

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.

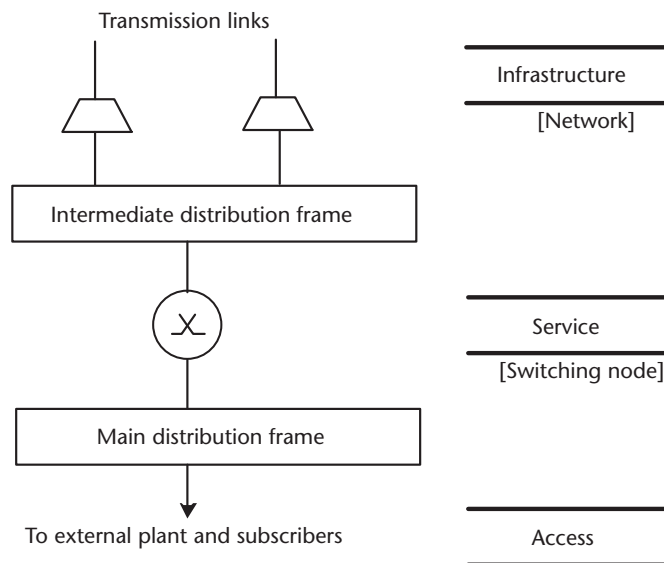


Figure 3.1 Three layers of the PSTN.

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

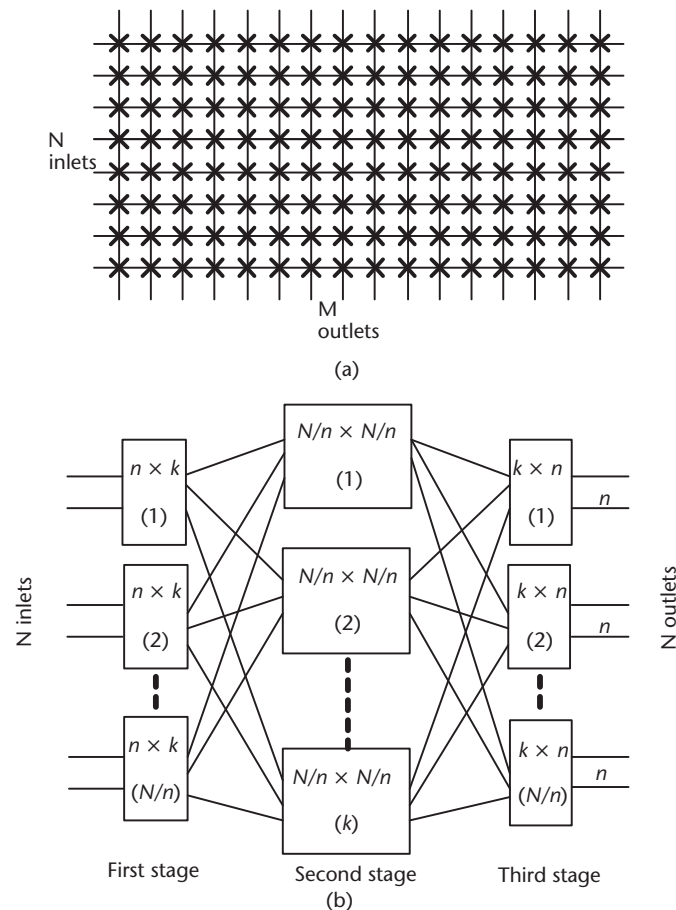


Figure 3.2 Switching matrices: (a) rectangular cross point array, and (b) three-stage switching matrix.

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.

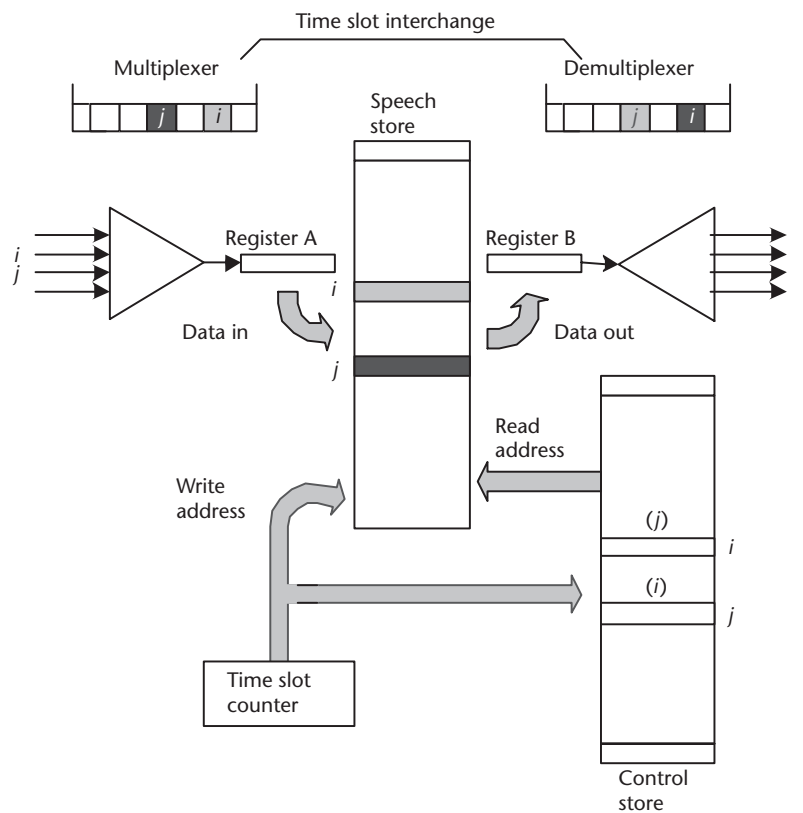


Figure 3.3 TSI with multiplexer/demultiplexer and memory switch.

Because a write and a read are required for each channel (sampled every $125 \mu\text{s}$) 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/E1 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.

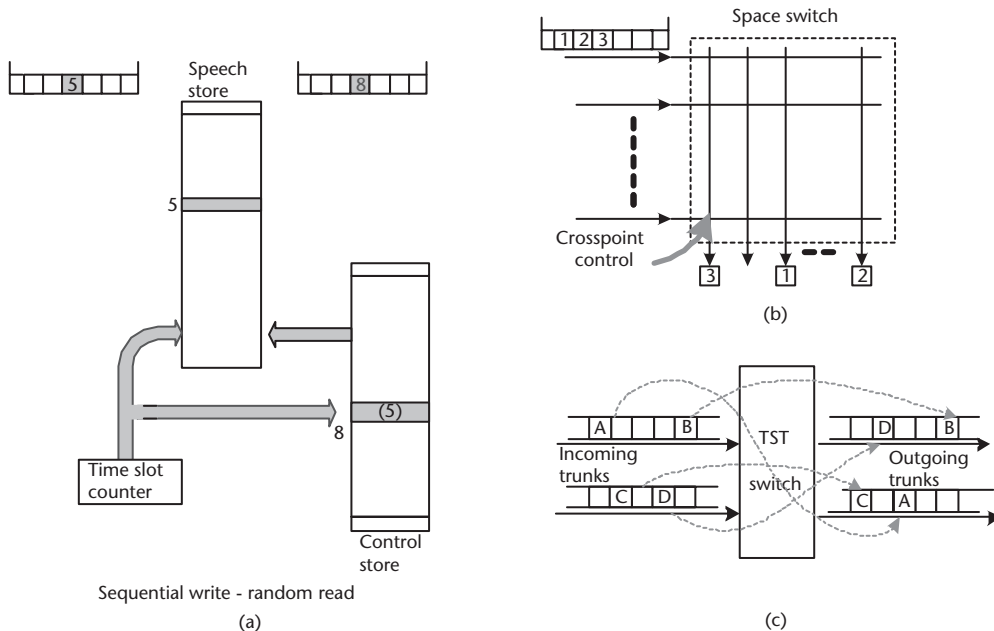


Figure 3.4 Switch modules: (a) time switch modules, (b) space switch, and (c) effect on the incoming and outgoing streams.

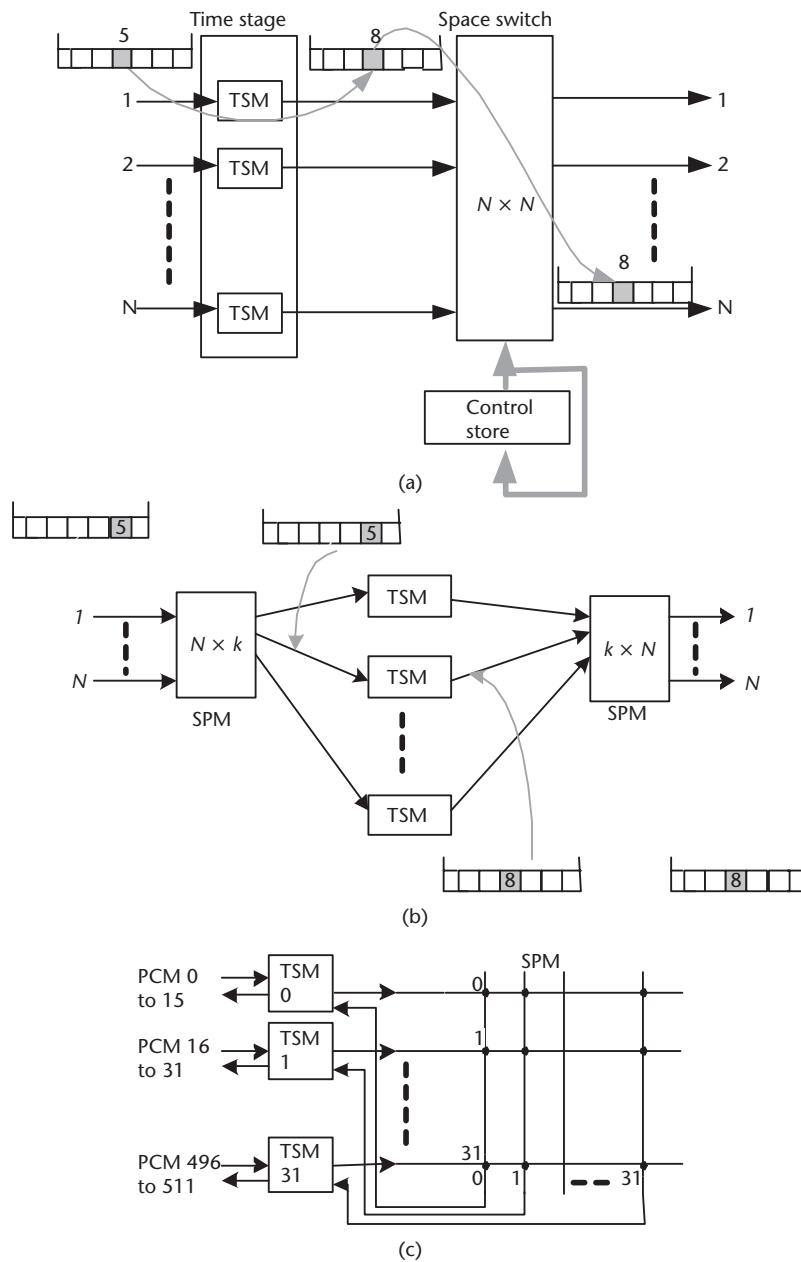


Figure 3.5 Multidimensional switches: (a) TS switch, (b) STS switch, and (c) TST with multiple PCM connections.

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 $16 \times 32 = 512$ multiple positions (MUP). Thirty two such modules connected to an SPM provide a capacity of $512 \times 32 = 16,384$ 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

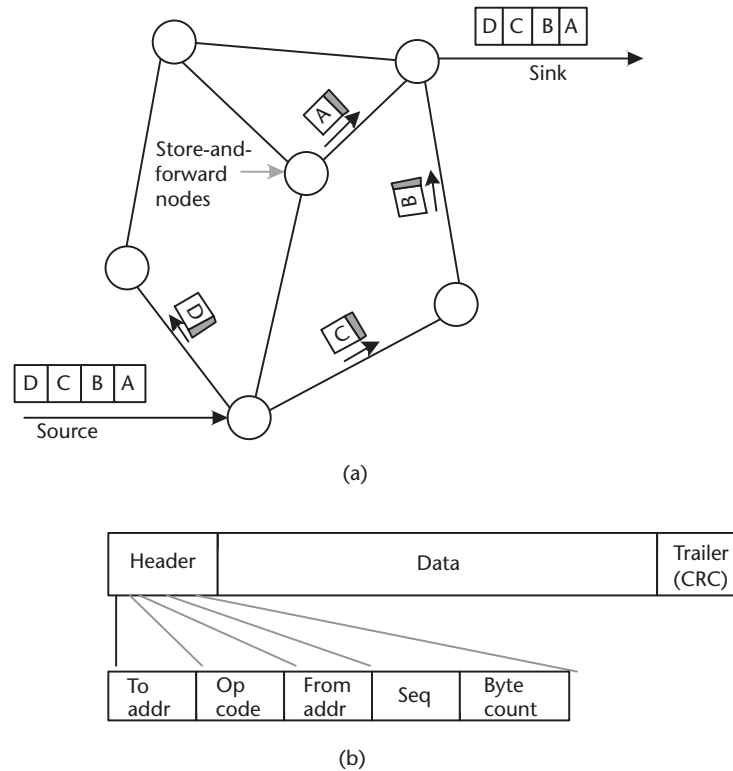


Figure 3.6 Packet switching concepts: (a) simplified network, and (b) typical packet format.

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

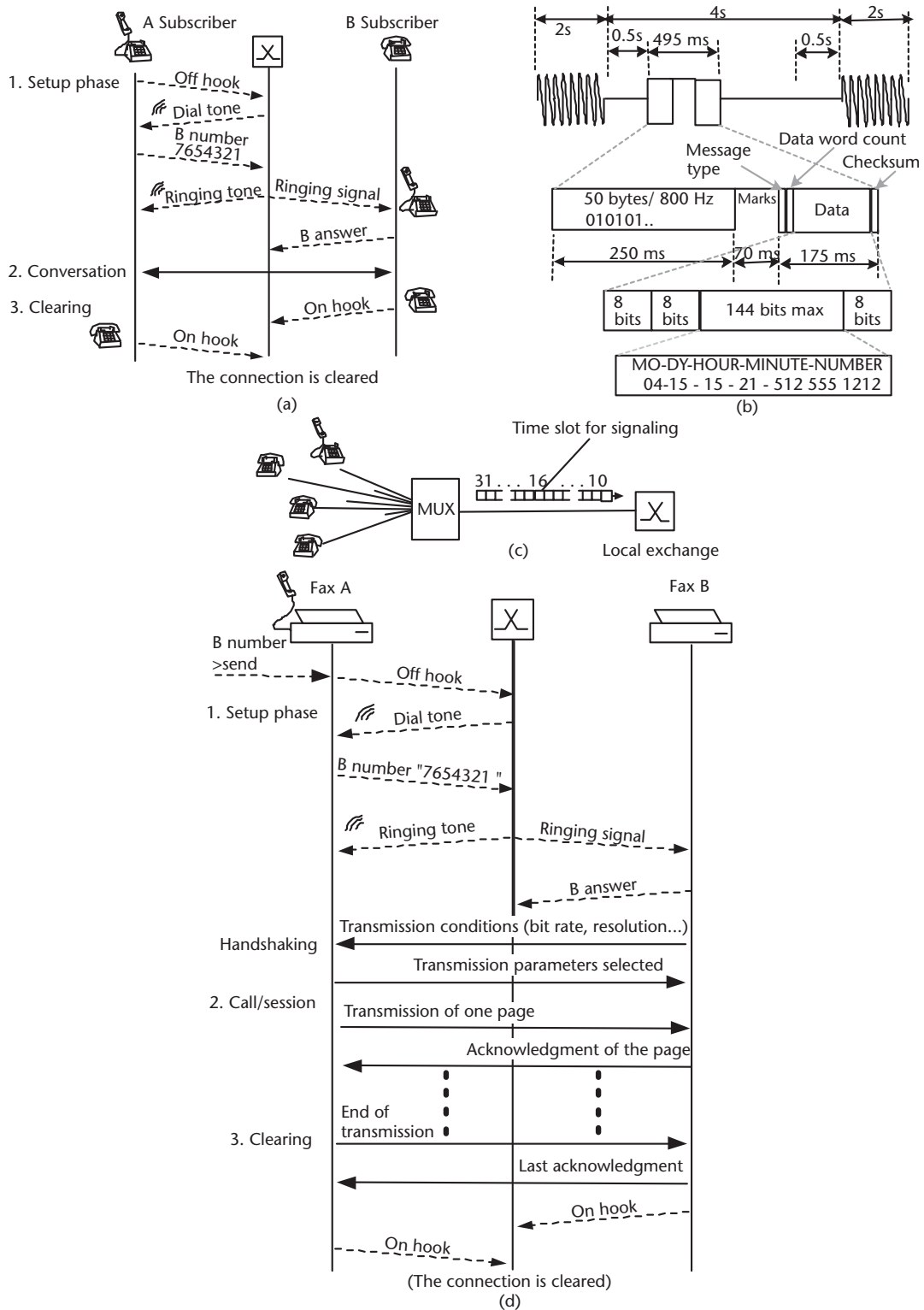


Figure 3.7 Signaling examples in voice systems: (a) establishing a basic voice connection, (b) a typical CLI signal, (c) signaling between a subscriber multiplexer and an LE, and (d) fax communication.

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

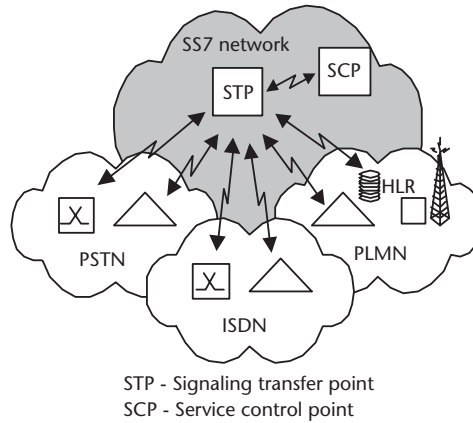
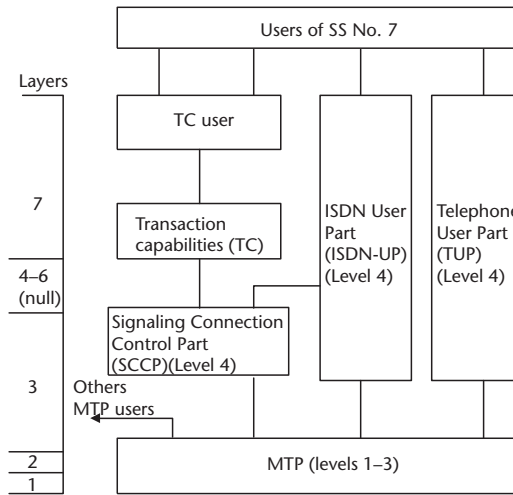
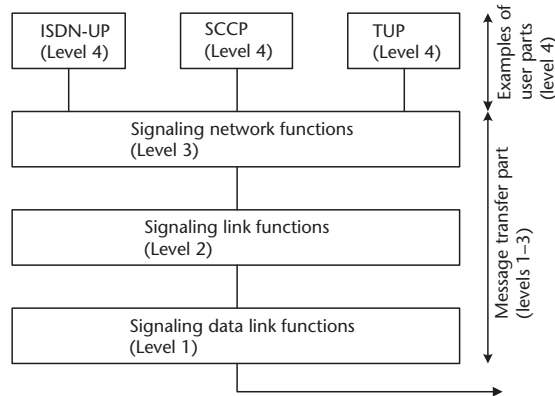


Figure 3.8 SS7 operation within the overall network.

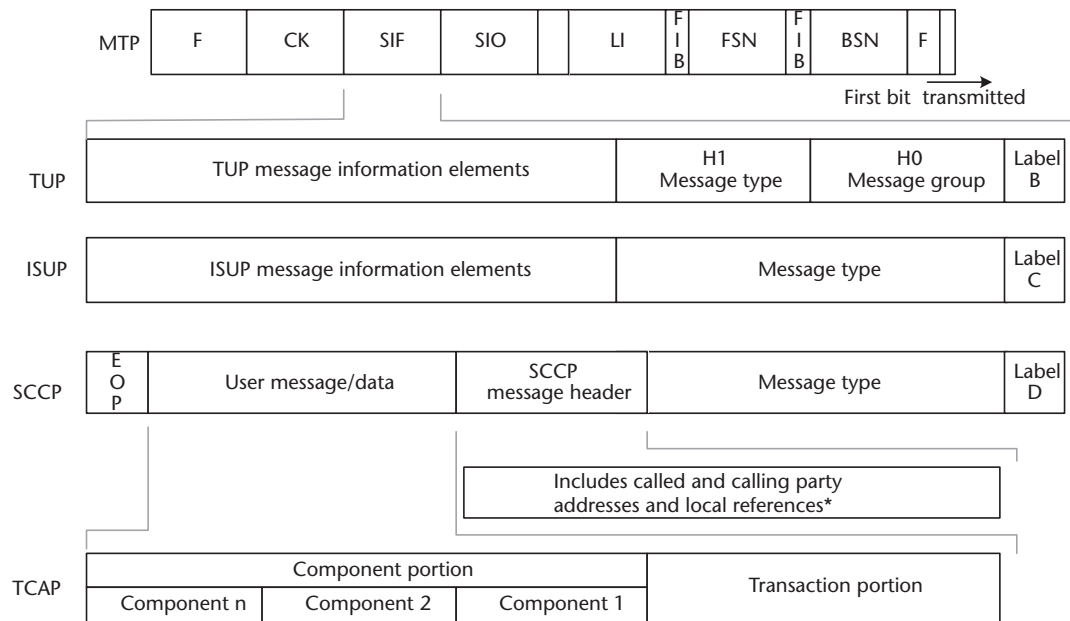


(a)



(b)

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



* 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

<i>User Part/Application Part</i>	<i>Application</i>	<i>ITU-T Recommendations</i>
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

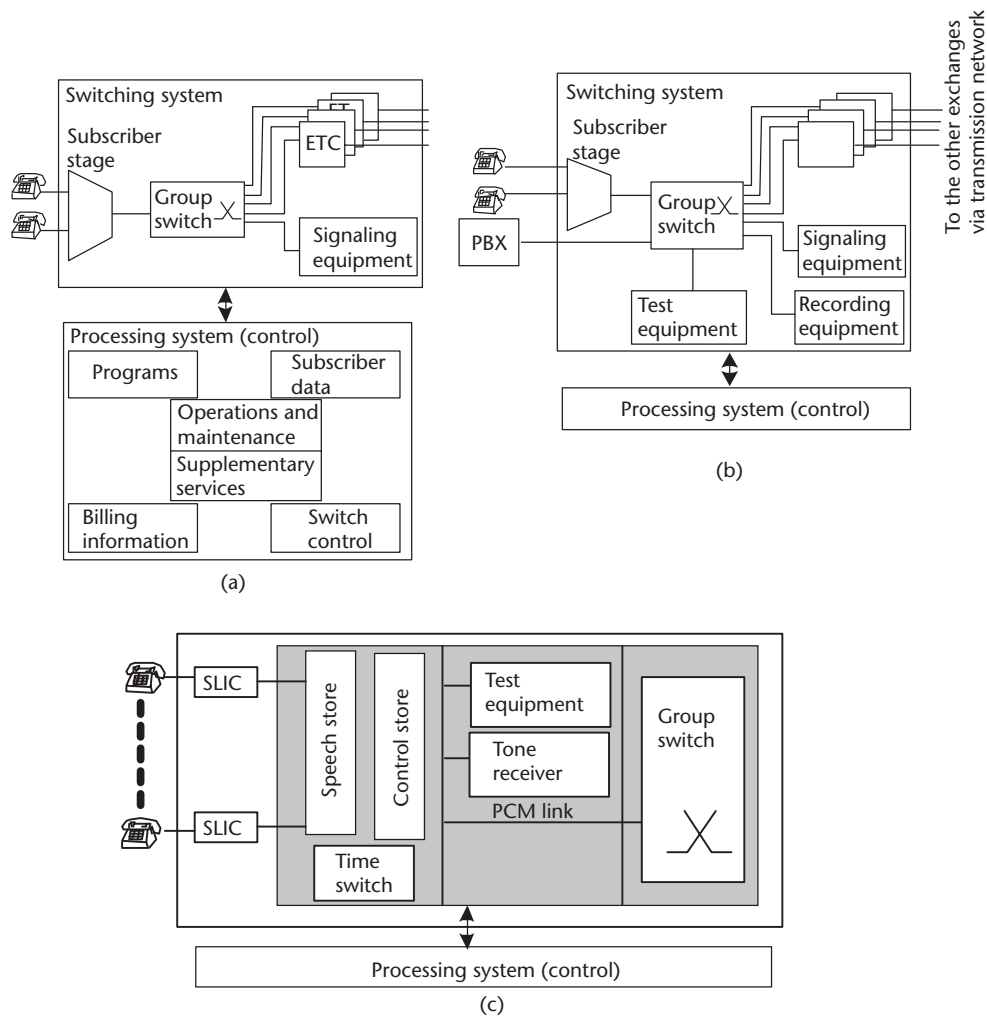


Figure 3.10 A simplified view of local exchange (a) concept, (b) equipment connected to the group switch, and (c) subscriber stage.

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 circuits (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

<i>Phase</i>	<i>Tasks</i>
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 <i>B</i> -subscriber number and associated connection information. Use the connection of a tone receiver for detecting DTMF signaling.
Phase 2—Figure 3.11(b): <i>B</i> -subscriber number detection	This requires analysis of the <i>B</i> -subscriber number for determining <i>B</i> -subscriber location, charging function, and length of the number dialed. If the <i>B</i> subscriber is within the local exchange, send a query to the database for the SLIC of the <i>B</i> subscriber, the level of incoming call barring of <i>B</i> , and special services allowed for <i>B</i> . If <i>B</i> 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 <i>A</i> subscriber's exchange indicating that <i>B</i> 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 <i>B</i> subscriber's exchange sends ringing signals/tones. On lifting of the <i>B</i> handset, the voice samples are switched and activation of charging and call monitoring begins.
Phase 4—Figure 3.12(d): Call release	If <i>A</i> concludes the call, the call is released without a noticeable delay and any time supervision (as <i>A</i> pays in general). If <i>B</i> hangs up first, a supervisory period is initiated allowing any other connections <i>A</i> 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.

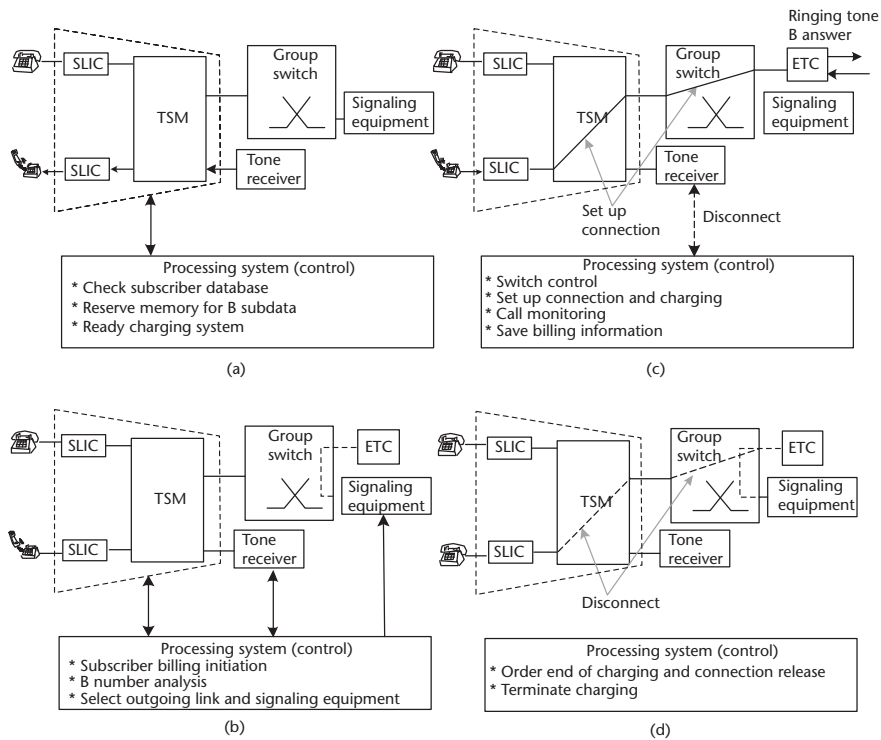


Figure 3.11 Process and connection stages of a phone conversation: (a) A-subscriber lifting a hand set, (b), B-number analysis for the outgoing connection, (c) outgoing call set up, and (d) release of the call.

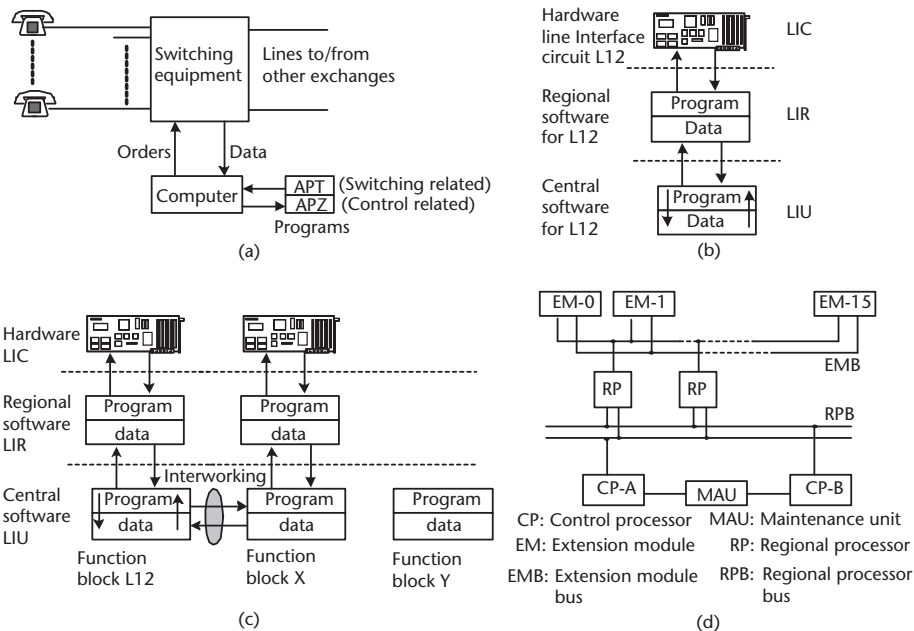


Figure 3.12 An example of switching and control systems in a typical exchange—AXE architecture from Ericsson: (a) basic concept, (b) an example of a function block, (c) interworking of function blocks, and (d) processor hierarchy. (Source: L. M. Ericsson, Sweden.)

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

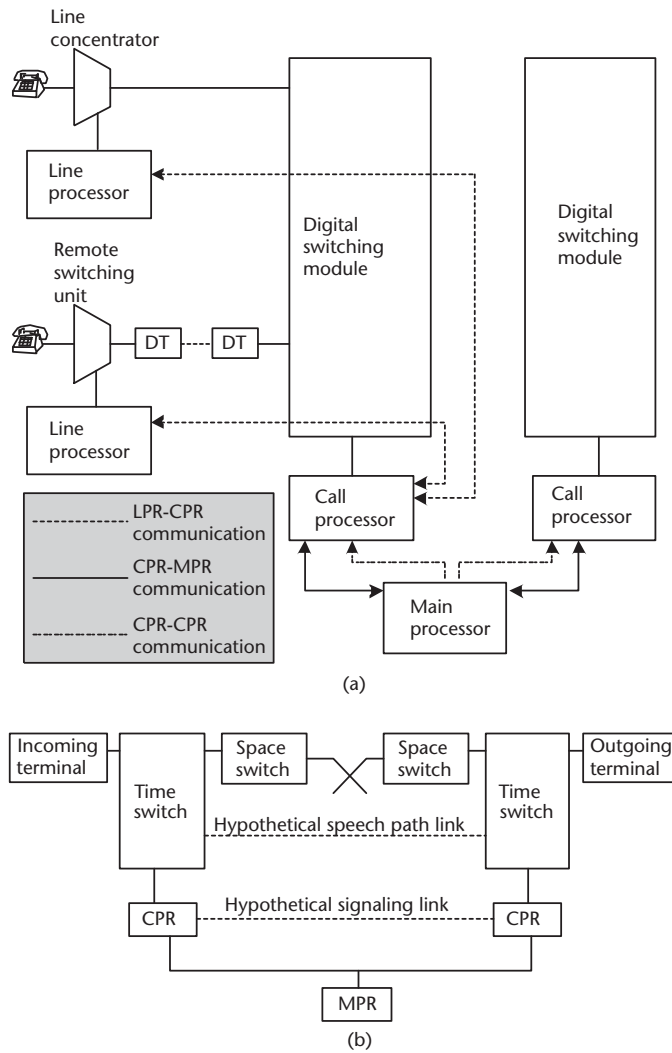


Figure 3.13 Processor hierarchy in Fujitsu FETEX-150 system: (a) basic concept, and (b) hypothetical independent exchange method. (Source: [9]. ©1990 IEEE.)

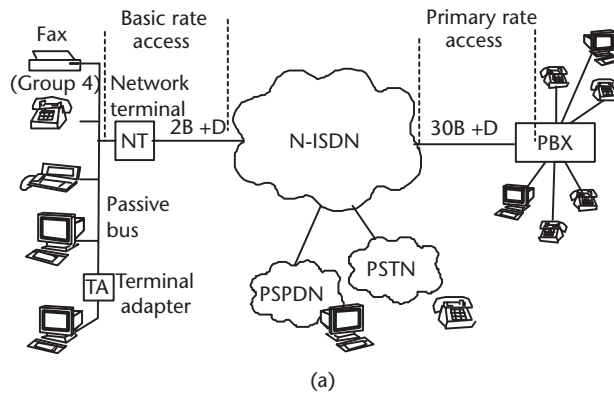


Figure 3.14 ISDN concept.

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

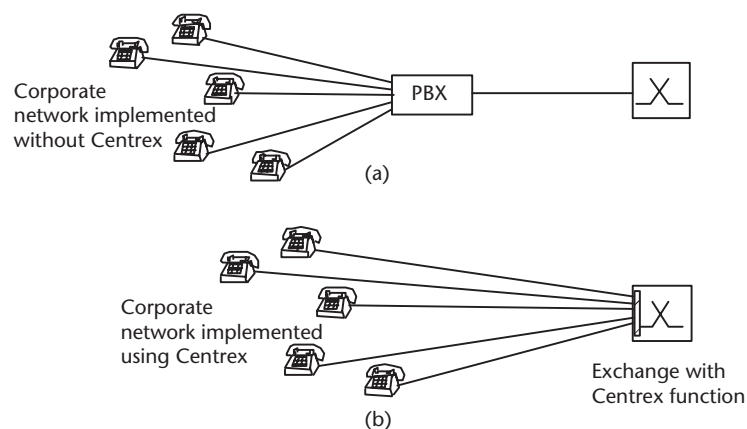


Figure 3.15 Comparison of PBX and Centrex systems: (a) PBX, and (b) Centrex.

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.

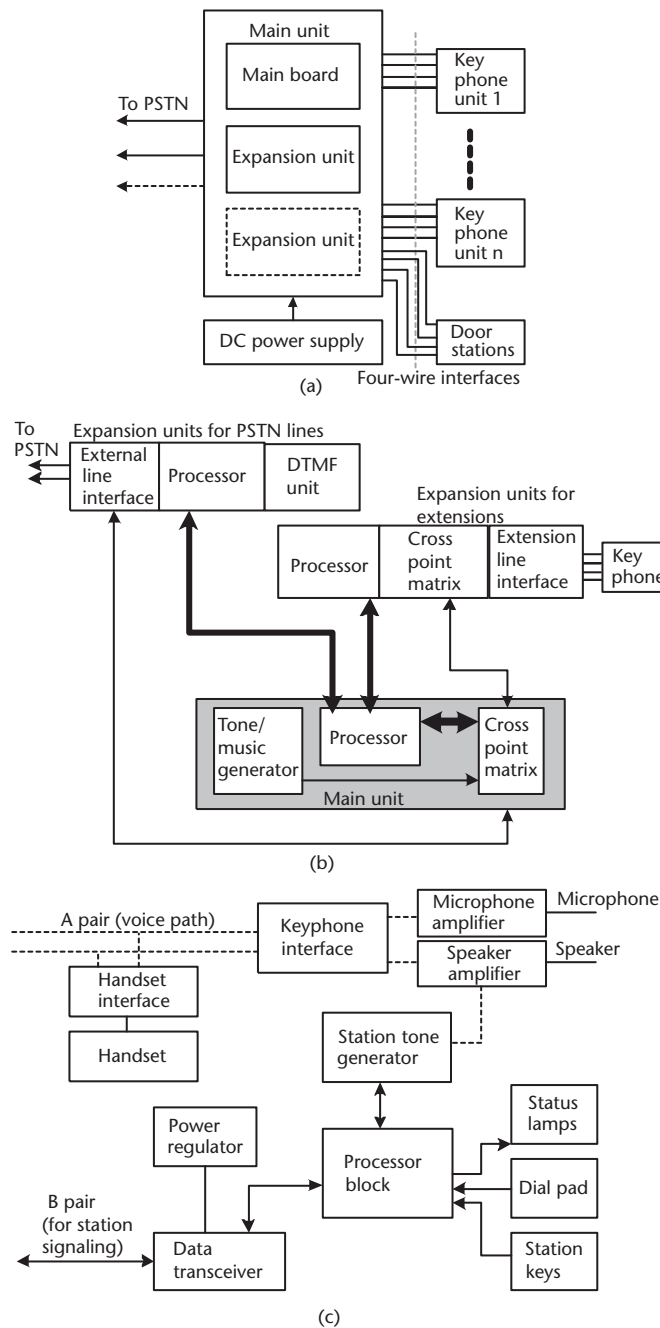


Figure 3.16 Components of a typical key phone system: (a) system interconnections, (b) block representation, and (c) key station block diagram.