CHAPTER 3

Switching and Signaling Systems

3.1 Evolution of Switching Systems

Subsequent to the invention of the telephone in 1876, by 1884 approximately 350,000 phones were in use, interconnected by a plethora of manual switchboards. Suspicions that the operators at switchboards were diverting the undertaker A. B. Strowger's potential business to competitors led him to develop the first automatic step-by-step exchange in around 1896. This was based on pulse dialing from the rotary phone handset. Variations of this type of switches proliferated and were widely in use up to the mid 1950s. With transistorized circuits in the late 1940s, tone-based signaling became a reality. More advanced control systems and common control of switching arrays progressed with digital control. As a result, crossbar systems and reed relay–based switches proliferated. In all of these systems, the basis of switching was to have a pair of wires selectively coupled to another pair of wires by mechanical means to complete a speech path between subscribers.

As digital hardware and processing techniques matured, engineers saw the advantage of placing a computer subsystem for common control of a switching subsystem. Hence, the stored program control (SPC) of the telephone switching systems evolved. The first SPC exchange trials were by the Bell Labs in around 1958. SPC systems were conducive to providing many special services such as call waiting, call forwarding, and specialized billing. By the early 1970s, there were large exchanges with capacities around 150,000 lines, processing over 750,000 calls per hour during peak periods. While the SPC subsystems provided better and more efficient call processing, the switching within an exchange core and the long-distance transmission systems remained predominantly analog. Crossbar-type exchanges with SPC resulted in reliable switches with lower maintenance. This in turn resulted in services growing rapidly, irrespective of local and national boundaries, via cable, submarine, and satellite transmission.

With semiconductor memories, digital components, and processing techniques becoming cost effective, switching system designers were able to develop the PCM-based fully digital switching systems. In these systems, the processor subsystem was able to tackle the signaling process as well as the switching of voice paths by organized exchanges of PCM voice samples. Such fully digital exchanges came into operation in the late 1970s.

Over the 1980s and 1990s most of the telecommunications systems providers were able to replace older types of exchanges with digital switching systems. This period also saw vast developments related to digital transmission systems. During the same period, various data communication applications made use of a mix of the telephone network's infrastructure as well as dedicated data networks.

Over 800 million subscribers worldwide use the PSTN today. During 1980 to 2000, there were rapid developments in mobile, Internet, multimedia, and broadband, services, which changed the overall scenario of telecommunication systems. This chapter provides an overview of digital switching systems and associated concepts such as signaling as applied to modern telecommunication systems. For more details on the historic development of switching systems, [1–4] are suggested.

3.2 Essentials of Switching, Signaling, and Control

Today's PSTN reference model can be simply depicted, as in Figure 3.1(a). Figure 3.1(b) indicates the same situation dividing the network into three layers: access, service, and infrastructure.

In modern switching systems, three basic concepts—signaling, switching, and control—are integrated to provide a physical or a virtual path between the subscribers for a communication session. In simple terms, a subscriber connected to the access network indicates the desire to access the network (for example, a telephone subscriber listens to dial tone and then dials digits) and then gets connected to another subscriber or a location with the aid of signaling. The nearest local switch, which maintains specific information about the particular subscriber, interprets the signaling information and creates a path to another subscriber, distance exchange, or a site using the concepts of switching. Modern switching concepts can be primarily divided into circuit switching and packet switching.

Modern switches are designed around a real-time processor-based control system, operating as the central element that controls the subscriber subsystem (access subsystem), switching subsystem (switch fabric through which the physical or logical connection is made), and the access to transport networks.
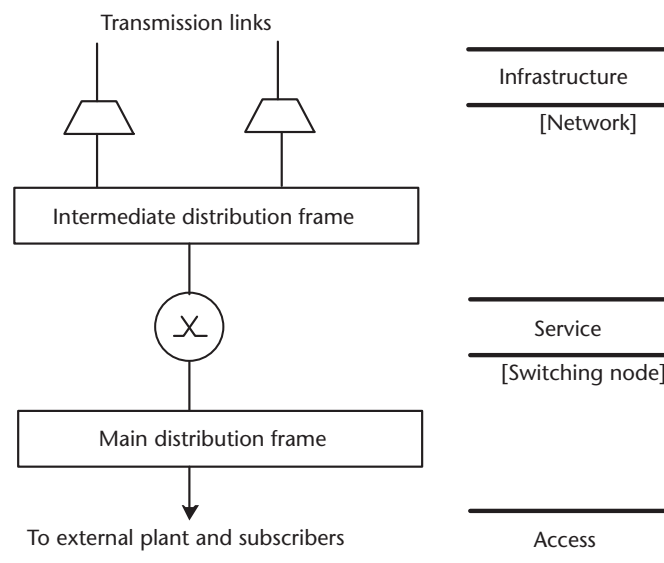

Figure 3.1   Three layers of the PSTN.

3.3   **Principles of Switching**

Early manual systems had the operator as its central controller. Manual control gradually evolved into fully automatic SPC systems. The processor system in an SPC exchange controls a complex set of switching matrices of wire pairs. In this process, various elements such as rectangular cross-point arrays and multistage switching modules are used. The simple principle maintained throughout the process was that there was always a physical path for a copper pair from the access network to get connected to another copper pair either within the local access area or to a distance exchange.

The principle of a single rectangular crosspoint array and multistage switching matrices as examples are shown in Figure 3.2. Figure 3.2(a) indicates the case of N inlets and M outlets. Figure 3.2(b) shows a multistage case for N inlets and N outlets. The first stage consists of multiple $n \times k$ matrices, coupled to a second stage of k matrices with N/n inputs and N/n outputs each (called the junctors) and finally connected to the third stage with $k \times n$ matrices. More details of such techniques with important parameters such as blocking, blocking probabilities, paths, and path
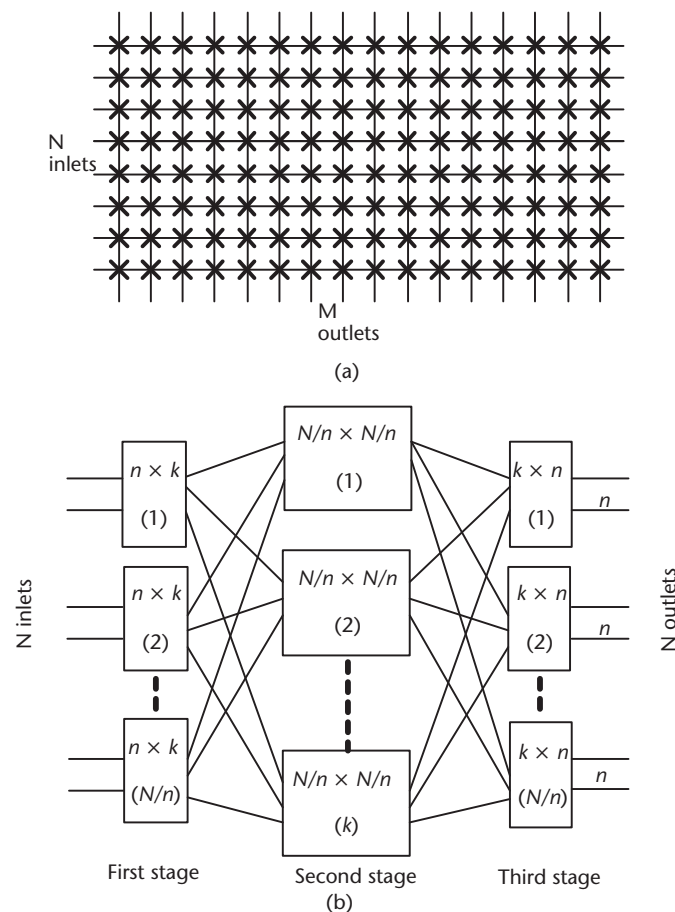
Figure 3.2   Switching matrices: (a) rectangular cross point array, and (b) three-stage switching matrix.

finding times are available in [4, 5]. This concept of having a physical path via a switch matrix is called *space division switching.*

The maturing of SPC techniques while semiconductor memories became less expensive allowed designers to adopt PCM to digitize the voice signals and store the samples in a semiconductor memory. With processor blocks controlling the overall operation in real-time, consecutive PCM, coded voice samples from a bank of memories (incoming voice) could be transferred to the output stage with suitable interchange of time slots. Selective switching of voice channels were thus reduced to a write-select-read sequence of PCM samples in a time switch interchange (TSI) module.

3.3.1 Time and Space Switching

By the early 1980s, the TSI concept had evolved well into practical switching systems. A situation with the multiplexer, demultiplexer, and associated memories is shown in Figure 3.3. Voice samples in the PCM bit stream from the multiplexer are received into register *A* and written into the speech store (SS) sequentially. Signaling related to the conversation updates the control store (CS). This in turn provides the read address sequence for reading the contents of the SS to register *B,* coupled to the outgoing bit stream, which separates into the individual voice channels after the demultiplexer. The time slot counter controls the timing related to the writing and reading processes.
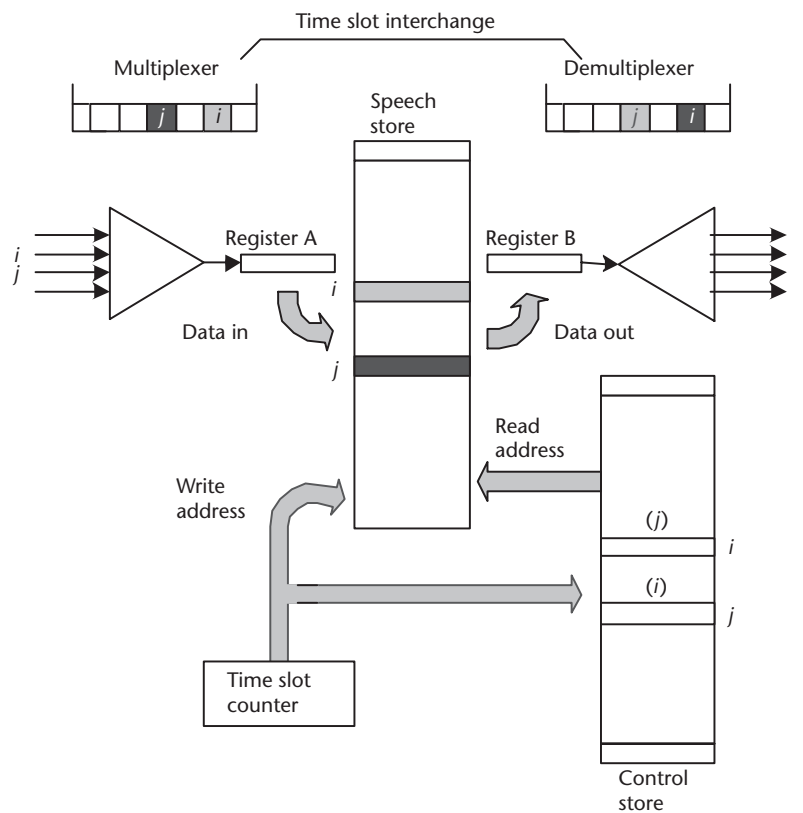

**Figure 3.3** TSI with multiplexer/demultiplexer and memory switch.

Because a write and a read are required for each channel (sampled every $125\,\mu$s) entering and leaving the memory, the maximum number of channels N that can be supported by a single TSI is $125/2t_c$, where t_c is the memory cycle time in microseconds.

Figure 3.3 clearly indicates that the speed of the memory chips govern the number of channels that can be switched on a nonblocking basis. In general, *time switch* modules can be configured as sequential write/random read or vice versa, as in Figure 3.4(a), showing a case of time slots 5 and 8 getting exchanged in sequential write/random read mode.

Similar to the time switch, a *space switch* can be configured as in Figure 3.4(b) with $n \times n$ switching points designed as electronic gates. Each column in the matrix is controlled by a CS, which is used to indicate the relationship between incoming time slots and the respective outgoing connections that allow the distribution of incoming time slots to various outgoing paths. Similarly, the time slots in incoming and outgoing T1/EI links in transit exchanges can be redistributed. The overall effect of time and space switching is illustrated in Figure 3.4(c). A *group switch* (GS) is a combination of time and space switches.

3.3.2   Multidimensional Switching

Large, versatile digital exchanges require combinations of space switch modules (SPM) and time switch modules (TSM). A large exchange could have combinations such as time-space (TS), space-time (ST), time-space-time (TST), and space-time-space (STS) switches. Figure 3.5(a) and (b) indicate TS and STS examples, respectively. It is important to realize that duplex voice requires bidirectional paths with corresponding elements in the SS in each direction.
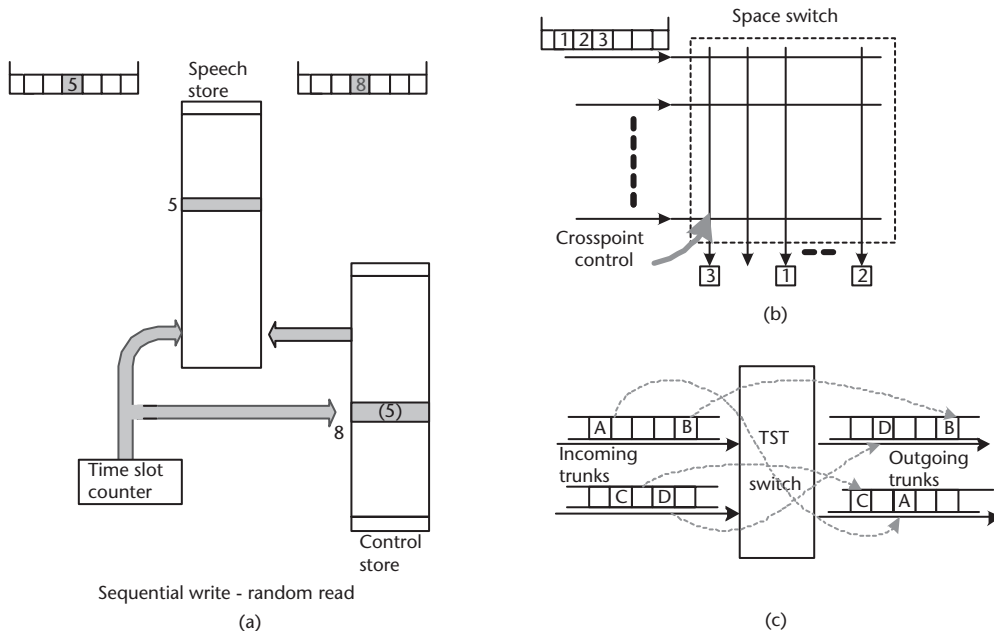
Figure 3.4   Switch modules: (a) time switch modules, (b) space switch, and (c) effect on the incoming and outgoing streams.
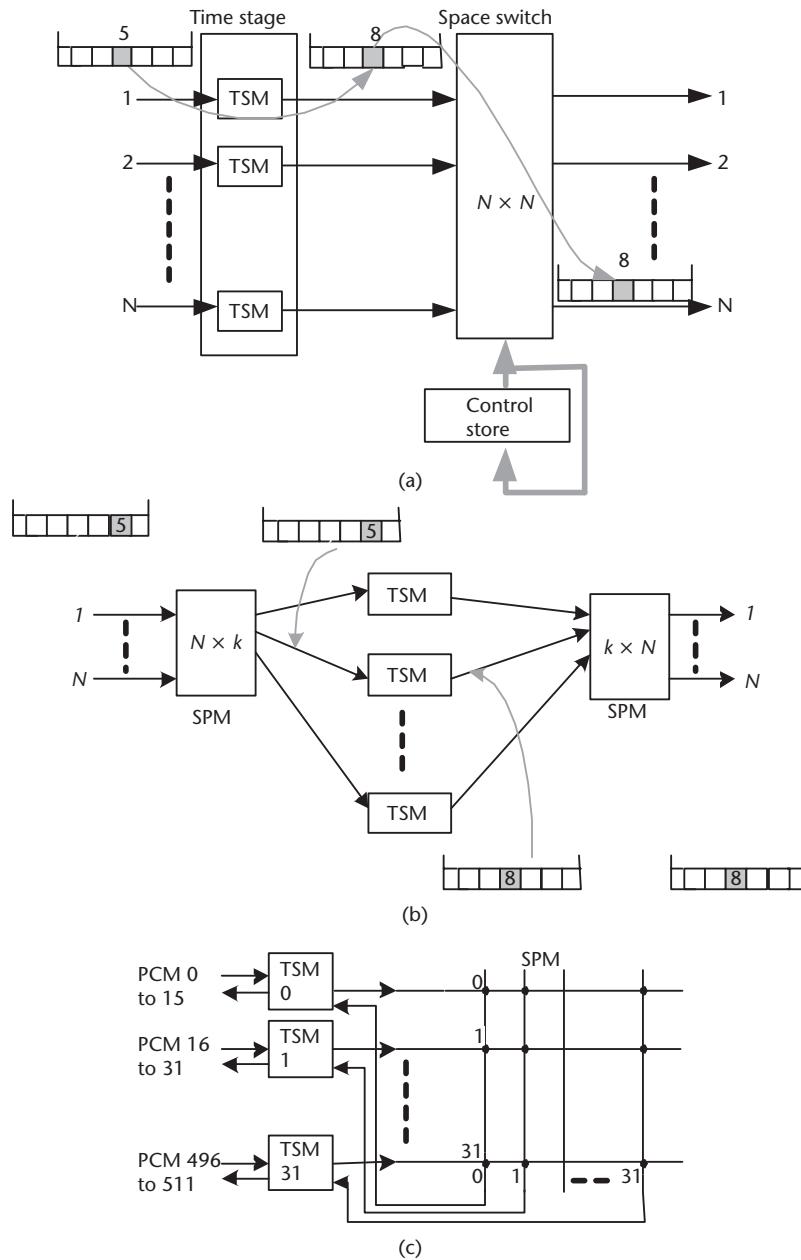
Figure 3.5　Multidimensional switches: (a) TS switch, (b) STS switch, and (c) TST with multiple PCM connections.

Figure 3.5(c) shows a typical group switch, having a TST switch with 32 TSMs and a 32 × 32 SPM. Each TSM can receive 16 PCM links giving 16 × 32 = 512 multiple positions (MUP). Thirty two such modules connected to an SPM provide a capacity of 512 × 32 = 16,384 MUPs. Because a telephone call always seizes two inputs and two outputs, such a group switch can theoretically switch 8,192 simultaneous calls. In practice this number will be less due to signaling and recorded announcements occupying some time slots. More details are available in [6].

3.4   Circuit Switching and Packet Switching

In telecommunications systems, three types of switching are generally possible—circuit switching, message switching, and packet switching.

Circuit switching is the case where a physical channel or its equivalent is established between the two parties, and it remains until the call is completed. Even if there is no conversation present, such as in the case of silent periods in a duplex conversation, an equivalent channel remains end to end until the call is completed. Selection of an appropriate path through the public systems is based on signaling between the two subscribers.

In message switching, a public network carries the entire message from one end to the other. A classic example is telegraphy. Though these systems are nearly extinct in public telecommunications networks, they are used in the aviation industry.

In packet switching, data to be transmitted is broken down into segments, or packets, and each packet is *enveloped* with address and other overhead information to ensure that it travels to the correct destination without errors. The packet travels across the network on a store-and-forward basis until it reaches the final destination. Packet-switched networks use the capacity only when there is something to be sent. This way, all available capacity can be filled with packets. Packet switching is more suited for data communication, where error-free transmission is important and variable delays are tolerable. Circuit switching is traditionally used for voice systems, where presence of errors is acceptable up to a limit but where real-time communication is a must.

3.4.1   Basics of Packet Switching

Figure 3.6(a) depicts a packet-switched network. The store-and-forward nodes are interconnected to each other via data links in a mesh. The nodes are processor systems that are capable of many tasks, such as receiving and transmitting files from the information sources, breaking files into smaller units, and forming packets with headers and trailers. This overhead information provides error checking for error-free transmission of packets, routing information regarding the available paths with alternatives, and priority information.

3.4.1.1   Packetizing

The format of a packet varies from one network to another. In the early packet-switched networks, the assumptions were that links were slow, subject to errors, and expensive. In such cases, the subscriber data is supplemented with many additional bits for control and error handling, resulting in increased overhead. In modern systems, where the transmission links are faster and error free, such overhead could be relatively small. Figure 3.6(b) indicates a typical packet format. A header contains destination and source addresses, an operation code (indicating whether the packet is data or control information), sequence information (to help assembly of packets at the destination), and a byte count. The trailer contains an error-detecting code, such as cyclic redundancy check (CRC). The delivery of packets is associated with an acknowledgment procedure, which allows the sender to
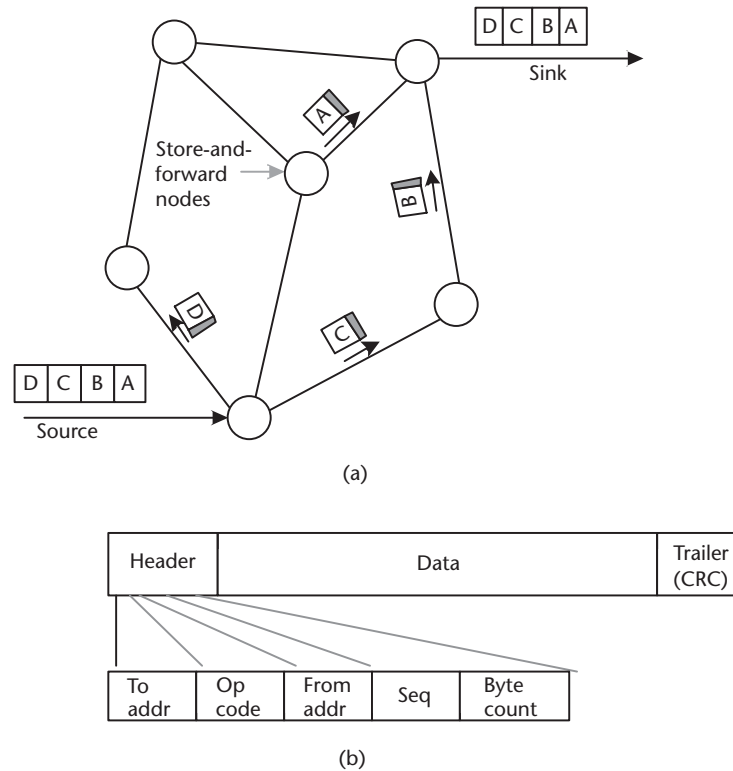
Figure 3.6 Packet switching concepts: (a) simplified network, and (b) typical packet format.

determine whether a packet has been delivered to the destination. If a packet has not been received correctly by the destination, a retransmission is initiated following one of several algorithms.

3.4.1.2   Virtual Circuits and Datagrams

The switching of packets may be via two basic methods, as *virtual circuits* or as *datagrams*. Virtual circuits adopt basic aspects of both circuit and packet switching. The transmission capacity is dynamically assigned on an "as needed" basis, but all packets of a multipacket information file follow the same route through the network. Before interactive communications begin, a route is established through the network and all participating nodes are informed of the connection path. The path exists only for the duration of the data transfer. These are also referred to as *switched virtual circuits* (SVCs), or *connection-oriented* packet switching. In essence, a virtual circuit is a logical concept involving addresses and pointers in the nodes of the network, but with no dedicated transmission resources.

A *permanent virtual circuit* (PVC) can also be established, where successive sessions between any two end points always use the same path. An attractive feature is the absence of the call-establishment phase of the virtual circuit, while the vulnerability to node or link failures is a disadvantage.

In datagram packet switching, a virtual circuit is not established prior to communication. Packets are switched individually based on their destination address.

Hence, packets corresponding to a single communication may travel via different paths over the network and may arrive out of order. The advantages are the absence of a call-establishment phase and the ability to route packets, avoiding network congestion and failures. This technique is called *connectionless* packet switching.

3.4.1.3   Routing

Routing is an important concept used in packet-switching systems. It enables a suitable path to be found for delivery of packets to the destination. The best routes are determined by switches based on criteria related to factors such as network traffic, link costs, and delays. All routing techniques allow for a certain amount of adaptation or dynamic routing, circumventing line or node failures and responding quickly to network failures or congestion.
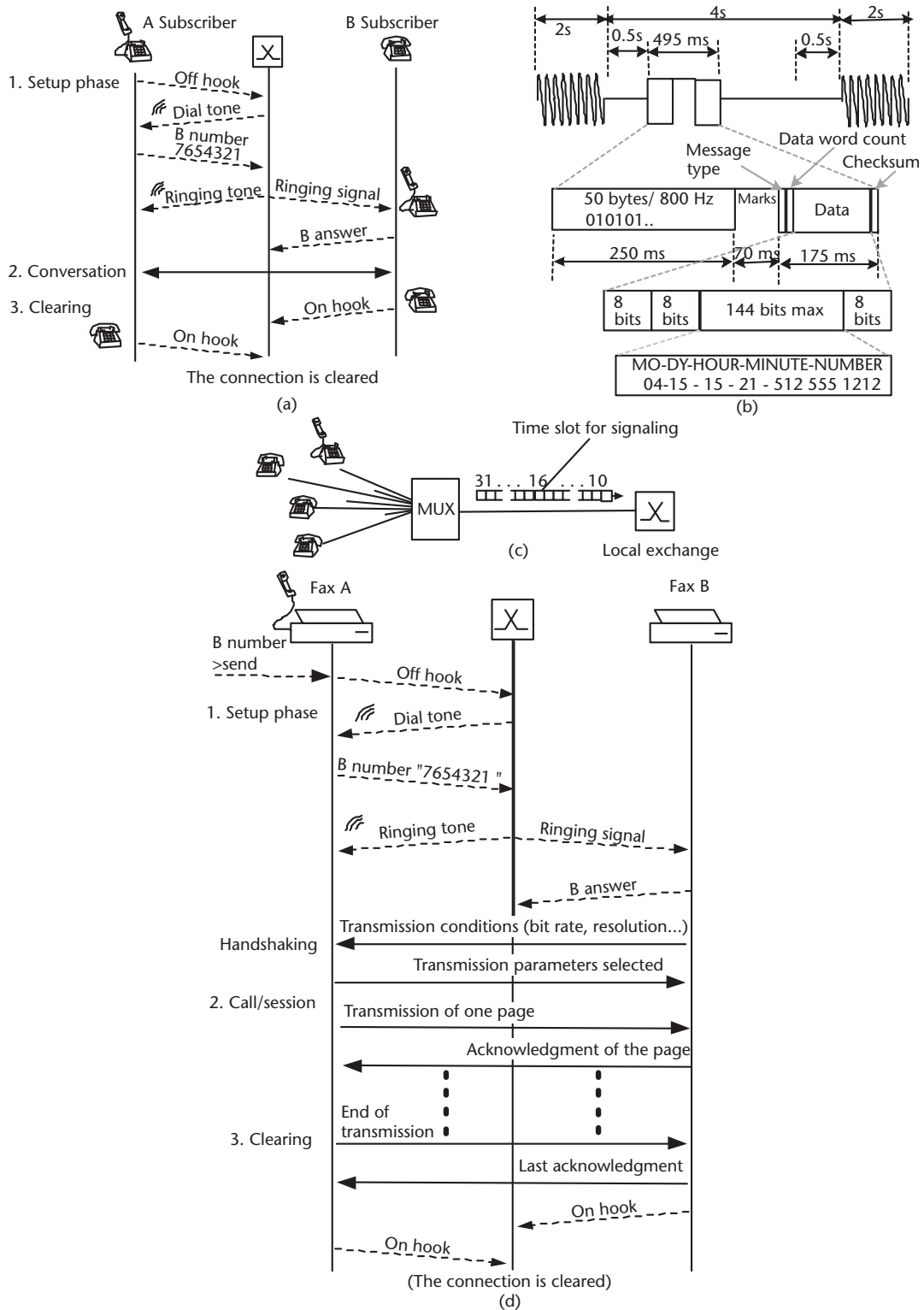
3.4.1.4   Flow Control

Routing and flow control are two closely related operational requirements of any communications network. The same basic principle for controlling the flow in circuit switched network also applies to packet networks. Whenever the network is experiencing stress due to loss of capacity or heavy demand for services, new service requests must be blocked at the periphery before they tie up the common resources at nodes, the store and forward buffers, and the transmission links. Flow control in a packet network is primarily concerned with buffer management. Transmitted packets are held in a buffer until acknowledgments are received. Flow control procedures ensure that these buffers do not overflow in the event of a network problem. For more details on the basics of packet switching, [7] is suggested.

3.5   Signaling Systems

Signaling functions of a communications network refer to the means for transferring network-related control information between various terminals, nodes, and users. Signaling functions can be broadly categorized into two primary types: supervisory and information bearing. Supervisory signals convey the status or control information of the network elements. Obvious examples are off-hook, dial tone, ringing, busy tone, and call terminating. Information bearing signals are calling/called party addresses and toll charges.

3.5.1   Subscriber Signaling in the PSTN

A simple example of signaling between the subscriber and the local exchange (LE) is shown in Figure 3.7(a). The process starts with supplying the dial tone to calling subscriber (*A*), followed by the number-sending process until the exchange detects the condition of the called subscriber (*B*). If *B* is free, the ringing tone and a ringing signal is sent to each party until *B* answers. This commences the conversation process. When one of the parties places the receiver on-hook, a clearing signal is sent and the call ends. When the calling line identification (CLI) service is present, *A*'s

A Subscriber B Subscriber

1. Setup phase
Off hook
Dial tone
B number 7654321
Ringing tone   Ringing signal
B answer

2. Conversation

3. Clearing
On hook
On hook
The connection is cleared
(a)

4s 2s
2s 0.5s 495 ms 0.5s

Message type   Data word count   Checksum

50 bytes/ 800 Hz 010101.. Marks Data

250 ms 70 ms 175 ms

8 bits | 8 bits | 144 bits max | 8 bits

MO-DY-HOUR-MINUTE-NUMBER
04-15 - 15 - 21 - 512 555 1212
(b)

Time slot for signaling

31 . . . 16 . . . 10

MUX

(c) Local exchange

Fax A Fax B

B number
>send
Off hook

1. Setup phase
Dial tone

B number "7654321 "

Ringing tone   Ringing signal

B answer

Transmission conditions (bit rate, resolution...)

Handshaking
Transmission parameters selected

2. Call/session
Transmission of one page

Acknowledgment of the page

End of transmission

3. Clearing
Last acknowledgment

On hook

On hook

(The connection is cleared)
(d)

Figure 3.7 Signaling examples in voice systems: (a) establishing a basic voice connection, (b) a typical CLI signal, (c) signaling between a subscriber multiplexer and an LE, and (d) fax communication.

number and other identifying information is sent typically using DTMF or FSK before or during the ringing period. Figure 3.7(b) shows a typical CLI signal format. When metering pulses are delivered from the exchange to subscribers where call meter services are used, exchange equipment connects an oscillator sending 12- or 16-kHz pulses to the subscriber end. In this case, filters are used to block the signals outside the voiceband from the subscriber and the exchange end voice equipment.

When a T1/E1 link is used between a multiplexer and an LE, a time slot or a bit in the voice time slot of each subscriber is used for signaling [see Figure 3.7(c)]. Chapter 7 provides further details on this.

Figure 3.7(d) indicates the simplified example of fax signaling. After the basic procedure of signaling between *A* and *B* to establish the call, and after *B* answers, the two faxes enter into the process of handshaking. This process exchanges several parameters to determine the transfer rate. A Group 3 fax machine can have a built-in modem with automatic bit-rate adaptation. The transfer rate will depend on the quality of the voice line, as described in [5].

3.5.2   Interexchange Signaling in the PSTN

If more than one LE is involved in setting up a connection or in activating a supplementary service, these exchanges must interchange information. Several systems for interexchange signaling are described in [6]. Two basic forms—*in-channel* or *channel associated signaling* (CAS) and *common channel signaling* (CCS)—are present. CAS (sometimes referred to as *per-trunk signaling*) uses the same transmission facilities or channel for signaling and voice. CCS uses one channel for all signaling functions of a group of voice channels. While most older systems use CAS, modern networks use CCS.

CAS systems can be further subdivided into inband and out-of-band techniques. While the former uses voiceband frequencies, the latter uses a range of frequencies outside the voiceband. CAS is further divided into subsets such as line signaling and register signaling [6, 8].

CCS is generally referred to as the *signaling system number* 7 (SS7) standardized by ITU-T. CCS requires a separate network as a bearer service for signaling only. The processor systems extract the signaling information relevant to groups of calls together with other network conditions and routes them through a separate data link between switches. The signaling network is an integral part of the overall network. Its task is to support other networks. It can carry traffic transparently in the form of short messages between exchanges.

3.5.3   An Overview of ITU-T SS7

Figure 3.8 indicates the way SS7 operates within the modern network, including the PSTN, ISDN, and the PLMN, carrying signaling message packets or message signal units (MSU) between processors in the network.

The SS7 uses the seven-layer open systems interconnect (OSI) concept as depicted in Figure 3.9(a). The protocols for processor communication reside above the OSI layer 3 and are called user parts (UP) and application parts (AP). As shown
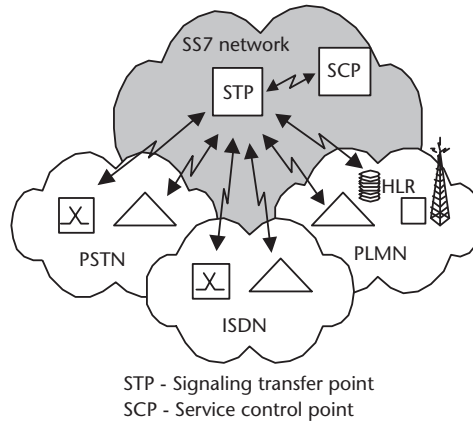
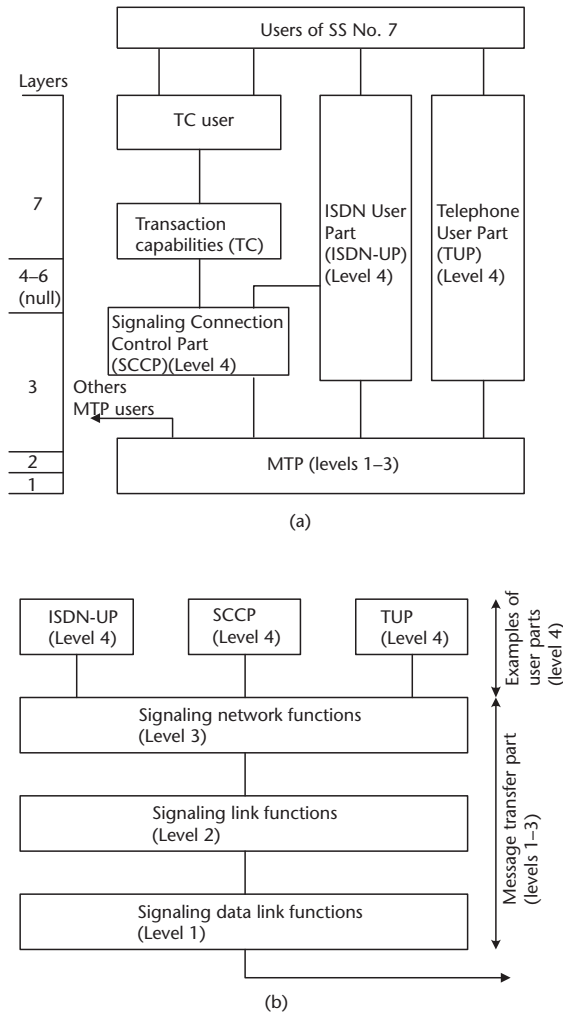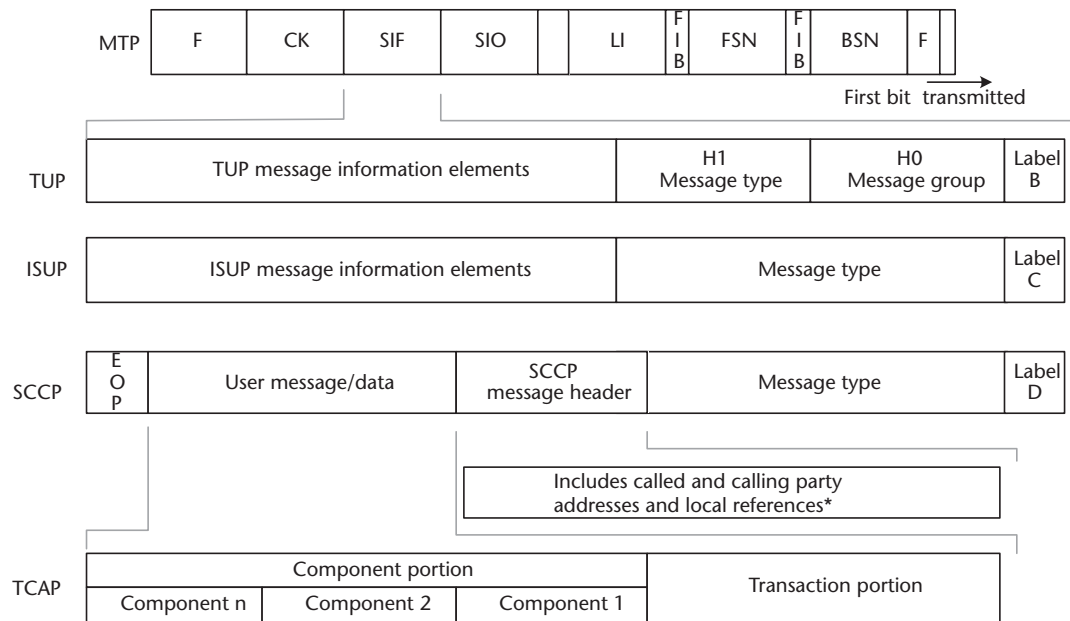**Figure 3.8** SS7 operation within the overall network.

Figure 3.9 SS7 concepts and message details: (a) architecture, (b) functional levels, and (c) message structure. (*Source:* ITU-T: Q.700.)

MTP | F | CK | SIF | SIO | | LI | F I B | FSN | F I B | BSN | F

First bit transmitted

TUP | TUP message information elements | H1 Message type | H0 Message group | Label B

ISUP | ISUP message information elements | Message type | Label C

SCCP | E O P | User message/data | SCCP message header | Message type | Label D

Includes called and calling party addresses and local references*

TCAP | Component portion / Component n | Component 2 | Component 1 | Transaction portion

* Co only.

BIB: Backward indicator bit
BSN: Backward sequence number
CK: Check bits
F: Flag
FIB: Forward indicator bit
FSN: Forward sequence number
LI: Length indicator
SF: Status field
SIF: Signaling information field
SIO: Service information octet

(c)

Figure 3.9 (continued).

in Figure 3.8, a switching node in the network is called a signal transfer point (STP). It handles traffic between nodes in different types of networks, notably the service control points (SCPs). Nodes connected to the signaling network are called signaling points (SPs). The user services, consisting of UPs and APs of the SS7 network, play an important role as functional building blocks in other networks. UPs and APs handle user information, such as control information for setting up and clearing voice and data connections, and information for centralized supplementary services [intelligent network (IN) services]. Table 3.1 indicates the relationship between UPs and APs with respective applications and associated ITU-T recommendations. They communicate with the same type and levels in the signaling network.

The message transfer part (MTP) and signaling connection control part (SCCP) are at the OSI layers 1 to 3 and form the network service part (NSP). The MTP forwards message signal units (MSUs) between user parts of the same type (e.g., in the case of the PSTN, TUPs between exchanges). TUP messages for the set up and clearing of a telephone connection pass all the exchanges along the traffic path with

Table 3.1 The User Services of the SS7 Network

User Part/Application Part	Application	ITU-T Recommendations
Telephony user part (TUP)	Signaling in PSTN	Q.721 to Q.725
ISDN user part (ISUP)	Signaling in N-ISDN and PSTN	Q.761 to Q.764
Mobile application part (MAP)	Signaling and database communication in PLMN	
Transaction capabilities application part (TCAP)	Support for communication with IN databases and for signaling in PLMN	Q.771 to Q.775
Intelligent network application part (INAP)	Communication with IN databases	
Operation and application maintenance part (OMAP)	Communication in management networks	Q.750 to Q.755

instructions for switch control for every participating exchange. Similarly, other user parts AP/TCAP or ISUP deal with the control of the participating exchange nodes. Figure 3.9(b) indicates the functional levels of SS7. Figure 3.9(c) indicates the message structure used in SS7. Signaling terminals exchange information via the PCM channels coupled through exchange terminal circuits (ETC) described later. More details are available in [37].

3.6   Digital Exchanges

By 1985, most telecom equipment manufacturers were able to supply fully digital exchanges due to the proliferation of inexpensive microprocessors, memories, and other semiconductor components. With modular software being bundled with telecom systems, many operators were able to introduce advanced services such as call forwarding, call waiting, multiparty conference, subscriber barring, and specialized billing services.

In the present-day networks, PSTN nodes can be subdivided into three main categories: LEs, transit exchanges, and international/gateway exchanges. LEs are used for connection of subscribers. Transit exchanges switch traffic within and between different geographical areas. International or gateway switches handle traffic between different operators or national boundaries.

3.6.1   Functional Groups of a Digital Exchange

A digital exchange can be logically divided into two separate functional parts as shown in Figure 3.10: the switching system and the processing (or control) system. The processing system, as discussed in Section 3.3, is a complex processor system that handles all control functions, subscriber data, billing information, and routing information, in addition to the control of the switching part.

The switching system contains two switching points: the central group switch and the subscriber stage. The former allows common equipment such as dial tone
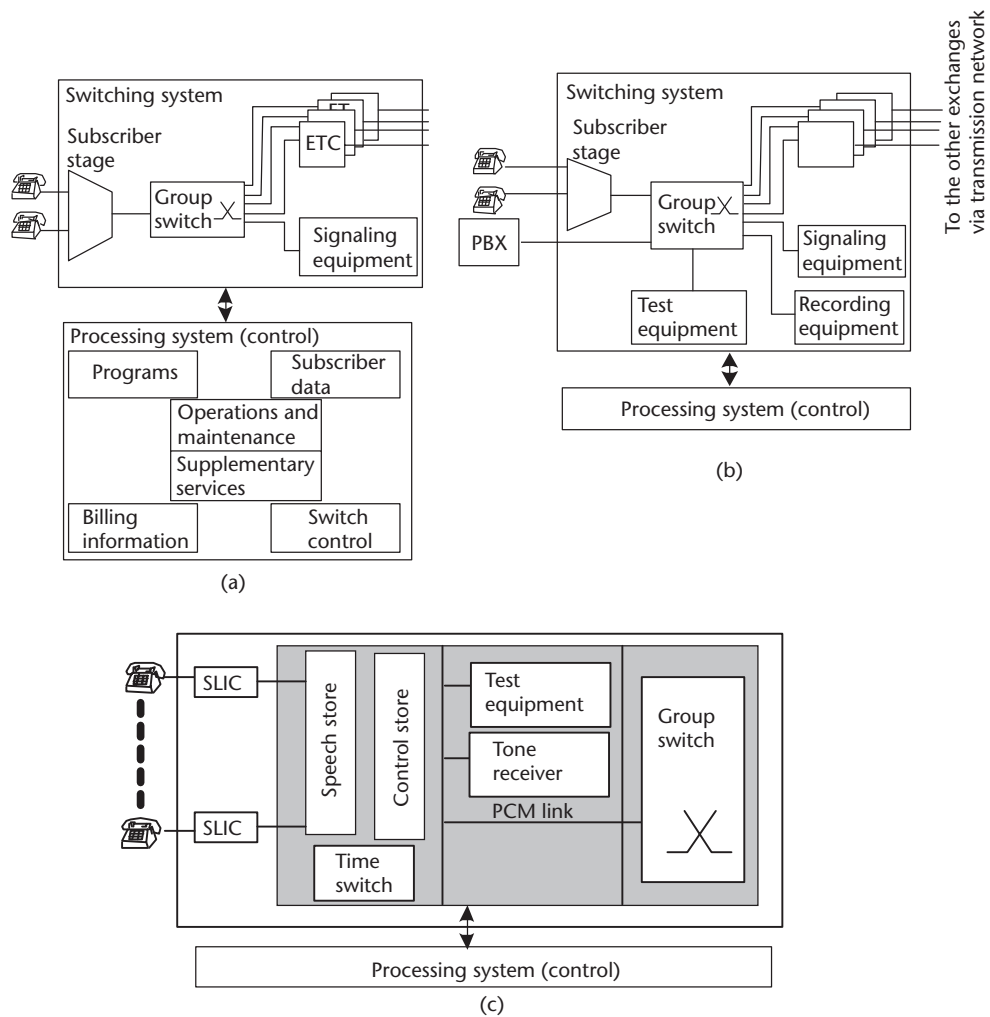
Figure 3.10   A simplified view of local exchange (a) concept, (b) equipment connected to the group switch, and (c) subscriber stage.

providers and ringers to be connected to subscribers and to concentrate traffic to the central group switch. The subscriber stage, lines to other exchanges (known as ETCs), signaling equipment, recording equipment and test gear are connected to the central group switch, as shown in Figure 3.10(b).

Connection set up is controlled by the processor system by reading a set of test points in the subscriber stage that detect off-hook conditions and provide dial tone through relays in the switching system. A characteristic feature of modern exchanges is their modular design, facilitating capacity expansion and the addition of new functionality. All exchanges include a number of basic functions within the core:

- A group switch for switching functions;
- A trunk stage including ETCs operating as interfaces to the transport network;

- Signaling functions for communicating with other exchanges (such as SS7-CCS);
- Operation and maintenance function;
- Control functions.

A local exchange sets up a connection from one of the subscribers to:

- Another subscriber in the same LE;
- A PBX via a time slot in a T1/E1-type PCM link to the PBX;
- A time slot of a PCM link to another exchange;
- The Internet via a modem pool and an access server;
- A recorded announcement;
- Test/measurement equipment needed for maintenance services.

Also, signaling equipment of the CAS kind is used for multifrequency compelled (MFC) signaling, and is connected to the same time slot that will be later used by the call. SS7-type signaling equipment is semipermanently connected to a time slot dedicated for signaling (see Chapter 7). All of these systems are controlled by the processing system, which is a multiprocessing, multitasking system. During the peak periods, congestion in the subscriber stage, group switch (GS), or on the ETC-connected outgoing lines can cause grade of service (GOS) deterioration. The processor system can keep a track of events related to calls for call tracing purposes or the estimation of GOS parameters and billing.

Figure 3.10(c) depicts the subscriber line interface cirucits (SLIC) and a small TSM and associated SS and CS. The SLIC takes care of the BORSCHT functions (see Chapter 2); TSM handles local calls, connection of various equipment, and grouping time slots for PCM links connected to the group switch. The degree of concentration is typically between 10:1 to 3:1, based on the traffic generated by subscribers. When a subscriber makes a call, the corresponding time slot contains speech samples; otherwise, they are empty. The control system analyzes and determines the order in which the samples are to be read for the desired connections among subscribers, and the corresponding values are written into CS. In this way, subscribers are connected to one another, to the group switch, or to other equipment. The group switch comprises of a set of TSMs and at least one SPM forming a TST stage.

In modern exchanges, the group switch is used both by the PSTN and the N-ISDN. N-ISDN services requiring connection of more than one timeslot are referred to as *wideband* or *n*.64 kbit/s. Another new service is the broadcast function for services such as weather, business, or public announcements. These services call for the connection of an announcement service to many simultaneous listeners.

The control function of an exchange ensures that the data needed in a connection is read and modified as necessary. This includes all data about subscribers or the connected exchanges and the services and facilities they use in addition to the equipment connected to the group switch. Modern exchanges can handle large busy hour call attempts (BHCA). The BHCA indicates the control capacity of the exchange. Most powerful control systems used in modern systems allow over

1 million BHCA [5]. Table 3.2 is a simplified explanation of the setting up of an outgoing call, as depicted by the steps in Figures 3.11(a–d).
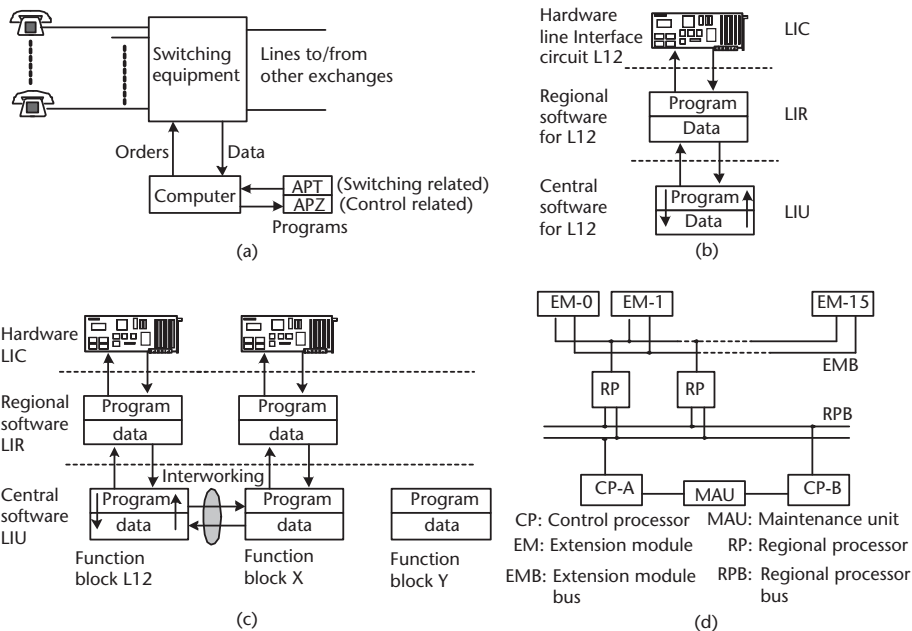
3.6.2 Processor Systems

Digital exchanges used in the PSTN today handle anything from a few thousand subscribers to nearly 100,000, or equivalent trunk groups in the case of transit or gateway systems. Large subscriber numbers and advanced services can only be handled by extremely complex real-time processor systems and modular software. Generally, duplicated processor systems with parallel or hierarchical structures operate in hot-standby modes for reliability. In addition to essential information processing, these systems handle special services as well as system supervision and fault reporting tasks. Most of the processor systems are coupled to maintenance centers to allow centralized maintenance and supervision. Figure 3.12(a) illustrates this concept, dividing the switching equipment, processing equipment, and the associated software. Switching equipment is coupled to the main processors via fault-tolerant lower level processor subsystems. Real-time software organized in a highly modular structure communicates with the switching equipment by sending orders and receiving data from the switching equipment. This example is for AXE systems where hierarchical processing is used.

Table 3.2 Setting Up of an Outgoing Call

Phase	Tasks
Phase 1—Figure 3.11(a): Off-hook detection	Check subscriber database for the barring level of the subscriber (the subscriber may be permitted to dial a limited number or all types of outgoing calls), hot line service activation (where no dial tone is sent), and whether any other services such as unconditional call forwarding are activated (where a different dial tone should be sent). Reserve a memory area in the control unit for storing the *B*-subscriber number and associated connection information. Use the connection of a tone receiver for detecting DTMF signaling.
Phase 2—Figure 3.11(b): *B*-subscriber number detection	This requires analysis of the *B*-subscriber number for determining *B*-subscriber location, charging function, and length of the number dialed. If the *B* subscriber is within the local exchange, send a query to the database for the SLIC of the *B* subscriber, the level of incoming call barring of *B*, and special services allowed for *B*. If *B* is in another exchange, perform routing analysis involving the subscriber category, alternative routings, and the conditions of the network. Charge for the service. After analysis, prepare set-up connection by reserving a time slot in the GS together with other needs for signaling.
Phase 3—Figure 3.11(c): Call set up	While a connection is set up from the far end (possibly via several intermediate exchanges), a signal is sent to *A* subscriber's exchange indicating that *B* is free. The GS reserves a path between the subscriber stage and the selected time slot on the outgoing PCM link. Enable appropriate disconnection of the tone receivers, allowing the time slot connection. The *B* subscriber's exchange sends ringing signals/tones. On lifting of the *B* handset, the voice samples are switched and activation of charging and call monitoring begins.
Phase 4—Figure 3.12(d): Call release	If *A* concludes the call, the call is released without a noticeable delay and any time supervision (as *A* pays in general). If B hangs up first, a supervisory period is initiated allowing any other connections *A* is continuing with. When the call is finally released, the control system disconnects the call charging. Disconnection takes place of all equipment used for the call, release of time slots, and signaling performed to release all other exchange equipment.
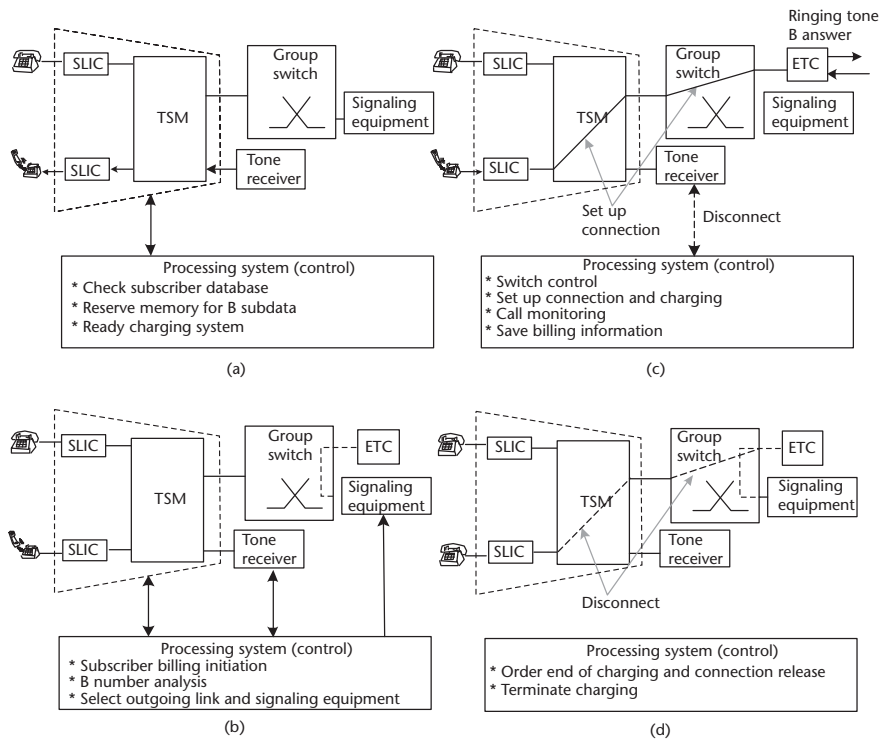
Figure 3.11 Process and connection stages of a phone conversation: (a) A-subscriber lifting a hand set, (b), B-number analysis for the outgoing connection, (c) outgoing call set up, and (d) release of the call.

Figure 3.12 An example of switching and control systems in a typical exchange—AXE architecture from Ericsson: (a) basic concept, (b) an example of a function block, (c) interworking of function blocks, and (d) processor hierarchy. (*Source:* L. M. Ericsson, Sweden.)

In a typical AXE exchange, the overall activities are divided into many function blocks, which consist of switching hardware, regional software, and central software. This is shown in Figure 3.12(b), which indicates the example of a LIC. As shown in Figure 3.12(c), several function blocks can communicate with each other through the central software using software signaling. Central operating system allows interworking between function blocks by a cyclic system that cycles fast, allowing a meaningful timeshare process.

Figure 3.12(d) indicates the processor interconnections. Duplicated central processors (CPs) communicate via a duplicated regional processor bus (RPB) with many regional processors (RPs). The duplication of buses and the processors maintain reliability. Duplicated RPs control the switching hardware via an extension module bus (EMB). The duplicated extension modules (EMs) and the EMB allows reliable switching system hardware functions. Duplicated CP pairs such as CP-A and CP-B, providing the necessary CP capacity, communicate with each other via a maintenance unit (MAU). This complex architecture divides a given function and its associated tasks into smaller units. The CPs make the major decisions, while more routine tasks are handled by the RPs. RP pairs carry specialized software, dependant on the connected switching equipment such as the group switch, LICs, or ETCs. EMs receive the RP output commands (in bytes), which are decoded and used by the switching hardware. For example, a set of subscribers and their associated LICs are grouped in such a way that when a given set of subscribers go off hook, the conditions are read by the associated RP pair, informing the CP to command the necessary other function blocks to commence the call establishment.

Figure 3.13(a) indicates the distributed control architecture of the Fujitsu FETEX-150 switching system. A main processor (MPR) system communicates with several call processors (CPRs) and the line processors (LPRs). The MPR manages the overall system and performs maintenance and operation functions. The CPR provides call processing functions using the LPR located in the line concentrators or remote switching units. Figure 3.13(b) shows the concept of the *hypothetical independent exchange method* with hypothetical speech path and signaling links. For further details, [9] is suggested.
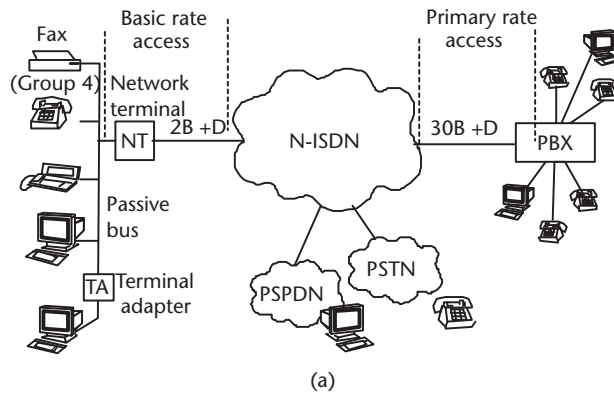
3.6.3   ISDN Systems

Traditional networks developed in separate subsets such as the telephony, telex, and packet switched networks, with basically one network for each service category. Transition from analog to digital systems paved the way for integrated digital networks (IDNs) with digital switching and transmission. Further developments led to ISDNs with advanced signaling for different signal bandwidths. ISDN is a circuit-switched concept, as shown in Figure 3.14.

As discussed in Chapter 6, ISDN makes use of the higher frequency capability of the copper pairs, allowing two basic (*B*) rate (64-kbit/s) channels for voice or data traffic and a 16-kbit/s data (*D*) channel for signaling, forming a basic rate access (BRA) situation as in Figure 3.14. For connection of PBX systems, primary rate access (PRI) is used, where 30 *B* channels and a *D* channel are present.

System consequences of an LE with PSTN/ISDN combination and its switch control in the LE will be more complex due to handling of more n.64-kbit/s channels. Usually optional packet handlers are installed in the switching part with access
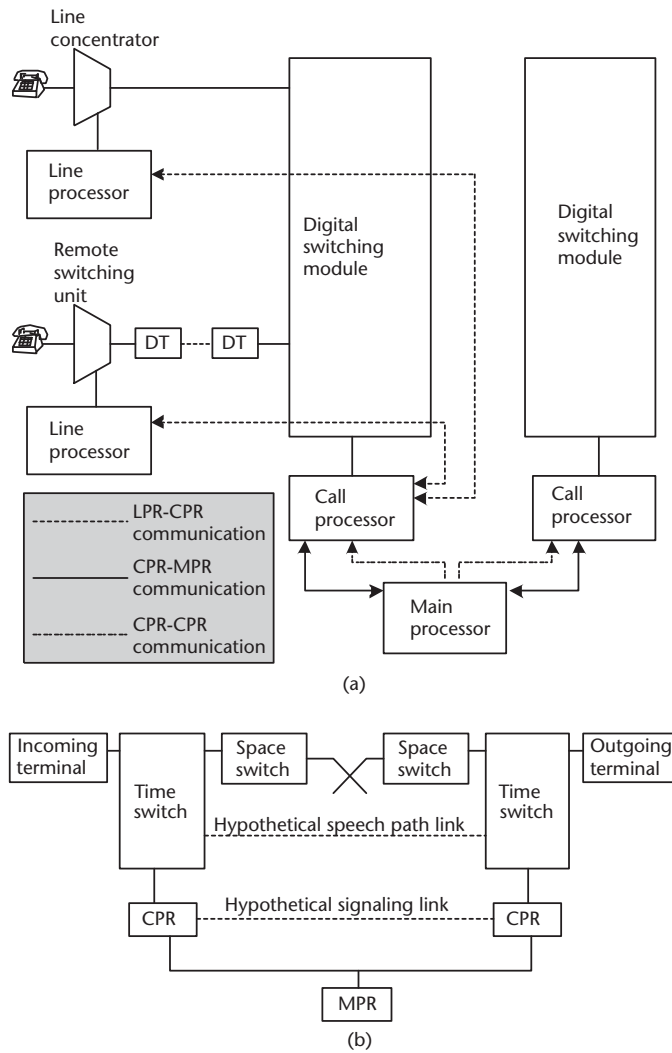
Figure 3.13 Processor hierarchy in Fujitsu FETEX-150 system: (a) basic concept, and (b) hypothetical independent exchange method. (*Source:* [9]. ©1990 IEEE.)

Figure 3.14 ISDN concept.

servers with statistical multiplexing capability. Narrowband ISDN (N-ISDN) is currently progressing towards broadband ISDN (B-ISDN) systems making use of newer ATM switches. For more details, [5] is suggested.

3.7   Private Branch Exchanges and Key Phone Systems

3.7.1   Private Branch Exchanges and Centrex

To facilitate intercommunication requirements within large organizations, the same concepts used in digital switches were adapted in private branch exchange (PBXs), dividing them into two clear subsets: the Centrex service and the PBX systems. Centrex is a service to simulate a PBX for an organization by means of software installed in a LE. This service is attractive for medium-sized businesses and eliminates the need for in-house service teams or contracts with PBX system providers. Figure 3.15(a) and (b) distinguish the PBX and the Centrex services.

Both cases can be multilocation systems, where branch offices are connected in such a way that the user dials only extension numbers, even though the sites are connected via T1/E1 trunks via the PSTN. For PBXs coupled via leased lines, capacity can be reserved in the PSTN, allowing permanent connections through respective group switches in participating public exchanges. Computer telephony integration (CTI) is another common service when a LAN is present.

With the Centrex function, the extensions at distant locations are translated to special numbers and sent via intergroup traffic. In the mid 1990s, many wireless PBX (WPBX) systems entered the market, providing mobility in an office environment. Another enhancement to PBX or WPBX systems was ISDN facilities. References [9–16] provide some development-oriented insight on this technology.
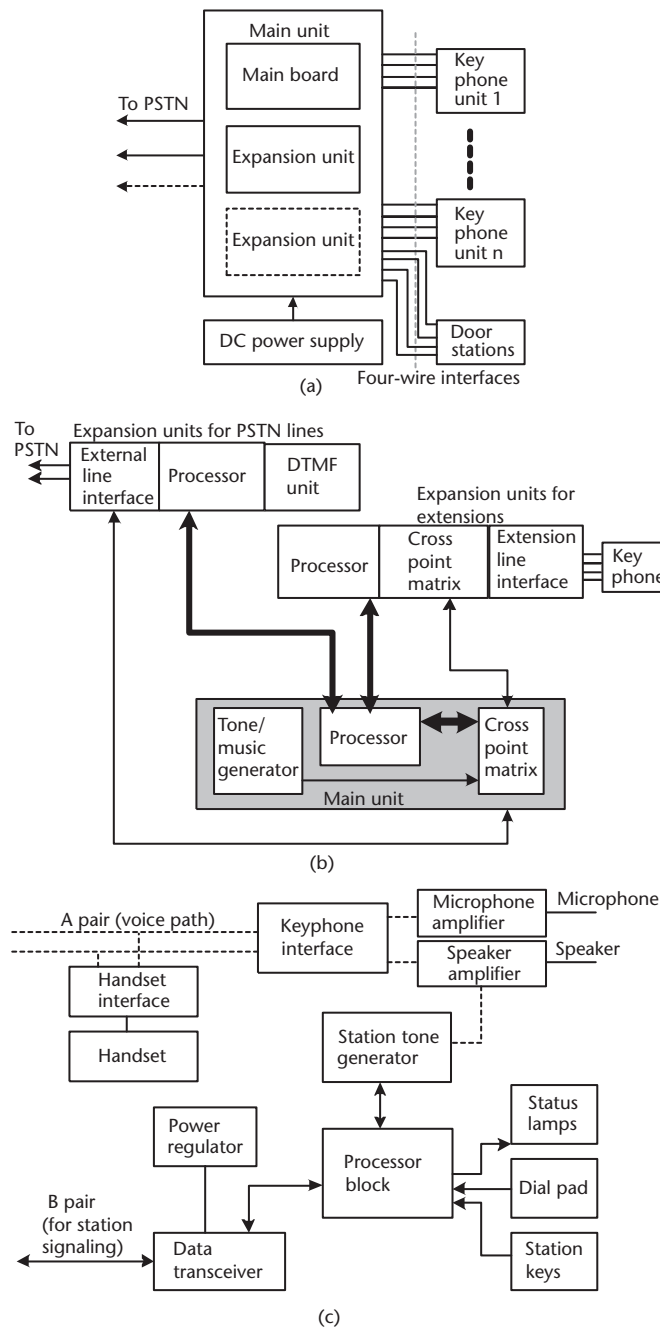
3.7.2   Key Phone Systems

Key phone systems are smaller cousins of PBX systems, where the cost, implementation, and facilities are not as sophisticated as PBX or Centrex systems. Most low-end systems (from few extensions to around 100 extensions) are designed around a
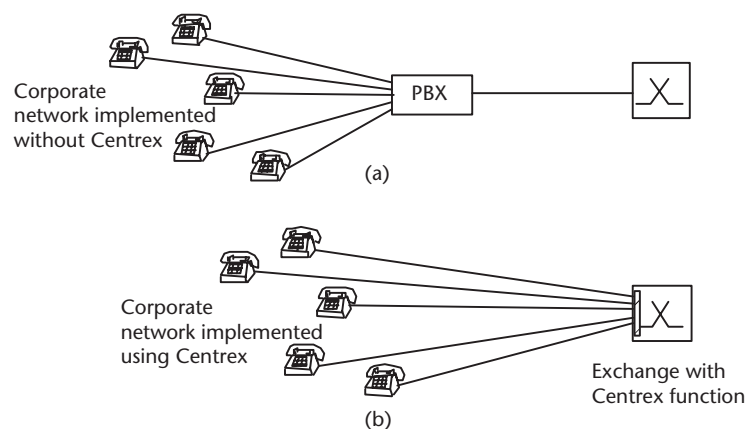
Figure 3.15   Comparison of PBX and Centrex systems: (a) PBX, and (b) Centrex.

few microprocessors and some switching hardware, allowing many users to access the PSTN with a few subscriber loops. Designed around simple cross-point switches, key phone extensions typically work with four-wire interfaces. One pair is used for voice, and the other pair is used for signaling (as well as power distribution to key stations). Figure 3.16 depicts elements of a typical key phone system.

Figure 3.16 Components of a typical key phone system: (a) system interconnections, (b) block representation, and (c) key station block diagram.

Figure 3.16(a) indicates the main unit with its main processor board and expansion boards with dedicated processors. Some of the units can have cross-point matrices and external line interfaces as required by the configuration. Figure 3.16(b) indicates a possible configuration with several expansion boards. A typical key station block diagram is shown in Figure 3.16(c).

For small and medium-size office systems, differences between the key phone system architectures and PBX are hardly visible. Many commercial variations with different capacities are available.

3.8   Merging Data and Voice

Modern telecommunications systems handle a complex mix of voice, data, video, and multimedia. In such situations, the information flowing through switches is fully digital. This scenario forces service providers to handle the following parameters carefully:

- Bandwidth;
- Burstiness (variation in the bandwidth requirements of a given information set);
- Bit errors and blocking;
- Delay;
- Information security.

Giving special attention to these aspects, modern switch cores are changing fast to handle high-speed digitized signals with appropriate packet-switching techniques. However, this development places special demands on applications needing real-time communications.

3.9   Packet Switching Systems

3.9.1   X-25 and Other Early Packet Switching Systems

Developing from the early 1970s, packet-switched public data networks (PSPDNs) have been installed all over the world. As summarized in Section 3.4.1, the concept is shown in Figure 3.17(a). As depicted here, the user's data terminal equipment (DTE) is connected to the packet-switched network via the data circuit terminating equipment (DCE). X.25 was the first international standard for wide area packet networks. The name refers to one of many recommendations, as seen in Table 3.3, for packet networks by the then CCITT, for both asynchronous and synchronous access. The application environment for these standards is shown in Figure 3.17(b).

X.25 is essentially a protocol for synchronous transmission across the DTE/DCE interface, as shown in Figure 3.17(b). However, the name X.25 has been applied to a suite of protocols covering three layers of a packet-switched network, as indicated in Table 3.4.
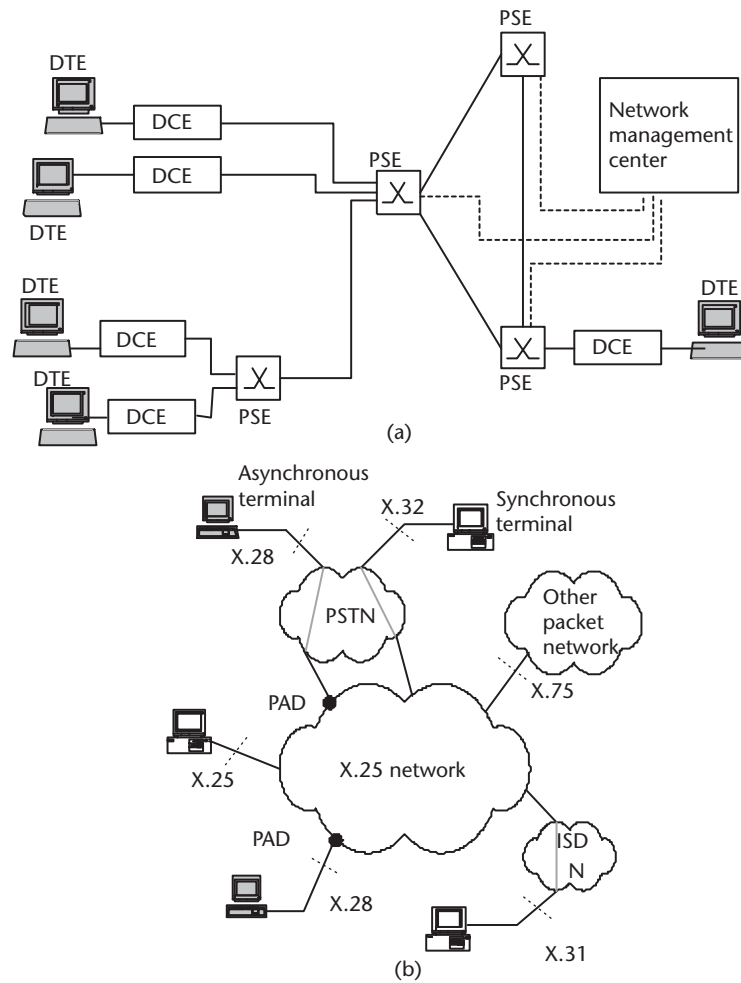
Figure 3.17   Packet networks: (a) basic structure, (b) early standards.

Table 3.3   Early Packet Switching Standards

| Standard	Description
X.3	Packet assembly and disassembly (PAD) for asynchronous data equipment
X.25	Interface functions and protocols for the connection of synchronous data equipment in packet-switched networks
X.28	Interface functions and protocols for the connection of asynchronous data equipment to PAD functions in the PSE
X.29	Exchange procedure for control information between a PAD and a synchronous data equipment in PSE
X.31	ISDN terminal connection
X.32	Interface between DTE and circuit-switched equipment such as DCE via PSTN channel
X.75	Interface between X.25 networks

Table 3.4 Layers and Details of the X.25 Protocol Suite

Layer	Function
Layer 1	This is the interface to the physical layer, DTE/DCE (X.21 and X.21 bis for digital and analog terminals, respectively, and X.31 for ISDN terminals).
Layer 2	This is the link layer, which defines the data flow over a link. It ensures error-free transport of packets to and from layer 3.
Layer 3	This defines the addressing and packaging (packet assembly). It handles virtual circuits (VCs) and defines packet types for call set up, data phase, and clearing. It also multiplexes logical channels to a single physical channel and addresses flow control and error handling.

3.9.1.1 Packetizing and Signaling in X.25

X.25 is a network that operates on a store-and-forward basis with variable length packets. Layer 2 in X.25, which ensures error-free transport of packets, applies a version of the high-level data link (HDLC) procedure. This is the basis for many packet data networks [7]. The variation used in X.25 is called link access procedure B or D. (LAPB/LAPD). The frame structure [8] formed in HDLC is illustrated in Figure 3.18(a). The information field is used by the data packets from layer 3. Only one packet can be transported in each frame. The frames are separated by one or more eight-bit flags. The signaling process is indicated in Figure 3.18(b).
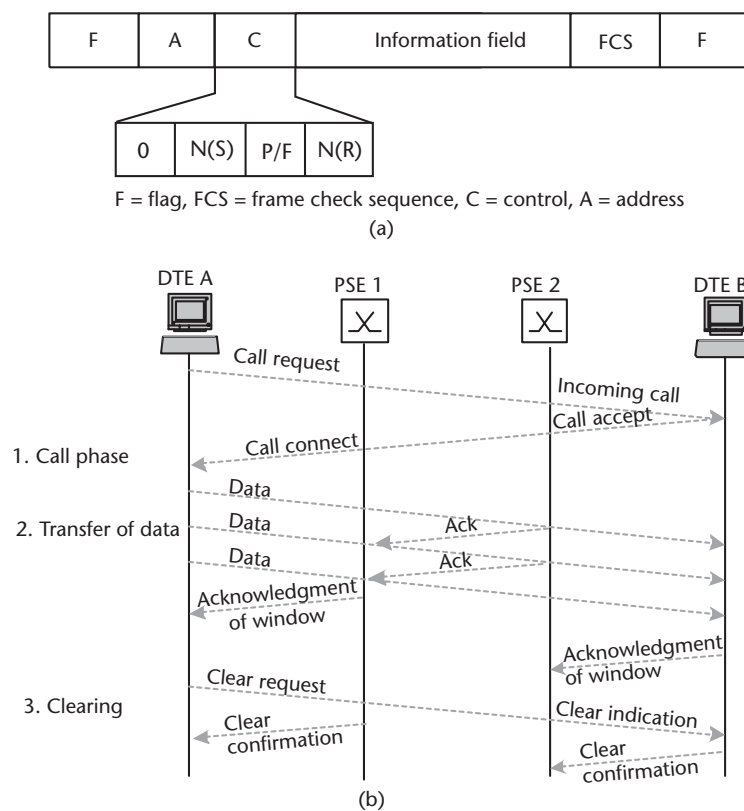

Figure 3.18 HDLC: (a) frame format, (b) signaling and data exchange.

3.9.2 Frame and Cell Relay

Up to the early 1990s, X.25 was the only technology to offer flexible bandwidth, or bandwidth on demand (BOD), services. Introduction of new data services in the early 1990s showed that the typical X.25 services using mostly 64-kbit/s links were creating bottlenecks. Also X.25 networks were designed with the assumption that BERs were in the order of 10^{-6} or worse. With fiber and digital microwave links developing with much lower BERs, in the order of 10^{-9} or better, while fast digital switches were developing in the PSTN and PSDPN, frame relay (FR) services were introduced. By reducing much overhead within the error correction and flow control blocks, frame relay was able to transport data much faster over packet networks than X.25.

Protocols for FR are relatively simple and active within the layers 1 and 2 of the OSI model, with only a small amount of overhead. High transmission capacities of 2 Mbps up to 50 Mbps are involved. Services are provided mostly via PVCs, while SVCs are gradually being introduced. LAN-to-LAN connections have resulted in a high rate of FR installations and services.

FR technology was developed using the X.25 experience for the demands of the new data-centric customer bases, and the standardization process commenced in 1990. Figure 3.19(a) indicates how an FR router connects a LAN to the network via an FR switch. In a popular LAN interconnection situation, an FR access device (FRAD) in a LAN is usually combined with a router that has specific protocols for an FR network [Figure 3.19(b)]. The other end of the user network interface (UNI) is
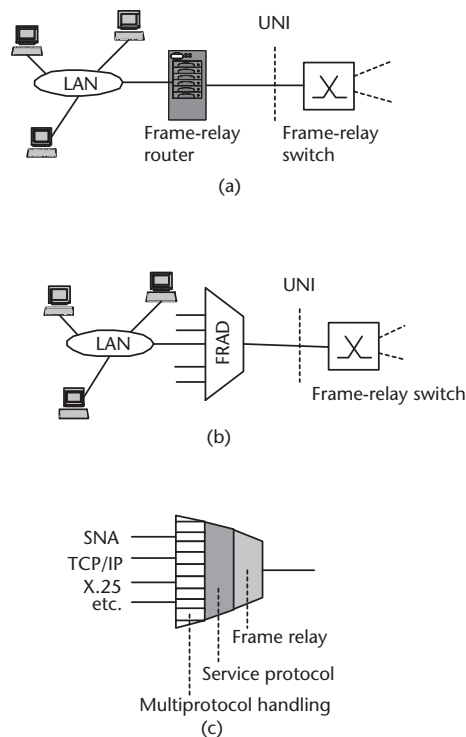

Figure 3.19   Frame relay: (a) LAN connection via FR, (b) FRAD, (c) multiprotocol handling via FRAD.

equipped with the FR switch, also known as an FR network device. FRAD is considered the equivalent of the PAD in the X.25, though the FRAD does not assemble data. As seen in Figure 3.19(c), the FRAD has a multiprotocol interface that permits interfacing of protocols other than FR variants. Any commercial FRAD contains three primary components: a multiprotocol handler, service protocol, and the FR component. The FRAD can assemble several multiprotocol information sources to a single logical connection. However each end user sees a transparent connection.

In FR, the physical layer and part of framing functions (as described by standard Q.922) are implemented by the network, while the upper part of the framing functions defined by Q.922 is implemented by the interface. By restricting to a two-layer protocol within the network, the overhead is much removed, compared to X.25. FR uses LAPF at layer 2, an adapted version of LAPD, which was originally designed for N-ISDN.

Figure 3.20(a) indicates the relationship of protocols in an FR network. The physical layer standards involved are derived from ISDN systems. The frame format as per Figure 3.20(b) follows the HDLC format. It consists of two flags delimiting the beginning and end of the frame, an address field [the data link connection identifier (DLCI)], the information field (which is variable in size), and a frame check sequence. The DLCI field is variable from 10 to 23 bits. The fields within these bytes are described in Table 3.5. Note that Figure 3.20(b) indicates only the case of two byte (default) header format. See [6].
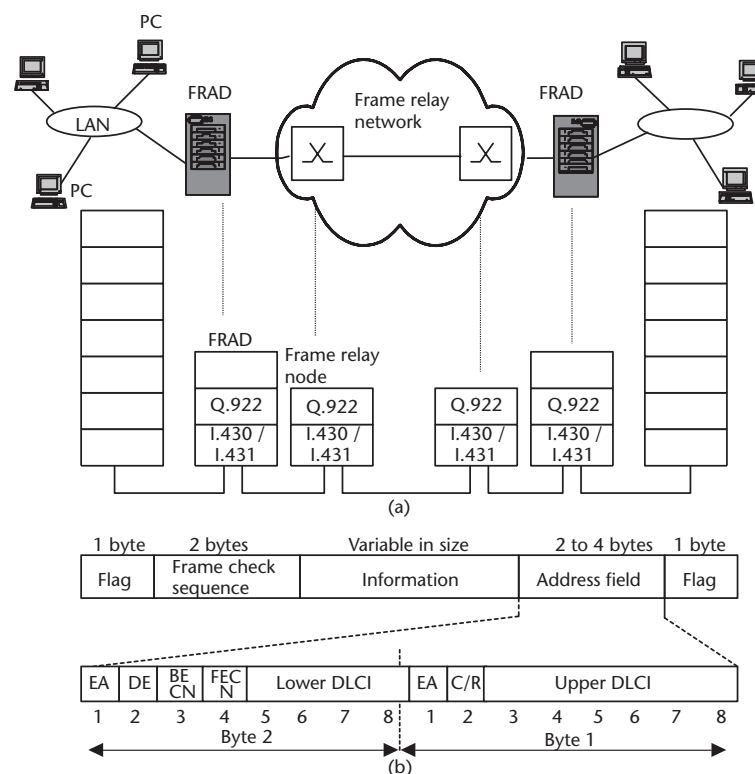

Figure 3.20   FR protocols and frame details: (a) protocols in an FR network, and (b) frame formats. (*After:* [6].)

Table 3.5   Details of the FR Frame

Field	Description	Remarks
DLCI	DLCI indicates the logical connection between different FR users.	DCLI defines the next destination to which the frame is transported. This identifies the virtual circuit.
	The first six bits of the byte 1 and the first four bits of the byte 2 form the DCLI.	There is a total of 2^{10} logical connections. Many are reserved for special tasks. By assigning different DLCI values to different frames, several simultaneous logical connections can be made on the same circuit.
Command / response (C/R)	CR is not used by the FR protocol.	It is sent transparently through the nodes to users.
Extended address (EA)	EA is used to indicate whether the DCLI is over 10 bits.	EA=0 means another byte is to follow.
Forward explicit congestion notification (FECN)	FECN indicates the overload in the network.	FECN is sent to the receiving end.
Backward explicit congestion notification (BECN)	BECN is the same as FECN, used in the direction of sender.	
Discard eligibility (DE)	DE is an indication of frame discard in case of congestion.	DE is used as a priority case for discarding. Frames without a DE can also be discarded
Information field	This is user data.	This has a maximum of 1,600 bytes. (The actual size is decided by the service provider.)
Frame check sequence (FCS)	FCS checks the frame for errors.	

Figure 3.21(a) indicates the use of FECN and BECN in the direction of congestion within the FR network, connecting to LANs. FECN and BECN are used in flow control, where the congested node sends the FECN and BECN to receive and transmit nodes, respectively. As shown in Figure 3.21(b), DLCI has a local significance only between two nodes, and each node can change these as needed along the path between two users.

FR networks operate with the assumption of very low BER and thus minimal need for error control. Elements of the end-user systems, rather than the network itself, guarantee error-free transmission. The FR interface specification provides the signaling and data transfer mechanism between the end points and the network based on a set of user parameters, the most important of which are the committed information rate (CIR) and the excess information rate (EIR). The operator can set the CIR value for individual VCs. For a typical case of a 2-Mbit/s network link, the operator may choose any value from 0, 8, 32, or $n.64$ kbit/s. In such a case, 2 Mbit/s is the speed of the physical link, and the CIR is the guaranteed data rate of the VC. The same physical connection can carry several VCs with different CIR values. However, at times during which some VCs are not active, the available capacity can be taken by the active VCs. The EIR is the maximum permissible transmission capacity for a VC during such short periods. For more information on FR, [17] is suggested.
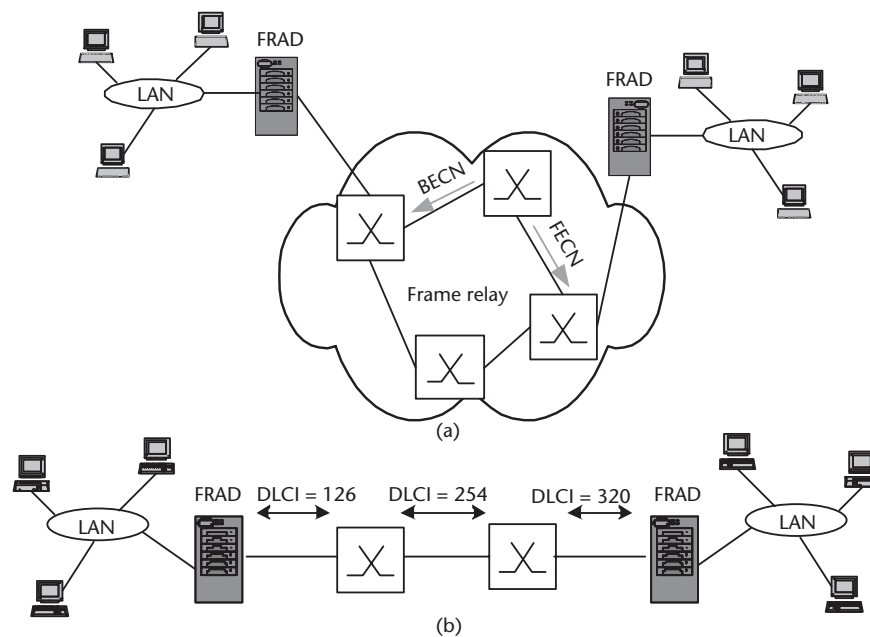
Figure 3.21   Use of FECN, BECN, and DLCI in an FR network: (a) congestion notification, and (b) local significance of DLCI in a connection.

3.9.3   ATM and Broadband-ISDN

X.25 and FR, both of which are packet-switching techniques, have one important common factor, which is the variable packet size. However in handling connections and traffic, the variable packet length can cause inefficiencies, particularly in situations such as VoIP, where the connection delays are important for QoS. With the networks carrying more data-centric traffic and with the need for wideband connections, fixed-length packet size becomes important. Cell relay techniques use such constant, small, fixed-size frames called *cells*. Hence, the fixed-length cell-relay techniques such as ATM and distributed queue dual bus (DQDB) were introduced. Our focus here will be only ATM. For the interested reader, DQDB is discussed in [6].

Evolution of broadband networks was synonymous with the evolution of ATM and B-ISDN techniques. B-ISDN is a further development of the N-ISDN concept, with the telecommunications community recognizing the need for wider bandwidth networks for handling voice, data, and video in the mid 1980s. While N-ISDN was based on switching bandwidths of n.64 kbit/s up to 2 Mbit/s, B-ISDN can switch different bit rates variable from 64 kbit/s to over 100 Mbit/s. At this point, Figure 3.22 provides a clear comparison of switching techniques evolved.

Figure 3.22(a) indicates the different connections in dedicated networks. In ISDN, not only can transmission between nodes be integrated, different transfer modes are also permitted (e.g., circuit mode and packet mode). Figure 3.22(b) illustrates this. When an ISDN subscriber makes a voice call, the circuit-mode switch in ISDN will be used, and the call will be routed via ISDN or PSTN. If the subscriber wants to send packet data, the circuit mode switch has to route the traffic to a
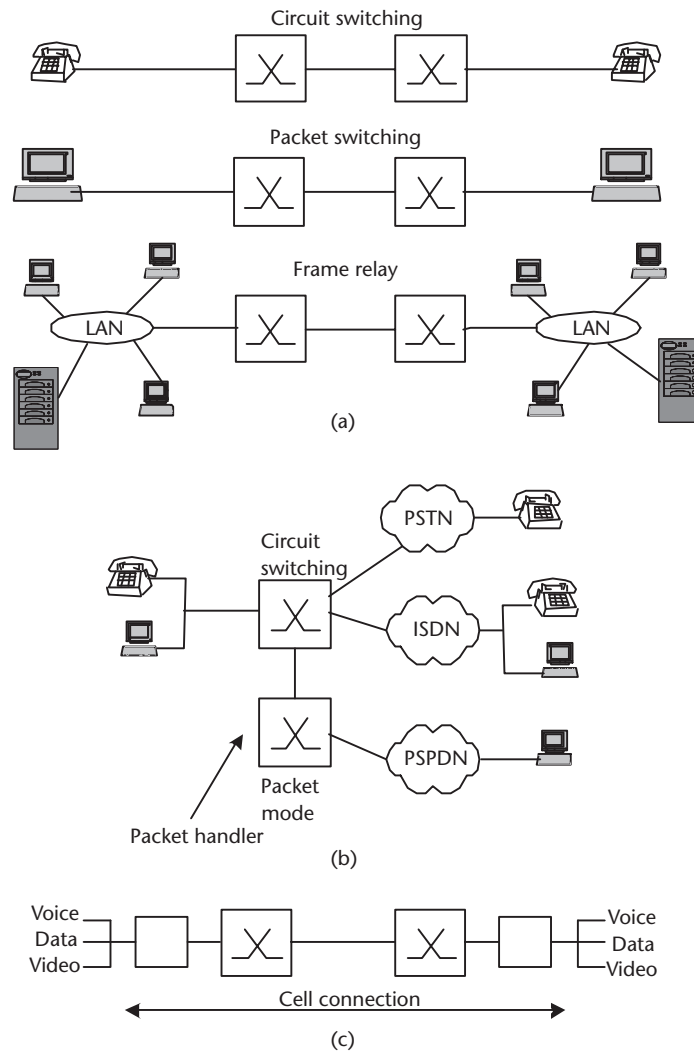
Figure 3.22 Networks and transfer modes: (a) different connections in dedicated networks, (b) integrated subscriber line and node, and (c) full integration in B-ISDN.

packet handler, whose task is to route the packets to the destination, either locally or a PSPDN.

By contrast, in B-ISDN, full integration is possible at a cell level, as per Figure 3.22(c). In this case voice, data, and (motion) video information is transferred via ATM switches, where an ATM cell is a small packet having a fixed length of 53 bytes, with five of them comprising a header. The 53-byte cell format was standardized as a compromise between the U.S. and European needs. The advantage of a short cell of this kind is for real-time services such as telephony, where there is a strong sensitivity to network delays of over 8 to 10 ms. Filling a cell of this size from an individual call connection takes about 6 ms in today's technology. However, in data communication services, longer cells are more effective due to less total overhead. Table 3.6 compares the characteristics of X.25, FR, and ATM.

Table 3.6   Characteristics of X.25, FR, and ATM

Characteristic	X.25	FR	ATM
Transfer mode	Connection oriented or connectionless	Connection oriented	Connection oriented or connectionless
Packet length	Variable	Variable	Fixed (53 bytes total)
Switching	Software oriented, based on the address field of the packet	Software oriented, based on the address field of the packet	Hardware oriented
Speed (bandwidth)	64 kbit/s to 2 Mbit/s	2–50 Mbit/s	Flexible (1.5 Mbit/s to over 2.5 Gbit/s)
Retransmission	Link by link	End to end	
Channel performance	High BER (e.g., leased lines)	Low BER (e.g., optical fiber)	Low BER
Primary application	Data or still picture exchange	Interconnect of LANs	All applications, including B-ISDN

3.9.3.1   General Features of ATM

ATM is a transfer mode to handle a variety of types of traffic. Data, digitized voice, or video information is broken into fixed-length cells of 53 bytes and transferred over a channel based on a *labeled multiplex* basis, enabling the flexible use of the bandwidth. Video information will occupy more cells, compared to voice information, due to the high information rate required. Figure 3.23(a) indicates the
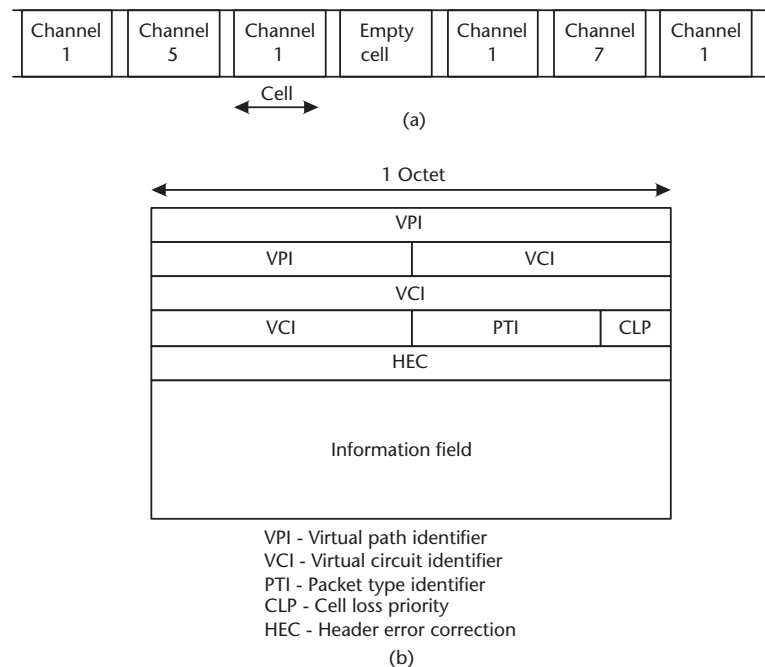
Figure 3.23   Labeled multiplexing and the cell format in ATM: (a) labeled multiplexing for flexible bandwidth, and (b) cell format and header details. (*Source:* ITU-T: I.371.)

principle where channel 1 is allowed more cells than the others. Figure 3.23(b) indicates the cell format, with details of the ATM cell header.

As shown in Figure 3.23(b) the five-byte cell header is divided into different fields:

- Address field in the form of virtual path identifier (VPI) and virtual channel identifier (VCI);
- Payload type identifier (PTI), specifying the contents of the cell (whether user information or network information);
- Cell loss priority (CLP) information, in case of congestion;
- Header error control (HEC) with a check value for detecting bit errors;
- Generic flow control (GFC) used for flow control information.

Classes of Service

ITU-T has standardized a reference model for ATM with three protocol layers corresponding to the OSI model. These are the physical layer (layer 1), ATM layer (layer 2), and ATM adaptation layer (AAL). In this reference model, five service classes are defined [35] based on the following parameters related to the information processed and sent:

- The need for synchronization between the sender and receiver;
- Constant bit rate (CBR) or VBR;
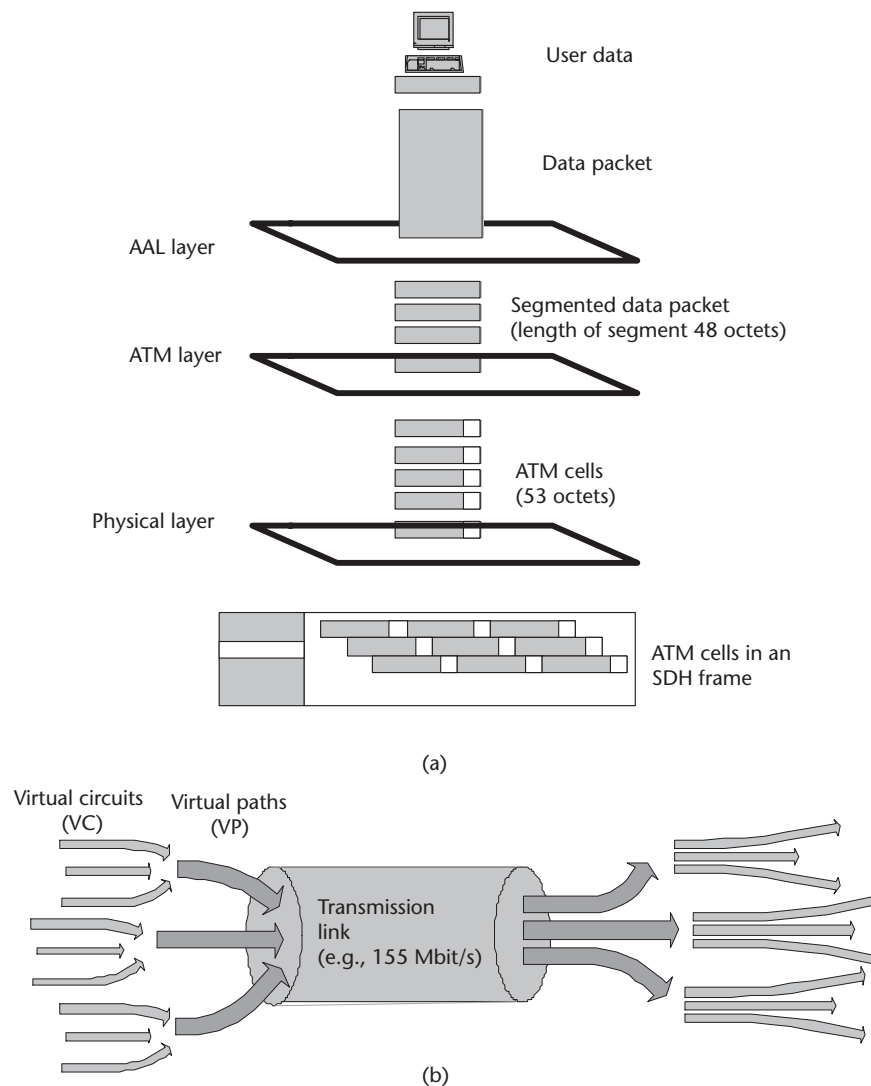- Connection-oriented or connectionless transfer.

Four protocols designated AAL 1, AAL 2, AAL3/4, and AAL 5 support the classes, A, B, C, and D, as summarized in Table 3.7.

Figure 3.24 indicates the relationships between processing layers and the relationship between VCs and virtual paths (VPs), via a transmission link such as 155-Mbps SDH link. More details on ATM and associated standards are well described in [7].

Table 3.7 Classes of Service in ATM

Type	AAL Protocol	Type of Services Handled	Example Applications
1	AAL 1	CBR services for connection-oriented transfers with end-to-end synchronization	PCM coded voice, circuit-emulated connections with bit rates of n.64 kbit/s or n.2 Mbit/s, and video signals coded for CBR
2	AAL 2	VBR-based connection-oriented transfers, which need end-to-end synchronization	VBR-coded video signals, compressed using MPEG 2 standard (when the motion elements in the picture is high, the bit rate needs to be increased), and ADPCM-based compressed voice
3/4	AAL 3/4 or 5	VBR-based, connection-oriented transfers, which do not need end-to-end synchronization	LAN interconnect, FR, and X.25
5	AAL 3/4 or 5	VBR-based, connectionless transfers without end-to end synchronization	LAN interconnect, TCP/IP, and SMDS

User data

Data packet

AAL layer

Segmented data packet
(length of segment 48 octets)

ATM layer

ATM cells
(53 octets)

Physical layer

ATM cells in an
SDH frame

(a)

Virtual circuits
(VC)

Virtual paths
(VP)

Transmission
link
(e.g., 155 Mbit/s)

(b)

Figure 3.24 ATM processing at different layers and VC/VP elements: (a) layer processing, and (b) VC/VP relationships in a 155-Mbit/s link.

3.9.3.2 ATM Switching

ATM is a packet-switching technology that directs traffic using a label contained in the packet's header. Using the information within the header, an ATM switch transports ATM cells from an incoming logical channel to one or more outgoing logical channels. A logical channel is identified by a combination of the number of the physical link and the identity of the channel (specified by the VPI and the VCI).

ATM switches operate on a similar principle to the circuit-mode (voice) switches. The logical channels correspond to time slots, and the identity is based on the VPI and VCI in the cell header. The second function is analogous to the space switch in a circuit-mode switch. Figure 3.25(a) illustrates these principles, where the labels indicate the VCI/VPI combinations of the cell headers, and the payload transfer paths are indicated by A and B.
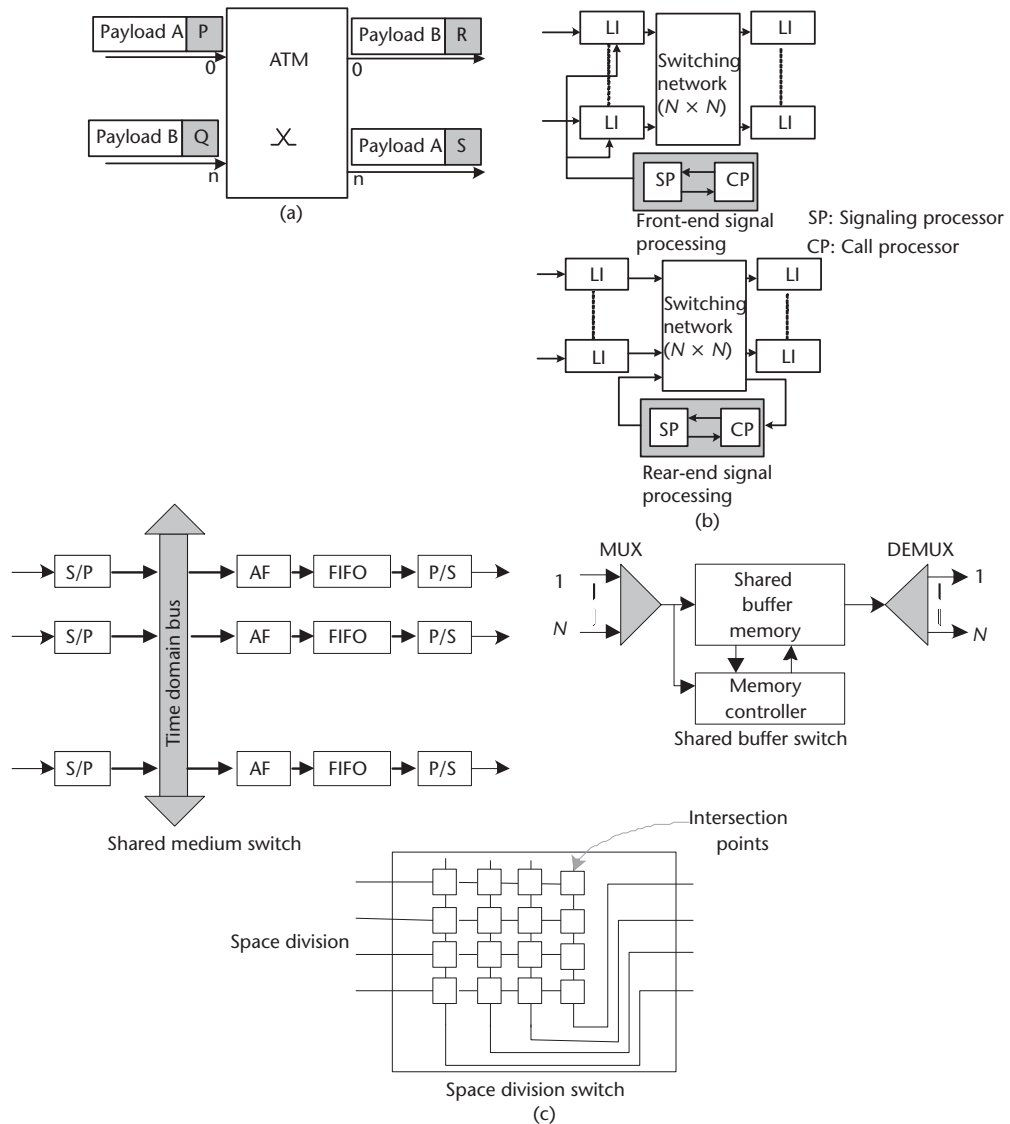
Figure 3.25 ATM switching: (a) principle, (b) stages of an ATM switch, and (c) different architectures. (*Source:* [36]. © 2002 Artech House. Reprinted with permission.)

When ATM cells reach the switch, the cell header is opened, the VCI/VPI parameters are read, and the cells are switched to the right receivers. To ensure that the cells are not lost or incorrectly routed, a bit error check is performed on the cell header based on the header error correction (HEC) indicated in Figure 3.23(b).

Figure 3.25(b) indicates the stages of an ATM switch. Different architectures of ATM switches are shown in Figure 3.25(c) with shared medium, shared buffer (or shared memory), or space division possibilities. For details, [36] is suggested. Switching time through an ATM node is almost 100 times faster than the normal packet switching, allowing wideband information to be handled easily.

3.9.3.3   ATM Standards

During the mid 1990s, the ITU completed a series of standards on ATM for operation and management. These incorporate network resource management, traffic management, congestion control, fault management, and performance management. ITU I series standards for B-ISDN, ITU G-78X series of SDH standards, and many others are included in the overall process, which allows the development of a global ATM network. References [18, 19] are recommended for details.
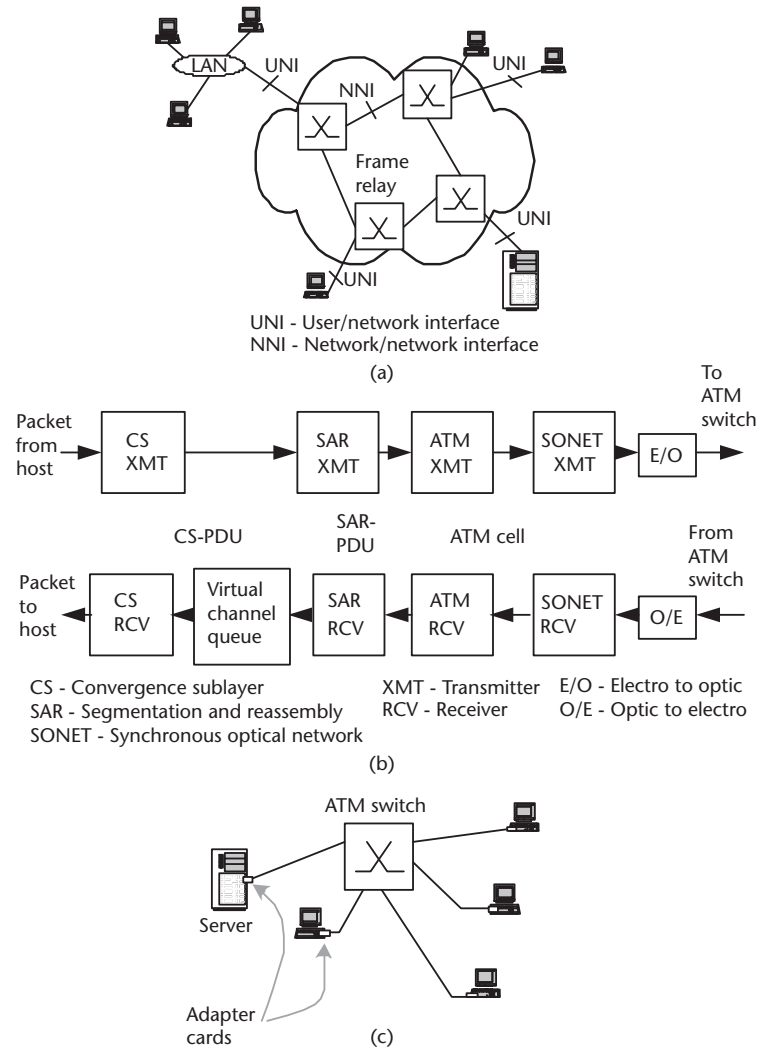
3.9.3.4   ATM-Based Enterprise Networks

ATM was seen in the early 1990s as the technology to meet bandwidth and application requirements across both LANs and WANs. As shown in Figure 3.26(a), various UNI and network node interface (NNI) connections could be carried via different physical media, such as plesiochronous digital hierarchy (PDH) or SDH (see Chapter 7). The structure shown there can be same for LANs, metropolitan area networks (MANs), or WANs. A virtual channel connection (VCC) is set up between any source or destination pair in the ATM network [17], and the network signaling allows setting up a VCC between the two end systems by the use of VCIs and VPIs.

In the latter part of the 1990s, user organizations seriously considered using ATM as a migration path from fiber distributed data interface (FDDI) and Fast Ethernet, encouraging users to couple their LANs and desktop computers to ATM switches. With the release of LAN emulation specification [17] and the availability of suitable software, many LAN vendors were actively developing ATM products. One common way to interconnect desktops [20] is to use an adapter card on the host computer, which allows the protocol conversion process and the electrical to optical (E/O) and optical to electrical (O/E) conversions required to connect to ATM networks. Figure 3.26(a) and (b) illustrate the ATM WAN and LAN network concepts coupling desktops and LANs and the block operation of such a card [20]. However, due to the limitations of many desktop computers with their software bottlenecks, direct coupling may not be the best solution, except for such cases as servers. A better way is to use an ATM switch and LAN emulation [20], as per Figure 3.26(c).

3.9.4   IP Switching and Label Switching

The last decade has seen several major changes in the Internet. Not only has there been an explosive growth in terms of size and volume, but there has also been an increase in the number of multimedia and real-time applications, which have put significant pressure on the network to support higher bandwidth and provide QoS guarantees. Though the Internet has been successful in terms of scalability due to its connectionless nature, the hop-by-hop packet forwarding paradigm has turned out to be insufficient in supporting these demands. As an improvement in traditional IP routing, *IP switching* techniques have been developed, which makes use of the label switching technology of ATM for packet forwarding. These techniques are outlined in [21].

UNI - User/network interface
NNI - Network/network interface

(a)

CS-PDU   SAR-PDU   ATM cell

CS - Convergence sublayer
SAR - Segmentation and reassembly
SONET - Synchronous optical network

XMT - Transmitter   E/O - Electro to optic
RCV - Receiver   O/E - Optic to electro

(b)

(c)

Figure 3.26 ATM WANs and LANs: (a) ATM network (WAN), (b) ATM adapter card operation (Source: [20], ©1996 IEEE.), and (c) ATM LAN.

3.9.4.1 Switching Versus Routing

X.25, FR, and ATM are all based on connection-oriented packet switching, where switching is based on VC/VP identifiers contained in the data-link layer frames. The switch looks up a table for the outgoing VCI/VPI for packets coming on incoming VCI/VPIs.

The IP [7] was the first wide area connectionless protocol used extensively. Each IP packet at layer 3 includes its source and destination and moves independently through the network. The technique by which the packets were forwarded to the destination was termed *routing*. Even though routing is a form of switching, we can distinguish routing by its connectionlessness. The responsibility for maintaining the *state* of an end-to-end connection lies with the end nodes, and the network does not track a circuit, connection, or flow. The network nodes will be indifferent to

anything but handing off packets to the next node [22]. On the other hand, in a switching network, the intermediate nodes keep track of the connection—the path and bandwidth (which relates to quality). This distinction applies on all types of data links—SDH/SONET, ATM, and Ethernet.

3.9.4.2   IP Switching

The goal of this technology is to make IP faster and offer the QoS support. The approach is to discard the connection-oriented ATM software and implement the connectionless IP routing directly on top of the ATM hardware. An IP switch shown in Figure 3.27(a) uses an ATM switch with all control software above AAL 5 removed and replaced by standard IP routing software. To gain the benefits of switching, a mechanism has been defined to associate *IP flows* (sequences of packets that get the same forwarding treatment at a router) with ATM labels [21]. This approach takes advantage of the robustness of IP routing, as well as speed, capacity, and scalability of ATM switching.

An IP flow is characterized by the fields in IP/TCP/user datagram protocol (UDP) headers such as source and destination addresses and port numbers and IP protocol type. When a packet is assembled and submitted to the controller in an IP switch, apart from forwarding it to the next node, it also does *flow classification*. This is a policy decision local to an IP switch. Depending on the classification, the switch decides to forward (e.g., for short-term flows such as database queries) or switch (e.g., long-term flows such as file transfers) the next packets of that flow. Each packet's next hop and output port are determined by a table look up with the packets destination IP address as the key. Classification is used to derive output port queuing and scheduling rules. The forwarding table is established and updated by the management engine based on the decisions of the active IP routing protocols [23]. This process is shown in Figure 3.28(a).
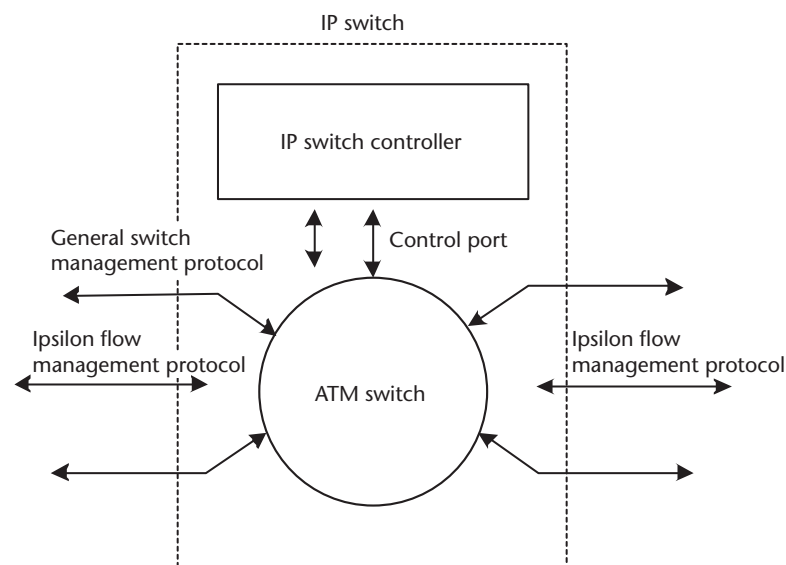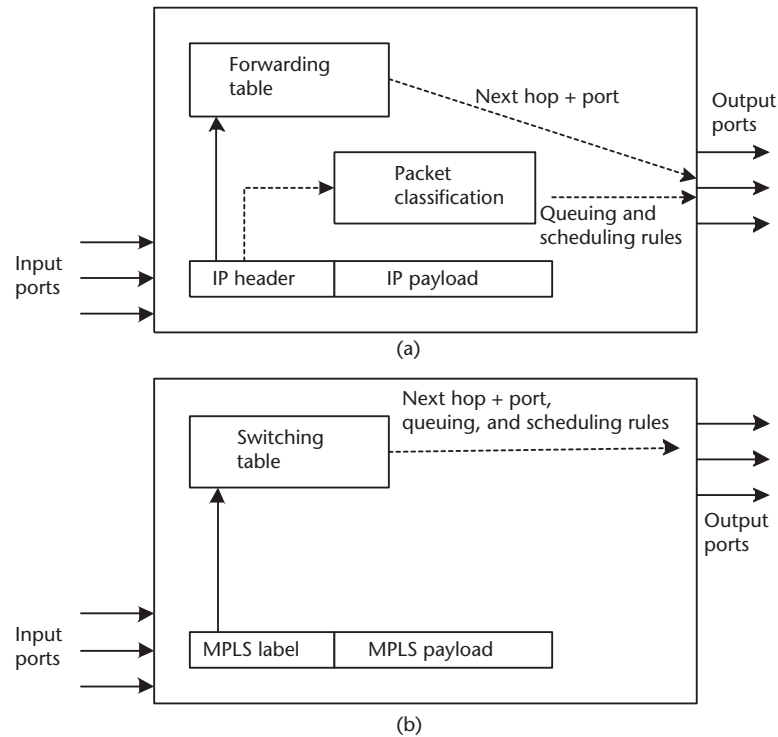

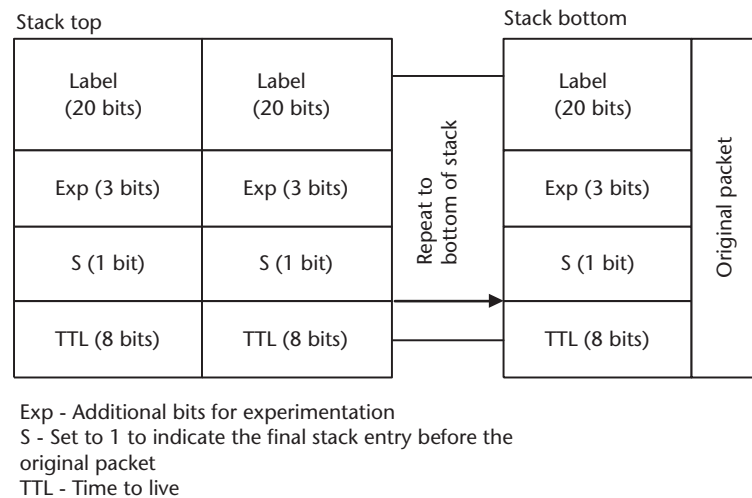**Figure 3.27**   Structure of an IP switch.

Figure 3.28   Operation of the forwarding engines in: (a) IP switching, and (b) MPLS. (*Source:* [23]. ©2000 IEEE.)

3.9.4.3   Label Switching

Label switching techniques such as multiprotocol label switching (MPLS) add the ability to forward packets over arbitrary nonshortest paths and emulate high-speed tunnels between IP-only domains. Each packet's forwarding treatment is entirely determined by a single index lookup into a switching table, as shown in Figure 3.28(b), using the packet's *label* and possibly the input port. The packet's label is replaced by a new next-hop label retrieved from the switching table, and the packet is queued at the appropriate output port for transmission. The switching table is loaded a priori with a unique next-hop label, output port, queuing, and scheduling rules for all current label values. This mapping information is established and managed by the management engine in response to external requests for a labeled path through the switch.

Figure 3.29 shows the structure of the generic MPLS frame. An MPLS label stack of one or more 32-bit entries precedes the payload (e.g., an IP packet). The label is 20 bits wide. Switching occurs on the label in the top (and possibly the only) stack entry. The stacking scheme allows label-switched paths (LSPs) to be tunneled through other LSPs. The separation of forwarding information from the content of the IP header allows MPLS to be used with devices such as optical cross-connects (OXCs), whose data plane cannot recognize the IP header [24].

MPLS forwarding is defined for a range of link layer technologies, some of which are inherently label switched, such as ATM, and others that are not, such as

Stack top   Stack bottom

Label (20 bits)	Label (20 bits)		Label (20 bits)	
Exp (3 bits)	Exp (3 bits)	Repeat to bottom of stack	Exp (3 bits)	Original packet
S (1 bit)	S (1 bit)		S (1 bit)	
TTL (8 bits)	TTL (8 bits)		TTL (8 bits)	

Exp - Additional bits for experimentation
S - Set to 1 to indicate the final stack entry before the
original packet
TTL - Time to live

Figure 3.29   The MPLS frame structure. (*Source:* [23]. ©2000 IEEE.)

packet over SONET/SDH. Reference [23] shows how the MPLS frames are placed within the link's native format in each case. ATM and FR switching is based on a link-layer copy of the stack top entry.

3.9.5   An Illustrative Modern Data Network

The FR network of David Pieris Information Technologies Ltd., Sri Lanka, provides extensive islandwide coverage. As seen in Figure 3.30, the FR backbone consists of five strategically located multiservice switches. These switches, with a switching capacity of 1.6 Gbit/s, can handle FR, ATM, and transparent HDLC traffic. The backbone operates on a digital microwave media running SDH at the physical layer and is configured as a ring for redundancy. The physical layer terminates in cross connects that interface the switches that provide FR, ATM, and transparent HDLC.

The last-mile connection from the closest network node to the customer premises is also wireless and are point-to-point or point-to-multipoint links connected through digital cross connects. The link speeds can vary from 64 kbit/s to 2 Mbit/s in $n \times 64$ kbit/s increments. At the customer premises, the access network terminates at a router or FRAD. The end-to-end VCs are configured and assigned CIRs in FR switches. Any type of multiprotocol traffic from one end point to the other can be routed through these VCs.

3.10   Switch Fabric Interfaces and ICs

Switch fabrics are fundamental building blocks in a wide range of switching systems. While many semiconductor manufacturers introduced basic semiconductors for voice switches in the early 1970s, ATM switch fabric ICs and interfaces were

introduced as relatively new components. Figure 3.31(a) indicates a generic over-view of a high-capacity switch fabric architecture. The ingress path connects a line card's network processing subsystem consisting of network processors and traffic mangers to the switching core, allowing the dynamic connection of ingress parts to the egress ports. The components for switching can be implemented in many ways, as indicated in Figure 3.31(b). Regardless of the architecture, however, most are input buffered where they queue the incoming cells or packets at the ingress stage until a scheduling mechanism signals them to traverse the switch core. In many implementations, the buffering occurs at the line card, and the switch cards contain limited memory.

A fabric interface chip (FIC) resides on each line card and interfaces to either a network processor or a traffic manager [25].
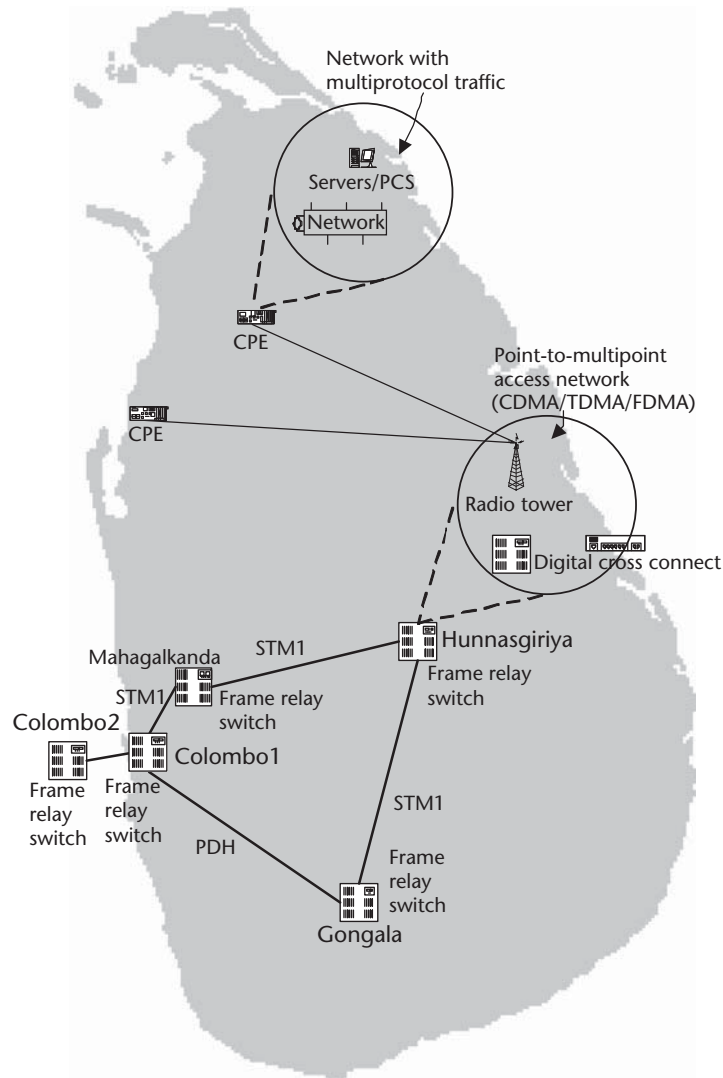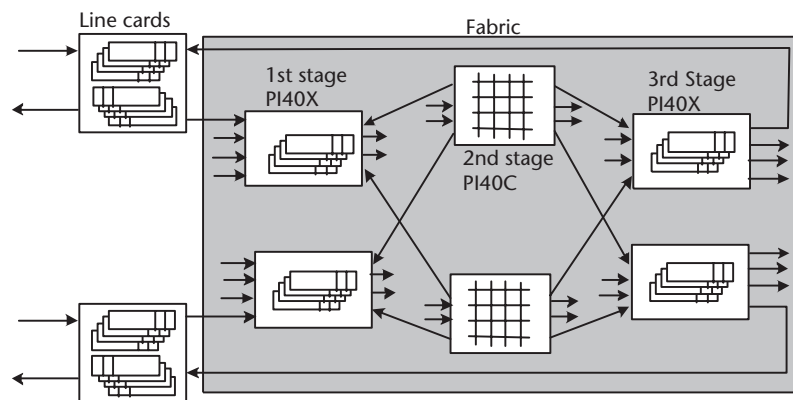

Figure 3.30 FR network provided by David Peiris Information Technologies Ltd. in Sri Lanka. (Reproduced with permission of David Pieris Information Technologies Ltd.)
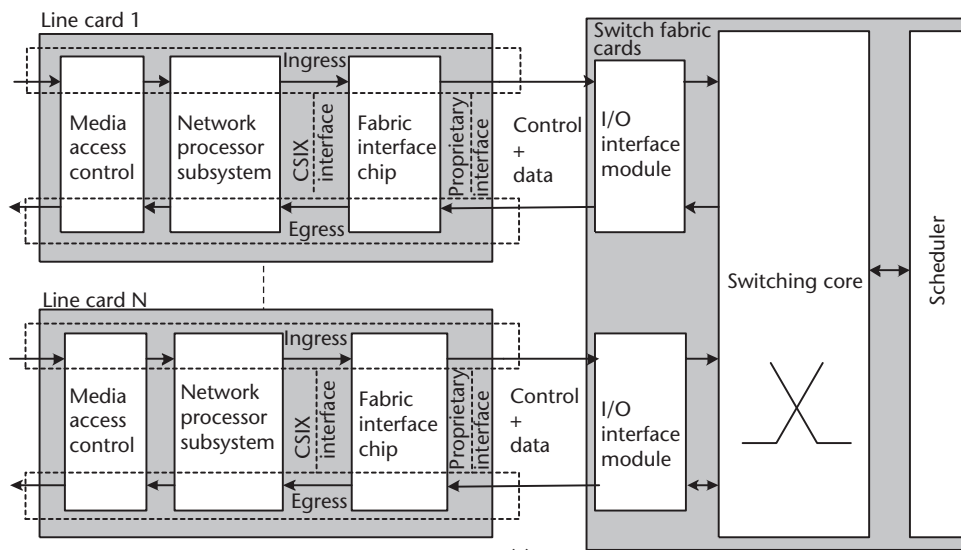
CSIX - Common switch interface
(a standard developed by Network Processing Forum)

(a)

(b)

Figure 3.31   Switch fabrics: (a) common switch fabric architecture, allowing multiple input ports to multiple output ports. (*Source*: [25], ©2003 IEEE), and (b) a switching systems based on Agere Systems switch fabric P140. (*Source*: [26].)

Semiconductor switch fabric ICs are gradually appearing from the product folios of the telecom component manufacturers, and time-division, space-division, or combination versions are common in the market. Some examples are indicated in [26–28]. Figure 3.31(b) indicates a 40-Gbit/s switch fabric configured using Agere Systems P140 switch fabric with line cards for an ATM or IP switching. Details are beyond the scope of the chapter and [20, 26–29] are recommended.

3.11   Optics and the Future

Switching speeds achievable using semiconductors cannot keep up with the transmission capacity offered by optical systems. Optical systems can achieve

tremendous transmission capacities. However, in every switching node, O/E conversion and vice versa (E/O conversion) need be carried out, posing a bottleneck to high-speed transmission. With the development of optical components and technologies, the transfer of switching function from semiconductors to optics will result in a massive overall improvement of information transfer. The main developments in this direction are optical packet switching (OPS), optical burst switching (OBS), and generalized multiprotocol label switching (GMPLS).

GMPSL, an emerging label-switching technology, supports not only devices that perform packet switching, but also those that perform switching in time, wavelength, and space domains [24].

While GMPLS provides bandwidth at a granularity of a wavelength, OPS can offer an almost arbitrary fine granularity, while OBS lies in between [30]. Several applications of optical switch fabrics are under research, and some important ones are OXCs, protection switching, optical add/drop multiplexing (OADM), optical signal (spectral) monitoring (OSM), and network provisions. Optical switch fabrics, where switching is to occur directly in the optical domain, avoids the need of O/E/O conversions. Some promising technologies for the future are optomechanical switches, microelectromechanical systems (MEMs), electro-optic switches, and thermo-optic switches. A discussion on these is beyond the limits of this chapter, and [30–33] are suggested for details.

Today's data networks have four layers: IP for carrying applications and services, ATM for traffic engineering, SONET/SDH for transport, and dense wavelength-division multiplexing (DWDM) for capacity. With such multilayer architectures, any one layer can limit the scalability of the network, as well as add to the cost. Figure 3.32 shows the evolution of networks towards all-photonic technologies.

Meanwhile leading telecom system manufacturers are developing futuristic architectures to couple multiservice networks, such as that by Ericsson discussed in [34].
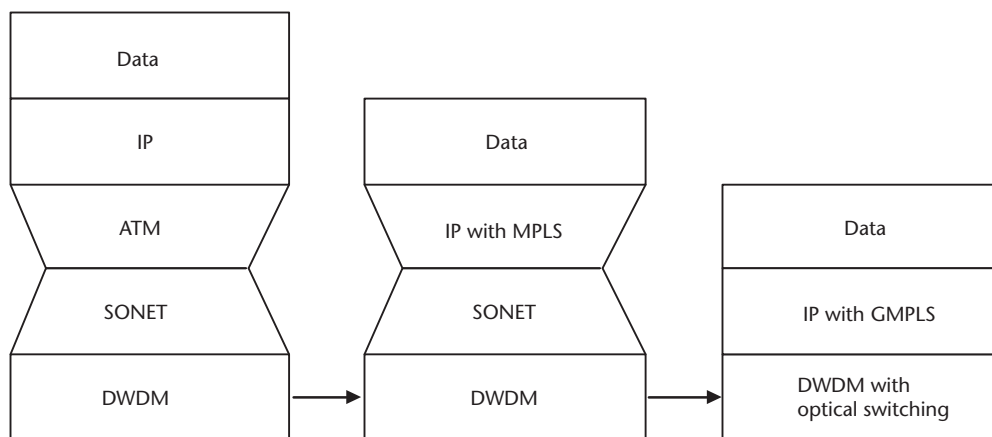
Figure 3.32   The evolution towards photonic networking. (*After:* [24].)

References

[1]   Reeves, A. H., "The 25th Anniversary of PCM," *IEEE Spectrum,* May1965, pp. 53–63.

[2]   Joel, A. E., "The Past 100 years in Telecommunications Switching," *IEEE Communications Magazine,* May 1984, pp. 64–83.

[3]   Joel, A. E., "Digital Switching—How It Has Developed," *IEEE Transactions on Communications,* July 1979, pp. 948–959.

[4]   Thompson, A.R., "Telephone Switching Systems," Artech House Publishers, 2000.

[5]   Ericsson and Telia, Ericsson Telecom AB, Telia AB, Studentlitteratur AB, 1998. *Understanding Telecommunications—Volume 2,* Lund, Sweden: Studentlitteratur, 1998.

[6]   Dixit, S. and Elby, S., "Frame Relay & ATM Networking," *IEEE Communications*, June 1996, pp. 64–82

[7]   Stallings, W., *Data and Computer Communications*, Seventh Edition, Englewood Cliffs, NJ: Prentice Hall, 2003.

[8]   Bellamy, J. C., *Digital Telephony,* Third Edition, New York: John Wiley, 2000.

[9]   Fujiyama,Y., H. Masuda, and K. Mano, "Architecture Enhancement of Digital Switching System," Vol. V, *Proc. XII International Switching Symp.*, Sweden, 1990, pp. 217–220.

[10] Milstead, R. M. and P. S. Warwick, "The Digital PBX: Current and Future Views of Applications for Information Environment," *Proc. Int'l. Conf. on Private Switching Systems and Networks,* June 1988, pp. 93–98.

[11] Foote, D. L. and M. W. Medin, "Extending ISDN Functional Layering Concepts in PBX Architectures," *Proc. Int'l. Conf. on Private Switching Systems and Networks*, June 1988, pp. 1–5.

[12] Kerr, G. W., "A Multisite Videophone PBX System," *Proc. Int'l. Conf. on Private Switching Systems and Networks,* June 1988, pp. 93–98, 215–219.

[13] Hamada, K. I., et al., "Digital Cordless Telephone Equipment for Wireless PBX Systems," *Proc. 43rd IEE Vehicular Technology Conf.,* May 1993, pp. 963–966.

[14] Bonzoano, L. and V. Palestini, "DECT Performance in the Wireless PBX Application," *Int'l. Conf. on Communication,* May 1993, Vol. 2, pp. 1269–1272.

[15] Lin, P., and Y. Lin, " Implementation and Performance Evaluation for Mobility Management of a Wireless PBX Network," *IEEE Journal on Selected Areas in Communications*, Vol. 19, No. 6, June 2001, pp. 1138–1146.

[16] Lai, W. R., and Y. B. Lin, "Resource Planning for Wireless PBX Systems," *Proc. First IEEE Enterprise Networking Mini Conf.,* June 1997, pp. 63–72.

[17] Frame Relay Forum, "ATM in Europe: The User Handbook," Ver. 1.0, July 1997.

[18] Anderson, J., et al., " Operations Standards for Global ATM Networks: Network Element View," *IEEE Communications Magazine,* December 1996, pp. 72–84.

[19] Jain, R., and G. Babic, "Performance Testing Effort at the ATM Forum: An Overview," *IEEE Communications,* August 1997, pp. 110–116.

[20] Hou, Y. T., L. Tassiulas, and H. J. Chao, "Overview of Implementing ATM Based Enterprise Local Area Network for Desk Top Multimedia Computing, *IEEE Communications Magazine,* April 1996, pp. 70–76.

[21] Agrawal, S., "IP Switching," August 1997, http://www.cis.ohio-state.edu/~jain/cis788-97/ip_switching/index.htm.

[22] "Switching Versus Routing," *Network Magazine,* http://www.networkmagazine.com/shared/article/showArticle.jhtml?articleId=8702661.

[23] Armitage, G., "MPLS: The Magic Behind the Myths," *IEEE Communications Magazine*, January 2000, pp. 124–131.

[24] Banerjee, A., et al., "Generalized Multiprotocol Label Switching: An Overview of Routing and Management Enhancements," *IEEE Communications Magazine*, January 2001, pp. 144–150.

[25] Elhany, I., K. Busch, and D. Chiou, "Switch Fabric Interfaces," *IEEE Computer*, September 2003, pp. 106–108.

[26] Agere Systems, "Protocol Independent Switch Fabrics," product brief, May 2002.

[27] Agere Systems, "-X/-C Protocol Independent (ATM/IP) Switch Elements," product brief, May 2001.

[28] Agere Systems, "Protocol Independent Switch Fabric (P140SAX)," advance product brief, October 2002.

[29] Goldberg, L., "Fractional Terabit ATM Switch Demonstrated: New Modular Architecture Accommodates User's Growing Bandwidth Needs," *Electronic Design,* February 1997, pp. 31–32.

[30] Papadimitriou, G. I., C. Papazoglou, and A. S. Pomportsis, "Optical Switching: Switch Fabrics, Techniques and Architectures," *IEEE Journal of Lightwave Technology,* Vol. 21, No. 2, February 2003, pp. 384–405.

[31] Massetti, F., et al., "High Speed, High Capacity ATM Optical Switches for Future Telecommunication Transport Networks," *IEEE Journal on Selected Areas in Communications,* Vol. 14, No. 5, June 1996, pp. 979–998.

[32] Forbes, M., J. Gourlay, and M. Desmulliez, "Optically Interconnected Electronic Chips: A Tutorial and Review of Technology," *Electronics and Communication Engineering Journal,* October 2001, pp. 221–232.

[33] Neukermans, A., and R. Ramasawamy, "MEMS Technology for Optical Networking Applications," *IEEE Communications Magazine,* January 2001, pp. 62–69.

[34] Hallenstal, M., U. Thune, and G. Oster, "ENGINE Server Network," *Ericsson Review,* No. 3, 2000, pp. 126–149.

[35] European Market Awareness Committee: "ATM in Europe: The Use Handbook," The ATM Forum, version 1.0, July 1997.

[36] Lee, B. G., and Kim, W., *Integrated Broadband Networks: TCP/IP, ATM, SDH/SONET, and WDM/Optics*, Norwood, MA: Artech House Inc., 2000.

[37] ITU-T: "Q.7000: Specifications of Signalling System No. 7" Revision 03/93.