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INTRODUCTION TO TELECOMMUNICATIONS

There are two types of communication networks: *circuit-switched* networks and *packet-switched* networks. In circuit-switched networks a dedicated physical (digital or analog) circuit between the calling and called party is set up at the start of a call and released when the call has ended. Traditional telephone networks are circuit-switched networks and collectively form the switched-circuit network (SCN). Today these networks are used for speech and other purposes, such as facsimile, and are usually referred to as *telecommunication* networks.

Initially, all communication networks were circuit-switched networks. *Data networks*, consisting of a number of nodes connected by digital links, made their appearance around 1970. In these networks, a call (or *session*) consists of a series of short data bursts (packets) followed by relatively long silent intervals. A physical circuit therefore does not have to be dedicated to a single data call but can be shared by several simultaneous calls. The Internet is an example of a data network.

The terms "telecommunication network" and "data network" usually imply circuit-mode and packet-mode, respectively. However, advances in packet technology are making possible voice communication in data networks, in what is called *convergence* of voice and data. The long-term trend is toward packet communication for voice, video, and data, so the word "telecommunication" is also used sometimes to denote converged networks.

This book is about signaling in communication networks. The first nineteen chapters are dedicated to signaling in telecommunication networks, with "telecommunication" used in the traditional sense. The last three chapters are dedicated to signaling in packet networks, with focus on the convergence of voice and data.

To understand signaling it is necessary to be familiar with some basic telecommunication concepts and terms. This chapter presents an overview of telecommunication

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networks (in the SCN sense). It is intended as an introduction and sets the stage for later chapters.

1.1 TELECOMMUNICATION NETWORKS

1.1.1 Introduction

Figure 1.1-1 shows a small part of a telecommunication network. It consists of *exchanges, trunks*, and *subscriber lines*. Trunks are circuits between exchanges, and the group of trunks between a pair of exchanges is known as a *trunk group* (TG). Subscriber lines (SLs) are circuits between a subscriber S and the *local* exchange (A, B, C). Exchanges D and E do not have subscriber lines and are known as *intermediate*, *tandem*, *toll*, or *transit* exchanges.

Calls. A call requires a communication circuit (connection) between two subscribers. Figure 1.1-2 shows a number of connections in the network of Fig. 1.1-1 that involve subscriber S_p. In Fig. 1.1-2(*a*), S_p is on a call with S_q who is attached to the same exchange. Calls of this type are known as *intraexchange* calls. The circuit for the call consists of the subscriber lines SL_p and SL_q and a temporary path in exchange A. Cases (*b*) and (*c*) are calls between S_p and subscribers attached to other local exchanges (*interexchange* calls). The circuit in case (*b*) consists of

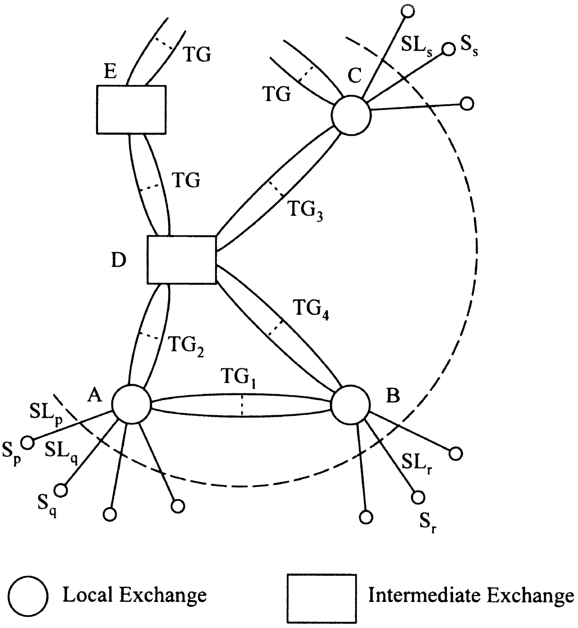

Figure 1.1-1. Partial view of a telecommunication network.

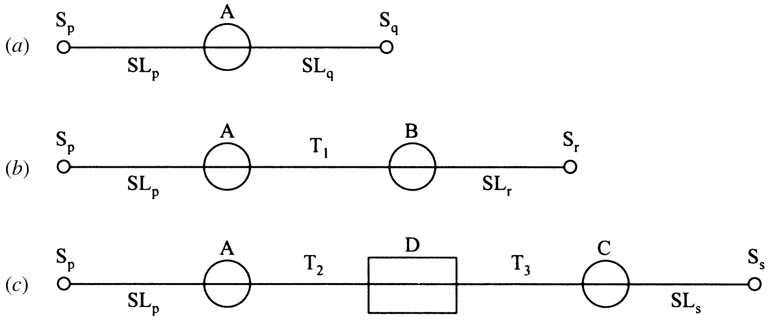

Figure 1.1-2. Connections involving subscriber S_p.

SL_p, a temporary path across exchange A, trunk T_1, a temporary path across exchange B, and SL_r. The connections of Fig. 1.1-2 are set up (switched "on") at the start of a call and released (switched "off") when the call ends.

Setup and Release. The setup and release of connections in telecommunication networks are triggered by *signals*. Starting and ending a call involve signaling between the subscribers and their local exchanges and, for interexchange calls, signaling between the exchanges along the connection.

Figure 1.1-3 shows the signaling for the setup of the connection of Fig 1.1-2(*b*). Subscriber S_p sends a request-for-service signal to exchange A (by lifting the handset of a telephone) and then signals the digits of the telephone number of S_r (with the dial or keyset of the telephone).

From the received number, exchange A determines that S_r is served by exchange B, and that the call is to be routed out on a trunk in group TG_1 (Fig. 1.1-1). It then searches for an idle trunk in this group and finds trunk T_1. Exchange A now seizes the trunk and sends a seizure signal, followed by signals that represent digits of the called number, to exchange B. It then sets up a path between SL_p and T_1.

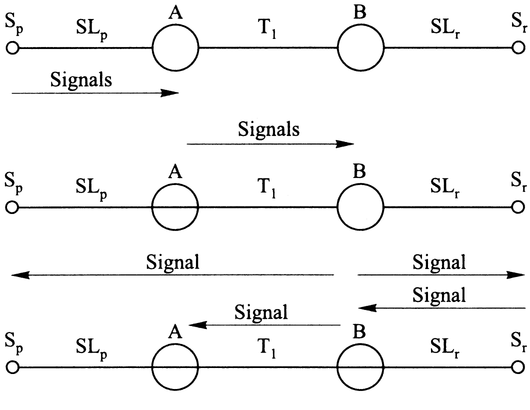

Figure 1.1-3. Setup of a connection.

When exchange B receives the seizure signal and the called number, it checks whether S_r is idle. If this is the case, it sends a ringing signal on SL_r, and a ringing-tone signal on T_1, to inform S_p. When S_r lifts the handset of the telephone, an answer signal is sent to exchange B, which then stops the ringing signal and ringing tone, sets up a path between T_1 and SL_r, and signals to exchange A that the call has been answered.

The connection is now complete and allows speech or other communications between the subscribers. At the end of the call, another signaling sequence takes place to release the connection.

One-Way and Bothway Trunk Groups. In Fig. 1.1-1, there is at most one trunk group between two exchanges. Let us consider the group TG_1. The network should allow calls originating at A with destination B and calls originating at B with destination A. Therefore, both exchanges are allowed to seize trunks in TG_1. A trunk group whose trunks can be seized by the exchanges at both ends is known as a *bothway* trunk group [1,2].

A pair of exchanges can also be interconnected by two *one-way* trunk groups. The trunks in one-way groups can be seized by one exchange only. For example, exchanges A and B could be interconnected by two one-way trunk groups TG_{1A} and TG_{1B}, whose trunks can be seized by A and B, respectively.

Both arrangements are used in actual networks. Two-way groups have an economic advantage because, for a given traffic intensity, the number of trunks of a bothway trunk group can be smaller than the total number of trunks in the one-way groups.

In bothway groups, it can happen that the exchanges at both ends of a trunk group seize the same trunk at the same time (double seizure). There are several alternatives to deal with a double seizure. For example, it can be arranged that one exchange continues the setup, and the other exchange backs off (tries to. seize another trunk for its call). The signaling on bothway trunks includes provisions to alert the exchanges when a double seizure occurs.

1.1.2 Networks

In everyday life, we think of "the" telecommunication network that allows us to speak, or send faxes and other data, to just about anybody in the world. In fact, the telecommunication network is an aggregation of interconnected networks of several types.

Networks can be classified as shown in Fig. 1.1-4. In the first place, the (global) telecommunication network consists of national networks and the international network. In turn, a national network is a combination of public and private networks. Public networks are for general use; private networks can be used only by employees of the organization (an airline company, the U.S. government, etc.) that owns the network. A public network consists of a "fixed" network and a number of "cellular mobile" networks. In the United States, the fixed public network—known as the *public switched telecommunication network* (PSTN)—consists of about 150

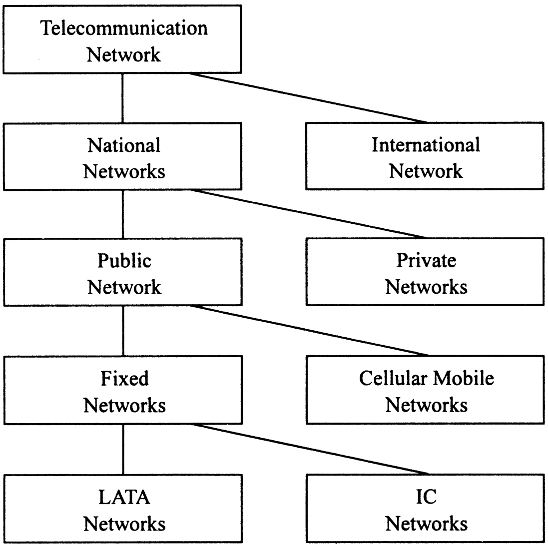

Figure 1.1-4. Networks.

LATA (*local access and transport area*) networks (the network of Fig. 1.1-1 is a LATA network), interconnected by networks that are known as IC (*interexchange carrier*), or long-distance, networks.

We now examine the interconnections of these networks. LATA and IC networks are interconnected by internetwork trunk groups—see Fig. 1.1-5. Some *local exchanges* (A) have a direct trunk group to an exchange of an IC, other exchanges (B, C, D, E) have access to the IC network via an intermediate (*tandem*) exchange in their respective LATAs.

A cellular network has one or more *mobile switching centers* (MSCs)—see Fig. 1.1-6. Each MSC is connected by an internetwork trunk group to a nearby tandem exchange T of a fixed (LATA) network. When a mobile station is making a call, it uses a radio channel of a nearby MSC.

Private Branch Exchange (PBXs). These are exchanges owned by government agencies, businesses, and so on and located in buildings that belong to these organizations. A PBX enables the employees in a building to call each other and to make and receive calls from subscribers served by the public network. A PBX is connected by an *access line group* (ALG) to a nearby local exchange (Fig. 1.1-7).

An organization with PBXs in several cities can establish a *private network* that consists of the PBXs and a number of *tie trunk groups* (TTG) between the public local exchanges to which their PBXs are attached. A TTG is a "private" group that is leased by the LATA operator and is dedicated to private-network calls. In Fig. 1.1-7, the connection for a call between public branch exchanges X and Y uses a trunk of TTG_1 and is switched in the public local exchanges A and B.

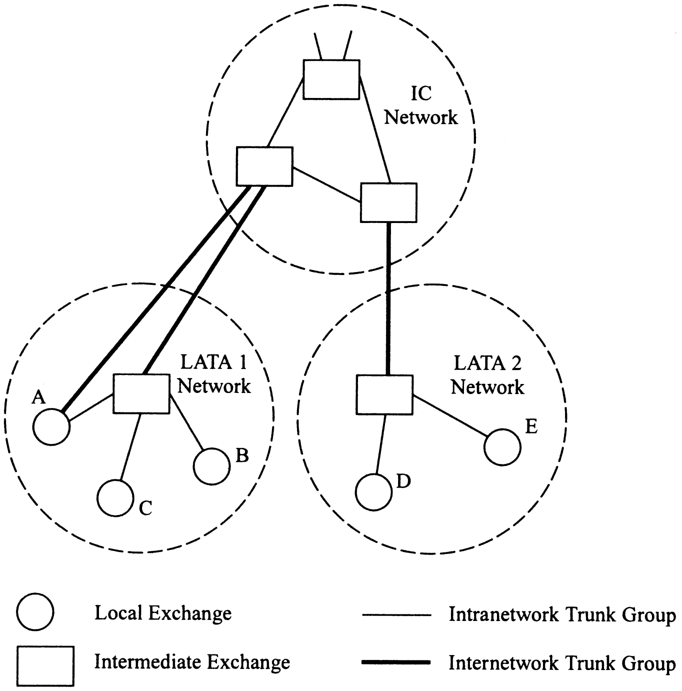

Figure 1.1-5. Interconnection of LATA and IC networks.

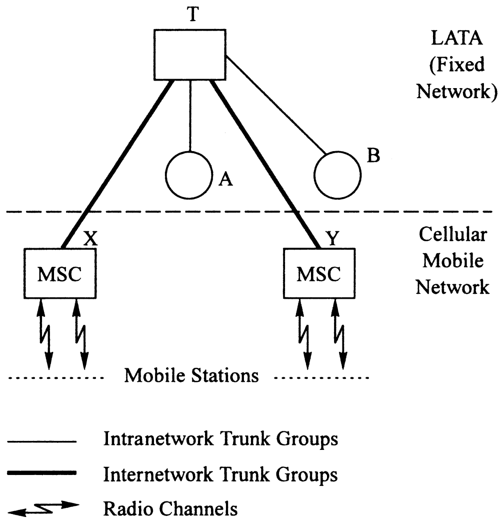

Figure 1.1-6. Interconnection of fixed and mobile networks. MSC: mobile switching center.

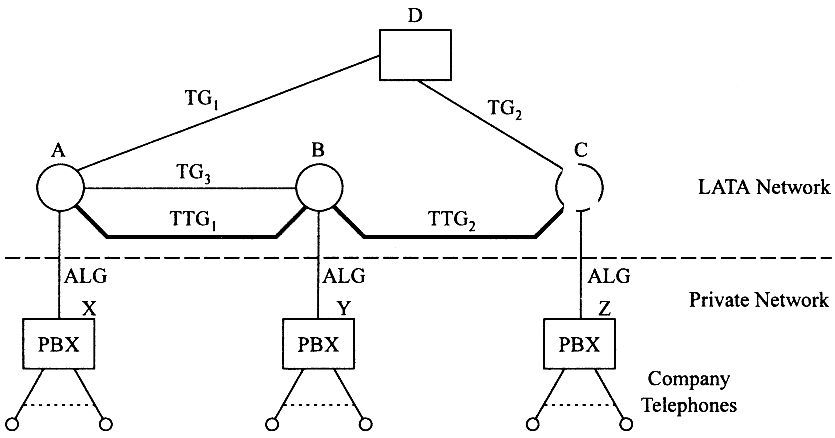

Figure 1.1-7. Interconnection of private network and a LATA network. ALG, access line group; TG, trunk group (public); TTG, tie trunk group (private); PBX, private branch exchange.

Today there are also *virtual private networks* (VPNs). They appear to a business as a private network but use the trunks of the public networks.

International Calls. Figure 1.1-8 shows the interconnection of long-distance (IC) networks in different countries. For a call from country A to country C, an IC network in country A routes the connection to an *international switching center* (ISC). An ISC has national trunk groups to exchanges of its IC and international trunk groups to ISCs in foreign countries.

The term "international network" refers to the combination of the ISCs and their interconnecting trunk groups.

1.1.3 Telecoms

We shall use the term *telecom* to denote a company that owns and operates a public telecommunication network. Until recently, the telecoms in most countries were government-owned monopolies that operated an entire national network. In recent years, a number of countries have started to privatize their telecoms and to allow competition by newly formed telecoms.

The networks in the United States are operated by investor-owned telecoms. Until 1984, the Bell System was the largest telecom, operating practically the entire long-distance network and many—but by no means all—local networks. Independent (non-Bell) telecoms, such as GTE, United Telecoms, and a host of smaller companies, operated the other local networks.

The division of the U.S. public network into *local access and transport areas* (LATAs) and *interexchange carrier* (IXC or IC) networks took place in 1984, due to government actions that broke up the Bell System [1]. As a result, competing IXCs started offering long-distance and international service, while *local exchange*

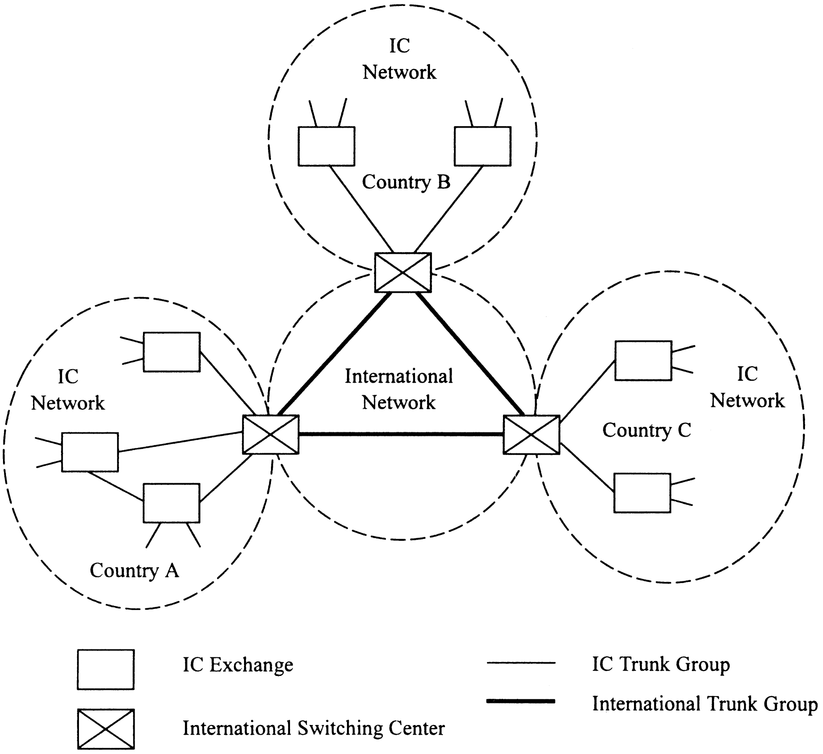

Figure 1.1-8. Interconnections between national IC networks and the international network.

carriers (LECs) provided service on a regional basis, in LATAs. A typical LEC owned a number of adjacent predivestiture local networks. Competition for local service was allowed but was slow in developing because of high entry barriers. The latest development is that IXCs and LECs are allowed to provide both local and long-distance service. In addition, new wireline, cable, and wireless companies can provide competing telephone service. All that has resulted in mergers between LECs, IXCs, cable companies, and wireless companies.

1.1.4 Synonyms

Since telecommunication terms originated rather independently in different countries, technical literature in English still uses different terms for the same concepts, depending on whether the authors are from the United States, from the United Kingdom, or from other English-speaking countries, or documents are translations from other languages. Some frequently used synonyms are listed below:

Subscriber, customer, user
Subscriber line, line, loop

Local exchange, local office, central office, end office
Intermediate exchange, tandem exchange, toll exchange, transit exchange
International switching center, gateway, international exchange
Trunk, junction, circuit
Telecom, administration, carrier, operating company, telephone company, telco, service provider
Exchange, switch
Switchblock, switch fabric

1.2 NUMBERING PLANS

This section explores the formats of the numbers (sometimes called addresses) that identify the subscribers of telecommunication networks.

Subscriber Numbers (Directory Numbers). The geographical area of a nation is divided into several *numbering areas*, and subscriber numbers (SNs) identify subscriber lines within a particular numbering area. A SN consists of an *exchange code* (EC) that identifies an exchange within a numbering area, followed by a *line number* (LN):

$$\mathrm{SN} = \mathrm{EC\text{-}LN}$$

National Numbers. Within a country, a subscriber is identified by a national number (NN), consisting of an *area code* (AC), which identifies the numbering area, followed by a subscriber number:

$$\mathrm{NN} = \mathrm{AC\text{-}SN} = \mathrm{AC\text{-}EC\text{-}LN}$$

International Numbers. Worldwide, a subscriber is known by an *international number* (IN) that consists of a country code (CC), followed by a national number:

$$\mathrm{IN} = \mathrm{CC\text{-}NN}$$

The generic format for the international numbering plan is specified by ITU-T in Rec. E.164 [2]. Three types of numbering schemes are supported by E.164, all with the CC component having up to three digits and the total IN number having a maximum of 15 digits:

1. Numbering plan for geographic areas (subscriber numbers)
2. Numbering plan for global services (e.g., Freephone numbers)
3. Numbering plan for networks (other than the telephone network)

For item 1 the CC may have one, two, or three digits and the NN breakdown is left open for national definitions. For global services and for networks, the CC must be three digits long. For networks, the NN is divided into an *identification code* (IC) of one to four digits and a subscriber number.

When subscriber S_1 calls a subscriber located in the same numbering area, the number dialed is a SN. If the called subscriber lives in the same country but in a different area, S_1 has to dial a NN. If the called party lives in another country, S_1 needs to dial an IN. To allow the local exchange to interpret what is being dialed, prefixes may need to be prepended to the numbers of a numbering plan. Prefixes are part of the *dialing plan*, described in Section 3.7.

National numbering plans define the formats of subscriber and national numbers. Most countries have their own numbering plans. However, the United States, Canada, and a number of Caribbean countries are covered by a common plan that was introduced in the mid-1940s.

1.2.1 North American Numbering Plan (NANP)

The North American territory [1] is divided into *numbering plan areas* (NPAs), which are identified by *three-digit numbering plan area codes*, most often called simply *area codes*, AC(3).

Each area covers a state, or part of a state, but never crosses a state boundary. Lightly populated states (Alaska, New Mexico, etc.) have one NPA, while more heavily populated states (Illinois, New York, etc.) are divided into several NPAs. The territory of a NPA is not identical to the service area of a LATA (LATA boundaries were established much later, after the breakup of the Bell System). Some (but not all) of the AC(3) numbering space in the 8XX and 9XX ranges is used for services where destinations are not directly linked to a geographical area ("800 service" and "900 service"). The 700 NPA is also available for individual carriers who want to introduce new services.

A subscriber number has seven digits: a three-digit exchange code, which defines an exchange within a NPA, followed by a four-digit line number:

$$\text{SN}(7) = \text{EC}(3)\text{-LN}(4)$$

Each EC can cover maximally 10,000 line numbers. Since local exchanges can serve up to some 100,000 subscribers, more than one exchange code may have to be assigned to a particular exchange. For example, in a particular NPA the subscriber numbers 357-XXXX, 420-XXXX, and 654-XXXX might be served by the same local exchange.

National numbers consists of ten digits: a three-digit area code AC(3), followed by a seven-digit subscriber number:

$$\text{NN}(10) = \text{AC}(3)\text{-SN}(7) = \text{AC}(3)\text{-EC}(3)\text{-LN}(4)$$

The NANP (for national numbers) is an example of a closed (uniform) numbering plan. In these plans, the lengths of all subscriber numbers, and of all national numbers, are constant.

Originally, the format of AC(3) was NXX with $N = 2$ or 3 and $X = 0$–9, and the format of EC(3) was NXX with $N = 4$–9 and $X = 0$–9. This allowed up to 160 area codes and up to 640 exchange codes. The increase in telephones in later years required a larger number of area codes, which led to a change resulting in the current format for both AC(3) and EC(3):

$$NXX \quad (N = 2\text{–}9 \text{ and } X = 0\text{–}9)$$

that has a capacity of 800 area codes and 800 exchange codes.

To support competition by new long-distance carriers after the 1984 divestiture, a four-digit *carrier identification code* CIC(4) may be inserted before the called party number to route the call through the long-distance carrier identified by the CIC(4) instead of going through the presubscribed IXC.

Prefixes. In order to expedite the setup of a connection, there are situations in which a subscriber has to dial one or more "prefix" digits ahead of the called party address. For example, since AC(3) and EC(3) codes use the same format, an exchange would have to perform a time-out after receiving the seventh digit to distinguish a subscriber number from a national number. Therefore, subscribers must dial the prefix "1" to indicate that the called address is a NN, while prefix "011" indicates that the address is an international number. Similarly, the prefix "101" must he dialed when a carrier identification code is inserted before the called party number. More details on prefixes are found in Section 3.7.1.

1.2.2 Other National Numbering Plans

Some countries have *open* numbering plans, in which subscriber numbers and area codes (sometimes called *trunk* or *city* codes) are not of fixed length. In these plans, the numbering areas usually have comparable geographical sizes. Heavily populated areas need subscriber numbers with six or seven digits, while four or five digits are sufficient in lightly populated areas. To limit the differences in length of national numbers, the area codes for areas with long subscriber numbers are usually shorter than those for areas with short subscriber numbers. An example of national numbers in an open numbering plan is shown below:

$$\text{NN}(8) = \text{AC}(2)\text{-SN}(6)$$
$$\text{NN}(9) = \text{AC}(2)\text{-SN}(7)$$
$$\text{NN}(8) = \text{AC}(4)\text{-SN}(4)$$
$$\text{NN}(9) = \text{AC}(4)\text{-SN}(5)$$

In order to allow exchange to interpret national numbers in this plan, the two initial digits of a four-digit area code cannot be the same as those of a two-digit area code. For example, the two-digit area code 70 precludes the use of four-digit area codes 70XX.

1.2.3 Country Codes

The country codes have been established by ITU-T [2] and consist of one, two or three digits. The first digit indicates the world zone in which the called party is located:

World Zone

1: North America
2: Africa
3: Europe
4: Europe
5: Latin America
6: Australia and Southern Pacific Region
7: Former Soviet Union
8: China and Northern Pacific Region
9: Middle East

Country codes starting with 1 and 7 are one-digit codes and represent, respectively, North America and the former Soviet Union. Country codes starting with 2 through 9 can have two- or three-digit codes, and the combinations of the first and second digit determine which is the case. For instance, in world zone 3, all combinations except 35 are two-digit codes, as shown by the following examples:

31	The Netherlands
354	Iceland
359	Bulgaria

These rules enable exchanges to separate the country code from the national number in a received international number.

Country code 1 represents the United States, Canada, and a number of Caribbean countries. Most area codes represent areas in the United States. Other codes represent areas in Canada and individual Caribbean nations.

1.2.4 Digit Deletion

In Fig. 1.2-1, calling party S_1 and called party S_2 are located in different NPAs of the United States. S_1 therefore dials the national number (NN) of S_2. As a general rule, exchanges send subscriber numbers to exchanges that are in the NPA of the destination exchange, and national numbers to exchanges outside the destination

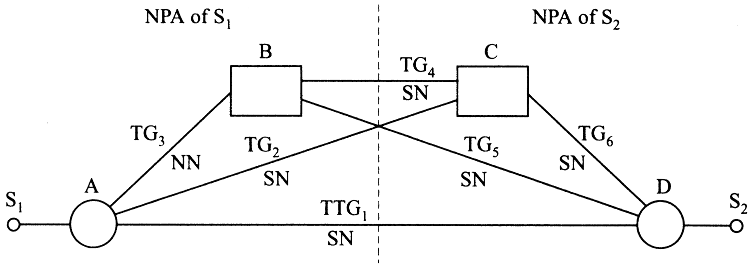

Figure 1.2-1. Called numbers. SN, subscriber number; NN, national number.

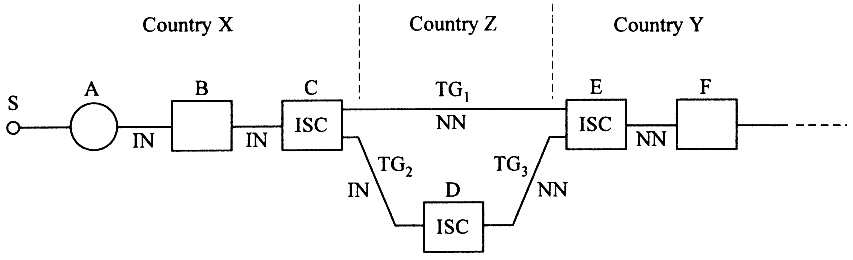

Figure 1.2-2. Called number formats on international calls. NN, national number; IN, international number.

NPA. In Fig. 1.2-1, exchange C is in the NPA of D, and exchanges A and B are not. Exchange A has received the NN of S_2. If A routes the call on a trunk of TG_1 or TG_2, it deletes the area code from the received number and sends the SN of S_2. However, if A routes the call on TG_3, it has to send the NN of S_2. Exchanges Band C always send the SN.

A similar digit deletion occurs on international calls. Figure 1.2-2 shows a call from subscriber S in country X to a subscriber in country Y. An ISC sends a national or international number, depending on whether it routes the call on a direct trunk on an ISC in the destination country, or on a trunk to an ISC in an intermediate country.

1.3 DIGIT ANALYSIS AND ROUTING

1.3.1 Destinations and Digit Analysis

Connections for interexchange calls are set up along paths that have been predetermined by the network operator. A *route* is a path to a particular *destination*. An exchange determines the call destination by analyzing the called number and then selects an outgoing trunk in a route to the destination.

We need to distinguish two destination types. The *final destination* (FDEST) of a call is the local exchange that serves the called party. An *intermediate destination*

(IDEST) is an exchange where the call path enters another network, on its way to the final destination. For a connection, the destination at the exchanges of the local network serving the called party is a FDEST. The destinations at exchanges in the other networks are IDESTs.

As an example, take a call from a calling party in local network $LATA_1$ to a called party in $LATA_2$. The interexchange carrier designated by the calling party is IC_D. In the exchanges of $LATA_1$, the call has an IDEST, namely, an exchange in the network of IC_D, predetermined by the telecoms of the $LATA_l$ and IC_D networks. In the IC_D exchanges, the call also has an IDEST: an exchange in $LATA_2$ predetermined by the telecoms of IC_D and $LATA_2$. In the exchanges of $LATA_2$, the call has final destination.

Digit analysis is the process that produces a FDEST or an IDEST from the called subscriber number (EC-LN), national number (AC-EC-LN), or international number (CC-AC-EC-LN).

In LATA exchanges, calls with subscriber and national numbers can have IDEST or FDEST destinations. Calls with international called numbers always have an IDEST. In IC exchanges, all calls have IDEST destinations. In calls with national called numbers, the IDEST is an exchange in the LATA network determined by the combination AC-EC.

For calls with international called numbers, the IDEST depends on whether the IC exchange is an ISC. If the exchange is not an ISC, the call destination is an ISC in the IC network, determined by the country code (CC) in the number. At an ISC, the destination is an ISC in the country identified by CC.

1.3.2 Routing of Intra-LATA Calls

Intra-LATA calls are handled completely by one telecom. In these calls, the FDEST is the local exchange of the called party. We examine a few routing examples for calls from a caller on local exchange A to a called party on local exchange D.

In Fig. 1.3-1(*a*), the telecom of the LATA has specified one indirect route, consisting of trunk groups TG_1 and TG_2:

Route A-TG_1-X-TG2-D

The fact that a TG belongs to a route does not mean that the TG is dedicated to the route. For example, TG_1 can also belong to routes from A to other destinations.

In Fig. 1.3-1(*b*), the telecom has specified a set of four routes:

A-TG_3-D

A-TG_4-Y-TG_6-D

A-TG_1-X-TG_2-D

A-TG_1-X-TG_5-Y-TG_6-D

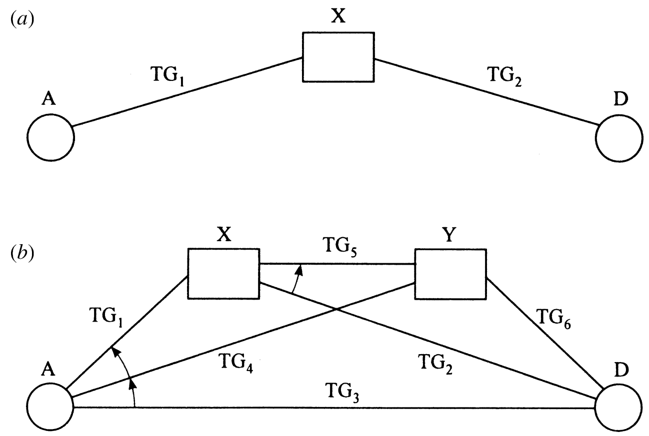

Figure 1.3-1. Routes for calls from A to D (intra-LATA calls).

As perceived by an exchange, a route to a destination is an outgoing trunk group (TG). In Fig. 1.3-1(*b*), the route set at exchange A for destination D consists of trunk groups TG_1, TG_3, and TG_4.

Each exchange has a list of routes that can be used for a destination. The lists for destination D at exchanges A, X, and Y are:

Exchange	Routes for Destination D
A	TG_3, TG_4, TG_1
X	TG_2, TG_5
Y	TG_6

Alternate routing is the procedure by which an exchange selects an outgoing trunk for a call when there are several routes to a destination. In this procedure, the order in which the routes are listed specifies the sequence in which an exchange checks the outgoing trunk groups for available trunks. In this example, exchange A first tries to find an available trunk in its first-choice route TG_3. If a trunk is available, A seizes the trunk. If not, it attempts to find an available trunk in its second-choice route TG_4, and so on. If none of these routes has an available trunk, exchange A aborts the setup of the call.

In alternate routing, the TGs to a destination are ordered such that the first-choice route is the most direct one (passing through the smallest number of intermediate exchanges), the second-choice route is the most direct one among the remaining routes, and so on. In Fig. 1.3-1(*b*), the arrows indicate the selection sequences at exchanges A and X.

1.3.3 Routing of Inter-LATA Calls

Figure 1.3-2 shows a routing example for a call originated by a subscriber attached to local exchange A of $LATA_1$, to a called party attached to local exchange Z in $LATA_2$.

In $LATA_1$, the IDEST of the call is exchange P or Q, in the IC network designated by the calling subscriber—Fig. 1.3-2(*a*). The routes for this destination at exchanges A, B, and C are:

Exchange	Routes for IDEST
A	TG_1, TG_2
B	TG_3, TG_4
C	TG_5, TG_6

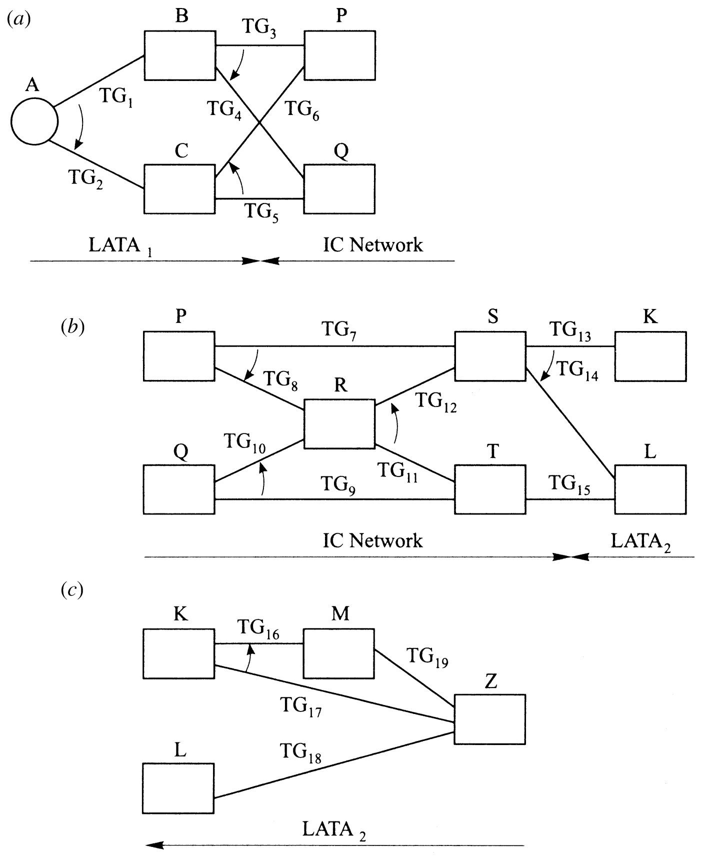

Figure 1.3-2. Routing of inter-LATA call: (*a*) routing in $LATA_1$, (*b*) routing in IC network, and (*c*) routing in $LATA_2$.

In the IC network, the IDEST for the call is exchange K or L $LATA_2$—Fig. 1.3-2(*b*)—and the routes for this destination at exchanges P, Q, R, S, and T are:

Exchange	Routes for IDEST
P	TG_7, TG_8
Q	TG_9, TG_{10}
R	TG_{11}, TG_{12}
S	TG_{13}, TG_{14}
T	TG_{15}

Finally, in $LATA_2$, the FDEST of the call is local exchange Z—Fig. 1.3-2(*c*)—and the routes at exchanges K, L, and M are:

Exchange	Routes for IDEST
K	TG_{17}, TG_{16}
L	TG_{18}
M	TG_{19}

1.3.4 Automatic Rerouting

Automatic rerouting (also called *crankback*) is a refinement of *alternate routing*. It is used in AT&T's long-distance network [3]. The procedure is illustrated with the example of Fig. 1.3-1(*b*).

Suppose that TG_3 is congested, and that A has seized trunk T in its second-choice route TG_4. The call setup arrives at exchange Y, which has only one route (TG_6) to destination D. Under alternative routing, if no trunk is available in TG_6, exchange Y abandons the setup and informs the calling party with a tone or recorded announcement.

Under automatic rerouting, exchange Y signals to exchange A that it is unable to extend the setup. A then releases trunk T and tries to route the call on its final route (TG_1). If trunks are available in TG_1 and TG_2, the connection can be set up.

Automatic rerouting depends on the ability of an exchange (Y) to signal the preceding exchange (A) that it is not able to extend the setup. We shall encounter signaling systems that have signals for this purpose.

1.4 ANALOG TRANSMISSION

Until 1960, analog transmission was the only form of transmission in telecommunication networks. Today, the telecommunication network is mostly digital, except for the subscriber lines.

This section outlines some basic aspects of the transmission of analog signals in telecommunications. In this section, the term *signal* refers to information (speech or voiceband data) exchanged between subscribers during a call (as opposed to *signaling*, which is the subject of this book).

1.4.1 Analog Circuits

An analog signal is a continuous function of time [4,5]. Telecommunication started out as telephony, in which a microphone (or transmitter, mouthpiece) produces an electrical *analog* signal, whose variations in time approximate the variations in air pressure produced by the talker's speech. A receiver (or earpiece) reconverts the electrical speech signal into air-pressure variations that are heard by the listener.

The pressure variations of acoustic speech are complex and not easily described. "Average" acoustic speech contains frequencies from 35 Hz to 10,000 Hz. Most of the speech power is concentrated between 100 Hz and 4000 Hz. For good-quality telephony, only the frequencies between 300 and 3400 Hz need to be transmitted. Analog communication channels in the network, which historically have been designed primarily for speech transmission, therefore accommodate this range of *voiceband* frequencies (Fig. 1.4-1).

Two-Wire and Four-Wire Circuits. Analog circuits can be two-wire or four-wire. Subscriber lines are two-wire circuits, consisting of a pair of insulated copper wires that transfer signals in both directions. Most analog trunks are four-wire circuits, consisting of two unidirectional two-wire circuits, one for each direction of transmission.

In the diagrams of this section, circuits are shown as in Fig. 1.4-2. The arrows indicate the directions of transmission. In general, bidirectional circuits (two- or four-wire) are shown as in (*a*), and a note indicates whether the circuit is

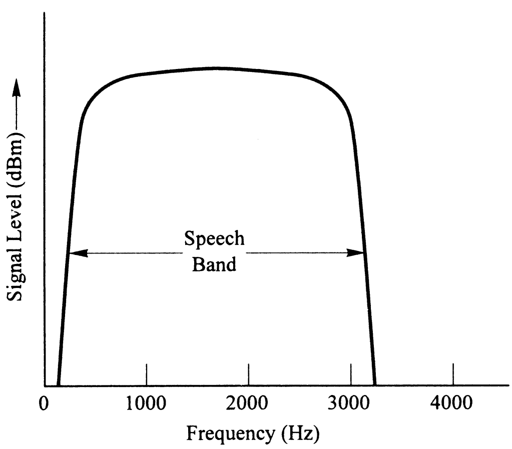

Figure 1.4-1. Frequency response of analog transmission circuits. (From R. L. Freeman, *Telecommunication System Engineering*, 2nd ed., Wiley, 1992.)

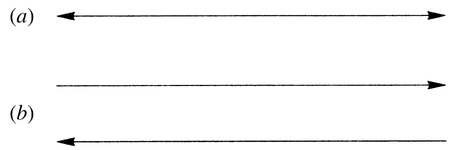

Figure 1.4-2. Circuit representations: (*a*) bidirectional two-wire or four-wire circuit and (*b*) two unidirectional two-wire circuits forming a bidirectional four-wire circuit.

two-wire or four-wire. When discussing the two unidirectional circuits of a four-wire circuit, representation (*b*) is used.

Data Transmission on Analog Circuits. In the 1960s, subscribers also began to use the telephone network for transmission of digital data, and the network has become a *telecommunication* network. Digital data are converted by *modems* into a form that fits within the 300–3400 Hz band of analog circuits. There are several modem types. *Frequency-shift keying* (FSK) modems convert the zeros and ones of the digital bit stream into two voiceband frequencies, for example, 1300 and 1700 Hz. With FSK, data can be sent at speeds of 600 or 1200 bits/second. The signal produced by a *differential phase-shift keying* (DPSK) modem is a single frequency with phase shifts. In the widely used V.26 modem [6], the frequency is 1800 Hz, and the phase shifts occur at a rate of 1200 shifts/second. The phase shifts can have four magnitudes, each of which represents the values of two consecutive bits in the digital signal. The modem thus transfers 2400 bits/second. More recently developed modems have transfer rates of up to 56 kb/second.

1.4.2 Analog Subscriber Lines [4,5]

Figure 1.4-3 shows a connection between two subscribers served by an analog local exchange. The subscriber lines (SLs) and the path (P) across the exchange are two-wire circuits.

The power of an electrical signal decreases as it propagates along the circuit. This attenuation becomes more severe with increasing circuit length.

The characteristics of microphones and receivers are such that a listener receives a sufficiently strong acoustical signal when at least 1% of the electrical signal power produced by the talker's microphone reaches the listener's receiver. This corresponds to the attenuation in a circuit of about 15 miles. Most subscriber

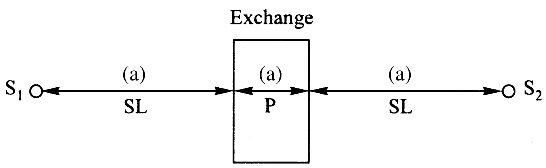

Figure 1.4-3. Circuit for intraoffice call. (a), Two-wire analog circuits.

lines are less than 4 miles long, and there are no signal-strength problems in intra-exchange calls.

1.4.3 Two-Wire Analog Trunks

Two-wire trunks are similar to subscriber lines and have similar attenuation characteristics. This limits the trunk length to about 10 miles.

1.4.4 Four-Wire Analog Trunks

Long-distance trunks require amplification to compensate the signal attenuation. Amplifiers are unidirectional devices, and this is why long-distance trunks are four-wire trunks. A four-wire circuit consists of two amplified unidirectional two-wire circuits. In Fig. 1.4-4, *hybrid* circuits (H) at both ends of the trunk convert a two-wire circuit into a four-wire circuit, and vice versa. Amplifiers (A) are located at regular intervals along the two unidirectional circuits. The unidirectional circuits at an exchange that transfer signals to and from the distant exchange are known as the send circuit (S) and the receive circuit (R).

1.4.5 Frequency-Division Multiplexing

Frequency-division multiplexing (FDM) [4,5] is a technique to carry the signals of a group of n analog four-wire trunks on a common four-wire analog transmission system. In each direction of transmission, the bandwidth of the transmission system is divided into n 300–3400 Hz channels, spaced at intervals of 4 kHz, for a total bandwidth of $n \times 4$ kHz (Fig. 1.4-5).

Figure 1.4-6 shows a group of n FDM trunks between exchanges X and Y. In this example, the trunks appear at the exchanges as two-wire circuits (a) that can transfer analog signals in the 300–3400 Hz range.

Frequency-division multiplexing (FDM) equipment is located at both exchanges. Hybrids (H) convert the two-wire circuits into four-wire circuits. For transmission in direction X $\rightarrow$ Y, multiplexer FDM-X shifts the 300–3400 Hz send channels (S) of

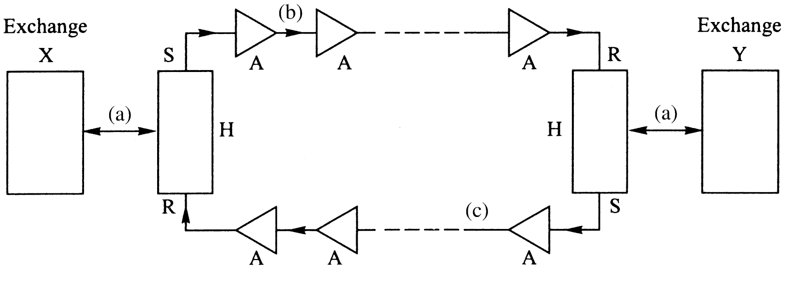

Figure 1.4-4. Amplified four-wire analog trunk. (a),Two-wire bidirectional analog circuits; (b,c), pair of unidirectional analog circuits forming a bidirectional circuit.

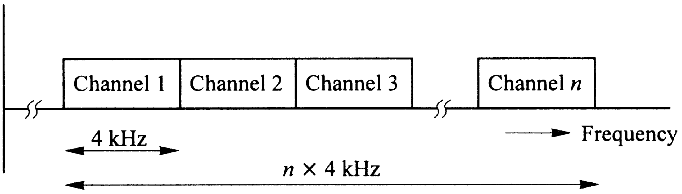

Figure 1.4-5. Frequency allocation of FDM transmission system.

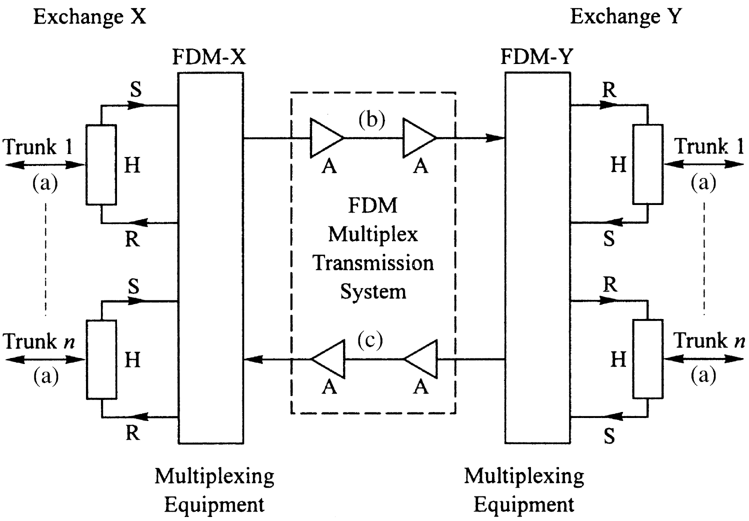

Figure 1.4-6. Multiplexed analog trunks. (a), Two-wire bidirectional circuit; (b,c), multiplexed unidirectional circuits (*n* channels), forming a four-wire FDM transmission system; A, amplifiers; S, R, two two-wire unidirectional circuits, forming a bidirectional circuit.

the trunks "up" in frequency (by different amounts) and then combines the channels on a unidirectional multiplex transmission circuit (b). FDM-Y demultiplexes the signal received on (b), by shifting the channels downward in frequency. The demultiplexed channels are the 300–3400 Hz receive channels (R) of the trunks at exchange Y.

In the other direction, the send channels at exchange Y are multiplexed by FDM-Y, combined on multiplex transmission circuit (c), and demultiplexed by FDM-X. Amplifiers A along the multiplexed circuits (b) and (c) compensate the signal attenuation.

Larger FDM multiplex systems can be formed by repeated multiplexing. This creates a FDM multiplex hierarchy, which has been standardized internationally [5]. In Fig. 1.4-7, the 60 two-wire trunks T are converted to four-wire circuits (b) by hybrids (H). The first-order FDM multiplexes convert 12 four-wire circuits into a four-wire *group* circuit (c) with 12 channels. The bandwidth of group circuits is $12 \times 4 = 48$ kHz in each direction (60–108 kHz).

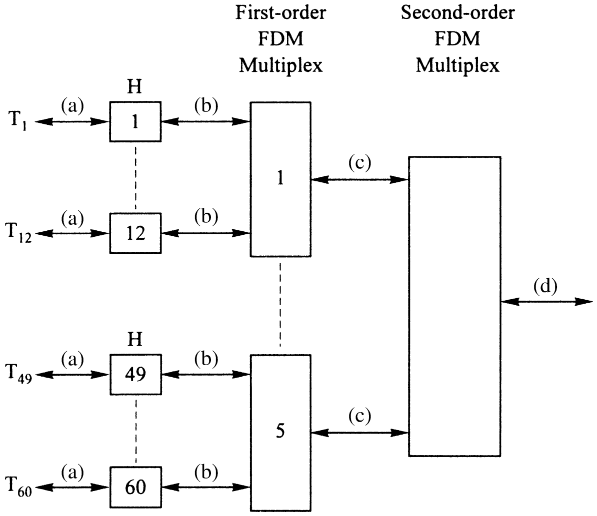

Figure 1.4-7. First- and second-order FDM multiplexes. (a), Two-wire bidirectional circuits; (b), four-wire circuits; (c), multiplexed four-wire circuits (12 channels); (d), multiplexed four-wire circuits (60 channels).

The second-order FDM multiplex combines five group circuits (c) into one four-wire *supergroup* circuit (d) with 60 channels. The bandwidth of a supergroup circuit is 5×48 kHz $= 240$ kHz in each direction (312–552 kHz).

Continuing in this way, ten supergroups can be combined into a 600 channel *mastergroup*, and six mastergroups form a 3600 channel *jumbogroup* [5].

1.4.6 FDM Transmission Systems

Group circuits can be carried on two amplified wire-pairs, with amplifiers at regular spacings. Signals of higher order multiplexes are carried on transmission systems of several types.

Cable transmission systems consist of two coaxial cables, with amplifiers at regular spacings. Cable systems are used on overland and underwater (transatlantic, transpacific) routes.

Microwave radio systems consist of a pair of unidirectional *radiofrequency* (RF) transmission links in the microwave region (2, 4, 6, 11, or 18 GHz). The output signal of a FDM multiplexer modulates the frequency of the microwave carrier. The RF signal travels in a narrow beam from the transmitting antenna to the receiving antenna.

In terrestrial systems, the microwave links are divided into several sections (hops) of, say, 20–40 miles. The transmission in each section is on a line-of-sight path. Repeater stations at the section boundaries amplify the RF signal received from one section and retransmit it to the next section. The repeater stations are land based, and terrestrial microwave systems can therefore be used on overland routes only.

Satellite microwave systems use communication satellites. These satellites are in *geosynchronous* orbits, some 22,300 miles above the equator. At this altitude, a satellite circles the earth at an angular velocity equal to the angular velocity of the earth's rotation. When observed from a point on earth, the satellite therefore appears in a fixed position, and only small and infrequent adjustments are needed to keep an antenna of a ground station aimed at the satellite. Each satellite link consists of an "uplink" from a ground station to the satellite, and a "downlink" from the satellite to another ground station. The satellite amplifies the signal received on the "uplink" and retransmits it on the "downlink."

Satellite microwave systems were introduced in the 1960s and have been used extensively on transoceanic routes. A disadvantage of satellite transmission is the long signal propagation time (in excess of 250 ms from ground station to ground station), which can cause problems in telephone conversations.

1.5 DIGITAL TRANSMISSION

Analog signals (a) can be converted to digital bit streams (d) and transmitted on digital transmission systems (Fig. 1.5-1).

Digital transmission is more robust and of better quality than analog transmission. While digital signals require more bandwidth, digital transmission systems are often less expensive than their analog counterparts, and digital transmission is rapidly replacing analog transmission in telecommunications.

1.5.1 Pulse Code Modulation

Pulse code modulation (PCM) is the earliest method for *analog-to-digital* (A/D) and *digital-to-analog* (D/A) conversions used in telecommunication networks [4,5]. The basic idea dates back to 1938 [7]. However, PCM requires a fair amount of digital signal processing and only became technically feasible after the

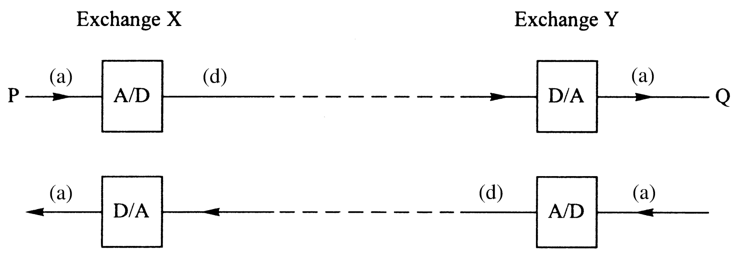

Figure 1.5-1. Digital transmission system. (a), Analog signals; (d), digital bit stream; A/D, analog-to-digital converter; D/A, digital-to-analog converter.

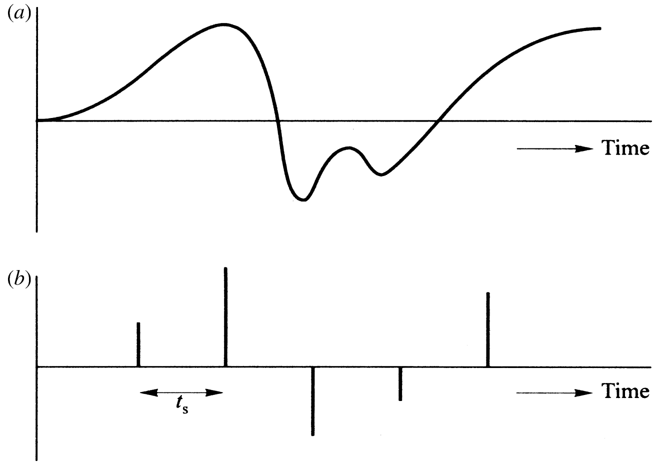

Figure 1.5-2. Sampler (*a*) input and (*b*) output.

invention of semiconductor devices. The first PCM systems were installed in the United States by the Bell System, in 1962 [1,4].

PCM is designed such that its frequency response, measured from analog input signal (P) to analog output signal (Q) of Fig. 1.5-1, is the same (300–3400 Hz) as for FDM analog transmission—see Fig. 1.4-1.

In PCM, the analog input signal (a) of an A/D converter is sampled at $t_s = 125$ microsecond intervals (8000 samples/second)—see Fig. 1.5-2. The magnitude of each sample is then converted into an *octet* (an eight-bit binary number). The digital signal (d) of Fig. 1.5-1 is a sequence of octets, transmitted at $8 \times 8 = 64$ kb/s (kilobits/second).

PCM Standards. In order to preserve the quality of low-level speech, the A/D and D/A conversions are nonlinear. Two coding rules are in existence: mu-law coding is the standard in the United States, Canada, and Taiwan, and A-law coding is the standard in the rest of the world [7,8].

1.5.2 Time-Division Multiplexing

PCM trunks are carried on four-wire *time-division multiplexing* (TDM) transmission systems [8]. Figure 1.5-3 shows a first-order pulse code modulation multiplex (PCM-X) for *m* trunks. In the example, the trunks are attached to the exchange as two-wire analog trunks (a) and are converted to four-wire analog circuits by hybrids (H).

A multiplexer PCM-X does the A/D conversions of the signals in the analog *send channels* (S) and then multiplexes the resulting octets into the outgoing bit stream (b). PCM-X also segregates the octets of the individual channels in the incoming

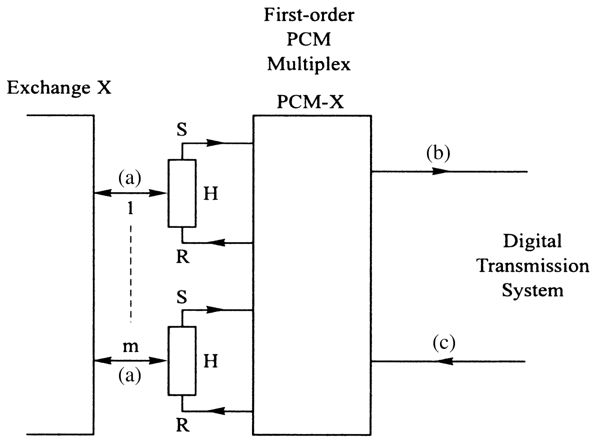

Figure 1.5-3. PCM first-order multiplex, attached to a two-wire analog exchange. (a), Analog two-wire bidirectional circuits; S, R, send and receive circuits of analog four-wire circuits; (b,c), outgoing and incoming circuits of a four-wire multiplexed digital circuit (*m* channels); H, hybrids.

bit stream (c) and does the D/A conversions that produce the voiceband analog signals in the *receive channels* (R).

T1 and E1 Digital Transmission Systems. There are two standards for first-order transmission systems. The T1 system, a development of Bell Laboratories, is used mainly in the United States [1,8]. The E1 system has been defined by the European Telecommunications Standards Institute (ETSI) and is used in most other countries [8].

In both systems, the bit stream is divided in *frames* that are transmitted at a rate of 8000 frames/second. Each frame is divided into a number of eight-bit time-division channels, known as *time slots* (TS), that contain the PCM samples of the individual trunks.

The frame format of the T1 system (known as the *DS1 format*) is shown in Fig. 1.5-4(*a*) [4,5,8]. It consists of $m = 24$ time slots (TS_1 through TS_{24}) and a framing bit (F). The F bits in successive frames form a fixed *synchronization* sequence that repeats every 12 frames. This enables a multiplexer to identify the start points of the individual frames in the incoming bit stream. The frame length is $1 + 8 \times 24 = 193$ bits, and the transmission rate—at points (b) and (c) of Fig. 1.5-3—is 193×8000 bits/second = 1544 kilobits/second (kb/s).

The El frame contains 32 eight-bit time slots, numbered TS_0 through TS_{31}—see Fig. l.5-4(*b*)—and serves $m = 30$ trunks. TS_1 through TS_{15} and TS_{17} through TS_{31} hold the octets for the samples of the trunks. Time slot TS_0 has a fixed eight-bit synchronization pattern, and time slot TS_{16} contains signaling information [4,8].

A E1 frame thus has $32 \times 8 = 256$ bits, and the bit rate on the transmission system is $8000 \times 256 = 2048$ kb/s.

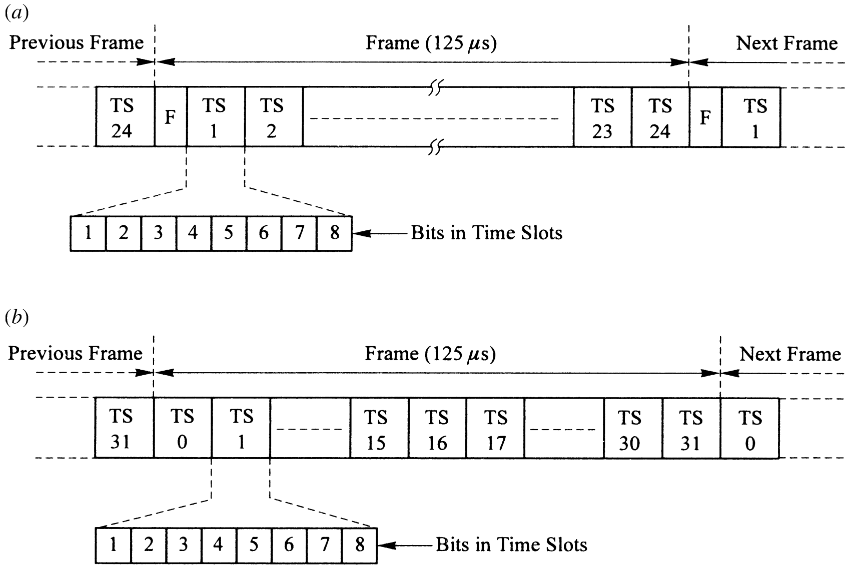

Figure 1.5-4. Formats of first-order PCM frames: (*a*) DS1 format ($m = 24$ channels) and (*b*) E1 format ($m = 30$ channels). TS, time slot.

Higher Order Multiplexes. As in FDM, higher order PCM multiplexes are formed by repeated multiplexing. The North American and the ETSI system have separate hierarchies [8].

1.5.3 Digital Transmission Systems [4,5]

First-order PCM multiplexes are carried on two copper wire-pairs in a conventional cable. The pulses attenuate and stretch as they traverse the cable and have to be regenerated by repeaters (R), located at intervals of 1 mile (Fig. 1.5-5). Since it is difficult to maintain transmission systems with large numbers of repeaters, the maximum length of these systems is about 200 miles.

The signals of higher order multiplexes can be carried on transmission systems of several types.

The North American T4M system carries a fourth-order PCM multiplex (4052 channels) on a pair of coaxial cables, with 1-mile repeater spacing. The T4M system is also limited to rather short trunks. Longer trunks are carried on microwave radio, or fiberoptic transmission systems.

In PCM microwave radio systems, the digital bit stream modulates a microwave carrier signal. *Phase-shift keying* (similar to DPSK modems—see Section 1.4.1) is a frequently used modulation form. Both terrestrial and satellite microwave transmission systems are in use. Like their analog counterparts, digital terrestrial radio links are divided into sections, with repeater stations at the section boundaries.

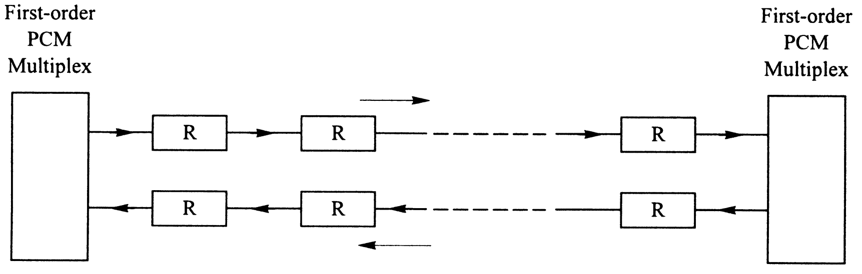

Figure 1.5-5. Transmission of first-order PCM multiplex on wire pairs. R, repeaters.

In optical fiber systems, the digital signal is in the form of lightwave pulses that propagate through a thin glass fiber (diameter on the order of 0.01 mm). At the sending end of the fiber, a laser diode converts the electrical pulses into "light" pulses. A photodiode at the receiving end converts the optical pulses back into electrical pulses. These transmission systems offer high-speed transmission with low attenuation and permit longer repeater spacing. Data rates of several gigabits/second are possible with repeater spacing of hundreds, even thousands, of kilometers, depending on the type of fiber.

Two standards for optical transmission exist:

1. Synchronous Optical Network (SONET)
2. Synchronous Digital Hierarchy (SDH)

Those standards specify a synchronous, framed transmission structure based on a hierarchical arrangement where lower order rates (called *tributaries*) are multiplexed into higher-order rates by byte interleaving, after adding overhead. SONET and SDH frames are transmitted at a rate of 8000 frames/second (every 125 μs), like T1 and E2 frames.

SONET is a North American standard, originally developed by Bellcore (now Telcordia), specified by ANSI [9,10], and fully compatible with T1 and multiples of T1 rates. SONET refers to an electrical signal as a *synchronous transport signal* (STS) and to its optical equivalent as an *optical carrier* (OC). Levels in the STS/OC hierarchy are called STS-n and OC-n with n = 1, 3, 12, 48, 192, and so on. The base rate is STS-1/OC-1 (51.84 Mbps) and can carry 28 DS-1 systems.

SDH, developed after SONET, is an international standard specified by ITU-T [11–13], which follows a similar hierarchical structure and is compatible with both T1 and E1 rates. Because most, if not all, SDH implementations carry only ETSI-defined data rates (E1 and multiples of E1 rates) as payload of the base rate, SDH is considered a European standard. SDH has been adopted by ETSI and is widely used outside North America. SDH signals are referred to as *synchronous transport modules* (STMs), and their hierarchical levels are called STM-n with n = 1, 4, 16, 64, and so on. The base rate is STM-l (155.52 Mbps) and can carry 63 E1 systems.

SONET is compatible with SDH starting with STM-3/OC-3, which corresponds to STM-1 (155.52 Mbps). STS-12/OC-12 then corresponds to STM-4, and so on.

1.5.4 Adaptive Differential Pulse Code Modulation

Since the introduction of PCM, several other coding techniques have been developed. The main objective of these developments is to lower the bit rate while maintaining the transmission quality. One of these methods, *adaptive differential pulse code modulation* (ADPCM), is being introduced in some telecommunication networks.

In ADPCM, four-bit digital samples, transmitted at 8000 samples/second, represent the *differences* in magnitude of two consecutive samples of the analog input signal. The coding is "adaptive": the relation between the magnitude of the analog quantity and the associated digital code is not fixed, but adapts automatically to the characteristics of the signal being encoded.

An ADPCM channel for a trunk requires $8000 \times 4 = 32$ kb/s and gives a speech quality almost equal to PCM. ADPCM doubles the channel (trunk) capacity of digital transmission systems. For example, by dividing the eight-bit octets of the E1 frame of Fig. 1.5-4(*b*) into two four-bit groups, a first-order E1 transmission system can serve 60 trunks.

1.6 SPECIAL TRANSMISSION EQUIPMENT

This section describes two types of special transmission equipment: *echo control* equipment and *circuit multiplication* equipment. Both of these exist in analog and digital versions.

1.6.1 Echoes

Figure 1.6-1 shows the transmission circuit (omitting the exchanges) for a typical long-distance connection with analog four-wire trunks. It consists of three parts. Parts 1 and 3 are two-wire circuits, containing the subscriber loops, and possibly two-wire trunks. Part 3 is a four-wire trunk. Hybrids H_1 and H_2 convert two-wire transmission into four-wire transmission.

When subscriber S_1 speaks, the speech signal leaves H_1 at port P, reaches port Q of H_2, and then travels on the two-wire circuit to listener S_2. However, a small part of the signal received at Q "leaks" to R, and thus returns to S_1, who hears an *echo* of his speech. The leakage occurs because hybrid circuits are "balancing" circuits. For leak-free operation, the impedance presented by the two-wire circuit (at port T of hybrid H_2) would have to match the design impedance of the hybrid for all voice-band frequencies. In each call, H_2 is connected to a different two-wire circuit, and impedances of these circuits (which vary with circuit length and physical cable characteristics) cannot be precisely controlled. Complete balance therefore never occurs in practice, and some echo is always present. When S_2 speaks, a similar echo is caused by reflections at H_1.

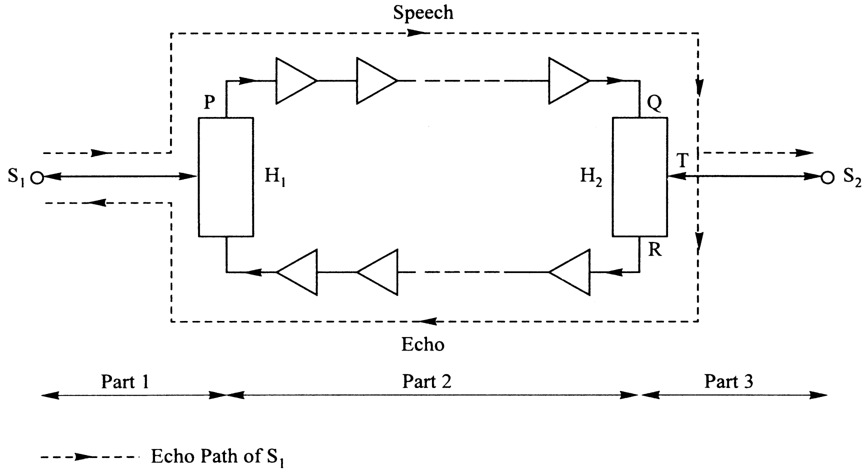

Figure 1.6-1. Long-distance connection.

The effect of echoes on a talker varies with the *echo delay* (the signal propagation time from talker S_1 to H_2 and back to S_1). Echoes delayed by less than 20 ms are barely noticeable, but echoes with larger delays rapidly become very disturbing to a talker. Echo delays increase with increasing length of the four-wire circuit, and trunks longer than 2000 miles (transcontinental, transoceanic, and satellite trunks) are equipped with echo control devices.

1.6.2 Echo Suppressors

Long four-wire analog trunks are equipped with echo suppressors [1,14]. Figure 1.6-2 shows echo suppressor units ES-A and ES-B, located at both ends of a trunk between exchanges A and B (the other exchanges in the connection are not shown).

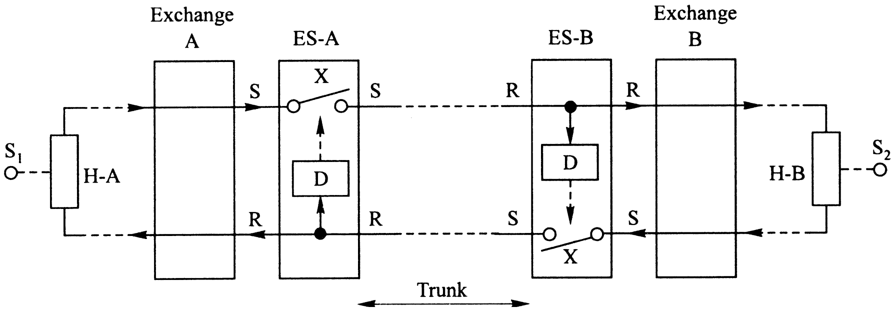

Figure 1.6-2. Echo suppressors on a long-distance analog trunk. D, Voiceband signal detector; X, switch; H, hybrid circuit; ES, echo suppressor.

Each suppressor unit has a detector (D) on its receive pair (R). When D detects the presence of a signal, the suppressor interrupts its send pair (S). For example, when S_1 speaks, ES-B interrupts its S path, and the leakage from hybrid H-B cannot return to S_1. ES-A protects S_2 from echoes in the same way.

When a suppressor has opened its send path, it "hangs over" (leaves the path open) for some 30–40 ms after the signal on the receive path has disappeared. This bridges the silent intervals between successive words of the distant speaker. Suppose now that S_1 has been speaking. The send path of ES-B is therefore open. When near-end subscriber S_2 starts to talk during a hangover, the initial part of the speech is clipped off. The effects of clipping become very noticeable on connections involving several echo-suppressed trunks in tandem. It is therefore desirable to avoid such connections. We shall see that some trunk signaling systems include indicators to make this possible.

Echo suppressor units are always installed in pairs—one at each end of a trunk. Some documents refer to the units as *half echo suppressors.*

1.6.3 Echo Cancelers

Echo cancelers [5,15], which are used on long PCM trunks, also provide protection against echoes but operate on a different principle. Figure 1.6-3 shows a pair of echo cancelers at the ends of a long-distance PCM trunk. The circuit between an echo canceler and its near-end subscriber is known as the "tail" and includes the subscriber line, hybrid circuit (H), an A/D and a D/A converter and can include other digital and analog trunks.

We assume that S_1 is speaking and explore the operation of canceler EC-B. The signals r, e, c, and s are PCM signals (sequences of octets). When S_1 speaks, a signal

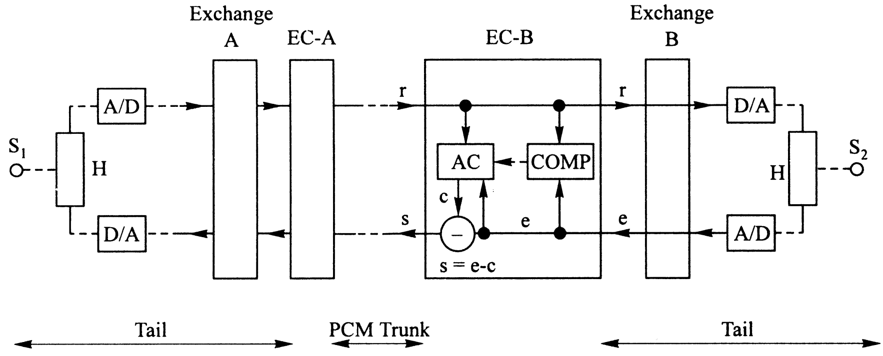

Figure 1.6-3. Echo cancelers on a PCM trunk. EC, Echo canceler; AC, adaptive circuit; COMP, comparator; A/D, analog-to-digital converter; D/A, digital-to-analog converter; H, hybrid circuit.

(r) is received on the receive pair of EC-B. During incoming speech, EC-B regards its tail as a two-port circuit that receives a signal (r) and returns an echo signal (e). Signal r also feeds the *adaptive circuit* (AC) in the canceler, which produces an output signal (c). Signal c is subtracted from signal e, and the results, $s = e - c$, is sent out on the send pair of EC-B.

The tail of an EC is different for each call. At the beginning of a call, the canceler automatically adjusts its adaptive circuit, AC, making the impulse transfer function of the circuit equal to that of the tail. This is done by comparing signals e and c. When the adaptation is complete, signals e and c are equal, and $s = 0$, which means that the echoes returning from the tail are being canceled. The adjustment takes place during the initial seconds of received far-end speech.

Comparator (COMP) compares the strengths of signals r and e. When $r > e$, the canceler is receiving speech from S_1 and can adjust circuit AC. However, when $e > r$, near-end subscriber S_2 is speaking, and no adjustments are made.

Echo cancelers have one significant advantage over echo suppressors: they do not clip the initial part of the near-end subscriber's speech. Therefore, more than one canceler-equipped trunk is allowed in a connection.

Digital Multiplexed Echo Cancelers. In the description above, the EC pair cancels echoes for one connection. Actual echo cancelers are digital devices that serve groups of *m* multiplexed PCM trunks. The canceler is connected to the output of a first-order PCM multiplex (Fig. 1.6-4) and processes the eight-bit PCM samples (octets) of the trunks, during their time slots in each PCM frame. The canceler has an adaptive circuit for each trunk, but its control circuitry is time-shared.

1.6.4 Circuit Multiplication

In a typical telephone conversation, each subscriber speaks during about 30% of the time, and both subscribers are silent during the remaining 40%. During the call, each channel of a four-wire trunk thus carries speech during only 30% of time. *Circuit multiplication equipment* allows a group of four-wire trunks to be carried on a four-wire transmission system with a smaller number of *bearer channels*, by making use of the "silent" intervals [5,14,16,17]. Circuit multiplication is economically

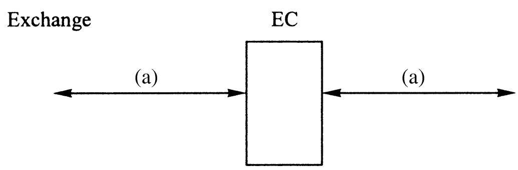

Figure 1.6-4. Echo cancelers for first-order digital time-division multiplexes (*m* channels). (a), Four-wire multiplexed circuits.

attractive in situations where transmission costs are high, for example, in transoceanic transmission systems.

Analog Circuit Multiplication Equipment. Circuit multiplication systems for analog four-wire trunks are generally known as *time assignment speech interpolation* (TASI) systems. Figure 1.6-5 shows two TASI units, at exchanges A and B, that concentrate m trunks to n (four-wire) bearer channels ($n < m$). In each TASI unit, the ($n \times m$) send switch allows up to n simultaneous unidirectional connections between the send circuits (S) of trunks and bearer channels. Likewise, the ($n \times m$) receive switch can connect up to n receive circuits (R) of bearer channels and trunks. Speech detectors are bridged across the S-pairs of the trunks. The controls in the TASI units communicate with each other on a pair of unidirectional control channels.

The assignments of bearer channels to trunks take place independently for each direction of transmission. The TASI operation for transmission from A to B is outlined below.

When a detector in TASI-A detects speech on the S-pair of a trunk, control- A seizes an available S-bearer channel and connects the trunk and the channel. It also informs control-B, identifying the trunk and the channel, and control-B then sets up a path between R-pair of the trunk and the R-bearer channel. This establishes a unidirectional path between the S-pair of the trunk at exchange A and the R-pair of the trunk at exchange B. At the end of a speech burst, an *overhang* timer, with an expiration time of about 400 ms, is started. If new speech energy is detected on the bearer channel while the timer is running, the timer is stopped. Expiration of the timer indicates a 400-ms silent interval on the channel. Control-A then releases the channel and informs control-B.

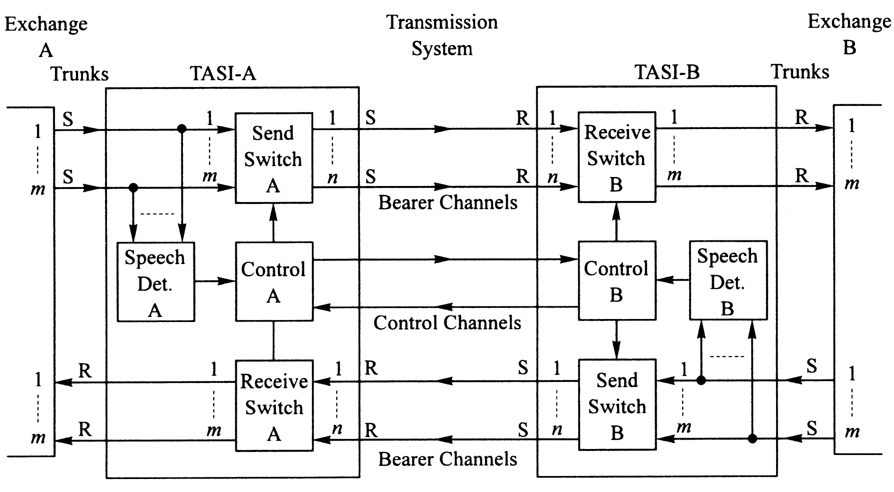

Figure 1.6-5. TASI equipment. R, Analog receive circuits; S, analog send circuits.

The setup of connections is very fast (typically, within 20 ms), and only a small part of the initial syllable of the first word in a sentence is "lost." This is barely noticeable to the listener.

Freeze-out. It can happen that all S-bearer chanels are occupied at the time that speech is detected on a previously silent trunk. In that case, the system has to wait until a bearer channel becomes available, and the initial part of a speech burst is "frozen out" (lost).

The ratio of bearer circuits to trunks is designed such that the probability of freeze-outs lasting more than 60 ms is below 1%. In TASI systems, typical concentration ratios m:n are on the order of 2:1.

TASI systems were designed during the years when speech was the only form of subscriber communication. Today, a small but increasing fraction of calls are used for facsimile and data transmission. During these calls, modem signals can be present continuously in one or both directions, thus occupying one or two bearer circuits on a full-time basis. To keep freeze-outs on the other trunks to an acceptably low level, it has become necessary to reduce the number of trunks that can be served by a TASI system.

Digital Circuit Multiplication Equipment. Digital circuit multiplication equipment operates on the same principle but has first-order multiplex interfaces with the exchange and the transmission system [17]. Moreover, most digital TASI systems convert the eight-bit PCM octets on the trunks to four-bit ADPCM groupings on the bearer channels. This doubles the channel capacity of the transmission system and reduces the per-channel transmission cost by another 50%. Figure 1.6-6 shows a typical configuration, where 150 PCM trunks (five first-order E1

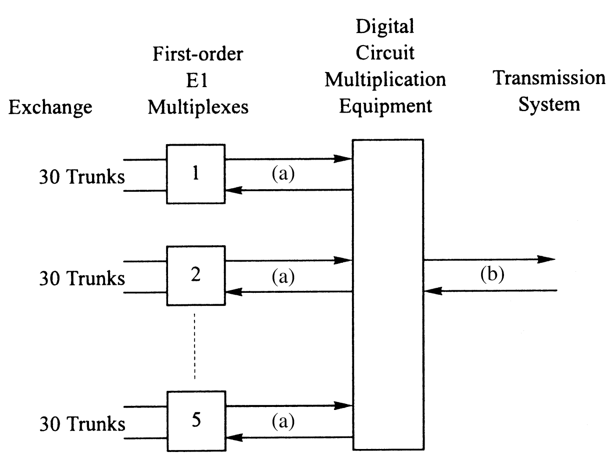

Figure 1.6-6. Digital circuit multiplication equipment. (a), 2048 kb/s E1 line (30 trunks); (b), 2048 kb/s E1 line (60 bearer channels).

multiplexes) are concentrated to 60 ADPCM bearer channels and carried on one E1 multiplex line.

1.7 EXCHANGES

1.7.1 Exchange Equipment

The primary function of an exchange is to establish and release temporary paths between two subscriber lines, between a subscriber line and a trunk, or between two trunks [4].

The two major equipment units in an exchange are shown in Fig. 1.7-1. The *switchblock* (or *switch fabric*) has a large number (up to some 100,000) of *ports* (P) to which subscriber lines and trunks are attached. The ports are interconnected by arrays of switches. By closing or opening appropriate sets of switches, paths across the switchblock can be set up or released. At any point in time, a multitude of these paths can be in existence. In Fig. 1.7-1, path 1 connects a line and a trunk, path 2 connects two lines, and path 3 connects two trunks.

The switchblock paths are set up and released on command from the *control equipment* (CE) of the exchange. CE sends commands to, and receives responses from, the switchblock via a *control channel* (CC) (shown as a dashed line).

1.7.2 Switchblocks

Analog Switchblocks. Until about 1975, all exchanges had analog switchblocks, implemented with electromechanical switching devices. In local exchanges, the switchblocks were two-wire (Fig. 1.7-2). The temporary paths in the switchblock, and the circuits attached to ports P, were two-wire bidirectional analog voiceband (300–3400 Hz) circuits (a).

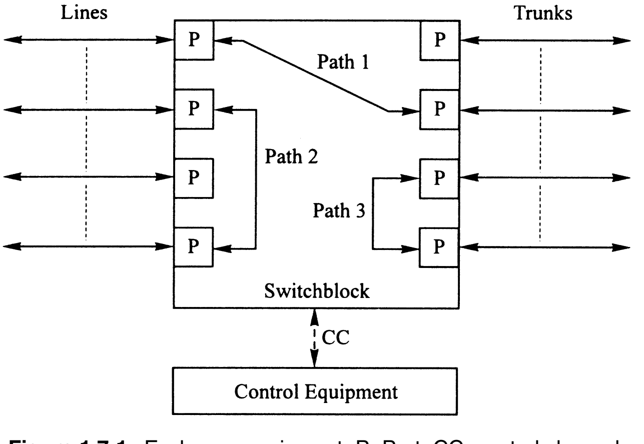

Figure 1.7-1. Exchange equipment. P, Port; CC, control channel.

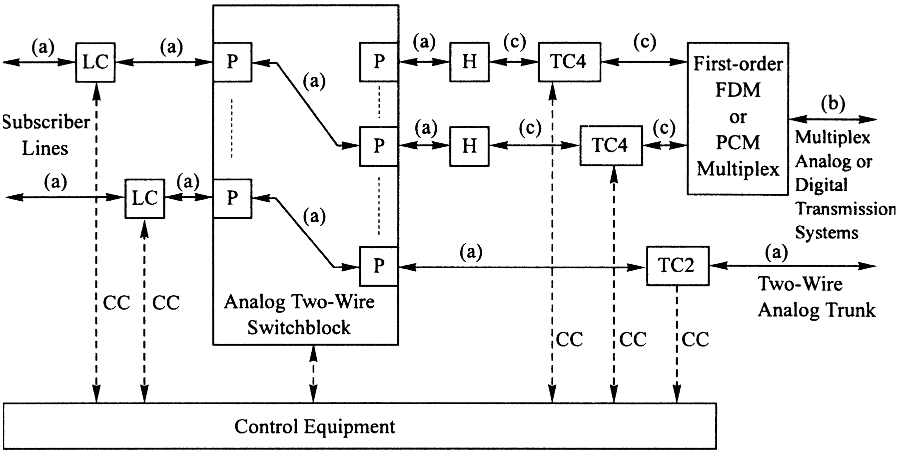

Figure 1.7-2. Local exchange with two-wire analog switchblock. P, Port; LC, line circuit; H, hybrid circuit; TC2, analog two-wire trunk circuit; TC4, analog four-wire trunk circuit; CC, control channel.

Subscriber lines and two-wire analog trunks (a) were attached to the ports of the switchblock via, respectively, two-wire *line circuits* (LCs) and two-wire trunk circuits (TC2). Multiplexed analog and PCM trunks (b) were converted, by first-order FDM and PCM multiplexes, to individual four-wire analog circuits (c). These circuits then passed through four-wire analog trunk circuits (TC4) and were converted by hybrids H into bidirectional analog two-wire circuits (a).

The switchblock and the line and trunk circuits were controlled by the control equipment of the exchange, via control channels (CCs). The line and trunk circuits play a role in the transmission and reception of signaling information (see Chapters 3 and 4).

The analog switchblocks in intermediate exchanges (tandem exchanges, toll exchanges, and international switching centers) provided four-wire analog paths between four-wire ports. That eliminated the need for hybrid circuits on multiplexed FDM and PCM trunks. Hybrid circuits are required on two-wire analog trunks, but the trunks attached to intermediate exchanges are predominantly four-wire trunks.

Digital Switchblocks. The introduction of digital time-division multiplex transmission systems has led to the development of exchanges with digital switchblocks. Most exchanges installed after 1980 have digital switchblocks that are implemented with integrated semiconductor circuits. These switchblocks are more compact, and more reliable, than their analog predecessors.

As shown in Fig. 1.7-3, the ports on digital switchblocks are usally four-wire first-order time-division digital multiplex ports (DMPs). The frame formats at points (b) are as shown in Fig. 1.5-4. The DMPs of exchanges in the United States have the DSI format (for $m = 24$ circuits). In most other countries, the DMPs have the E1 frame format (for $m = 30$ circuits).

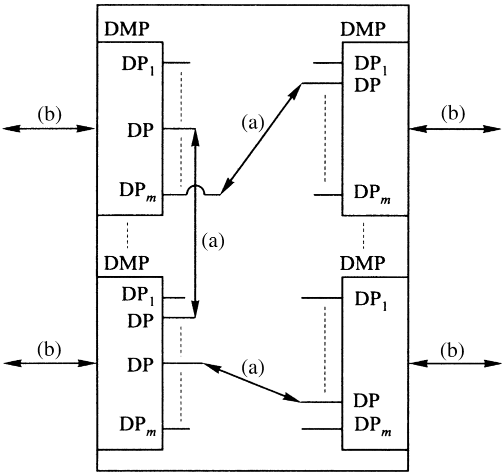

Figure 1.7-3. Paths in digital switchblock. (a), 64 kb/s four-wire paths; (b), first-order digital multiplex circuits (*m* channels); DMP, digital multiplex port.

The paths in the switchblock are 64 kb/s, four-wire circuits (a), The *m* multiplexed circuits entering a digital multiplex port DMP are segregated into *m* individual 64-kb/s digital ports (DPs), and the switchblock paths connect pairs of these ports.

Attachment of Lines and Trunks. (See Fig. 1.7-4.) T1 or E1 transmission systems for digital trunks (b) can be attached directly to the DMPs. Each DMP has a control channel (CC), on which the control equipment of the exchange sends and receives signaling information for the *m* multiplexed channels.

Subscriber lines arrive at the exchange as two-wire analog circuits (c). They pass through two-wire line circuits (LCs) and hybrid circuits (H) and enter first-order PCM multiplexers (Section 1.5.2). The outputs of the multiplexers are four-wire digital multiplexed circuits (b) that serve *m* lines and are connected to DMPs.

Each line circuit also has a control channel (CC) to the control equipment of the exchange.

Analog trunks arriving on FDM transmission systems are first demultiplexed by FDM multiplexes (Section 1.4.5) into individual four-wire analog circuits (d). These circuits then pass four-wire analog trunk circuits (TC4) and enter PCM multiplexes, where they are converted to a first-order digital multiplex circuit (b).

1.7.3 Control Equipment

The control equipment [1,4] of early exchanges was implemented with electromechanical devices. Around 1955, it became clear that digital computers could be used to control exchanges. The speed and sophistication of computers at that time already greatly exceeded the capabilities of electromechanical exchange control

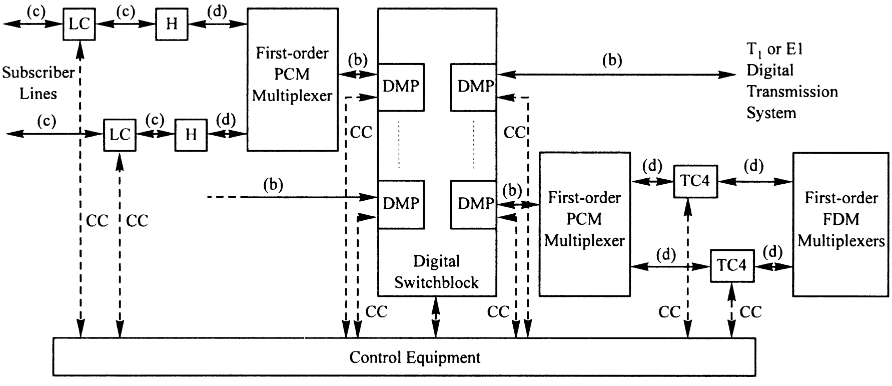

Figure 1.7-4. Exchange with digital switchblock. LC, Line circuit; H, hybrid circuit; DMP, digital multiplex port; TC4, analog four-wire trunk circuit; CC, control channel.

equipment. The first *stored-program controlled* (SPC) exchange was placed in commercial operation in 1965 [1].

The main elements of a stored-program exchange control system are shown in Fig. 1.7-5. A processor (or a group of processors) performs all exchange actions, controlling the switchblock, the line and trunk circuits, and other exchange equipment, by executing instructions in its program. The control channels for these equipment units are connected to a bus on the processor.

Processor memory is divided into semipermanent memory and temporary memory. The semipermanent memory stores the programs for call processing, charging, and exchange maintenance procedures, and tables with data for digit analysis, call routing and charging, and so on.

Temporary memory stores data that are changed frequently: for instance, information on the status of the subscriber lines and trunks, or records of currently existing calls.

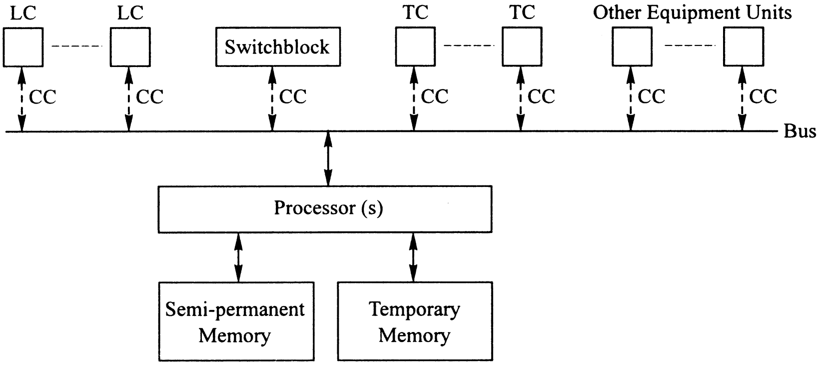

Figure 1.7-5. Stored-program exchange control. LC, Line circuits; TC, trunk circuits; CC, control channels.

The increased logical power of SPC exchanges has allowed the inclusion of features that were not available in earlier exchanges. These fall into two main categories:

- New services for subscribers (call forwarding, call waiting, three-way calling, etc.).
- Procedures to facilitate the administration, operation, and maintenance of the exchange by the telecom. These include routine checks on the exchange equipment, periodic testing of attached lines and trunks, the generation of data on the traffic handled by the exchange and its trunk groups, and the production of equipment trouble reports.

1.8 ACCESS NETWORKS AND LINE CONCENTRATORS

An *access network* (AN) is the part of the telephone network that connects the subscriber premises to the *local exchange* (LE). Initially, access networks consisted of subscriber lines (copper wire pairs) that terminated at the individual subscriber premises on one side and at the local exchange on the other side. Because of the cost involved in deploying the massive volume of copper wires needed to connect all local users to the LE, *remote line concentrators* (RLCs), sometimes known as *pair gain systems* or *loop carriers*, are often interposed between the subscriber premises and the LE. Concentrators, located close to the subscriber premises, collect a large number of line loops and reduce the volume of wires that go all the way from subscribers to the LE.

Various techniques are used for concentration, all taking advantage of the fact that not all subscribers are involved in a call at the same time. The earliest types of RLCs were simple devices that served N subscriber lines with a pool of M wire pairs ($M < N$) between the concentrator and the LE, assigning an idle pair from the pool to a subscriber for the duration of a call and then returning the pair to the pool at the end of the call (Fig. 1.8-l).

The telephone traffic for a residential line during the busy hour has been traditionally around 10% or 0.1 erlangs (although dial-up Internet service has tended to raise average holding times), and the concentration ratio N:M is typically around 4:1.

The more recent line concentrators are called *digital line concentrators* or *digital loop carriers* (DLCs), which multiplex the calls of N subscribers onto a digital multichannel link, such as a T1 ($M = 24$) or E1 ($M = 30$) line connecting the concentrator to the LE (Fig. 1.8-1(a)). Each call is assigned to a channel (time slot—TS). With this technique, two types of multiplexing are possible:

- Fixed assignment of time slots to user interfaces
- Dynamic (statistical) assignment of time slots to user interfaces

With fixed assignment, the number of channels on the digital link is equal to or greater than the number of line interfaces ($M \geq N$), and there is no blocking since each user interface is permanently connected to a channel. The “pair gain” is due

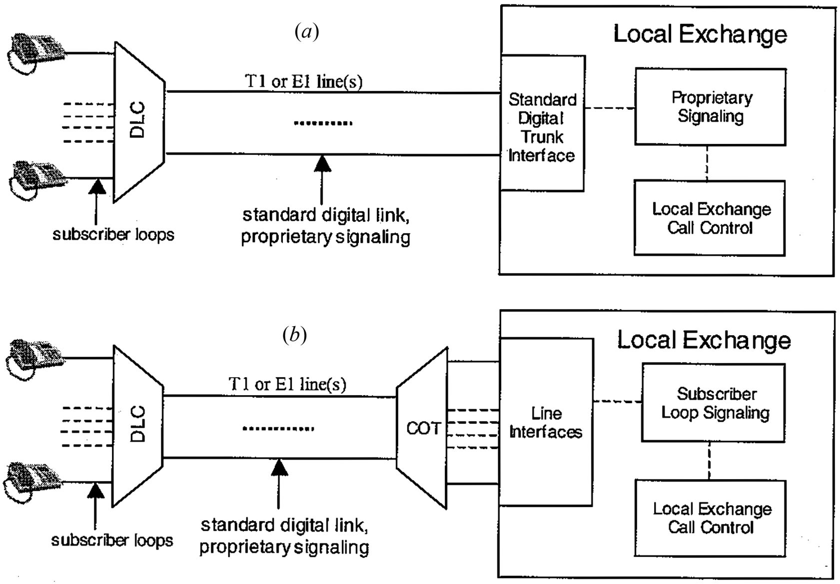

Figure 1.8-1. Digital line concentrators.

to the fact that one physical link (the T1 or E1 line) is carrying the traffic of 24 or 30 concurrent connections. With statistical assignment, fewer TSs are available than the number of line interfaces ($M < N$) and further savings in facilities are achieved through concentration. The drawback is that blocking is possible in the case of unusually high traffic. Multiple E1/T1 lines may be used on one RLC/DLC interface for reliability or if the traffic warrants it.

At the local exchange, the E1/Tl lines can be connected directly to a digital trunk interface, or to a demultiplexer that recreates the subscriber line interfaces as they would appear if the loops had gone all the way from the subscriber premises to the LE. The demultiplexer is known in North America as a *central office terminal* (COT—Fig. 1.8-1(*b*)).

Signaling is required between the concentrator and the LE or COT to handle DLC functions. In the past, each equipment supplier provided its own proprietary signaling protocols, and the COT was the only way to ensure that a DLC would interwork with any type of local exchange, regardless of the supplier. The situation started to change in the 1980s with standardization efforts to “open” the interface between DLC and LE and to eliminate the need for the COT. The new types of DLCs are known as *access systems*, a term that is associated with a standard digital trunk interface and a standard signaling protocol (Fig. 1.8-2). Signaling for access systems is described in Chapter 6.

Access systems have led to a local exchange configuration where the LE retains the call control function for lines but has no physical line interfaces. All physical interfaces are standard TDM digital trunks and the physical interface to local loops is handled by

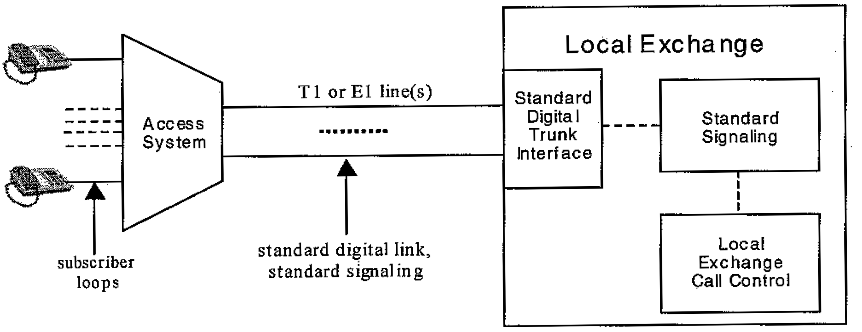

Figure 1.8-2. Access systems.

access systems. Functionally, access systems can be viewed as open-interface line units, which can be deployed remotely or collocated with the LE (when loop lengths are very short), and which can be procured independently of the LE.

1.9 ACRONYMS

A/D	Analog-to-digital
AC	Area code, adaptive circuit
ADPCM	Adaptive differential pulse code modulation
ALG	Access line group
AN	Access network
CC	Country code, control channel
CE	Control equipment
CIC	Carrier identification code
COT	Central office terminal
D/A	Digital-to-analog
DLC	Digital line concentrator, digital loop carrier
DMP	Digital multiplexer port
DPSK	Differential phase-shift keying
DS1	North American first-order pulse code modulation multiplex
EC	Exchange code, echo canceler
ES	Echo suppressor
ETSI	European Telecommunications Standards Institute
FDM	Frequency-division multiplexing
FSK	Frequency-shift keying
H	Hybrid circuit
IC	Interexchange carrier, identification code
IXC	Interexchange carrier
IN	International number
ISC	International switching center

ISDN	Integrated Services Digital Network
ITU	International Telecommunication Union
ITU-T	Telecommunication Standardization Sector of ITU
LATA	Local access and transport area
LC	Line circuit
LE	Line equipment, local exchange
LEC	Local exchange carrier
LN	Line number
MP	Multiplex port
MSC	Mobile switching center
NANP	North American numbering plan
NN	National number
NPA	Numbering plan area
OC	Optical carrier
P	Single port
PBX	Private branch exchange
PCM	Pulse code modulation
PSTN	Public switched telecommunications network
R	receive circuit of four-wire circuit
RF	Radiofrequency
RLC	Remote line concentrator
S	Subscriber, send circuit of four-wire circuit
SCN	Switched-circuit network
SDH	Synchronous Digital Hierarchy
SL	Subscriber line
SN	Subscriber number
SONET	Synchronous Optical Network
SPC	Stored-program controlled
STM	Synchronous transport module
STS	Synchronous transport signal
T	Trunk
TASI	Time assignment speech interpolation
TC	Analog trunk circuit
TC2	Analog two-wire trunk circuit
TC4	Analog four-wire trunk circuit
TDM	Time-division multiplexing
TG	Trunk group
VPN	Virtual private network

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