CHAPTER 1

Synchronous Hierarchical Networks

1.1 INTRODUCTION

Historically, the first method of telecommunications after the industrial revolution depended on telegraphy. Telegraphy, however, was a point-to-point, one-way wired connection, and it used a simplistic on–off modulation method by which an electric current was manually interrupted to provide Morse-like electric pulses that were decoded by a skilled operator into characters at the receiving end. The telecommunications revolution began with the ability to modulate DC electric current with a microphone. When analog voice impinges on its diaphragm, the modulated signal is transmitted over copper wires and converted into an acoustic signal, reproducing voice (Figure 1.1). The ability to simultaneously talk and listen at any time, created such a demand that the first communication network was created with switching nodes in between (albeit operated manually) and telephony was created. The evolving network, however, reached a point of growth that was becoming uneconomical due to labor involved in the manual switching from node to node, becoming particularly noticeable in geographically large countries with city centers located apart from each other, such as the United States. As a result, the old analog network was not profitable and new methods were needed to automate connectivity, increase networking capability, and transport more traffic over the same copper cables. The result was digital telephony over a synchronous network.

However, voice still remains an analog signal, and the analog transducer, the microphone, still generates an analog electrical signal, as it did more than 100 years ago. In fact, the analog signal generated by a telephone is transmitted over a twisted pair of wires (TP 2W), and separated by electrical signals flows over the pair in both directions simultaneously, circuits known as "hybrids." An alternate method uses two pairs of wires (TP 4W), one pair per direction, without hybrids.

In a 2W system (popular in narrowband), where the transmitted and received signals are separated by a "hybrid," part of the signal from the microphone, a *sidetone or echo,* leaks from the hybrid and finds its way to the earpiece (receiver); this is eliminated with an *echo-canceller.*

At the receiving line unit (or channel unit), the analog signal is converted to a pulse-code modulated (PCM) digital signal by means of a coder/decoder (CODEC) (Figure 1.2).

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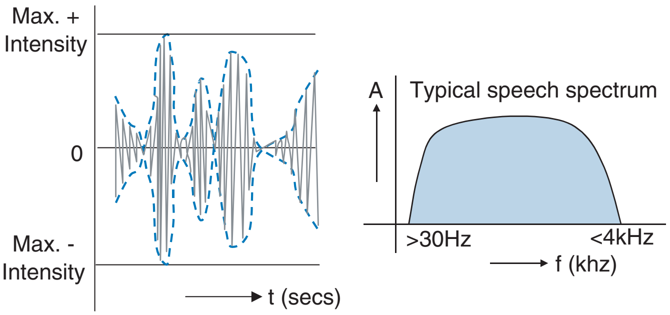

Figure 1.1 The voice signal is by nature analog. The filtered voice signal used in telephony has a frequency spectrum between 30 Hz and 4 kHz.

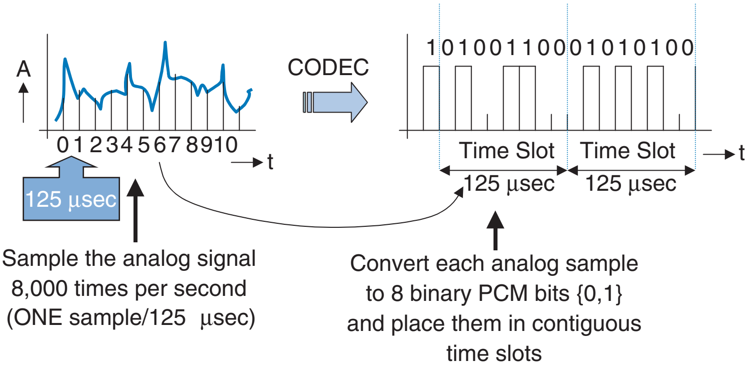

Figure 1.2 Sampling analog voice 8000 times per second and converting each sample to 8 bits pulse-coded modulated (PCM) is how the narrowband DS0 signal at 64 Kbps is generated.

Because the current that flows around the loop at the instance when the phone goes off-hook is DC, and the thickness of the copper wire is between 19 and 26 gauge, the loop length is limited by its ohmic resistance ($V = IR$).* Thus, it turns out that the best-case loops are as long as 28 kft (approximately 9 km). In most cases, cities in which population is dense, 28 kft is more than adequate. In fact, old European demographics did not require more than 12 kft, and 28 kft was almost exclusively used in the United States, where farms are located several kilometers away from telephone switching centers. So the US market prompted the development of

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*When the phone goes off-hook, it closes the loop and current flows. This flow is sensed by the channel unit of the switch. If the current cannot be sensed due to Ohmic resistance, then the switch will never know that the phone is off-hook. For DSL applications, ANSI standard T1.601-1988 equated an 18 kft (5.5 km) loop of 1300 ohm resistance to 42 dB loss at 40 kHz. In 1988, 18 kft represented 99% coverage of all loops in the United States.

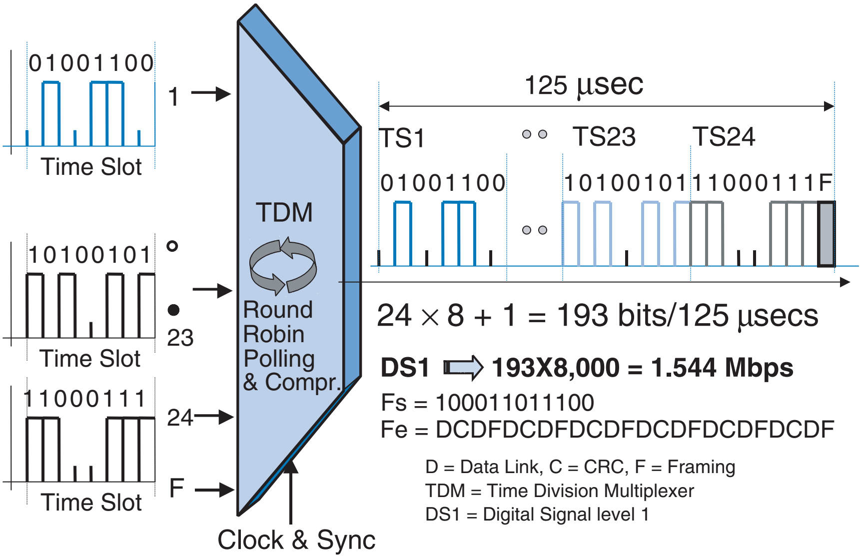

Figure 1.3 24 DS0s are synchronously round-robin polled by a M1 multiplexer to generate the DS1 signal at 1.544 Mbit/s.

the pair-gain systems. However, as soon as the analog signal was converted into digital, and with the evolution of electronics, digital multiplexing was greatly simplified, and the quality of transmission increased. Thus, a hierarchical multiplexing scheme evolved that was based on synchronous sequential or "round-robin" polling, and time division multiplexing (TDM) emerged, in the United States from digital signal level-0 or DS0 to DS3 (Figures 1.3 and 1.4), and in Europe and elsewhere from E0 to E31 (Figures 1.5 and 1.6).

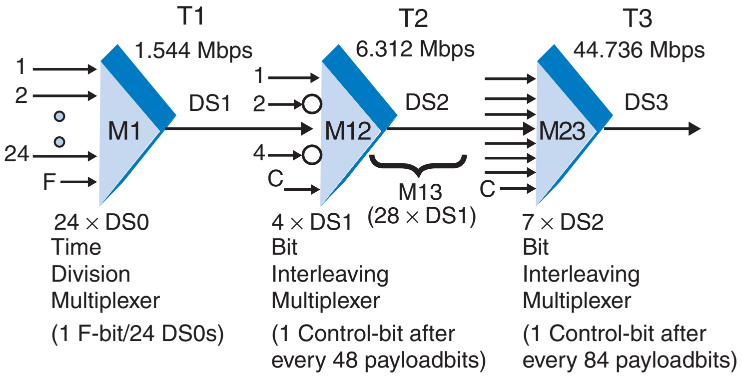

Figure 1.4 Synchronous hierarchy in the United States, from DS0 to DS3.

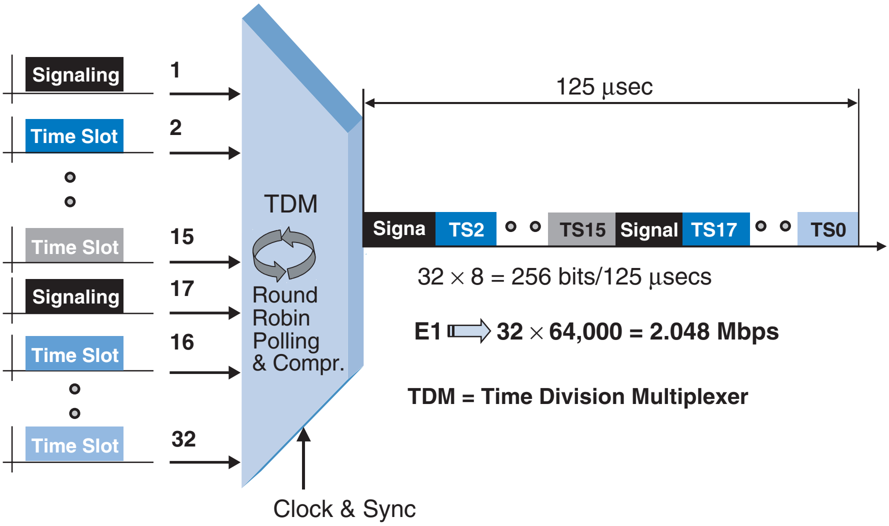

Figure 1.5 In Europe and elsewhere, the synchronous round-robin multiplexing included 30 DS0s and two bytes for signaling to construct the E1 signal at 2.048 Mbit/s.

1.2 SWITCHING HIERARCHY

Because of the very large number of telephones and data terminals, cable patching and ducting required good management. Thus, the cabling structure was separated into major responsibility domains. The subscriber lines connected the "plain old telephone service" (POTS) with a pair-gain system or directly with a *local exchange*. The local exchange was connected with *toll switches* by cable systems

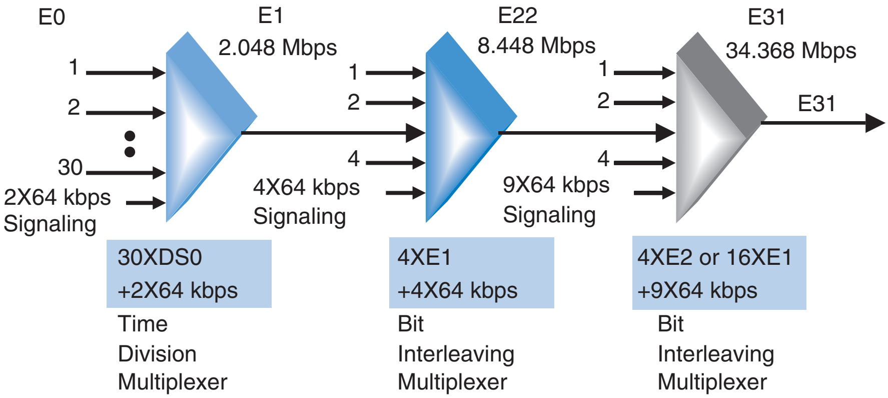

Figure 1.6 European (CEPT) synchronous hierarchy from E0 to E31.

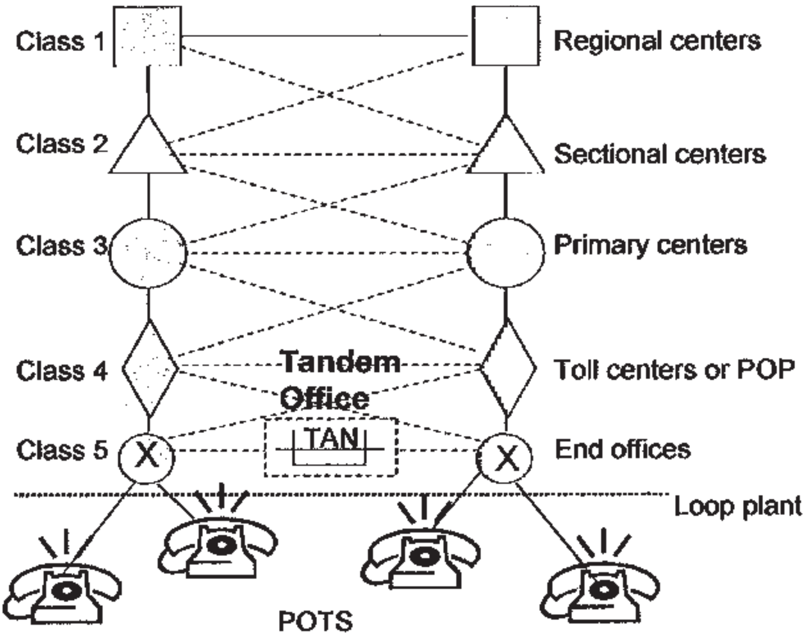

Figure 1.7 Switching hierarchy and classes of switching nodes. A tandem office provides connectivity between two low-end offices (Class 5) in the same local service area, avoiding toll centers.

known as *trunk lines,* and so on. Similarly, a hierarchical switching network evolved with switching nodes having different responsibilities in the overall networks (Figure 1.7). Edge switches were called local exchanges and they were responsible for call initiation and terminations as well as signaling for path or circuit setup. The toll switching centers were responsible for keeping records of which telephone was connected with which, for how long, and so on.

The switching hierarchy also establishes a network hierarchy within a city (if there are different service providers) or between different cities (Figure 1.8). In Europe, the nomenclature differed, and trunk lines were called *junctions.*

In general, trunks operate at higher bit rates, transporting what became known as Broadband traffic to differentiate it from DS0, known as narrowband. Table 1.1 summarizes the DSn and En data rates.

1.3 DIGITAL SUBSCRIBER LINES

In addition to analog access to the public switched network, over the last 20 years or so digital access has been increasingly penetrating the subscriber loop, hence called the digital subscriber loop (DSL). The first attempt to transmit data over the loop

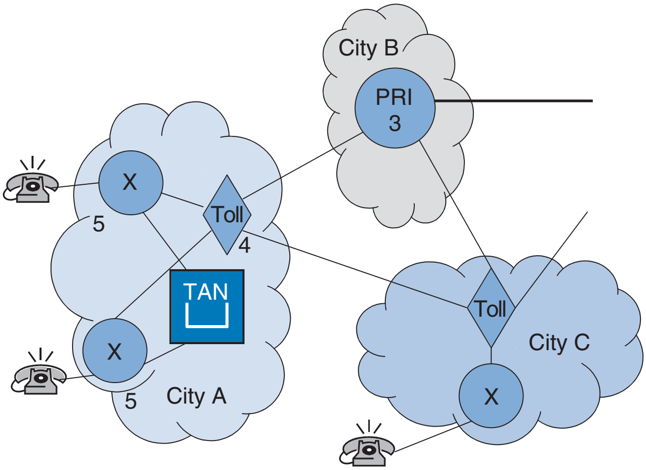

Figure 1.8 The network hierarchy is determined by the switching hierarchy within it. (TAN indicates a tandem switch.)

was a method initially known as time compression multiplexing (TCM), and officially as public switched digital capability (PSDC). This method used a TP-2W loop and transmitted 80 kbit/s of data, alternate mark inversion (AMI) modulated, in each direction, but periodically alternating directions. Hence it was nicknamed "ping-pong." However, this method was never standardized nor provided the needed bandwidth. The response to this was the standardized integrated services digital network (ISDN), which transmitted over a TP-4W at 144 kbit/s (2 × 64 + 16); this rate is known as basic rate ISDN (BRI). Clearly, digital transmission over the loop

Table 1.1 Traditional DS-n and E-n bit rates of the synchronous hierarchy

Facility	United States	Europe	Japan
DS0/E0	64 kbps	64 kbps	64 kbps
DS1	1,544 kbps		1,544 kbps
E1		2,048 kbps	
DS1c	3,152 kbps		3,152 kbps
DS2	6,312 kbps		6,312 kbps
E22		8,448 kbps	
			32,064 kbps
E31		34,368 kbps	
DS3	44,736 kbps		
DS3c	91,053 kbps		
			97,728 kbps
E4		139,264 kbps	
DS4	274,176 kbps		
			397.2 Mbps

implies that certain design concessions must be made. First, POTS cannot be used. Instead, specially designed telephone/data terminals that contain CODECS are used. Second, the digital loop plant can support a limited length of loops without induction coils (induction coils were placed on analog loops to suppress noise).* Third, the channel units or line units that interface the digital loop must be able to support ISDN.

The momentum of digital subscription rapidly increased in the last few years with the demand for higher bandwidth on the loop. Thus, more advanced methods were developed, all having the generic name of DSL. However, DSL is an umbrella of several technologies, collectively referred to as xDSL, where x denotes the particular DSL technology, format, and rate. For example:

- VDSL is very-high-bit-rate DSL.
- HDSL is high-bit-rate DSL.
- ADSL is asymmetric-bit-rate DSL; the downstream bit rate is much higher than the upstream.
- SDSL is symmetric-bit-rate DSL.
- RADSL is rate-adaptive-DSL; RADSL-based systems typically run in auto-rate adaptive or manual mode to adapt to a variety of bit rates as required by the user.
- MSDSL is multirate symmetric DSL; MSDSL-based systems are built on the single-pair SDL technology and offer one of many rates, thus, one of many loop lengths. For example, MSDSL on a 24 gauge unloaded copper pair can provide service at 64/128 kbit/s up to 29 kft (8.9 km), or 2 Mbps up to 15 kft (4.5 km).

Not all DSLs have the same modulation scheme or bit rate. Some modulation schemes are the 2B1Q, DMT, and CAP (Figure 1.9).

1.3.1 2B1Q

The two-bits-to-one quaternary (2B1Q) method translates each of the four states of a two-bit binary code in one of four voltage levels—–3, –1, +1, and +3—hence, *two binary, one quaternary.* Its transmitting power is superior to AMI (alternate mark inversion) that is used in T1 lines at 1.544 Mbps but its bit rate is limited to 392 kbps, which is suitable for upstream transmission on the loop. 2B1Q coding is used for BRI signals (basic rate ISDN).

1.3.2 DMT

The discrete multitone (DMT) method divides the bandwidth into frequency channels onto which traffic is overlaid. With DMT, when a certain (frequency) channel

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*Hybrid splitting may also be used to separate ADSL (digital) from POTS (analog) signals.

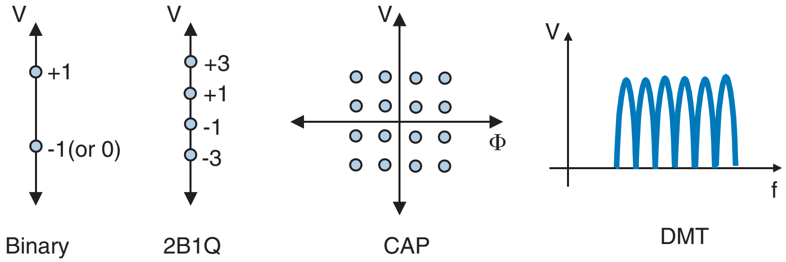

Figure 1.9 Three modulation methods in DSL, and binary transmission.

is detected to have inferior transmission characteristics, the traffic is assigned another frequency channel, a technique known as *frequency hopping*. DMT is the official standard of the ANSI T1E1.4 Working Group, supporting up to 6 Mbps services (this includes up to four MPEG-1 or a single MPEG-2 compressed video data; MPEG stands for Motion Picture Experts Group).

1.3.3 CAP

The carrierless amplitude phase (CAP) modulation is a derivative of the Quadrature Amplitude Modulation method (QAM). CAP translates a four-bit code in one of sixteen voltage-phase points. One may think of CAP as a 2B1Q two-dimensional approach, where the vertical axis is amplitude and the horizontal is phase. Its transmitting power is superior to AMI and 2B1Q; however, its effective bit rate is in the range of 10–175 kbps. Although DMT has been the standard of choice, CAP has been a de facto standard, which by 1996 dominated (by ~ 97%) all ADSL lines.

In summary, each DSL method has different characteristics and applicability (Figure 1.10). Some examples are:

- HDSL dual pair: The downstream bit rate is 2.048 Mbit/s, and the upstream bit rate is 2.048 Mbit/s. The maximum length of loop is 13,000 feet. Compare with a T1 line that requires a repeater every 6000 feet.
- HDSL single pair: The downstream bit rate is 768 kbit/s, and the upstream bit rate is 768 kbit/s. The maximum length of loop is 12,000 feet.
- ADSL DMT single pair: The downstream bit rate is 1.5 Mbit/s, and the upstream bit rate is 176 kbit/s. The maximum length of loop is 12,000 feet.
- ADSL CAP single pair: The downstream bit rate is 6 Mbit/s, and the upstream bit rate is 640 kbit/s. The maximum length of loop is 12,000 feet.
- ADSL CAP single pair: The downstream bit rate is 1.5 Mbit/s, and the upstream bit rate is 64 kbit/s. The maximum length of loop is 18,000 feet.
- ISDN two pairs: The downstream bit rate is 144 kbit/s, and the upstream bit rate is 144 kbit/s. The maximum loop length is 18,000 feet; 144 kbit/s is (2 × 64 + 16) kbit/s or 2B+D channels.

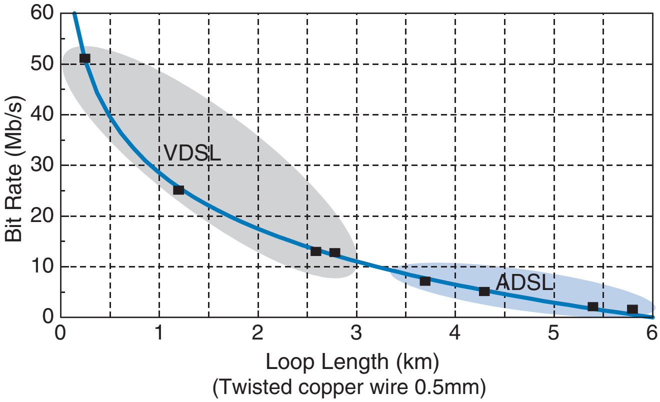

Figure 1.10 Bit rate versus distance for VDSL and ADSL.

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