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Fixed Telephone Networks

This book starts its story from telephone networks, since the telephone network was the one and only globally available wide area communications network approximately over a hundred years. In the first half of the twentieth century, the communications systems supported analog voice with telephony and text with telegraph and telex systems. The era of data communications did not begin until after the Second World War, when the development of electronics and computer technology took giant steps decade by decade. At first, remote data traffic could only be transported over the telephone network with help of modems, but before long fully digital leased lines were also being used. The technology used for such leased data lines was originally developed for telephone network trunks to transport voice traffic converted into digital form. The first digital voice encoding method, **pulse code modulation (PCM)**, had a profound effect on the design of the telephony network and transmission systems for which the basic bitrate 64 kbps equals the PCM codec bitrate.

Unless the reader is already familiar with basic telecommunications concepts such as amplitude, digital systems, encoding, modulation, protocols, frames, and so on, it is recommended to at first study *Online Appendix A*, which provides an introduction to various methods commonly used in telecommunications systems. The text within the book expects the reader to know those methods and also makes references to the Appendix, as appropriate.

1.1 Telephone Network

1.1.1 Analog and Digital Representation of Voice

Voice is the vibration of air caused by speech. The human ear detects sound waves since they make the eardrum vibrate by the alternating air pressure inside the ear. Figure 1.1 is a representation of a sample voice waveform, where time flows from left to right and the vertical peaks represent the intensity of the air pressure.

Telephones reproduce voice by converting this waveform, as captured from air, to an electric format with a microphone and then back to sound waves with a loudspeaker. The early telephone networks transported the analog electric waveform as such between the caller and the callee. The drawback of this scheme was rather low voice quality due to different sources of distortion, such as noise or echo, impacting the analog signal in the network. This problem was eventually solved by converting the analog waveform to a digital format, as transporting digital signals over long distances without errors became possible by the progress of electronics.

Pulse code modulation (PCM) was the first widely deployed method to convert voice into digital format in telephone networks. The PCM method globally used for the circuit switched telephony is specified in ITU-T Telecommunication Standard Recommendation G.711 [1]. Pulse code modulation works as follows [2]:

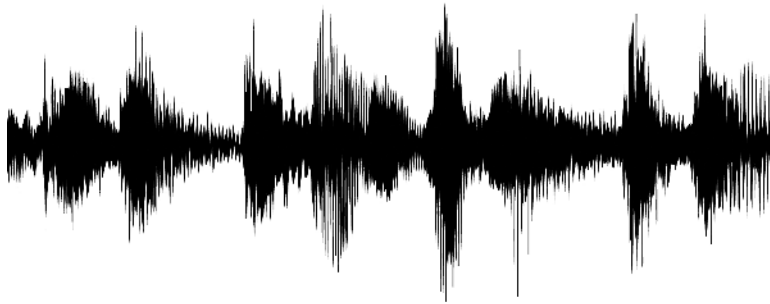


Figure 1.1 Time representation of voice waveform.

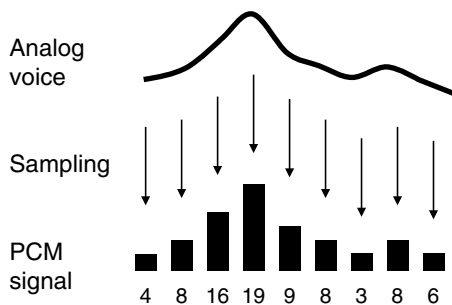


Figure 1.2 Concept of PCM sampling.

- Voice waves are converted with a microphone to an electrical analog signal, which has the same waveform as the sound waves had in the air.
- The amplitude of the electrical waveform is measured in regular short intervals. This **sampling** process is depicted in Figure 1.2.
- Each measured value of the amplitude is represented as a binary number.
- The sequence of the generated binary numbers is transported to the remote end. The numbers are sent with the same frequency as used for taking samples. When the numbers arrive to the receiver in a timely manner with constant latency, the receiver is able to reconstruct an analog electrical signal from these samples. By

interpolating the waveform between the samples, the receiver can produce a waveform rather close to the one which was created by the microphone.

- The reconstructed electrical waveform is fed to a loudspeaker, which converts the signal to acoustic sound waves.

To convert an analog waveform to digital form accurately enough, the amplitude of the wave shall be sampled with a frequency at least twice as high as the maximum frequency of the analog signal. Majority of the power in a voice signal is conveyed within a sound frequency range between 400 and 3400 Hz, which was agreed to be the frequency range to be represented with PCM. Even if the low and high audio frequencies outside of this range would be filtered away, the reproduced speech can be well understood and the unique voice of the person speaking was also recognized. This fundamental agreement behind PCM is the reason why the highest voice frequencies transported over the fixed telephone network are below 4000 Hz. Consequently, PCM sampling frequency is 8000 Hz, which means samples are taken 8000 times per second and the interval between two samples is 125 microseconds.

The 8000 Hz sampling frequency is used with all the narrowband telephone network implementations around the world. Regardless of what the data rate of a digital link is, the frames of the telephone network always take 125 microseconds. On higher data rates, bigger frames are able to transport multiple voice channels simultaneously.

In the digital telephone network, the PCM modulation represents the value of measured amplitude as a binary number of 8 bits. Such a number may have 256 discrete values. As the actual value of the measured analog amplitude can be arbitrarily high or low, the PCM standards must specify how the available 256 values are mapped to a continuous range of amplitude value. This process is called **quantization**. One possibility would be to define the maximum and minimum values of the amplitude, which can be represented, and thereafter divide the range between those two limits to 256 equal parts, called **quantization intervals**. A measurement value which falls into one of the intervals is represented by the single discrete number assigned to that

interval. Quantization introduces a rounding error, as every interval represented by a single number is in reality a range of amplitude values within the interval. In PCM, the difference between the actual sampled amplitude value and the amplitude represented as a single point defined by the interval number is called **quantization error**.

To divide the full range of voice waveform amplitudes to 256 equally spaced intervals is not a good approach. Its drawback is that on the intervals used for smallest amplitudes, the proportional quantization error would be largest, as measured in percentages of the actual value. As a consequence, quiet voices would be distorted most by the quantization process, when the loud voices would be represented reasonably well. While the human ear is most sensitive on quiet rather than loud sounds, the perceived voice quality would be low.

A better solution is to divide the range to quantization intervals or segments using logarithmic scale so that when the amplitude grows, the width of the interval also proportionally grows. This process is known as **companding**, with which the voice signal is compressed before quantization and expanded after decoding at the remote end. ITU-T has specified two companding schemes: A-law is used in European markets and μ -law used in North America and Japan. In practice, the companding approach can be approximated by splitting the whole range of amplitude to a few linear segments of different sizes and thereafter dividing those equally to smaller intervals, as shown in Figure 1.3.

For A-law, the eight PCM coding bits of a **PCM code word** are used as follows:

- Bit 1 means the polarity of the sample either as positive or negative.
- Bits 2–3 mean the linear segment within which the sample is located.
- Bits 5–8 mean the value of the sample within its segment.

The PCM circuits at first compress the sampled values for logarithmic scaling and thereafter apply the analog-to-digital conversion for producing the code words. The data rate of PCM coded signal is constant 64 kbps, as the sampling produces 8-bit code words for each of the 8000 samples taken in a second. This data rate is the basic rate of all digital fixed telephone networks.

1.1.2 Telephone Network Elements

The traditional fixed telephone network is a system which allows two persons to set up a voice call between each other, regardless of the location of their telephones. The network consists of telephones as voice terminals, exchanges or switches for connecting the calls, and links between all those devices to transmit voice and signaling necessary for call setup and release [3]. The endpoints of the call are identified with telephone numbers, each of them uniquely identifying one single terminal within the global telephone network.

Two terms are widely used for traditional fixed telephony:

- **Public switched telephone network (PSTN)** means the network itself.
- **Plain old telephone service (POTS)** means the telephony service provided by the fixed telephone network over a subscriber line to an analog telephone device.

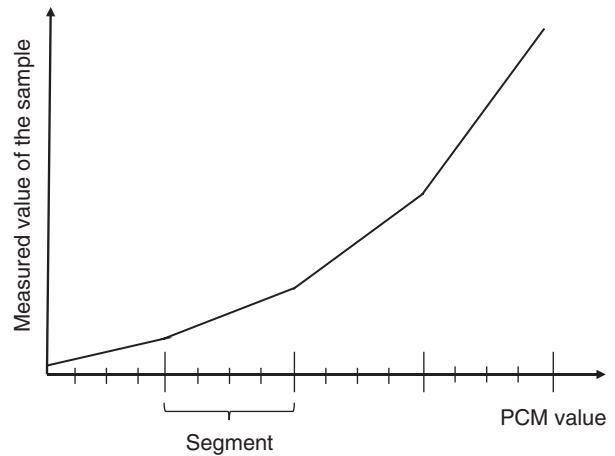


Figure 1.3 Operation of PCM quantization.

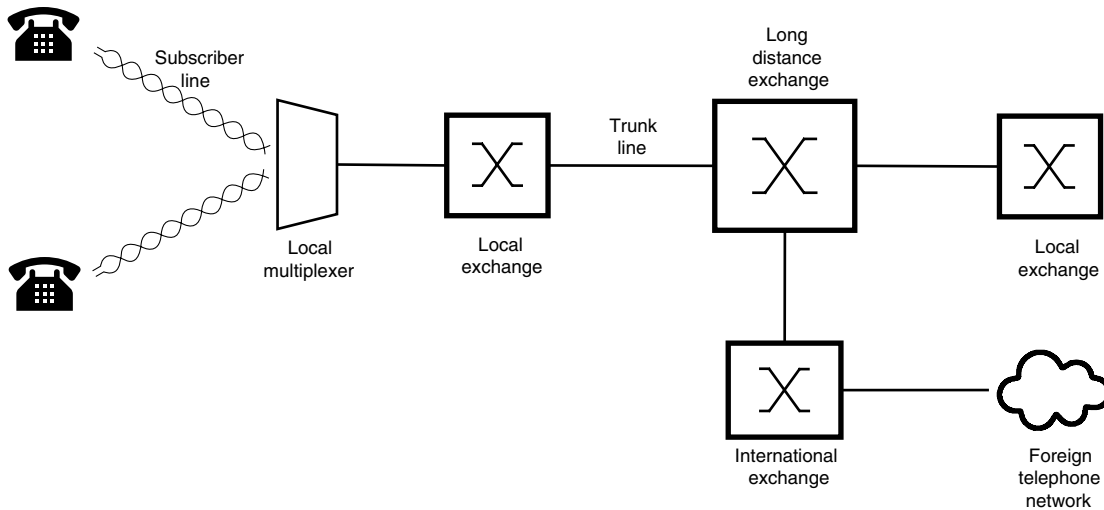


Figure 1.4 Telephone network structure.

The PSTN network consists of the following types of elements:

- Analog **telephones** at the subscriber premises
- **Subscriber lines** as a twisted pair of copper wires, connecting telephones to local multiplexers or in some cases directly to local exchanges
- **Local multiplexers** to terminate subscriber lines and to multiplex calls from them to trunk lines toward exchanges
- **Local telephone exchanges**, which connect calls within a region such as a town or city
- **Long-distance exchanges** connecting calls between different regions
- **International exchanges** to connect international calls
- **Trunk network lines** between exchanges and multiplexers

As can be seen from Figure 1.4, the global telephone network is a strongly hierarchical system. Logically, the telephone network can be divided into the access network and core network. The access network consists of subscriber lines and multiplexers. Its topology is a tree. The core network consists of exchanges, connected with trunks to form a mesh (or tree) network.

1.1.3 Evolution of the Fixed Telephony Network from Analog to Digital

Until the 1960s, the telephone signals were completely analog end-to-end on all the links and interconnected devices. All the related hardware (cables, exchanges, and multiplexers) just contributed to creating and maintaining the physical electric circuits between the two telephones which were engaged to a call and to amplifying the signal to support long-distance calls. The only exception were the trunks, which used frequency division multiplexing. For those trunks, the fundamental frequency of the analog voice signal was transposed to an available frequency division multiplexing (FDM) subband. Nevertheless, the nature of the signal was always analog at every part of the end-to-end circuit. Only the development of electronics and mass production of inexpensive components like transistors made it possible to start replacing analog transmission systems with more efficient **digital transmission systems** [4].

The evolution of the fixed telephone network from the 1800s to modern times took place over the following steps:

- The first subscriber lines had only a single iron wire so that the circuit consisted of that wire and the ground as the “return wire.” Such a connection was very sensitive to external disturbances and the quality of the sound was awkward, except for very short connections. The first telephones were directly connected between each other, but when the number of phones grew, it was soon found that exchanges were needed to keep the number of lines reasonable.
- The earliest telephone exchanges introduced during the 1870s were operated manually. When someone wanted to initiate a call, the caller had to rotate a handle of the telephone to generate an electric signal toward the exchange. At the exchange, an indicator of the calling telephony line changed its position and a tone was produced to notify the operator to check which line had initiated the call attempt. After locating the line, based on its indicator, the operator connected her own phone to that line and asked the caller the identity of the callee. The electricity needed for the call was provided by the exchange to the connected circuit. Thereafter, the operator called the callee with a similar procedure to notify the callee to pick up the phone, which started ringing when receiving electric current from the exchange. When both the caller and callee had their phones ready for the conversation, the operator eventually interconnected the two lines to join the two originally independent calls. The operator then regularly checked from the line if the conversation was still going on. When no voices could be heard, the operator deemed the call to have ended and disconnected the lines. Later on, it became possible for the caller to notify the exchange about the call end by rotating the handle of the phone just like when initiating a call. With such manually operated exchanges, telephony was mainly a local service. The exchange was able to connect calls between phones directly connected to the local exchange, which was thus called the **central office** [2].
- In the 1880s, it was found that using two insulated wires for the subscriber line significantly reduced cross talk and increased the quality of the connection. A second improvement was to replace iron with copper for the wires. Compared to iron, copper had better conductive properties and was not subject to corrosion. Eventually it was found that twisting the copper wire pair made the wire more resistant to noise and decreased attenuation as the inductance between the wires grew. Progress was also made for exchanges, as the first electro-mechanical exchanges were introduced to replace the manual exchanges. Long-distance calls were introduced, but setting up such a call over multiple exchanges was still a time-consuming process. The caller had to wait for the operator to call back after the callee has been reached and all the necessary connections had been created for the call.
- In the early 1900s, the local exchanges were interconnected with automated long-distance exchanges and trunk lines. In the United States, the first transcontinental call, between New York and San Francisco, was placed in 1915. The structure and the operation of trunk lines in those days were similar to the subscriber lines. The trunk line was just a bunch of twisted copper wires, insulated with paper inside a protective rubber cover. When the distances were long enough, analog amplifiers were installed between trunk cables to keep the level of the signal good enough. Most typically, a single trunk line had two pairs of wires so that there was one wire pair to each of the directions. In that way, the amplifier could be unidirectional. If the city had multiple local exchanges, they were also interconnected with trunk lines. Since the number of calls between exchanges were just a small fraction of the number of local calls within an exchange, relatively few trunk lines were needed.
- Since installation of long trunk lines was expensive, creative new solutions were invented to transport multiple simultaneous calls over one cable. The bandwidth of a single analog voice signal over a telephone network is approximately 3 kHz. An analog trunk line made of copper provides at least 36 kHz of effective bandwidth. During the early 1900s, a method was used to modulate each separate voice signal on the trunk line with carriers separated from each other at 4 kHz intervals. In this way, one wire pair was able to transport 12

frequency-multiplexed calls on parallel. When using a coaxial cable instead of a twisted pair, the number of simultaneous calls could be increased even more, as the bandwidth supported by coaxial cable is much higher than that of a twisted pair.

- At the 1930s, the means to convert an analog voice signal into digital form with pulse code modulation (PCM) was invented. The invention was used to improve the quality of voice over long transmission links. At the time when the invention was made, the electronics that could be used for digitalization were so expensive that no wide PCM deployments were economically viable.
- At the 1950s, deployment of the automated exchanges was continued. Direct long-distance calling without operator intervention became possible in various regions in this period. Compared to the earlier manual approaches, direct calling significantly speeded up call connection time.
- At the 1960s, the development of new electronics and the falling prices of electronics made it finally feasible to introduce devices capable of multiplexing a number of voice signals into a single cable with PCM. By changing the multiplexing method on trunk lines from frequency multiplexing of analog signals to time division multiplexing of digital PCM signals, it was possible to increase the number of simultaneous calls over twisted pair from 12 to 24 or even 30. Such digital links were at first used on trunks between the long-distance exchanges but later on also between local exchanges. The digital signal was created and terminated at both ends of the trunk before connecting the individual calls to exchanges as analog signals in the traditional manner. Direct international and intercontinental calls between operators were also introduced in this period.
- A few years later, the digital line deployment reached also the trunks between exchanges and local multiplexers. The analog subscriber lines were terminated at the multiplexer, which took care of the pulse code modulation and passed the PCM signals toward the trunk. Consequently, the number of wires between exchanges and multiplexers could be reduced. Later on, coaxial cables replaced copper wire pairs on trunks.
- In the 1970s, the first digital exchanges were introduced for small towns, where the needed switching capacity was small. In those exchanges, the relays used by the old exchanges were replaced by microchips and signal buses. The digital switches were able to connect voice signals in digital PCM form between trunks used toward multiplexers and other exchanges, so the need to convert signal back to analog form for the exchange was avoided. Additionally, the digital exchanges were able to automatically process the digitally transported information about the state of the connected telephones and calls, such as the callee number and busy status.
- In the 1980s, the capacity of digital switches grew and the rate of increasing old analog exchanges with newer digital ones was increased. At the end of the decade, it became possible to transport the voice or data in digital form from the telephone device to the local multiplexer. To support that, the old analog telephone could be replaced with a newer ISDN phone capable of performing PCM process already at the customer premises rather than at the multiplexer. ISDN subscriber lines were able to support two voice channels and one data channel (2B+D). ISDN also supported using the voice channels to transport data. As the exchanges grew, the transmission capacity requirements increased between the exchanges. The capacity of trunk lines was increased at first with PDH trunks capable of multiplexing signals in a hierarchical manner and later on with optical SDH networks, to achieve even higher bitrates and direct access to sub-signals for demultiplexing.
- In the 1990s, the fixed telephone networks reached their peak as the cellular mobile phone technologies started to take off. During the first two decades of the twenty-first century, the mobile phones have nearly completely replaced fixed telephones so that we are currently living in the era of mobile telephony. Still, the telephone core networks continued to serve mobile phones, but their architecture was gradually upgraded toward packet switched rather than traditional circuit switched protocols. The twisted pair subscriber lines continued their life as carriers for DSL data connections while analog telephony was phased out. In enterprise contexts, table phones may still be used, but they typically use Voice over IP (VoIP) data connections instead of the traditional telephone network.

1.1.4 Telephone Numbering

Telephones are identified by their telephone numbers, unique within the network. As explained in *Online Appendix A.8.1*, a telephone number consists of the following parts [5]:

- Country code used for international calls
- Area code or national destination code, which defines the regional network within a country
- Subscriber number, which is unique within the region identified by the area code

When initiating a call in a fixed telephone network, the user has to provide the called number in one of the following ways:

- When calling a phone within the same area with the caller, only the subscriber number is needed.
- When calling a phone in a different area within the same country as the caller, both area code and subscriber number are needed.
- When calling a phone in a different country, all three parts are needed. An international prefix should be supplied before the country code to indicate to the exchange that the call is to another country.

Originally, the subscriber number was fixedly tied to a specific subscriber line. The house or apartment to which the subscriber line was connected owned the number. In the 1980s, when number portability was introduced in intelligent networks (see later this chapter, 1.2.2), this fixed mapping was broken. The semi-permanent relationship between the subscriber number and a subscriber line was recorded to a database rather than hardcoded to the exchange. This allowed the subscriber to keep the number even when moving to another apartment.

When mobile phones were introduced, the numbering system experienced a few changes:

- Providing the area code became necessary when calling from a mobile phone, since the mobile phones did not belong in a fixed way to any area even if they stayed being country-specific.
- Instead of the area code, mobile phones themselves had an operator code. Later on, with number portability, this fixed relationship was broken so that this code did not necessarily identify the serving operator.

1.1.5 Tasks and Roles of Telephone Exchanges

A telephone exchange has three major tasks in the network. First of all, with help of an exchange, the number of lines between all the phones in the network can be dramatically reduced. Instead of building a full mesh network where every pair of phones would have a direct connection between them, it is sufficient to connect a phone only to the exchange. Secondly, the exchange enables two phones to be connected just for the duration of the call so that the subscriber lines and trunk lines are released for other calls when the call is terminated. Since the early exchanges set up physical circuits for calls and released them after the calls were finished, the connections managed by exchanges are known as **circuit-switched** connections. Circuit-switched connections are created and released dynamically in the switching matrix of the exchange, based on control signals received from telephones [4]. Switching matrices of analog exchanges consisted of mechanical switches or relays used to connect and disconnect electrical circuits between the phones engaged to calls. With digital telephone networks, switching matrices become specialized integrated digital circuits, which were able to connect digital channels between each other. Finally, the exchange has, or is able to retrieve, the knowledge of how to route the calls from the caller to the callee, based on the telephone number provided at the call setup phase. Routing decisions may be static or depend on some dynamic conditions like the time of a day, as described in ITU-T E.170 [6]. In addition to transporting the end user traffic over circuits, exchanges collect and process charging data, make various measurements, and take care of timing and synchronization, as specified in ITU-T Q.521 [7].

Telephone exchanges can be categorized as follows:

- Local exchanges provide network connectivity to the subscriber lines within a certain geographical area, such as a town or suburb. The exchange is able to connect calls between phones in that area or forward the call attempts to another exchange, in case the callee resides in another region. In practice, there are local multiplexers between the subscriber lines and exchanges, so that the exchange receives the signals from subscribers within trunk lines from the multiplexers.
- Long-distance exchanges interconnect the local exchanges of different regions. A long-distance exchange may either be a separate standalone exchange or just a specialized function within a local exchange.
- An international telephone exchange interconnects national telephone networks over international trunk lines. An international exchange is connected either to the local or long-distance exchanges of the national networks.

1.1.6 The Subscriber Line

The traditional fixed telephones were analog devices. The microphone of a telephone vibrates from the voice air waves and converts the vibration into an electric signal with similar waveform. The telephone then passes the analog electric signal to the subscriber line leading to an exchange or a local multiplexer. After being carried over the telephone network to the telephone of the callee, the earpiece of the telephone converts the received electrical signal to sound, matching its waveform with the received electrical signal. The traditional telephone network has been designed to pass analog signals of frequencies within human voice with minimal distortion between two telephones.

Subscriber lines behind the telephone connectors on the walls are so called **unshielded twisted pairs (UTP)** of thin copper wires inside plastics insulation. The width of the cross-section of the wire is different in different countries and installations. In the United States, the options are defined within **American wire gauge (AWG)** standard, while in Europe the wires are according to the **international wire gauge (IWG)** standard maintained by the European Telecommunications Standards Institute (ETSI). When an analog voice call is set up, switches are closed at the telephone device and the exchange to form an electric circuit for the call. As this circuit is physically a loop, the subscriber line is also called **local loop** [8].

The subscriber lines were traditionally built (especially before the 1970s) so that a bunch of twisted pair cables were dug underground and run from a local exchange to a local multiplexer located in a cabinet along a street or road. The maximum span of such twisted pairs was a few kilometers. At the exchange, these wires were connected to a **main distribution frame (MDF)**, from where each wire could be connected to a **subscriber line unit** of the exchange or to a data multiplexer serving leased lines. When new houses or apartments were built along with the street, spare wires were taken into use at the local multiplexers and connected to new twisted pair subscriber lines toward the new house or apartment. If a house was torn down and a new house built nearby, in some cases the unused subscriber line was left in place but a new branch connected to it. A joint was created to the middle of the wire at a location from the new wire that could be built to the new house. Such branches or wiretaps did not cause any harm to the analog voice signals below 4 kHz but would cause strong attenuation of certain higher frequencies used later for **digital subscriber line (DSL)** data connections [9].

The subscriber line attenuates the analog voice signal, which can travel in a twisted pair at maximum a few kilometers. If the subscriber premises were still farther away from the closest local multiplexer or exchange, special arrangements were needed to avoid the signal being faded out. A commonly used technique was to build the subscriber line from a chain of wires with different widths. In the United States, the subscriber lines in the rural areas were often extremely long, and loading coils were added to the line with regular intervals to keep the inductance of the line in the right level compared to the capacitance of the line. Such a solution minimized the attenuation on the analog voice frequencies but unfortunately greatly increased the attenuation for higher frequencies. On such lines, the digital subscriber line data technologies do not really work without removing the loading coils.

When relying only on twisted pairs, the local exchanges could not be located too far away from the subscriber premises. In large cities, the telephone operator had to build and maintain a large number of local exchanges. Digital transmission technologies, however, changed the game. A digital PCM trunk line extended the signal reach between a local multiplexer and an exchange to tens of kilometers, which allowed the operator to rely on much fewer local exchanges, still supporting a high number of subscribers. That was beneficial for the operators, as costs for exchange maintenance and rents paid for the central office premises could be decreased. As telephone service has recently been largely moved to mobile phones, the fixed subscriber lines have either been abandoned or reused as carriers of Internet data over DSL connections. With the progress of optical cable technology, operators have started to replace the copper wires leading to customer premises with optical trunk cables, so that the copper wires may be still used between an apartment and a local optical cable terminal in a street cabinet or in buildings' basements.

1.1.7 Telephony Signaling on the Analog Subscriber Line

Figure 1.5 represents an analog telephone system. An analog telephone gets its operating voltage via the subscriber line from the local multiplexer or exchange. Since the analog phone is not connected to a power socket, it continues operating as long as the exchange or the multiplexer is powered, even if the customer premises would otherwise have an electricity power outage. When the phone is unused, its earpiece disconnects the electrical circuit over the subscriber line. The phone is said to be in the **on-hook state**. When the user of the phone lifts the earpiece, the phone goes to **off-hook state** and closes the electrical circuit toward the multiplexer [3]. When the exchange finds out that the circuit is closed, it starts to send a continuous sound signal to the earpiece, informing the user that the exchange is ready for receiving the called number from the phone.

When the user thereafter presses the numeric keys or turns the rotary dialer of the telephone, the phone generates signals to the subscriber line to inform the exchange about the called number. Analog telephones used two different methods for encoding the number:

- Using **pulse dialing**, where the called number was expressed by a sequence of direct current pulses (100 ms) and idle break periods (60 ms) when the circuit was opened. The selected number (0–9) was indicated as the number of pulses transmitted. Old phones generated the pulses with a disc that rotated with help of a string. During its rotation, the disc connected the circuit for every pulse and disconnected it for every break. The newer phones were able to generate similar pulses according to the keypresses of the user.
- Using **dual-tone multifrequency (DTMF)** signal, which consists of two audible frequencies. These frequencies were selected so that they were within the 400–3400 Hz band supported by the telephone network, as shown in Table 1.1. DTMF tones could also be used to provide selections for automatic call processing machines and voice mail systems. The machines requested that the user press a certain key for a specific choice, such as selection between different types of services or the language that the customer wanted to speak. The frequencies used in the dual tones are given in the following table:

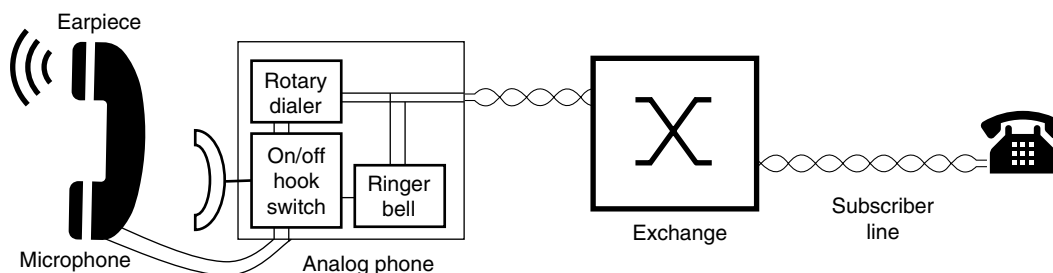


Figure 1.5 Analog phone system.

Table 1.1 DTMF dual frequency plan.

Number	1	2	3	4	5	6	7	8	9	0
Lower tone/Hz	697	697	697	770	770	770	852	852	852	941
Upper tone/Hz	1209	1336	1477	1209	1336	1477	1209	1336	1477	1336

After the called number is chosen, the exchange informs the user about the progress of connecting the call with different types of tones, as specified in ITU-T E.180 [10]. The tones used in a network are specific to the national conventions. Examples of such tones are the following:

- **Ringing tone** to tell that phone in the remote end is ringing. Exchange sends tones in a certain frequency over 0.67–1.5 seconds, after which there is a pause of 3–5 seconds.
- **Busy tone** to tell that the phone in the remote end is engaged to another call. Exchange sends tones followed by a pause, so that the duration of such a complete cycle is between 0.3–1.1 seconds.

To get the phone to start ringing at the callee, the exchange sends it a special **ringing signal**, which the phone connects to a ringer bell. The bell sound alerts the callee about the incoming call. Ringers are of two types, mechanical or electronic. Both types are activated by a 20-hertz, 75-V alternating current generated by the exchange. The ringer is commonly activated with 2-second pulses separated by a pause of 4 seconds.

1.1.8 Trunk Lines

Before digitalization of exchanges and trunks, end-to-end telephone signals were also analog. The analog voice signals were connected by exchanges to analog trunk lines. A single trunk line was able to carry multiple voice signals by using frequency division modulation. The bandwidth of the trunk cable was divided into subbands, each capable of carrying an analog voice signal with 3400 Hz bandwidth. The voice signal was used to modulate a carrier in the middle of the subband.

Since the 1970s, the trunk lines of a fixed telephone network were typically converted to digitally multiplexed links between exchanges and local multiplexers. To use a digital trunk, the multiplexer or the exchange itself terminates analog subscriber lines and multiplexes signals from them to a digital trunk line. PCM is used to encode the voice signal for the trunk. Any trunk line connected to a multiplexer is able to support tens of subscriber lines so that each of those has its own dedicated 64 kbps PCM channel on the trunk toward the telephone exchange. The higher order trunk lines between exchanges are able to support even thousands of simultaneous calls, depending on the type and capacity of the trunk.

The smallest capacity digital trunk lines were structured in slightly different ways in different continents. The basic American trunk was called the T1 link, supporting 24 voice channels and 1.5 Mbps total bit rate. The basic European trunk was called the E1 link, supporting 30 voice channels and 2.048 Mbps total bitrate. As described in Section 1.3.1, a 64 kbps voice channel is carried within a single timeslot of the time division multiplexed frame structures of T1 and E1 signals. As capacity needs increased, the digital technology was evolved to support further multiplexing of T1 and E1 signals to PDH or SDH/Sonet links of higher capacity. The physical media used for the trunk lines was typically twisted pair wire or coaxial cable for the smaller capacity links and coaxial or optical cable for the bigger links. Optical cables were preferred for long link spans. In some cases, microwave radios could be used instead of cable—for instance, when building a cable would be too expensive in difficult terrain. For further details about these digital transmission technologies, please refer to Section 1.3.

In addition to the actual voice channels, a fraction of the link capacity is reserved for the telephone network internal purposes such as transmission of notifications about remote error conditions or forwarding signaling and

network management messages between the network devices. Such messages are processed within a signaling network, which consists of the following types of devices [11]:

- **Service switching point (SSP)**, which fundamentally is a telephone exchange creating and releasing the circuit switched connections between end user terminals.
- **Service control point (SCP)**, which is either a database storing information about subscribers or a centralized server with call processing logic.
- **Signaling transfer points (STP)**, which is a network device used for routing signaling messages between service switching points (SSPs) and service control points (SCPs).

Two different network architectures have been deployed for the transfer of signaling and management messages [8]:

- In the American model, the signaling and network management messages were transported in a dedicated signaling data network, which was separated from the voice network. SSP and STP were two separate types of network equipment.
- In other countries, the signaling and network management messages were sent directly between the telephone exchanges over the trunks used for voice traffic. A small part of the trunk frame structure was reserved for signaling traffic while the rest was used for the voice traffic. The telephone exchanges had both the roles of STP and SSP for signaling.

1.1.9 Telephone Networks and Data Communications

Traditionally, the capacity of trunk lines and exchanges has been dimensioned with the assumption that the length of an average call is between 3 and 5 minutes. When analog modems were placed into wide use during the first Internet boom of the 1990s, this old assumption became invalid. The duration of a data call could extend to hours, which caused overload for both trunk lines and telephone exchanges so that no new calls could be made, as all resources were already busy. To solve the problem, in theory three different approaches could have been used:

- 1) Increase the capacity of telephone exchanges and the interconnecting trunk lines. This solution was neither economically viable nor technically justified. Running packet switched data connections over a circuit switched infrastructure is fundamentally a bad idea. The telephony network lacks the flexibility needed for packet switched data. Telephony network was temporarily used for data connectivity due to the lack of better options, until data-centric network architectures became commonly available.
- 2) Separate the data traffic transported over subscriber lines into a dedicated data network and connect only the voice traffic to the telephone exchange. This solution became common in the early 2000s when asymmetric digital subscriber line (ADSL) and later on very high-speed digital subscriber line (VDSL) technologies were deployed. The basic idea of DSL was simple: the capacity of the subscriber line can be divided into two frequency bands to be used for different purposes. The low frequencies under 3400–4000 Hz were used for analog voice and any higher frequencies for DSL data carriers. In both ends of the subscriber line, there is a splitter which allows a single pair of wires to be connected to two devices, one for voice and another for data. In the customer premises, the devices would be an analog telephone and a DSL data modem. At the local multiplexer, the devices would be a voice multiplexer and a DSL data multiplexer, which would forward the traffic to different trunks and networks dedicated for voice and data.
- 3) Use totally separate networks for data traffic versus the voice traffic, already from customer premises. There are various ways of how to apply this approach to network configurations:
 - Using a cable modem over cable television network infrastructure for data and an analog telephone over the fixed telephone network for voice calls
 - Using DSL over the subscriber line or optical fiber to home for data and mobile telephony for voice calls

1.2 Telephone Exchange and Signaling Systems

1.2.1 Operation and Structure of a Telephone Exchange

A telephone exchange processes call attempts from analog subscriber lines with the following steps:

- 1) Detect a subscriber to have initiated a call attempt.
- 2) Collect the digits of the called telephone number.
- 3) Analyze the called number.
- 4) Route the call signaling to a trunk line toward another exchange or a local subscriber line of the callee.
- 5) Give an indication of an incoming call to the callee.
- 6) Create connections between digital 64 kbps channels (such as E1/T1 timeslots; see Section 1.3.1) used for the call at the switching matrix [4].
- 7) Provide the caller with ringing tone or other announcements depending on the progress of the call attempt.
- 8) Pass through voice media when the call is connected.
- 9) Release the call and collect charging records for it.

The specific functions of a telephone exchange are defined in ITU-T recommendation Q.521 [7]. A modern digital telephone exchange is a large system controlled by special computers or microprocessors embedded into the system. Although the exchange designs are vendor specific, this chapter describes a typical architecture of a digital telephone exchange.

The exchange consists of subracks fitted into a rack as shown in Figure 1.6. Each subrack contains a number of circuit boards or units, each of which has its own specific tasks to server the complete operation of the exchange. The units are able to communicate with each other over the communication channels available on the backplane of the subrack. The subracks are interconnected with cabling, providing data links internal to the exchange. The format of the data transported between the units and subracks may either be some vendor-specific proprietary format or any suitable standard format. Typically, the voice data is transported using the standard T1 and E1 links. In addition to the hardware, the exchange is controlled by a complex software system, which is distributed in one to the different units and the microprocessors running on those.

The parts of an exchange have been dimensioned in such a way that the exchange is capable of supporting simultaneous calls only from a subset of the connected subscriber lines, but not all of them. The capacity of internal links and the switching matrix of the exchange is just a fraction of the number of served subscribers. The capacity of the exchange should be sufficient for supporting the telephone traffic during typical rush hours. In exceptionally busy cases, it is possible that the exchange may run out of capacity so that any new call attempts are simply blocked.

The main functions of an exchange can be divided between units as shown in Figure 1.7:

- The units taking care of subscriber lines, when individual twisted pair lines are connected directly from the MDF to the exchange. The unit terminates the twisted pair lines, converts the analog voice signal to digital PCM signal (and vice versa) and takes care of the analog signaling of the line. With ISDN subscribers, the unit terminates the ISDN 2B+D channels. One unit is typically able to terminate 30–120 subscriber lines. Note that typically, these kinds of units are part of the local multiplexer or concentrator rather than the exchange so that the exchange receives the traffic aggregated from subscriber lines within trunks from the multiplexer or concentrator. When using concentrators, only a subset of the subscriber lines can be connected simultaneously to the trunk toward the exchange.
- The units taking care of trunk lines from local multiplexers, concentrators, or other exchanges. The task of such a unit is to process calls and signaling between the subscribers and the rest of the telephone network. The unit typically processes the channel associated signaling or the lower layer protocols of the common channel

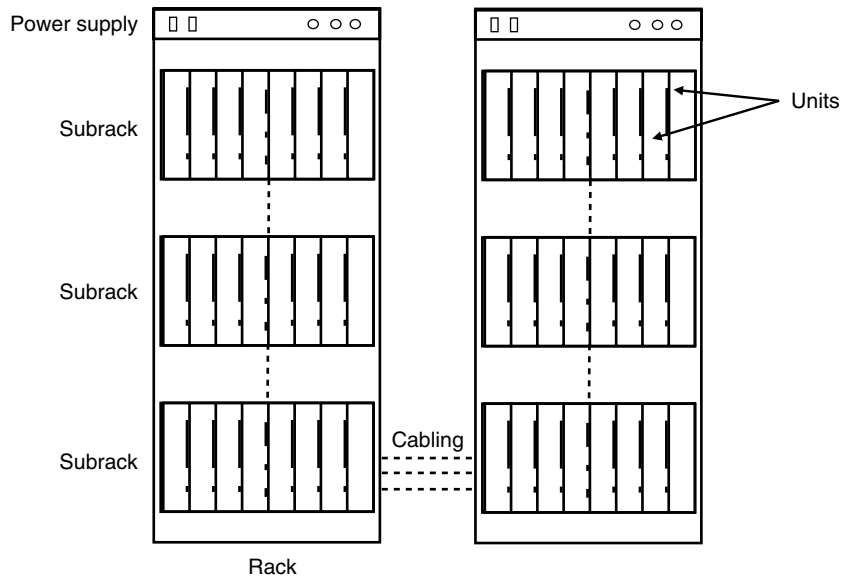


Figure 1.6 Physical structure of a telephone exchange.

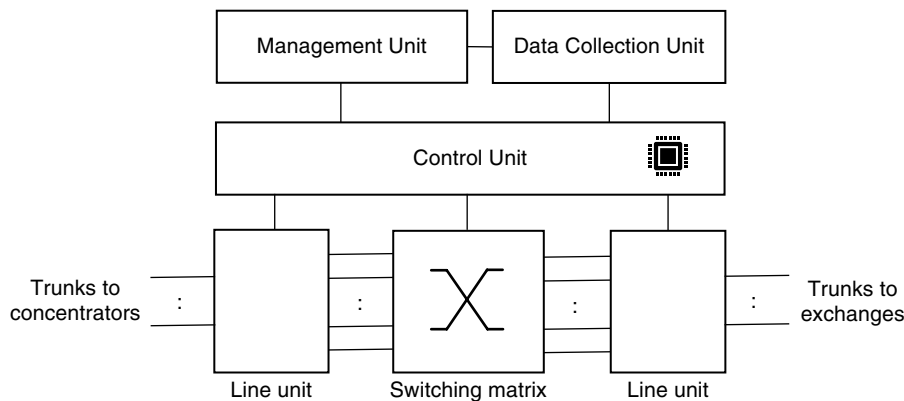


Figure 1.7 Logical structure of a telephone exchange.

signaling protocol stack (see Section 1.2.3). All the 64kbps channels received from other exchanges can be simultaneously connected to the switching matrix of the exchange, while the same may not apply to the trunks received from local multiplexers.

- The cross-connect units hosting switching matrices where the calls are connected between the subscriber lines and/or 64kbps channels of the trunk lines, as controlled by signaling. The switching matrix processes the voice channels in the same way, whether it gets them from a subscriber line unit or trunk line unit. Between the units, the signals are typically transported as T1 or E1 frames. When connecting a voice channel from a unit to another unit, the switching matrix does switching in two dimensions, space and time, as shown in Figure 1.8:
 - Space: to connect a timeslot between line units
 - Time: to rearrange the connected timeslots within a T1/E1 frame toward a line unit so that the timeslot is connected to the correct subscriber line at the line unit where the mapping between timeslots and subscriber lines is fixed

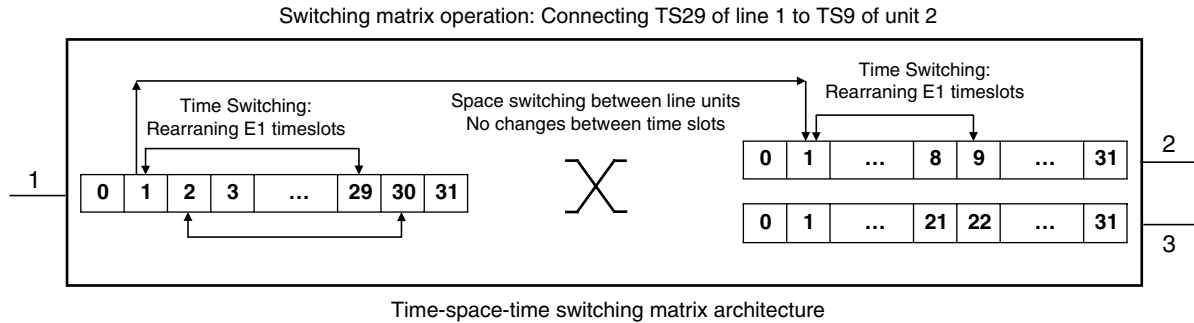


Figure 1.8 Operation of the switching matrix in time and space dimensions [8].

- The control unit, which processes the signaling data and controls the switching done on the switching matrix. The control unit may either be a separate unit or a function of the cross-connect unit in smaller exchanges. The control unit tells the switching matrix how the time slots of different line units should be connected. The control unit drives creating and releasing connections according to the signaling used to set up and clear calls. The control unit may also request the line units to perform line testing and will pass the results within signaling messages to other exchanges. The control unit may also take care of routing the signaling messages and the maintenance of the routing tables.
- The subscriber database unit is responsible for connecting the exchange to a separate database server. The database is used to help in routing calls to other exchanges and in taking care of different exceptional situations.
- The management unit of the telephone exchange. This unit monitors the whole telephone exchange, informs the network management system about fault conditions, and takes care of actions needed to recover from different types of faults. The management unit may also support a management console and a log printer.
- The data collection unit collects usage statistics needed for subscriber charging and network maintenance purposes. For each call, data is collected for the billing system. The unit provides statistics to support network planning and maintenance. From such statistics, network operators may find out that average load has increased to such a level that additional capacity would be needed.

1.2.2 Intelligent Networks

The earliest digital exchanges contained all the logic, databases, and programs needed for processing and connecting calls. During the 1980s, the technology was evolved toward distributed architectures where the logic, data, and connection processing were split to different pieces of equipment. The evolution was driven by the need to change and extend the call handling procedures due to the following reasons:

- The competitive situation between incumbent and challenger operators made the operators demand possibilities for quicker development of operator-specific services. Part of the call control logic was moved from the exchange to a separate computer system so that the operator was able to modify the logic and add new additional services used for service differentiation, such as
 - Directing calls toward a single nationwide hotline number to different call centers, depending on the time of a day.
 - Setting up a virtual private telephone network between different sites of an enterprise. Even if the calls would be connected over the public telephone network, the enterprise can define its own telephone number space for internal calls. Within a virtual network, the numbers can be freely distributed between different sites, regardless of the physical location of the sites and the related area codes of the public telephone network.

- Public authorities began to require operators to provide new regulated services, such as number portability. Number portability means that when a subscriber moves from one apartment to another, the telephone number provided by the fixed telephone network still stays the same for the subscriber. Consequently, the operators can no longer encode geographical areas such as suburbs into the telephone numbers. Instead, the mapping between numbers and individual subscriber lines must be defined in a separate subscriber database. When connecting calls, the exchange shall check this mapping from the database to route the call to the right destination.
- If the call control logic and related databases would be kept within the exchanges, any updates to the logic should be done simultaneously to all the exchanges of the network. But if the logic and databases are separated from exchanges, a single database server may be sufficient for the whole network and all its exchanges. The software and database updates would impact only the separate server, reducing the amount of maintenance effort needed and risks for having discrepancies in data stored in multiple locations. Naturally, for increased safety and reliability, the server should be duplicated so that if the server or its software upgrade process fails, a redundant server is readily available to support correct operation of the network.

The standardization of so-called **intelligent network (IN)** methods was started in the 1980s in the United States by Bellcore. In the 1990s, ITU took a role in the standardization work and published standards Q.1200 [12]–Q.1400 [13] which closely follow the American standards. Eventually, ETSI defined their recommendations about how those ITU standards should be applied in European telephone networks. According to ITU-T Q.1219 [14] the intelligent network concept had the following targets:

- Efficient use of network resources
- Modularization of network functions
- Integrated service creation and implementation with reusable standard network functions
- Portability and flexible allocation of network functions to physical entities
- Standardized communication between network functions.

Various interrelated protocols were specified for the communication between intelligent network exchanges and external call control and database systems. The protocols IN/1, AIN, and IANP were used in the different intelligent network standard versions [8]. Each of these protocols specify what kind of data the exchange may query from other systems. To support standard patterns of communications, intelligent network (IN) call models were also standardized. Please refer to the *Online Appendix B.2* for a brief description of ITU-T CS-2 version of the IN call model defined in the ITU-T Q.1224 [15] specification. Since the standards were defined and enhanced over time, this common state model also evolved gradually as more detailed and complex. The unfortunate consequence was that in many cases the telephone exchange and call control server, which supported different versions of the standard, were not always able to interoperate correctly. IN protocols also specify the format of the query and response messages sent in well-defined steps of the call model. The messages compliant to these IN protocols were transported between devices with the TCAP protocol of the SS7 protocol suite (which is introduced in Section 1.2.3.2).

Support for IN features could be added to existing digital exchanges as a software upgrade. To complement such upgrades, new external call control and database systems were deployed for a complete solution. Interestingly, introduction of intelligent network technology took place during the 1980s when liberalization started to change the game for incumbent telephony operators. The 1980s and 1990s were a period of deregulation. Earlier, the telecom operators had a monopoly for providing telephony services within their countries. But from the 1980s onwards it became possible for challenger operators to establish and operate their own networks to compete with the incumbents. Together with the introduction of mobile telephony mass market, this caused the boom of the telecommunication industry during the 1990s and paved the way toward the mobile Internet era of today.

Intelligent network architecture made it possible for operators to specify additional supplementary services, supported by the logic in the external servers. Over time, those services proliferated. Eventually, ITU-T defined the various standard supplementary services, such as the following:

- Calling line identification presentation, ITU-T Q.731.3 [16]
- Calling line identification restriction, ITU-T Q.731.4 [17]
- Connected line identification presentation, ITU-T Q.731.5 [18]
- Connected line identification restriction, ITU-T Q.731.6 [19]
- Call diversion services: Call forwarding busy, Call forwarding no reply, Call forwarding unconditional, Call deflection, ITU-T Q.732.2 [20]
- Explicit call transfer, ITU-T Q.732.7 [21]
- Call waiting (CW), ITU-T Q.733.1 [22]
- Call hold (HOLD), ITU-T Q.733.2 [23]
- Terminal portability (TP), ITU-T Q.733.4 [24]
- Conference calling, ITU-T Q.734.1 [25]
- Closed user group (CUG), ITU-T Q.735.1 [26]

1.2.3 Signaling between Exchanges

As described in 1.1.7, an analog phone delivers the selected telephone numbers to the exchange either by cutting and closing the analog subscriber loop or transmitting special dual tone audio signals on a closed loop. The subscriber line is terminated either at the local multiplexer or the exchange. The signaling data such as the called number is transported in digital format between different types of digital telephone network equipment. Channel associated signaling and common channel signaling are methods used to transport signaling over digital trunks.

1.2.3.1 Channel Associated Signaling

The earliest method for transporting signaling over digital links was **channel associated signaling (CAS)** [8]. C5 signaling standards (CCITT Signaling System No. 5) defined the following CAS approaches:

- With C5 single frequency (SF) signaling, the telephone exchanges use the voice channels (T1/E1 timeslots) of a trunk to carry signaling tones while those channels are not used for calls. When the channel is not occupied, a certain constant audio tone is sent over it. Depending on the implementation, the tone is either within the range of 400–3400 Hz (used for carrying speech) or outside of it in the range of 3400–4000 Hz. In the former case, signaling cannot be transported on a channel when a call is connected to it, but in the latter case the signaling tones could be sent on parallel with a call. When the exchange requests to set up a call for a channel, it stops sending the signaling tone over the same channel. The remote exchange stops sending signaling tone to the same channel when it is ready for receiving the called number.
- With C5 multifrequency (MF) signaling, the telephone number selected by the caller is transported with 60-millisecond signals, which consist of two audible tones. Thus, the method is similar to DTMF tones, except for the strictly defined length of the signal and the frequencies used for the tones.

On digital PCM links, the channel associated signaling can be sent within the same channels as used for the calls or also over a separate signaling channel, if such exists in the frame structure. The former method is used on T1 links, while on E1 links the timeslot 16 is used for signaling. In the case of E1, the capacity of TS 16 is divided between the other 30 voice timeslots so that 16 consecutive E1 frames form a multiframe. To control voice channels, E1 multiframe provides 4 bits for signaling so that once within every multiframe half of the timeslot 16 is allocated for signaling associated to a specific voice timeslot. In this case, the signaling channel has lower capacity than audio channels and CAS signaling uses some binary indications instead of audio tones. The relation between one audio timeslot and the related four TS 16 bits of the multiframe used for CAS signaling is fixed. This makes the signaling still “channel associated.” (For further details about T1 and E1 frame structures, please refer to 1.3.1.3.)

1.2.3.2 Common Channel Signaling and SS7

The drawback of CAS signaling is that its tone system allows only very limited types of states to be expressed. CAS can be used just to tell if a timeslot is occupied or not and to pass the called number between exchanges. When using voice frequencies for CAS tones, signaling is not available at all during the call. When different kinds of supplementary services were developed, new needs emerged for signaling, which CAS could not support. To facilitate the new use cases and to optimize the capacity used for signaling, a new signaling method **common channel signaling (CCS)** [8] was developed during the 1970s. CCS was introduced by the CCITT Signaling System No. 6 standard, published in the year 1972.

Common channel signaling used an approach to transport signaling in digital rather than analog format. A set of CCS protocols were specified to be able to transport all the signaling information needed for various services of the telephone network. Common channel signaling protocols are members of the famous **SS7 protocol suite**. The **Signaling System Number 7 (SS7)** standards were specified and maintained during the 1980s and 1990s. The name SS7 originally referred to CCITT Signaling System No. 7 standard. Later, the SS7 protocol specifications were published within ITU-T Q.7xx standard series, after CCITT was renamed as ITU-T in 1993.

ITU-T standards define a number of SS7 protocol options, usage of which is determined within the national standardization organizations as profiles for local deployment. The SS7 protocol stack, as defined in ITU-T recommendation Q.700 [27], is depicted in Figure 1.9.

SS7 stack has the following protocol layers:

- **Message Transfer Part (MTP):** Set of protocols used to transport signaling messages in the right order and without errors from the source to the destination, such as between two telephone exchanges to set up a call. According to ITU-T Q.701 [28], MTP consists of three protocol layers on top of each other:
 - MTP level 1: Physical layer functions to transport bits with constant rate over a bidirectional link.
 - MTP level 2: Link layer functions such as alignment of the link, link monitoring, transport of SS7 frames, flow control, error correction, and retransmissions.
 - MTP level 3: Network layer functions such as routing of signaling messages, monitoring of links and routes, as well as route rearrangement in fault conditions. SS7 does not support dynamic routing protocols; thus, the network management system shall modify the contents of routing tables in each of the exchanges that perform routing.
- **Signaling Connection Control Part (SCCP):** Network layer protocol to complement features of MTP3. SCCP is used to transport such signaling messages which are not circuit related. The special task of SCCP protocol is to transport queries and notifications toward SCP network elements, such as intelligent network subscriber database servers.
- **Transaction Capabilities Application Part (TCAP):** Protocol used to deliver different kinds of commands and database queries related to telephone services. The specific commands and queries are defined in other national or international standards. TCAP is used to provide intelligent exchanges with information needed for call setup, while the actual call setup process is done with ISUP protocol.
- **ISDN User Part (ISUP):** An application layer protocol which is used to control the call setup and release processes between the exchanges. The predecessor of ISUP protocol in SS7 stack was called telephone user part (TUP). TUP did not yet support ISDN, while ISUP can and is used for both traditional voice call and ISDN call control.

In North American SS7 networks, the signaling messages use their own network, which is separated from the voice network. Totally separate links have been dedicated for signaling while voice traffic uses their own T1 links. Routing of signaling messages in the American model is done with dedicated equipment known as **signaling transfer point (STP)**. In the rest of the world, the signaling messages are transported between exchanges

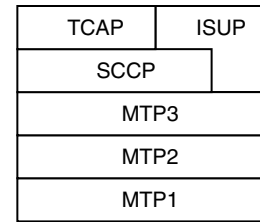


Figure 1.9 SS7 protocol stack.

along the same links used for voice circuits [8]. On the E1 links, the timeslot 16 of the frame is reserved for common channel signaling. The exchanges deployed outside of the American market take care of routing the signaling messages.

More information of the above mentioned SS7 protocols can be found from the *Online Appendix B*.

1.2.4 ISDN

Narrowband Integrated Services Digital Network (ISDN) was developed during the 1970s and the 1980s after most of the trunk network and exchanges were digitalized [4]. ISDN call control relied on the SS7 protocol suite. The following goals were defined for the narrowband ISDN system:

- The system should be able to transport 64kbps digital signals to subscribers over the existing telephone network cabling.
- Simultaneous usage of voice and data should be possible with a single ISDN terminal.
- The interface to the network should be such that it would enable connecting multiple terminals to a single cable connection, regardless of whether they all would use the same service or different ISDN services.
- The system should allow the end user to manage different supplementary services (such as call transfer, call hold, restriction of international calls etc.) with a numeric telephone keyboard.

The structure and functions of the ISDN system were defined within the I-series standards of CCITT. The specifications were published from 1984 onwards and later they were maintained by ITU-T. The concept and architecture of ISDN are specified in the following ITU-T recommendations:

- I.120 [29] Integrated services digital networks (ISDNs)
- I.310 [30] ISDN network functional principles
- I.324 [31] ISDN network architecture
- I.412 [32] ISDN user-network interfaces

For further details of narrowband ISDN, please refer to *Online Appendix C.1*.

The narrowband ISDN system was deployed at the first time in Great Britain in 1985. In the end of the 1980s, ISDN service was opened in the United States, France, and Germany. ISDN services were introduced in a slow pace for other countries so that the technology was in its widest use in the 1990s. In the early 2000s, ISDN became obsolete and ISDN connections were often replaced with DSL technologies.

Soon after introducing the narrowband ISDN, it was recognized during the 1990s that the narrowband 64kbps data rate was too slow for high-performance use of the Internet. ITU-T started activities to specify **broadband ISDN** version to support higher bitrates. Broadband ISDN was designed to use a new asynchronous transfer mode protocol, described in 3.2.2 and *Online Appendix H*.

1.3 Transmission Networks

1.3.1 E1 and T1

1.3.1.1 Standardization of E1 and T1

In order to transport multiple 64kbps PCM signals on a single twisted pair or coaxial cable link, two different standards based on synchronous time division multiplexing were developed at the 1960s:

- T1, where the TDM frame is divided into 24 timeslots to provide 1.544 Mbps total capacity for the link. T1 standard has been deployed in North America and Japan.
- E1, where the TDM frame is divided into 32 timeslots to provide 2.048 Mbps total capacity for the link. E1 standard has been deployed in the majority of the world, except in North America and Japan.

Since a 64 kbps channel (a single 8-bit timeslot of a T1/E1 frame) is the basic unit of transmission that may carry either PCM voice or data, it is called digital signal level 0 (DS0) user data channel [11]. The DS0 channel may also be referred as E0 (for E1 links) or T0 (for T1 links). T0 and E0 channels are typically used to carry PCM encoded voice. Due to the market specific requirements, T0 channels of T1 links typically carry PCM μ -law encoding while E0 channels of E1 links carry PCM voice encoded with A-law (see 1.1.3).

The frame structures of E1 and T1 as well as the functions of equipment processing those signals are specified in the following ITU-T standards:

- G.703 [33]: physical and electrical characteristics of PCM signals
- G.704 [34]: frame structures for signals with 1544 and 2048 kbps bitrates
- G.705 [35]: PDH equipment functional block model
- G.706 [36]: frame synchronization and CRC checksums of E1 and T1
- G.732 [37] and G.736 [38]: functions of E1 multiplexer generating the 2048 kbps signal
- G.796 [39]: functions of E1 cross-connect equipment

E1 and T1 links were designed to transport multiple 64 kbps PCM voice channels over trunk connections. While the technology was at first used for interconnecting exchanges, later it was used in various other kinds of contexts, such as connecting exchanges with local multiplexers, GSM base stations with base station controllers, as leased data lines to enterprises, or as data links within the early Internet core network. Later, such kinds of trunk lines were built with higher capacity link types, described in Section 1.3.

1.3.1.2 Endpoints of E1 and T1 Lines

The networks deploying T1 and E1 links use the following types of network equipment:

- **Regenerator (REG)** used on long transmission links to increase the length of the link. The regenerator interprets the received E1 or T1 signal and reproduces it to the next span of the link.
- **Multiplexer (MUX)**, which composes a single E1 or T1 signal from multiple DS0 channels and vice versa.
- **Add-drop multiplexer (ADM)**, which forwards one E1 or T1 signal from a link to another like a regenerator, but locally drops one or multiple DS0 channels from it and replaces them with other DS0 channels.
- **Digital cross-connect switch (DXC)**, which is able to interconnect the DS0 channels between different links and timeslots of multiple T1 and E1 links attached to the DXC.

Such pieces of equipment may either be standalone network devices or parts of a more complex telecom equipment, such as a mobile phone base station or microwave radio link.

1.3.1.3 Frame Structures

The structures of T1 and E1 signals are as follows (see Figure 1.10):

- T1: All the 24 timeslots are used for voice channels. In the beginning of a frame there is a framing bit which is used on every other frame to carry a signaling data channel. Further on, in the T1 multiframe structure every sixth frame has only 7 bits per timeslot used for user data or voice while 1 bit per timeslot is used for signaling.
- E1: Timeslot 0 is dedicated for frame synchronization and timeslot 16 for signaling. Other 30 timeslots are used for transporting voice channels.

Both types of frame structures are defined in ITU-T standard G.704 [34]. The length of one T1/E1 frame is 125 μ s, which is the sampling interval used in PCM. When the PCM process produces 8 bits of information every 125 μ s, it is possible to transport a PCM encoded voice signal over a single timeslot of the frame. Since the frame has either 24 or 30 voice timeslots, it is capable of transporting as many voice channels on parallel over a single link.

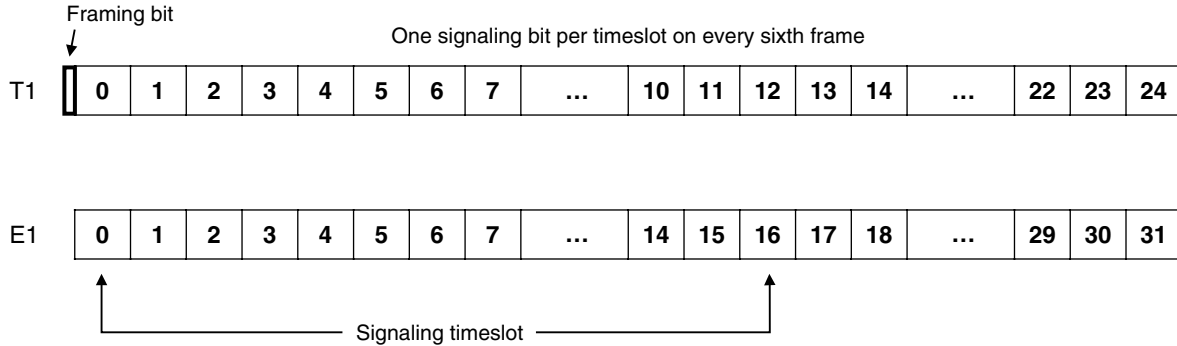


Figure 1.10 Structures of T1 and E1 frames.

The bitrates of the links in these two standards are as follows:

- T1: 1.544 Mbps = 24×64 kbps
- E1: 2.048 Mbps = 32×64 kbps

The length of T1 frame is 193 bits, transmitted in 125 microseconds, resulting 1544 kbps bitrate. The first bit of the frame is so called F bit, and the other 192 bits are used for 24 pcs of 64 kbps timeslots (TS); 24 consecutive T1 frames form a T1 multiframe.

The structure of T1 frame is as follows:

- F bit divided over the multiframe as follows:
 - The F bits of every other T1 frame (starting from the very first frame of the multiframe) belong to the 4-kbps network management channel. This channel is able to transport regular reports about the link status within LAPD protocol messages. Additionally, certain bit sequences have been defined, which are used to send commands or information related to reconnecting the link during the link test process, protecting the link with a redundant link or using the link as a synchronization source. The synchronization status messages (SSM) used for the latter purpose are similar to the SSM messages used in SDH, as described in *Online Appendix D.2.4*.
 - The F bits of every fourth T1 frame (starting from the second frame of the multiframe) belong to the CRC-6 checksum calculated from the information bits within the multiframe.
 - The F bits of every fourth T1 frame (starting from the fourth frame of the multiframe) are used to transport a constant frame alignment signal bit sequence 001011. This signal is used by the receiver to recognize the start of the multiframe.
- 24 timeslots as T0 channels. In the T1 multiframe structure, every sixth frame has only 7 bits per timeslot used for user data or PCM voice while 1 bit per timeslot is used for signaling.

The bits of a T1 signal are transmitted over the wire using AMI line coding [33]. T1 signal is transported over a coaxial cable or two unidirectional twisted pairs. The reason for using two pairs of wires is that the AMI line coding method used attenuates on twisted pair so quickly that unidirectional amplifiers are needed at every kilometer of the cable. Coaxial cable is a better option for long distances due to the lower attenuation.

The length of E1 frame is 256 bits, transmitted in 125 microseconds, resulting in 2048 kbps total bitrate. All the 256 bits of the frame are used for 32 pcs of 8-bit timeslots (TS). The first timeslot 0 is dedicated for frame synchronization and timeslot 16 for signaling. Other 30 timeslots are used as E0 channels to carry PCM voice or data. In GSM networks, a single E0 timeslot could be used to carry multiple voice channels encoded with GSM voice codecs. For instance, the GSM Full Rate voice codec produces a bitstream of 16 kbps, thus four GSM voice channels can be multiplexed into a single E0 channel. Sixteen consecutive E1 frames make up an E1 multiframe, which is divided into two submultiframes, both with eight E1 frames.

The structure of the E1 frame is as follows:

- Timeslot 0
 - In the eight frames of a submultiframe, the first bit of TS0 in every other frame may be used to carry a CRC-4 checksum calculated over the submultiframe. The usage of this checksum is optional. Bits 2–8 of the TS0 in those frames contain the constant bit sequence 0011011 used as a frame alignment word.
 - The bit 1 of TS0 of the other four frames also carry CRC-4 checksum while bit 2 has a constant value one. The bit 3 is used to pass information about link fault conditions detected in the remote end. The rest of the bits are reserved for other purposes, such as sending SSM or network management messages.
- Timeslot 16 is used as a signaling channel in one of these two ways:
 - As 30 separate CAS signaling channels, one for each voice/data timeslot. Each of the signaling channels use 4 bits per multiframe, so that TS16 of each frame carries two of the signaling channels.
 - As a single 64 kbps channel used for common channel signaling (CCS), which replaced CAS during the 1980s.
- Timeslots 1–15 and 17–31 are E0 channels with 64 kbps bitrate. It is possible to allocate multiple timeslots for a single $N \times 64$ kbps data signal. Such aggregation of E0 channels has been used for leased data lines.

The T1 and E1 signals are synchronous so that the receiver shall constantly maintain the bit and frame synchronization with the received signal. Each network node processing T1 or E1 signals can derive its internal clock frequency either from an external precision clock signal or from the received T1 or E1 signal. In the older implementations, each link had to be synchronized separately, whereas newer implementations are able to forward the clock frequency from a link to another. In that way, all the devices of the network could be synchronized to one shared clock frequency.

1.3.2 V5

When the telephone exchange sites were close enough to subscribers, it was possible to build direct copper wire pairs from the exchange to telephones. Since end of the 1970s, the approach was changed and the copper wires were often terminated at a local multiplexer. T1 or E1 links were used to connect the local multiplexer with the exchange. Usage of the digital technology made it possible to decrease cabling costs as a single T1 or E1 cable was able to carry tens of simultaneous calls between the multiplexer and the exchange. On the other hand, the digital technology helped increasing the distance between telephone exchanges and subscribers, dramatically reducing the number of exchanges needed.

Initially, there was no standard specified as to how to connect subscriber lines via multiplexers to a digital trunk line. Instead, vendor-specific solutions were used. This typically caused a vendor lock and challenges for the operator to introduce new telephony services, which the vendor might not yet support. To ensure the compatibility between their systems, operators had to buy both their exchanges and multiplexers from the same vendor. To address these problems, during the 1990s, ETSI specified two different European interface standards, V5.1 and V5.2. While V5.1 specified a standard way of connecting subscriber links to trunks in a fixed manner, V5.2 supported dynamic connections allowing the operators to use statistical multiplexing to reduce the number of needed trunk lines. From the device perspective, V5.1 relied on multiplexers and V5.2 on concentrators. The first of these standards, V5.1, was widely deployed. The introduction of V5.2 took place so late that operators at that time were not too eager to make investments to their narrowband telephone networks. Instead, telephone operators started to invest in DSL to grow their Internet access business to complement the traditional voice business.

Please refer to *Online Appendix D.1* to find more information about V5 standards.

1.3.3 PDH

1.3.3.1 Standardization of PDH

Plesiochronous digital hierarchy (PDH) provided a way to multiplex many E1 or T1 signals to one cable and increase the transmission capacity over a trunk link for circuit switched connections. Plesiochronous

refers to almost synchronous. PDH signals are not completely synchronous as the network does not have a common clock frequency.

In the trunk network, it was often necessary to transport a much larger number of voice channels as available on E1 and T1 links. The most straightforward way to solve this problem was to multiplex a number of E1 or T1 signals to one cable with TDM. In the next step, those signals could once again be multiplexed to higher-order signals. E1 and T1 signals themselves are products of TDM multiplexing, so the outcome of this process is nested TDM signals for the trunk lines. A single high-order signal may then consist of two to four nested TDM levels of lower-order signals. This plesiochronous digital hierarchy (PDH) solution was specified by CCITT in the following standards, which later became ITU-T documents:

- G.704 [34]: PDH frame structures with bit rates between 1.5–45 Mbps
- G.754 [40]: PDH frame structures with 140 Mbps bit rate

1.3.3.2 PDH Signal Hierarchy and Operation

The very basic dilemma of PDH comes from the lack of common clock frequency for individual signals multiplexed together. When building a new network or digitalizing old trunks, deployment of new digital links was done cable by cable. Since building of new cables was considered expensive, often the old cables were used and only the signal format within the cable was changed to increase bitrates. In early digital trunk networks, on each link the clock synchronization was done with help of a clock within the local receiver circuit. Since the accuracy of such local oscillators was limited to 50 ppm (i.e., 0.005%), the consequence was that when multiplexing different lower order signals into a higher order signal, there was no common clock frequency to be used for the combined signal. Because of this, the bitrate of the higher order signal was chosen to be slightly higher than the combined bitrates of the multiplexed lower order signals. This was done to ensure that the higher order signal could carry all the data from the multiplexed signals, even if their actual clock frequencies (and bitrate) would exceed the nominal one of the multiplexer. When the higher order signal had some extra capacity, the amount of which depended on the differences between the clock frequencies of the signals, the gap was filled with **justification bits** carrying no data. These justification bits were removed at the demultiplexing phase. As mentioned in the beginning of the chapter, the name plesiochronous digital hierarchy refers to almost synchronous hierarchy where the justification (or stuffing) bits are the exception.

Initially, only one level of multiplexing was used. In that case, the presence of justification bits did not cause any major issues. When the transmission technology and circuits were evolved and the need for capacity grew, higher bitrates were called for. Those needs were met by repeating the TDM multiplexing process to create higher order signals from a number of lower order signals. At each layer of multiplexing, justification bits had to be used as the network would not have a common clock frequency for any of the signals. The final hierarchical structure of PDH signals specified by ITU [41], with either E1 or T1 being the base rate, were as shown in Table 1.2:

The T/DS part of the hierarchy was used in American markets while the E part of the hierarchy was defined for the European market, but was deployed also on other continents.

Each level of the hierarchy has its own frame structure with a fixed length. PDH frames do not only contain the multiplexed sub-signals and justification bits, but also some amount of additional data bytes over which the link endpoints can provide information about detected fault situations or measured performance of the link [42]. Some of the extra bytes are used for a signaling channel used to transport LAPD protocol signaling messages. To organize the usage of those extra bytes, a multiframe structure has been defined over a number of consecutive PDH frames, similar to how multiframes are used for E1 and T1.

The type of cabling required to transport PDH signals depended on the distance and the type of the signal. Twisted pair wires could be used to carry E1/T1 signals over 1–2 km. For longer distances, repeaters or coaxial cables were needed. E2/T2 could also use twisted pairs, if the pairs would be properly insulated from each

Table 1.2 PDH signal multiplexing hierarchy.

	Multiplexing structure	Voice channels total	Bitrate
E1	32 x E0	30	2.048 Mbps
E2	4 x E1	120	8.448 Mbps
E3	4 x E2	480	34.368 Mbps
E4	4 x E3	1920	139.264 Mbps
T1/DS1	24 x T0	24	1.544 Mbps
T2/DS2	4 x T1	96	6.312 Mbps
T3/DS3	7 x T2	672	44.736 Mbps
T4/DS4	6 x T3	4032	274.176 Mbps

other in a cable bunch. Coaxial or optical cables were needed for higher order PDH signals, such as E3/T3 and E4/T4.

1.3.3.3 PDH Network Architecture

The only standard type of PDH equipment is multiplexer, which simply multiplexes four to seven lower order signals into one higher order PDH signal to be transmitted over one single cable. The basic structure of a PDH network is a tree of multiplexers, the depth of the tree depending on the type of the highest order signal transported over the trunk. The internal structure and functionality of PDH equipment is specified in ITU-T standard G.705 [43]. The following ITU-T standards define multiplexers for the different levels of PDH multiplexing hierarchy:

- G.733 [44], G.734 [45]: T1 multiplexers
- G.743 [46], G.752 [47]: T2 multiplexers
- G.732 [37], G.735 [48]: E1 multiplexers
- G.742 [49], G.744 [50], G.745 [51]: E2 multiplexers
- G.753 [52]: E3 multiplexers
- G.754 [40]: E4 multiplexers

Equipment vendors, however, produced other types of PDH devices, such as add-drop multiplexers and cross-connects. Those devices were typically able to demultiplex the input signal by one level, connect the demultiplexed signals as desired, and multiplex the signals back to the original level for transmission. For instance, an E4 add-drop multiplexer could demultiplex the E4 signal to four E3 signals, drop one of them and multiplex the rest into another E4 signal to be transmitted onwards within the network. Some devices were even able to support 1 + 1 protection of signals, as described for SDH in *Online Appendix D.2.6*.

The main problem with PDH is that each level of multiplexing and demultiplexing must be implemented with a separate piece of equipment. Because of the presence of stuffing bits and slightly different bit rates of the multiplexed signals, it is simply not possible to find a single E1 signal from an E4 signal, without demultiplexing the whole hierarchy of signals. If there is a need to drop one signal of low bitrate (like E1 or even an individual E0 voice channel) from a high-speed E4 or E3 signal, the whole signal must be demultiplexed to its E1 components. After dropping the needed low-order signals, the higher order signal must thereafter be hierarchically multiplexed once again before it can be transported over the next trunk. For every level, the stuffing bits are removed by the demultiplexing process and restored once again in the multiplexing process. The stack of equipment needed for this was known as PDH multiplexer mountain (see Figure 1.11).

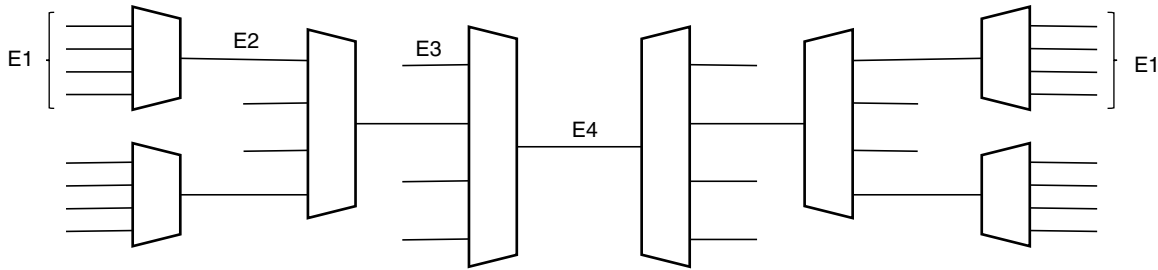


Figure 1.11 PDH multiplexer mountain.

It was very expensive for the network operators to acquire and maintain all the equipment needed by PDH transmission. In addition to the direct cost of the equipment, the building or rental costs of the premises were high as well when much space was needed for the equipment racks. Managing connections and cables between different multiplexers was complex and prone to human errors. Further problems were caused by the way PDH was specified. There was no global PDH market, since the American and European standards were not compatible. Even worse, the standards did not cover either PDH optical interfaces or network management aspects. During the 1980s, it became clear that for connections with higher bitrates than 140 Mbps, a better totally synchronous transmission standard was needed.

1.3.4 SDH

1.3.4.1 Standardization of SDH

Synchronous Digital Hierarchy (SDH) provides a way to multiplex many E1, T1, or PDH signals synchronously to one single electric or optical cable without additional justification bits. SDH increases the bit rates of a single signal even higher than the highest order PDH signals. SDH is an international standard, while it originated from an American SONET standard. SONET/SDH standardization had the following goals when the specifications were created in the end of the 1980s:

- SDH had to provide higher bit rates over an optical cable compared to what was available with PDH.
- SDH had to provide methods for adding and dropping individual lower order signals from a higher order signal, just using one single piece of equipment rather than a stack of multiplexers.
- SDH had to support multiplexing E1, T1, and any PDH signals.
- SDH was expected to provide methods for monitoring the SDH links as well as automatic reporting of faults detected.
- SDH was expected to provide methods for protecting links or individual lower order connections multiplexed to the links and taking the redundant capacity automatically into use in fault cases.
- SDH had to support centralized network management over a standard Q3 network management interface.

SDH multiplexing hierarchy, frame structures, functions and architecture are defined in the following ITU-T standards:

- G.703 [33]: Electric connectors of SDH equipment
- G.707 [53]: SDH multiplexing hierarchy and frame structures
- G.774 [54]: SDH management model
- G.781 [55]: SDH network synchronization
- G.783 [56]: SDH functional equipment model
- G.803 [57]: SDH network architecture
- G.957 [58]: Optical connectors of SDH equipment

ETSI has refined the SDH equipment model and functions in the specification ETS 300 147 [59], which is used as a reference for the latest versions of ITU-T standard G.783 [56].

This chapter provides high-level architectural overview of SDH, but further details of various SDH mechanisms can be found from the *Online Appendix D.2*.

1.3.4.2 Basic Principles of SDH Multiplexing

The very basic approach used for SDH is that all the SDH equipment within a single network is synchronized with one single **primary reference clock (PRC)** of the network. The clock frequency of the PRC is conveyed between the SDH equipment in the network over the SDH signals or with separate clock lines. The common clock frequency guarantees that SDH signals of certain order always have exactly same frame durations and bitrates, regardless from which link they are received. Because of this, the payload bit rate of a higher order SDH signal is exactly the sum of the multiplexed lower order SDH signals and no stuffing bits are needed. All that makes the SDH demultiplexing process straightforward even over multiple layers of nesting the signals.

An SDH frame is called **synchronous transport module (STM)**. When an external signal such as E1, T1, or any PDH signal is multiplexed into an STM, the clocking frequencies and bitrates of those external signals (based on their own clocking) must be adjusted to the common clock rate of the SDH network. Thus, while all the SDH signals of the SDH network have a common clock rate, the carried PDH signals do not follow it. Because of this, the transmission capacity reserved in SDH for a PDH signal is slightly higher than the nominal bitrate of the PDH signal. Extra additional bitrate is no longer needed for multiplexing SDH signals within the SDH multiplexing hierarchy. SDH uses stuffing bits only for adjusting the incoming PDH signals to the common SDH network bitrate. The transport capacity reserved within STM to carry PDH signals is called **container**. The received PDH signal and SDH stuffing bits in predefined positions are carried by the container. In addition to those bits, the container has a set of control bits telling if the stuffing bits carry any data or not. The number of stuffing bits depends on the ratio of the SDH primary clock frequency to the clock frequency of the carried PDH signal. As stuffing bits are no longer needed for multiplexing lower-order SDH frames into higher-order frames, it is possible to locate a single container from a nested SDH frame structure without demultiplexing the whole SDH signal. Figure 1.12 shows how higher-order SDH frame structures can be built from lower-order containers.

The purposes of containers within Figure 1.12 is as follows:

- C-11: Designed to carry T1 signals over SDH network
- C-12: Designed to carry E1 signals over SDH network
- C-2: Designed to carry T2 signals over SDH network
- C-3: Designed to carry T3 and E3 signals over SDH network
- C-4: Designed to carry E4 signals over SDH network
- C-4-4c: Transport capacity equaling to four concatenated C-4 containers to transport any very high bitrate data streams over SDH

ITU-T recommendation G.707 defines mappings for also other types of container payloads, such as stream of ATM cells and protocols such as DQDB and FDDI, which were used for local data transport while SDH was under specification.

A **virtual container (VC)** is a frame structure created by adding some **overhead (OH)** header bytes to the front of the **container (C)**. The location of a virtual container is not fixed within the SDH frame carrying the containers, but the start of a VC can be found with help of pointers located to fixed positions of the SDH frame. This mechanism is needed because there is no common frame synchronization at the edge of SDH network (due to processing of E1, T1, and PDH signals not synchronized with each other) even if the SDH network has a common clock frequency. While the length of each virtual container of the same order is exactly the same in the SDH network, the starting positions of the virtual containers as generated by different SDH edge equipment will differ. As an STM frame may multiplex virtual containers coming from different

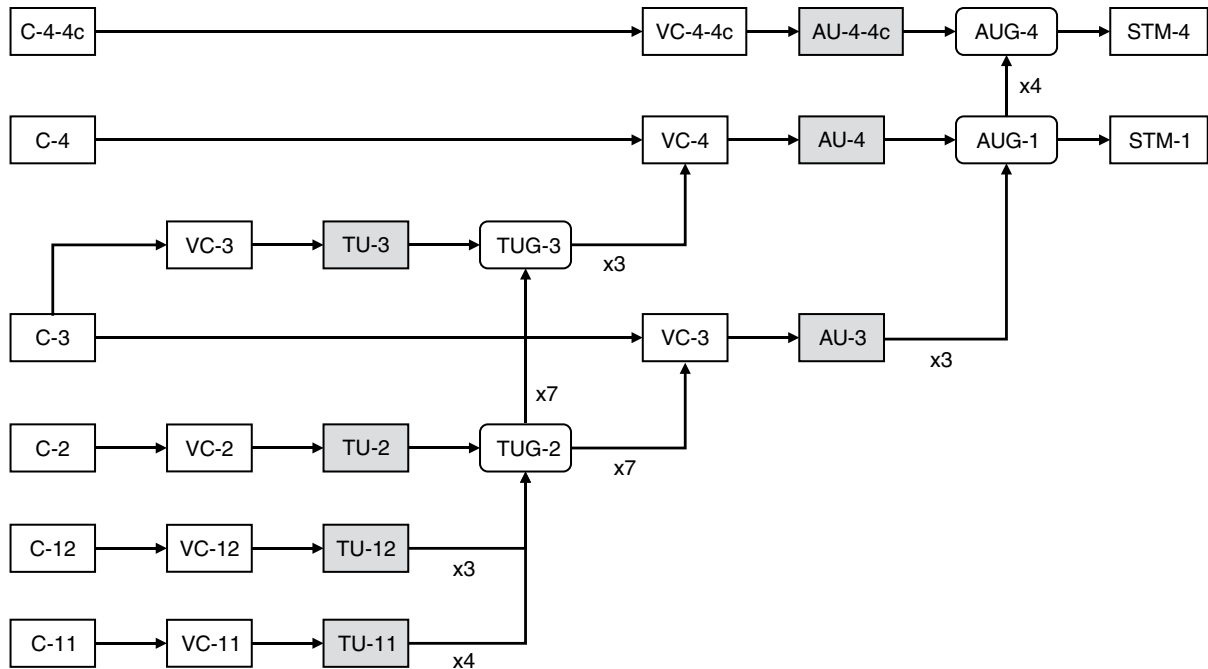


Figure 1.12 SDH multiplexing structure up to STM-4. *Source:* Adapted from ITU-T G.707 [53].

sources, the starting position of any virtual container may be arbitrary as compared to the start of the STM frame. Pointers are used to overcome this problem.

For each such SDH pointer there is a mechanism to adjust small differences between the clocking signals of the multiplexed signals. This mechanism is needed when virtual containers are transported between two different SDH networks, which are relying on different PRC clock sources. The starting positions of virtual containers multiplexed into one single higher-order SDH signal can be nudged by a few bits or bytes forwards or backwards. When moving the start position of VC backwards, a few bits can be taken from the pointer area to carry the virtual container data. When moving the start position of VC forwards, a few stuffing bits are added to the end of the pointer area.

Virtual containers are divided into two classes depending on the location of the VC in the SDH multiplexing hierarchy. The lowest-order virtual containers together with the pointers to VC frame start are called **tributary units (TU)**. A group of multiplexed tributary units are called **tributary unit group (TUG)**. TUG may carry either one high-capacity virtual container or multiple smaller capacity ones. The higher-order virtual containers together with the pointers to the VC frame start are called **administrative units (AU)**, and the group of them are **administrative unit group (AUG)**, as shown in Figure 1.13.

SDH and SONET standards define frame structures for different bitrates. Since SONET is based on the PDH DS hierarchy used in American markets, it supports the related additional bitrates missing from the basic SDH. Table 1.3 describes the hierarchy of signals in the different standards:

The highest-order virtual container of SDH multiplexing hierarchy, VC-4 provides 140Mbits bitrate for its payload to carry an E4 PDH signal. If bigger transmission capacity is needed for one connection, it is possible to concatenate multiple VC-4 virtual containers connected along with the same route as one logical VC-4-Xc channel, if all the equipment along with the path supports such concatenation. The value “X” is the number of concatenated VC-4 signals and is typically four, but larger values may also be used. The concatenation is done so that information about the concatenated VC-4 virtual containers is provided with the AU pointers. It must be noted

that the path of the complete VC-4-Xc signal has to be kept always together end-to-end. If any of the VC-4 components or underlying transmission links break due to a fault then the whole concatenated VC-4-Xc signal is lost unless it can be moved to a redundant link.

1.3.4.3 SDH Network Architecture

It is possible to form topologically different types of networks from available SDH network element types (ITU-T G.803 [57]). Typically, at the network edge, a three-shaped structure is used. When moving toward the core of the network there is a need to protect links with redundant ones. Ring topology may be used to provide connection between two elements over two routes. In the very core of the network there is a need to build direct connections between all the elements as mesh, to support provisioning of both capacity allocation and link protection. In the network core, the transmission rates of the links are much higher compared to the network edge. See Figure 1.14 as an example of a small SDH network.

The following types of equipment are used in SDH networks:

- **Regenerator (REG)** is used on long trunks to extend their reach. The regenerator receives the attenuated optical signal, converts it to electric digital format, and regenerates the signal with its original transmission power to the next span of the link.
- **Multiplexer (MUX)** takes care of multiplexing lower-order signals to a higher-order signal to an optical (or an electric STM-1E) transmission link. Depending on its line units, SDH multiplexer is able to multiplex signals of different order and bitrates. For instance, an STM-4 multiplexer is able to compose an STM-4 signal by multiplexing 2 x STM-1, 1 x E4, and 63 x E1 signals into it.
- **Add-drop multiplexer (ADM)** forwards one STM signal from an SDH link to another but locally drops one or multiple lower-order sub-signals from the main STM signal and replaces them with other sub-signals.
- **Digital cross-connect switch (DXC)** is an equipment that is able to connect signals of SDH hierarchy (or the related virtual containers) received from different links to other links without any significant limitations, as depicted in Figure 1.15.

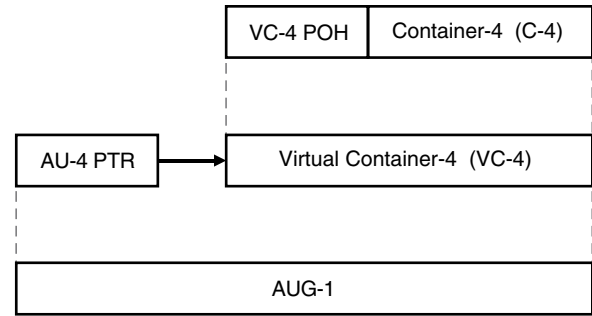


Figure 1.13 Building AUG-1 from C-4. *Source:* Adapted from ITU-T G.707 [53].

Table 1.3 SDH multiplexing hierarchy.

SDH signal	SONET electrical	SONET optical	Bitrate
STM-1	STS-3	OC-3	155.22 Mbps
STM-4	STS-12	OC-12	622.08 Mbps
STM-16	STS-48	OC-48	2488.32 Mbps
STM-64	STS-192	OC-192	9953.28 Mbps
STM-256	STS-768	OC-768	39813.12 Mbps
–	STS-1	OC-1	51.84 Mbps
–	STS-9	OC-9	466.56 Mbps
–	STS-18	OC-18	933.12 Mbps
–	STS-24	OC-24	1244.16 Mbps

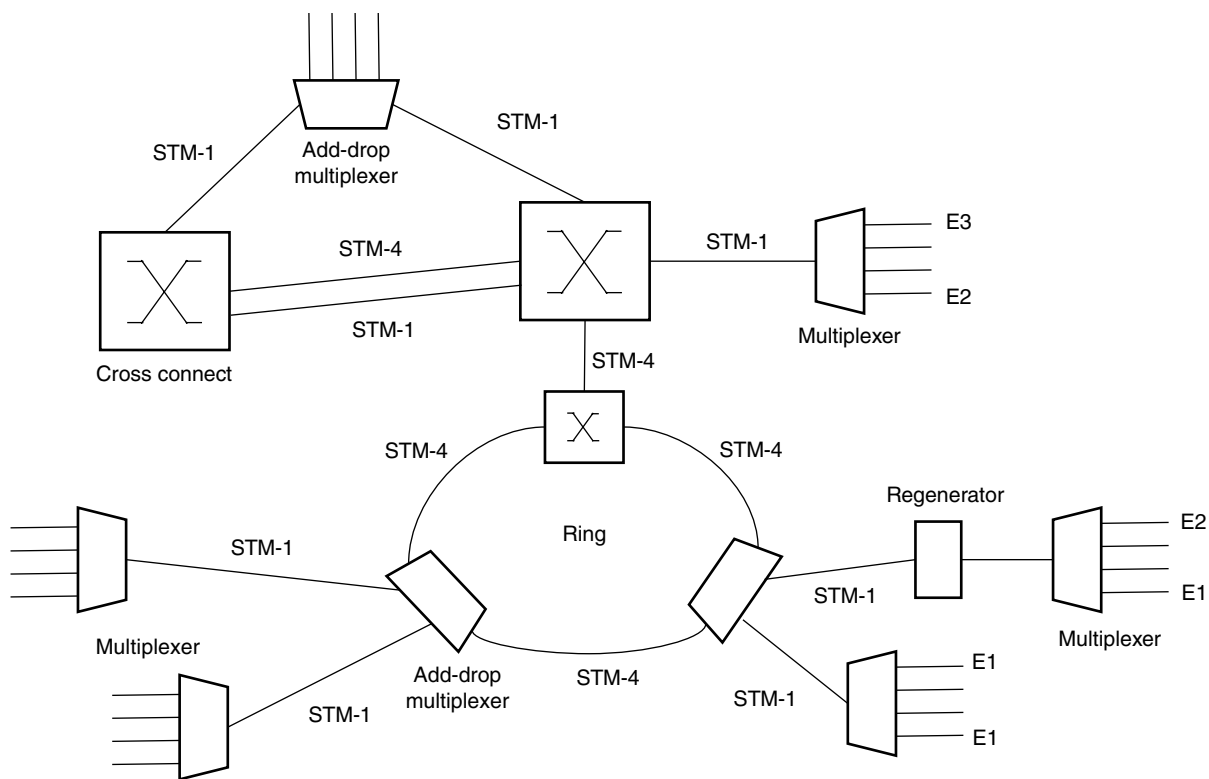


Figure 1.14 SDH network example.

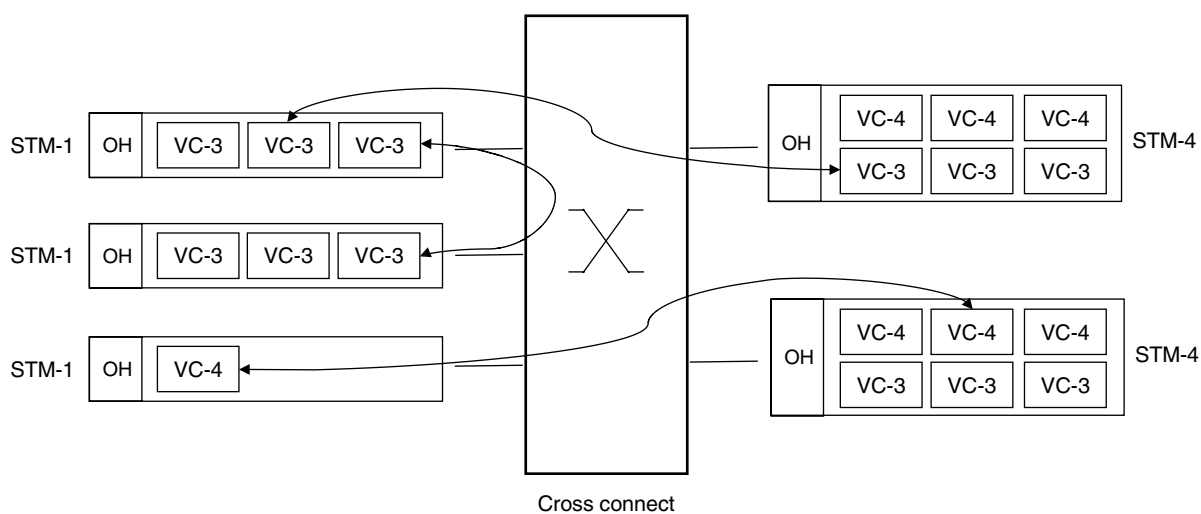


Figure 1.15 Connecting virtual containers in SDH cross-connect.

1.3.5 Microwave Links

1.3.5.1 Standardization of Microwave Link Systems

Microwave links can be used instead of cables on a point-to-point link, for instance, in the following cases:

- In an environment where installation of cables is difficult or expensive, such as rough terrain
- When setting up new connections quickly to areas where infrastructure is underdeveloped

Microwave links typically use a direct line-of-sight radio connection between two parabolic or horn antennas fixedly mounted to a tower or mast. The parabolic antenna has a dish shape, while horn antennas used for very high frequencies look like megaphones. Other types of antennas, such as microstrip patch or MIMO antennas, may also be used. The length of a radio link is from a few kilometers to 50 km. The microwave radio links are able to carry PDH or SDH frames over radio. They can be used as a part of a telecommunications network, for instance, to provide connectivity for a mobile phone base station antenna mast toward the core network.

Microwave links typically use licensed frequencies in frequency area 1–40 GHz. Additionally, the unlicensed 58 GHz band may be used. Each link connection uses a frequency band fixedly allocated for it. The directional antennas used at the end of the link have a direct line-of-sight radio connection between each other.

ETSI standard EN 302 217 [60] specifies some of the characteristics to be used for microwave radio links while it lacks complete standardization of signals transmitted.

1.3.5.2 Architecture of Microwave Radio Links

The architecture of a typical microwave radio link is as follows:

- An inside unit, which contains connectors for PDH or SDH trunks, an optional cross-connect unit, and a network management function.
- An outdoor unit, which contains the radio modem and equipment blocks for processing radio frequencies. The modem generates the frame sent over the radio link and takes care of line coding as well as carrier generation. The radio frequency processing blocks perform signal filtering and amplification.
- Antennas used for transmitting and receiving microwave radio signals over the link.
- Any cabling used between the units.

The signal to be transported is amplified both before transmission and after reception. Since the gain of the amplifier is typically constant, the transmit power is adjusted with adjustable attenuators.

1.3.6 Wavelength Division Multiplexing (WDM)

1.3.6.1 Standardization of WDM Systems

The Internet boom of the 1990s started exponential growth of global data consumption. The capacity needs for transporting data in access and core network rapidly increased. Supporting the requirements for ever higher link bitrates became more difficult than ever before. Extending the SDH signal hierarchy onwards from STM-64 was technically challenging due to the very large frequencies used on both the optical and electrical components. Since the problem was not the transmission rate of a single signal but the capacity available on optical cables, another approach was chosen to solve the issue. **Wavelength division multiplexing (WDM)** was the emerging solution to support multiplexing of different optical wavelengths to one optical cable [2]. Each wavelength was able to carry a single STM-16, STM-64, or Gigabit Ethernet data transport signal. This approach for the optical cables was similar to applying FDM to increase the capacity available over the electric cables. Both FDM and WDM allow transporting similar signals on parallel, using either different frequencies or wavelengths to carry the signals.

When using lasers with stable and narrow enough spectrum together with very precise optical filters, even 10–20 different wavelengths of light could be multiplexed to one single cable. In such a case, the cable could provide a total transmission capacity of tens or hundreds of Gigabits per second, depending on the bitrates of the individual signals multiplexed by WDM. One benefit of the WDM technology is that every separate wavelength could carry whatever upper layer protocol stack desired, making WDM solution a perfect platform for multiplexing different transport and link protocols.

WDM systems typically use bands that are 100 nm wide around the wavelength areas of 1300, 1530, 1550, and 1623 nm. In standards approximately 50 such wavelengths are defined. These wavelengths experience small enough attenuation in an optical cable to support spans with tens of kilometers for WDM links.

The functions of WDM systems are described in the following ITU-T standards:

- G.662 [61]: Characteristics of optical amplifiers and subsystems
- G.680 [62]: Physical functions of optical network elements
- G.694 [63]: Spectral grids for WDM lasers
- G.872 [64]: Architecture of optical transmission networks
- G.873.1 [65]: Linear protection in optical transmission networks
- G.874 [66]: Management of optical transmission networks

In addition to those individual documents, ITU-T has the following series of standards relevant for WDM networking:

- G.65x series recommendations define the characteristics of different types of optical fiber and cables.
- G.66x series covers various standards related to optical amplifiers, components, and subsystems.
- G.709.x [67] series defines architecture and multiplexing hierarchies for optical transport network (OTN). Following the SDH approach, OTN provides OTU frame structures with overhead information supporting forward error correction and performance monitoring. Success of OTN has yet been limited as many operators prefer to transport Gigabit Ethernet over raw WDM channels, instead of investing in the additional OTN layer to increase network reliability [2].

1.3.6.2 WDM System Building Blocks

WDM equipment can be built from the following types of optical components:

- Laser transmitters with narrow and adjustable wavelengths. Narrow enough wavelength can be produced by cooled distributed feedback lasers. The wavelength may be adjusted either mechanically by moving the mirrors of the transmitter or adjusting the filters used after the transmitter. The filters may be adjusted either electrically or with help of sound waves. Instead of adjustable transmitters, a matrix of fixed wavelength transmitters can be used so that the right transmitter instance is selected to produce the desired wavelength.
- Optical couplers, which are able to combine multiple optical input signals into one fiber. Couplers are used within WDM multiplexers and amplifiers.
- Optical receivers, which either convert the received optical power to electricity or modulate a locally generated laser with the received optical pulses so that the resulting microwave signal can be recognized.
- Optical filters and switches, which can be used to pick a specific wavelength from the received spectrum in an WDM add-drop multiplexer. The wavelength passing a filter may be fixedly configured or adjusted by changing the optical properties of the crystals in the filter electrically or acoustically.
- Optical amplifiers, which can amplify all the wavelengths received within a certain range of wavelengths. Operation of the amplifiers may be based on semiconductors or using erbium-doped fiber amplifier (EDFA), shown in Figure 1.16. In the latter case, the erbium in the fiber is driven with local laser pump to a state where

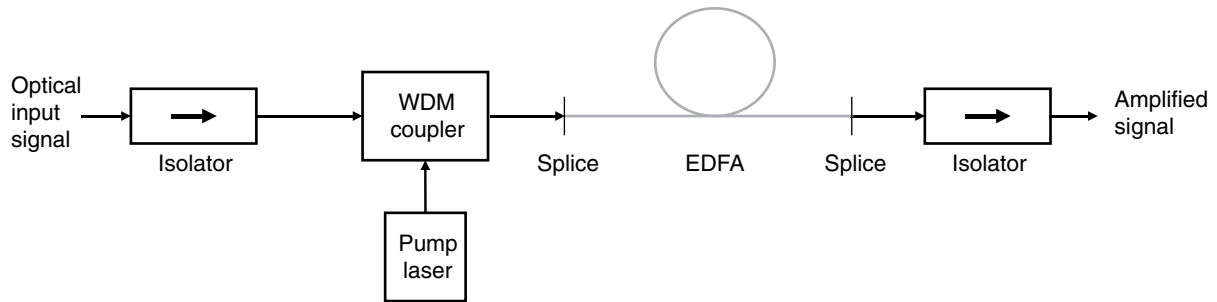


Figure 1.16 EDFA amplifier.

it is very close to emit light. When the amplifier receives light power also from the network, the total power absorbed by erbium makes it to emit light according to the received light pulses. The energy received from the input link is amplified with the energy erbium gets from the local unmodulated laser pump. Amplifiers may be used either at transmitters or receivers to amplify the signal before arriving to the receiver circuitry. It is also possible to use stand-alone amplifiers in the middle of a link to increase the span of the link.

- Wavelength converters, which may use only optical or both optical and electrical signals. The converter may change the received optical signal to an electrical one and thereafter back to an optical signal of another wavelength. Full optical converter may change the wavelength of the signal using a set of modulation, amplification, and optical filtering techniques.

The following techniques have been used for building optical cross-connects:

- It is possible to embed very small mirrors or hatches to a semiconductive board so that their positions can be adjusted electrically. With a very simple approach, such a hatch would either let or stop an optical signal to pass a hole in the board. A more complex implementation is able to adjust mirrors to different positions so that the mirrors can be used to guide the signal to a specific path. With modern technology it is possible to locate more than 100 mirrors to 1 square centimeter of the board.
- A bubble switch uses bubbles in liquid to connect and switch optical signals between optical waveguide fibers. The straight waveguides are attached to a circuit board and cut so that they are terminated at crossings. At the crossings, the waveguides meet with small angle between their paths. In each crossing there is a small pool of liquid. In basic configuration, the light passes the liquid directly so that the direction of the light ray is not changed. The light ray continues propagating over the crossing to a codirectional waveguide. With help of a bubble technique used for inkjet printers, bubbles can be created in the liquid. Such a bubble causes light to be reflected and the ray to change its path. In this way, the path of the light can be changed from one waveguide fiber to another one connected to the same crossing.
- Thermo-optical switch is able to switch the rays of light in the fiber crossings from one fiber to another with help of a thermo-optic phase shifter. When heating up the shifter electrically, the optical properties of the shifter can be adjusted so that the phase of the light wave changes within the shifter. Such a phase change causes the ray to change its path when passing through the shifter.
- Liquid crystal switch uses technique developed for liquid crystal displays. In the switch, the polarization state of a liquid crystal is electrically changed so that the ray is polarized between two directions. Based on its polarization state, the ray will pass one of the polarization filters used in front of the two fibers.
- Optical signal can be converted as an electrical signal before the switch and back to optical signal after the switch. Such a cross connect may be based, for instance, on an SDH cross-connect architecture.

1.3.6.3 WDM Network Architecture

The types of equipment used for WDM networks are similar to those used in SDH networks even if the actual technology is quite different:

- An optical amplifier or regenerator is used on long links to increase the span from a few tens of kilometers up to 100–200 km. A regenerator receives the attenuated optical signal from an upstream link, converts it to digital form, and re-creates the signal to the other downstream link. An amplifier just amplifies the optical signal, but it is much more inexpensive than a regenerator. However, spontaneous emissions from EFDA add extra noise to the signal, so the number of amplifiers that can be added to a link is limited. To reach very long distances, over 600–1000 km, typically regenerators are used instead of amplifiers.
- A multiplexer adds wavelengths from multiple sources to one optical cable.
- An add-drop multiplexer passes one WDM signal from a link to another, but locally drops and adds one or a few wavelengths.
- Optical cross-connect is able to interconnect wavelengths between links. The connections may be either permanent or they can be changed by the management system.

ITU-T recommendation G.872 [64] describes the architecture of optical transport network as a chain of links and connections between them. The links carry multiplexed wavelengths. The end-to-end path of the optical signal is called a trail. Network equipment functions, such as trail termination, adaptation, and connection functions similar to the SDH equipment, are defined in the specifications.

The simplest possible WDM network is one in which a wavelength is transported over only one single link between the source and destination. To build such a network, two WDM multiplexers are needed, one for both ends of the link. The topology of a WDM network may also be a ring so that the ring consists of WDM add-drop multiplexers (ADM) and interconnecting links between them. At each ADM, some of the multiplexed wavelengths exit and enter the ring. The dropped signals are connected to SDH or Gigabit Ethernet links. The other wavelengths continue their paths within the WDM ring. The WDM network can also be built as a partial mesh so that each of the WDM equipment within the network has a sufficient number of direct links with other endpoints.

When the topology of a WDM network becomes more complex, planning and building the network become more difficult. The planner will have cases where a path across many links cannot use a single wavelength, as none of the wavelengths are available on all the links along the path, even when every link may have some spare wavelengths. The solution to such problems is to use wavelength converters or transponders where necessary. When designing such networks, the following challenges must be solved:

- How to reserve necessary capacity and wavelengths from each link of the network?
- How to locate the expensive wavelength converters to the network?
- How to configure the connections of the network so that there are no loops in any unused wavelengths? This is a necessary condition, since loops would cause problems when the signals in the loop would be amplified forever.

1.4 Questions

- 1 What are the tasks of a telephone exchange?
- 2 How does the pulse code modulation work?
- 3 How does a telephone network benefit from signaling?

- 4 What were the reasons for introducing intelligent networks?
- 5 Which improvements did SDH bring over PDH?
- 6 What is meant by wave division multiplexing?

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