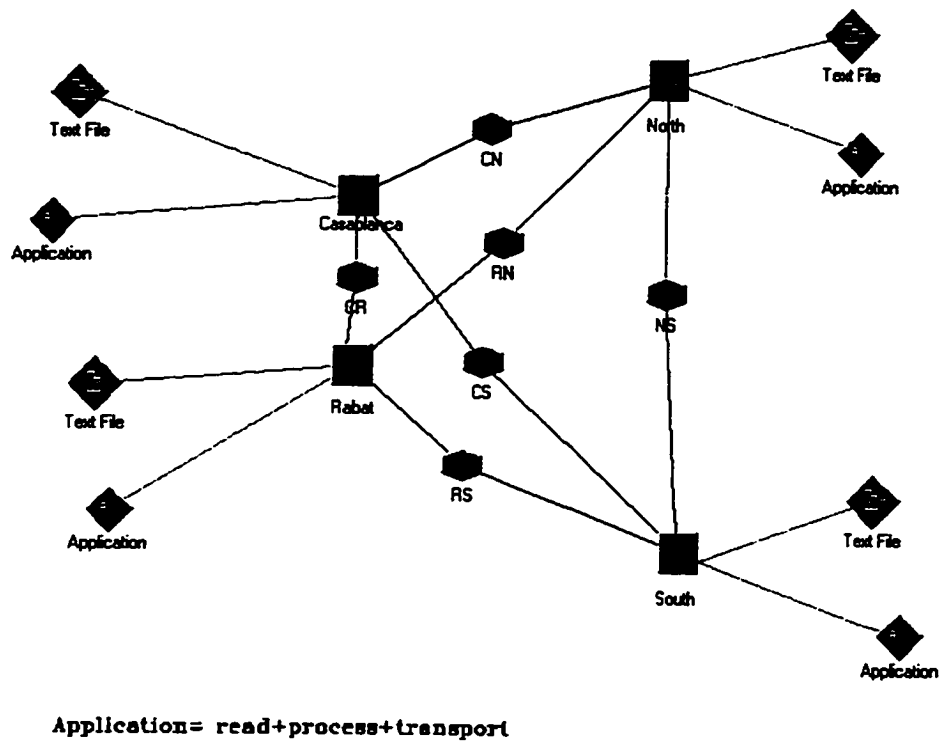
**Figure 8.19: National Network providing ATM Services**

On each of the subnetworks on the national network, ATM nodes have been placed, and new services based on the cell technology have been defined. The topology of this subnetworks will be similar to the following topology.



**Figure 8.20: CTN Rabat topology based on ATM technology and providing cell relay technology services.**

The Computer Freezes while running the previous simulation, or saving the results. A simplest version only cell switching and based on the 4 major nodes for the regions of Casablanca, Rabat, North, and South is simulated. This version is more realistic looking at the development curve in the Moroccan Network.



**Figure 8.21: A four Node based ATM network covering the National Network.**

The results related to CasaBlanca including the application and messages generated at that node are shown in the following figures. The size of the files used is very large, and the applications need a lot of processing. The results shown are based on ATM nodes supporting all kind of communication schemes.

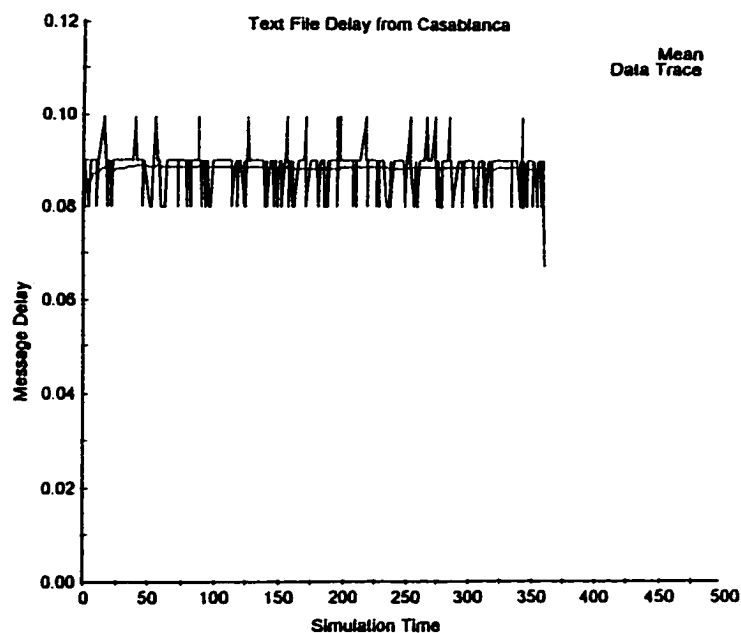
The interarrival time of the text files at the node Casablanca are based on an exponential distribution, and the size on a Poisson distribution (Poi(400.0, 10)). The reason we used a very high size text files is to force the computer to finish the execution. Our best alternative was to have more frequent arrivals an

frequent text files, but since we were not able to get any results due to the capacity of the IBM ThinkPad used for the simulations.

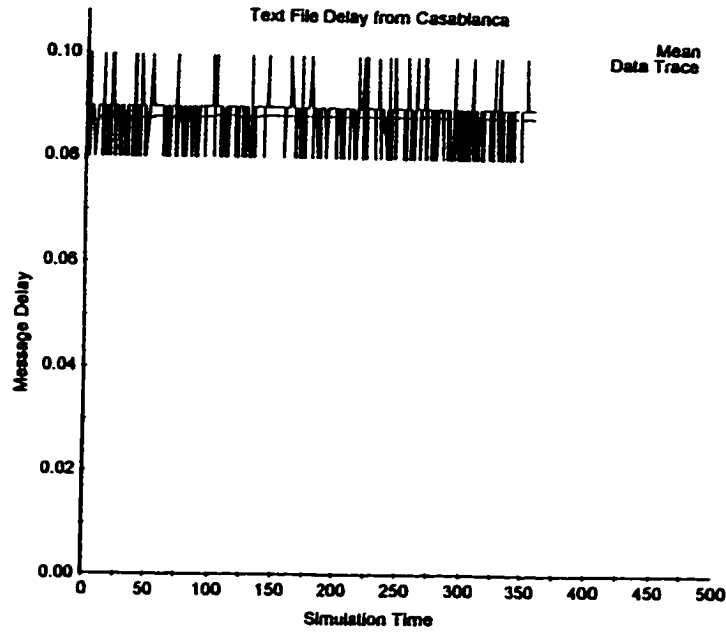
The experiments are based on changing the interarrival time, and keeping everything else the same to see the influence of congestion of the network.

Delays related to the text file as well as the application are collected.

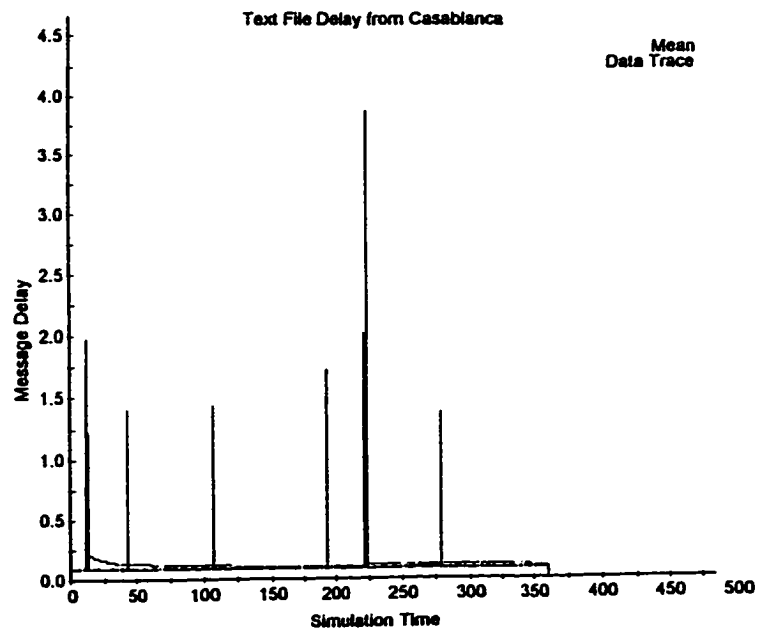
**Figure 8.22: Text file delay as function of the interarrival time.**



### 8.22.a. IAT=1.0 based on Exponential Distribution

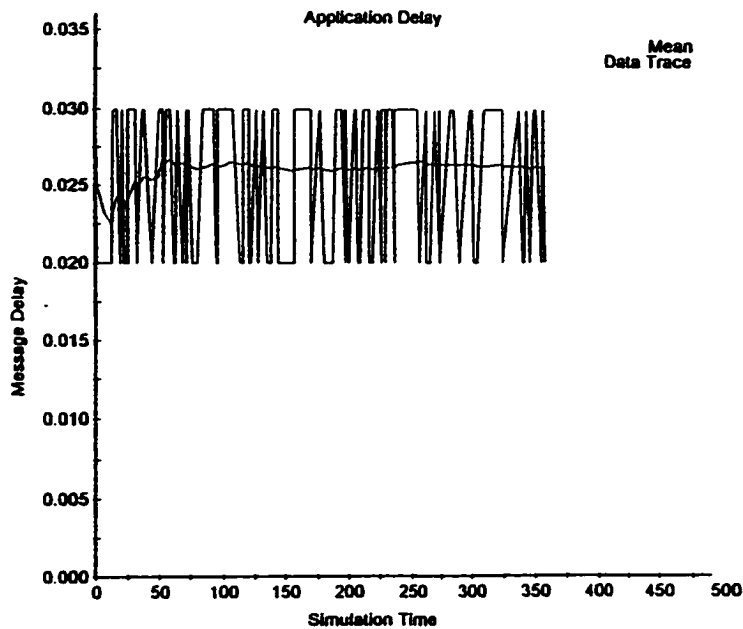


### 8.22.b. IAT=0.5 based on Exponential Distribution

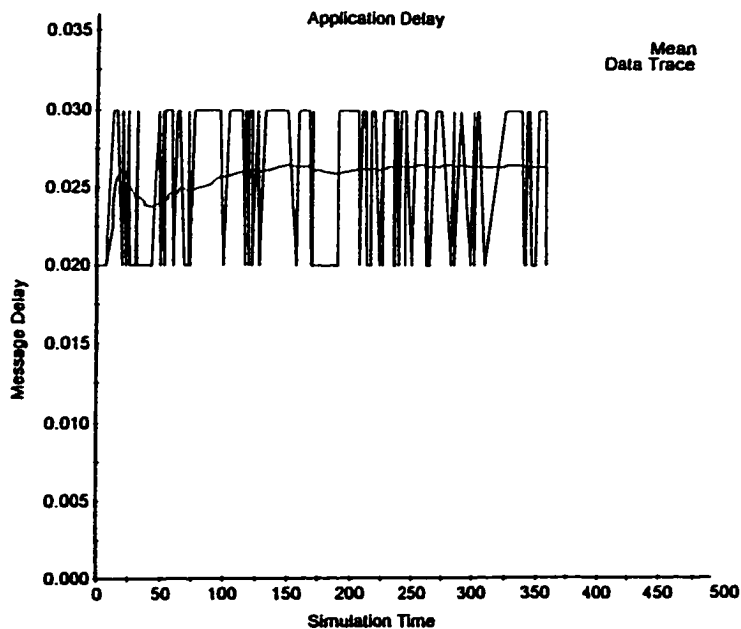


### 8.22.c. IAT=0.1 based on Exponential Distribution

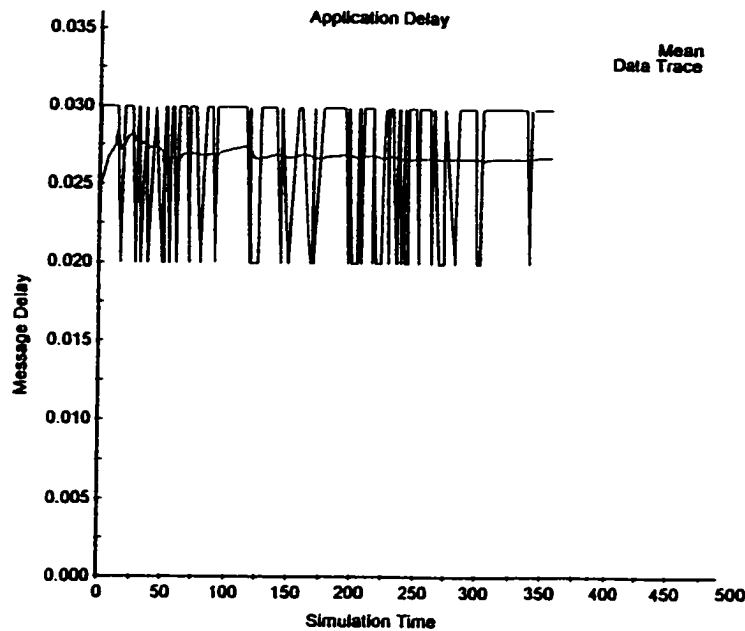
**Figure 8.23: Application as function of the interarrival time of text files the same node.**



**8.23.a. IAT=1.0 based on Exponential Distribution**



**8.23.b. IAT=0.5 based on Exponential Distribution**



### 8.23.c. IAT=0.1 based on Exponential Distribution

The decrease of the interarrival time will increase the number of text files to be transported from node 1. The delay of this text file is minimal taking in consideration the size each ( $Poi(400.0,20.0)$ ), and will increase by decreasing the interarrival time. The application is also affected by this change on the interarrival time for the text file, and the delay is becoming more important.

Processing will remain the key to solve this type of problems, as well as good compression techniques for ATM networks.

### CONCLUSION

The largest component of the bandwidth in BISDN/ATM networks is expected to carry video traffic. This is due to the introduction of many video/multimedia

services and also to the large amount of bandwidth needed to transport real time digital video. To increase the number of video streams that can be simultaneously carried over a link, compression schemes can be employed in order to reduce the amount of data required by video frames. Compression converts video into a variable bit rate signal (VBR). Unlike conventional data networks, ATM networks efficient support for VBR traffic.

To minimized the delay at the switch level, the architecture of ATM switches need to be looked at. Parallel processing is recommended at the input of the switch, and also just before the trunks. Call processing will be distributed between both interfaces. The first interface will deal with UNI and the second with NNI. Once the this architecture is on place, the delay will be minimized, and even more messages will go through at once. This processing elements, all with corresponding memories, and software applications should be able to consider each of the ATM layers independently.

### **8.3.**

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## **SIGNALING NETWORKS SUPPORTING ATM BASED TRANSPORT NETWORKS**

Currently, telecommunication networks are based on SS7 signaling. The Common Channel Signaling Networks are using X.25 protocol to transfer signals between SSPs and SCPs. In the future, all communication Networks will also provide Intelligent Services all with typical type of IN services. Then, the signaling network may get to be congested, and the need of signaling networks

based on ATM may become a necessity. Other than congestion, we may need to provide the quality of service to the signaling network. This will be the case of a accessing to a file server to find the information concerning a patient by using his finger prints for example.

### **8.3.1.**

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#### **Intelligent Networking and delay of services**

The actual simulation is almost the same as simulating a client server environment where the SCP is the server and any node within a network who has the capabilities of providing intelligent services is the client. The client is requesting information from the server. Two measurement points are needed—the client local network and the server local network. At the client side, we can measure client delay or “think time”. At the packet level, this is the delay between the time the client (switching node) receives a frame of data until it requests the next frame. Also at the client level, we can measure the combined (network +server) delay. The second part of the measurement is to go to the server network and measure the server delay.

Let Client think time =  $t_1$ ,

(network + Server )delay =  $t_c$

Server think time =  $t_3$

Calculate the network delay:

Network Delay =  $t_2 = t_c - t_3$

Total transaction delay =  $t_1 + t_2 + t_3$

The client which represents the switching node will need to have the ability to detect the type of call, access to the trigger database in the Intelligent Peripheral to check if the call needs access to the SCP for more detail about the call, etc. The server should have the ability to check the information from the local database, and send the details about the call routing and type back to the original node, or the receiving node depending on the type of intelligence. A connection will then be made between different nodes, and the data will be processed or transported within the network.

The analysis we are doing here is to get an idea if whether the bottlenecks are in the switching node (Client), SCP (Server) or CCISS Network. The node and SCP have both three potential bottlenecks: memory, CPU cycles, and I/O (both for network and disk). The SCP performance must scale with other Switching nodes or traffic load; the switching node is supposed to be supporting only one user (related with IN calls).

In case of communication between SCP and SSP, the hosts (SCP and SSP) are constrained by three factors: -Memory, CPU, and I/O- and networks are limited by two fundamental constraints: bandwidth and latency. The bandwidth is determined by the physical. Data link layers of the network infrastructure, and the latency is more subtle and has more components: propagation delay, store

and forward delay in routers and switches, access delay for shared media. Each network component will add latency. Bandwidth and latency combine to determine the bottom line throughput of the communication channel.

In the simplest case, we will have a node transmitting a request (to the SCP for example) in blocks of data of size  $W$  bytes.  $W$  is the smaller of the transport layer protocol window or the request block size. This information is transmitted over the network of bandwidth  $BW$ . Each block transmitted is followed by an acknowledgment (case of X.25 which is not needed for ATM). The data rate for this transaction will be:

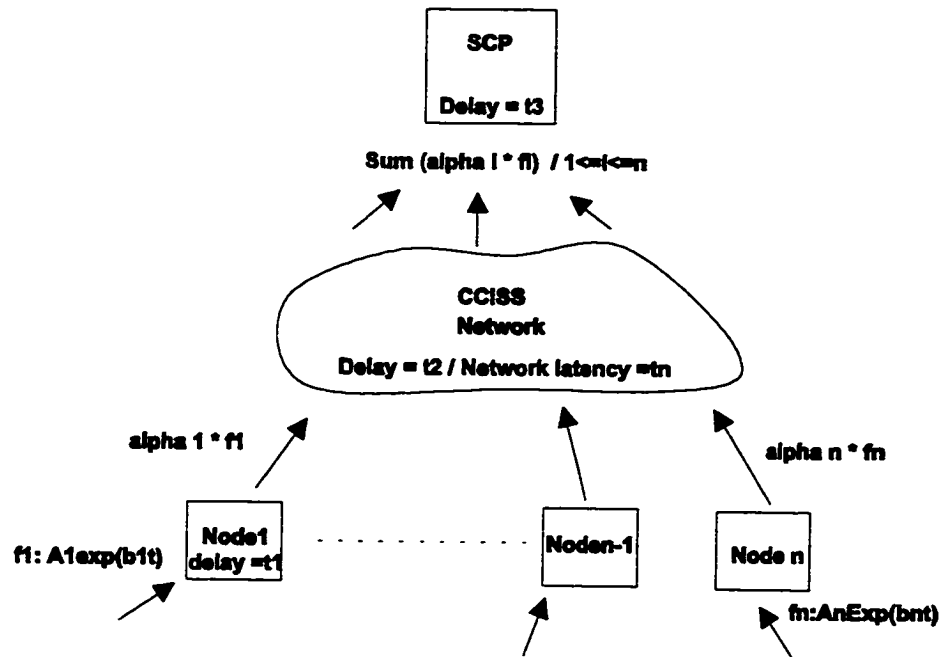
Data rate =  $W / (W / WB + 2t_n + t_s + t_c)$ ; where  $t_n$  is the network latency,  $t_s$  is the server think time and  $t_c$  is the client think time.  $BW$  is what the network bandwidth.  $W / WB$  is the time to actually transmit the block of data over the transmission media. The data rate is simply the size of the block of transmitted ( $W$ ) divided by the total time it takes to send, receive and acknowledge this data.

Note that it appears twice in the formula.

In such cases a formula is very easy to get. Once we have the formula, we will decide which part of the network is to study. For example, in our case we need to decide which is better for signaling? ATM or X.25?. If , we assume that  $t_s = t_c = 0$ , then normalized bandwidth:

$$\text{Data Rate} / BW = 1 / [1 + (2t_n BW / W)].$$

The idea presented thus far not have not taken into account the fact that the user data or requests may simultaneously make demands on the SCP. Figure 8.24 shows an example of requests exponentially arriving to the nodes, a percentage of this requests will require some communication with the SCP.



**Figure 8.24: Delay at the SCP level as a function of requests, type of requests, and processing elements**

In this case a queue of the requests will form as individual transactions wait their turn to be processed. queuing is another important concept that must be taken into account.

The number of packets requiring an attention from the SCP I equal to:

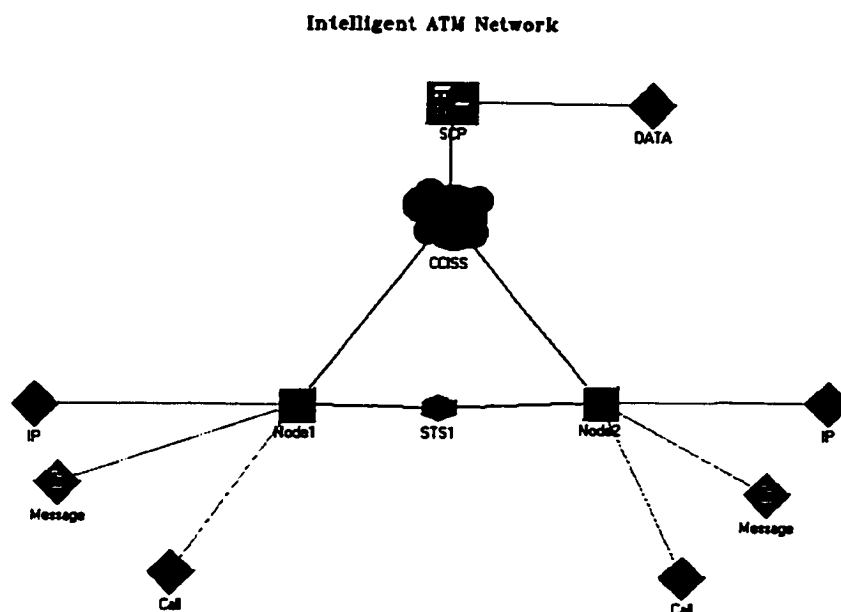
$$\sum (\alpha_i f_i) / 1 \leq i \leq n$$

After a delay due to the latency and propagation this packets will arrive to the input queue of the SCP for processing. The processing time of a given packet will depend on the service. The delay at the SCP level will then depend on the the local processing equation and the interarrival time of packets requesting special attention.

### 8.3.2.

## Intelligent ATM Network Simulation Results

In this part, we will simulate both cases by using an ATM based transport network. Figure 8.25 shows the architecture used to simulate the intelligent ATM network.



**Figure 8.25: Intelligent ATM Network Simulation;**

The previous architecture may be the case of the National network studied before since we don't need more than one SCP in it. Because of problems encountered while running the national network, interarrival time of different packets as well as the size of message and the number of possible ATM nodes has been changed. At the current stage, only two ATM nodes can be placed, one at the region of Casablanca and one the region of Rabat the more populated regions in Morocco. The northern regions can use the ATM node installed in Rabat as a first choice, and the southern regions can use the node of Casablanca as a first choice.

### **8.3.2.1. SIGNALING NETWORK BASED ON X.25**

The following simulation results is based on X.25 based SS7.

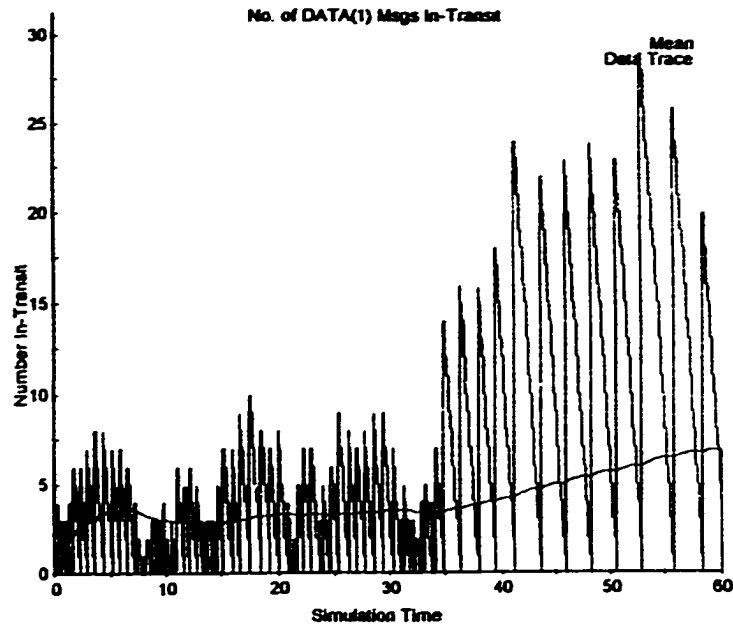
#### **SIMULATION FOR DIFFERENT NUMBER OF PROCESSORS AT THE**

#### **SCP LEVEL**

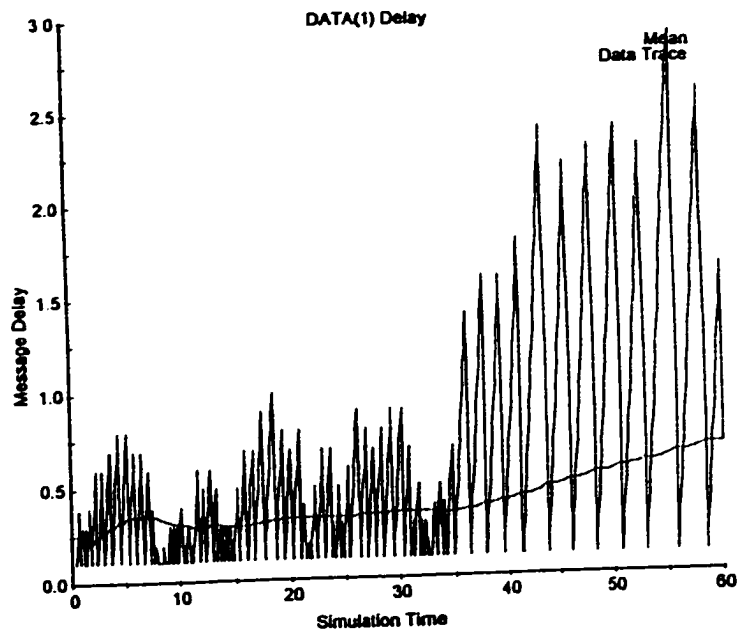
Figure 8.26, and figure 8.27 show the simulation results for a signaling system based on X.25. In the second case we increased the number of processing elements from P1 to P2.

For the same intelligent request packets, when the number of processors increase at the SCP level, the number of message in transit at the SCP level goes down with the delay.

All the delay are supposed to be constant except the control messages delay.

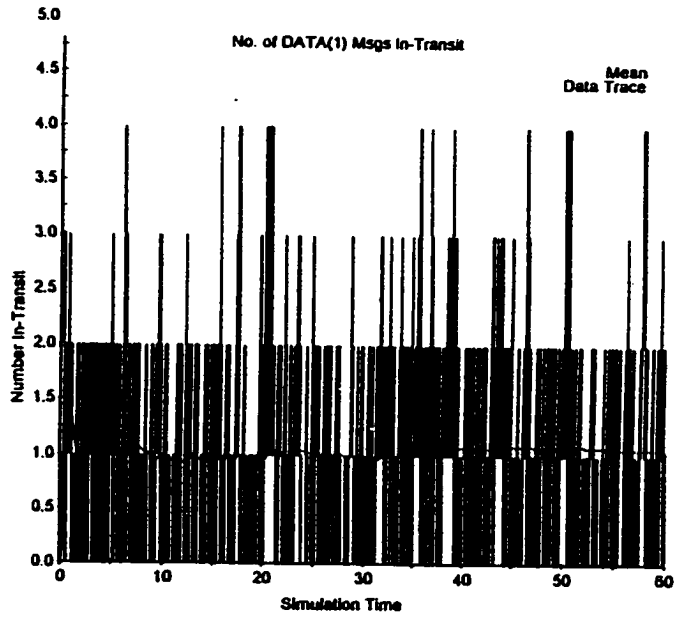


8.26.a. Number of signaling messages

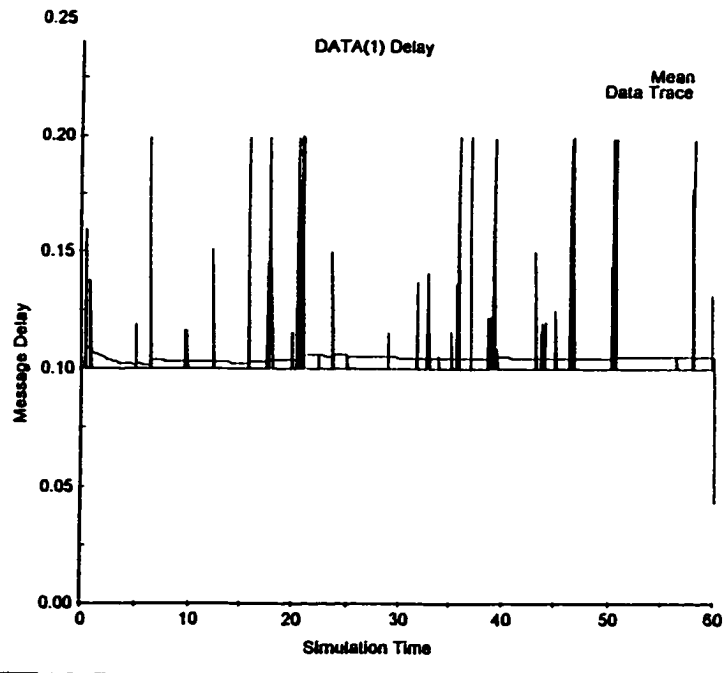


8.26.b. Signaling Message delay

**Figure 8.26: SS7 based on X.25 for Signaling; Number of signaling messages followed by the Signaling Message delay for P=5.**



8.27.a. Number of signaling messages



8.27.b. Signaling Message delay

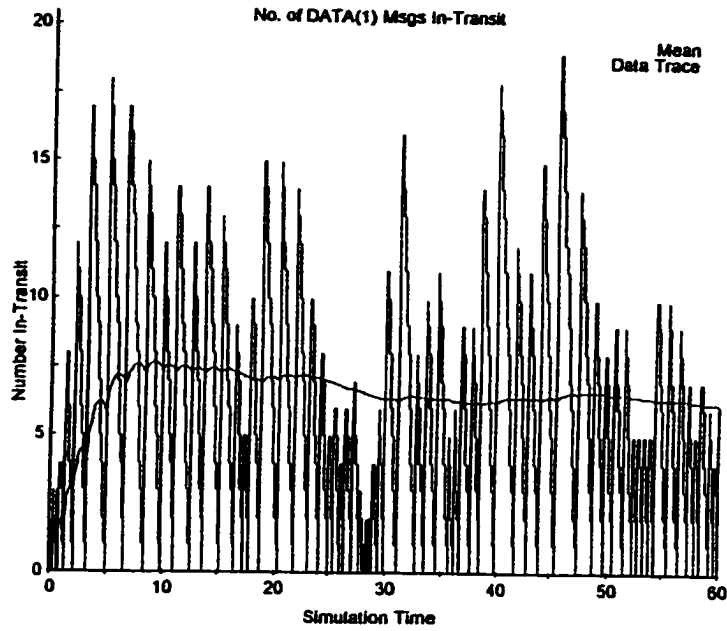
Figure 8.27: SS7 based on X.27 Signaling; Number of signaling messages followed by the Signaling Message delay for P=15.

When using the X.25 for SS7, the delays are minimum. A large number of processing elements are needed at the SCP level to decrease the delay. This delay are increasing with the control messages in transit. Once we reach the optimum number of processing elements, we get to a minimum delay, and the number of messages in transit decreases.

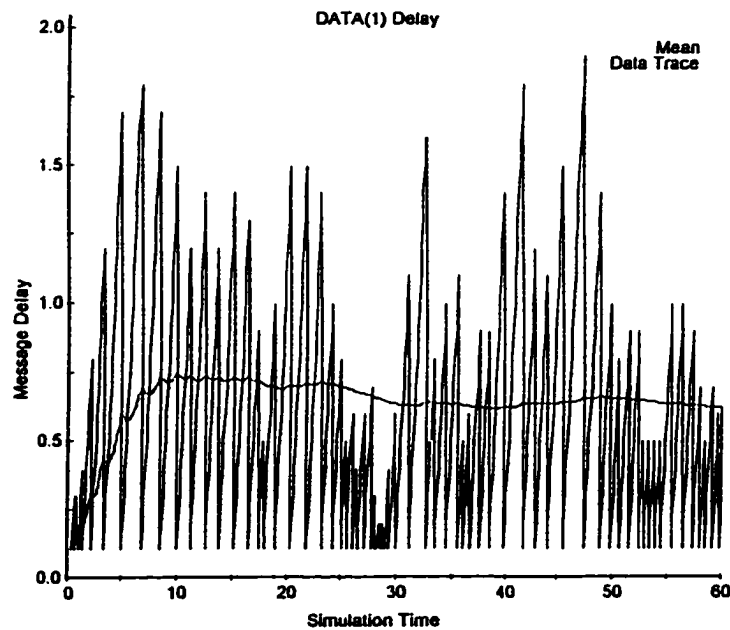
### **8.3.2.2. SIGNALING NETWORK BASED ON ATM**

The Common Channel Signaling Network should not show any congestion and the delays should minimum. Once the signaling network is congested, then user of a Signaling protocol based on ATM will come to stage. A Front End processor is placed before the SCP level to convert ATM cells into packets that can be processed by a computer node.

Figure 8.28, 8.29, and 8.30 will show the delay of control messages, as well as the number of signaling messages in transit at the SCP level. The total value of delays is 0.10 seconds. To find out the delay at the signaling network we will need to subtract the delay of 0.10 control messages from the delay of control messages.

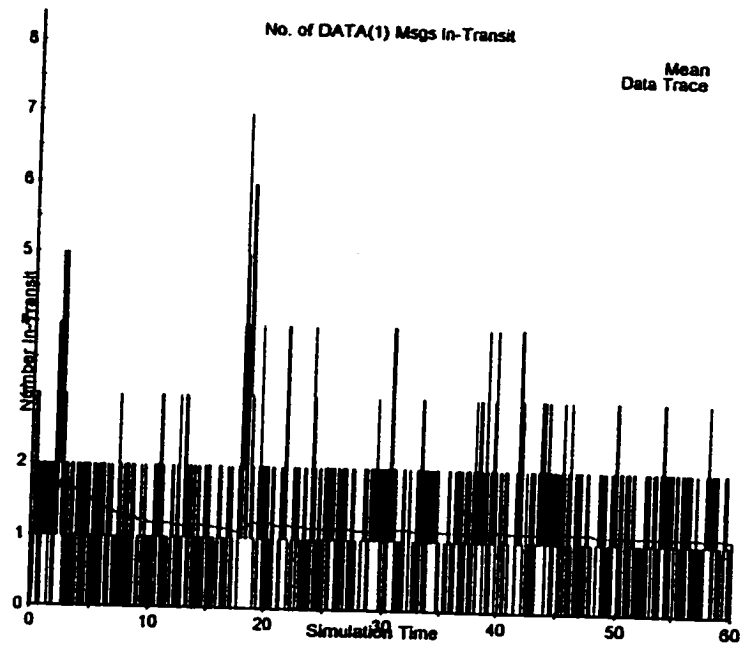


8.28.a. Number of signaling messages

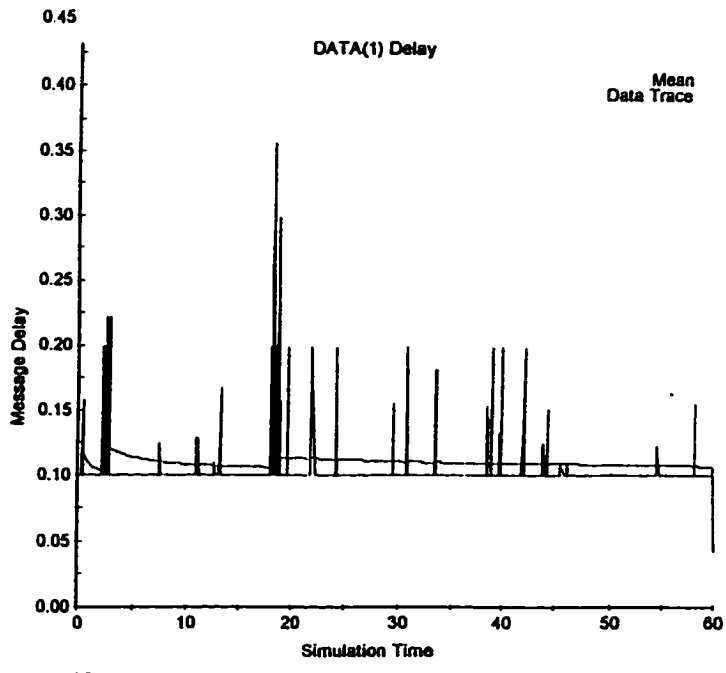


8.28.b. Signaling Message delay

**Figure 8.28: ATM Signaling; Number of signaling messages followed by the Signaling Message delay for  $P=1$ .**

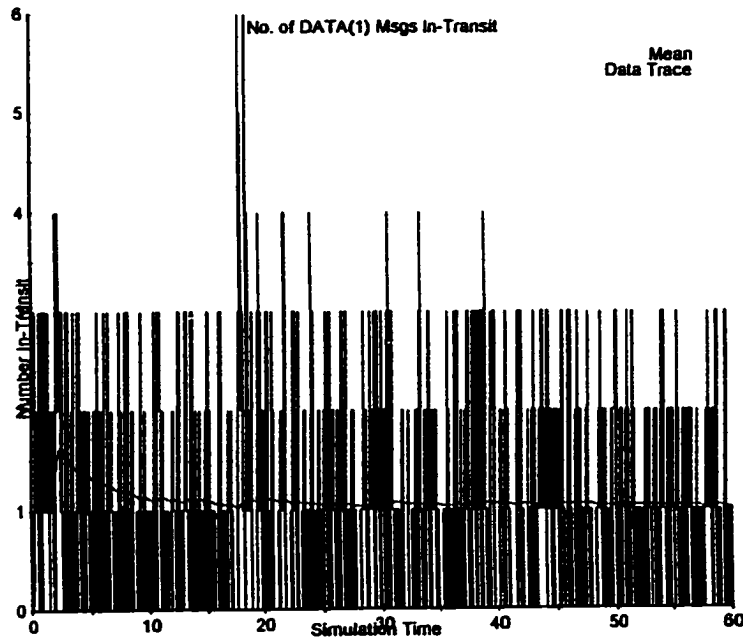


8.29.a. Number of signaling messages

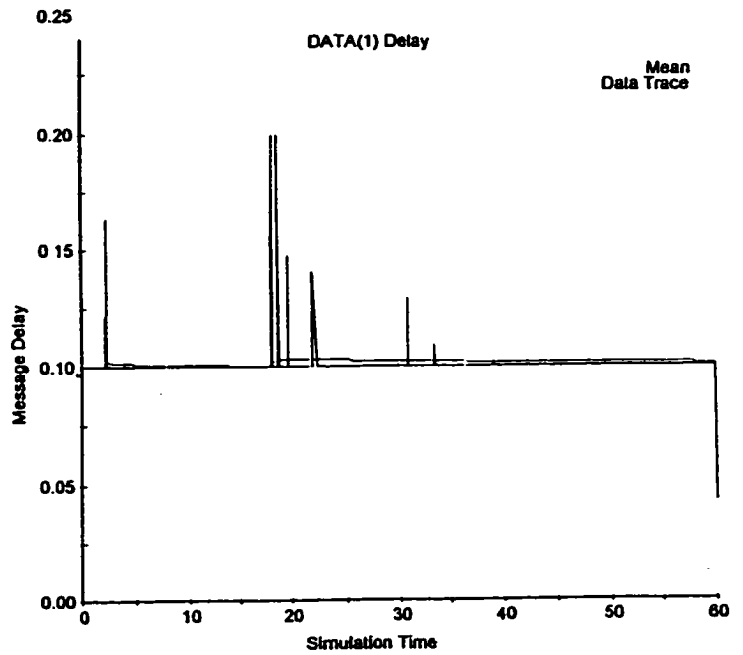


8.29.b. Signaling Message delay

Figure 8.29: ATM Signaling; Number of signaling messages followed by the Signaling Message delay for P=5.

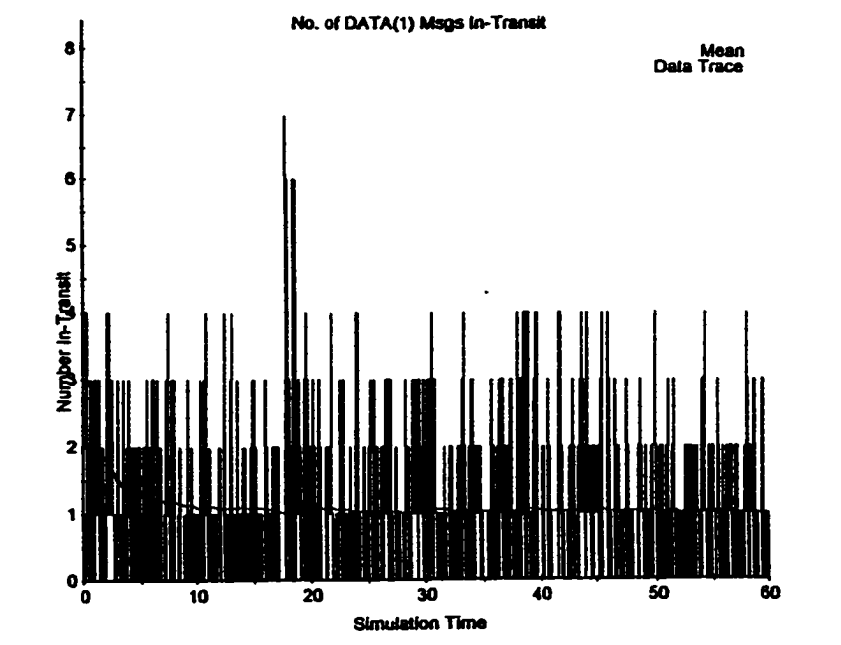


8.30.a. Number of signaling messages

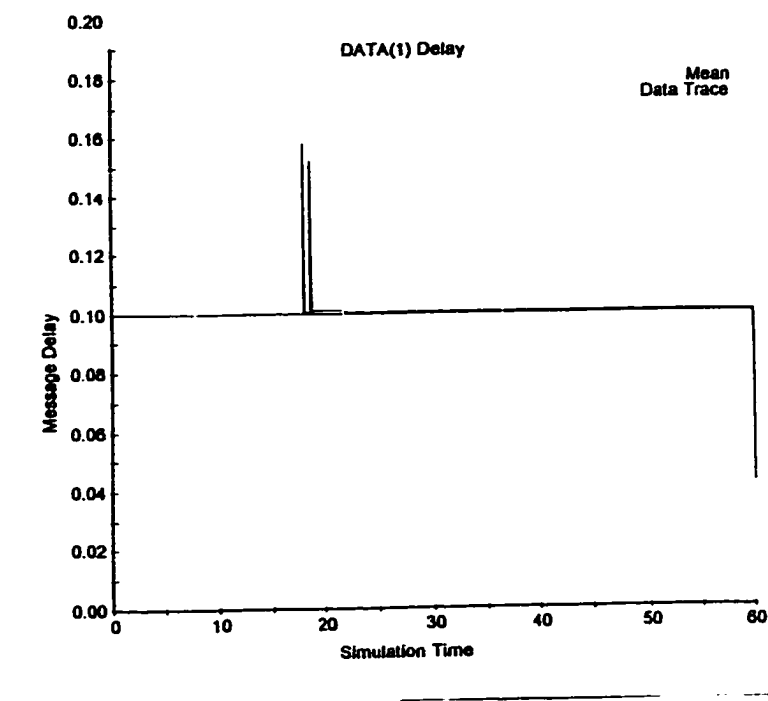


8.30.b. Signaling Message delay

**Figure 8.30: ATM Signaling; Number of signaling messages followed by the Signaling Message delay for P=10.**



8.31.a. Number of signaling messages



8.31.b. Signaling Message delay

**Figure 8.31: ATM Signaling; Number of signaling messages followed by the Signaling Message delay for P=15.**

When using the ATM for SS7, the delays are minimum. A large number of processing elements are needed at the SCP level to decrease the delay. This delay are increasing with the control messages in transit. Once we reach the optimum number of processing elements, we get to a minimum delay, and the number of messages in transit decreases. For an optimum value of processors at the SCP level and at the IP and switch level, delays on the signaling network will be practically null.

If we try to compare X.25 with ATM, we can see from the previous diagrams, when the number of processing elements is  $P=5$  for example, that the delays are increasing with the time in the case of X.25 while they are varying within the same interval when using ATM for transport. This shows that ATM are better for signaling in case of congestion.

### CONCLUSION

The signaling using ATM network gives better throughput than the actual X.25. But, since the current signaling network is not congested yet, then the use of SS7 the way it is should be sufficient. In the future, when the signaling network is congested, then the use of ATM at the signaling level will become a must. Then, A front end processor should be placed to convert ATM cells into X.25 to be processed by the SCP.

## 8.4.

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### CONCLUSION

In case of voice communication, the simulation results for ATM nodes shows the same results as the current electronic switching systems, at the time where the ATM switches offer better quality of service, support optical carriers, and mainly less expensive. The ATM switching systems are a better alternative for voice communication. In this case, the signaling system number 7 can remain the way it is now for the ATM networks to provide Intelligent services for the voice based communications.

In the case of data or video communication, ATM networks with optical transport provide very high quality of services with low delay. This delays can be minimized if the architecture of the ATM switches is based on a distributed processing. In that case, an egress processing module should be added to take care of the routing. This processing module will deal with NNI signals.

During the beginning of the installation, ATM can be placed to provide routing services, and then better throughput. Latter, ATM nodes can be placed as access nodes to offer a very high quality service and very low delay to the users who pay for it.

To provide the Intelligent communication services, the intelligent peripheral on the IN/2 can be upgraded to process call signals and decide if the call is intelligent or not. This IP module will receive from the switch calls that need special attention, and check on its trigger table for more information to route the

call through the signaling network to the specific file server containing the information about the type of the call or application. The signaling network can still be based on SS7 which provide very low delay. In case of Intelligent Broadband services, the control signal should contain information about BISUP, and this fits the current SS7 protocol, but for better service, this signal can be sent through an ATM network.

## Chapter 9

### **CONCLUSIONS**

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ATM Networks have the capability of supporting multiple Services such as: Constant Bit Rate, Variable Bit Rate, Connection Oriented data Service, and Connectionless Services on the same platform. They also provide a wide range access (digital signals and optical carriers) with a high performance that includes a low latency and high throughput. The Intelligent ATM Networks promise to provide this capabilities for the futuristic, still to define, intelligent services. This new intelligent services will not only be based on the dialed number scheme, but also may be implemented for any type of communication services by using an ATM address for example, or even better deciding on the type of intelligence from the user premises when requesting information by using a specific menu.

Networks such as medical and educational networks should take very good advantage of the Intelligent ATM networks. The intelligent medical network have been under study, and promise better service. Once implemented in

medical Networks, ATM networks will provide better service to the medical environment, and should provide further services if this networks are intelligent.

As a starting point, Intelligent ATM Networks can still use the SS7 as a signaling protocol through the common channel network (non-associated mode), but once Intelligent Boadband services are defined, the signaling system will be using ATM cells to transfer control signals on either the common channel signaling system, or through the transport network (associated mode). In the far futur two major problems will need to be taken in consideration:

- the signaling network will be traitned by congestion, and
- the number of file servers containing the control information will become very large.

To solve this problems, the signaling and transport can be joined using a non-associated mode which fits perfectly the ATM networks.

The future Service Control Point will then be presented by a large number of File Servers that have the ability to communicate with each other using databases with common fields. The signaling network, then, will be based on a cell command (as in a computer instruction) and a fiber for transmission (as in the case of a bus for input and output in the case of a computer system). The voice communication intelligent services can still be based on the current SS7, and the future intelligent Broadband services as well as Intelligent Communication Services can be based on the newly defined protocol. Then, the

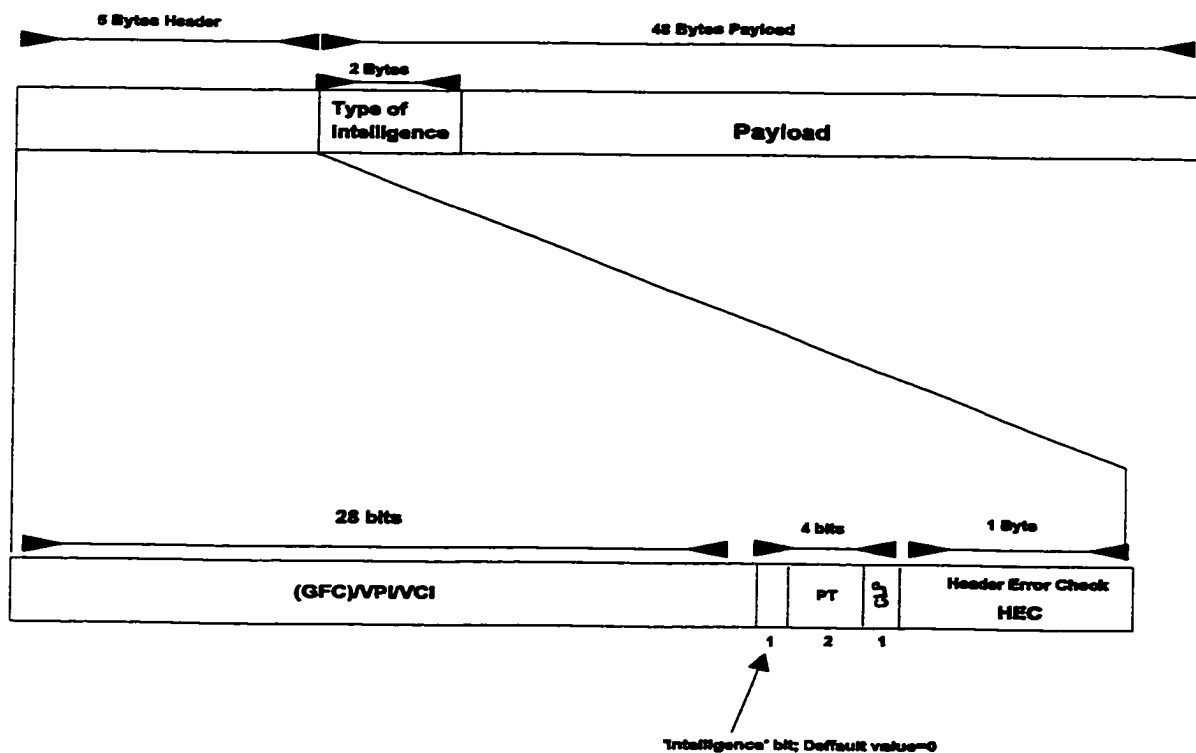
use of ATM protocol on a fiber transmission (in both transmission and signaling) will ensure a high transmission security.

Innovations in Signaling, Accessing, Switching, Routing, etc. will emerge to take the role of leadership. Wired and wireless, IN tries to deploy the ATM concept in defining the high-speed wireless mobile access system, while developing the excellent Quality of Service (QoS) guarantee solutions. By developing an open intelligent ATM platform (wired and wireless) including generic signaling system, management system, a true virtual community will be provided -where any broadband real-time information can be reachable making the words "Anytime anywhere" more real.

Today many companies are introducing ATM for routing in existing telecommunication networks supporting voice telephony and constant bit rate video services. The use ATM in such networks has proven a better service and big savings. The ATM facilities will allow B-ISDN to support greatly enhanced variety of services. This will include the possibility to integrate a wide range of existing and future services such as voice, image, text and high speed data. New techniques for data compression will need to be defined. This algorithms should be powerful enough to maintain the quality of service.

It's time for start building a public Intelligent Communication Network, and ATM networking is the technology to be used. Before then, the signaling protocol will need to be standardized to support Intelligent Services including the Intelligent Broadband Services.

The current study, assume that the reserved bit in the ATM cell is used for Intelligent services. This means that the network will distinguish between normal communication and an intelligent communication (requiring more details from one or more file servers containing control information) by checking up the reserved bit (which we can call now: the 'intelligence' bit). When the 'intelligence' bit holds a 1, then the communication or application is of the intelligent type, and the network will take the necessary steps to perform the requested application depending on the pre-defined rules.



**Figure 9.1: The use of the reserved bit in the ATM cell header as an 'intelligence' bit may open the doors to new payload types to be defined in the future.**

Once the intelligence bit is defined as a part of the ATM protocol (and more specifically a part of the payload type), the payload type will distinguish  $2^3$  possibilities instead of the current  $2^2$ . Once needed information for the type of service (as well as the possibility of completing the service), is provided from the specific file server, an ATM cell containing the type of intelligence can be send first ( ATM is connection oriented).

The same ideas can be used to implement the newly proposed knowledge systems and knowledge networks. In the case of knowledge processing systems, the ATM cell can be treated as knowledge instruction containing Knowledge Opcode and a knowledge Operand (as in computer instruction/ Opcode and Operand) to answer a knowledge query. In this case, several new elements are needed to be implemented:

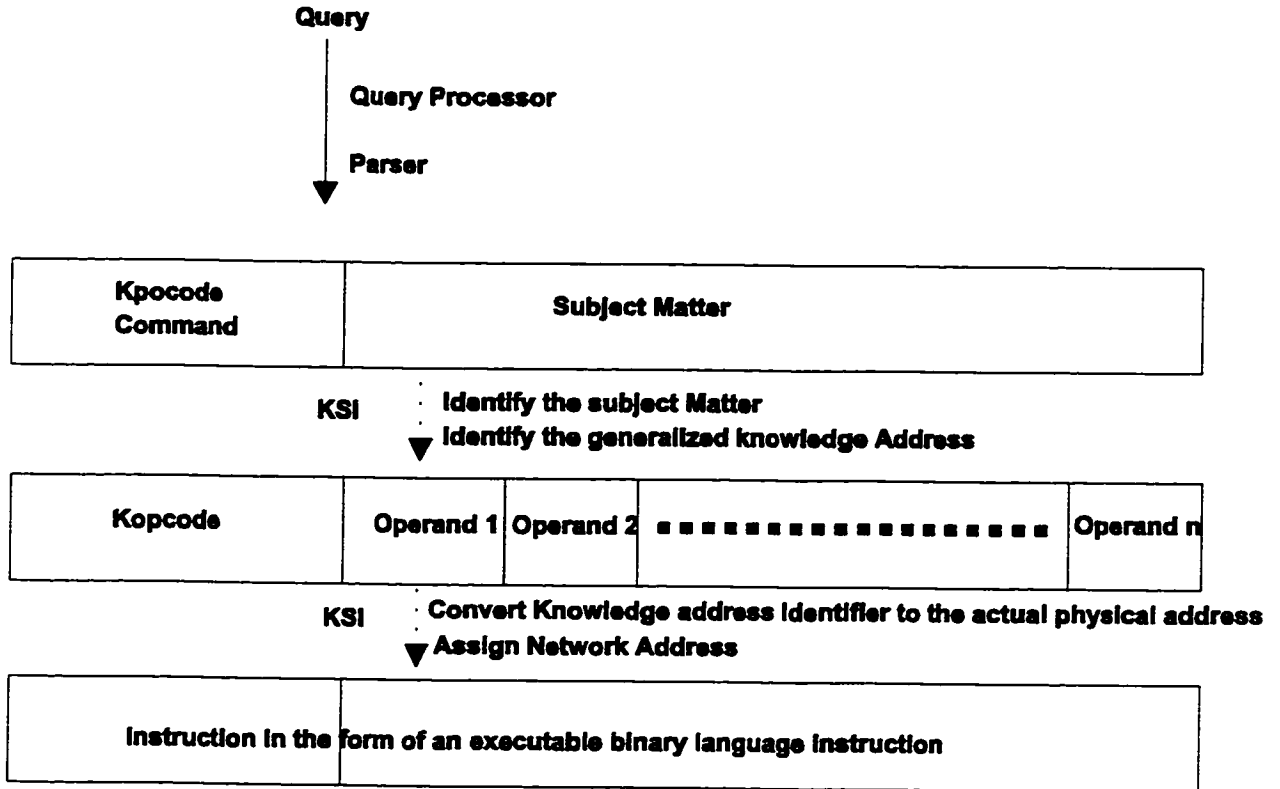
- A memory for storing knowledge in a plurality of knowledge modules in a knowledge bank, in which the knowledge includes data associated with pre-existing solutions to previous problems.
- A knowledge processing unit (KPU), operating knowledge-based software, and including a knowledge processing unit for generating a plurality of artificial intelligence based modules, and for executing the plurality of artificial intelligence base modules to process input data, including the problem related data, for generating a plurality of solution-related modules from the data representing the pre-existing solutions;

- a numerical processing unit for generating a confidential level associated with a respective solution to the current problem; and
- a database processing unit for controlling the storing of the solution-related modules in the memory

The commands in this newly invented knowledge processing system can be based on the ATM cells; the header will represent the kopcode and the payload will represent the Operand.

The Intelligent ATM Networks can also help the implementation of the Knowledge Networks which architecture will look like the IATM architecture with the capability of routing knowledge control messages by using ATM cells. The knowledge networks have been recently invented, but the signaling protocol is not defined yet. In this networks, knowledge messages (commands) can be routed through the networks to perform knowledge management (setup, maintenance, termination) and network function management functions. These commands are to be based on the cell technology where the ATM cells must be routed through the network. Thus, the network to be controlled is ATM based, and the knowledge control signaling is also to be implemented using cell technology. Let name the signaling protocol for the knowledge networks 'KSI' for Knowledge Signaling Interconnection. The KSI should define the functions that are performed in the cell-switching network, but should not dictate any hardware implementation (software to convert ATM protocol to other protocols are becoming more and more available). Knowledge signaling network can be

based on ATM cells commands. KSI may define three major entities: Knowledge Point (which can be any knowledge query point), Knowledge Transfer Points, and Interconnection Links.



**Figure 9.2: Steps to take in consideration while defining the Knowledge System Interconnection reference model.**

In this networks, a bank of super computers will have the ability to process any kind of query, knowledge or only an intelligent service request, create a cell based command, and then access to a large number of file servers through a localized ATM network for more details about the query. To answer the query, the access to remote systems may be a possibility.

In case of queries related to knowledge, a new protocol is to be defined. In the case of a query related to intelligent services, the already existent SS7 can be used, or a new Signaling System based on ATM for transport of the signals can be defined. In any of these cases, the switching system should have the ability to trigger the query, decide if the access to the file servers is a must, and connect to other systems. The following deals with answering queries related to IN services in an IN environment.

The newly proposed networks may take good advantage of the Intelligent ATM Network defined during the phase of this research. These proposed networks will be using ATM routing and ATM access for better quality of service as well as higher bandwidth. A faster processing will need to be defined taking in consideration silicon chips which continue to increase in function, power, and speed.



## GLOSSARY

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### #

<b>1B</b>	type of Switching processor.
<b>1ESS, 1A ESS, 2B ESS, 3ESS</b>	precursor ESS systems introduced before the current 4ESS and 5ESS.
<b>2B1Q</b>	Two Binary to One code for BRISDN.
<b>3B20</b>	type of specialized switching processor.
<b>4A ETS</b>	4A Toll Crossbar with Electronic Translator System.
<b>4ESS</b>	Number 4 Electronic Switching System.
<b>5 ESS</b>	a generalized Electronic Switching System for most types of CS, PS, ISDN, subscriber loop carrier and trunk switching.
<b>5ESS 2000</b>	a new 5ESS platform and concept to contend with wireless, intelligent network functions and almost all of the routine 5ESS functions.

### A

<b>AAL</b>	ATM Adaptation Layer.
<b>ACD</b>	Automatic Call Distributor.
<b>ACP</b>	Action Point.
<b>ADM</b>	Asymmetric Digital Subscriber.
<b>ADSL</b>	Asymmetric Digital Subscriber Line.
<b>ADSL/HDSL</b>	Asymmetric Digital Subscriber Line/High-Speed Digital Subscriber Line.
<b>AIN</b>	Advanced Intelligent Network.
<b>AL</b>	Application Layer.
<b>ALE</b>	ATM Local Exchange.
<b>ALI</b>	Automatic Location Identification.
<b>AM</b>	Administrative Module; also Amplitude Modulation.
<b>AMPS</b>	Advanced Mobile Phone Service.
<b>ANI</b>	Automatic Number Identifier.
<b>ANSI</b>	American National Standards Institute.

<b>AP</b>	ATM Point.
<b>ARPANET</b>	Advanced Research Project Agency Network.
<b>ASE</b>	Application Service Elements.
<b>ASN</b>	Adjunct Service Node.
<b>ASP</b>	Adjunct Service Point.
<b>ASP</b>	Advance Services Platform.
<b>ASP</b>	ATM Signaling Point.
<b>ASP</b>	Adjunct Service Processor.
<b>AT</b>	Access Tandem.
<b>AT&amp;T</b>	American Telephone and Telegraph.
<b>ATC</b>	ATM Transit Exchange.
<b>ATM</b>	Asynchronous Transfer Mode.
<b>AUC</b>	Authentication Center.
<b>AXC</b>	ATM cross-connect.
<b>AXE</b>	A Public telecommunications Switching System built by Ericsson.

## **B**

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<b>B-NT</b>	network terminator for broadband ISDN.
<b>B-TA</b>	terminal adapter for broadband ISDN.
<b>B-TE</b>	terminal equipment for broadband ISDN.
<b>B CHANNEL</b>	A 64-kbit per second bearer or information-carrying channel in the ISDN context.
<b>B1 Channel</b>	the number one B channel of BRISDN service that is used to carry 64 kbps PCM voice data for telephone service.
<b>B2 Channel</b>	the number two B channel of BRISDN service that can be used to carry packet-switched data.
<b>BECN</b>	Backward Explicit Congestion Notification.
<b>BER</b>	Bit Error Rate.
<b>BHCA</b>	Busy Hour Call Attempts.
<b>BISDN</b>	Broadband ISDN.
<b>BRISDN</b>	Basic Rate ISDN.
<b>BS</b>	Base Station.
<b>BVA</b>	Billing Validation Application.
<b>BW</b>	Bandwidth.
<b>BX.25</b>	a specialized Bellcore protocol to communicate service management information and provides an interface for maintenance, security, and operations of the SCP.

## **C**

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<b>C-Link</b>	the communications link for connecting STPs in the same region with specially connected wire pairs.
<b>CAD</b>	Computer Aided Design.

<b>CATV</b>	Cable Television.
<b>CBR</b>	Constant Bit Rate.
<b>CCIS</b>	Common Channel Interoffice Signaling.
<b>CCIS6</b>	Common Channel Interoffice Signaling 6.
<b>CCISS</b>	Common Channel Inteoffice Signaling System.
<b>CCITT</b>	International Telegraph and Telephone Consultative Committee, now called ITU-T.
<b>CCS</b>	Common Channel Signaling.
<b>CCS7</b>	Common Channel Signaling 7
<b>CCSS7</b>	Common Channel Signaling System 7.
<b>CD-ROM</b>	Compact Disk Read Only Memory.
<b>Centrex</b>	a widespread telephone-company switching service that uses Central Office switching equipment and to which customers connect via individual extension access lines.
<b>Channel</b>	a transmission path between two points.
<b>Channel Bank</b>	Channel terminal equipment used for combining (multiplexing) channels on a frequency-division or time-division basis.
<b>Circuit-mode</b>	Type of switching that causes a one-to-one correspondence between a call and a circuit.
<b>CLASS</b>	Custom Local Area Signaling Services.
<b>CLP</b>	Cell Loss Priority.
<b>CM</b>	Control Memory; also, Communication Module.
<b>CMAC</b>	Customer Mobile Access Control, in UMTS.
<b>CO</b>	Central Office; a telephone company switching facility or center, at which a diversity of transmission channels, including subscribers' local loops terminate.
<b>COS</b>	Corporation for Open Systems.
<b>CP</b>	Control Point.
<b>CPCS</b>	Common-Part Convergence Sublayer.
<b>CPE</b>	Customer Premises Equipment.
<b>CPI</b>	Customer Premises Interface (with ISDN).
<b>CPM</b>	Central Processing Module.
<b>CPU</b>	Central Processing Unit.
<b>CS</b>	Convergence Sublayer.
<b>CSDC</b>	Circuit Switched Digital Capability.
<b>CSMA/CD</b>	Carrier Sense Multiple Access/Collision Detect.
<b>CSPDN</b>	Circuit Switched Public Data Network.
<b>CT2</b>	Second-generation Cordless Telephone system.
<b>CT3</b>	Third-generation Cordless Telephone system.
<b>CTI</b>	Centre de Transit Internationale.
<b>CTN</b>	Centre de Transit nationale.
<b>CTU</b>	Centre de Transit Urbaine.

## D

<b>D CHANNEL</b>	the Delta Channel; in ISDN, a 16 kbps signaling channel for basic rate access, or a 64 kbps signaling channel with other access rates.
<b>DCE</b>	Data Circuit-terminating Equipment.
<b>DE</b>	Eligibility Bit.
<b>DECnet</b>	Digital Electronics Corporation's Network.
<b>DLCI</b>	Data Link Connection Identifier.
<b>DLL</b>	Data Link Layer.
<b>DQDB</b>	Dual Queue Dual Bus (IEEE 802.6).
<b>DS0</b>	universal 64kbps channel (or rate); same in the CEPT and Japanese digital hierarchy.
<b>DS1</b>	Digital Signal 1, a formatted signal transmitted at 1.544 Mbps.
<b>DS3</b>	DS3 Carrier Systems; the standard third level digital carrier system used in the United States at 44.736 Mbps.
<b>DS4</b>	fourth level digital carrier system used in the North America at 274.176 Mbps; also known as T4 signal (Japanese equivalency at 397.2 Mbps).
<b>DSDC</b>	Direct Services Dialing Capabilities.
<b>DSL</b>	Digital Subscriber Line.
<b>DSU/CSU</b>	Data Service Unit/Channel Service Unit.
<b>DTE</b>	Data Terminal Equipment
<b>DTMF</b>	Dual Tone MultiFrequency.
 <b>E</b>	
<hr/> <b>E1</b>	European transmission link with 2.048 Mbps of bandwidth capacity.
<b>EDSL</b>	Extended Digital Subscriber Lines.
<b>EIA</b>	Electronic Industries Association.
<b>EIA/TIA</b>	Electronic Industries Association/Telecommunication Industries Association.
<b>EIR</b>	Equipment Identifier Register.
<b>EPD</b>	Early Packet Discard.
<b>ESS</b>	Electronic Switching System.
<b>ET</b>	Exchange-Termination Equipment.
<b>EWSD</b>	An Electronic Switching System built by Siemens Corporation.
 <b>F</b>	
<hr/> <b>FCC</b>	Federal Communications Commission.
<b>FCS</b>	Frame Checking Sequence field.
<b>FCs</b>	Functional Components.
<b>FDDI</b>	Fiber Distributed Data Interface.
<b>FDM</b>	Frequency Division Multiplexing.
<b>FECN</b>	Forward Explicit Congestion Notification.

<b>FEP</b>	Front End Processor..
<b>FIFO</b>	First In First Out.
<b>FM</b>	Frequency Modulation.
<b>FO</b>	Fiber Optic.
<b>FPD</b>	First Packet Discard.
<b>FR</b>	Frame Relay.

## **G**

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<b>G.C.</b>	Global Control.
<b>Gbps</b>	Gigabits Per Second.
<b>GFC</b>	Gneric Flow Control.
<b>GHZ</b>	GigaHertz.
<b>GSM</b>	Group Special Mobile; also, Global System for Mobile Communication.

## **H**

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<b>H0-channel</b>	in ISDN, a 384-kilobits per second information carrying channel (5B+D service; also know as C6).
<b>H1-channel</b>	in ISDN, a 1.536 Mbps information-carrying channel (23B+D service, aslo known as C24)..
<b>HDLC</b>	High-Level Data Link Control.
<b>HDSL</b>	High-Speed Digital Subscriber Line.
<b>HDTV</b>	High Definition Television.
<b>HEC</b>	Header Error Control.
<b>HFC</b>	Hybrid Fiber Coax.
<b>HL</b>	Higher layer of the OSI model.
<b>HLR</b>	Home Location Register.

## **I**

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<b>IATMN</b>	Intelligent ATM Networks.
<b>IDB</b>	INWATS Data Base.
<b>IM&amp;M</b>	Information Movement and Management.
<b>IN</b>	Intelligent Network.
<b>IN/1</b>	The initial Bellcore release of the intelligent network.
<b>IN/1+</b>	The evolution of the IN/1 architecture and some IN/1 elements.
<b>IN/2</b>	The final architecture of an intelligent network.
<b>INAP</b>	Intelligent Network Access Point.
<b>INWATS</b>	INwards Wide Area Telecommunication Service.
<b>IOP</b>	Input/Output Processor.
<b>IP</b>	Intelligent Peripheral.
<b>ISDN</b>	Integrated Services Digital Network.
<b>ISO</b>	International Organization for Standards.
<b>ISUP</b>	ISDN User-Part.
<b>ITU-T</b>	International Telegraph and Telephone Consultative Committee.

**IXC** Interexchange Carriers.

## **K**

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**KB** Knowledge Base; also, Knowledge Bank.  
**KCP** Knowledge Control Point.  
**KD** Knowledge Domain.  
**KLI** Knowledge Logic Interpreter  
**KM** Knowledge Module; also, Knowledge Machine.  
**KMS** Knowledge Maintenance System or Knowledge Management System.  
**Kopcode** Knowledge Domain Opcode  
**KQP** Knowledge Query Point.  
**KQR** Knowledge Query Ring.  
**KR** Knowledge Ring.  
**KTP** Knowledge Transfer Point.

## **L**

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**L.C.** Local Control.  
**LAN** Local Area Network.  
**LAP** Link Access Procedure/Protocol.  
**LAPB** Link Access Procedures-balanced.  
**LAPD** Link Access Protocol for the D channel.  
**LASS** Local Area Signaling Service.  
**LATA** Local Access and Transport Area.  
**LEC** Local Exchange Carriers.  
**LIDB** Line Information Database.  
**LT** Line-Termination.  
**LTE** Line Terminating Equipment.

## **M**

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**MAC** Media Access Control.  
**MAN** Metropolitan Area Network.  
**MBES** Machine Based Educational Network.  
**MBG** Multifrequency Business Group.  
**MCPN** Mobile Customer Premises Network.  
**MF** MultiFrequency.  
**MHZ** Million Hertz.  
**MPEG** Motion Picture Experts Group.  
**MPU** Medical Processor Unit.  
**MS** Mobile Station.  
**MSC** Mobile Switching Center.  
**MTP** Message Transfer Part.  
**MTS** Mobile Telephone Service.

**MTSO** Mobile Telephone Switching Office.

## **N**

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**N-ISDN** Narrow Band ISDN.  
**NAP** Network Access Point.  
**NCP** Network Control Point.  
**NCP** Network Service Part.  
**NCTE** Network Channel Termination Equipment.  
**NE** Near End.  
**NID** Network Information Database  
**NL** Network Layer.  
**NM** Node Manager.  
**NNI** Network-to-Network Interface.  
**No. 1 ESS** the number one ESS switching System.  
**NRM** Network Resource Manager.  
**NSCX** Network Service Complex,  
**NSP** Network Service Control Point Corresponding to a SCP.  
**NT** Network Termination,  
**NT1** Network Termination 1, which typically consists of the network channel termination equipment and is located between the U and T reference points at the customer premises.  
**NT2** Network Termination 2, Which typically consists of a PBX and/or a terminal controller following the T interface.  
**NYNEX** New York telephone operating company (now merged with Bell Atlantic).

## **O**

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**OA&M** Operations, Administration, and Maintenance.  
**OAM&P** Operations, Administration, Maintenance, and Provisioning.  
**OC** Optical Carrier.  
**OC-n** Optical Carrier nth level of multiplex.  
**OE&M** Operations, Engineering, and Maintenance.  
**OH** Overhead.  
**ONU** Off Network Unit.  
**OSI** Open System Interconnection.  
**OSI-RM** Open System Interconnection- Reference Model.  
**OSN** Operations System Network.  
**Out-of-Band Signaling** Signaling outside the frequency band; opposite of in-band signaling.

## **P**

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**PA** Physical Access.  
**PABX** Private Automatic Branch Exchange.

<b>PBX</b>	Private Branch Exchange.
<b>PCM</b>	Pulse Code Modulation.
<b>PCN</b>	Personal Communication Network.
<b>PCS</b>	Personal Communication Services.
<b>PHY</b>	Physical layer Protocol.
<b>PLCP</b>	Physical Layer Convergence Protocol.
<b>POH</b>	Payload Overhead
<b>POTS</b>	Plain Old Telephone.
<b>PSN</b>	Public Switched Network.
<b>PSN</b>	Packet Switched Network; also, Public Switched Network.
<b>PSPDN</b>	Packet Switched Public Data Network.
<b>PSTN</b>	Public Switched Telephone Network.
<b>PT</b>	Payload Type.
<b>PVN</b>	Private Virtual Networks.

## **Q**

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<b>Q.700</b>	A standard Series of protocol definitions published by CCITT (now ITU-T).
<b>Q.900</b>	A standard Series of protocol definitions published by CCITT (now ITU-T).
<b>QoS</b>	Quality of Service.

## **R**

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<b>R, S, T, U</b>	the four ISDN reference Points at the customer premises.
<b>RAM</b>	Random Access Memory.
<b>RBOCs</b>	Regional Bell Operating Companies.
<b>RC</b>	Regional Center.
<b>RF</b>	Radio Frequency.
<b>RG</b>	Cable Type.
<b>RHC</b>	Regional Holding Companies.
<b>RS-232</b>	An EIA standard for connecting business machines with serial interfaces.
<b>RT</b>	Remote Terminal.

## **S**

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<b>S/N</b>	Signal to Noise.
<b>S-AAL</b>	Signaling.
<b>SAP</b>	Service Action Point.
<b>SAR</b>	Segmentation and Reassembly.
<b>SCCP</b>	Signaling Connection Control Part.
<b>SCE</b>	Service Creation Environment.
<b>SCN</b>	Service Circuit Nodes.
<b>SCN</b>	Service Circuit Node.

<b>SCP</b>	Service Control Point.
<b>SDDN</b>	Software Defined Data Network.
<b>SDH</b>	Synchronous Digital Hierarchy.
<b>SDLC</b>	Synchnous Data Link Control.
<b>SDN</b>	Software Defined Network.
<b>SF</b>	Single Frequency.
<b>SIP</b>	SMDS Interface Protocol
<b>SL</b>	Session Layer.
<b>SLC</b>	Subscriber Loop Carrier.
<b>SLEE</b>	Service Logic Sexecution Environment
<b>SLI</b>	Service Logic Interpreter.
<b>SM</b>	Switching Module.
<b>SMDS</b>	Switched Multimegabit Data Service.
<b>SMS</b>	Service Management System.
<b>SMS/2</b>	Service management System/2.
<b>SN</b>	Service Node.
<b>SNA</b>	Systems Network Architecture (from IBM).
<b>SNA/ATM</b>	Systems Network Architecture / ATM Cell Relay.
<b>SNA/FR</b>	Systems Network Architecture /Frame Relay.
<b>SONET</b>	Synchronous Optical Network.
<b>SPC</b>	Stored Program Control.
<b>SPE</b>	Synchronous Payload Envelop.
<b>SPM</b>	Subsciber Processing Module.
<b>SRAM</b>	Stack Random Access Memory.
<b>SS7</b>	Signaling System 7.
<b>SSCS</b>	Service-Specific Convergence Sublayer.
<b>SSP</b>	Service Switching Point.
<b>SSP/2</b>	Service Switching Point/2.
<b>STM</b>	Synchronous Transport Module.
<b>STM-I</b>	Synchronous Transport Module level I (I=N/3=1, 4, 16, 64).
<b>STP</b>	Signaling Transfer Point.
<b>STS</b>	Synchronous Transport Signal/System.
<b>STS-N</b>	Sysnchronous Transport Signal level N ( N= 1, 3, 12, 48, or 192).

## **T**

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<b>T1</b>	A digital carrier used to transmit a DS-a formatted digital signal at 1.544 Mbps, also called T1 carrier system.
<b>T1 or E1</b>	Priamary rates for North American or European environment (see DS-n levels).
<b>TA</b>	Terminal Adapter.
<b>TDM</b>	Time Division Multiplex.
<b>TDMA</b>	Time Division Multiplex Access.
<b>TE</b>	Terminal Equipment.
<b>TE1</b>	Terminal Equipment specifically meeting ISDN equipments.

<b>TE2</b>	Terminal equipment type 2, not specifically meeting ISDN requirements, which is typically connected to the network at the R interface reference point separated from the S reference point TA.
<b>TELENET</b>	A terminal Emulation protocol or an application program that permits the logging option into another computer system.
<b>TIA</b>	Telecommunication Industry Association.
<b>TOH</b>	Transport Overhead.
<b>TPM</b>	Trunk Processing Module.
<b>TUP</b>	Telephony User Part
<b>TWP</b>	Twisted Wire-Pair.

## **U**

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<b>UISN</b>	Universal Information Service Network.
<b>UNI</b>	User-to-Network Interface.

## **V**

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<b>VBR</b>	Variable Bit Rate.
<b>VCC</b>	Virtual Channel Connection.
<b>VCI</b>	Virtual Channel Identifier.
<b>VCR</b>	Video cassette Recorder.
<b>VFN</b>	Vendor Feature Node.
<b>VHSDN</b>	Very High Speed Digital Network.
<b>VLR</b>	Visitor Location register.
<b>VPI</b>	Virtual Path Identifier.
<b>VSM</b>	Virtual Switching Machine.
<b>VT</b>	Virtual Terminal..

## **W**

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<b>WAN</b>	Wide Area Network.
<b>WATM</b>	Wireless ATM.
<b>WIN</b>	Wireless intelligent Network.

## **X**

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<b>X.21</b>	Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for synchronous operation on public circuit-switched networks.
<b>X.25</b>	A CCITT (now ITU-T) recommendation that specifies the interface between user data terminal equipment (DTE) and packet-switching data circuit-terminating equipment (DCE).
<b>X.28</b>	DTE/DCE interface for start-stop mode data terminal equipment accessing the packet assembly /disassembly facility (PAD) in

- a public data network situated in the same country.
- X.29** Procedures for the exchange of control information and user data between a packet assembly/disassembly facility (PAD) and a packet-mode DET or another PAD.
- X.75** Terminal and Transit call-control procedure and data transfer mechanisms on (typically) international circuits between packet-switched data networks.

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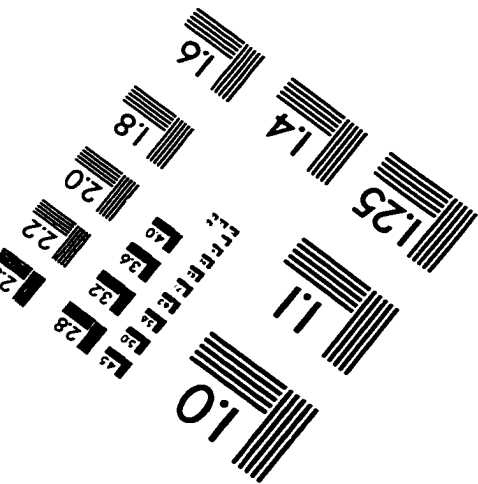
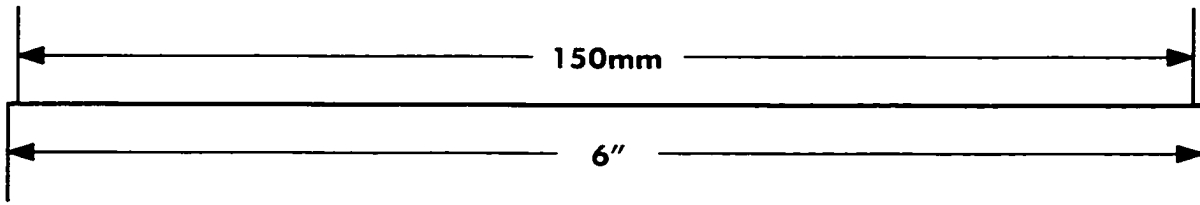
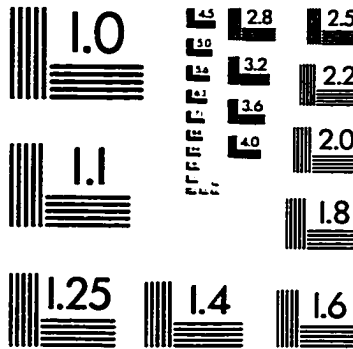
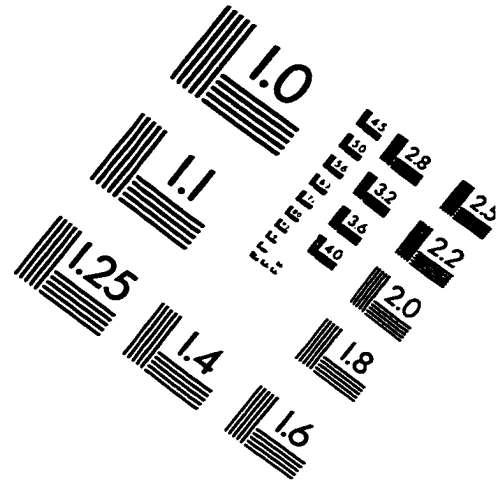
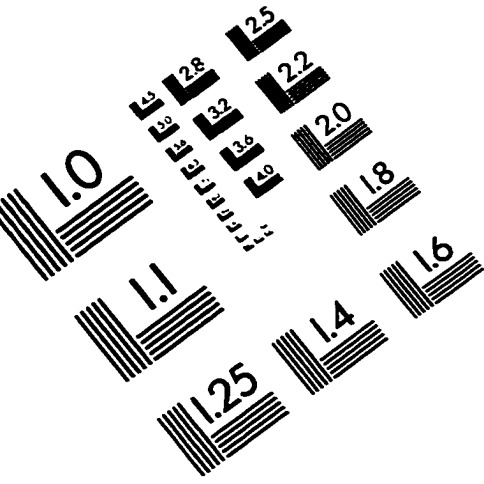
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