(iii) local event,
(iv) user services,
(v) used services.

6.37 Using the abbreviated names listed in Figure 6.32, show how the idle RQ primary can be specified in the form of:
(i) a state transition diagram,
(ii) an extended event–state table,
(iii) a high-level pseudocode program.

6.38 With the aid of a time sequence diagram, show a typical set of link layer service primitives assuming the link layer provides
(i) a reliable service and
(ii) a best-effort service.

Section 6.8

6.39 In relation to the HDLC frame format shown in Figure 6.36, explain the meaning and use of the following terms:
(i) supervisory frames,
(ii) unnumbered frames,
(iii) poll/final bit,
(iv) command and response frames,
(v) extended control field bit definitions,
(vi) piggyback acknowledgment,
(vii) unnumbered information frame.

6.40 With the aid of the frame sequence diagram shown in Figure 6.37, explain how a corrupted I-frame is overcome in the normal response mode. Include in your explanation the use of the P/F bit.

6.41 With the aid of the frame sequence diagram shown in Figure 6.38, explain how the piggyback acknowledgment procedure works in the asynchronous balanced mode.

6.42 With the aid of the frame sequence diagram shown in Figure 6.39, describe the window flow control procedure used with the UDLC protocol.

7

Circuit-switched networks

7.1 Introduction

Although in Chapter 1 we discussed the operation and applications of a PSTN and an ISDN separately, in practice, as we shall expand upon later, both network types utilize the same core transmission and switching network to provide the related services and the only difference between the two networks is the way subscribers gain access to the core network. In this chapter, therefore, we shall discuss the operation of both a PSTN and an ISDN under the general heading of public circuit-switched networks.

All public circuit-switched networks consist of three hierarchical subnetworks:

- a relatively large number of **local access and switching networks**: these connect subscribers within a localized area to their nearest local exchange (LE) or end office (EO) and are concerned with the transmission and switching of calls within their own area;
- one or more **interexchange trunk/carrier networks**: these are national networks concerned with the transmission and switching of calls between different regional and national exchanges/offices;

- an interconnected set of **international networks**: associated with each national network is an international gateway exchange (IGE) and collectively these are concerned with the transmission and switching of calls internationally between the different national networks.

This general architecture is shown in Figure 7.1.

In some countries, the total national network is owned and managed by a single operator. In most countries, however, the various parts of the network are privately owned and managed by a number of operators. In some cases, the total local access and switching network is owned by a single operator and the various interexchange trunk networks are each owned by different operators. In others, there is one set of operators that own and run different parts of the local access network – known as **local exchange carriers (LXCs)** – and a different set of operators that run their own interexchange trunk networks – **interexchange carriers (IXCs)**. Nevertheless, the same basic architecture that we show in Figure 7.1 can be used to explain the principle of operation of this type of network. In practice, the overall network consists of three interrelated systems: transmission, switching, and signaling and hence we shall describe their operation under these headings.

Transmission system

Each subscriber within a localized area is connected to an LE/EO in that area by a dedicated circuit which, in the case of most homes, is a twisted-pair wire cable. This is known as the **customer line** or, more generally, the **subscriber line** (**SL**) as it is used exclusively by that subscriber. In the case of a PSTN, analog transmission is used over each subscriber line and hence, as we described in Section 2.5.1, all the signals relating to a call are analog signals within a bandwidth of 200 Hz through to 3.4 kHz. With an ISDN, digital transmission is used over the subscriber line which is then known as a **digital subscriber line** (**DSL**) since all the signals relating to a call are digital.

In the case of a customer premises which contains a PBX, as we explained earlier in Section 1.2, multiple calls can be in progress concurrently. Hence a digital subscriber line is used with a bit rate sufficient to support multiple simultaneous calls. Typical bit rates are 1.5 Mbps or 2 Mbps – or multiples of these – the first supporting 24 calls and the second 30 calls. A similar set of transmission circuits is used to interconnect the various exchanges within the network except, since the number of simultaneous calls can be much larger, optical fiber cables are used which operate at very high bit rates and hence can support many thousands of simultaneous calls.

As we explained in Section 1.3.1, in addition to telephony, a PSTN also provides a number of digital services including fax, access to the Internet, and high-speed access to various entertainment servers. In order to do this, modems are used to convert the digital signals associated with these applications into and from the analog signals used over the subscriber line. In the case of fax and Internet access, since these involve a switched connection similar to that used for telephony, low bit rate modems of up to 56 kbps are used. In the case of entertainment applications that do not utilize the switching network, however, modems supporting in excess of 1.5 Mbps are used.

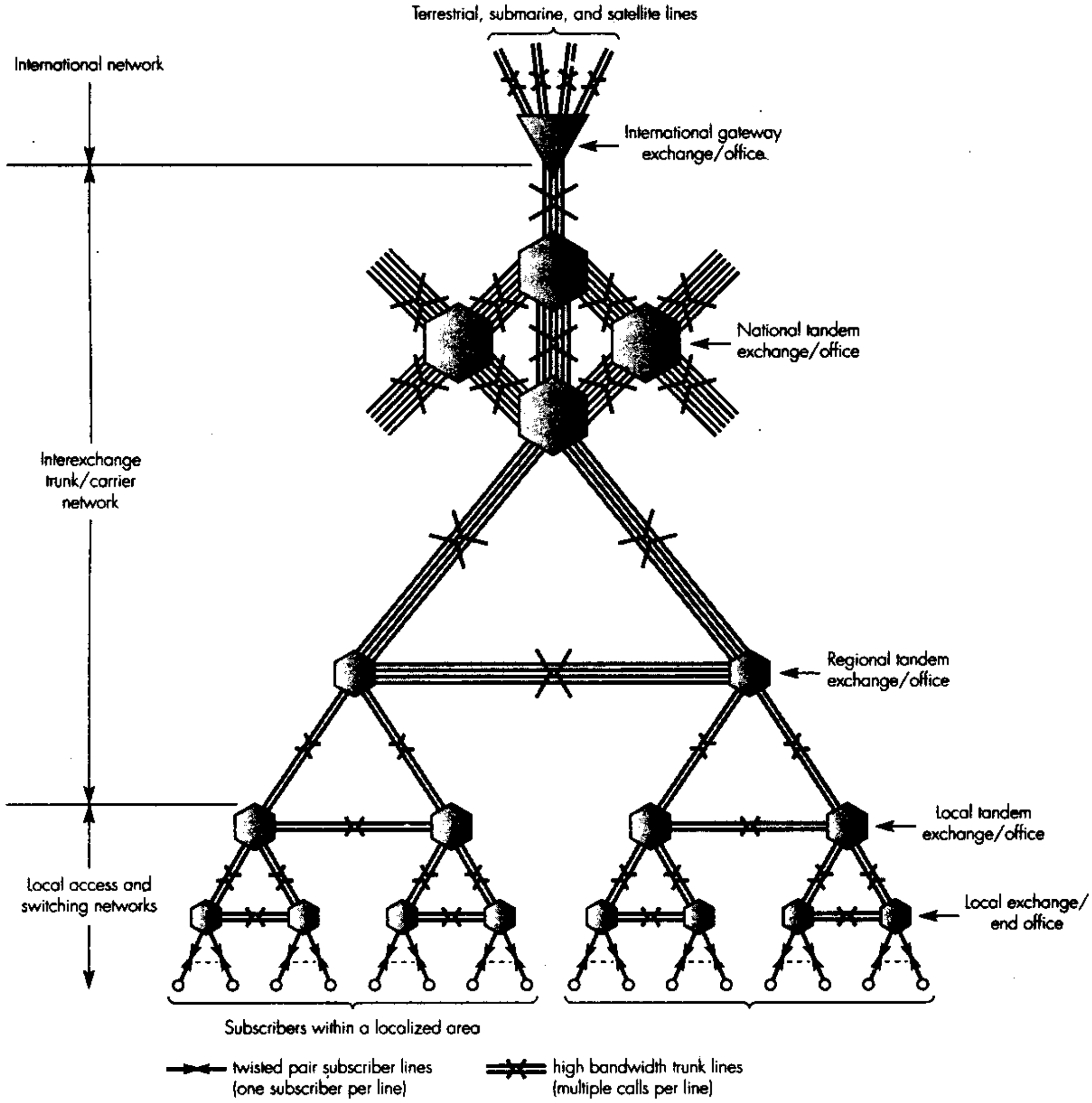


Figure 7.1 General architecture of a national circuit-switched network.

Switching system

Each LE/EO within a local area has sufficient switching capacity to support a defined number of simultaneous calls/connections. In some instances, the calls are between two subscribers that are connected to the same LE/EO while in others they are between two subscribers connected to different exchanges.

In the first case, as we can see from Figure 7.1, a connection can be set up directly by the LE/EO without any other exchanges being involved. In all other cases, however, additional exchanges are involved, the number determined by the location of the LE/EO to which the called subscriber is connected.

- If the LE/EO is within the same local area, the connection involves just the two interconnected LEs/EOs.
- If the LE/EO is in a neighboring local area, the connection is through a local tandem exchange/office.
- If the LE/EO is in a different region, the connection is through either a pair of neighboring regional tandem exchanges/offices or, one or more additional higher-level national tandem exchanges/offices.
- If the LE/EO is in a different country, the connection is through the complete set of exchanges in each country including the two international gateway exchanges involved.

As we can deduce from the interconnection structure illustrated in Figure 7.1, within the total switching network, there are a number of alternative paths/routes between any two exchanges. The additional lines are provided both to increase capacity and to improve the resilience of the total network to exchange and/or line failures.

Signaling system

As we saw earlier in Section 1.2, both PSTNs and ISDNs operate in a circuit-switched mode. This means that, prior to a call taking place, a connection through the network between the two subscribers must be set up and, on completion of the call, the connection closed down. The setting up and closure of connections is carried out by the transfer of a defined set of control messages – known as **signaling messages** – between the calling and called subscriber and their LE/EO and also between the various exchanges that are involved.

The signaling messages used over the subscriber line are different from those used within the core transmission and switching (trunk) network. In the case of an analog subscriber line, the signaling messages are analog signals such as single-frequency audio tones. With a digital line, the signaling messages are also digital and, because an ISDN can support two (or more) calls simultaneously, the signaling messages are allocated a dedicated portion of the bandwidth/bit rate of the line. Within the trunk network, however, the signaling messages relating to both types of network are digital and use a common format and signaling protocol.

Thus, as we have stated, public circuit-switched networks comprise three interrelated systems: transmission systems, switching systems, and signaling systems. Hence in the remainder of this chapter we shall describe the underlying principles associated with each of these systems. In addition, since modems play an important role in the provision of multimedia communication services with public circuit-switched networks, we shall also discuss the principle of operation of the different types of low bit rate modem. We shall defer discussion of high bit rate modems until Chapter 11 where we discuss the different types of entertainment networks.

7.2 Transmission systems

As we explained in the last section, the transmission system comprises two parts: that used in the local access network and that used in the trunk network. The type of transmission used in the local access network can be either analog (PSTN) or digital (ISDN). Also, although all-digital transmission (and switching) is used in the trunk network, for historical reasons, there are two different types of digital transmission system used: one called the **plesiochronous digital hierarchy (PDH)** and the other the **synchronous digital hierarchy (SDH)**, the latter also known as the **synchronous optical network** or **SONET**. We shall discuss the operation of each type of system separately.

7.2.1 Analog subscriber lines

As we mentioned in the introduction to the chapter, each subscriber line comprises a single twisted-pair wire that connects the subscriber network termination to a **line termination unit (LTU)** in the LE/EO. In practice, each subscriber line is made up of multiple cable sections with a combined length of up to about 5 miles (8 km) depending on the telecommunications operating company (teleco). In order to enable new customers to be added and faults on individual lines to be located, each subscriber line within a localized area is terminated at a **junction box**. Normally, this is located within a few hundred yards/meters of the customer premises within that area and, inside the box, the individual wire pair from each subscriber premises is joined to a second pair within a larger cable containing multiple pairs.

In addition, for cable lengths greater than a mile (1.5 km), the individual pairs within these cables are joined to a third set of pairs within an even larger cable. This is done in a road-side cabinet known as a **cross-connect**. Typically, the cable from a junction box to a cross-connect contains in the order of 50 pairs and that from a cross-connect to the LE/EO several hundred pairs.

Telephone basics

The various components that are present in a telephone are shown in Figure 7.2(a).

A d.c. voltage of 48 *V* is permanently applied to the subscriber line by the LTU and, when the handset is lifted, contacts in the *cradle switch unit* close which causes a current to flow from the LTU to the handset. This flow of current is detected by the LTU and, as a result, it applies a pair of low-frequency tones – collectively known as *dial-tone* – to the line. On hearing this,

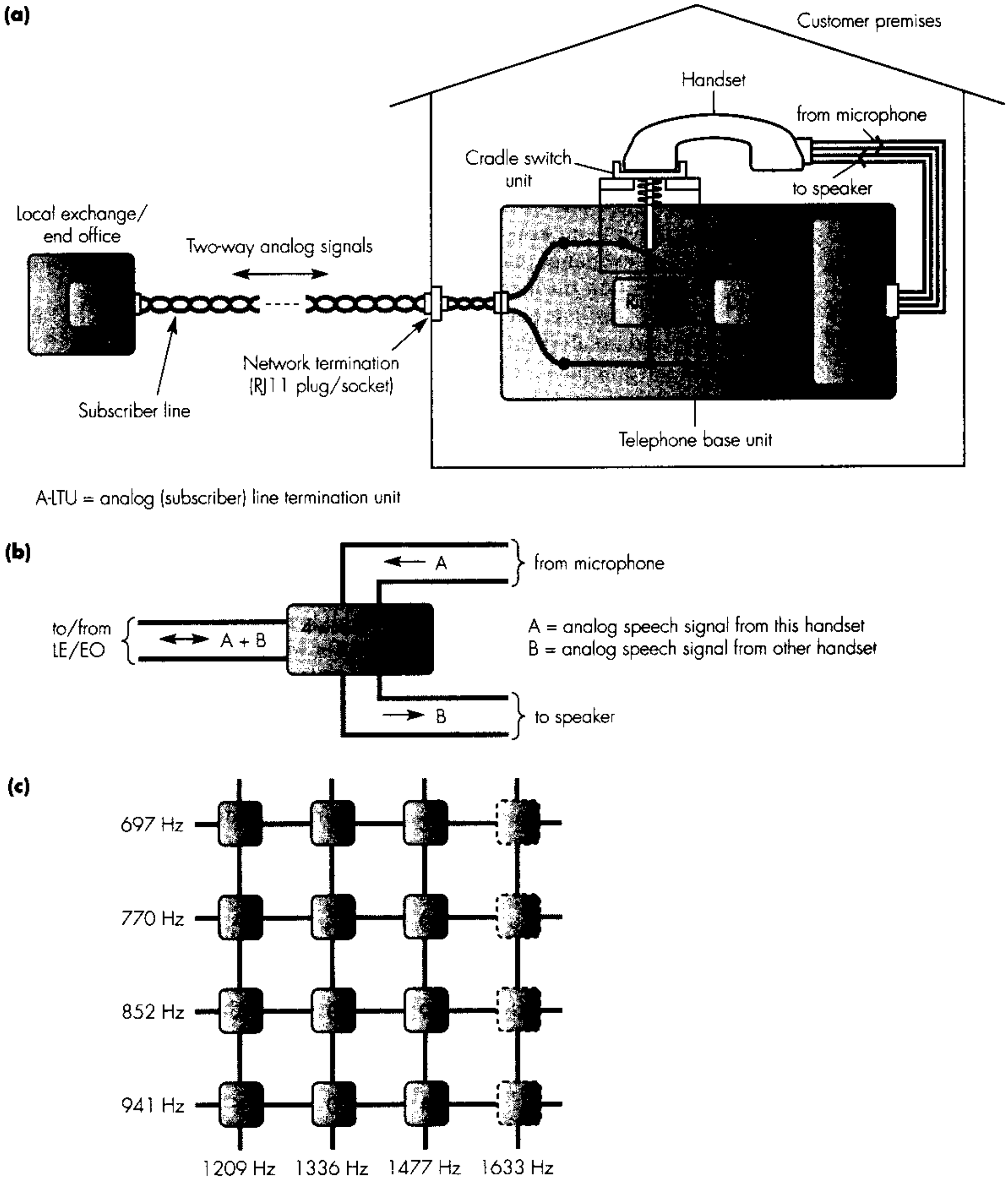


Figure 7.2 Analog subscriber line principles: (a) telephone components; (b) 4-wire-to-2-wire hybrid; (c) dual-tone multifrequency keypad.

the subscriber proceeds to enter the number of the called party using the telephone keypad which is connected to the *dialer*.

As we show in Figure 7.2(c), when each key on the keypad is pressed, a pair of (single-frequency) tones is applied to the line by the dialer; for example, pressing the digit 5 causes two tones to be applied, one of frequency 770 Hz and the other 1336 Hz. This type of dialing is known as **dual-tone multifrequency (DTMF) keying**. At the exchange end, a bank of filters – each of which detects just one of the tones – is used to determine the string of dialed digits that have been entered by the subscriber. The called number is then passed to the exchange *control processor* which proceeds to initiate the setting up of a connection through the switching network to the called party.

The *ringer circuit* is connected across the subscriber line before the *cradle switch unit* and, to alert the called subscriber of an incoming call, the LTU of the called party applies a series of short bursts of a pair of low frequency (*ringing*) *tones* to the line. The lifting of the handset by the called subscriber causes a current to flow as before and, in response, the LTU removes the ringing tone. Both subscribers are aware of this, and the conversation then starts.

As we indicated earlier, all transmission and switching within the trunk network is performed digitally. Hence as the analog speech signal from the calling subscriber is received, it is immediately sampled and converted into a (PCM) digital signal as we described in Section 2.5.1. Similarly, the received digital signal from the called subscriber is converted back into an analog form for onward transmission over the subscriber line. However, since the subscriber line comprises only a single pair of wires, this means that the same pair of wires must be used to transfer the two analog speech signals associated with the call. Hence in order for each subscriber not to hear their own voice when speaking, a unit known as a 4-wire- to 2-wire-hybrid is present in each telephone, the principle of which is shown in Figure 7.2(b).

Essentially, the output from the microphone in the handset (A) is passed to the subscriber line for onward transmission to the other party but, simultaneously, within the hybrid the same signal is subtracted from the combined signal received from the line (A + B). This means that only the signal received from the other party (B) is fed to the speaker in the handset. In practice, the hybrid is a transformer and, since a transformer will not pass a DC signal, it is the presence of the transformer that dictates the use of an analog signal. In addition, imperfections in the hybrid transformer often result in an attenuated version of the received signal – that is, from the distant subscriber – being coupled into the line from the microphone. Hence this signal is returned to the distant subscriber handset as if it was from the local subscriber but with a delay equal to twice the signal propagation delay time between the two subscribers. This is known as the **echo signal** and, providing it is received within less than 24 ms, it is not discernible to the remote subscriber. Above this value it is necessary to introduce a circuit known as an echo canceler to remove the echo signal.

Finally, the connection is closed down when either subscriber replaces the handset, the loss of current flow to the handset being detected by the related LTU which, in turn, initiates the clearing of the network connection.

Remote concentrator units

The distance between the subscriber premises and the LE/EO is limited by the attenuation that occurs in the subscriber line. Hence in order to provide a connection to subscribers that are beyond the maximum allowable distance, a device known as a **remote concentrator unit (RCU)** is used. The general layout of the access network is then as shown in Figure 7.3.

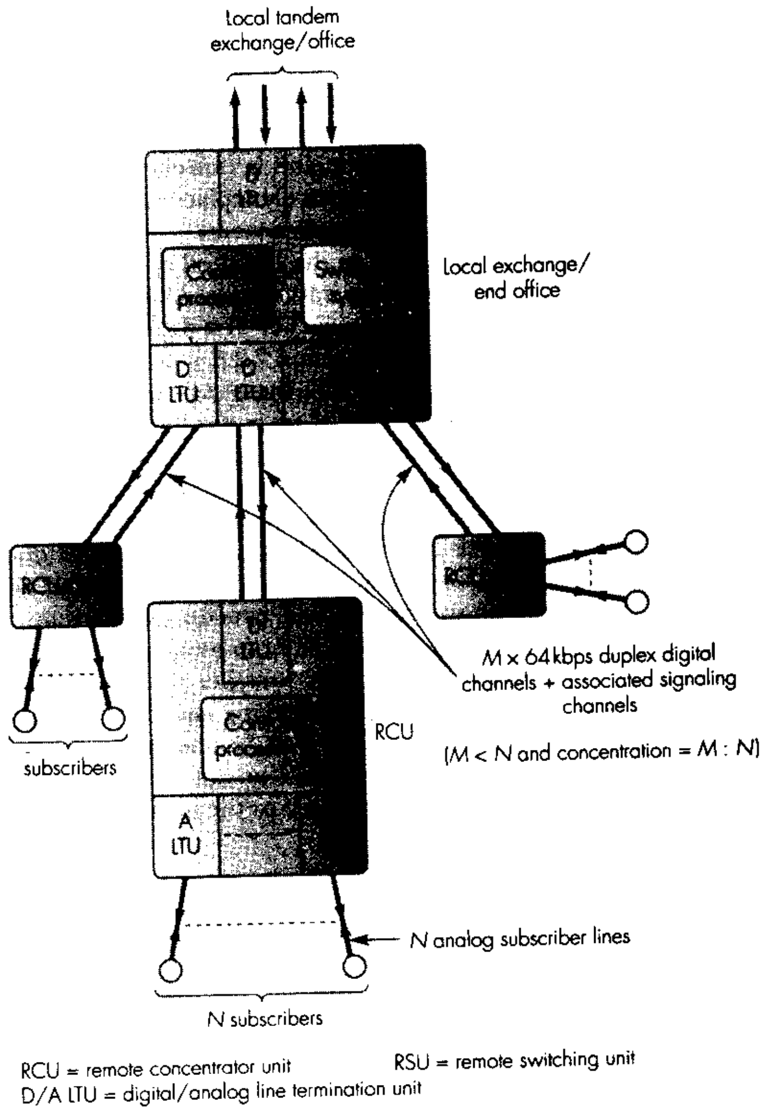


Figure 7.3 Access network structure with remote concentrator/ switching units.

Each subscriber line connected to an RCU is terminated by an LTU which performs a similar set of functions to those carried out by an A-LTU within an LE/EO. However, although the subscriber line used to connect each subscriber to the RCU operates using analog transmission, the circuit that connects the RCU to the LE/EO operates in a digital mode. As we indicated earlier, the two speech signals associated with a call are digitized to produce a corresponding pair of 64 kbps (PCM – pulse code modulation) digital signals. Hence, in theory, the digital circuits that are used to link the RCU to the LE/EO must each operate at a bit rate that supports $N \times 64$ kbps where N is the number of subscriber lines connected to the RCU. Normally, however, the number of calls that take place concurrently, M, is much less than N. Hence in order to reduce the bit rate of the digital circuit, the bit rate is made equal to $M \times 64$ kbps rather than $N \times 64$ kbps. This is the origin of the term **concentration** and is expressed as $N{:}M$, a typical figure being 8:1. Also, because RCUs effectively replace multiple (twister-pair) subscriber lines, they are known as **pair-gain systems**.

In addition to the digitized speech signals associated with each (active) call, the signaling information (dialed digits) associated with a call must also be passed to the LE/EO in a digital form. Normally, therefore, as we shall expand upon in Section 7.2.3, a portion of the bandwidth of the digital line is used to exchange the signaling messages associated with all of the currently active calls. As we show in the figure, this bandwidth is used to produce a channel known as the **signaling channel**.

To perform the various signaling functions associated with each call – off-hook detection, dial-digit collection, ringing, and so on – an RCU has a separate control processor within it that communicates with the control processor within the LE/EO to set up and release connections. Hence by adding some additional processing functions, it is also possible to allow the processor within an RCU to set up calls between any two subscribers that are connected directly to it rather than through the LE/EO. The RCU is then known as a **remote switching unit** (**RSU**) and, as we can deduce from Figure 7.3, the effect of using RCUs and RSUs is that each LE/EO can then operate with all-digital transmission and switching similar in principle to the various tandem exchanges used in the trunk network.

7.2.2 PSTN modems

As we explained in Section 6.1, in order to transmit a digital signal over an analog subscriber line, modulated transmission must be used; that is, the electrical signal that represents the binary bitstream output by the source equipment must first be converted into an analog signal that is compatible with a (telephony) speech signal. As we saw in Section 2.5.1, the range of signal frequencies that a public circuit-switched network passes is from 200 Hz through to 3400 Hz. This means that an analog subscriber line will not pass the low-frequency signals that could occur if, for example, the bitstream to be transmitted is made up of a very long string of binary 1s or 0s. For this reason, it is not possible simply to apply two voltage levels to the telephone line, since zero output will be obtained for both levels if the binary stream is all 1s or all 0s. Instead, we must convert the binary data into a form compatible with a speech signal at the sending end of the line and reconvert this signal back into its binary form at the receiver. The circuit that performs the first operation is known as a **modulator**, and the circuit performing the reverse function a **demodulator**. Since the two communicating devices normally both send and receive data, the combined device is known as a **modem**.

Using modems, data can be transmitted through the network either by setting up a switched path through the network as with a normal telephone call, or by leasing a **dedicated** (or **leased**) **line** from the network operator. Since leased lines bypass the normal switching equipment (exchange) in the network and are set up on a permanent or long-term basis, they are economically justifiable only for applications having a high utilization factor. An added advantage of a leased line is that its operating characteristics can be more accurately quantified than for a short-term switched circuit, making it feasible to operate at higher bit rates. Figure 7.4 shows the two alternative operating modes.

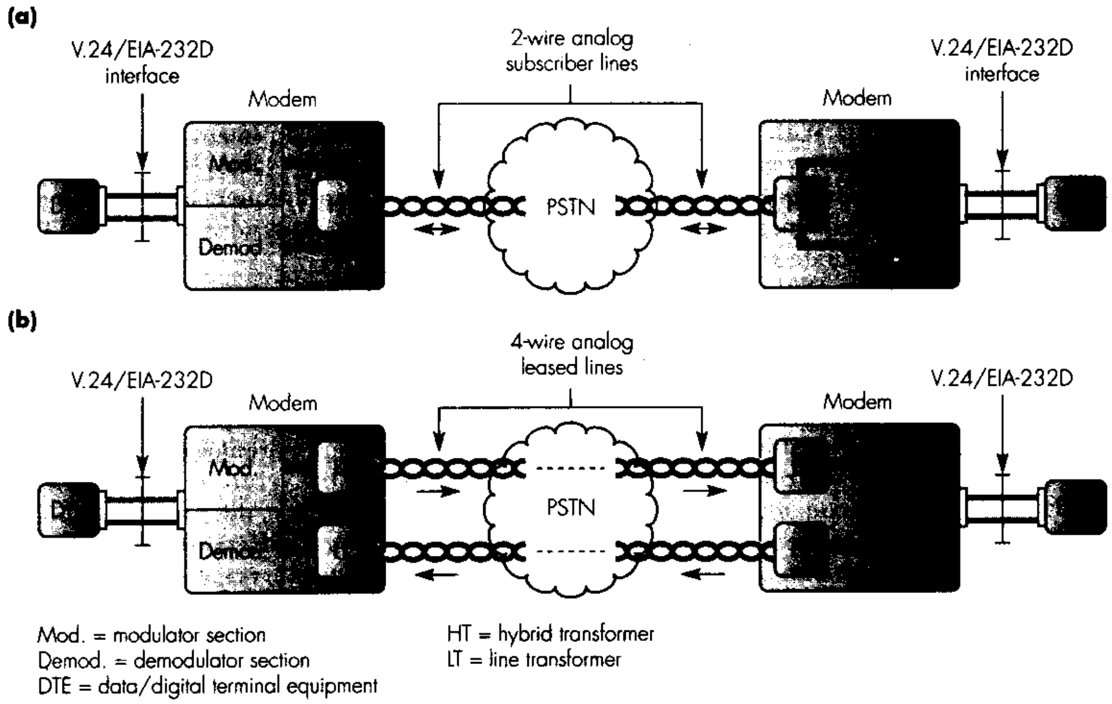


Figure 7.4 Modem operating alternatives: (a) 2-wire switched connections; (b) 4-wire leased circuits.

As we can see, in the case of a switched connection, the two analog signals that carry the transmitted and received bitstreams must share the use of the single twisted-pair subscriber line. Hence, as in a telephone, a hybrid transformer is used. In the case of a leased circuit, however, normally a 4-wire – two pairs – line is used. We shall explain the principle of operation of the modem and the terminal interface to the modem separately.

Modem principles

Three basic types of modulation are used: amplitude, frequency, and phase. Since binary data is to be transmitted, in the simplest modems just two signal levels are used. The signal then switches (shifts) between these two levels as the binary signal changes (keys) between a binary 1 and 0. The three basic modulation types are known, therefore, as **amplitude shift keying (ASK)**, **frequency shift keying (FSK)** and **phase shift keying (PSK)** respectively. The essential components that make up the modulator and demodulator sections of a modem are shown in Figure 7.5(a) and example waveforms relating to the three modulation types in Figure 7.5(b).

With ASK, the amplitude of a single-frequency audio tone is keyed between two levels at a rate determined by the bit rate of the transmitted binary signal. The single-frequency audio tone is known as the **carrier signal** (since it effectively carries the binary signal as it passes through the network) and its frequency is chosen to be within the band of frequencies that are allowed over the access circuit. The amount of bandwidth required to transmit the binary signal is then determined by its bit rate: the higher the bit rate, the larger the required bandwidth.

With FSK, the amplitude of the carrier signal remains fixed and its frequency is keyed between two different frequency levels by the transmitted binary signal. The difference between these two frequencies is known as the frequency shift and the amount of bandwidth required is determined by the bit rate and the frequency shift.

With PSK, the amplitude and frequency of the carrier remains fixed and transitions in the binary signal being transmitted cause the phase of the carrier to change. As we can see in the figure, two types of PSK are used. The first uses two fixed carrier signals with a 180° phase difference between them to represent a binary 0 and 1. Since one signal is simply the inverse of the other, it is known as **phase-coherent PSK**. The disadvantage of this scheme is that a reference carrier signal is required at the receiver against which the phase of the received signal is compared. In practice, this requires more complex demodulation circuitry than the alternative **differential PSK**. With this scheme, phase shifts occur at each bit transition irrespective of whether a string of binary 1 or 0 signals is being transmitted; a phase shift of 90° relative to the current signal indicates a binary 0 is the next bit while a phase shift of 270° indicates a binary 1. As a result, the demodulation circuitry need determine only the magnitude of each phase shift rather than its absolute value. In practice, PSK is the most efficient modulation scheme in terms of the amount of bandwidth it requires and hence is the one used in modems that provide a bit rate in excess of 4.8 kbps.

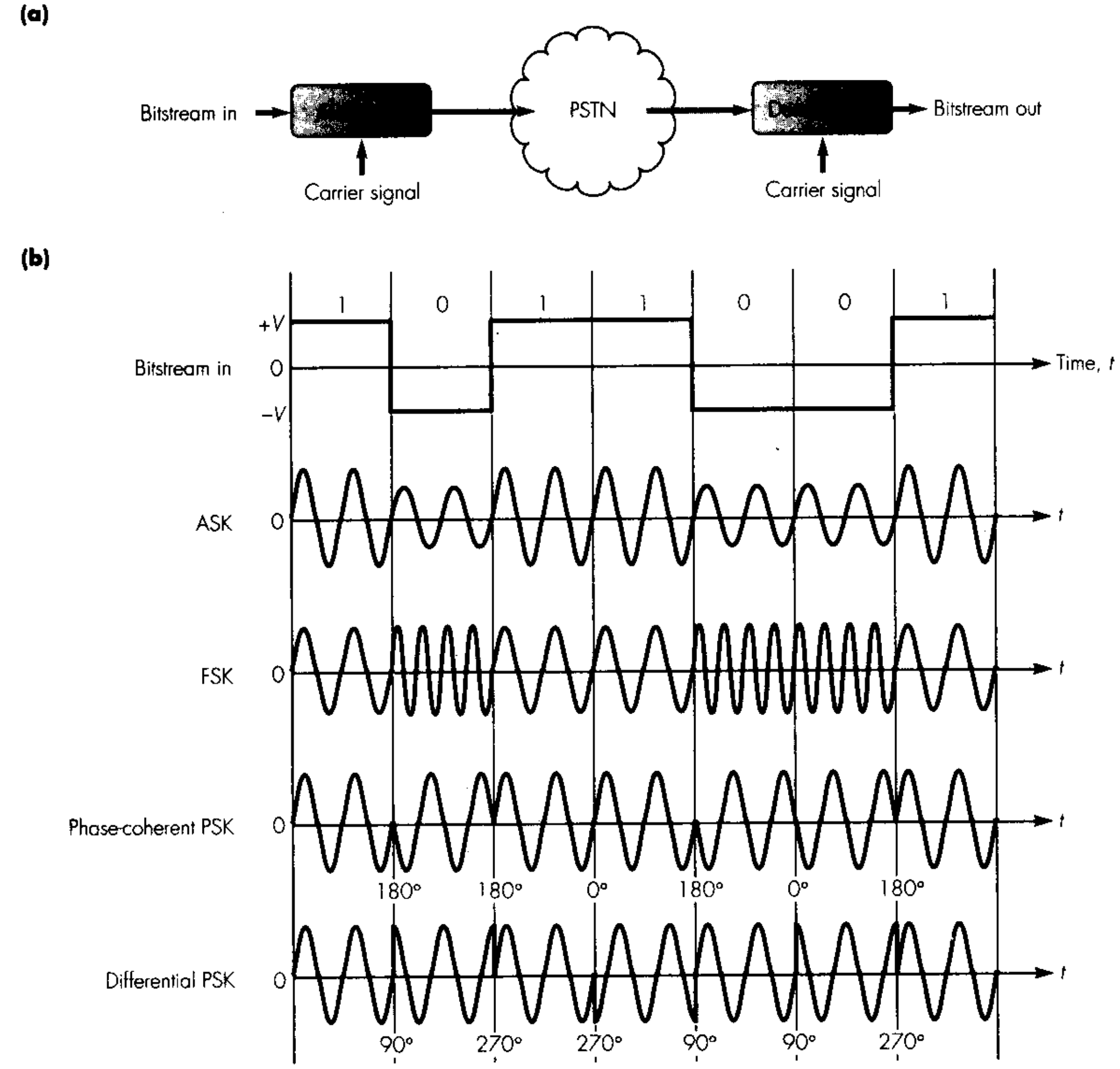


Figure 7.5 Modem principles: (a) modulator/demodulator schematic; (b) waveforms of basic modulation methods.

Multilevel modulation

As we can deduce from Figure 7.5, with the basic modulation methods just two different signal changes are used – in either amplitude, frequency, or phase – to represent the binary bitstream being transmitted. This means, therefore, that the maximum rate of change of the transmitted signal – that is, the baud rate – is equal to the bit rate of the input bitstream. Since the bandwidth of the access circuit is fixed, however, in order to obtain higher bit rates multiple signal levels are used. Hence instead of the transmitted signal changing at the same rate as the input bitstream, it changes at a lower rate.

For example, with PSK, if four phase changes are used – 0°, 90°, 180°, 270° – this enables each phase change to represent a pair of bits from the input bitstream, as we show in Figure 7.6(a). Because this scheme uses four phases, it is known as **quadrature PSK (QPSK)** or 4-PSK. Higher bit rates are achieved using larger numbers of phase changes. In practice, however, there is a limit to how many different phases can be used, as the reducing phase differences make it progressively more prone to the noise and phase impairments introduced during transmission. In order to minimize the number of phase changes required, two separate carriers are used, each of which is separately modulated.

As we show in Figure 7.6(b), the two carriers have the same frequency but there is a 90° phase difference between them. They are known, therefore, as the **in-phase carrier** (I) and the **quadrature carrier** (Q). The two modulated carriers are transmitted concurrently and, as we show in the figure, with just a single phase change per carrier – 0° and 180° – there are four combinations of the two modulated signals which means that each combination can represent 2 bits from the input bitstream. This type of modulation is known as **quadrature amplitude modulation** (QAM) and, because there are four combinations, **4-QAM**.

In order to increase the bit rate further, a combination of ASK and PSK is used. This means that as well as the I and Q carriers changing in phase, their amplitude also changes. The complete phase diagram showing all the possible combinations of amplitude and phase is known as the **constellation diagram** and an example is shown in Figure 7.6(c). As we can see, this uses 16 combinations of two amplitude and two phase changes per carrier and is known therefore as **16-QAM**.

As we can see from the above, there are many combinations of the different modulation schemes. Hence for each application, it is essential that both modems utilize the same bit rate and modulation method. For this reason the ITU-T has defined a set of international standards for modems. These are known as the **V-series** set of standards and a selection of these is given in Figure 7.7. In this way, a person who buys a modem that adheres to, say, the V.29 standard, can readily use it to communicate with a V.29-compatible modem from a possibly different manufacturer.

As we can see, some of the standards relate to modems to be used with leased circuits and others with switched circuits/connections. Also most of the modems can operate at more than one rate. Hence once a connection has been set up and prior to transmitting any user information, the two communicating modems go through a **training phase** to determine the highest of the available bit rates that can be supported by the connection. This is done by one modem sending a standard bitstream and the other measuring the

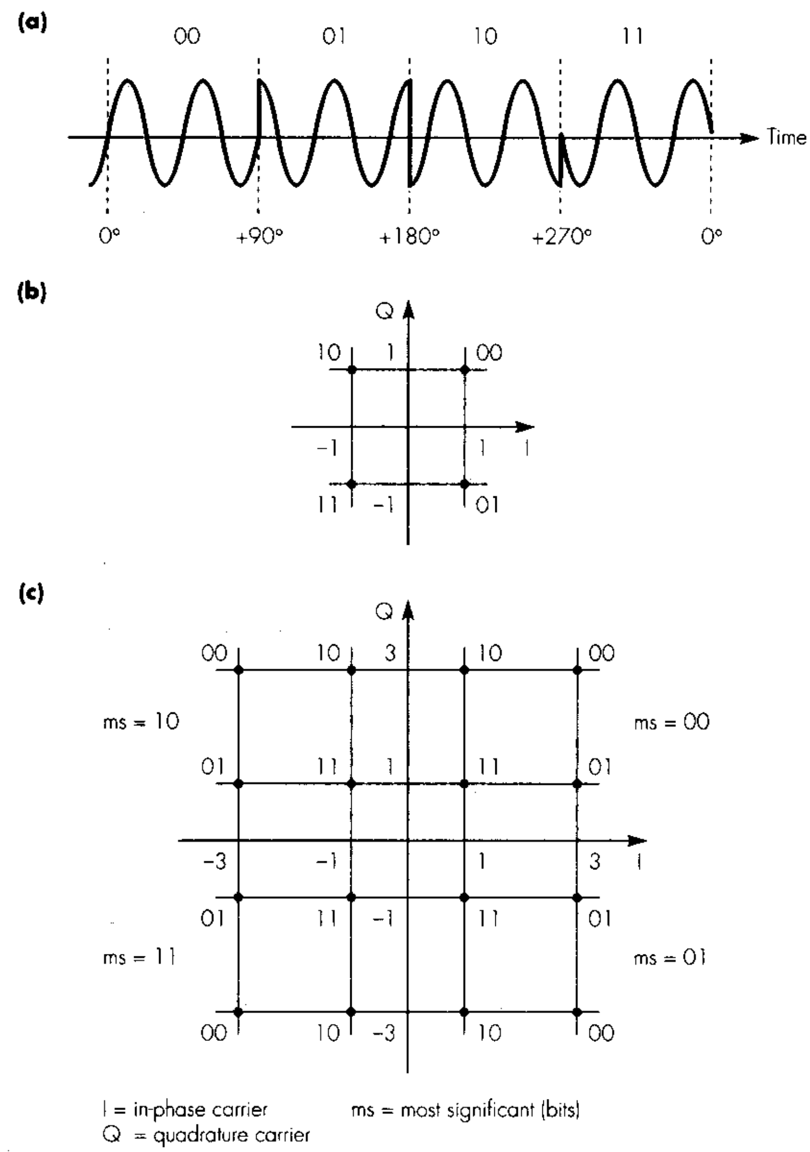


Figure 7.6 Multilevel modulation: (a) 4-PSK using a single carrier; (b) 4-QAM and (c) 16-QAM using two carriers, one at 90° (Q) out of phase with the other (I).

BER of the received stream. Normally, the lowest available bit rate is selected first and the rate is then increased progressively until the measured BER reaches a defined threshold. Typically this is in the region of 10^{-5} and 10^{-6} and, once this has been reached, both modems agree to operate at this rate.

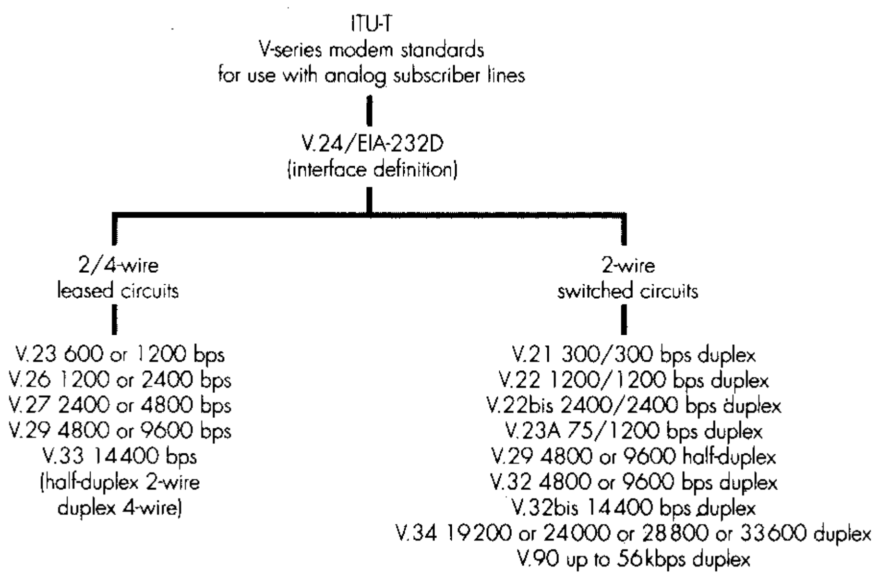


Figure 7.7 A selection of ITU-T V-series modem standards.

The choice of operating bit rate – and hence baud rate – is transparent to the user and the only observable effect of a lower bit rate is a slower response time in an interactive application, for example, or an inferior speech/video quality in an interpersonal application. Also, it should be remembered that the analog signal output by the modulator is converted into an equivalent digital signal for transmission and switching within the trunk network in just the same way that an analog speech signal is converted. Indeed, whether the source is speech or data is transparent to the network.

V.24/EIA-232D interface standard

As we showed earlier in Figure 7.4, a standard interface is used for connecting the serial part of a data terminal equipment (DTE) – a computer for example – to a PSTN modem. This is defined in ITU-T Recommendation **V.24** which is the same as the EIA standard **EIA-232D**, the latter being the latest version of the earlier RS-232A, B, and C standards. In the standards documents the modem is referred to as the **data circuit-terminating equipment (DCE)** and a diagram indicating the position of the interface in relation to two communicating DTEs/DCEs is shown in Figure 7.8(a).

The connector used between a DTE and a modem is a 25-pin connector of the type shown in Figure 7.8(b). It is defined in standard ISO 2110 and is known as a **DB25 connector**. Also shown is the total set of signals associated with the interface together with their names and pin assignments. In most cases, however, only a subset of the signals are required.

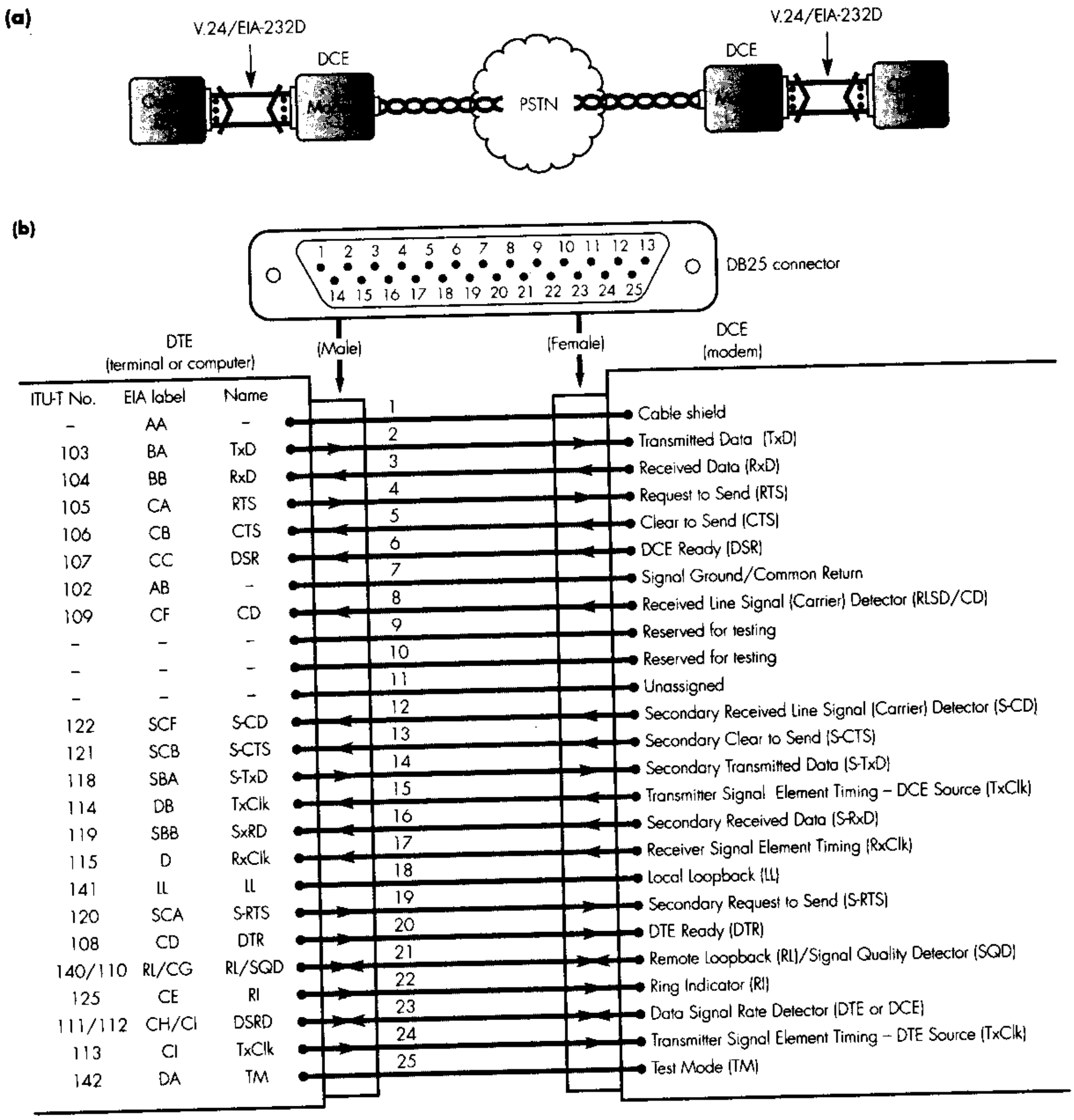


Figure 7.8 V.24/EIA-232D interface standards: (a) interface function; (b) connector, pin, and signal definitions.

The transmit data (TxD) and receive data (RxD) lines are used by the DTE to transmit and receive data respectively. The other lines collectively perform the timing and control functions associated with the setting-up and clearing of a switched connection through the PSTN and with performing selected test operations. The second (secondary) set of lines allows two data transfers to take place simultaneously over the one interface.

The timing control signals are concerned with the transmission (TxClk) and reception (RxClk) of the data on the corresponding data line. As we explained in Sections 6.4.1 and 6.5.1, data is transmitted using either an asynchronous or a synchronous transmission mode. In the asynchronous mode, the transmit and receive clocks are both generated by an independent clock source and fed directly to the corresponding pins of the DTE. In this mode, only the transmit and receive data lines are connected to the modem. In the synchronous mode, however, data is transmitted and received in synchronism with the corresponding clock signal and these are normally generated by the modem. The latter is then known as a **synchronous modem** and, when the signaling (baud) rate is less than the data bit rate – that is, multiple signal levels are being used – the transmit and receive clocks generated by the modem operate at the appropriate fraction of the line signaling rate.

We can best see the function and sequence of the various call-control lines by considering the setting-up and clearing of a call. Figure 7.9 shows how a connection (call) is first set up, some data is exchanged between the two DTEs in a half-duplex (two-way alternate) mode and the call is then cleared. We assume that the calling DTE is a user at a personal computer and its modem has automatic dialing facilities. Typically the called DTE is a server computer and its modem has automatic answering facilities. Such facilities are defined in **Recommendation V.25**. When a DTE is ready to make or receive data transfer requests, it sets the data terminal ready (DTR) line on and the local modem responds by setting the DCE ready (DSR) line on.

A connection is established by the calling DTE sending the telephone number of the modem (line) associated with the called DTE. On receipt of the ringing tone from its local switching office/telephone exchange, the called modem sets the ring indicator (RI) line to on and the called DTE responds by setting the request-to-send (RTS) line on. In response, the called modem sends a carrier signal – the data tone for a binary 1 – to the calling modem to indicate that the call has been accepted by the called DTE and, after a short delay to allow the calling modem to prepare to receive data, the called modem sets the clear-to-send (CTS) line on to inform the called DTE that it can start sending data. On detecting the carrier signal, the calling modem sets the carrier detect (CD) line on. The connection is now established and the data transfer phase can begin.

Typically, the called DTE (computer) starts by sending a short invitation-to-send message over the setup connection. When this has been sent, it prepares to receive the response from the calling DTE by setting the RTS line

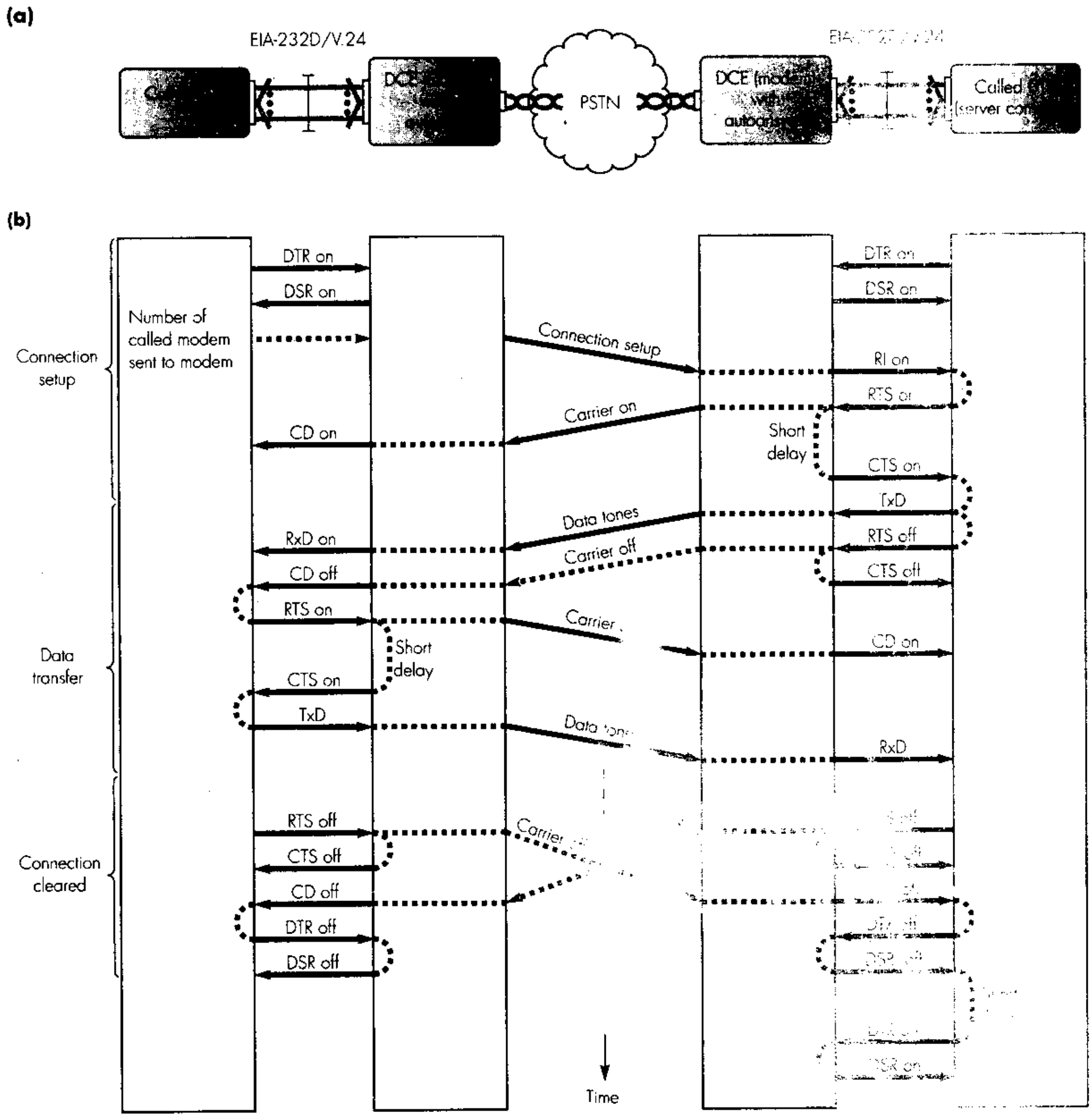


Figure 7.9 V.24/EIA-232D connection setup, two-way alternate data transfer and connection clearing sequences.

off and, on detecting this, the called modem stops sending the carrier signal and sets the CTS line off. At the calling side, the removal of the carrier signal is detected by the calling modem and, in response, it sets the CD line off. In order to send its response message, the calling DTE (PC) sets the RTS line on and, on receipt of the CTS signal from the modem, starts to send the message. This procedure then repeats as messages are exchanged between the two DTEs. Finally, after the complete transaction has taken place, the call is cleared. This is accomplished by both DTEs setting their RTS lines off which, in turn, causes the two modems to switch their carriers off. This is detected by both modems and they set their CD lines off. Both DTEs then set their DTR lines off and their modems respond by setting the DSR lines off thereby clearing the call. Typically, the called DTE (the server computer) then prepares to receive a new call by resetting its DTR line on after a short delay.

We have described the use of a half-duplex switched connection to illustrate the meaning and use of some of the control lines available with the standard. In practice, however, the time taken to change from the receive to the transmit mode in the half-duplex mode – known as the **turnaround time** – is not insignificant. It is preferable to operate in the duplex mode whenever possible, even when half-duplex working is required. In the duplex mode, both RTS lines are permanently left on and both modems maintain the CTS line on and a carrier signal to the remote modem.

When two DTEs are communicating and a fault develops, it is often difficult to ascertain the cause of the fault – the local modem, the remote modem, the communications line, or the remote DTE. To help identify the cause of faults, the interface contains three control lines: the local and remote loopback (LL and RL) and the test mode (TM). Their function is shown in Figure 7.10(a): in (i) a local loopback test is used and in (ii) a remote loopback.

The DTE (modem) always sets its DSR line on when it is ready to transmit or receive data. To perform a test on its local modem, the DTE sets the LL line on and, in response, the modem internally connects the output from the modulator circuit back to the input of the demodulator circuit. It then sets the TM line on and, when the DTE detects this, it transmits a known test (data) pattern on its TxD line and simultaneously reads the data from its RxD line. If the received data is the same as the test data, then the DTE assumes the local modem is working satisfactorily. If it is not – or no signal is present at all – then the local modem is assumed faulty.

If the local modem is deemed to be working correctly, then the DTE proceeds to test the remote modem by this time setting the RL control line on. On detecting the RL line going on, the local modem sends a predefined command to the remote modem which, in turn, performs a remote loopback as shown. The remote modem then sets its TM line on to inform the remote DTE it is involved in a test – and hence cannot transmit data – and returns an acknowledgment command back to the modem originating the test. The modem, on receipt of this, sets its TM line on and, on detecting this, the local DTE starts to transmit the test data pattern. Again, if this data is received

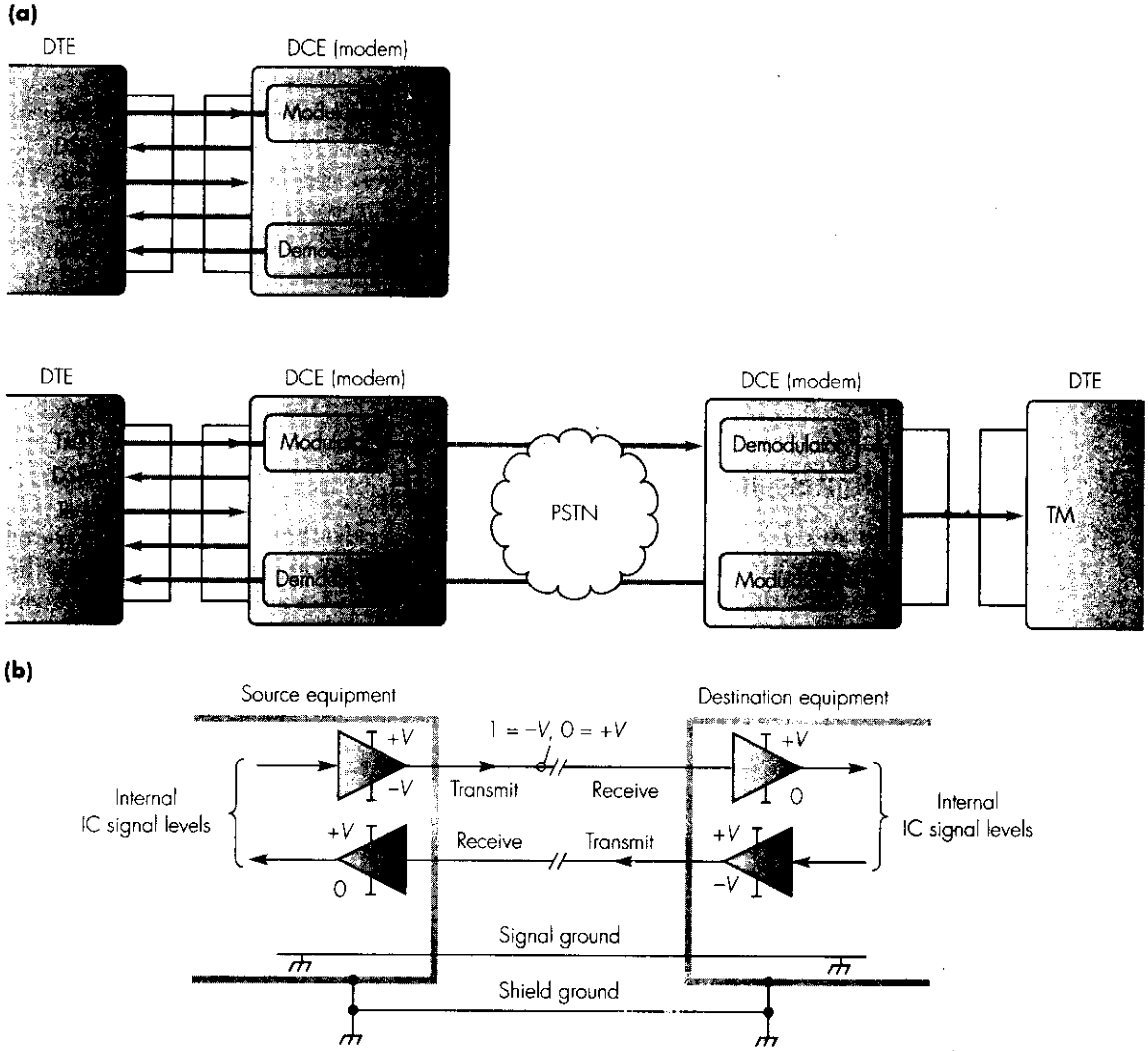


Figure 7.10 V.24/EIA-232D interface: (a) local and remote loopback tests; (b) V.28/RS.232A signal levels.

correctly, then both modems are assumed to be working correctly and the fault lies with the remote DTE. Alternatively, if the received data is badly corrupted then the remote modem is assumed to be faulty or, if no signal is received at all, then the PSTN line is assumed faulty.

The V.24/EIA-232D interface uses either a flat-ribbon or a multiple-wire cable which includes a single ground reference wire. However, because of the short distances – less than a few centimeters – between neighboring integrated circuits within a DTE computer, the signal levels used to represent binary data are very low power and, as a result, cannot be used directly for transferring signals outside of the computer. Hence, as we show in Figure 7.10(b), associated with each signal line of the V.24/EI-232D interface is a matching **line driver** and **line receiver** circuit.

The electrical signal levels are defined in standards **V.28/RS.232A.** The signals used on the lines are symmetric with respect to the ground reference signal and are at least $3V$: $+3V$ for a binary 0 and $-3V$ for a binary 1. In practice, the actual voltage levels used are determined by the supply voltages applied to the interface circuits, $\pm 12V$ or even $\pm 15V$ not being uncommon. The transmit line driver circuits convert the low-level signal voltages used within the equipment to the higher voltage levels used on the connecting lines. Similarly, the line receiver circuits perform the reverse function.

7.2.3 Digital subscriber lines

In an ISDN, all the signals associated with a call – both the two speech signals and the associated signaling messages – are transmitted over the subscriber line in a digital form. Also, as we saw in Figure 1.4 and in the accompanying text, an ISDN has a number of different subscriber line interfaces:

- a basic rate interface (BRI) that provides two independent 64 kbps duplex channels;
- a primary rate interface (PRI) that provides either 23 or 30 64 kbps duplex channels;
- a primary rate interface that provides a single duplex channel of $p \times 64$ kbps where p can be 1–23 or 1–30.

We shall discuss each interface separately.

Basic rate interface

As we explained in Section 1.3.4, the basic rate interface – and the associated **network termination unit** (**NTU**) – allows for two calls of 64 kbps duplex to be in progress concurrently. Hence, since the two calls can be set up independently – that is, the second call can be set up while the first is in progress – an additional duplex channel of 16 kbps is used for the exchange of the signaling messages relating to the two calls. This arrangement is known as **out-of-band signaling**. Each 64 kbps user channel is known as a **bearer** or **B-channel** and the 16 kbps signaling channel the **D-channel**. Hence the combined bit rate associated with this interface is 2B + D or 144 kbps duplex. Two examples of an NTU associated with an ISDN BRI are shown in Figure 7.11.

As we show in Figure 7.11(a), in the first example the NTU has two digital ports and two analog ports associated with it. The two analog ports are provided to enable the subscriber to utilize existing analog equipment such

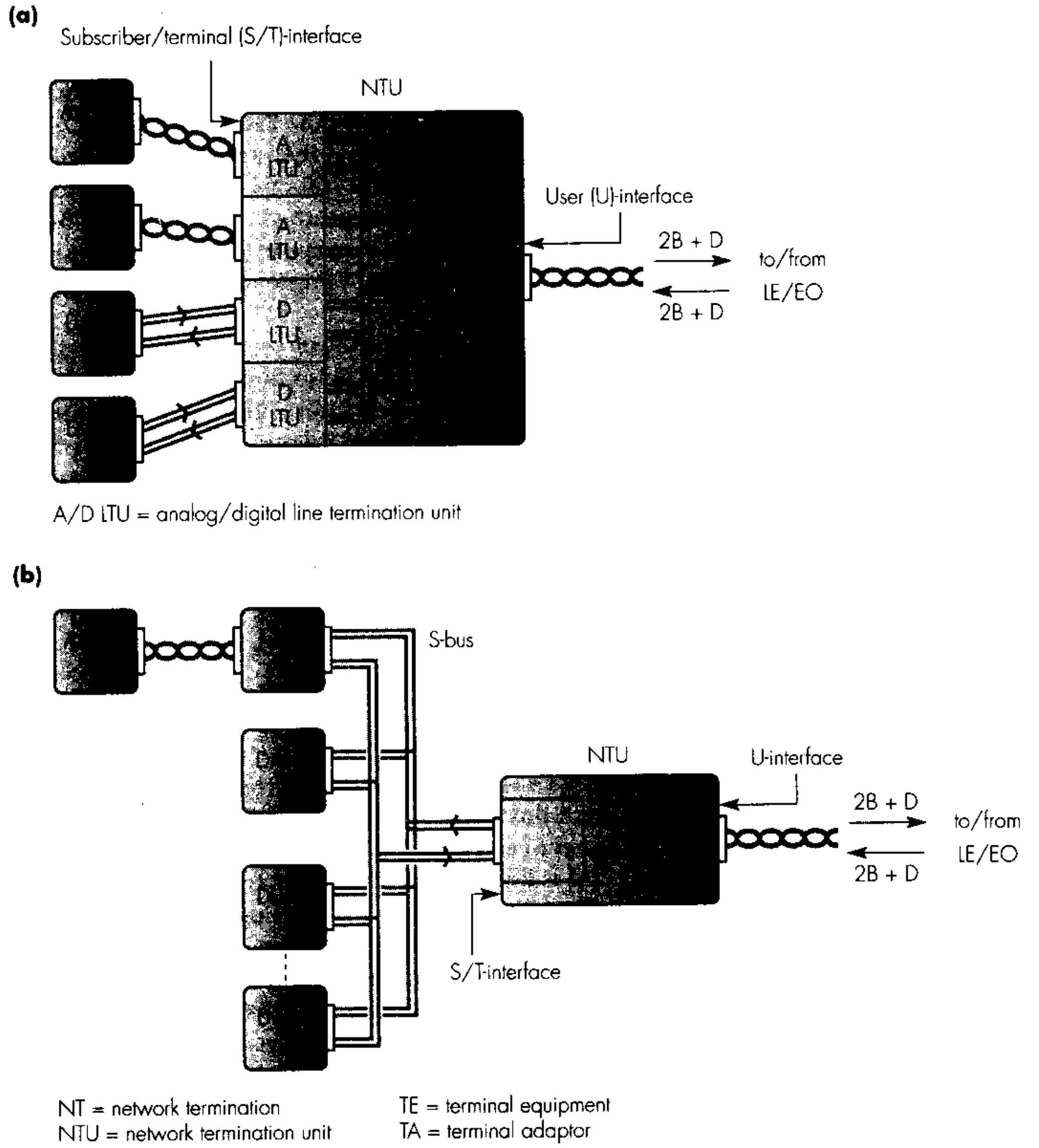


Figure 7.11 ISDN network termination alternatives: (a) 4-port NTU; (b) S-bus NTU.

as an analog phone, a fax machine, or a PC with modems. The two digital ports are provided to enable newer digital equipment with an ISDN interface to be used. In the case of the two analog ports, the conversion of the analog signals into and from their digital form is performed within the NTU. This means that the subscriber can use any mix of the four ports with any two active at one time.

As we show in Figure 7.11(b), a second mode of working is also used which allows from one up to eight devices to (time) share the use of the two B-channels. This mode of working is defined in **ITU-T Recommendation I.430**. In this mode the NTU has a single port associated with it to which is connected a duplex bus known as the subscriber or **S-bus**. The various terminal equipments (TEs) then gain access to the bus – and hence B-channels – using a defined interface and associated protocol. In the case of an existing analog TE, a device known as a **terminal adaptor (TA)** must be used to convert the analog signals associated with the TE into and from the digital signals used over the S-bus.

The S-bus must support the duplex flow of two (64 kbps) B-channels and the 16 kbps D-channel together with the contention resolution logic for time-sharing the use of the D-channel. To achieve this, the bitstream in each direction is divided into a stream of 48-bit frames each of which contains 16 bits for each of the two B-channels and 4 bits for the shared D-channel multiplexed in the order 8B1, 1D, 8B2, 1D, 8B1, 1D, 8B2, 1D. The remaining 12 bits are then used for various functions including:

- the start-of-frame synchronization pattern,
- contention resolution of the shared D-channel,
- the activation and deactivation of the interface of each TE,
- DC balancing.

The duration of each 48-bit frame is 250 microseconds which yields a bit rate of 192 kbps in each direction. As we show in Figure 7.12, an 8-pin connector is used to connect each user TE to the NTU. This is defined in the **ISO 8877** standard and is known as a **RJ45**. The relatively high bit rate and physical separations associated with the S-bus mean that **differential line driver and receiver circuits** must be used. As we show in the figure, this requires a separate pair of wires for each of the transmit and receive signals. This type of signal is defined in standards **V.11/RS-422A**.

A differential transmitter produces twin signals of equal and opposite polarity for every binary 1 or 0 signal to be transmitted. As the differential receiver is sensitive only to the difference between the two signals on its two inputs, noise picked up by both wires will not affect receiver operation. Differential receivers, therefore, are said to have good **common-mode rejection** properties. A derivative of the RS-422A, the **RS-423A/V.10**, can be used to accept single-ended (**unbalanced**) voltages output by an EIA-232D interface with a differential receiver.

An important parameter of any transmission line is its **characteristic impedance** (Z_0) because a receiver absorbs all of the received signal power only if the line is terminated by a resistor equal to Z_0. If this is not the case, **signal reflections** occur which further distort the received signal. Normally, therefore, lines are terminated by a resistor equal to Z_0, with values from 50 to 200 ohms being common.

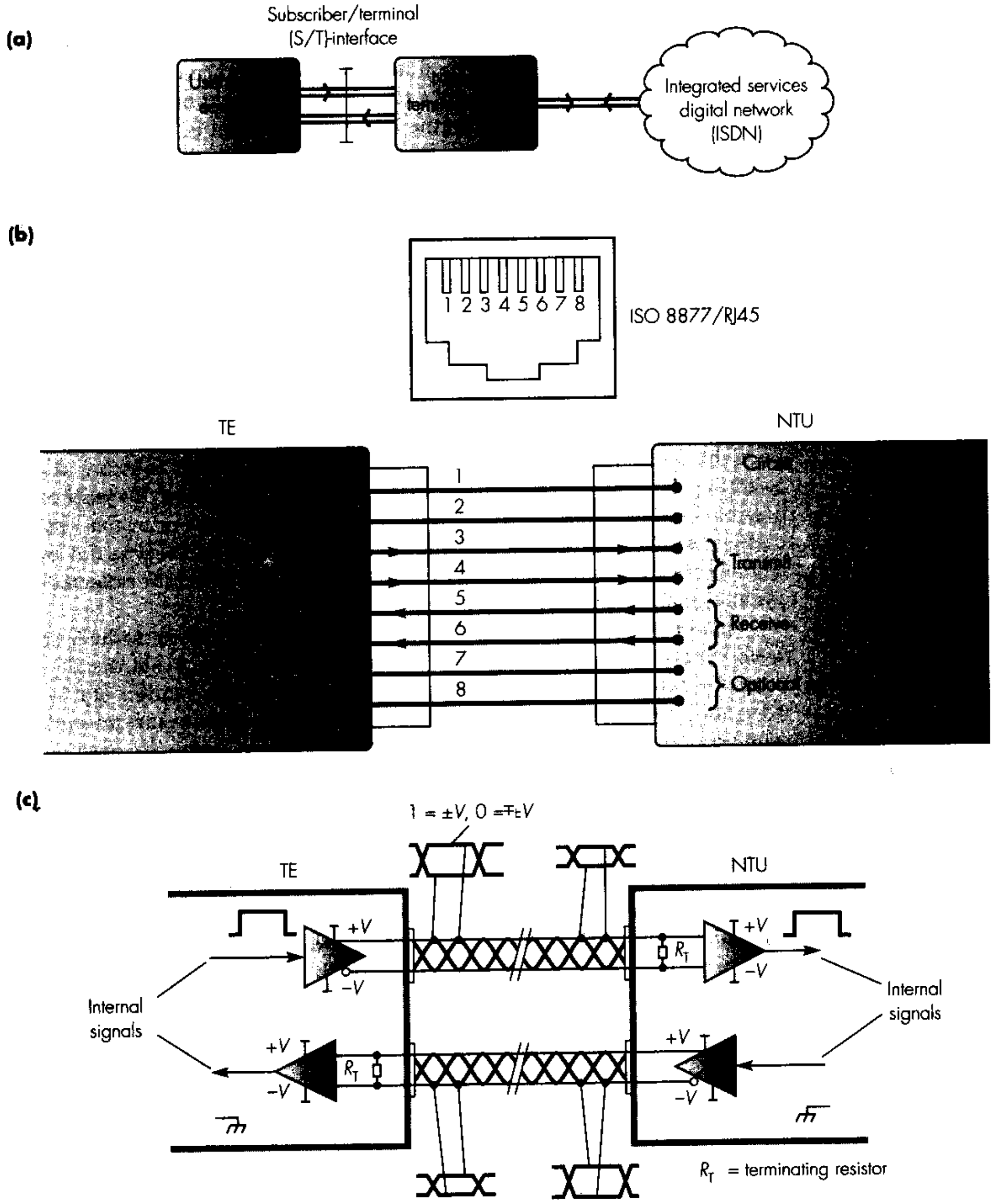


Figure 7.12 ISDN subscriber/terminal (S/T)-interface: (a) interface location; (b) socket, pin and signal definitions; (c) signal levels.

The line code used over the S-bus is known as **alternate space inversion (ASI)** the principle of which is shown in Figure 7.13(a). As we can see, this is a 3-level code: $+V$, 0, and $-V$. The 0 level is used to indicate the transmission of a binary 1 and either $+V$ or $-V$ a binary 0: for every 0 bit transmitted, the line signal changes polarity from the last 0-bit level; that is, either from $+V$ to $-V$ or vice versa. This type of line signal is called **pseudoternary**.

At the start of each 48-bit frame is the 2-bit frame synchronization pattern of +V, –V. The line signal then changes according to the transmitted bitstream with the first 0-bit encoded as –V. As we can deduce from the figure, however, the line signal will not be balanced if, overall, the number of 0 bits is odd. Hence, since each TE is connected to the bus by means of a transformer, the polarity of the various DC balancing bits present in each frame is chosen so that the mean DC level of the line is always zero.

Although the bus can have up to 4–8 TEs connected to it, only two can be active at any point in time. A scheme is required, therefore, to enable all the TEs to contend for access to the two B-channels in a controlled way. However, since the two calls must be set up by means of the D-channel (which is shared by all the TEs) the contention occurs for use of the D-channel. To resolve any possible contention, the NTU reflects the four D-bits present in the (48-bit) frame it is currently receiving (from the one or more TEs) back in the frame it is currently transmitting out in the reverse direction. The four reflected bits are known as the **echo** or **E-bits**.

When no TE is using the D-channel, the four D-bits are set to the 0 signal level. Hence prior to sending a request message to set up a call, the TE first reads the four E-bits from the frame currently being transmitted out by the NTU and only if they are all at the 0 level does it proceed to start to send the request message in the four D-channel bits in the next frame. In addition, to allow for the possibility of one or more other TEs starting to send a request message at the same time, each TE that is trying to send a message monitors the (reflected) E-bits in the frame currently being received to check that these are the same as the D-bits that it has just transmitted. If they are the same, then it continues to send the remaining bits in the message; if they are different, then it stops transmitting and tries again later. The principle of the scheme is shown in Figure 7.13(b) and, as we can see, because of the line code used, the winner of a contention will always be the TE whose message contains a 0 bit when the other(s) contains a 1 bit. Also, in the event of both B-channels being in use when a (successful) new request is made, then a busy response message will be returned to the requesting TE. If the call can be accepted, however, then an acceptance message will be returned and the signaling procedure continues.

The interface between the NTU and the subscriber line is known as the **U-interface** and this is defined in **ITU-T Recommendation G.961**. As we showed earlier in Figure 7.11, as with an analog subscriber line, a single pair of wires is used to connect the subscriber NTU to the LE/EO. This means, therefore, that both the transmitted and received digital bitstream of 144 kbps (2B + D)

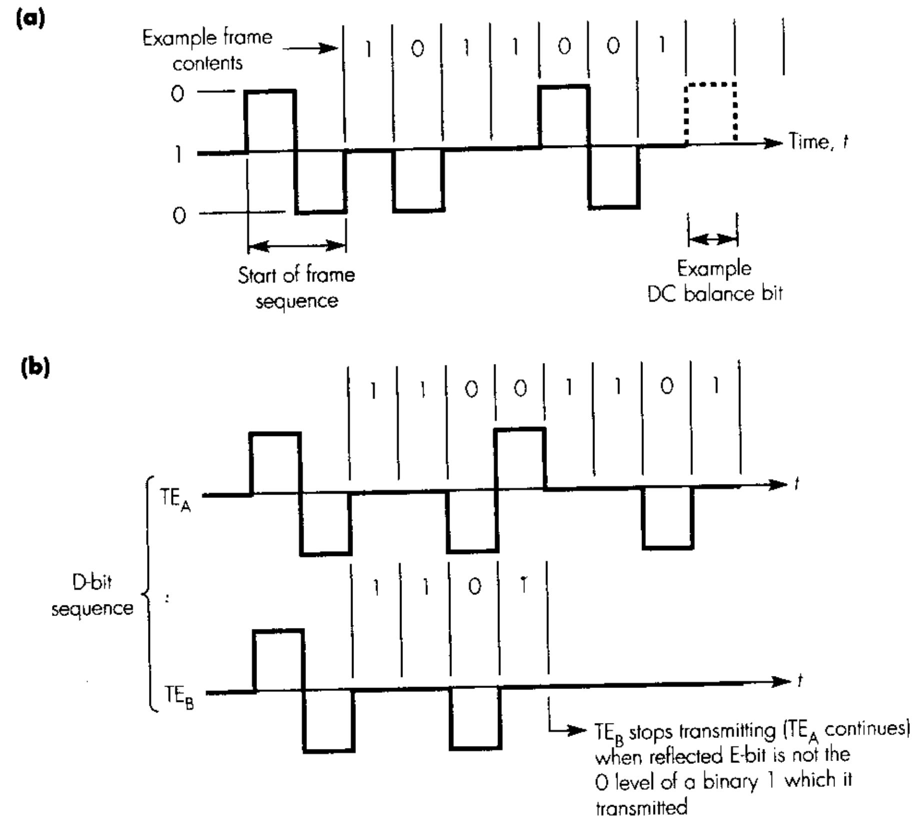


Figure 7.13 ISDN basic rate access S-bus line code principles: (a) alternate space inversion (ASI) line code; (b) example of contention resolution.

must be transmitted over the same pair of wires simultaneously. Hence in order for the receiver part of the NTU to receive only the incoming digital bitstream, an electronic version of the hybrid transformer used with an analog line is incorporated into the NTU, as we show in Figure 7.14(a).

As we can see, in addition to the hybrid, the network termination includes a circuit known as an **adaptive echo canceler**. In practice, the hybrids present at each end of a subscriber line – there is also one in the line termination unit at the LE/EO – are not perfect and, as a result, an attenuated version of the transmitted signal (A) is echoed back from the remote hybrid and hence is passed to the receiver part of the NTU together with the wanted received signal (B). Essentially, the adaptive echo canceler circuit estimates the magnitude of the attenuated version of its own transmitted signal – that is, the echo signal – and subtracts this from the combined signal output by the hybrid.

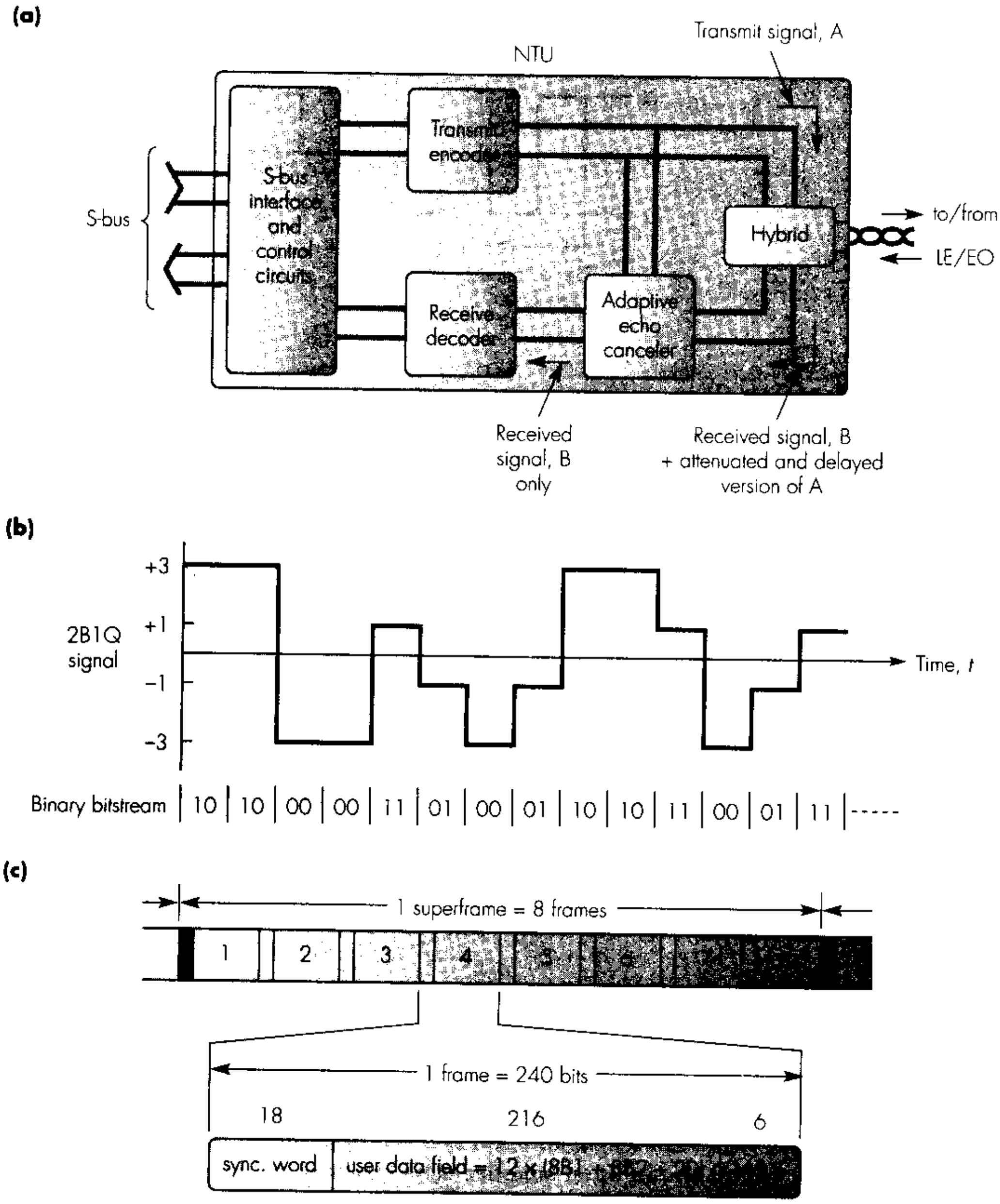


Figure 7.14 ISDN subscriber line principles: (a) NTU schematic; (b) 2B1Q line signal example; (c) frame and superframe format.

In order for the distance from the LE/EO to the subscriber NTU to be as large as possible, the line code used over the subscriber line is a 4-level code known as **two binary, one quaternary (2B1Q)**, the principles of which are shown in Figure 7.14(b). As we can deduce from the example bitstream shown in the figure, with this code the maximum rate of change of the line signal – the baud rate – is one half of the bit rate. As a result, the bandwidth required for this code is one half that required for a 2-level bipolar code.

The total 2B + D bitstream is divided into a sequence of frames of 240 bits the format of which is shown in Figure 7.14(c). As we can see, each frame consists of:

- an initial 18-bit *synchronization word* comprising the quaternary symbol sequence of +3 +3 −3 −3 +3 +3 −3 −3 ... to enable the receiver to determine the start of each frame;
- the *user data* field comprising a set of 12 groups of 18 bits, 8 bits for each of the two B-channels and 2 bits for the D-channel (multiplexed in the order 8B1, 8B2, 2D);
- a 6-bit field that relates to a separate M-channel that is used for maintenance messages and for other purposes.

A further structure known as a **superframe** which comprises eight frames is then defined by inverting the symbols in the synchronization word of the first frame. The resulting 48 M-bits are then used to carry a number of fields including a 12-bit CRC that is used to monitor the quality of the line.

Primary rate interface

In the case of a **primary rate interface (PRI)**, only a single TE can be connected to the NTU. As we explained in the introduction, however, the TE can be a PBX, for example, which, in turn, supports multiple terminals each operating at 64 kbps or a reduced number of terminals operating at a higher bit rate. To provide this flexibility, the transmitted bitstream contains a single

Example 7.1

Deduce the bit rate and baud rate of the subscriber line of an ISDN basic rate access circuit assuming the 2B1Q line code and the frame format shown in Figure 7.14(c).

Answer:

Each 240-bit frame comprises 12 × 8 = 96 bits per B-channel. Hence, since this is equivalent to a bit rate of 64kbps, the total bit rate is 64 × 240/96 = 160 kbps.

Since there are 2 bits per signal element, the signaling rate = 80 kbaud.

D-channel which is used by the TE to set up the required call(s). The number of 64 kbps channels present in the bitstream is either 23 or 30 depending on the type of interface being used. These correspond to the 1.544 Mbps interface and 2.048 Mbps interface respectively. Since each operates in a different way, we shall discuss them separately.

1.544 Mbps interface The line code used with this interface is known as **alternate mark inversion (AMI)** with **bipolar and eight zeros substitution (B8ZS)**. The principle of both coding schemes is shown in Figure 7.15(a).

As we can see, AMI is a 3-level code: $+V$, 0, and $-V$. The 0 level is used to indicate the transmission of a binary 0 and either $+V$ or $-V$ a binary 1: for every 1 bit transmitted, the line signal changes polarity from the last 1 bit level; that is, either from $+V$ to $-V$ or vice versa. Normally, as we explained in Section 6.5.1, a DPLL is used to obtain clock/bit synchronization. The disadvantage of AMI on its own, therefore, is that a long string of binary 0s will have no associated signal transitions. Consequently, the DPLL may lose bit synchronization whenever a string of 0s is present.

To overcome this limitation, the additional B8ZS coding scheme is used. As we can see, when B8ZS is used with AMI, the line code is the same as that with AMI on its own except that when a string of eight 0 bits is detected in the string these are encoded as 000VB0VB prior to transmission, where V is a violation (same polarity) transition and B a normal (opposite polarity) transition. With B8ZS present, therefore, the maximum string of 0 bits that can be present is seven, which is acceptable for the DPLL. We note also that both AMI and the combined scheme produce differential signals which allows longer cable lengths to be used with transformers at each end.

The transmitted bitstream is divided into a sequence of 193-bit frames the format of which is shown in Figure 7.15(b). As we can see, each frame comprises a single **framing** or **F-bit** followed by 24 8-bit time slots. The duration of each frame is 125 microseconds and hence each of the 8-bit time slots forms a 64 kbps channel. Also, 193 bits every 125 microseconds produces a bit rate of 1.544 Mbps. Normally, one of the time slots – and hence 64 kbps channels – is used as a signaling (D) channel and the messages carried over this relate to the setting up and closing down of the calls carried over the remaining 23(B) channels. The 23 time slots are used either singularly or in groups to provide the required bit rate.

In order for the receiver to detect the start of each frame using a single bit, a group of 24 frames – known as a **multiframe** – is defined. The F-bits from frames 4, 8, 12, 16, 20, and 24 are set to the bit sequence 0, 0, 1, 0, 1, 1 respectively and this is then known as the **frame alignment signal (FAS)**. Six of the remaining 18 F-bits are then used for a 6-bit CRC which is used to monitor the quality of the line.

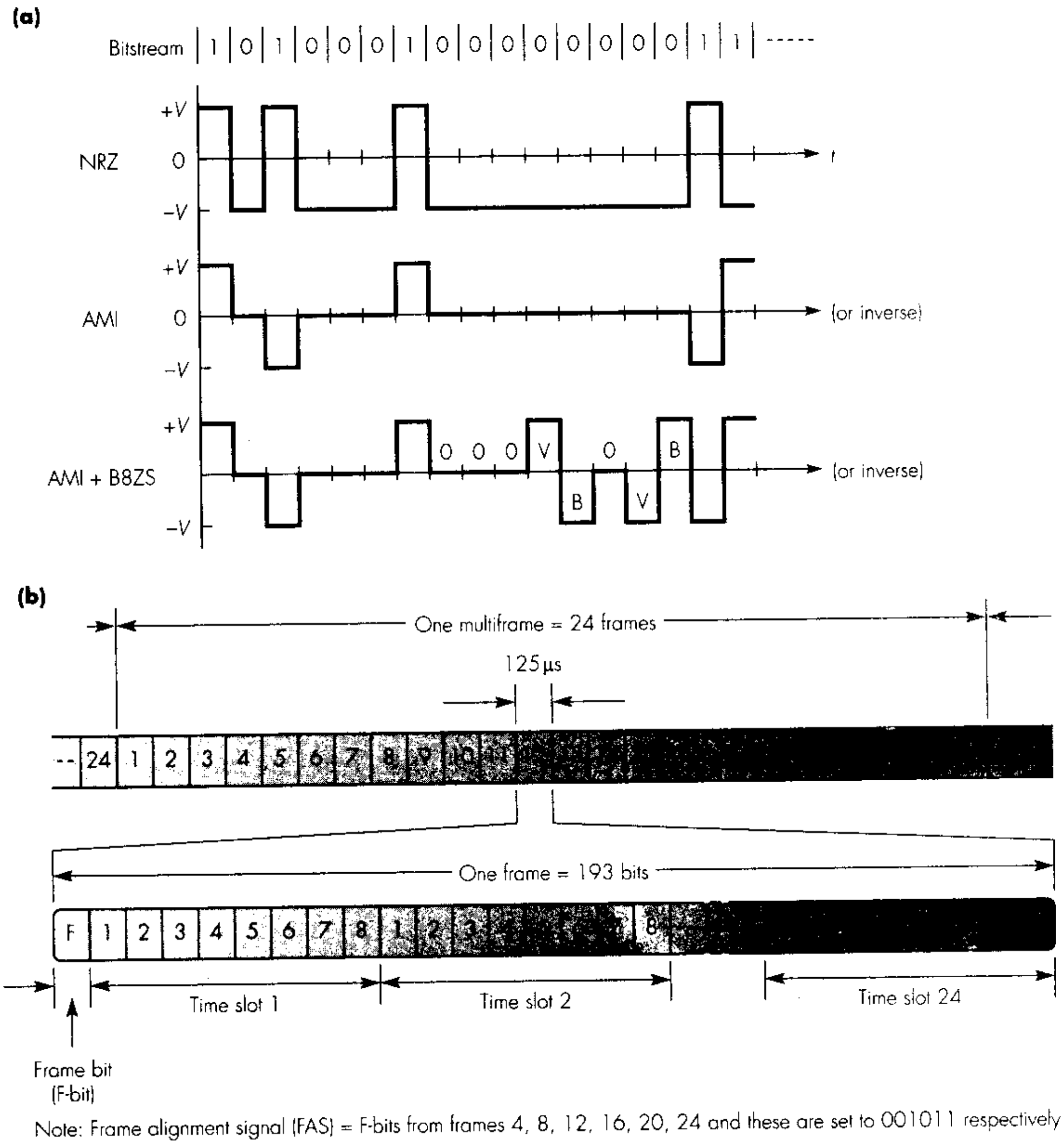


Figure 7.15 ISDN 1.544 Mbps primary rate interface principles: (a) line code; (b) frame and multiframe structure.

2.048 Mbps interface The principles of the line code and the framing structure used with the 2.048 Mbps interface are shown in parts (a) and (b) of Figure 7.16 respectively. As we can see, the line code is also AMI but the additional coding scheme used to obtain signal transitions when strings of 0 bits are being transmitted is the **high density bipolar 3 (HDB3)** scheme.

This operates by replacing any string of four 0 bits by three 0 bits followed by a bit encoding violation; that is, a transition which is of the same polarity as the previous transition. Hence, as we can see, the first string of four 0 bits is replaced by 000V. With this basic rule, however, the presence of a long string of 0 bits would lead to a mean DC level being introduced into the signal as each set of four 0 bits is encoded in the same way. To overcome this, when transmitting a bitstream that contains a long string of 0 bits, after the first four 0 bits have been encoded, each successive set is changed to B00V. As we can see, this produces a signal of alternating polarity which removes the DC level that would have been present so allowing transformers to be used.

As we can see in Figure 7.16(b), the transmitted bitstream is divided into a sequence of 256-bit frames of duration 125 microseconds. Hence each time slot produces a 64 kbps channel and the bit rate of the bitstream is 2.048 Mbps. Time slot 0 is used for frame alignment and other maintenance functions. To achieve frame alignment, the contents of time slot 0 in alternate frames is as shown in the figure. The remaining bits – shown as x in the figure – are then used to carry a 4-bit CRC for line quality monitoring and other functions. Normally, time slot 16 is used as a signaling (D) channel and the remaining 30 time slots (1–15 and 17–31) are used either singly or in groups to provide the required bit rate. Note that the establishment of a superframe – comprising 16 frames – is optional and is done, for example, to obtain an added level of line quality monitoring. To achieve this, an additional frame alignment word is present in time slot 16 of frame 0.

7.2.4 Plesiochronous digital hierarchy

As we saw earlier in Section 7.1, within the trunk network digital transmission (and switching) is used throughout. For historical reasons, there are two types of transmission system used, one based on what is known as the plesiochronous (nearly synchronous) digital hierarchy (PDH) and the other on a synchronous digital hierarchy (SDH). We shall explain the principles of the PDH in this section and those of the SDH in the next section.

As we saw in Figure 7.1, a national circuit-switched network is made up of a hierarchy of digital transmission and switching systems. And as we explained in the previous two sections, in terms of transmission systems, multiplexing starts within the access network where, typically, the 64 kbps channels derived from the 24/32 time slots are multiplexed together. At higher levels in the hierarchy, however, the transmission systems must support progressively larger numbers of channels/simultaneous calls. Hence this

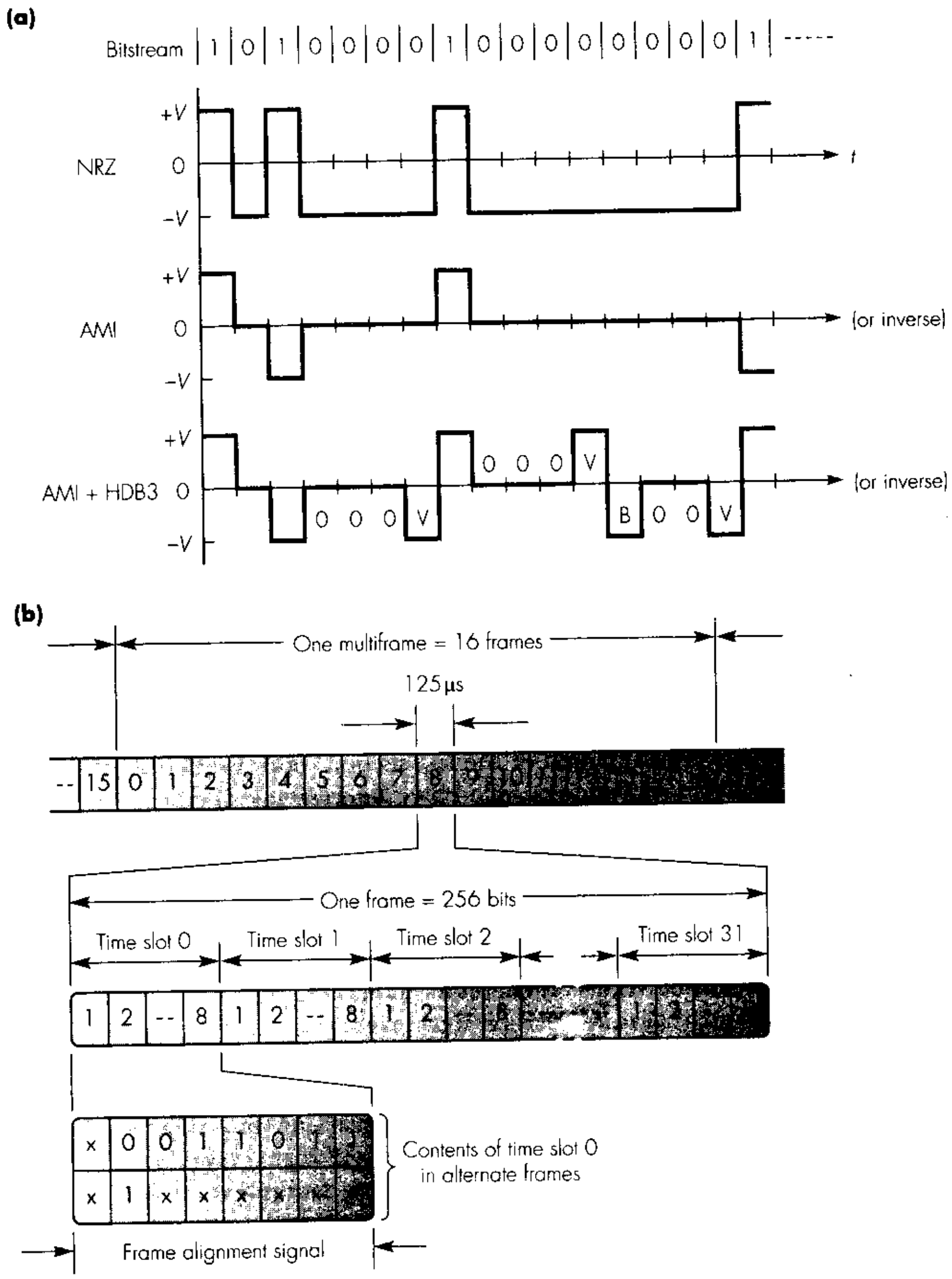


Figure 7.16 ISDN 2.048 Mbps primary rate interface: (a) line code; (b) frame and multiframe structure.

also is carried out in a hierarchical way by progressively multiplexing together multiple lower-level multiplexed streams.

The early multiplexers used in the trunk network operated in an analog mode and the newer digital multiplexers were introduced in an incremental way as these were upgraded. As a result, although all the replacement digital multiplexers operate at nominally the same bit rates, small variations in the timing circuits used in each multiplexer mean that, in practice, there are small differences in their actual bit rates. Hence when multiplexing together two or more lower-order multiplexed streams, steps have to be taken to compensate for the small differences in the timing of each stream. To overcome such differences, an output (multiplexed) bit rate that is slightly higher than the sum of the combined input bit rates is used. Any bits in the output bitstream that are not used are filled with what are called **justification bits**. The resulting set of higher-order multiplexed rates form the plesiochronous digital hierarchy.

The two alternative primary rate access circuits we described in the last section – 1.544 and 2.048 Mbps – form what is called the **primary multiplex group** of a related PDH. In the case of the 1.544 Mbps multiplex this is called a **DS1** or **T1** circuit and in the 2.048 Mbps multiplex an **E1 circuit**. Each is at the lowest level of the related hierarchy and hence all of the higher-level multiplexed groups contain multiples of either 24 or 32 64 kbps channels. The bit rate and derivation of the two sets of multiplexed groups are summarized in parts (a) and (b) of Figure 7.17.

As we explained in the last section, both primary multiplex groups are derived using what is called **byte interleaving** since the multiplexed stream comprises an 8-bit byte from each channel. This is done since a PCM sample of a speech signal is 8 bits and hence it is convenient electronically to multiplex together the complete set of 8-bit samples from each channel. In contrast, when multiplexing a number of primary-rate groups together, since each bitstream is independent of the others and arrives at the multiplexer bit serially, the various higher-level multiplex groups are formed using **bit interleaving**; that is, as each bit from each group arrives – 1 bit per group – they are transmitted out immediately in the output bitstream.

In the same way that additional bits (to the user data bits) are required in each primary multiplex group for framing and maintenance purposes, so additional bits are present in the various higher-level bitstreams for framing – to enable the corresponding receiving multiplexer to interpret the received bitstream on the correct multiplexed group boundaries – and maintenance. Hence the bit rates of all the higher-level multiplexed streams shown in Figure 7.17 contain additional bits for framing and maintenance purposes. For example, an E2 circuit contains the bitstreams from four 2.048 Mbps E1 circuits. Hence, since $4 \times 2.048 = 8.192$ Mbps and the actual bit rate is 8.448 Mbps, 0.256 Mbps are used. The various bit rates shown in the figure are often abbreviated to 1.5, 3, 6, 44, 274, 565 and 2, 8, 34, 140, 565 respectively.

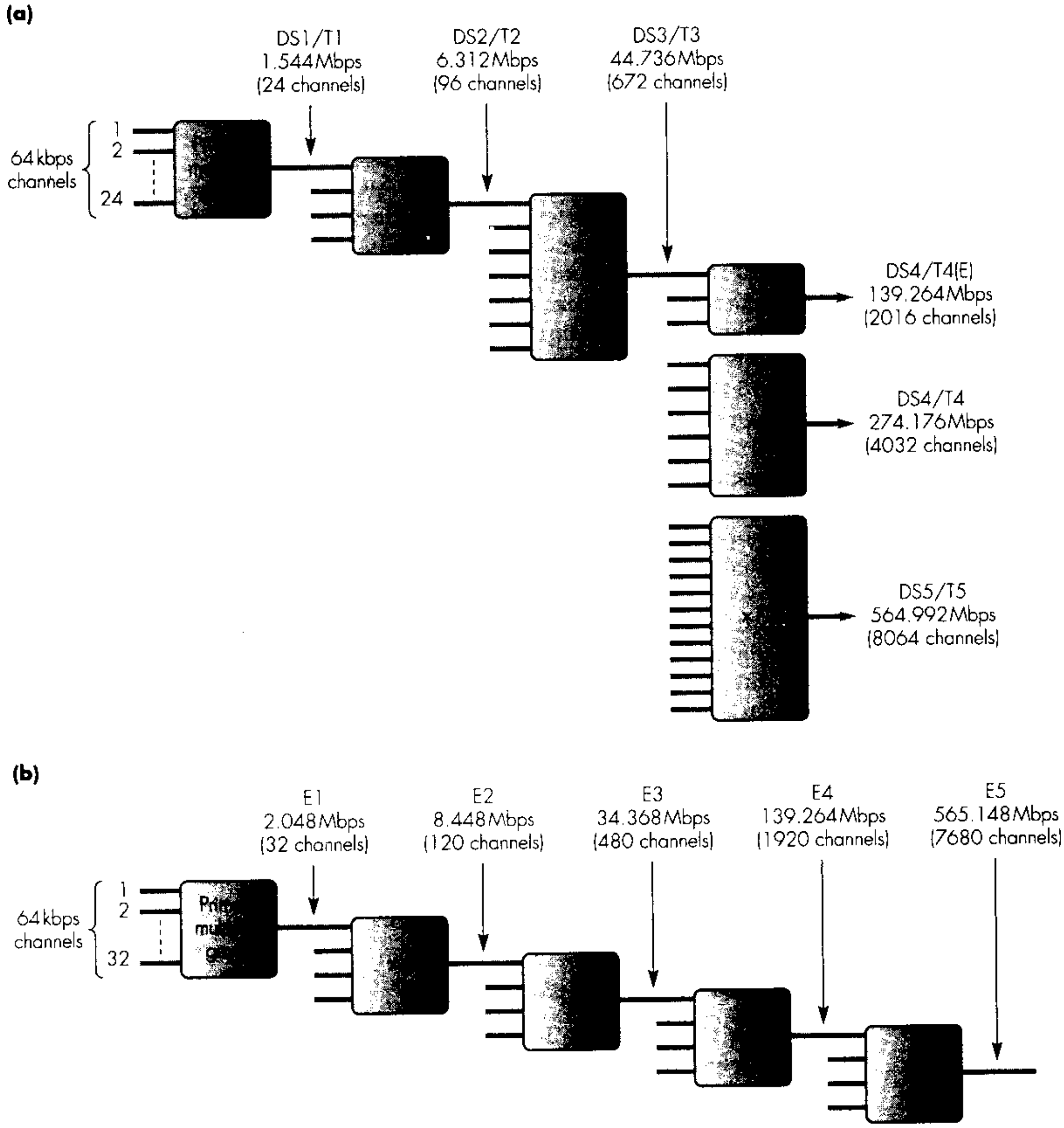


Figure 7.17 Plesiochronous digital hierarchies: (a) 1.544 Mbps derived multiplex hierarchy (b) 2.048 Mbps derived multiplex hierarchy.

Although the use of justification bits at each level in the hierarchy does not in itself pose a problem, their presence means that we cannot identify precisely the start of a lower-level multiplex bitstream within a higher-order stream. The effect of this is best seen by considering a typical operational requirement. Assume three switching centers/exchanges located in different towns/cities are interconnected by 140 Mbps (PDH) trunk circuits as shown in Figure 7.18(a). A business customer, with sites located somewhere between them, makes a request to link the sites together with, say, 2 Mbps leased circuits to create a private network. This is shown in schematic form in Figure 7.18(b). Because it is not possible to identify a lower bit rate channel from the higher-order bitstream, the operator must fully demultiplex the 140 Mbps stream down to the 2 Mbps level before this can be allocated to the customer. This stream must then be remultiplexed back into the 140 Mbps stream for onward transmission. This type of demultiplexing/multiplexing operation is performed by a device called a **drop-and-insert** or **add-drop multiplexer (ADM)** and, as we can deduce from Figure 7.18(c), the equipment required to meet this relatively simple request is very complicated.

Although it is not shown in the figure, at each switching office/exchange the allocated 2 Mbps leased circuit must be similarly identified and the switch bypassed in order to form a direct link between the customer sites. Hence when leased circuits are provided for customers in this way, careful records must be kept of the circuits and equipment being used for each customer so that if a fault is reported, appropriate remedial action can be taken. In practice, the provision of only basic performance monitoring within the frame formats of the PDH means that normally, it is the customer who has to alert the provider of the occurrence of faults.

To overcome these limitations, the more flexible synchronous digital hierarchy (SDH) is now used for all new installations. As we shall explain below, in addition to providing a more flexible transmission network which can be readily reconfigured to meet ever changing and expanding requirements, SDH equipment can be configured remotely and has a richer set of maintenance and error reporting functions.

7.2.5 Synchronous digital hierarchy

SDH was developed by Bellcore in the United States under the title of **synchronous optical network (SONET)**. All SDH equipment is synchronized to a single master clock. The basic transmission rate defined in the SDH is 155.52 Mbps – abbreviated to 155 Mbps – and is known as a **synchronous transport module level 1** signal or simply **STM-1**. Higher rates of **STM-4** (622 Mbps) and **STM-16** (2.4 Gbps) are also defined. In the SONET hierarchy the term **synchronous transport signal (STS)** or sometimes **optical signal (OC)** is used to define the equivalent of an STM signal. In SONET the lower rate of 51.84 Mbps forms the first level signal – **STS-1/OC-1**. An STM-1 signal is produced by multiplexing three such signals together and hence is equivalent to an STS-3/OC-3 signal.

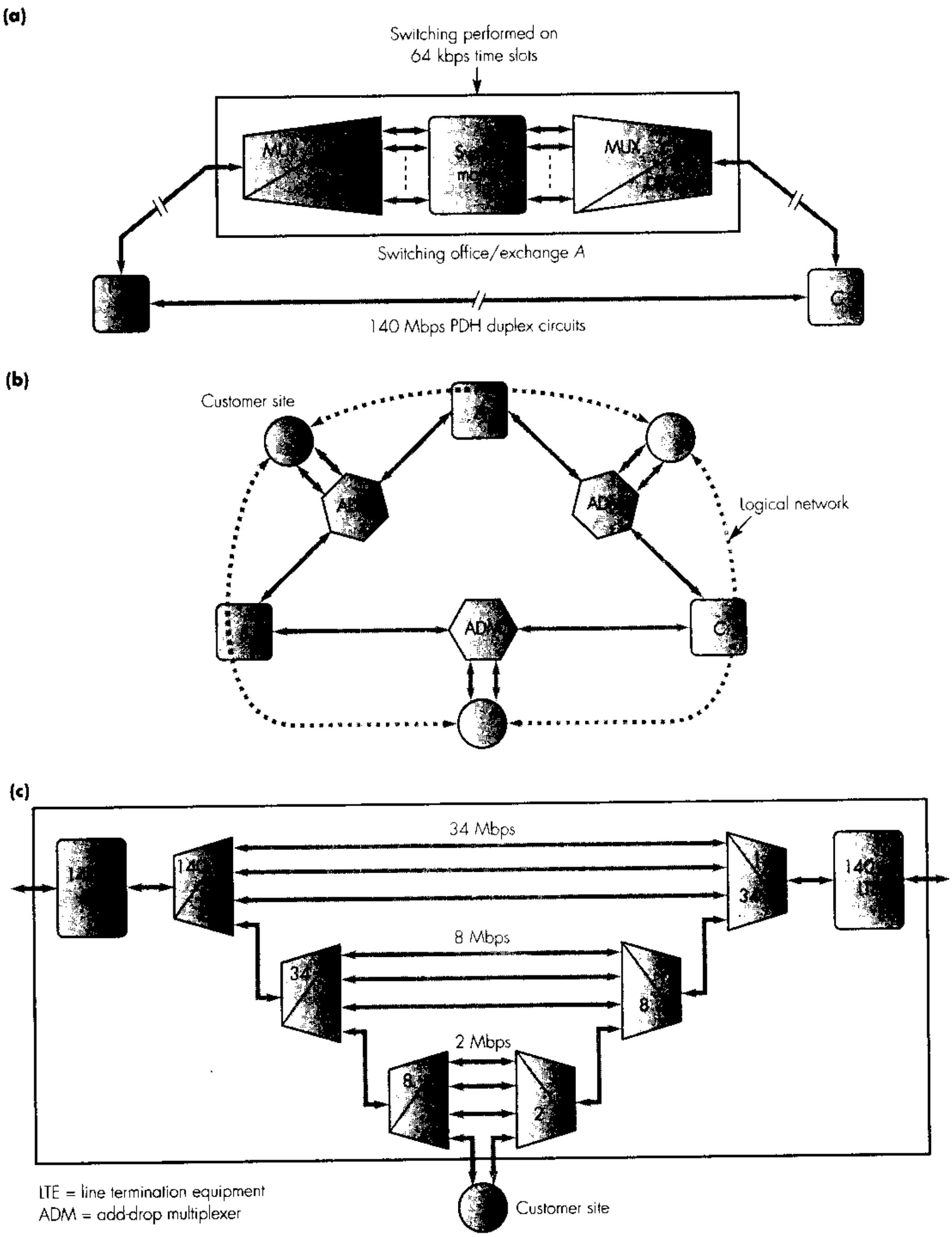


Figure 7.18 Private network provision with a PDH transmission network: (a) existing network; (b) modified network; (c) ADM principles.

As with the PDH, the STM-1 signal consists of a repetitive set of frames which repeat with a period of 125 microseconds. The information content of each frame can be used to carry multiple 1.5/2/6/34/45 or 140 Mbps PDH streams.

Each of these streams is carried in a different **container** which also contains additional **stuffing bits** to allow for variations in actual rate. To this is added some control information known as the **path overhead** which allows such things as the bit error rate (BER) of the associated container to be monitored on an end-to-end basis by network management. The container and its path overhead collectively form a **virtual container (VC)** and an STM-1 frame can contain multiple VCs either of the same type or of different types. Some example multiplexing alternatives are shown in Figure 7.19. Note that the first digit of the lowest level container – and hence VC – is a 1 and the second digit indicates whether it contains a 1.5 Mbps PDH signal (1) or 2 Mbps (2).

The higher-order transmission rates are produced by multiplexing multiple STM-1 (STS-3/OC-3) signals together. For example, an STM-16 (STS-48/OC-48) signal is produced by multiplexing either 16 STM-1 (STS-3/OC-3) signals or 4 STM-4 (STS-12/OC-12) signals. To provide the necessary flexibility for each higher-order signal, in addition to the overheads

SONET	SDH	Bit rate (Mbps)
STS-1/OC-1		51.84
STS-3/OC-3	STM-1	155.52
STS-9/OC-9		466.56
STS-12/OC-12	STM-4	622.08
STS-18/OC-18		933.12
STS-24/OC-24		1244.16
STS-36/OC-36		1866.24
STS-48/OC-48	STM-16	2488.32

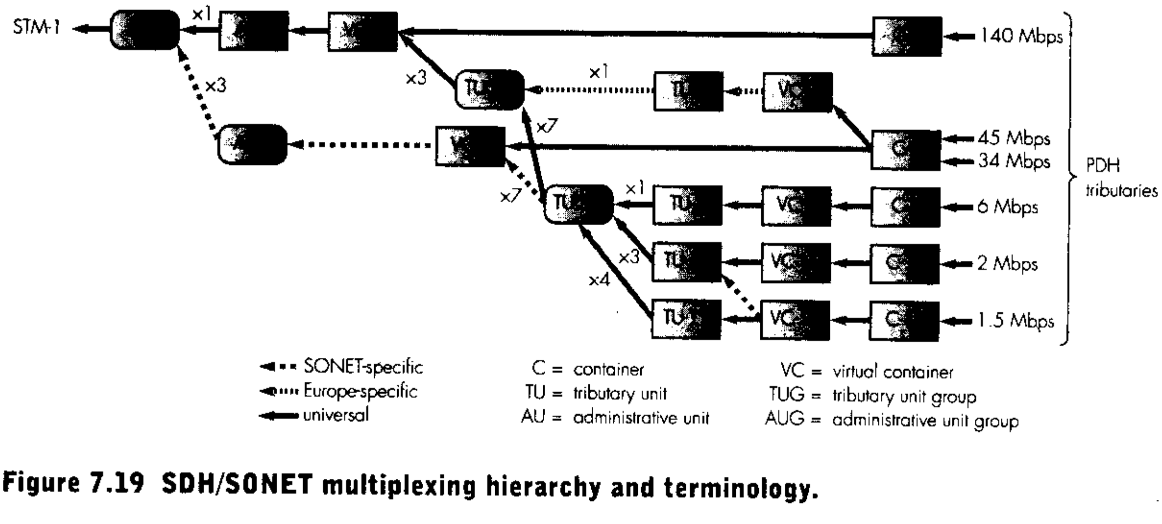


Figure 7.19 SDH/SONET multiplexing hierarchy and terminology.

at the head of each lower-level STM frame, a pointer is used to indicate the position of the lower-level STM frame within the higher-order frame.

We can best describe the structure of an SDH/SONET frame in relation to a complete synchronous transmission system since each frame contains management information relating to each constituent part. These parts include **sections**, **lines** and **paths** and their interrelationships are shown in Figure 7.20(a).

A section is a single length of transmission cable and both ends of the cable are terminated with a **section termination equipment (STE)**. An example of an STE is a repeater which regenerates the optical/electrical signals

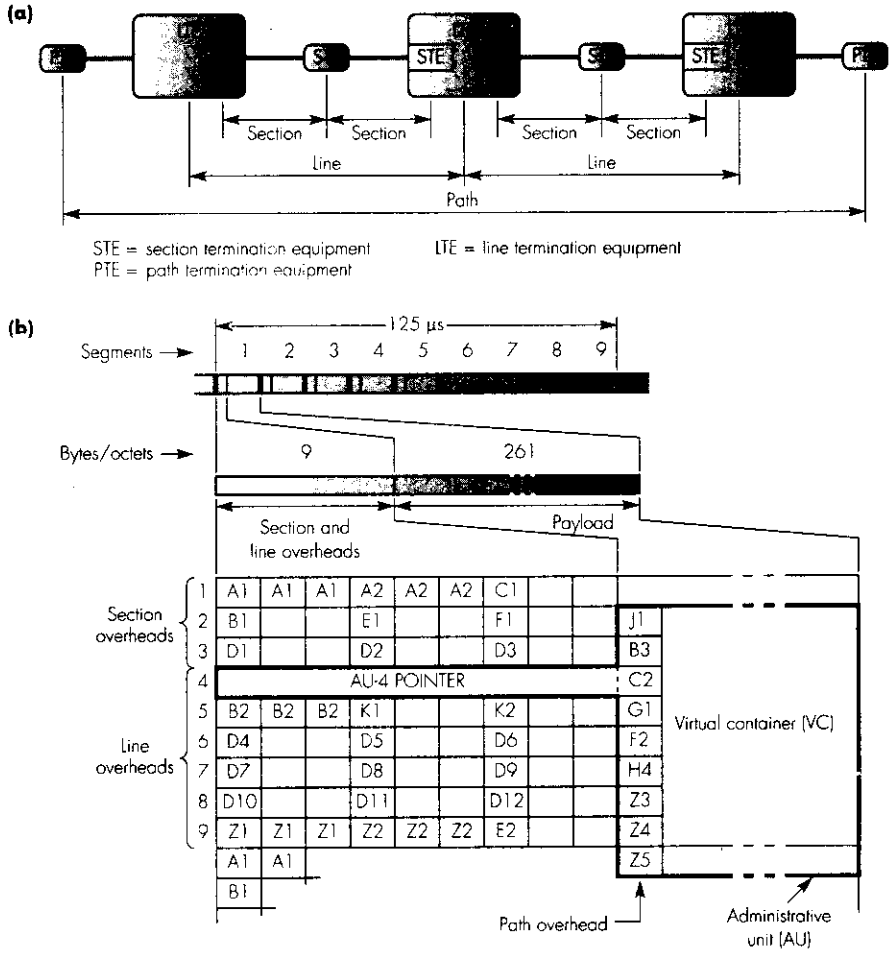


Figure 7.20 SDH/SONET detail: (a) managed entities; (b) frame format; (c) example VC mapping.

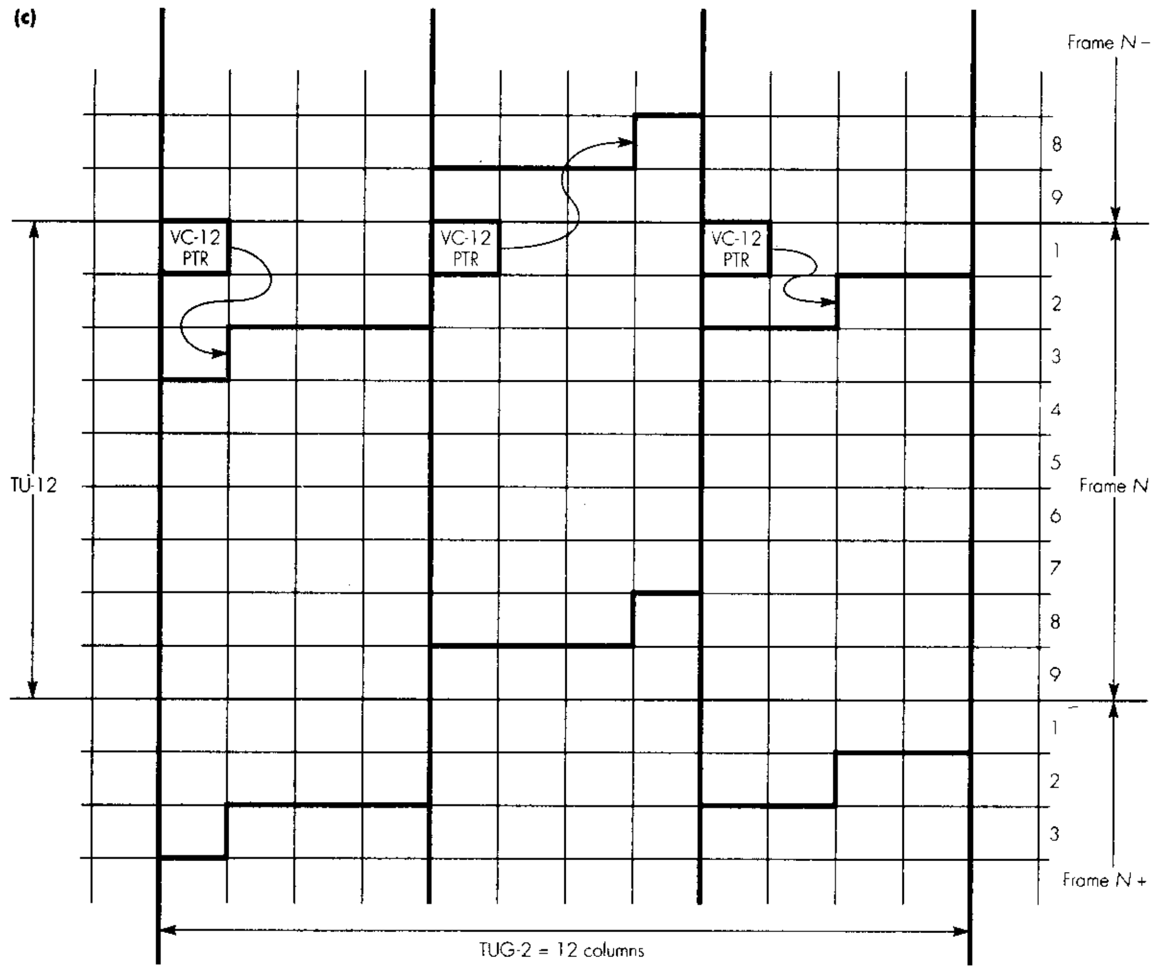


Figure 7.20 Continued.

being transmitted on this section of cable. A line extends across multiple cable sections and is terminated by a **line termination equipment** (LTE). Examples of LTEs are multiplexers and switching nodes. A path is an end-to-end transmission path through the complete transmission system. Each end of the path is terminated by a **path termination equipment** (**PTE**).

The structure of an STM-1 (STS-3/OC/3) frame is shown in Figure 7.20(b). As we can see, each frame comprises 2430 bytes/octets and repeats every 125 microseconds producing a bit rate of 155.52 Mbps. A frame comprises nine 270-byte **segments** each of which has a 9-byte header associated with it and a 261-byte **payload**. The header bytes are known as **overheads**. Normally, the segments making up a frame are shown with the segments one on top of the other since specific bytes in each header relate to one another.

The **section overhead** bytes relate to the management of a specific section. As we can see in Figure 7.20(b), some bytes are replicated for error protection purposes. The use of each byte is as follows:

A1–A2	Always the first bytes transmitted; used for framing.
B1	An 8-bit parity check used to monitor bit errors on the section.
C1	Identifies a specific STM-1 frame in a higher-order (STM-n) frame.
D1–D3	Form a data communication channel for network management messages relating to the section.
E1	Used for orderwire channels which are voice channels used by maintenance personnel.
F1	User channels, available for the management of customer premises equipment.

The **line overhead** bytes relate to the management of a complete line. The use of each byte is as follows:

B2	An 8-bit parity check used to monitor bit errors on the line.
D4–D12	Form a data communication channel for network management messages relating to the line.
E2	Orderwire channels relating to the complete line.
K1–K2	Form a signaling channel for automatic protection switching of the complete line.
Z1–Z2	Reserved for national use.

The columns in the payload field can be assigned in various ways to carry lower bit rate signals. To transport lower-level PDH streams – known as **tributaries** – the payload capacity in each container is allocated in integral numbers of columns. Each container has a column of path overhead bytes assigned to it and collectively these form the VC.

The identity of each byte given in Figure 7.20(b) and their uses are as follows:

J1	This byte verifies the VC path connection.
B3	8-bit parity for monitoring the bit error rate of the path.
C2	Indicates the composition of the VC payload.
G1	Used by the receiver to return the status of the received signal back to the transmitter.
F2	Provides a user data communication channel.
H4	Indicates whether the payload is part of a multiframe.
Z3–Z5	Available for national use.

A pointer is placed at the head of each VC and is used to indicate the start of the VC relative to the start of each frame. Note that if the container contains a PDH tributary, then the pointer value may change from frame to frame owing to possible timing differences. Different combinations of VCs can be used to fill up the payload area of a frame with smaller VCs being carried within larger ones. The VC and its pointer are known as a **tributary unit** (**TU**) if it carries a lower-order tributary or a **tributary unit group** (**TUG**) if it carries a number of lower-order tributaries. The largest VC in an STM-1 frame is known as an **administrative unit** (AU) and, as we show in Figure 7.20(b), the pointer to the start of this is written into the first octet position of the line overhead.

The example given in Figure 7.20(c) shows how three VC-12s can be carried within a TUG-2 frame. Each VC-12 comprises four columns of the STM-1 payload area and hence the TUG-2 comprises 12 columns. The VC-12 and its pointer forms a TU-12. The pointer always occupies the first byte position but, if the timing of the VC-12 contents relative to the STM-1 frame varies, then the position of the VC-12 is allowed to slip to accommodate this with the value in the pointer changing so that it always points to the first byte in the VC. A VC-12 will accommodate $4 \times 9 = 36$ bytes. Hence, since a VC-12 comprises 33 bytes – 32 bytes for the E1 frame plus 1 byte for the pointer – the remaining bytes are filled with what is referred to as **fixed stuff**.

In order to carry other signals which do not have a container defined to accept them, two or more TUs can be combined together using a technique known as **concatenation**. For example, five TU-2s can be concatenated to carry a single 32 Mbps signal. Four such signals can then be carried in a VC-4 instead of three if a standard C-3 container has been used. This technique is also used for the transport of ATM traffic.

All SDH equipment has software associated with it known as a **network management** (**NM**) **agent** and the communication channels in the overhead bytes are used by this to report any malfunctions of sections, lines, or paths to a central network management station. They are also used for the latter to download commands to change the allocation of the payload field associated with each STM-1 frame. For example, SDH ADMs can be configured – and reconfigured – remotely to provide any desired bandwidth mix without the need for demultiplexing. The general principle is shown in Figure 7.21. Redundant (standby) links are used between each pair of SDH multiplexers and these can be brought into service using commands received from a remote network management station.

7.3 Switching systems

As we showed earlier in Figure 7.1 and explained in the accompanying text, the total switching system comprises a hierarchical set of exchanges of varying switching capacity: local exchanges/end offices, regional tandem exchanges, and national tandem exchanges. Also, as we explained in Section 7.2.1, although within the local access network some analog transmission is

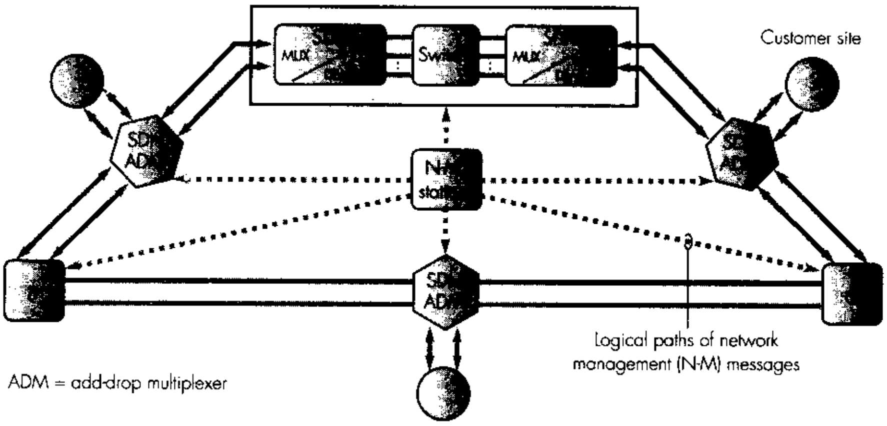


Figure 7.21 Service provision with SDH equipment using network management.

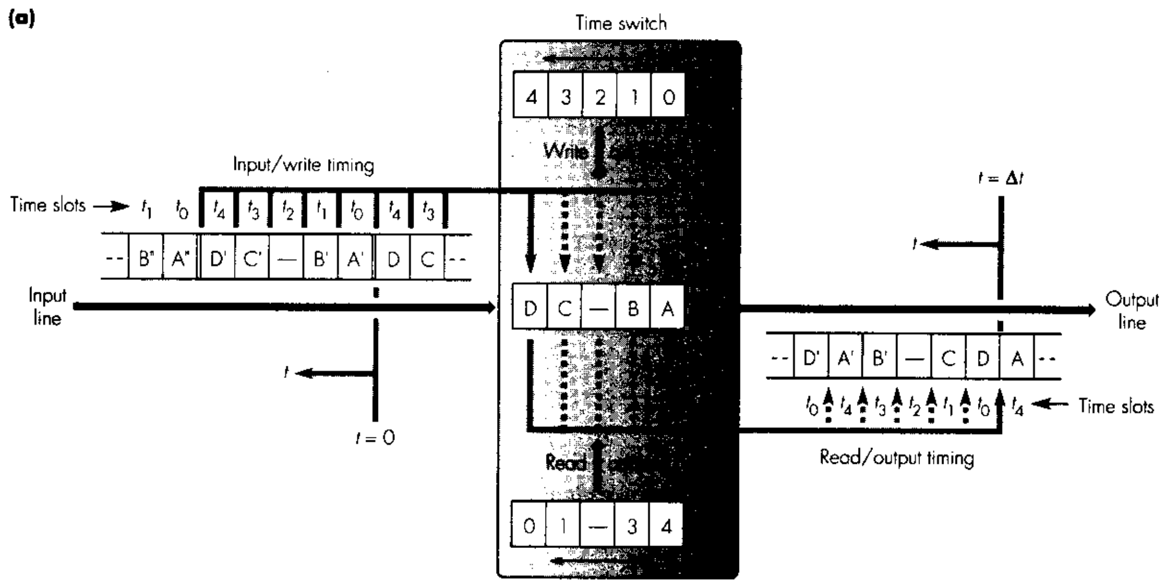
still in use, the use of remote concentrators and multiplexers means that all levels of exchange involve the switching of 125 μ/64 kbps time slots/channels from one incoming (PCM) line to another. Hence all switching exchanges operate in a similar way and the major difference between them is the volume of traffic that is switched. We shall restrict our discussion here to the basic principles on which all digital switching exchanges are based.

As we have just indicated, the basic requirement of a digital switching exchange is to switch the contents of each set of time slots in each of its incoming lines to any time slot position in any of its outgoing lines. Normally, the time slots in each input and output line are allocated independently and hence most connections involve a different time slot position in the input and output lines. As a result, a digital switch involves two different types of switching function: one to perform the switching between the input and output lines involved – **space switching** – and the other to perform the switching between the two time slot positions involved – **time-slot interchange** or **time switching**. We shall explain the principles of each first and then how they are combined to form a digital switching unit.

7.3.1 Time switching

The principle of operation of a time switch is shown in Figure 7.22. Assuming each PCM frame comprises N time slots/channels, the role of the time switch is to switch the contents of each time slot position read from the input line – an 8-bit byte/octet – and output it to a different time slot position in the output line. Hence, as we can deduce from the figure, since the contents of each time slot position in the input line needs to be switched to a later time

(a)



(b)

Time, t	Input byte	FS contents	Output byte	Delay
0	D	DC–BA	A	
t_0	A'	DC–BA'	D	
t_1	B'	DC–B'A'	C	
t_2	–	DC–B'A'	–	
t_3	C'	DC'–B'A'	B'	2ST
t_4	D'	D'C'–B'A'	A'	4ST
t_0	A"	D'C'–B'A"	D'	1ST
t_1	B"	D'C'–B"A"	C'	3ST

MC = modulo-N counter FS = frame store CS = connection store
Δt = switching delay = $T_{write} + T_{read}$ ST = slot time

Figure 7.22 Time switch principles: (a) switching schematic; (b) timing.

slot position in the output line, it is necessary to store the byte read from each time slot in a memory containing the same number of locations as there are time slots in the PCM frame. The memory is known, therefore, as the **frame store (FS)**.

As we show in the figure, because both the input and output (PCM) lines operate bit-serially, on input there is a serial-to-parallel conversion and on output a parallel-to-serial conversion. This means, therefore, that the contents of each of the time slots from the input line are written into the frame store after the last bit of the time slot has been received; that is, at the end of the time slot period. Conversely, the contents from each location in the frame store awaiting output are read at the start of the appropriate time slot period. As we can deduce from this, there is a time delay associated with a time switch which varies between 1 slot time, if the byte is to be output in the immediately following time slot, and $N - 1$ slot times if it is to be output N time slots later; both times being in addition to the basic write/read cycle time of the frame store, Δt.

The addresses used to write into the frame store are obtained from a modulo-N counter that increments from 0–($N - 1$) in time synchronism with the arrival of each byte. Hence in the example, since $N = 5$, the counter increments from 0–4 and then repeats and, for each count value, the contents of the time slot just arrived are written into the frame store at the address currently held in the counter.

A second store (also containing N locations) called the **connection store** (**CS**) is then used to hold the set of addresses – and hence time slot positions – that are to be used for reading from the frame store and hence the order in which the current set of bytes in the frame store are to be transmitted out. So in the example, since time slot 0 in the output line is to contain the contents of time slot 4 from the input line, then the address 4 is present in the connection store at location 0. Again, the contents of the frame store are read in time synchronism with the start of each time slot period.

7.3.2 Space switching

A basic space switch comprises an array of M input and M output lines and an associated set of crosspoint switches. The signal present on each of the input lines is switched to a different output line by activating the appropriate

Example 7.2

A simple time switch is to be used [illegible] associated with (i) a 24-channel and [illegible] the size of the frame store, address [illegible] each case.

Answer:

(i) 24 channels:
Frame store = 24 locations [illegible]
Address counter = Modulo-24 [illegible]
Connection store = 24 locations [illegible]

(ii) 32 channels:
Frame store = 32 locations [illegible]
Address counter = Modulo-32 counter (00000–1111 [illegible]
Connection store = 32 locations each of 5 bits (00000–1111 [illegible]

crosspoint switch. The set of crosspoints to be activated are held in a connection store and an example is shown in Figure 7.23(a). As we can see, in this example there are four input and four output lines and hence the connection store contains the output line – and thus crosspoint switch to be activated – for each of the four input lines. The crosspoints remain activated for the duration of the call.

In the case of a PCM space switch, the contents of each of the N time slots from each of the M input lines may need to be switched to a different output line. Hence each position in the connection store shown in Figure 7.23(a) has N entries, one for each time slot position. The crosspoints are activated in time synchronism with the arrival of each new set of time slots on the M input lines. A simple example is shown in Figure 7.23(b).

As we can see, in this example the switch comprises four input and four output lines as before but the signal present on each input line is a 4-channel PCM signal. Hence in this case, each entry in the connection store contains a set of four values which collectively indicate the crosspoints that are activated during each of the four time slot positions. The entries in the connection store for each time slot position are determined by first noting the time slot contents on a particular output line and then tracing back through the switch to determine the input line on which this arrives. The corresponding entry in the connection store is then set to the output line number which, in turn, indicates the crosspoint to be activated during this slot time. For example, for time slot position t_2, output line 1 contains contents C. Hence, tracing back through the switch, we find that this arrives on input line 0 and hence the connection store entry for this line is 1.

Example 7.3

A 7-input, 7-output line space switch is to be used to switch the contents of each set of time slot positions associated with (i) a 24-channel and (ii) a 32-channel PCM system. Derive the size of the connection store and the number of bits required for each entry in the store for both systems.

Answer:

(i) 24-channels:
Connection store has $7 \times 24 = 168$ locations
Each entry must have 3 bits: entries 000–110 to indicate the output line number (0–6) and hence the crosspoint position to be activated. Entry 111 is then used to indicate no crosspoint is to be activated; that is, the time slot is not in use.

(ii) 32-channels:
Connection store has $7 \times 32 = 224$ locations
Each entry must have 3 bits as for (i).

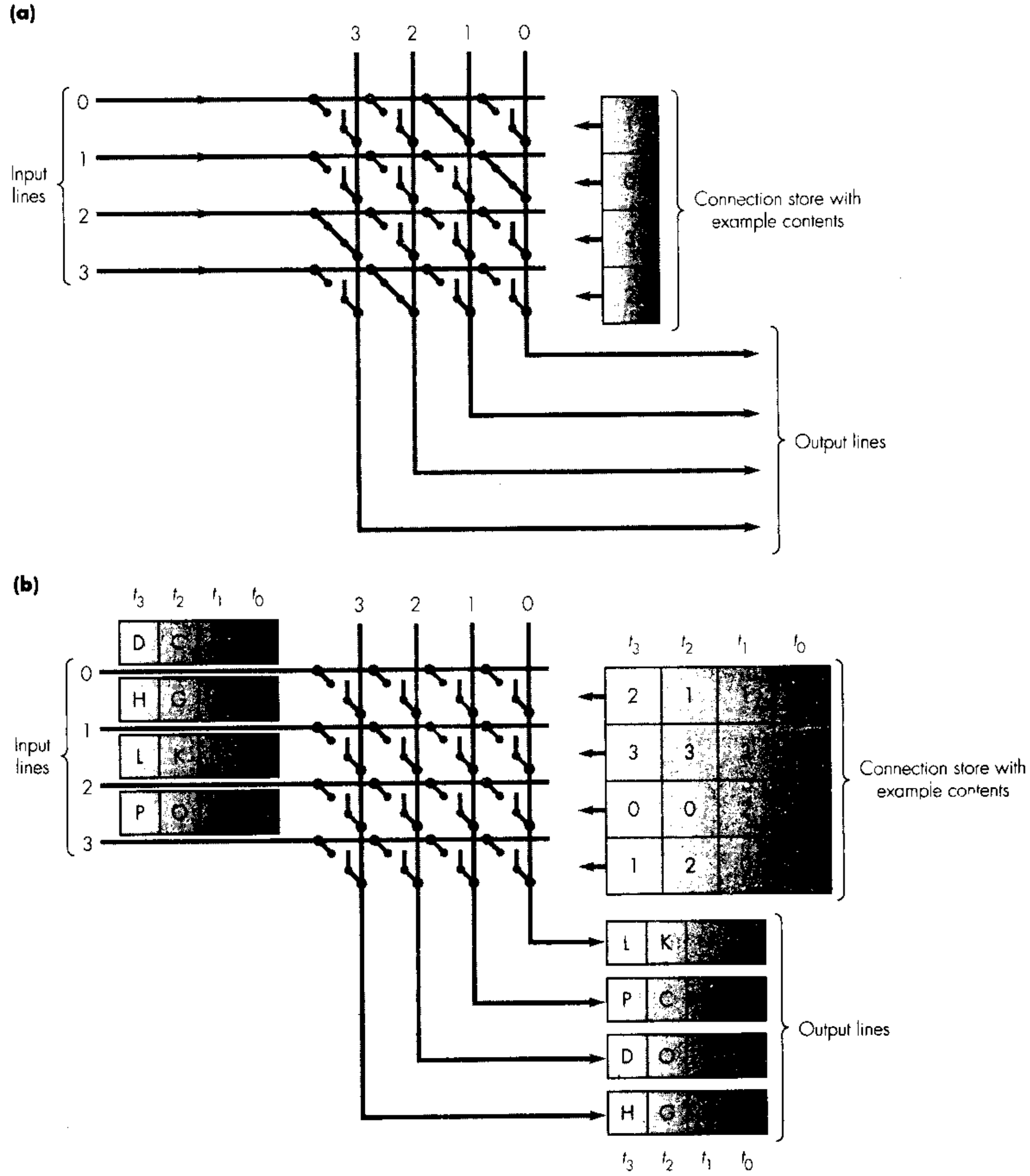


Figure 7.23 Space switch principle of operation: (a) basic space switch; (b) PCM space switch.

As we can deduce from the contents of the two sets of time slots associated with the input and output lines shown in Figure 7.23(a), a space switch simply changes the line on which each is transmitted and hence there is no delay associated with a space switch.

7.3.3 Digital switching units

As we explained earlier in Section 7.3, the requirement of a digital switching exchange is to be able to switch the contents of each time slot position in each of the input lines to a possibly different time slot position in a different output line. Therefore the overall switching operation involves a combination of both time and space switching, a simple example of which is shown in Figure 7.24(a).

As we can see, associated with each input line is a time switch which performs the time slot switching operation as defined by the contents of the related connection store. The single space switch then switches the rearranged set of time slots from one input line to the other.

With this simple configuration, since there are only two input/output lines there is always a time slot available in the required output line. As more lines are added, however, this is not necessarily the case. Although in terms of transmission, the time slots in each line can be allocated in a random way – during the call setup/signaling procedure – as existing calls terminate and new calls start, it is required also to ensure that the switching unit has the required capacity to switch the resulting number of calls. Hence for all but the smallest switches, switching units comprising multiple stages are used. As an example, a three-stage **time-space-time switch** is shown in Figure 7.24(b).

As we can see, each input line is passed through a separate time switch and each output line is preceded by a second time switch. The two sets of input and output time switches are then interconnected by a space switch which performs the required line switching operation. To perform the switching function, it is necessary to choose a time slot which is free in both the connection store of the input time switch and the frame store of the required output time switch. An example is shown in the figure which involves the switching of time slot 1 in input line 0 to time slot 8 in output line 2. To do this, the intermediate time slot chosen is 5.

As we can deduce from the figure, this carries out the switching operation in one direction only. However, since a duplex connection is required, a separate connection path in the reverse direction must also be established from input line 2 to output line 0. Hence, during the setting up of a new call/connection, the time slots to be used for the forward and reverse directions are reserved at the same time. Normally, to simplify the control of the switching network, the two time slots associated with each input/output line have a fixed separation between them; for example, 12 in the case of a 24-channel system and 16 in a 32-channel system. The same intermediate time slot is then used in both directions. In the example in Figure 7.24(b), the time slot separation is 12 and the intermediate time slot is 5 in both directions.

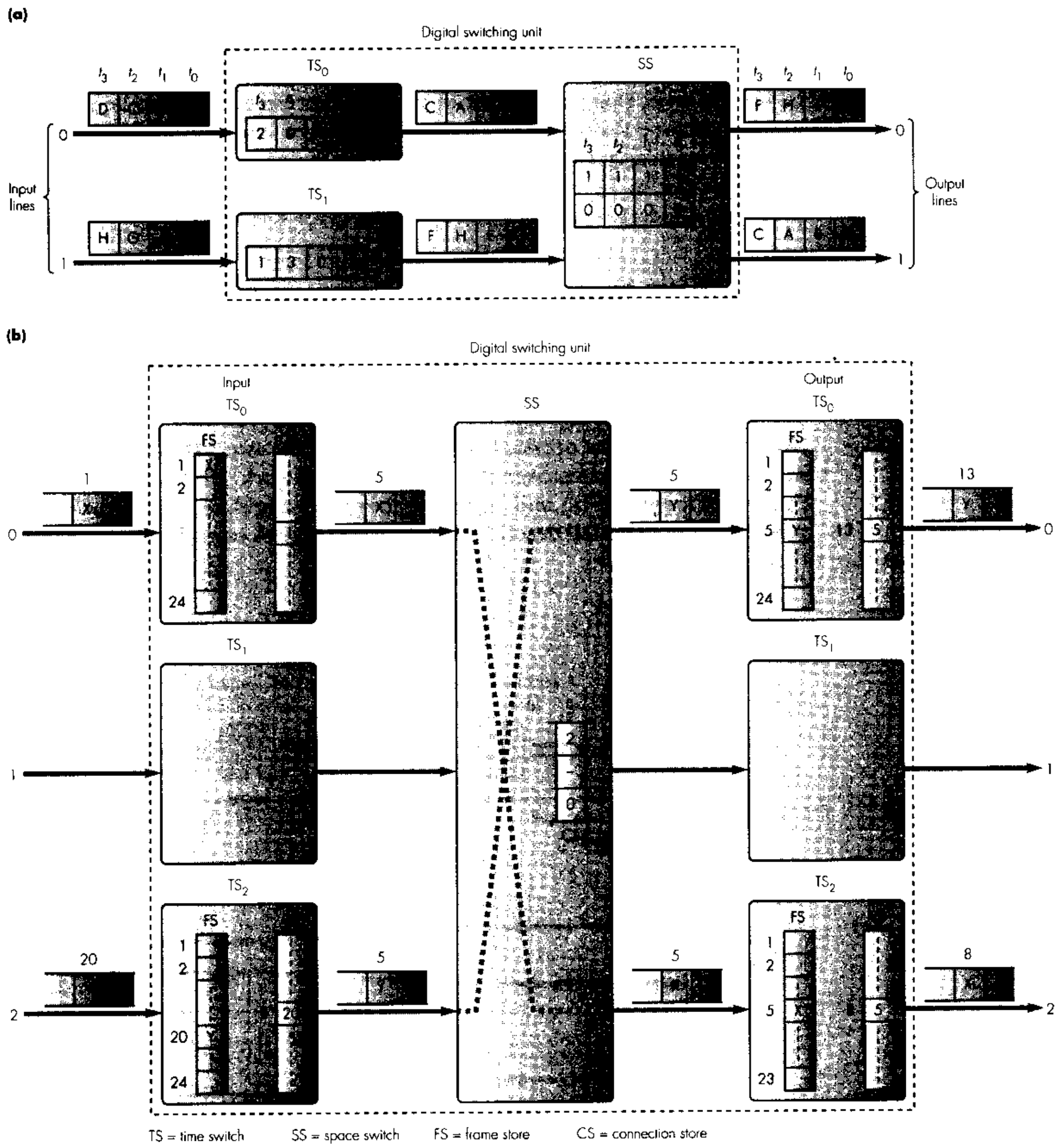


Figure 7.24 Digital switching units: (a) time-space switch; (b) time-space-time switch.

We can see from this example that since there is only a single space switch, a key factor in obtaining a free path through the switching network is the availability of a free intermediate time slot. In practice, traffic analysis can be used to show that the number of intermediate time slots must be twice the number of time slots in each input/output frame minus 1. This is achieved either by operating the space switch at twice the time slot arrival rate of the input lines or by having more than one space switch; for example, a large switch may operate with three intermediate space switches.

7.4 Signaling systems

The signaling operations associated with the setting up and clearing – also known as closing or tearing down – of a connection between two subscriber equipments involves two separate signaling systems: the first which operates over the local access networks associated with the two subscribers and the second which operates over the core trunk network. The two systems are shown in Figure 7.25(a) and, although they are interrelated, since each operates differently we shall describe the essential features of each separately.

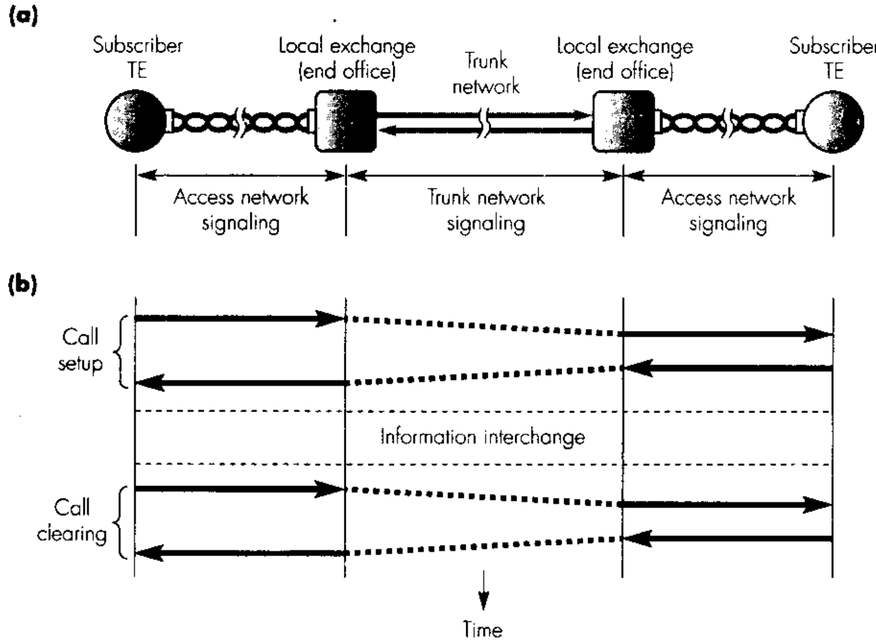


Figure 7.25 Signaling system components: (a) access and core trunk network components; (b) access network components.

7.4.1 Access network signaling

The basic operations associated with the setting up and closing down of a connection over the access network are shown in diagrammatic form in Figure 7.25(b). The steps involved are:

- call setup: this includes the dialing, ringing, and answer stages;
- interformation interchange: this is concerned with the exchange of information – speech/data – between the two subscribers;
- call clearing: this results in the disconnection/release of the connection and can be initiated by either subscriber.

In practice, these basic operations are carried out differently over the various types of access network. These include both analog (PSTN) and digital (ISDN) access networks, the latter including the signaling operations that are required between an RCU/RSU/PBX and a local exchange/end office. We shall explain each separately.

Analog access circuits

We explained the basic features of an analog access circuit in the text associated with Figure 7.2. As you may recall, most signaling operations involve the transmission of one or more single-frequency audio tones. A selection of these is shown in Figure 7.26(a) and an example of their use in the setting up and clearing of a call/connection is given in Figure 7.26(b).

As we can see, in this example the call is successful and a connection would be set up. If the called subscriber line was busy, however, then the busy tone would be returned to the calling subscriber who would then replace the handset.

In the case of a modem, the same call setup and clearing sequences are followed by the incorporation of appropriate circuits within the modem. In addition, some modems – for example the V.32 – use an error detection and correction protocol during the information interchange phase in order to achieve a more reliable transfer of information. This is known as **link access procedure for modems (LAPM)** and these modems transmit the source information in frames using bit-oriented synchronous transmission and an HDLC-based error correcting protocol the principles of which we explained in the last chapter. The applicability of LAPM is shown in Figure 7.27(a).

Each modem comprises two functional units: a **user** (DTE) **interface part (UIP)** and an **error correcting part (ECP)**. The LAPM protocol is associated with the latter while the UIP is concerned with the transfer of single characters/bytes across the local V.24 interface and with the interpretation of any flow control signals across this interface.

The UIP communicates with the ECP using a defined set of service primitives, as shown in the time sequence diagram in Figure 7.27(b). The different HDLC frame types used by the LAPM protocol entity to implement the various services are also shown.

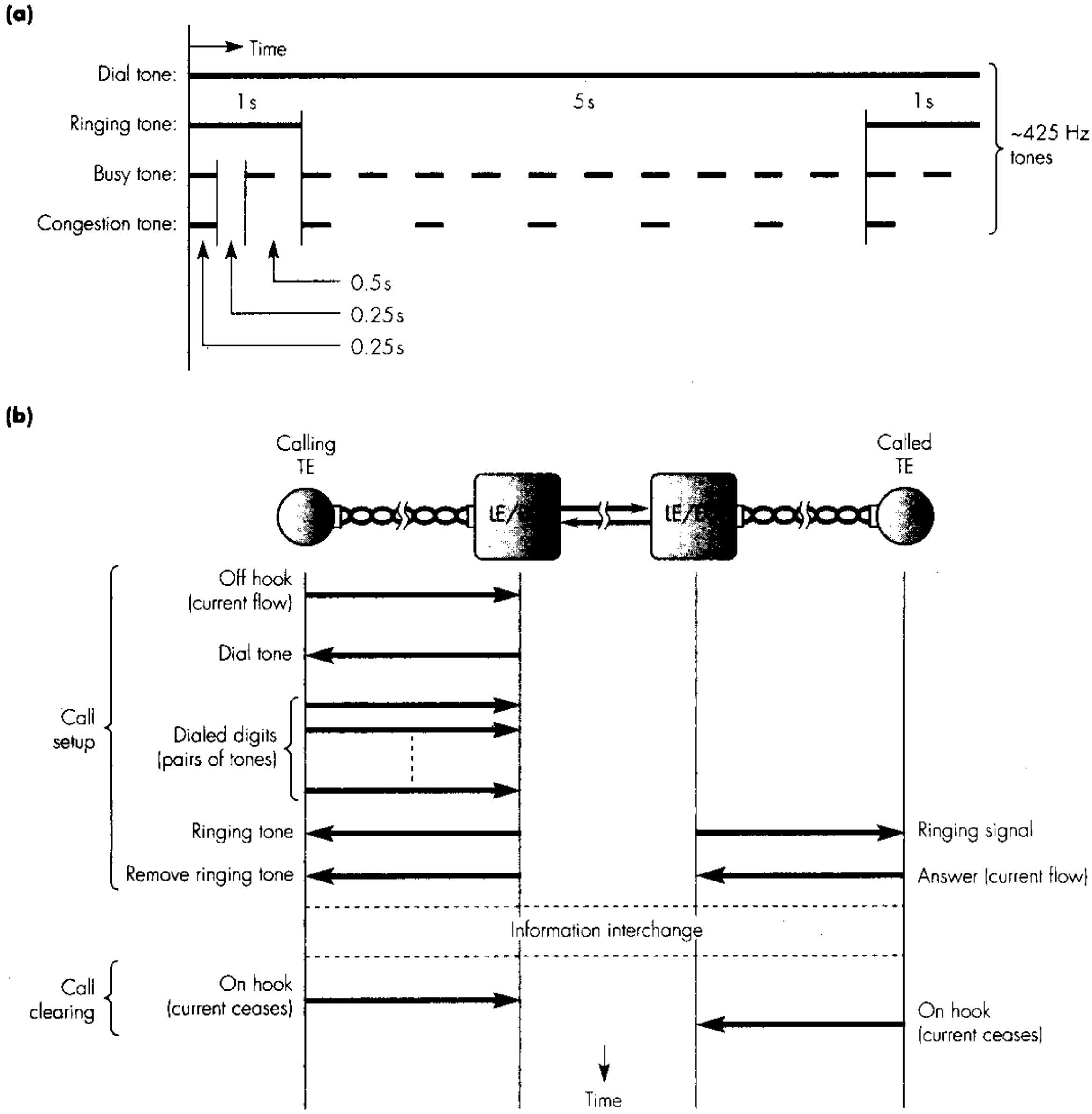


Figure 7.26 Analog access signaling: (a) a selection of the signals used; (b) sequence of signals exchanged to set up and clear a call.

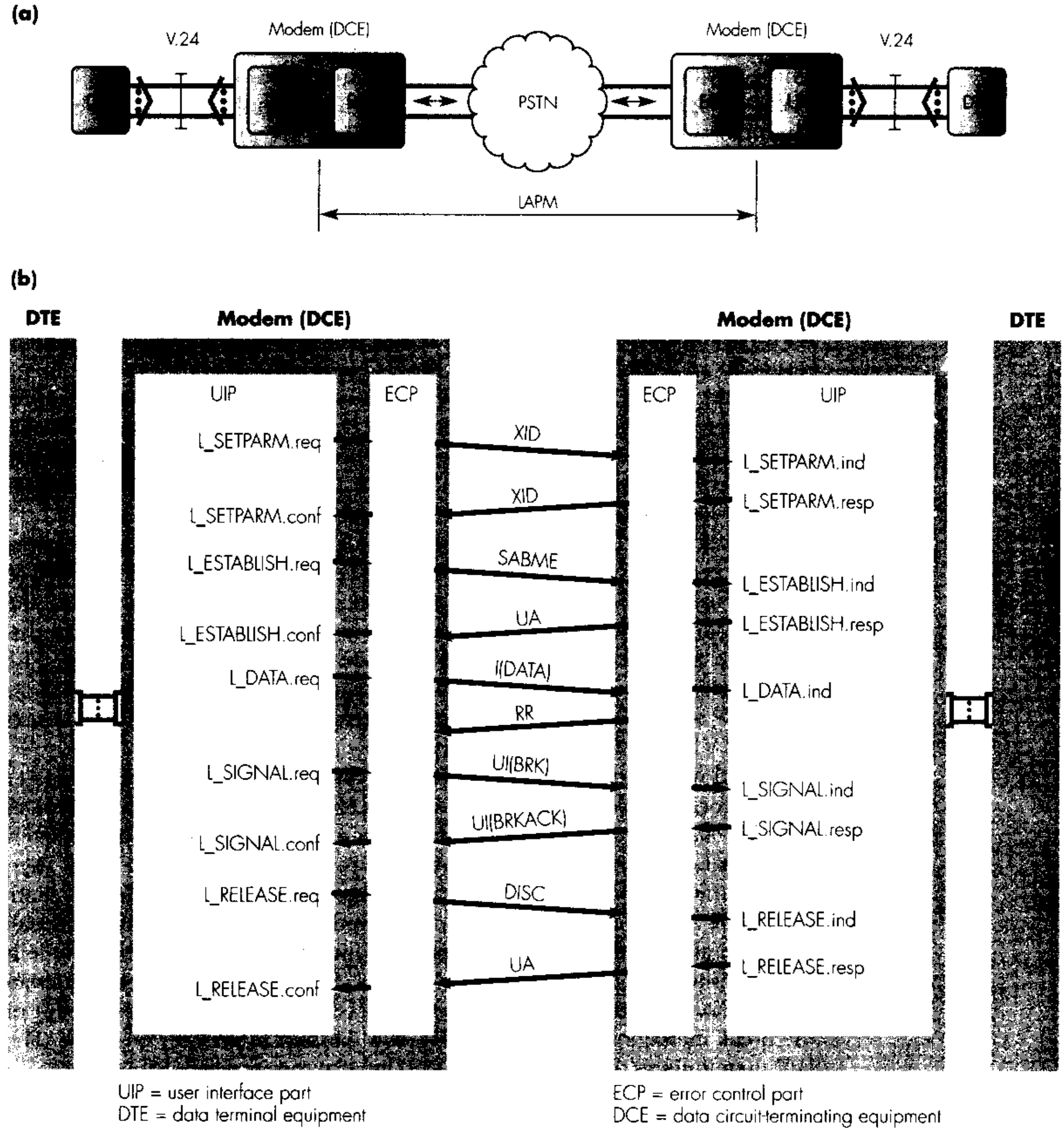


Figure 7.27 LAPM: (a) operational scope; (b) user service primitives and corresponding frame types.

Before establishing a (logical) link, the originating and responding ECPs must agree on the operational parameters to be used with the protocol. These parameters include the maximum number of octets in I-frames, the acknowledgment timer setting, the maximum number of retransmission attempts, and the window size. Default values are associated with each of these, but if they are not used, the originating UIP must issue an L_SET-PARM.request primitive with the desired operational parameter values. The values are negotiated when the two ECPs exchange two special unnumbered frames – known as **exchange identification (XID)** – one as a command and the other as a response.

Once the operational parameters have been agreed, a link can then be set up when the UIP issues an L_ESTABLISH.request primitive. This, in turn, results in an SABM (normal) or SABME (extended) supervisory frame being sent by the ECP. The receiving ECP then issues an L_ESTABLISH.indication primitive to its local UIP and, on receipt of the response primitive, the receiving ECP returns a UA-frame. On receipt of this frame, the originating ECP issues a confirm primitive and the (logical) link is now set up. Data transfer can then be initiated using the L_DATA service.

Typically, the UIP first assembles a block of data, comprising characters or bytes received over the V.24 interface, then passes the complete block to the ECP using an L_DATA.request primitive. The ECP packs the data into the information field of an I-frame as a string of octets and transfers this using the normal error correcting procedure of the HDLC protocol. The receiving ECP then passes the (possibly error corrected) block of data to its local UIP which transfers it a character (byte) at a time bit-serially across the local V.24 interface.

If a flow control (break) condition is detected during the data transfer phase – for example, an X-OFF character is received or the DTR line becomes inactive – then the UIP stops outputting data to the local DTE and immediately issues an L_SIGNAL.request primitive to its local ECP. The local ECP then informs the distant ECP to (temporarily) stop sending any more data by sending a BRK (break) message in an unnumbered information (UI) frame. Recall that this, as the name implies, does not contain sequence numbers since it bypasses any error/flow control mechanisms. The receiving ECP then issues an L_SIGNAL.indication primitive to its local UIP and acknowledges receipt of the break message by returning a BRKACK message in another UI-frame. The UIP then initiates the same flow control signal across its own V.24 interface.

Finally, after all data has been transferred, the link is cleared when the originating UIP issues an L_RELEASE.request primitive. Again this is a confirmed service and the associated LAPM frames are DISC and UA.

ISDN digital access circuits

As we explained in Section 7.2.3, there are two alternative physical interfaces to an ISDN: basic rate and primary rate, the latter being either 1.544 Mbps or 2.048 Mbps. The basic rate interface provides a separate 16 kbps D-channel for signaling – in addition to the two 64 kbps user B-channels – and the two alternative primary rate interfaces include a 64 kbps signaling channel. Since both interfaces are digital, the setting up and clearing of calls/connections is carried out by the exchange of (signaling) messages over the respective D-channel. This mode of operation is called **channel associated signaling** (**CAS**).

The signaling system associated with an ISDN digital access circuit is known as **digital subscriber signaling number one** (**DSS1**) and its composition is shown in Figure 7.28. Since the signaling messages must be received free of any transmission (bit) errors, a reliable data link protocol known as **link access procedure D-channel** (**LAPD**) is used to control their transfer over the interface. This is based on the HDLC protocol and is defined in **ITU-T Recommendation Q.921**. The format of the actual signaling messages and the protocol that is used to control their transfer is defined in **ITU-T Recommendation Q.931**. We shall describe the basic features of both protocols separately.

Q.921 (LAPD) Two types of service have been defined for use with LAPD. A time sequence diagram showing the two sets of service primitives is shown in Figure 7.29. As we can see, both an unacknowledged (connectionless) and an acknowledged (connection-oriented) service are supported. The connection-oriented service is used to transfer call setup messages between an item of user equipment — a telephone or a DTE – and the local exchange. The associated protocol incorporates error control The connectionless service is used for the transfer of management-related messages and the associated protocol uses a best-effort unacknowledged approach.

As we explained earlier in Section 7.2.3, multiple items of terminal equipment can time-share the use of the access circuit. However, all the layer 3 signaling messages are sent to a specific terminal equipment using the address field in the header of each LAPD frame the general structure of which is shown in Figure 7.30.

Two octets are used for the address field. These consist of two subaddresses: a **service access point identifier** (**SAPI**) and a **terminal endpoint identifier** (**TEI**). Essentially, the SAPI identifies the class of service to which

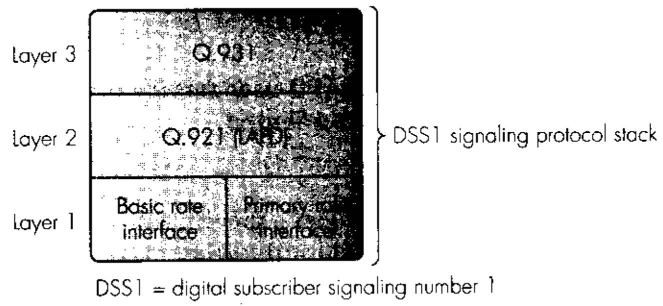


Figure 7.28 ISDN digital access signaling protocol set.

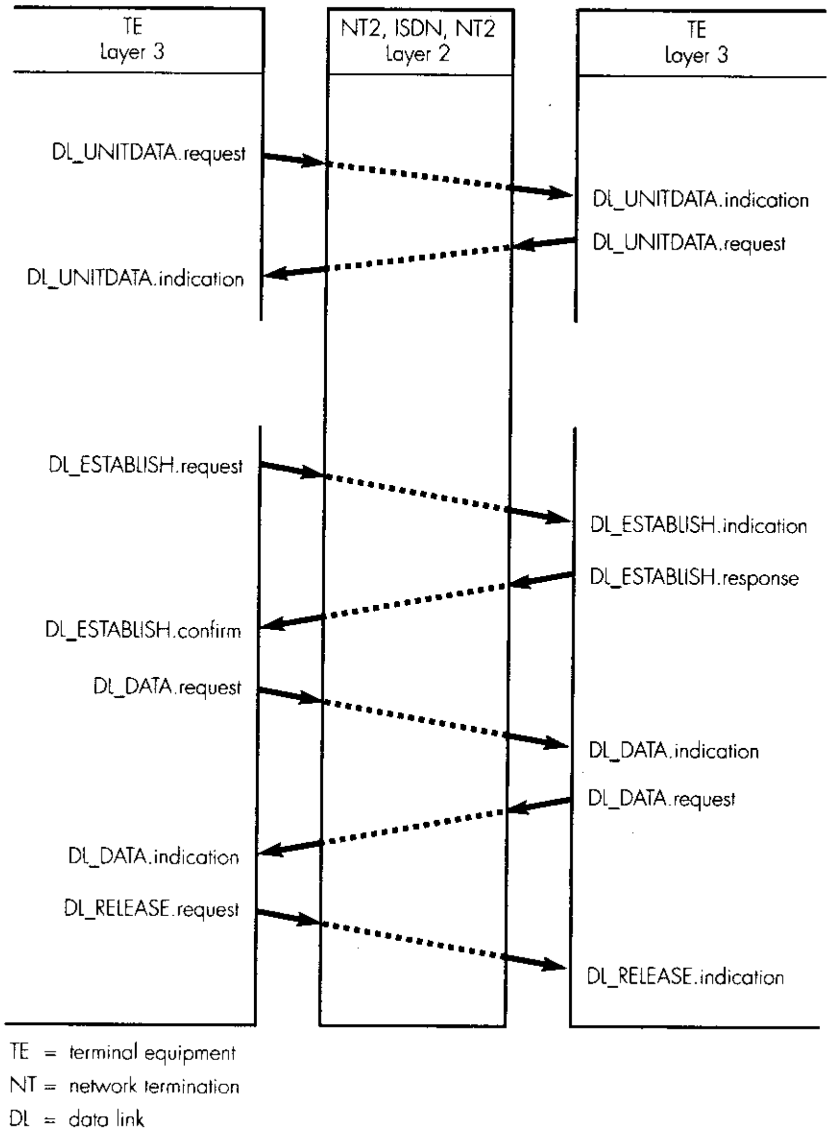


Figure 7.29 LAPD user service primitives: (a) connectionless; (b) connection-oriented.

the terminal relates – speech, video, and so on – and the TEI uniquely identifies the terminal within that class. There is also a broadcast address – all binary 1s – that allows a message to be sent to all terminals in a class. This can be used, for example, to allow all telephones to receive an incoming call setup request message.

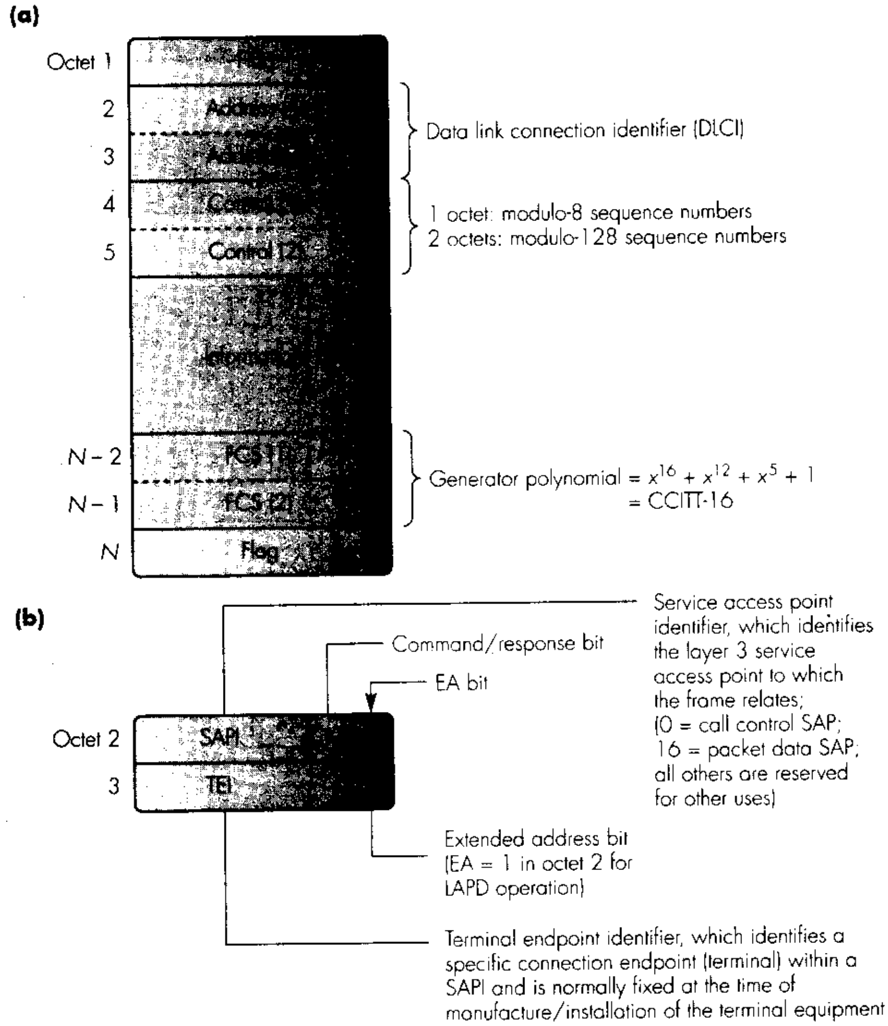


Figure 7.30 LAPD: (a) frame format; (b) address field usage.

The various control field formats – octets 4 and 5 – associated with LAPD are summarized in Figure 7.31, which also shows which frames can be sent as command frames and which as response frames.

In LAPD, as with LAPM, the additional unnumbered information (UI) frame is used. LAPD uses this with the connectionless service. Since there is no error control associated with this service (best-effort), all information is sent with a single control field with neither an N(S) nor an N(R). Such frames do have an FCS field however and, should this fail, the frame is simply discarded. Normally with this service the higher (user) layer must then detect the discarded frame – for example, by the lack of a suitable response (also in a UI-frame) – and make another attempt.

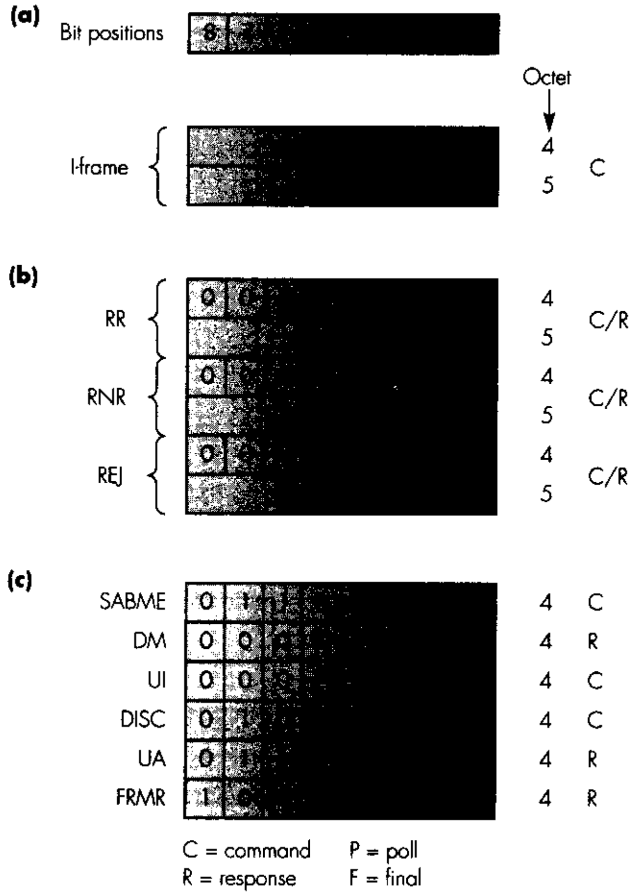


Figure 7.31 LAPD control field bit definitions: (a) information; (b) supervisory; (c) unnumbered.

Q.931 The Q.931 protocol is concerned with the sequence of the signaling messages (packets) that are exchanged over the D-channel to set up a call. An abbreviated list of the message types used is as follows:

- call establishment:
  - ALERTing
  - CALL PROCeeding
  - CONnect ACKnowledge
  - SETUP
  - Others;

- information transfer:
  - USER INFOrmation
  - Others;
- call clearing:
  - DISConnect
  - RELEASE
  - RELease COMPlete
  - Others.

Some of these messages have local significance (TE/LE) while others have end-to-end significance (TE/TE). However, all the messages are transferred across the interface in layer 2 (LAPD) I-frames. An example illustrating the use of some of these messages in setting up a conventional telephone call is shown in Figure 7.32(a).

During the setting up of a conventional telephone call over a PSTN, it is assumed that the called telephone operates in a standard way and hence the call setup phase involves only the setting up of a connection through the network. With an ISDN, however, since it was designed to support a range of services, it is necessary not only to set up a connection but also to establish an agreed set of operational parameters for the call between the two TEs. To do this, the call SETUP message, in addition to the address/number of the called TE – required to set up a network connection – also includes the proposed operational parameters for the call. The general format of all layer 3 messages is shown in Figure 7.32(b).

In addition to signaling messages, the ISDN D-channel can also be used for other purposes such as a low bit rate packet-switched facility. The *protocol discriminator* field is used to specify the protocol to which the message relates which, for ISDN signaling messages, is DSS1. The *call reference* field indicates the type of call involved and the *message type* the type of message the packet contains such as SETUP, CONNECT, and so on . The *message parameters* field contains other information relating to this message type such as the address/number of the called TE and the proposed operational parameters for the call.

PCM circuits

As we showed earlier in Figure 7.3 and explained in the accompanying text, in many instances remote concentrator units (RCUs) are used in the access network to terminate analog subscriber lines and relay the equivalent digital signals to the LE/EO. In practice, the circuit that links the RCU to the LE/EO is a PCM circuit similar to that used for the primary rate access to an ISDN. As we showed earlier in Figures. 7.15 and 7.16, there are two alternative PCM multiplexing structures in use, one comprising 24 channels and the other 32 channels. Also, in addition to each set of 24/32 time slots being multiplexed into a frame, a multiframe structure is established comprising 24/16 frames respectively.

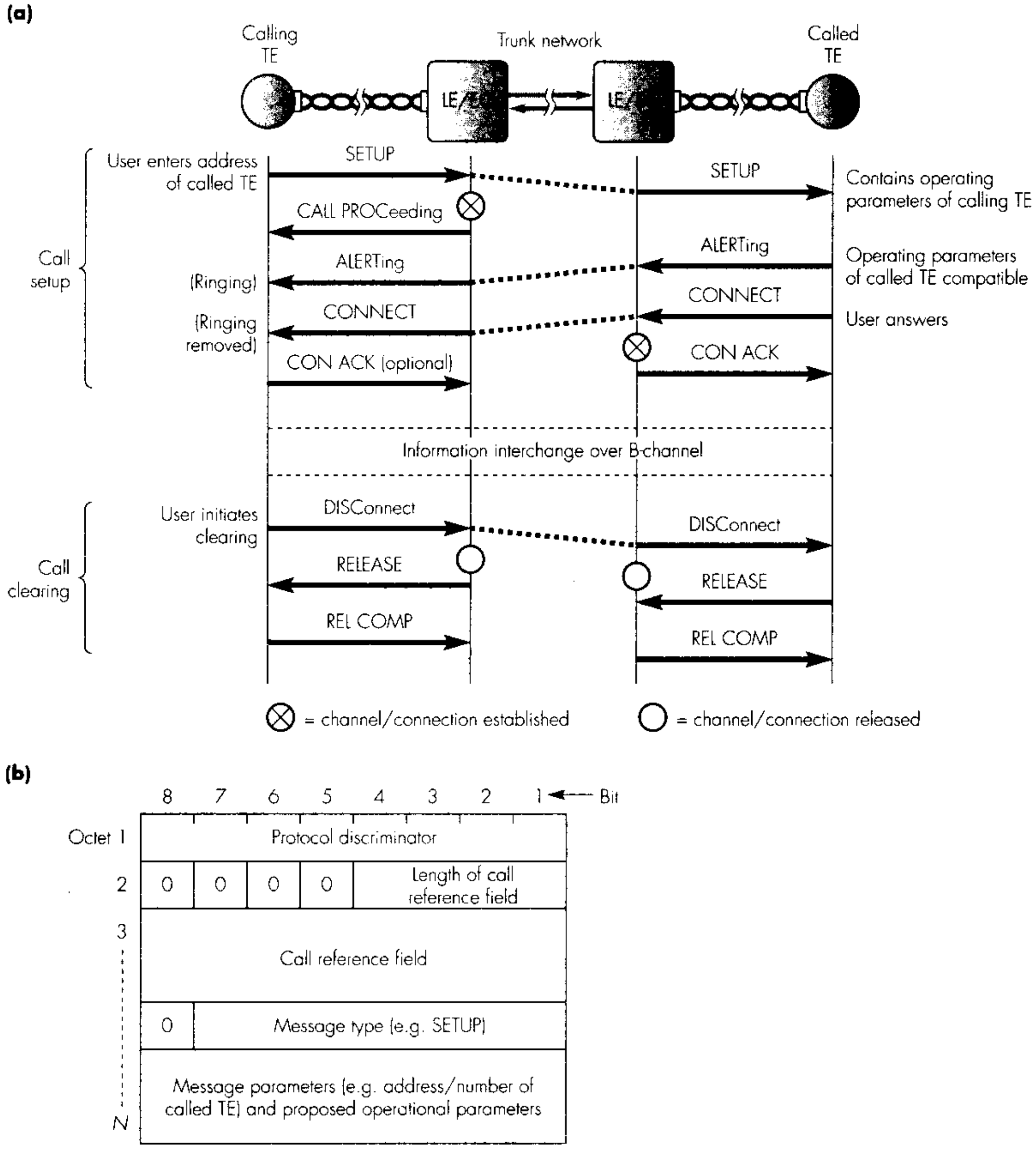


Figure 7.32 ISDN layer 3 signaling: (a) example message sequence to set up a conventional telephone call; (b) message format.

The RCU converts the analog speech signal relating to each active call into a 64 kbps PCM signal and uses a preassigned time slot in the PCM circuit to relay this to the LE/EO. Similarly, each PCM signal received from the preassigned time slot in the return direction is converted back into an analog signal by the RCU and relayed over the analog access circuit to the related subscriber TE. Hence as part of the call setup phase, on receipt of each new call request the RCU first determines the availability of a free time slot in the PCM circuit. Then, assuming one is available, the RCU proceeds to use selected bits in each multiframe associated with the time slot both to inform the LE/EO of the time slot number and also to relay the subsequent signaling information associated with the call.

The bits used for signaling in a 1.544 Mbps 24-channel PCM circuit are identified in Figure 7.33(a). As we can see, the least significant (eighth) bit from each time slot in frames 6 and 12 of each multiframe is used for the transfer of the signaling information associated with the corresponding time slot/channel. Therefore each time slot in every sixth frame contains seven information bits rather than eight. Although this is acceptable for speech, for data and other multimedia applications it is normal to operate all the channels at 56 kbps with this type of circuit.

The bits used in a 2.048 Mbps 32-channel PCM circuit are identified in part (b) of the figure. As we can see, with this type of circuit, except for frame 0, time slot 16 in each of the remaining 15 frames in a multiframe is used for the transfer of the signaling information associated with each of the 30 time slots/channels that are used for application information. As the names imply, this type of signaling is known as **channel associated signaling** (**CAS**).

Example 7.4

Assuming channel associated signaling is being used, derive the bit rate of the signaling channel associated with each of the time slots in (i) a 24-channel PCM circuit and (ii) a 32-channel circuit.

Answer:

(i) As we can deduce from Figure 7.33(a), for each time slot, 1 bit in every sixth frame is used for signaling. Hence, since each frame is of 125 μs duration, we have 1 bit every $6 \times 125 = 750\,\mu s$.
Thus the signaling rate per channel = $1/750 \times 10^{6} = 1.333$ kbps.

(ii) As we can deduce from Figure 7.33(b), for each time slot, 4 bits every 16 frames are used for signaling. Hence we have 4 bits every $16 \times 125 = 2000\,\mu s$.
Thus the signaling rate per channel = $4/2 \times 10^{-3} = 3$ kbps.

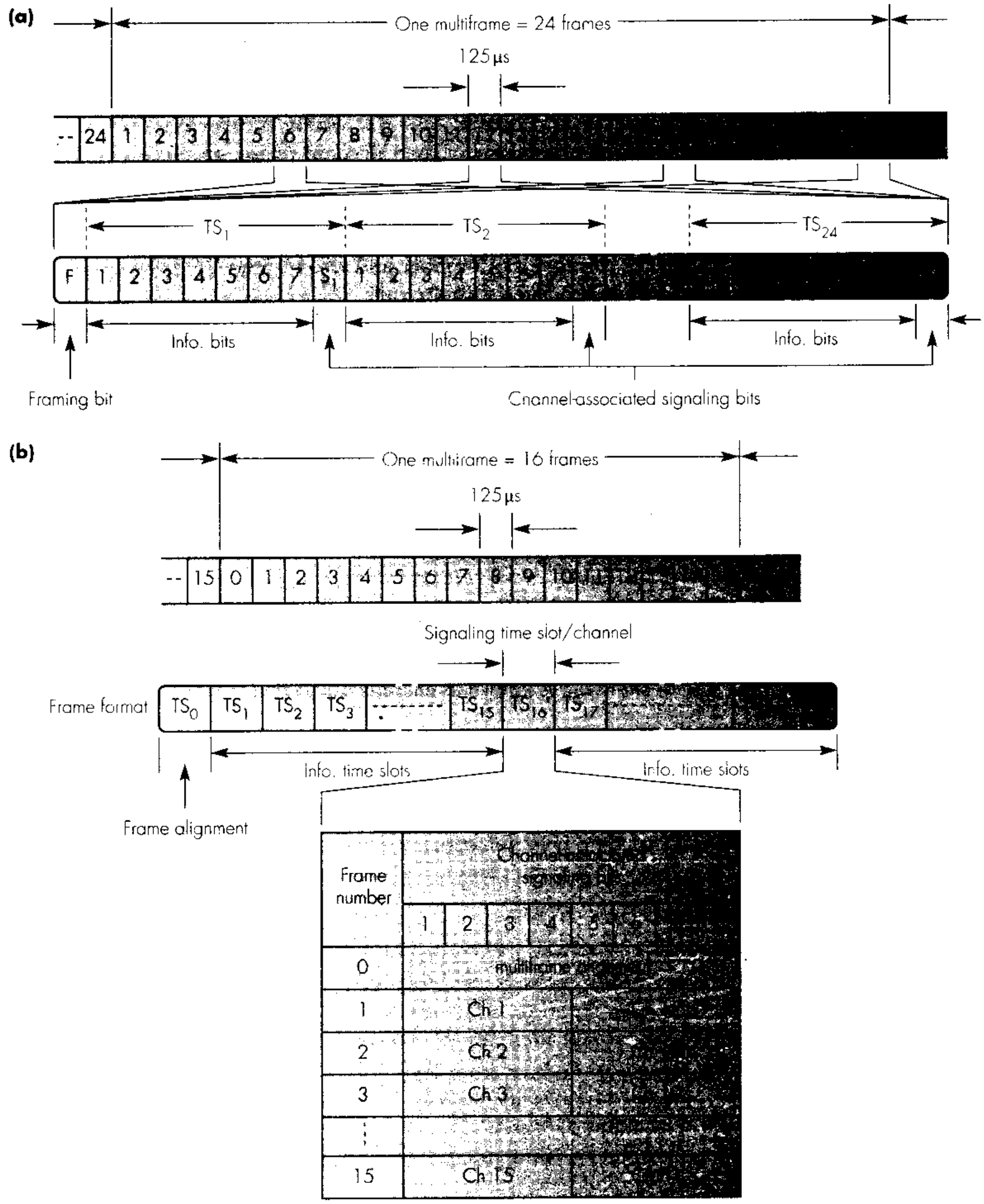


Figure 7.33 Signaling in PCM circuits: (a) 24-channel system; (b) 32-channel system.

An example set of signals/signaling messages associated with the setting up and clearing of a call/connection through an RCU is shown in Figure 7.34. The example relates to a conventional speech-only call and hence can be compared with the example we described earlier in Figure 7.26(b). It is assumed that the signaling messages are all transferred using the signaling channel associated with the assigned PCM time slot. Also, although not shown, the digitized speech signals associated with the call are transferred over the assigned PCM time slot. A similar set of signals/signaling messages is used if the intermediate device is a PBX or an RSU rather than an RCU. In the case of a PBX and RSU, however, local calls are processed directly and the procedure shown in Figure 7.34 is only invoked for calls external to their local switching domain.

7.4.2 Trunk network signaling

In early networks, channel associated signaling over PCM circuits was also used within the core trunk network. As the range of services supported by the network increased, so a more flexible form of signaling was introduced. For example, with a basic telephony service, a standard phone number is used which includes a country (if required), area, and local part. Then, when all calls use only these numbers, each exchange in the switching hierarchy can readily select one of a number of preallocated routes through the switching network using the various parts of the dialed number/address. With the advent of services based on non-standard numbers such as free-phone and local-charge, however, the number dialed does not contain these same parts. Hence before such calls/connections can be set up, the LE/EO must first obtain the standard number giving the location where the related service is being provided. However, since these types of number are introduced and changed quite frequently, it is not feasible for every LE/EO to have this information. Normally, therefore, this type of information is held only at a small number of locations within the network. On receipt of a call request involving, say, a free-phone number, prior to setting up the connection, the LE/EO sends an appropriate address-resolution signaling message to one of these locations and this responds with the standard number where the service is being provided.

As we can deduce from this, although the use of a small number of locations for this type of information means its management is made much easier, as a consequence, a faster and more flexible way of transferring signaling information is required. The solution adopted is to provide a separate network for the transmission and routing of signaling messages from that used for the actual call information interchange. This (signaling) network is then used to route and transfer all the signaling messages relating to all calls. This mode of working is known as **common channel signaling** (**CCS**) and the protocol stack that is associated with the signaling network, the **common channel signaling system number 7** or simply **SS7**. We shall restrict our discussion of network signaling to descriptions of the basic features of SS7.

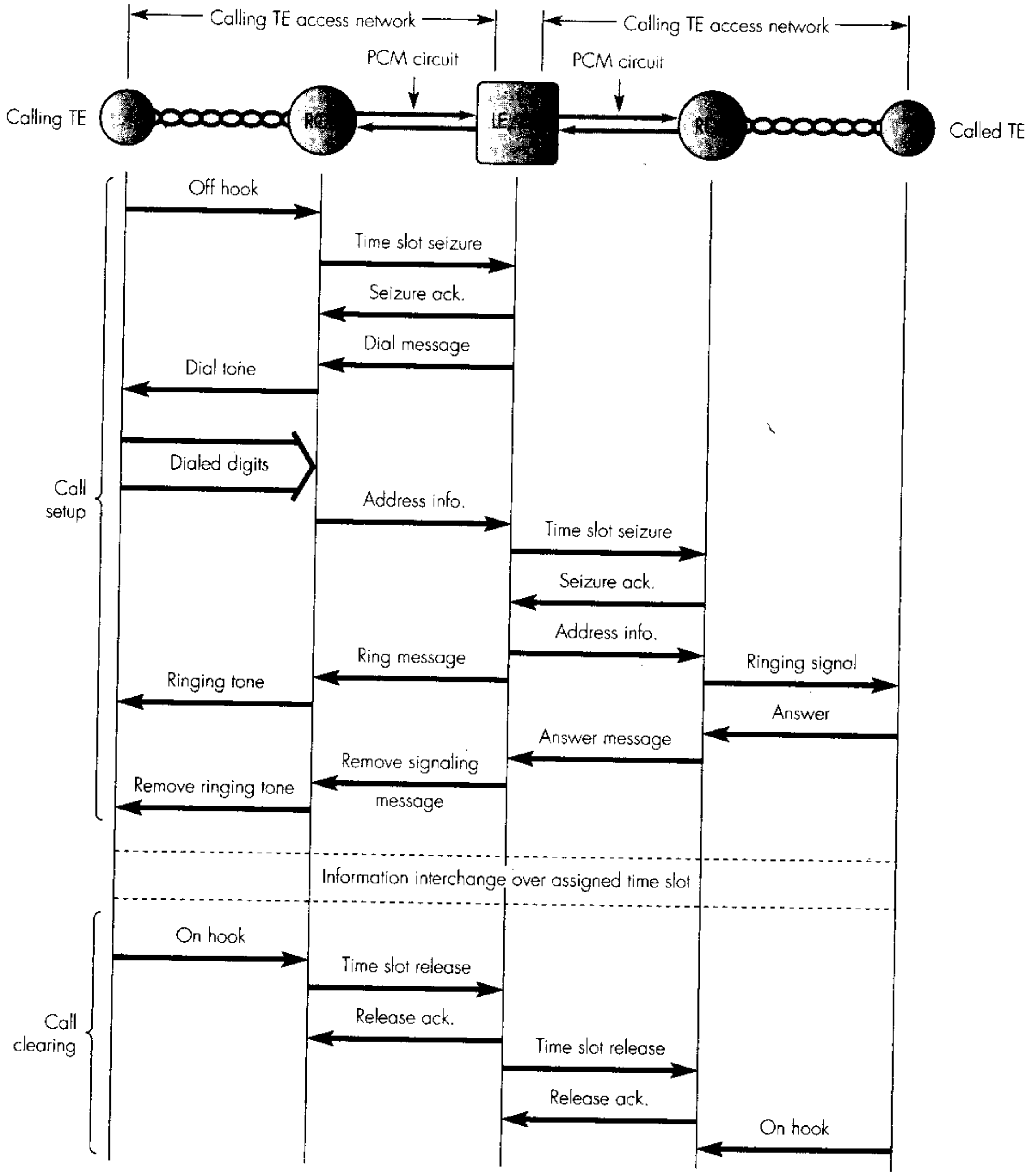


Figure 7.34 Analog access signaling through an RCU and PCM circuit.

SS7: components and terminology

Figure 7.35 shows a selection of the components and terminology associated with SS7 and, as we show in the figure, within each switching exchange/office is a collection of application protocols known as *parts* or *application service elements.* These include the **user part (UP)**, the **address resolution part (ARP)**, and the **operations, maintenance, and administration part (OMAP)**. Each performs a separate function by exchanging signaling messages with a peer part in another system using the services provided by the **message transfer part (MTP)**.

Each LE/EO contains one or more user parts, an ARP, and an OMAP. The role of the various user parts is, given a standard number/address, to set up a transmission path through the switching unit to the destination indicated in the number using one of the alternative paths/lines available. Examples of user parts include:

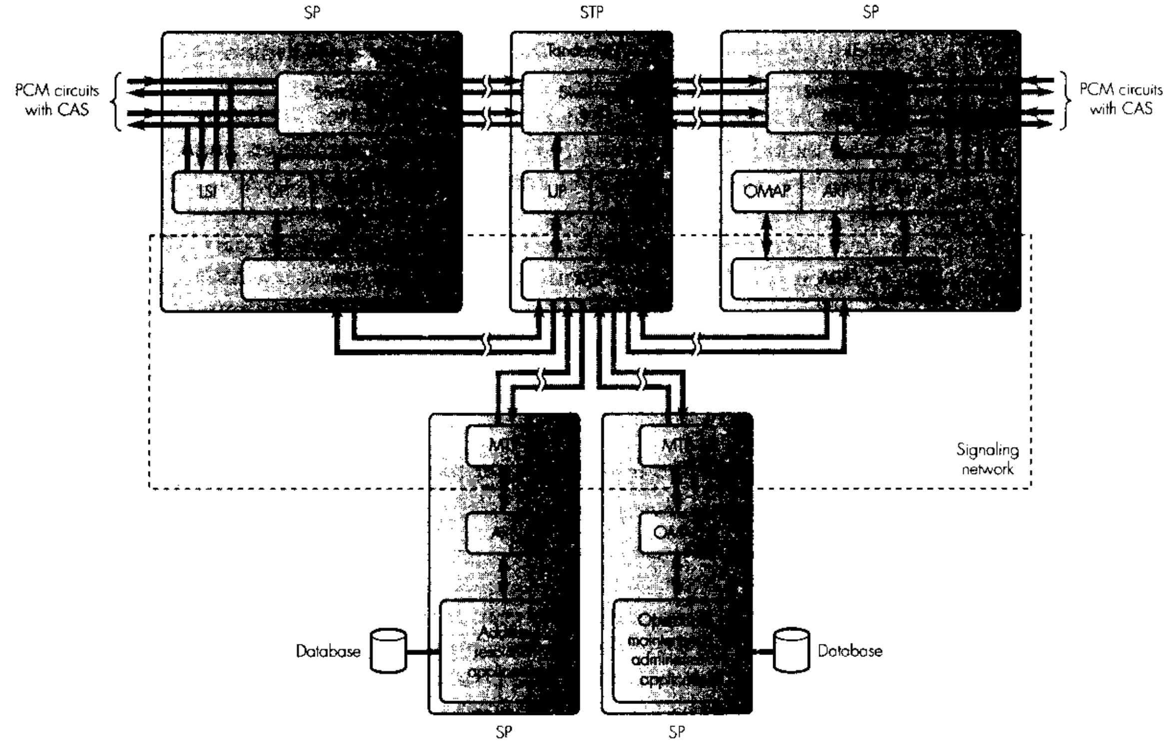
- TUP: the telephone (PSTN) user part,
- DUP: the data user part,
- ISUP: the ISDN user part.

As the name implies, the role of the ARP is, given a non-standard number/address (such as a free-phone number), to obtain the standard number where the related service is being provided.

The same signaling network is also used for the transfer of operational, maintenance, and administrative information relating to the total network. The OMAP in each exchange, therefore, is responsible for reporting fault conditions, receiving operational information, and so on to and from the nearest operations and management node.

In order to gain an understanding of the interactions between the various UPs and the ARP in each LE/EO, we shall consider the setting up of a connection using both a standard and a non-standard number. In the case of a standard number, the called number/address received from the subscriber TE – using CAS for example – is passed to the appropriate UP by the **local signaling interface (LSI)**. The UP then proceeds to reserve a time slot/channel in a line to the destination LE/EO – using the international (if appropriate) and area parts of the number and the contents of a routing table held by each exchange – and relays the number in a signaling message to the next exchange along the selected path using the services provided by the message transfer part (MTP). This procedure then repeats until the signaling message is received by the UP in the destination LE/EO. Then, assuming the called TE is free and is prepared to accept the call, the UP returns an accept message over the signaling network and the information interchange then proceeds.

In the case of a call involving a non-standard number, prior to setting up a connection through the switching network, the UP passes the entered number to the ARP and this proceeds to send an address-resolution request (signaling) message to one of the higher-level tandem exchanges/offices in



ARP = address resolution part
UP = user part
OMAP = operations, maintenance, and administration part
LSI = local signaling interface
MTP = message transfer part
SP = signaling point
STP = signaling transfer point

Figure 7.35 SS7 components and terminology.

the switching hierarchy using the MTP and the signaling network. Routing information relating to the ARP (and OMAP) signaling messages is held only in the higher-level exchanges. In the header of the messages relating to these parts is the identity of the required service, ARP, or OMAP, and also that of the initiating exchange. On receipt of such messages, the MTP in the higher-level tandem exchanges uses this to route the message to the network node providing this service using a previously assigned route. The address resolution application in this node then obtains the standard number of where the related service is being provided from a database and returns this in a message to the ARP in the LE/EO that originated the request, again using the signaling network. On receipt of the response message, the ARP in the LE/EO passes the standard number to the UP that initiated the request and the UP then proceeds to set up a network connection as previously described.

As we show in Figure 7.35, all LE/EOs and service nodes are known as **signaling points (SPs)** as they simply initiate and receive signaling messages.

The higher-level tandem exchanges/offices that can also route signaling messages are known as signaling transfer points (STPs). Hence, since only STPs can route signaling messages, the management of the related routing tables is much simpler. Also, although the signaling links are shown separate from the links that are used for the information relating to calls, normally, both share the same physical lines that interconnect the various exchanges shown earlier in Figure 7.1.

SS7: protocol architecture

A selection of the protocols associated with the SS7 protocol stack is shown in Figure 7.36(a). As we can see, the lowest three protocol layers make up the MTP and collectively they provide a basic reliable message transfer service through the signaling network. The function of each protocol is:

- **signaling data link**: this is the physical layer interface to the transmission channel(s) being used;
- **signaling link**: this is a connection-oriented (reliable) data link control protocol based on HDLC that incorporates error and flow control;
- **signaling network**: this is a connectionless protocol that provides message routing across the signaling network.

The format of the protocol data units associated with the signaling link and signaling network protocols is shown in Figure 7.36(b).

As we have just indicated, the signaling link protocol is based on HDLC and the *forward* and *backward sequence number* fields in the header of each I-frame are used for error and flow control purposes using a go-back-N error control scheme and a window flow control method. The *length indicator* is used to indicate the number of octets/bytes in the information field and the *frame check sequence* field is used to detect the presence of transmission errors.

The *service information octet* (*SIO*) and *signaling information field* (*SIF*) in the protocol control field of the signaling network protocol enable the application data to be routed through the signaling network to the destination node and, once there, to the intended application part. The SIO consists of two fields:

- *service indicator*: this specifies the application part to which the data in the message relates – TUP, DUP, ISUP, ARP, and so on;
- *subservice field*: this specifies the type of network (national/international) over which the message unit is traversing.

The SIF consists of three subfields:

- the *destination point code* is used by the signal network protocol in STPs to route the message unit to its intended destination exchange/node;

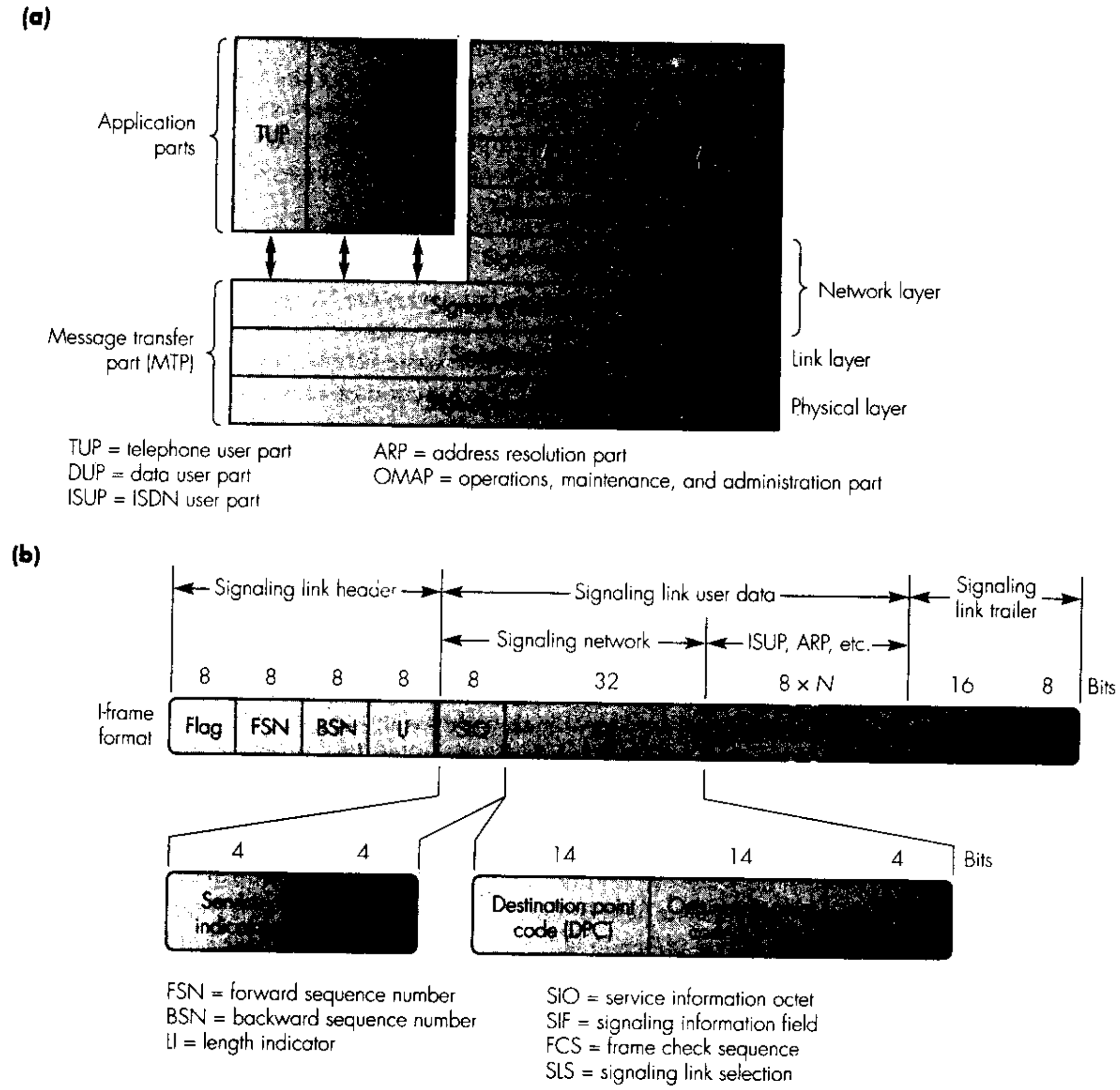


Figure 7.36 SS7 protocol architecture: (a) protocol components; (b) format of MTP message units.

- the *originating point code* is used to route the reply message to the originating exchange/node;
- the *signaling link selection* field is used to indicate the link being used when a choice is available, the aim being to share the signaling traffic over those available.

Example application part message sequence

In order to describe a typical application part message sequence, a selection of the messages involved in the setting up and clearing of a channel is shown in Figure 7.37. The messages used in the figure relate to the ISDN user part and include:

- *Initial address message* (*IAM*): this is sent by the ISUP in the calling LE/EO to initiate the setting up of a call in the forward direction. Hence it contains the ISDN number/address of the calling and called party within it and also other information;

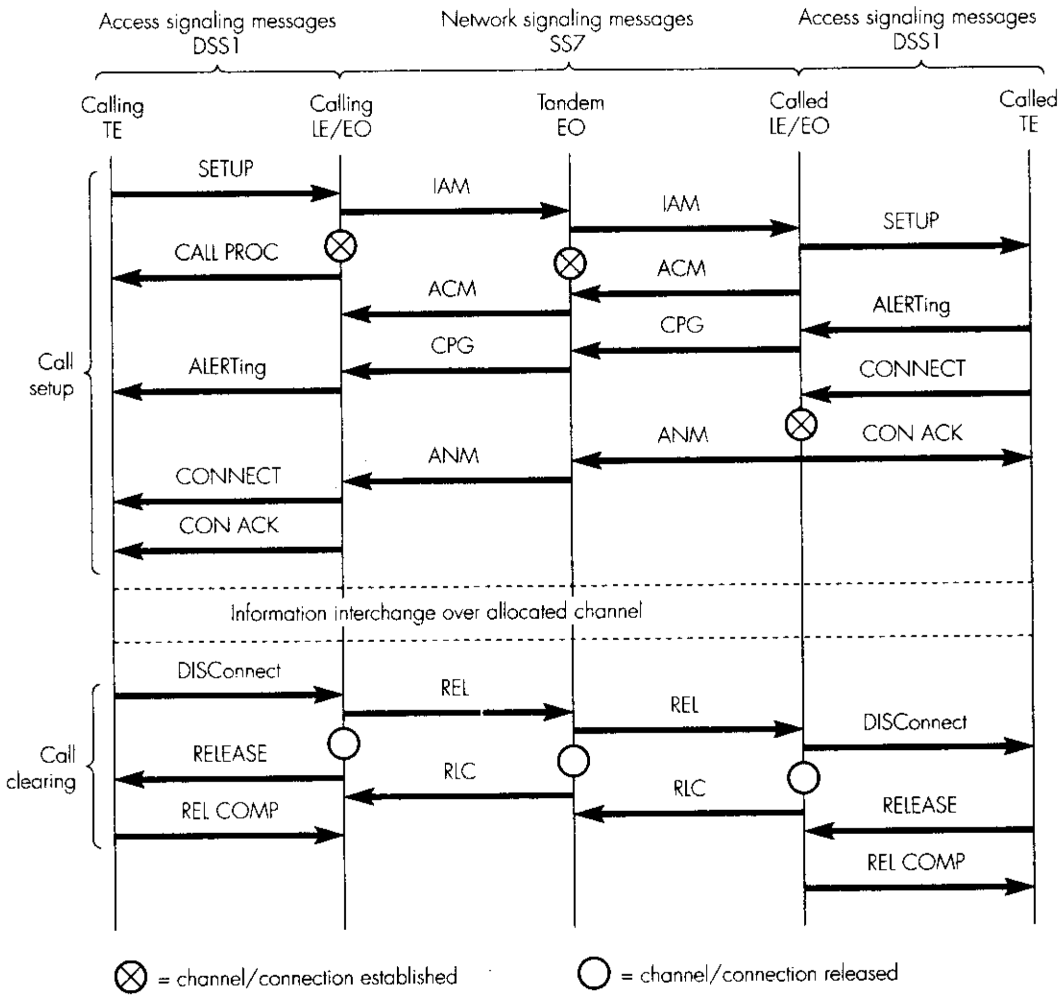


Figure 7.37 Network signaling message sequence to set up an ISDN channel/connection using SS7.

- *Address complete message* (*ACM*): this is sent by the ISUP in the called LE/EO to indicate to the calling ISUP that all the address information required for routing (and hence establishing) the call to the called party has been received;
- *Call progress* (*CPG*): this is sent by the called ISUP to indicate to the calling ISUP that an event has occurred that should be relayed to the calling TE;
- *Answer message* (*ANM*): this is sent by the called ISUP to the calling ISUP to indicate the call has been answered and that charging should commence;
- *Release* (*REL*): this can be sent in either direction to release the connection and to inform the other ISUP that this has been done;
- *Release complete* (*RLC*): this is sent in response to the release message.

The example builds on the ISDN D-channel message interchange sequence associated with the DSS1 access protocol that we showed earlier in Figure 7.32(a).

Summary

In this chapter we have described the basic operational characteristics of public circuit-switched networks. We explained how the networks consist of three interrelated systems: transmission, switching, and signaling. Also, how the basic core switching network associated with these networks supports the services offered by both a PSTN and an ISDN and that the only difference between the two networks is the way subscribers gain access to the core network.

A summary of the topics discussed in relation to the two types of access network is given in Figure 7.38(a). As we can see, in the case of a PSTN, these utilize analog transmission within the access network to link the TEs of subscribers to the nearest network termination point and we described the operation of both the signaling system used and the operational characteristics of low bit rate modems. In the case of an ISDN, these utilize digital transmission and we described both the transmission and signaling schemes used with the basic rate interface (2B + D) and the two types of primary rate interface (23B + D and 30B + D).

The core transmission and switching network operates digitally and a summary of the topics we discussed is given in Figure 7.38(b). In terms of transmission, the interexchange lines utilize a mix of the older PDH and newer SDH/SONET multiplexing hierarchies to multiplex several thousands of channels/calls over these lines and we explained the operational characteristics of both systems. In terms of switching, we explained the basic principles of time switching and space switching and how these are combined to form a digital switching unit.

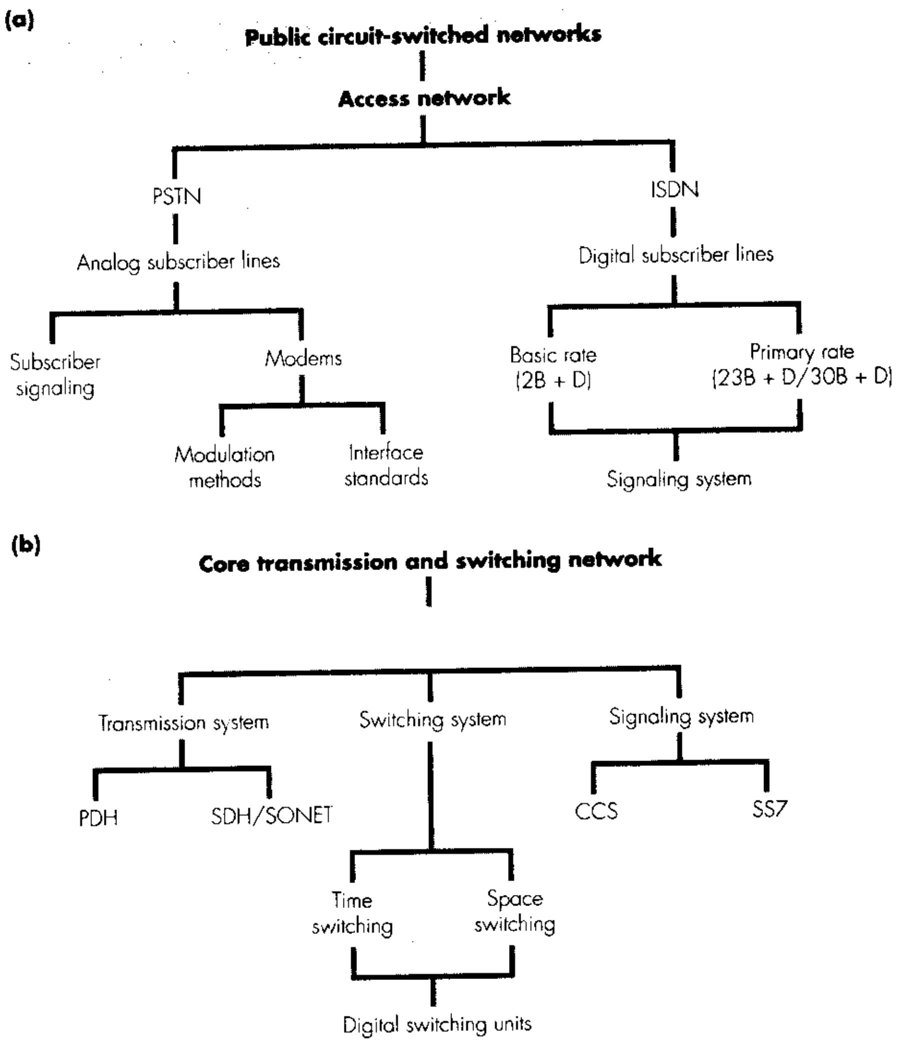


Figure 7.38 Public circuit-switched network summary: (a) access network; (b) core transmission and switching network.

The signaling system in the core network utilizes a separate network to transfer all the signaling messages between the trunk switching exchanges and the various supervisory nodes used for such functions as address resolution and network management. The network is known as the signaling network and this mode of operation common channel signaling (CCS). The protocol stack used in each exchange/node to control the transfer of signaling messages over the signaling network is known as SS7, the basic principles of which we also described.

Exercises

Section 7.1

7.1 With the aid of Figure 7.1, describe the role of the following:
(i) the local access and signaling networks,
(ii) the interexchange trunk/carrier networks,
(iii) international network.

7.2 Explain the meaning of the following terms relating to the total transmission system:
(i) analog subscriber line,
(ii) digital subscriber line,
(iii) low bit rate modems,
(iv) high bit rate access lines,
(v) interexchange trunk lines.

7.3 With the aid of examples, identify the switching exchanges involved in providing the following types of call between two subscribers connected to:
(i) the same LE/EO,
(ii) different LE/EOs within the same region,
(iii) different regions,
(iv) different countries.
Why are alternative lines/paths present?

7.4 State the role of the signaling system
(i) over the access lines,
(ii) over the interexchange trunk lines.

Section 7.2

7.5 Explain the meaning of the following terms relating to the analog access network of a PSTN:
(i) junction box,
(ii) cross-connect,
(iii) maximum cable length.

7.6 With the aid of the diagrams in Figure 7.2, explain the roles of the following:
(i) cradle switch unit,
(ii) ringer circuit,
(iii) dialer,
(iv) 4-wire to 2-wire hybrid,
(v) dual-tone multifrequency keypad,
(vi) echo signal,
(vii) echo canceler.

7.7 With the aid of Figure 7.3, explain the role of a remote concentrator unit in the access network of a PSTN. Include in your explanation the meaning of the terms "concentration" and "pair-gain". How is a remote switching unit different from a remote concentrator unit?

7.8 With the aid of Figure 7.4, explain the following modem operating modes:
(i) 2-wire switched connections,
(ii) 4-wire leased circuits.
What is the advantage of the latter and when is it used?

7.9 With the aid of the schematic diagram and waveform sets shown in Figure 7.5, explain the principle of operation of the three basic types of modulation used in modems. Include the role of the carrier signal and the difference between phase-coherent PSK and differential PSK.

7.10 With the aid of the diagrams shown in Figure 7.6, explain the operation of the following multilevel modulation schemes:
(i) single-carrier 4-PSK,
(ii) two-carrier 4-QAM,
(iii) two-carrier 16-QAM.

7.11 Assuming an input bit rate of 56 kbps, derive the line signaling rate in baud for each of the three modulation schemes identified in Exercise 7.9.

7.12 Explain the meaning of the term "training phase" and how the bit rate for a connection is established.

7.13 Identify the subset of lines from the V.24/EIA-232D interface that are used to carry out the setting up of a connection through the PSTN and the exchange of some data. With the aid of the time sequence diagram in Figure 7.9, explain how these are used.

7.14 With the aid of the two modem connections shown in Figure 7.10(a), explain how
(i) a local loopback test and
(ii) a remote loopback test is carried out.

7.15 Explain why line driver and receiver circuits must be used to connect a modem to the serial port of a computer. Describe the operation of such circuits.

7.16 With the aid of the two diagrams shown in Figure 7.11, explain the operation of the following ISDN network terminal alternatives:
(i) a 4-port NTU,
(ii) an S-bus NTU.
Include in your explanation the meaning of the term "out-of-band" and how the use of the two B-channels is shared between the attached terminal equipments.

7.17 With the aid of the schematic diagrams shown in Figure 7.12, describe the operation of a differential line driver and receiver circuit and why these are used. Include the meaning of the term "common mode rejection" and "characteristic impedance".

7.18 In relation to the line signals shown in Figure 7.13, explain how
(i) the DC balancing bits associated with each transmitted frame ensure the mean DC level of the line is always zero and
(ii) contentions for use of the shared signaling channel are resolved.

7.19 With the aid of Figure 7.14(a), explain how the NTU supports duplex transmission over a single twisted-pair. Include in your explanation the operation of the adaptive echo canceler.

7.20 Explain why a 4-level code is used over the twisted-pair access line from an NTU. Also explain how the code works.

7.21 With the aid of the frame structure shown in Figure 7.14(c), explain the framing structure used over the access line. Include the use of the synchronization word.

7.22 With the aid of the waveform set and frame format details relating to the 1.544 Mbps primary rate interface shown in Figure 7.15, explain
(i) the AMI line code and why this is supplemented with the B8ZS code,
(ii) the frame structure used and why a superframe is defined for use with this.

7.23 With the aid of the waveform set and frame format details relating to the 2.048 Mbps primary rate interface shown in Figure 7.16, explain
(i) the operation of the HDB3 line code,
(ii) the frame structure, including how the start of each frame is determined.

7.24 Explain the meaning of the following terms relating to the plesichronous digital hierarchies:
(i) justification bits,
(ii) primary multiplex,
(iii) DS1/T1 and E1,
(iv) byte interleaving,
(v) bit interleaving.

7.25 With the aid of the schematic diagrams shown in Figure 7.18, explain how a portion of a higher bit rate stream in a PDH can be derived and used using an add-drop-multiplexer. Identify the disadvantages of this approach.

7.26 In relation to the SDH/SONET, explain the meaning of the terms:
(i) container,
(ii) virtual container,
(iii) path,
(iv) line,
(v) section,
(vi) fixed stuff,
(vii) concatenation.

7.27 With the aid of Figure 7.21, explain how a lower bit rate portion of a higher bit rate stream is derived using a SDH/SONET add-drop multiplexer.

Section 7.3

7.28 Discriminate between the following terms relating to switching systems: "space switching" and "time switching".

7.29 With the aid of Figure 7.22(a), explain the role of:
(i) the frame store,
(ii) the modulo-N/address counter,
(iii) the connection store.

What is the maximum delay for a 24-channel system?

7.30 With the aid of the space switch diagrams shown in Figure 7.23, explain why the connection store requires four sets of entries. What would be the entry in the connection store if a time slot in one of the input lines was not in use?

7.31 Repeat Example 7.3 but this time assume an 8-input, 8-output space switch.

7.32 With the aid of the time-space-time switch shown in Figure 7.24(b), assume that time slot 5 in input line 1 is to be switched to time slot 10 in output line 2. Assuming an intermediate time slot is available, derive the contents of each connection store (CS) to do this.

7.33 Explain how duplex connections are set up through the T-S-T switch shown in Figure 7.24(b). State the number of intermediate time slots that should be provided with this switch to ensure non-blocking.

Section 7.4

7.34 With the aid of a diagram, discriminate between access networking signaling and core network signaling.

7.35 In relation to the diagrams shown in Figure 7.26 relating to analog access signaling:
(i) state the use of the congestion tone signal,
(ii) state how the exchange determines a wrongly keyed number and its response to this.

7.36 With the aid of the time sequence diagram shown in Figure 7.27(b), explain how
(i) error correction and
(ii) flow control are achieved during the information transfer phase.

7.37 With the aid of the signaling protocol set identified in Figure 7.28, explain the role of:
(i) LAPD (Q.921),
(ii) Q.931.

7.38 In relation to the LAPD/Q921 protocol, state the roles of the service access point identifier and terminal endpoint identifier in the frame header. Explain the uses of the various control fields.

7.39 In relation to the Q.931 protocol:
(i) give examples of the information present in the SETUP message,
(ii) explain the role of the protocol discriminator field in the message header.

7.40 Explain the meaning of the term "channel associated signaling" and give an example of its use.

7.41 With the aid of the diagrams shown in Figure 7.33, explain how the signaling messages relating to a specific time slot are assembled in
(i) a 24-channel system,
(ii) a 32-channel system.

State the bit rate of the signaling channel in each case.

7.42 In relation to the signaling messages shown in Figure 7.34, state the affect of the RCU being a PBX or RSU.

7.43 Explain the meaning of the term "common channel signaling" and how it is different from channel associated signaling.

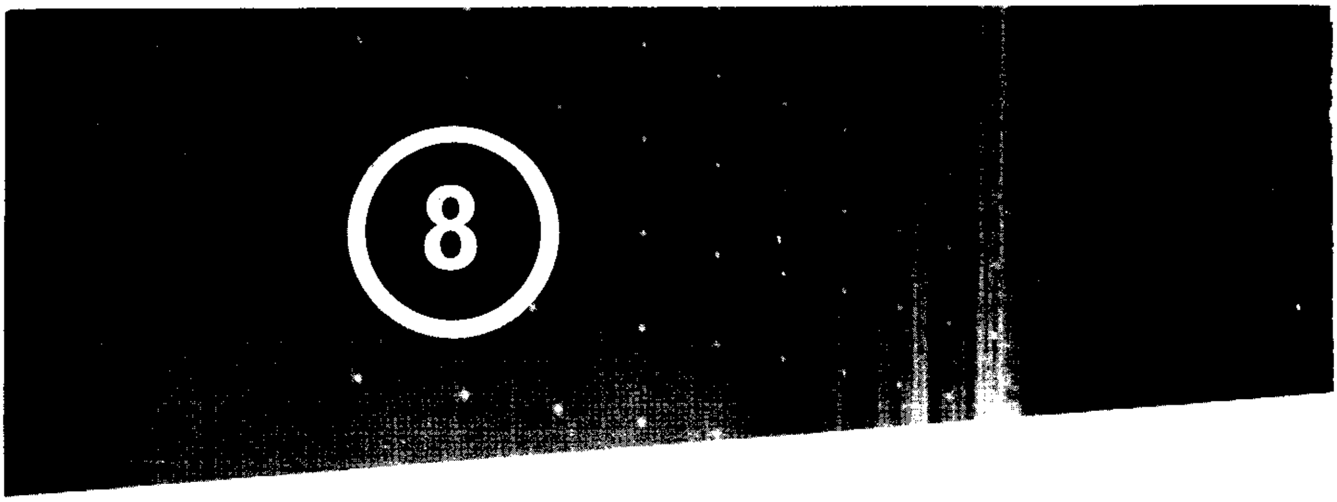
7.44 With the aid of Figure 7.35, explain the meaning/use of the following SS7 components ad terminology:
(i) UP,
(ii) ARP,
(iii) OMAP,
(iv) LSI,
(v) MTP,
(vi) SP,
(vii) STP.

7.45 With the aid of Figure 7.35, explain how a connection is established using
(i) a standard, and
(ii) a non-standard number.

7.46 With the aid of Figure 7.36(a), explain the role of the following protocols:
(i) signaling data link,
(ii) signaling link,
(iii) signaling network.

Hence, with the aid of the message format shown in Figure 7.36(b), describe the role of the following fields:
(i) FSN and BSN,
(ii) SID and SIF.

7.47 With the aid of Figures 7.26 and 7.37, produce a diagram that shows the exchange of the SS7 network signaling messages to set up a standard telephone channel/connection initiated using analog access signaling.



Enterprise networks

8.1 Introduction

When a person is at home, all calls relating to interpersonal and interactive applications must be made using a PSTN, an ISDN, or a cable distribution network. The calls are then charged at a rate determined by the call duration and the distance involved. In the case of a person in a business or large enterprise, however, the majority of the calls made are to other members of the same business/enterprise and only a small percentage are to people outside of the enterprise. Hence for all but the smallest businesses, in order to reduce call charges, most enterprises install their own private networks to handle those calls that are internal to the enterprise. Normally, the network comprises both a private branch exchange (PBX) and a local area network (LAN) and, collectively, these support all the interpersonal and interactive communications within the enterprise.

For an enterprise that occupies just a single site/establishment – for example, a small to medium-sized company, a hospital complex, a university campus – the PBX and LAN at the establishment handle all internal calls and only those calls that are external to the site are made using an appropriate public network such as a PSTN, an ISDN, or an Internet service provider