We call the rate at which the signal changes the **symbol rate** to distinguish it from the **bit rate**. The bit rate is the symbol rate multiplied by the number of bits per symbol. An older name for the symbol rate, particularly in the context of devices called telephone modems that convey digital data over telephone lines, is the **baud rate**. In the literature, the terms "bit rate" and "baud rate" are often used incorrectly.

Note that the number of signal levels does not need to be a power of two. Often it is not, with some of the levels used for protecting against errors and simplifying the design of the receiver.

Clock Recovery

For all schemes that encode bits into symbols, the receiver must know when one symbol ends and the next symbol begins to correctly decode the bits. With NRZ, in which the symbols are simply voltage levels, a long run of 0s or 1s leaves the signal unchanged. After a while it is hard to tell the bits apart, as 15 zeros look much like 16 zeros unless you have a very accurate clock.

Accurate clocks would help with this problem, but they are an expensive solution for commodity equipment. Remember, we are timing bits on links that run at many megabits/sec, so the clock would have to drift less than a fraction of a microsecond over the longest permitted run. This might be reasonable for slow links or short messages, but it is not a general solution.

One strategy is to send a separate clock signal to the receiver. Another clock line is no big deal for computer buses or short cables in which there are many lines in parallel, but it is wasteful for most network links since if we had another line to send a signal we could use it to send data. A clever trick here is to mix the clock signal with the data signal by XORing them together so that no extra line is needed. The results are shown in Fig. 2-20(d). The clock makes a clock transition in every bit time, so it runs at twice the bit rate. When it is XORed with the 0 level it makes a low-to-high transition that is simply the clock. This transition is a logical 0. When it is XORed with the 1 level it is inverted and makes a high-to-low transition. This transition is a logical 1. This scheme is called **Manchester** encoding and was used for classic Ethernet.

The downside of Manchester encoding is that it requires twice as much bandwidth as NRZ because of the clock, and we have learned that bandwidth often matters. A different strategy is based on the idea that we should code the data to ensure that there are enough transitions in the signal. Consider that NRZ will have clock recovery problems only for long runs of 0s and 1s. If there are frequent transitions, it will be easy for the receiver to stay synchronized with the incoming stream of symbols.

As a step in the right direction, we can simplify the situation by coding a 1 as a transition and a 0 as no transition, or vice versa. This coding is called **NRZI** (**Non-Return-to-Zero Inverted**), a twist on NRZ. An example is shown in

Fig. 2-20(c). The popular **USB** (**Universal Serial Bus**) standard for connecting computer peripherals uses NRZI. With it, long runs of 1s do not cause a problem.

Of course, long runs of 0s still cause a problem that we must fix. If we were the telephone company, we might simply require that the sender not transmit too many 0s. Older digital telephone lines in the U.S., called **T1 lines**, did in fact require that no more than 15 consecutive 0s be sent for them to work correctly. To really fix the problem we can break up runs of 0s by mapping small groups of bits to be transmitted so that groups with successive 0s are mapped to slightly longer patterns that do not have too many consecutive 0s.

A well-known code to do this is called **4B/5B**. Every 4 bits is mapped into a5-bit pattern with a fixed translation table. The five bit patterns are chosen so that there will never be a run of more than three consecutive 0s. The mapping is shown in Fig. 2-21. This scheme adds 25% overhead, which is better than the 100% overhead of Manchester encoding. Since there are 16 input combinations and 32 output combinations, some of the output combinations are not used. Putting aside the combinations with too many successive 0s, there are still some codes left. As a bonus, we can use these nondata codes to represent physical layer control signals. For example, in some uses "11111" represents an idle line and "11000" represents the start of a frame.

Data (4B)	Codeword (5B)	Data (4B)	Codeword (5B)
0000	11110	1000	10010
0001	01001	1001	10011
0010	10100	1010	10110
0011	10101	1011	10111
0100	01010	1100	11010
0101	01011	1101	11011
0110	01110	1110	11100
0111	01111	1111	11101

Figure 2-21. 4B/5B mapping.

An alternative approach is to make the data look random, known as scrambling. In this case it is very likely that there will be frequent transitions. A **scrambler** works by XORing the data with a pseudorandom sequence before it is transmitted. This mixing will make the data as random as the pseudorandom sequence (assuming it is independent of the pseudorandom sequence). The receiver then XORs the incoming bits with the same pseudorandom sequence to recover the real data. For this to be practical, the pseudorandom sequence must be easy to create. It is commonly given as the seed to a simple random number generator.

Scrambling is attractive because it adds no bandwidth or time overhead. In fact, it often helps to condition the signal so that it does not have its energy in dominant frequency components (caused by repetitive data patterns) that might radiate electromagnetic interference. Scrambling helps because random signals tend to be "white," or have energy spread across the frequency components.

However, scrambling does not guarantee that there will be no long runs. It is possible to get unlucky occasionally. If the data are the same as the pseudorandom sequence, they will XOR to all 0s. This outcome does not generally occur with a long pseudorandom sequence that is difficult to predict. However, with a short or predictable sequence, it might be possible for malicious users to send bit patterns that cause long runs of 0s after scrambling and cause links to fail. Early versions of the standards for sending IP packets over SONET links in the telephone system had this defect (Malis and Simpson, 1999). It was possible for users to send certain "killer packets" that were guaranteed to cause problems.

Balanced Signals

Signals that have as much positive voltage as negative voltage even over short periods of time are called **balanced signals**. They average to zero, which means that they have no DC electrical component. The lack of a DC component is an advantage because some channels, such as coaxial cable or lines with transformers, strongly attenuate a DC component due to their physical properties. Also, one method of connecting the receiver to the channel called **capacitive coupling** passes only the AC portion of a signal. In either case, if we send a signal whose average is not zero, we waste energy as the DC component will be filtered out.

Balancing helps to provide transitions for clock recovery since there is a mix of positive and negative voltages. It also provides a simple way to calibrate receivers because the average of the signal can be measured and used as a decision threshold to decode symbols. With unbalanced signals, the average may be drift away from the true decision level due to a density of 1s, for example, which would cause more symbols to be decoded with errors.

A straightforward way to construct a balanced code is to use two voltage levels to represent a logical 1, (say +1 V or −1 V) with 0 V representing a logical zero. To send a 1, the transmitter alternates between the +1 V and −1 V levels so that they always average out. This scheme is called **bipolar encoding**. In telephone networks it is called **AMI** (**Alternate Mark Inversion**), building on old terminology in which a 1 is called a "mark" and a 0 is called a "space." An example is given in Fig. 2-20(e).

Bipolar encoding adds a voltage level to achieve balance. Alternatively we can use a mapping like 4B/5B to achieve balance (as well as transitions for clock recovery). An example of this kind of balanced code is the **8B/10B** line code. It maps 8 bits of input to 10 bits of output, so it is 80% efficient, just like the 4B/5B line code. The 8 bits are split into a group of 5 bits, which is mapped to 6 bits, and a group of 3 bits, which is mapped to 4 bits. The 6-bit and 4-bit symbols are then concatenated. In each group, some input patterns can be mapped to balanced output patterns that have the same number of 0s and 1s. For example, "001" is mapped to "1001," which is balanced. But there are not enough combinations for all output patterns to be balanced. For these cases, each input pattern is mapped to two output patterns. One will have an extra 1 and the alternate will have an extra 0. For example, "000" is mapped to both "1011" and its complement "0100." As input bits are mapped to output bits, the encoder remembers the **disparity** from the previous symbol. The disparity is the total number of 0s or 1s by which the signal is out of balance. The encoder then selects either an output pattern or its alternate to reduce the disparity. With 8B/10B, the disparity will be at most 2 bits. Thus, the signal will never be far from balanced. There will also never be more than five consecutive 1s or 0s, to help with clock recovery.

2.5.2 Passband Transmission

Often, we want to use a range of frequencies that does not start at zero to send information across a channel. For wireless channels, it is not practical to send very low frequency signals because the size of the antenna needs to be a fraction of the signal wavelength, which becomes large. In any case, regulatory constraints and the need to avoid interference usually dictate the choice of frequencies. Even for wires, placing a signal in a given frequency band is useful to let different kinds of signals coexist on the channel. This kind of transmission is called passband transmission because an arbitrary band of frequencies is used to pass the signal.

Fortunately, our fundamental results from earlier in the chapter are all in terms of bandwidth, or the width of the frequency band. The absolute frequency values do not matter for capacity. This means that we can take a **baseband** signal that occupies 0 to B Hz and shift it up to occupy a **passband** of S to $S+B$ Hz without changing the amount of information that it can carry, even though the signal will look different. To process a signal at the receiver, we can shift it back down to baseband, where it is more convenient to detect symbols.

Digital modulation is accomplished with passband transmission by regulating or modulating a carrier signal that sits in the passband. We can modulate the amplitude, frequency, or phase of the carrier signal. Each of these methods has a corresponding name. In **ASK (Amplitude Shift Keying)**, two different amplitudes are used to represent 0 and 1. An example with a nonzero and a zero level is shown in Fig. 2-22(b). More than two levels can be used to represent more symbols. Similarly, with **FSK (Frequency Shift Keying)**, two or more different tones are used. The example in Fig. 2-21(c) uses just two frequencies. In the simplest form of **PSK (Phase Shift Keying)**, the carrier wave is systematically shifted 0 or 180 degrees at each symbol period. Because there are two phases, it is called **BPSK (Binary Phase Shift Keying)**. "Binary" here refers to the two symbols, not that the symbols represent 2 bits. An example is shown in Fig. 2-22(c). A better scheme that uses the channel bandwidth more efficiently is to use four shifts, e.g., 45, 135, 225, or 315 degrees, to transmit 2 bits of information per symbol. This version is called **QPSK** (**Quadrature Phase Shift Keying**).

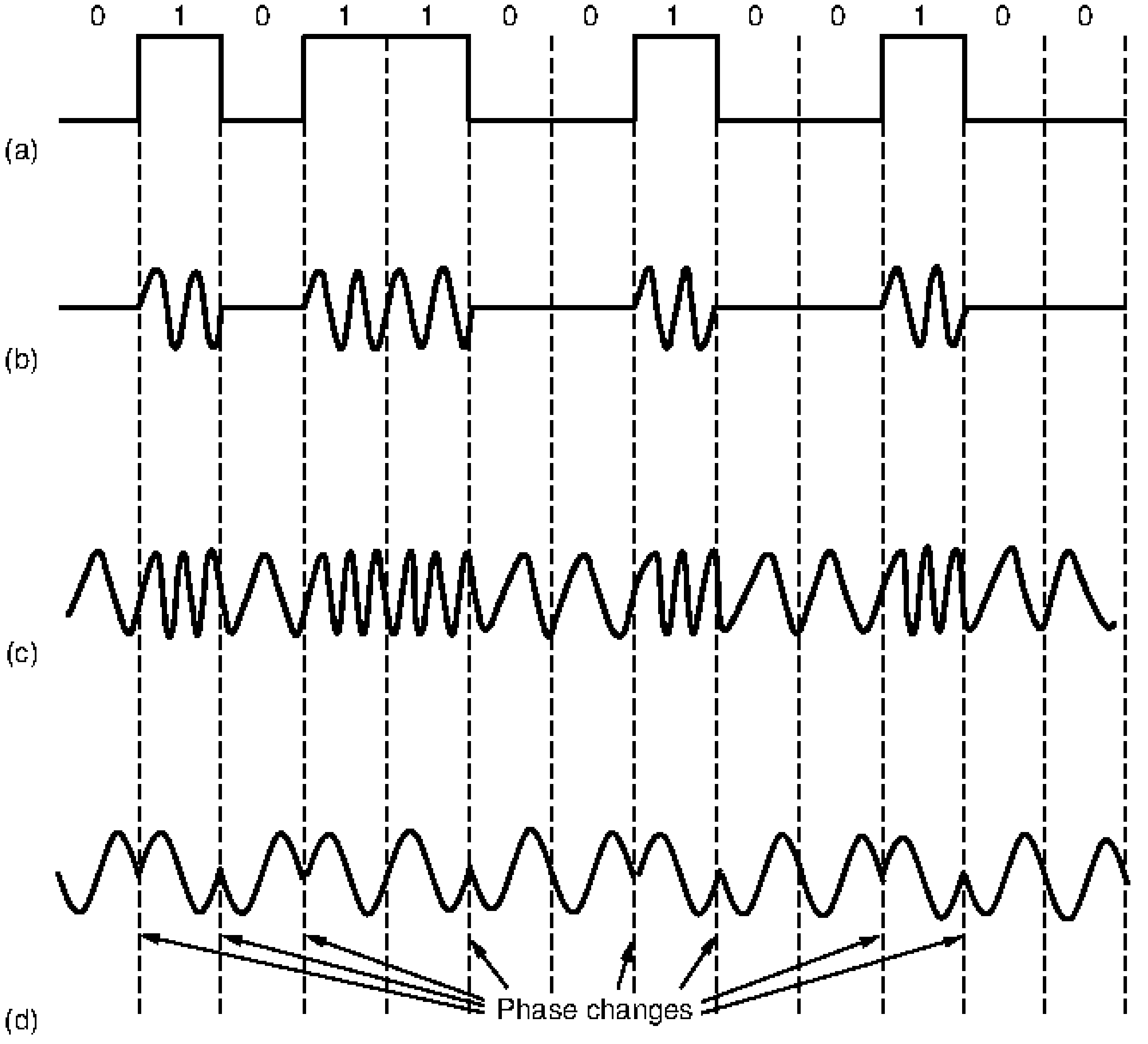


Figure 2-22. (a) A binary signal. (b) Amplitude shift keying. (c) Frequency shift keying. (d) Phase shift keying.

We can combine these schemes and use more levels to transmit more bits per symbol. Only one of frequency and phase can be modulated at a time because they are related, with frequency being the rate of change of phase over time. Usually, amplitude and phase are modulated in combination. Three examples are shown in Fig. 2-23. In each example, the points give the legal amplitude and phase combinations of each symbol. In Fig. 2-23(a), we see equidistant dots at 45, 135, 225, and 315 degrees. The phase of a dot is indicated by the angle a line from it to the origin makes with the positive x-axis. The amplitude of a dot is the distance from the origin. This figure is a representation of QPSK.

This kind of diagram is called a **constellation diagram**. In Fig. 2-23(b) we see a modulation scheme with a denser constellation. Sixteen combinations of amplitudes and phase are used, so the modulation scheme can be used to transmit

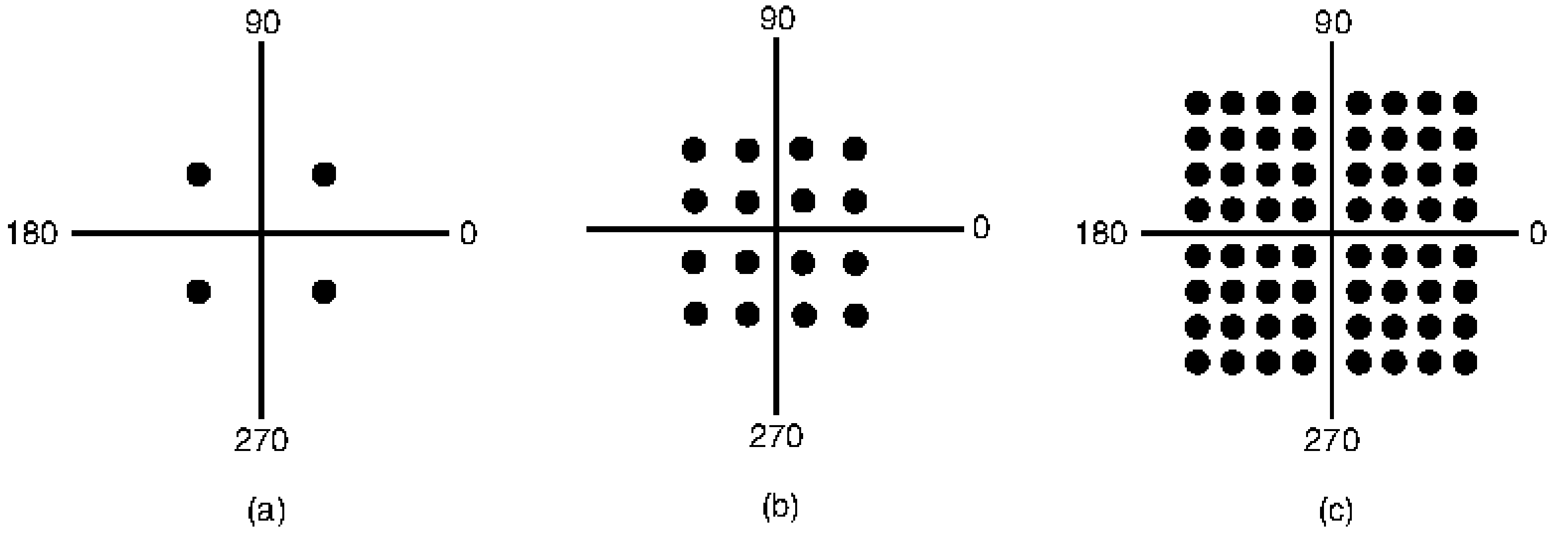


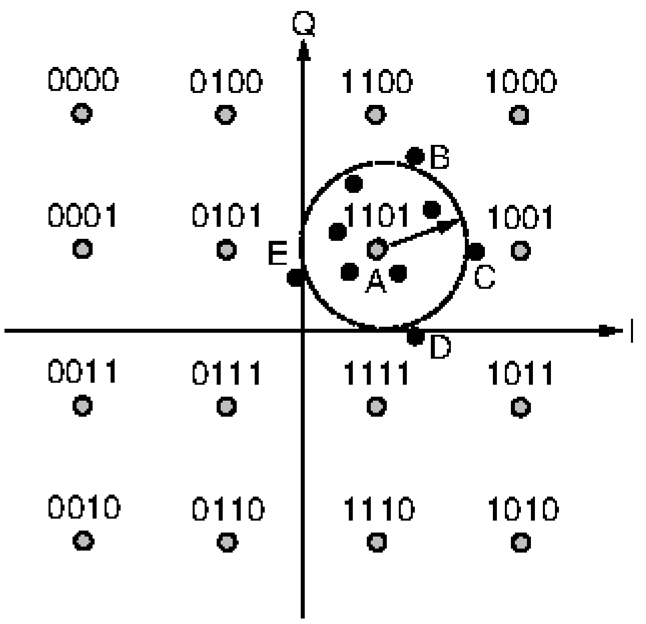
Figure 2-23. (a) QPSK. (b) QAM-16. (c) QAM-64.

4 bits per symbol. It is called **QAM-16**, where QAM stands for **Quadrature Amplitude Modulation**. Figure 2-23(c) is a still denser modulation scheme with 64 different combinations, so 6 bits can be transmitted per symbol. It is called **QAM-64**. Even higher-order QAMs are used too. As you might suspect from these constellations, it is easier to build electronics to produce symbols as a combination of values on each axis than as a combination of amplitude and phase values. That is why the patterns look like squares rather than concentric circles.

The constellations we have seen so far do not show how bits are assigned to symbols. When making the assignment, an important consideration is that a small burst of noise at the receiver not lead to many bit errors. This might happen if we assigned consecutive bit values to adjacent symbols. With QAM-16, for example, if one symbol stood for 0111 and the neighboring symbol stood for 1000, if the receiver mistakenly picks the adjacent symbol it will cause all of the bits to be wrong. A better solution is to map bits to symbols so that adjacent symbols differ in only 1 bit position. This mapping is called a **Gray code**. Fig. 2-24 shows a QAM-16 constellation that has been Gray coded. Now if the receiver decodes the symbol in error, it will make only a single bit error in the expected case that the decoded symbol is close to the transmitted symbol.

2.5.3 Frequency Division Multiplexing

The modulation schemes we have seen let us send one signal to convey bits along a wired or wireless link. However, economies of scale play an important role in how we use networks. It costs essentially the same amount of money to install and maintain a high-bandwidth transmission line as a low-bandwidth line between two different offices (i.e., the costs come from having to dig the trench and not from what kind of cable or fiber goes into it). Consequently, multiplexing schemes have been developed to share lines among many signals.



When 1101 is sent:

Point	Decodes as	Bit errors
A	1101	0
B	1100	1
C	1001	1
D	1111	1
E	0101	1

Figure 2-24. Gray-coded QAM-16.

FDM (Frequency Division Multiplexing) takes advantage of passband transmission to share a channel. It divides the spectrum into frequency bands, with each user having exclusive possession of some band in which to send their signal. AM radio broadcasting illustrates FDM. The allocated spectrum is about 1 MHz, roughly 500 to 1500 kHz. Different frequencies are allocated to different logical channels (stations), each operating in a portion of the spectrum, with the interchannel separation great enough to prevent interference.

For a more detailed example, in Fig. 2-25 we show three voice-grade telephone channels multiplexed using FDM. Filters limit the usable bandwidth to about 3100 Hz per voice-grade channel. When many channels are multiplexed together, 4000 Hz is allocated per channel. The excess is called a **guard band**. It keeps the channels well separated. First the voice channels are raised in frequency, each by a different amount. Then they can be combined because no two channels now occupy the same portion of the spectrum. Notice that even though there are gaps between the channels thanks to the guard bands, there is some overlap between adjacent channels. The overlap is there because real filters do not have ideal sharp edges. This means that a strong spike at the edge of one channel will be felt in the adjacent one as nonthermal noise.

This scheme has been used to multiplex calls in the telephone system for many years, but multiplexing in time is now preferred instead. However, FDM continues to be used in telephone networks, as well as cellular, terrestrial wireless, and satellite networks at a higher level of granularity.

When sending digital data, it is possible to divide the spectrum efficiently without using guard bands. In **OFDM (Orthogonal Frequency Division Multiplexing)**, the channel bandwidth is divided into many subcarriers that independently send data (e.g., with QAM). The subcarriers are packed tightly together in the frequency domain. Thus, signals from each subcarrier extend into adjacent ones. However, as seen in Fig. 2-26, the frequency response of each subcarrier is

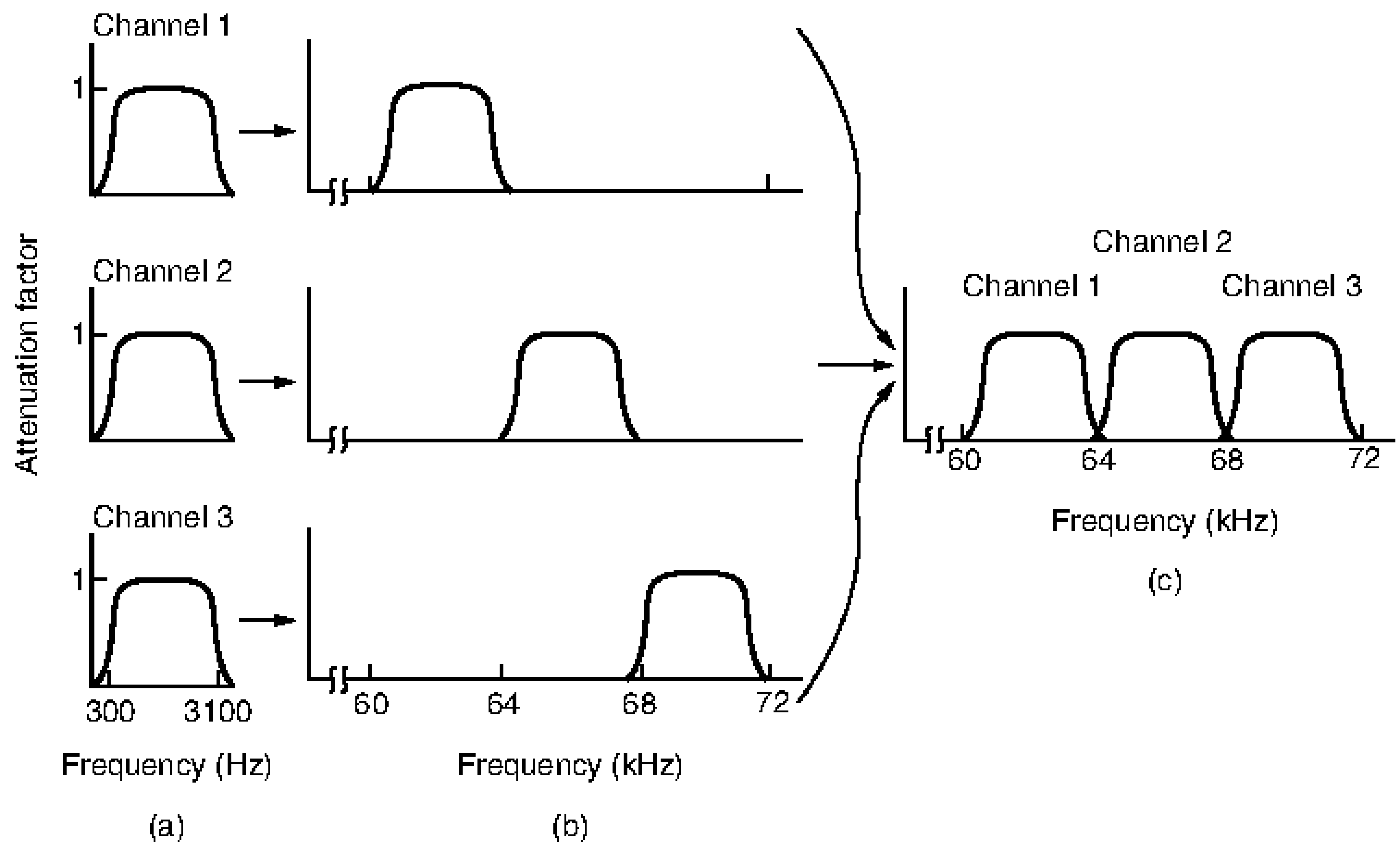


Figure 2-25. Frequency division multiplexing. (a) The original bandwidths. (b) The bandwidths raised in frequency. (c) The multiplexed channel.

designed so that it is zero at the center of the adjacent subcarriers. The subcarriers can therefore be sampled at their center frequencies without interference from their neighbors. To make this work, a guard time is needed to repeat a portion of the symbol signals in time so that they have the desired frequency response. However, this overhead is much less than is needed for many guard bands.

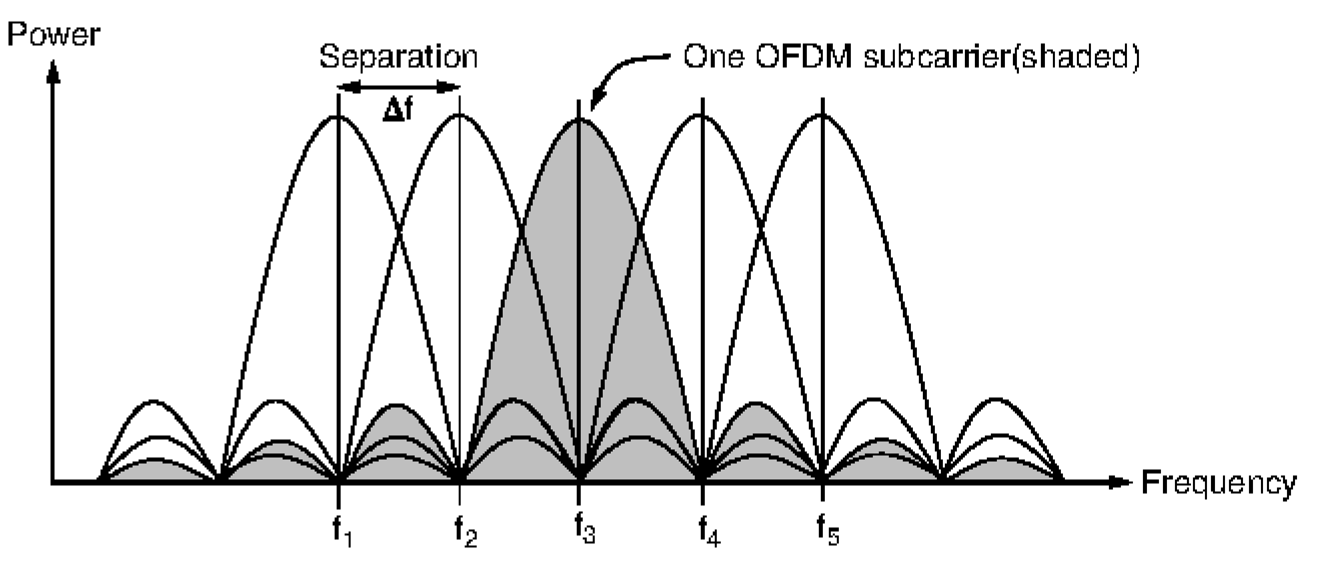


Figure 2-26. Orthogonal frequency division multiplexing (OFDM).

The idea of OFDM has been around for a long time, but it is only in the last decade that it has been widely adopted, following the realization that it is possible to implement OFDM efficiently in terms of a Fourier transform of digital data over all subcarriers (instead of separately modulating each subcarrier). OFDM is used in 802.11, cable networks and power line networking, and is planned for fourth-generation cellular systems. Usually, one high-rate stream of digital information is split into many low-rate streams that are transmitted on the subcarriers in parallel. This division is valuable because degradations of the channel are easier to cope with at the subcarrier level; some subcarriers may be very degraded and excluded in favor of subcarriers that are received well.

2.5.4 Time Division Multiplexing

An alternative to FDM is **TDM** (**Time Division Multiplexing**). Here, the users take turns (in a round-robin fashion), each one periodically getting the entire bandwidth for a little burst of time. An example of three streams being multiplexed with TDM is shown in Fig. 2-27. Bits from each input stream are taken in a fixed **time slot** and output to the aggregate stream. This stream runs at the sum rate of the individual streams. For this to work, the streams must be synchronized in time. Small intervals of **guard time** analogous to a frequency guard band may be added to accommodate small timing variations.

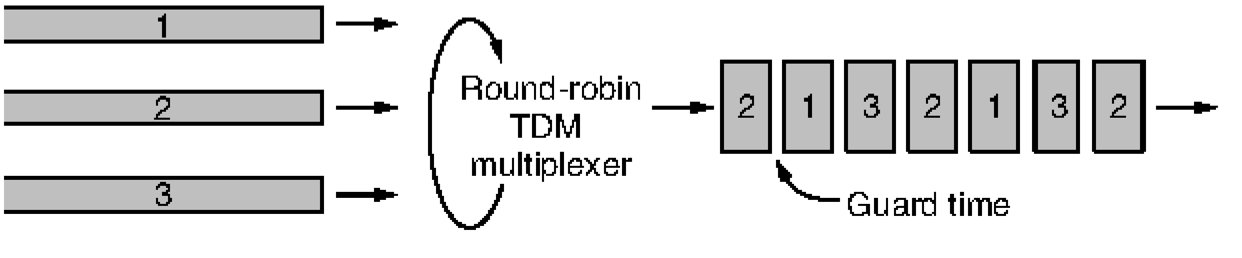


Figure 2-27. Time Division Multiplexing (TDM).

TDM is used widely as part of the telephone and cellular networks. To avoid one point of confusion, let us be clear that it is quite different from the alternative **STDM** (**Statistical Time Division Multiplexing**). The prefix "statistical" is added to indicate that the individual streams contribute to the multiplexed stream *not* on a fixed schedule, but according to the statistics of their demand. STDM is packet switching by another name.

2.5.5 Code Division Multiplexing

There is a third kind of multiplexing that works in a completely different way than FDM and TDM. **CDM** (**Code Division Multiplexing**) is a form of **spread spectrum** communication in which a narrowband signal is spread out over a wider frequency band. This can make it more tolerant of interference, as well as allowing multiple signals from different users to share the same frequency band. Because code division multiplexing is mostly used for the latter purpose it is commonly called **CDMA** (**Code Division Multiple Access**).

CDMA allows each station to transmit over the entire frequency spectrum all the time. Multiple simultaneous transmissions are separated using coding theory. Before getting into the algorithm, let us consider an analogy: an airport lounge with many pairs of people conversing. TDM is comparable to pairs of people in the room taking turns speaking. FDM is comparable to the pairs of people speaking at different pitches, some high-pitched and some low-pitched such that each pair can hold its own conversation at the same time as but independently of the others. CDMA is comparable to each pair of people talking at once, but in a different language. The French-speaking couple just hones in on the French, rejecting everything that is not French as noise. Thus, the key to CDMA is to be able to extract the desired signal while rejecting everything else as random noise. A somewhat simplified description of CDMA follows.

In CDMA, each bit time is subdivided into m short intervals called **chips**. Typically, there are 64 or 128 chips per bit, but in the example given here we will use 8 chips/bit for simplicity. Each station is assigned a unique m-bit code called a **chip sequence**. For pedagogical purposes, it is convenient to use a bipolar notation to write these codes as sequences of −1 and +1. We will show chip sequences in parentheses.

To transmit a 1 bit, a station sends its chip sequence. To transmit a 0 bit, it sends the negation of its chip sequence. No other patterns are permitted. Thus, for $m = 8$, if station A is assigned the chip sequence (−1 −1 −1 +1 +1 −1 +1 +1), it can send a 1 bit by transmiting the chip sequence and a 0 by transmitting (+1 +1 +1 −1 −1 +1 −1 −1). It is really signals with these voltage levels that are sent, but it is sufficient for us to think in terms of the sequences.

Increasing the amount of information to be sent from b bits/sec to mb chips/sec for each station means that the bandwidth needed for CDMA is greater by a factor of m than the bandwidth needed for a station not using CDMA (assuming no changes in the modulation or encoding techniques). If we have a 1-MHz band available for 100 stations, with FDM each one would have 10 kHz and could send at 10 kbps (assuming 1 bit per Hz). With CDMA, each station uses the full 1 MHz, so the chip rate is 100 chips per bit to spread the station's bit rate of 10 kbps across the channel.

In Fig. 2-28(a) and (b) we show the chip sequences assigned to four example stations and the signals that they represent. Each station has its own unique chip sequence. Let us use the symbol $\mathbf{S}$ to indicate the m-chip vector for station S, and $\overline{\mathbf{S}}$ for its negation. All chip sequences are pairwise **orthogonal**, by which we mean that the normalized inner product of any two distinct chip sequences, $\mathbf{S}$ and $\mathbf{T}$ (written as $\mathbf{S} \bullet \mathbf{T}$), is 0. It is known how to generate such orthogonal chip sequences using a method known as **Walsh codes**. In mathematical terms, orthogonality of the chip sequences can be expressed as follows:

$$\mathbf{S} \bullet \mathbf{T} \equiv \frac{1}{m} \sum_{i=1}^{m} S_i T_i = 0 \qquad (2\text{-}5)$$

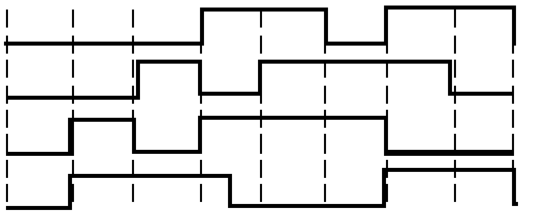
In plain English, as many pairs are the same as are different. This orthogonality property will prove crucial later. Note that if $\mathbf{S}\bullet\mathbf{T} = 0$, then $\mathbf{S}\bullet\overline{\mathbf{T}}$ is also 0. The normalized inner product of any chip sequence with itself is 1:

$$\mathbf{S}\bullet\mathbf{S} = \frac{1}{m}\sum_{i=1}^{m} S_i S_i = \frac{1}{m}\sum_{i=1}^{m} S_i^2 = \frac{1}{m}\sum_{i=1}^{m}(\pm 1)^2 = 1$$

This follows because each of the m terms in the inner product is 1, so the sum is m. Also note that $\mathbf{S}\bullet\overline{\mathbf{S}} = -1$.

A = (−1 −1 −1 +1 +1 −1 +1 +1)
B = (−1 −1 +1 −1 +1 +1 +1 −1)
C = (−1 +1 −1 +1 +1 +1 −1 −1)
D = (−1 +1 −1 −1 −1 −1 +1 −1)

(a)



(b)

$S_1 = C$ = (−1 +1 −1 +1 +1 +1 −1 −1)
$S_2 = B+C$ = (−2 0 0 0 +2 +2 0 −2)
$S_3 = A+\overline{B}$ = (0 0 −2 +2 0 −2 0 +2)
$S_4 = A+\overline{B}+C$ = (−1 +1 −3 +3 +1 −1 −1 +1)
$S_5 = A+B+C+D$ = (−4 0 −2 0 +2 0 +2 −2)
$S_6 = A+B+\overline{C}+D$ = (−2 −2 0 −2 0 −2 +4 0)

(c)

$S_1\bullet C$ = [1+1−1+1+1+1−1−1]/8 = 1
$S_2\bullet C$ = [2+0+0+0+2+2+0+2]/8 = 1
$S_3\bullet C$ = [0+0+2+2+0−2+0−2]/8 = 0
$S_4\bullet C$ = [1+1+3+3+1−1+1−1]/8 = 1
$S_5\bullet C$ = [4+0+2+0+2+0−2+2]/8 = 1
$S_6\bullet C$ = [2−2+0−2+0−2−4+0]/8 = −1

(d)

Figure 2-28. (a) Chip sequences for four stations. (b) Signals the sequences represent (c) Six examples of transmissions. (d) Recovery of station C's signal.

During each bit time, a station can transmit a 1 (by sending its chip sequence), it can transmit a 0 (by sending the negative of its chip sequence), or it can be silent and transmit nothing. We assume for now that all stations are synchronized in time, so all chip sequences begin at the same instant. When two or more stations transmit simultaneously, their bipolar sequences add linearly. For example, if in one chip period three stations output +1 and one station outputs −1, +2 will be received. One can think of this as signals that add as voltages superimposed on the channel: three stations output +1 V and one station outputs −1 V, so that 2 V is received. For instance, in Fig. 2-28(c) we see six examples of one or more stations transmitting 1 bit at the same time. In the first example, C transmits a 1 bit, so we just get C's chip sequence. In the second example, both B and C transmit 1 bits, so we get the sum of their bipolar chip sequences, namely:

$$(-1\ -1\ +1\ -1\ +1\ +1\ +1\ -1) + (-1\ +1\ -1\ +1\ +1\ +1\ -1\ -1) = (-2\ \ 0\ \ 0\ \ 0\ +2\ +2\ \ 0\ -2)$$

To recover the bit stream of an individual station, the receiver must know that station's chip sequence in advance. It does the recovery by computing the normalized inner product of the received chip sequence and the chip sequence of the station whose bit stream it is trying to recover. If the received chip sequence is $\mathbf{S}$ and the receiver is trying to listen to a station whose chip sequence is $\mathbf{C}$, it just computes the normalized inner product, $\mathbf{S}\bullet\mathbf{C}$.

To see why this works, just imagine that two stations, A and C, both transmit a 1 bit at the same time that B transmits a 0 bit, as is the case in the third example. The receiver sees the sum, $\mathbf{S} = \mathbf{A} + \overline{\mathbf{B}} + \mathbf{C}$, and computes

$$\mathbf{S} \bullet \mathbf{C} = (\mathbf{A} + \overline{\mathbf{B}} + \mathbf{C}) \bullet \mathbf{C} = \mathbf{A} \bullet \mathbf{C} + \overline{\mathbf{B}} \bullet \mathbf{C} + \mathbf{C} \bullet \mathbf{C} = 0 + 0 + 1 = 1$$

The first two terms vanish because all pairs of chip sequences have been carefully chosen to be orthogonal, as shown in Eq. (2-5). Now it should be clear why this property must be imposed on the chip sequences.

To make the decoding process more concrete, we show six examples in Fig. 2-28(d). Suppose that the receiver is interested in extracting the bit sent by station C from each of the six signals S_1 through S_6. It calculates the bit by summing the pairwise products of the received $\mathbf{S}$ and the $\mathbf{C}$ vector of Fig. 2-28(a) and then taking 1/8 of the result (since $m = 8$ here). The examples include cases where C is silent, sends a 1 bit, and sends a 0 bit, individually and in combination with other transmissions. As shown, the correct bit is decoded each time. It is just like speaking French.

In principle, given enough computing capacity, the receiver can listen to all the senders at once by running the decoding algorithm for each of them in parallel. In real life, suffice it to say that this is easier said than done, and it is useful to know which senders might be transmitting.

In the ideal, noiseless CDMA system we have studied here, the number of stations that send concurrently can be made arbitrarily large by using longer chip sequences. For 2^n stations, Walsh codes can provide 2^n orthogonal chip sequences of length 2^n. However, one significant limitation is that we have assumed that all the chips are synchronized in time at the receiver. This synchronization is not even approximately true in some applications, such as cellular networks (in which CDMA has been widely deployed starting in the 1990s). It leads to different designs. We will return to this topic later in the chapter and describe how asynchronous CDMA differs from synchronous CDMA.

As well as cellular networks, CDMA is used by satellites and cable networks. We have glossed over many complicating factors in this brief introduction. Engineers who want to gain a deep understanding of CDMA should read Viterbi (1995) and Lee and Miller (1998). These references require quite a bit of background in communication engineering, however.

2.6 THE PUBLIC SWITCHED TELEPHONE NETWORK

When two computers owned by the same company or organization and located close to each other need to communicate, it is often easiest just to run a cable between them. LANs work this way. However, when the distances are large or there are many computers or the cables have to pass through a public road or other public right of way, the costs of running private cables are usually prohibitive.

Furthermore, in just about every country in the world, stringing private transmission lines across (or underneath) public property is also illegal. Consequently, the network designers must rely on the existing telecommunication facilities.

These facilities, especially the **PSTN** (**Public Switched Telephone Network**), were usually designed many years ago, with a completely different goal in mind: transmitting the human voice in a more-or-less recognizable form. Their suitability for use in computer-computer communication is often marginal at best. To see the size of the problem, consider that a cheap commodity cable running between two computers can transfer data at 1 Gbps or more. In contrast, typical ADSL, the blazingly fast alternative to a telephone modem, runs at around 1 Mbps. The difference between the two is the difference between cruising in an airplane and taking a leisurely stroll.

Nonetheless, the telephone system is tightly intertwined with (wide area) computer networks, so it is worth devoting some time to study it in detail. The limiting factor for networking purposes turns out to be the "last mile" over which customers connect, not the trunks and switches inside the telephone network. This situation is changing with the gradual rollout of fiber and digital technology at the edge of the network, but it will take time and money. During the long wait, computer systems designers used to working with systems that give at least three orders of magnitude better performance have devoted much time and effort to figure out how to use the telephone network efficiently.

In the following sections we will describe the telephone system and show how it works. For additional information about the innards of the telephone system see Bellamy (2000).

2.6.1 Structure of the Telephone System

Soon after Alexander Graham Bell patented the telephone in 1876 (just a few hours ahead of his rival, Elisha Gray), there was an enormous demand for his new invention. The initial market was for the sale of telephones, which came in pairs. It was up to the customer to string a single wire between them. If a telephone owner wanted to talk to n other telephone owners, separate wires had to be strung to all n houses. Within a year, the cities were covered with wires passing over houses and trees in a wild jumble. It became immediately obvious that the model of connecting every telephone to every other telephone, as shown in Fig. 2-29(a), was not going to work.

To his credit, Bell saw this problem early on and formed the Bell Telephone Company, which opened its first switching office (in New Haven, Connecticut) in 1878. The company ran a wire to each customer's house or office. To make a call, the customer would crank the phone to make a ringing sound in the telephone company office to attract the attention of an operator, who would then manually connect the caller to the callee by using a short jumper cable to connect the caller to the callee. The model of a single switching office is illustrated in Fig. 2-29(b).

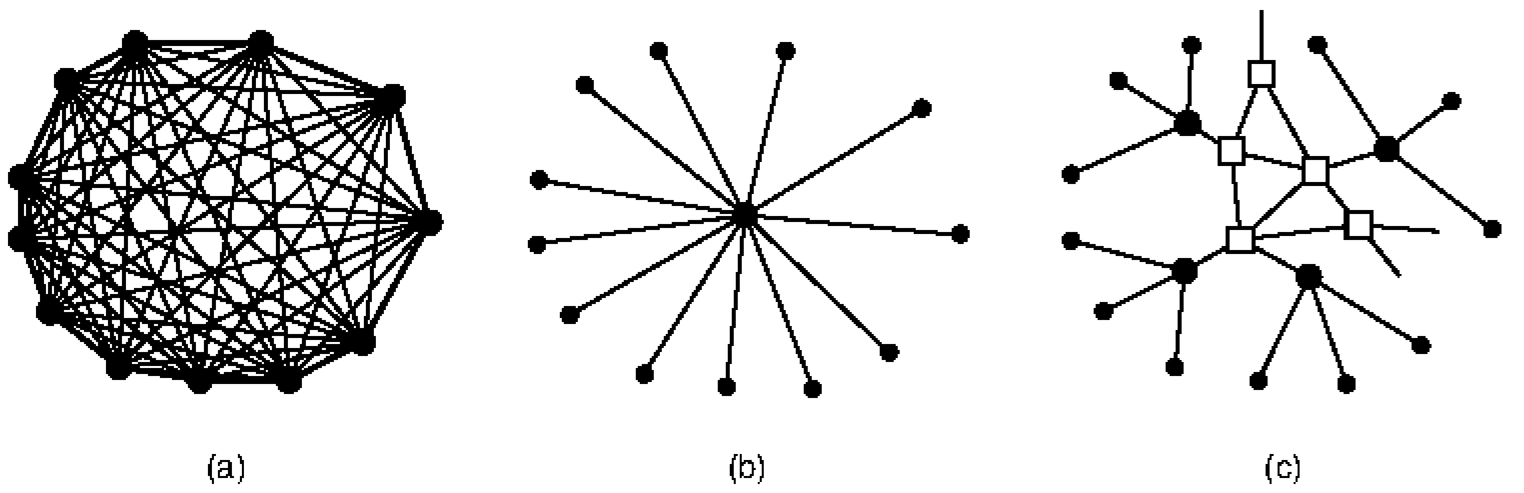


Figure 2-29. (a) Fully interconnected network. (b) Centralized switch. (c) Two-level hierarchy.

Pretty soon, Bell System switching offices were springing up everywhere and people wanted to make long-distance calls between cities, so the Bell System began to connect the switching offices. The original problem soon returned: to connect every switching office to every other switching office by means of a wire between them quickly became unmanageable, so second-level switching offices were invented. After a while, multiple second-level offices were needed, as illustrated in Fig. 2-29(c). Eventually, the hierarchy grew to five levels.

By 1890, the three major parts of the telephone system were in place: the switching offices, the wires between the customers and the switching offices (by now balanced, insulated, twisted pairs instead of open wires with an earth return), and the long-distance connections between the switching offices. For a short technical history of the telephone system, see Hawley (1991).

While there have been improvements in all three areas since then, the basic Bell System model has remained essentially intact for over 100 years. The following description is highly simplified but gives the essential flavor nevertheless. Each telephone has two copper wires coming out of it that go directly to the telephone company's nearest **end office** (also called a **local central office**). The distance is typically 1 to 10 km, being shorter in cities than in rural areas. In the United States alone there are about 22,000 end offices. The two-wire connections between each subscriber's telephone and the end office are known in the trade as the **local loop**. If the world's local loops were stretched out end to end, they would extend to the moon and back 1000 times.

At one time, 80% of AT&T's capital value was the copper in the local loops. AT&T was then, in effect, the world's largest copper mine. Fortunately, this fact was not well known in the investment community. Had it been known, some corporate raider might have bought AT&T, ended all telephone service in the United States, ripped out all the wire, and sold it to a copper refiner for a quick payback.

If a subscriber attached to a given end office calls another subscriber attached to the same end office, the switching mechanism within the office sets up a direct electrical connection between the two local loops. This connection remains intact for the duration of the call.

If the called telephone is attached to another end office, a different procedure has to be used. Each end office has a number of outgoing lines to one or more nearby switching centers, called **toll offices** (or, if they are within the same local area, **tandem offices**). These lines are called **toll connecting trunks**. The number of different kinds of switching centers and their topology varies from country to country depending on the country's telephone density.

If both the caller's and callee's end offices happen to have a toll connecting trunk to the same toll office (a likely occurrence if they are relatively close by), the connection may be established within the toll office. A telephone network consisting only of telephones (the small dots), end offices (the large dots), and toll offices (the squares) is shown in Fig. 2-29(c).

If the caller and callee do not have a toll office in common, a path will have to be established between two toll offices. The toll offices communicate with each other via high-bandwidth **intertoll trunks** (also called **interoffice trunks**). Prior to the 1984 breakup of AT&T, the U.S. telephone system used hierarchical routing to find a path, going to higher levels of the hierarchy until there was a switching office in common. This was then replaced with more flexible, nonhierarchical routing. Figure 2-30 shows how a long-distance connection might be routed.

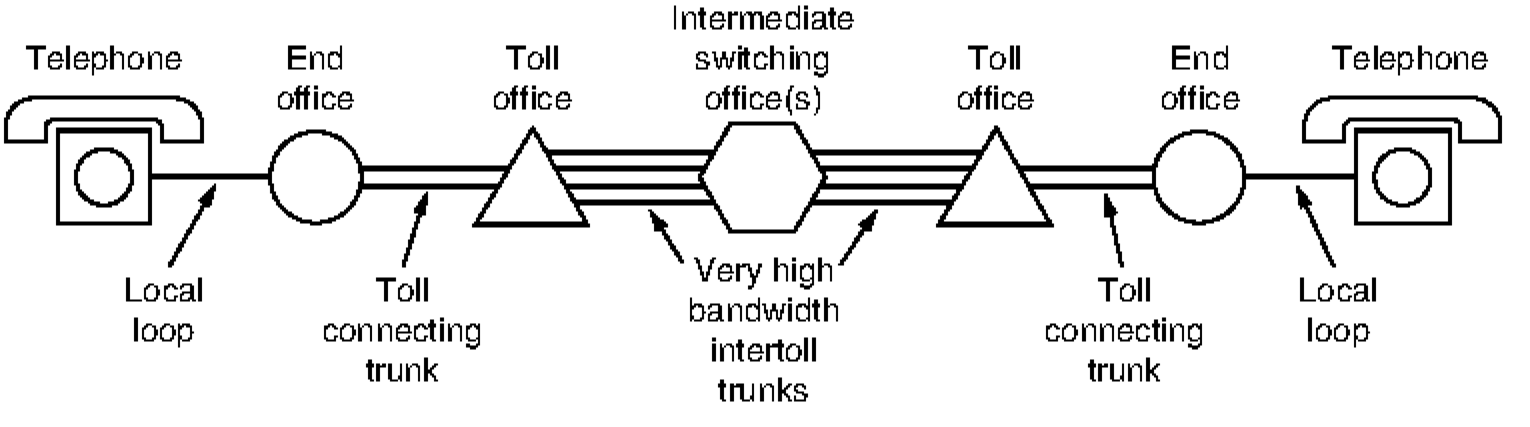


Figure 2-30. A typical circuit route for a long-distance call.

A variety of transmission media are used for telecommunication. Unlike modern office buildings, where the wiring is commonly Category 5, local loops to homes mostly consist of Category 3 twisted pairs, with fiber just starting to appear. Between switching offices, coaxial cables, microwaves, and especially fiber optics are widely used.

In the past, transmission throughout the telephone system was analog, with the actual voice signal being transmitted as an electrical voltage from source to destination. With the advent of fiber optics, digital electronics, and computers, all the trunks and switches are now digital, leaving the local loop as the last piece of analog technology in the system. Digital transmission is preferred because it is not necessary to accurately reproduce an analog waveform after it has passed through many amplifiers on a long call. Being able to correctly distinguish a 0 from a 1 is enough. This property makes digital transmission more reliable than analog. It is also cheaper and easier to maintain.

In summary, the telephone system consists of three major components:

1. Local loops (analog twisted pairs going to houses and businesses).
2. Trunks (digital fiber optic links connecting the switching offices).
3. Switching offices (where calls are moved from one trunk to another).

After a short digression on the politics of telephones, we will come back to each of these three components in some detail. The local loops provide everyone access to the whole system, so they are critical. Unfortunately, they are also the weakest link in the system. For the long-haul trunks, the main issue is how to collect multiple calls together and send them out over the same fiber. This calls for multiplexing, and we apply FDM and TDM to do it. Finally, there are two fundamentally different ways of doing switching; we will look at both.

2.6.2 The Politics of Telephones

For decades prior to 1984, the Bell System provided both local and long-distance service throughout most of the United States. In the 1970s, the U.S. Federal Government came to believe that this was an illegal monopoly and sued to break it up. The government won, and on January 1, 1984, AT&T was broken up into AT&T Long Lines, 23 **BOCs** (**Bell Operating Companies**), and a few other pieces. The 23 BOCs were grouped into seven regional BOCs (RBOCs) to make them economically viable. The entire nature of telecommunication in the United States was changed overnight by court order (*not* by an act of Congress).

The exact specifications of the divestiture were described in the so-called **MFJ** (**Modified Final Judgment**), an oxymoron if ever there was one—if the judgment could be modified, it clearly was not final. This event led to increased competition, better service, and lower long-distance rates for consumers and businesses. However, prices for local service rose as the cross subsidies from long-distance calling were eliminated and local service had to become self supporting. Many other countries have now introduced competition along similar lines.

Of direct relevance to our studies is that the new competitive framework caused a key technical feature to be added to the architecture of the telephone network. To make it clear who could do what, the United States was divided up into 164 **LATAs** (**Local Access and Transport Areas**). Very roughly, a LATA is about as big as the area covered by one area code. Within each LATA, there was one **LEC** (**Local Exchange Carrier**) with a monopoly on traditional telephone service within its area. The most important LECs were the BOCs, although some LATAs contained one or more of the 1500 independent telephone companies operating as LECs.

The new feature was that all inter-LATA traffic was handled by a different kind of company, an **IXC** (**IntereXchange Carrier**). Originally, AT&T Long Lines was the only serious IXC, but now there are well-established competitors such as Verizon and Sprint in the IXC business. One of the concerns at the breakup was to ensure that all the IXCs would be treated equally in terms of line quality, tariffs, and the number of digits their customers would have to dial to use them. The way this is handled is illustrated in Fig. 2-31. Here we see three example LATAs, each with several end offices. LATAs 2 and 3 also have a small hierarchy with tandem offices (intra-LATA toll offices).

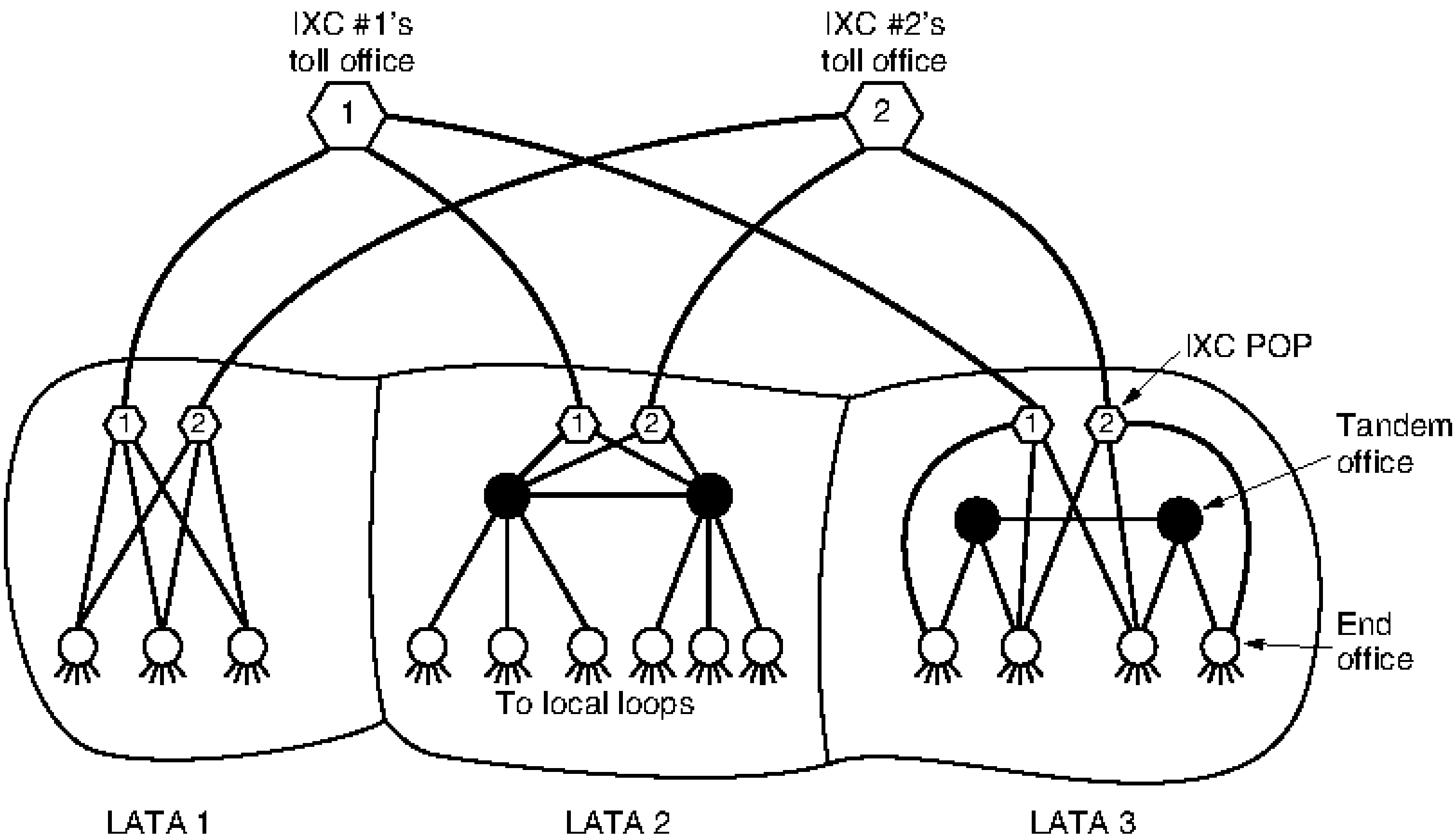


Figure 2-31. The relationship of LATAs, LECs, and IXCs. All the circles are LEC switching offices. Each hexagon belongs to the IXC whose number is in it.

Any IXC that wishes to handle calls originating in a LATA can build a switching office called a **POP** (**Point of Presence**) there. The LEC is required to connect each IXC to every end office, either directly, as in LATAs 1 and 3, or indirectly, as in LATA 2. Furthermore, the terms of the connection, both technical and financial, must be identical for all IXCs. This requirement enables, a subscriber in, say, LATA 1, to choose which IXC to use for calling subscribers in LATA 3.

As part of the MFJ, the IXCs were forbidden to offer local telephone service and the LECs were forbidden to offer inter-LATA telephone service, although both were free to enter any other business, such as operating fried chicken restaurants. In 1984, that was a fairly unambiguous statement. Unfortunately, technology has a funny way of making the law obsolete. Neither cable television nor mobile phones were covered by the agreement. As cable television went from one way to two way and mobile phones exploded in popularity, both LECs and IXCs began buying up or merging with cable and mobile operators.

By 1995, Congress saw that trying to maintain a distinction between the various kinds of companies was no longer tenable and drafted a bill to preserve accessibility for competition but allow cable TV companies, local telephone companies, long-distance carriers, and mobile operators to enter one another's businesses. The idea was that any company could then offer its customers a single integrated package containing cable TV, telephone, and information services and that different companies would compete on service and price. The bill was enacted into law in February 1996 as a major overhaul of telecommunications regulation. As a result, some BOCs became IXCs and some other companies, such as cable television operators, began offering local telephone service in competition with the LECs.

One interesting property of the 1996 law is the requirement that LECs implement **local number portability**. This means that a customer can change local telephone companies without having to get a new telephone number. Portability for mobile phone numbers (and between fixed and mobile lines) followed suit in 2003. These provisions removed a huge hurdle for many people, making them much more inclined to switch LECs. As a result, the U.S. telecommunications landscape became much more competitive, and other countries have followed suit. Often other countries wait to see how this kind of experiment works out in the U.S. If it works well, they do the same thing; if it works badly, they try something else.

2.6.3 The Local Loop: Modems, ADSL, and Fiber

It is now time to start our detailed study of how the telephone system works. Let us begin with the part that most people are familiar with: the two-wire local loop coming from a telephone company end office into houses. The local loop is also frequently referred to as the "last mile," although the length can be up to several miles. It has carried analog information for over 100 years and is likely to continue doing so for some years to come, due to the high cost of converting to digital.

Much effort has been devoted to squeezing data networking out of the copper local loops that are already deployed. Telephone modems send digital data between computers over the narrow channel the telephone network provides for a voice call. They were once widely used, but have been largely displaced by broadband technologies such as ADSL that. reuse the local loop to send digital data from a customer to the end office, where they are siphoned off to the Internet.

Both modems and ADSL must deal with the limitations of old local loops: relatively narrow bandwidth, attenuation and distortion of signals, and susceptibility to electrical noise such as crosstalk.

In some places, the local loop has been modernized by installing optical fiber to (or very close to) the home. Fiber is the way of the future. These installations support computer networks from the ground up, with the local loop having ample bandwidth for data services. The limiting factor is what people will pay, not the physics of the local loop.

In this section we will study the local loop, both old and new. We will cover telephone modems, ADSL, and fiber to the home.

Telephone Modems

To send bits over the local loop, or any other physical channel for that matter, they must be converted to analog signals that can be transmitted over the channel. This conversion is accomplished using the methods for digital modulation that we studied in the previous section. At the other end of the channel, the analog signal is converted back to bits.

A device that converts between a stream of digital bits and an analog signal that represents the bits is called a **modem**, which is short for ''*mo*dulator *dem*odulator.'' Modems come in many varieties: telephone modems, DSL modems, cable modems, wireless modems, etc. The modem may be built into the computer (which is now common for telephone modems) or be a separate box (which is common for DSL and cable modems). Logically, the modem is inserted between the (digital) computer and the (analog) telephone system, as seen in Fig. 2-32.

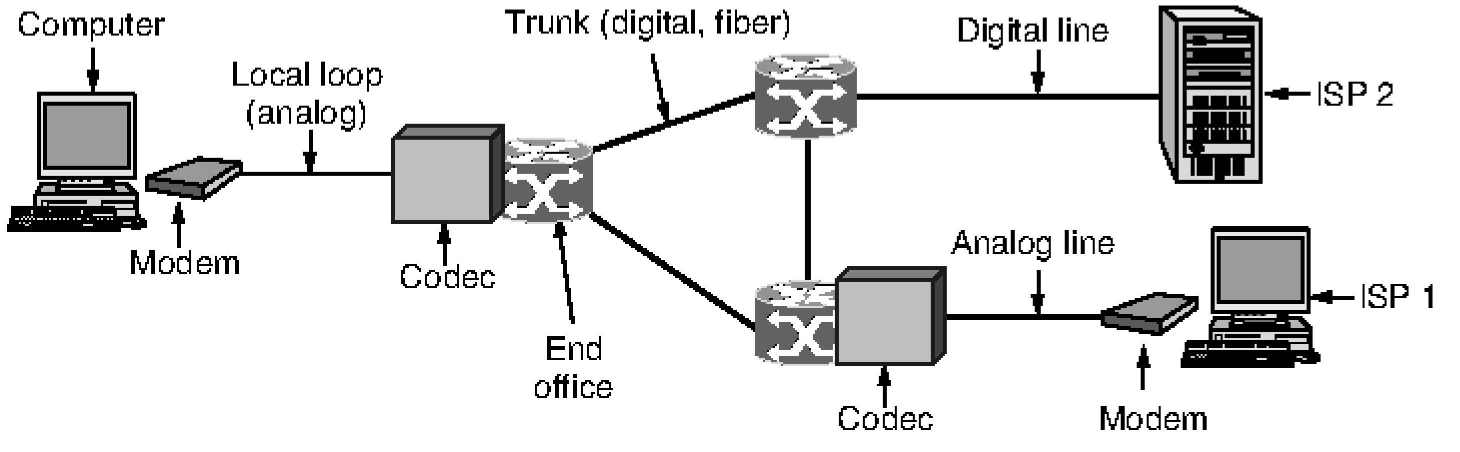


Figure 2-32. The use of both analog and digital transmission for a computer-to-computer call. Conversion is done by the modems and codecs.

Telephone modems are used to send bits between two computers over a voice-grade telephone line, in place of the conversation that usually fills the line. The main difficulty in doing so is that a voice-grade telephone line is limited to 3100 Hz, about what is sufficient to carry a conversation. This bandwidth is more than four orders of magnitude less than the bandwidth that is used for Ethernet or

802.11 (WiFi). Unsurprisingly, the data rates of telephone modems are also four orders of magnitude less than that of Ethernet and 802.11.

Let us run the numbers to see why this is the case. The Nyquist theorem tells us that even with a perfect 3000-Hz line (which a telephone line is decidedly not), there is no point in sending symbols at a rate faster than 6000 baud. In practice, most modems send at a rate of 2400 symbols/sec, or 2400 baud, and focus on getting multiple bits per symbol while allowing traffic in both directions at the same time (by using different frequencies for different directions).

The humble 2400-bps modem uses 0 volts for a logical 0 and 1 volt for a logical 1, with 1 bit per symbol. One step up, it can use four different symbols, as in the four phases of QPSK, so with 2 bits/symbol it can get a data rate of 4800 bps.

A long progression of higher rates has been achieved as technology has improved. Higher rates require a larger set of symbols or **constellation**. With many symbols, even a small amount of noise in the detected amplitude or phase can result in an error. To reduce the chance of errors, standards for the higher-speed modems use some of the symbols for error correction. The schemes are known as **TCM** (**Trellis Coded Modulation**) (Ungerboeck, 1987).

The **V.32** modem standard uses 32 constellation points to transmit 4 data bits and 1 check bit per symbol at 2400 baud to achieve 9600 bps with error correction. The next step above 9600 bps is 14,400 bps. It is called **V.32 bis** and transmits 6 data bits and 1 check bit per symbol at 2400 baud. Then comes **V.34**, which achieves 28,800 bps by transmitting 12 data bits/symbol at 2400 baud. The constellation now has thousands of points. The final modem in this series is **V.34 bis** which uses 14 data bits/symbol at 2400 baud to achieve 33,600 bps.

Why stop here? The reason that standard modems stop at 33,600 is that the Shannon limit for the telephone system is about 35 kbps based on the average length of local loops and the quality of these lines. Going faster than this would violate the laws of physics (department of thermodynamics).

However, there is one way we can change the situation. At the telephone company end office, the data are converted to digital form for transmission within the telephone network (the core of the telephone network converted from analog to digital long ago). The 35-kbps limit is for the situation in which there are two local loops, one at each end. Each of these adds noise to the signal. If we could get rid of one of these local loops, we would increase the SNR and the maximum rate would be doubled.

This approach is how 56-kbps modems are made to work. One end, typically an ISP, gets a high-quality digital feed from the nearest end office. Thus, when one end of the connection is a high-quality signal, as it is with most ISPs now, the maximum data rate can be as high as 70 kbps. Between two home users with modems and analog lines, the maximum is still 33.6 kbps.

The reason that 56-kbps modems (rather than 70-kbps modems) are in use has to do with the Nyquist theorem. A telephone channel is carried inside the telephone system as digital samples. Each telephone channel is 4000 Hz wide when the guard bands are included. The number of samples per second needed to reconstruct it is thus 8000. The number of bits per sample in the U.S. is 8, one of which may be used for control purposes, allowing 56,000 bits/sec of user data. In Europe, all 8 bits are available to users, so 64,000-bit/sec modems could have been used, but to get international agreement on a standard, 56,000 was chosen.

The end result is the **V.90** and **V.92** modem standards. They provide for a 56-kbps downstream channel (ISP to user) and a 33.6-kbps and 48-kbps upstream channel (user to ISP), respectively. The asymmetry is because there is usually more data transported from the ISP to the user than the other way. It also means that more of the limited bandwidth can be allocated to the downstream channel to increase the chances of it actually working at 56 kbps.

Digital Subscriber Lines

When the telephone industry finally got to 56 kbps, it patted itself on the back for a job well done. Meanwhile, the cable TV industry was offering speeds up to 10 Mbps on shared cables. As Internet access became an increasingly important part of their business, the telephone companies (LECs) began to realize they needed a more competitive product. Their answer was to offer new digital services over the local loop.

Initially, there were many overlapping high-speed offerings, all under the general name of **xDSL** (**Digital Subscriber Line**), for various *x*. Services with more bandwidth than standard telephone service are sometimes called **broadband**, although the term really is more of a marketing concept than a specific technical concept. Later, we will discuss what has become the most popular of these services, **ADSL** (**Asymmetric DSL**). We will also use the term DSL or xDSL as shorthand for all flavors.

The reason that modems are so slow is that telephones were invented for carrying the human voice and the entire system has been carefully optimized for this purpose. Data have always been stepchildren. At the point where each local loop terminates in the end office, the wire runs through a filter that attenuates all frequencies below 300 Hz and above 3400 Hz. The cutoff is not sharp—300 Hz and 3400 Hz are the 3-dB points—so the bandwidth is usually quoted as 4000 Hz even though the distance between the 3 dB points is 3100 Hz. Data on the wire are thus also restricted to this narrow band.

The trick that makes xDSL work is that when a customer subscribes to it, the incoming line is connected to a different kind of switch, one that does not have this filter, thus making the entire capacity of the local loop available. The limiting factor then becomes the physics of the local loop, which supports roughly 1 MHz, not the artificial 3100 Hz bandwidth created by the filter.

Unfortunately, the capacity of the local loop falls rather quickly with distance from the end office as the signal is increasingly degraded along the wire. It also depends on the thickness and general quality of the twisted pair. A plot of the potential bandwidth as a function of distance is given in Fig. 2-33. This figure assumes that all the other factors are optimal (new wires, modest bundles, etc.).

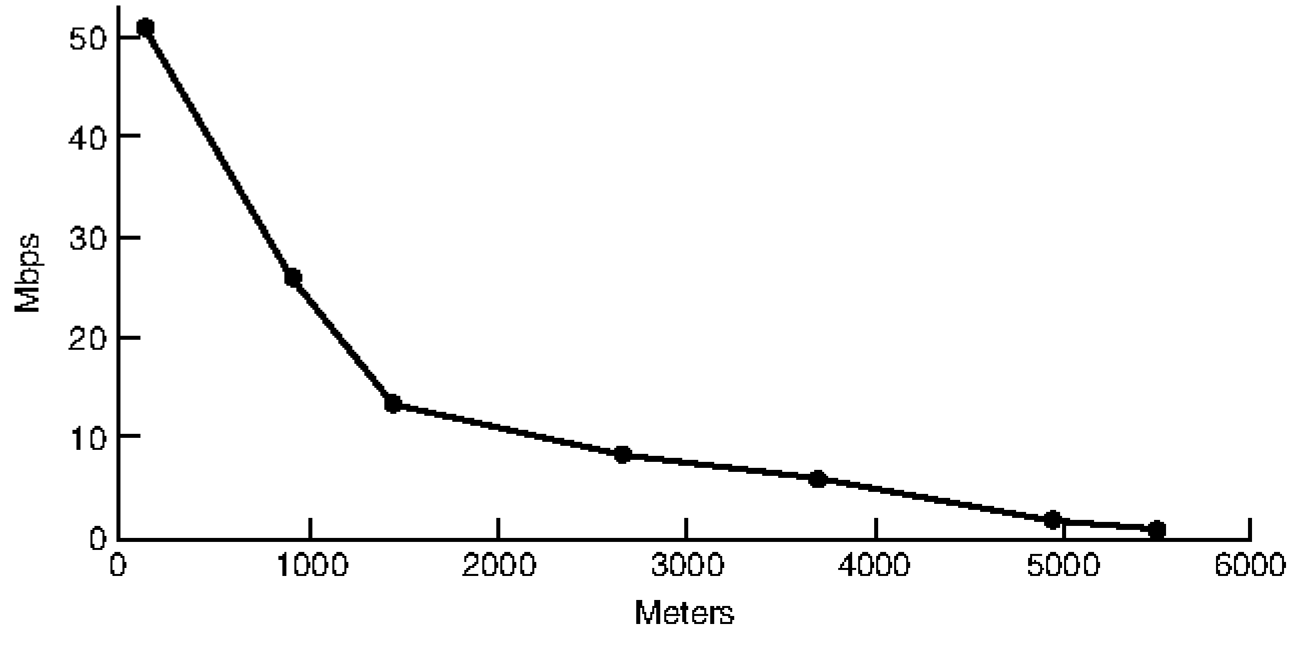


Figure 2-33. Bandwidth versus distance over Category 3 UTP for DSL.

The implication of this figure creates a problem for the telephone company. When it picks a speed to offer, it is simultaneously picking a radius from its end offices beyond which the service cannot be offered. This means that when distant customers try to sign up for the service, they may be told "Thanks a lot for your interest, but you live 100 meters too far from the nearest end office to get this service. Could you please move?" The lower the chosen speed is, the larger the radius and the more customers are covered. But the lower the speed, the less attractive the service is and the fewer the people who will be willing to pay for it. This is where business meets technology.

The xDSL services have all been designed with certain goals in mind. First, the services must work over the existing Category 3 twisted pair local loops. Second, they must not affect customers' existing telephones and fax machines. Third, they must be much faster than 56 kbps. Fourth, they should be always on, with just a monthly charge and no per-minute charge.

To meet the technical goals, the available 1.1 MHz spectrum on the local loop is divided into 256 independent channels of 4312.5 Hz each. This arrangement is shown in Fig. 2-34. The OFDM scheme, which we saw in the previous section, is used to send data over these channels, though it is often called **DMT** (**Discrete MultiTone**) in the context of ADSL. Channel 0 is used for **POTS** (**Plain Old Telephone Service**). Channels 1–5 are not used, to keep the voice and data signals from interfering with each other. Of the remaining 250 channels, one is used for upstream control and one is used for downstream control. The rest are available for user data.

In principle, each of the remaining channels can be used for a full-duplex data stream, but harmonics, crosstalk, and other effects keep practical systems well

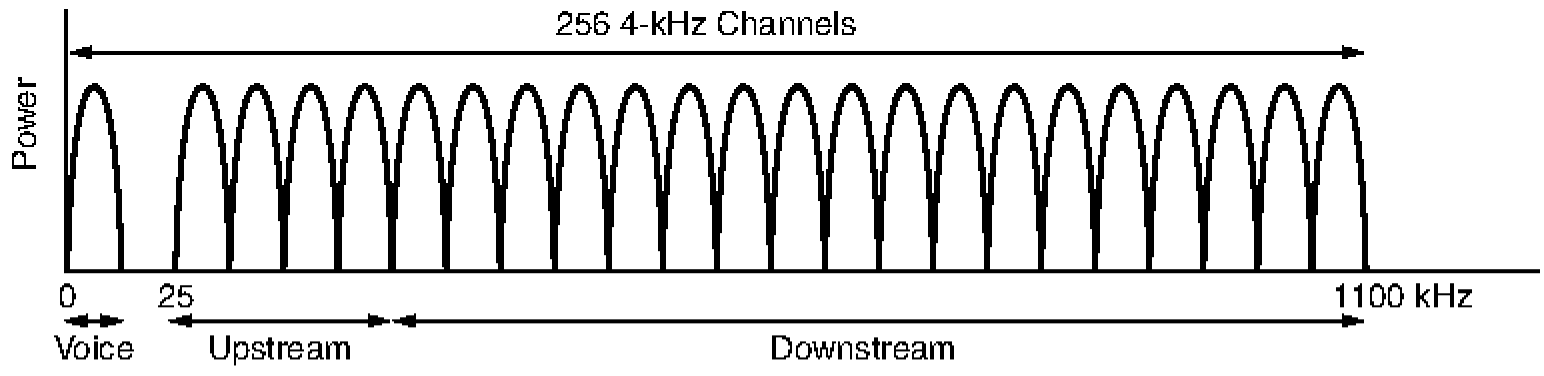


Figure 2-34. Operation of ADSL using discrete multitone modulation.

below the theoretical limit. It is up to the provider to determine how many channels are used for upstream and how many for downstream. A 50/50 mix of upstream and downstream is technically possible, but most providers allocate something like 80–90% of the bandwidth to the downstream channel since most users download more data than they upload. This choice gives rise to the "A" in ADSL. A common split is 32 channels for upstream and the rest downstream. It is also possible to have a few of the highest upstream channels be bidirectional for increased bandwidth, although making this optimization requires adding a special circuit to cancel echoes.

The international ADSL standard, known as **G.dmt**, was approved in 1999. It allows speeds of as much as 8 Mbps downstream and 1 Mbps upstream. It was superseded by a second generation in 2002, called ADSL2, with various improvements to allow speeds of as much as 12 Mbps downstream and 1 Mbps upstream. Now we have ADSL2+, which doubles the downstream speed to 24 Mbps by doubling the bandwidth to use 2.2 MHz over the twisted pair.

However, the numbers quoted here are best-case speeds for good lines close (within 1 to 2 km) to the exchange. Few lines support these rates, and few providers offer these speeds. Typically, providers offer something like 1 Mbps downstream and 256 kbps upstream (standard service), 4 Mbps downstream and 1 Mbps upstream (improved service), and 8 Mbps downstream and 2 Mbps upstream (premium service).

Within each channel, QAM modulation is used at a rate of roughly 4000 symbols/sec. The line quality in each channel is constantly monitored and the data rate is adjusted by using a larger or smaller constellation, like those in Fig. 2-23. Different channels may have different data rates, with up to 15 bits per symbol sent on a channel with a high SNR, and down to 2, 1, or no bits per symbol sent on a channel with a low SNR depending on the standard.

A typical ADSL arrangement is shown in Fig. 2-35. In this scheme, a telephone company technician must install a **NID** (**Network Interface Device**) on the customer's premises. This small plastic box marks the end of the telephone company's property and the start of the customer's property. Close to the NID (or sometimes combined with it) is a **splitter**, an analog filter that separates the

0–4000-Hz band used by POTS from the data. The POTS signal is routed to the existing telephone or fax machine. The data signal is routed to an ADSL modem, which uses digital signal processing to implement OFDM. Since most ADSL modems are external, the computer must be connected to them at high speed. Usually, this is done using Ethernet, a USB cable, or 802.11.

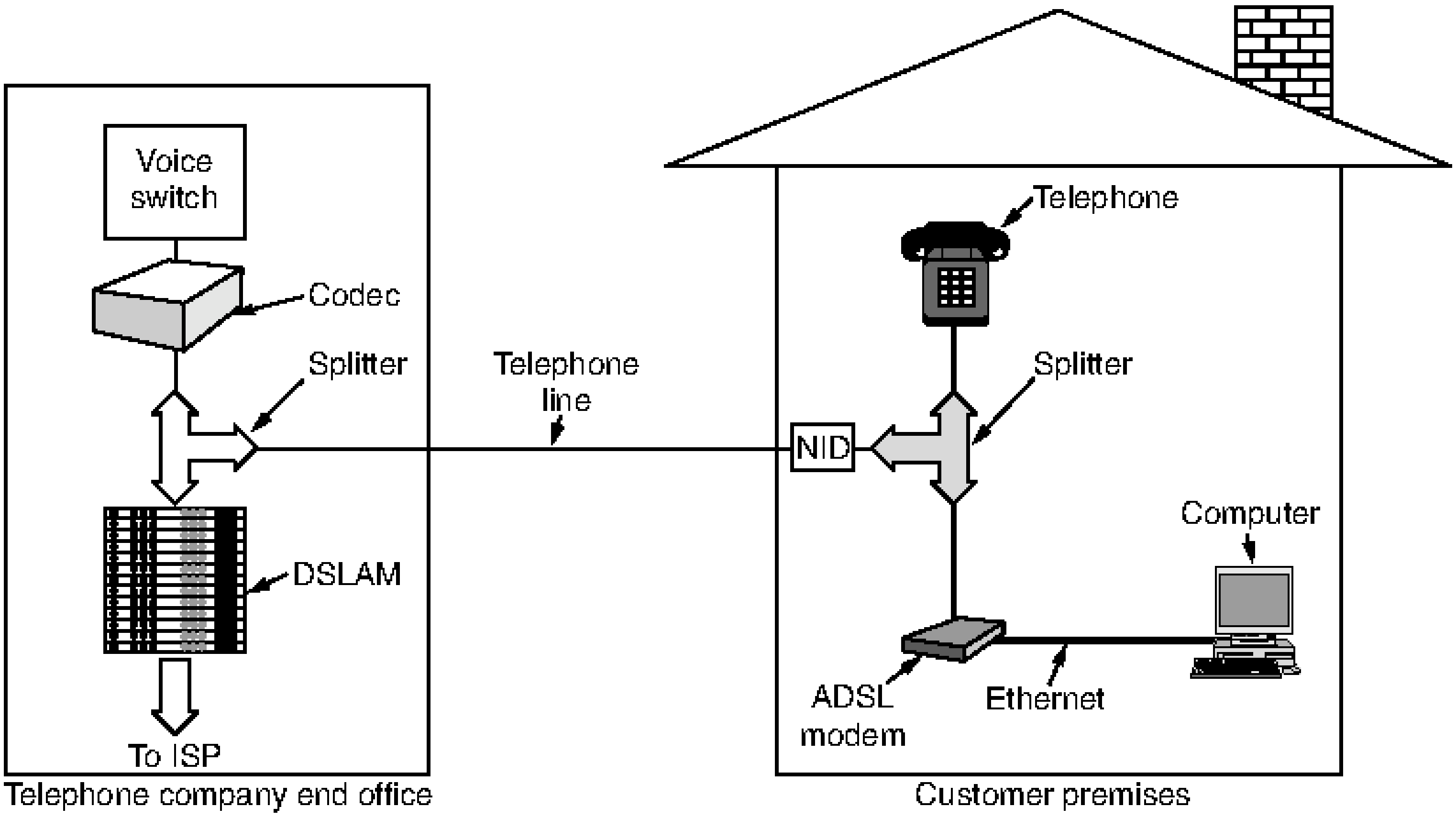


Figure 2-35. A typical ADSL equipment configuration.

At the other end of the wire, on the end office side, a corresponding splitter is installed. Here, the voice portion of the signal is filtered out and sent to the normal voice switch. The signal above 26 kHz is routed to a new kind of device called a **DSLAM** (**Digital Subscriber Line Access Multiplexer**), which contains the same kind of digital signal processor as the ADSL modem. Once the bits have been recovered from the signal, packets are formed and sent off to the ISP.

This complete separation between the voice system and ADSL makes it relatively easy for a telephone company to deploy ADSL. All that is needed is buying a DSLAM and splitter and attaching the ADSL subscribers to the splitter. Other high-bandwidth services (e.g., ISDN) require much greater changes to the existing switching equipment.

One disadvantage of the design of Fig. 2-35 is the need for a NID and splitter on the customer's premises. Installing these can only be done by a telephone company technician, necessitating an expensive "truck roll" (i.e., sending a technician to the customer's premises). Therefore, an alternative, splitterless design, informally called **G.lite**, has also been standardized. It is the same as Fig. 2-35 but without the customer's splitter. The existing telephone line is used as is. The only difference is that a microfilter has to be inserted into each telephone jack between the telephone or ADSL modem and the wire. The microfilter for the telephone is a low-pass filter eliminating frequencies above 3400 Hz; the microfilter for the ADSL modem is a high-pass filter eliminating frequencies below 26 kHz. However, this system is not as reliable as having a splitter, so G.lite can be used only up to 1.5 Mbps (versus 8 Mbps for ADSL with a splitter). For more information about ADSL, see Starr (2003).

Fiber To The Home

Deployed copper local loops limit the performance of ADSL and telephone modems. To let them provide faster and better network services, telephone companies are upgrading local loops at every opportunity by installing optical fiber all the way to houses and offices. The result is called **FttH** (**Fiber To The Home**). While FttH technology has been available for some time, deployments only began to take off in 2005 with growth in the demand for high-speed Internet from customers used to DSL and cable who wanted to download movies. Around 4% of U.S. houses are now connected to FttH with Internet access speeds of up to 100 Mbps.

Several variations of the form "FttX" (where *X* stands for the basement, curb, or neighborhood) exist. They are used to note that the fiber deployment may reach close to the house. In this case, copper (twisted pair or coaxial cable) provides fast enough speeds over the last short distance. The choice of how far to lay the fiber is an economic one, balancing cost with expected revenue. In any case, the point is that optical fiber has crossed the traditional barrier of the "last mile." We will focus on FttH in our discussion.

Like the copper wires before it, the fiber local loop is passive. This means no powered equipment is required to amplify or otherwise process signals. The fiber simply carries signals between the home and the end office. This in turn reduces cost and improves reliability.

Usually, the fibers from the houses are joined together so that only a single fiber reaches the end office per group of up to 100 houses. In the downstream direction, optical splitters divide the signal from the end office so that it reaches all the houses. Encryption is needed for security if only one house should be able to decode the signal. In the upstream direction, optical combiners merge the signals from the houses into a single signal that is received at the end office.

This architecture is called a **PON** (**Passive Optical Network**), and it is shown in Fig. 2-36. It is common to use one wavelength shared between all the houses for downstream transmission, and another wavelength for upstream transmission.

Even with the splitting, the tremendous bandwidth and low attenuation of fiber mean that PONs can provide high rates to users over distances of up to 20 km. The actual data rates and other details depend on the type of PON. Two kinds are common. **GPONs** (**Gigabit-capable PONs**) come from the world of telecommunications, so they are defined by an ITU standard. **EPONs** (**Ethernet PONs**)

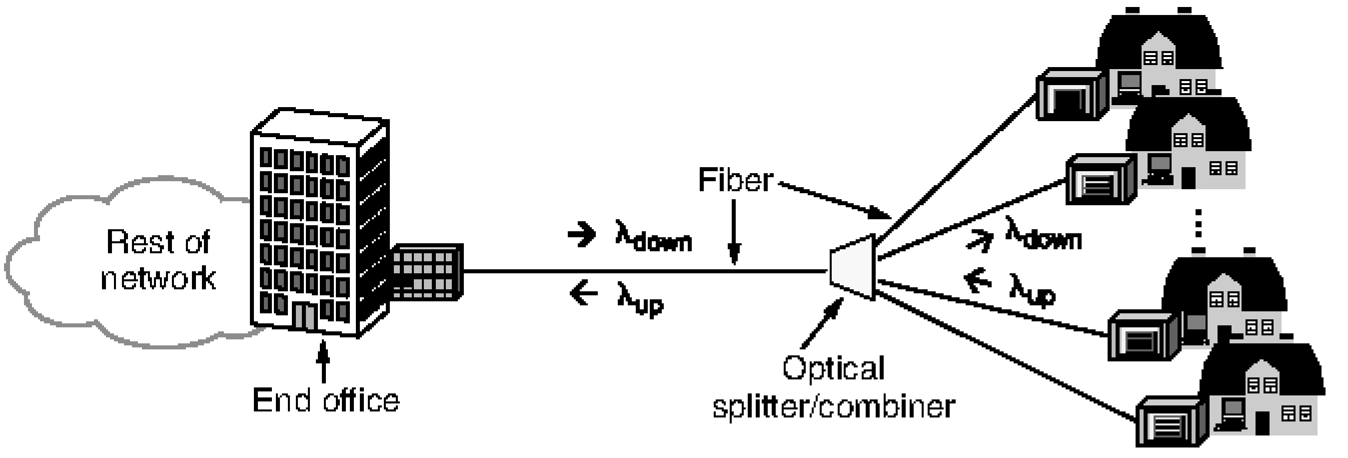


Figure 2-36. Passive optical network for Fiber To The Home.

are more in tune with the world of networking, so they are defined by an IEEE standard. Both run at around a gigabit and can carry traffic for different services, including Internet, video, and voice. For example, GPONs provide 2.4 Gbps downstream and 1.2 or 2.4 Gbps upstream.

Some protocol is needed to share the capacity of the single fiber at the end office between the different houses. The downstream direction is easy. The end office can send messages to each different house in whatever order it likes. In the upstream direction, however, messages from different houses cannot be sent at the same time, or different signals would collide. The houses also cannot hear each other's transmissions so they cannot listen before transmitting. The solution is that equipment at the houses requests and is granted time slots to use by equipment in the end office. For this to work, there is a ranging process to adjust the transmission times from the houses so that all the signals received at the end office are synchronized. The design is similar to cable modems, which we cover later in this chapter. For more information on the future of PONs, see Grobe and Elbers (2008).

2.6.4 Trunks and Multiplexing

Trunks in the telephone network are not only much faster than the local loops, they are different in two other respects. The core of the telephone network carries digital information, not analog information; that is, bits not voice. This necessitates a conversion at the end office to digital form for transmission over the long-haul trunks. The trunks carry thousands, even millions, of calls simultaneously. This sharing is important for achieving economies of scale, since it costs essentially the same amount of money to install and maintain a high-bandwidth trunk as a low-bandwidth trunk between two switching offices. It is accomplished with versions of TDM and FDM multiplexing.

Below we will briefly examine how voice signals are digitized so that they can be transported by the telephone network. After that, we will see how TDM is used to carry bits on trunks, including the TDM system used for fiber optics

(SONET). Then we will turn to FDM as it is applied to fiber optics, which is called wavelength division multiplexing.

Digitizing Voice Signals

Early in the development of the telephone network, the core handled voice calls as analog information. FDM techniques were used for many years to multiplex 4000-Hz voice channels (comprised of 3100 Hz plus guard bands) into larger and larger units. For example, 12 calls in the 60 kHz–to–108 kHz band is known as a **group** and five groups (a total of 60 calls) are known as a **supergroup**, and so on. These FDM methods are still used over some copper wires and microwave channels. However, FDM requires analog circuitry and is not amenable to being done by a computer. In contrast, TDM can be handled entirely by digital electronics, so it has become far more widespread in recent years. Since TDM can only be used for digital data and the local loops produce analog signals, a conversion is needed from analog to digital in the end office, where all the individual local loops come together to be combined onto outgoing trunks.

The analog signals are digitized in the end office by a device called a **codec** (short for "*co*der-*dec*oder"). The codec makes 8000 samples per second (125 μsec/sample) because the Nyquist theorem says that this is sufficient to capture all the information from the 4-kHz telephone channel bandwidth. At a lower sampling rate, information would be lost; at a higher one, no extra information would be gained. Each sample of the amplitude of the signal is quantized to an 8-bit number.

This technique is called **PCM** (**Pulse Code Modulation**). It forms the heart of the modern telephone system. As a consequence, virtually all time intervals within the telephone system are multiples of 125 μsec. The standard uncompressed data rate for a voice-grade telephone call is thus 8 bits every 125 μsec, or 64 kbps.

At the other end of the call, an analog signal is recreated from the quantized samples by playing them out (and smoothing them) over time. It will not be exactly the same as the original analog signal, even though we sampled at the Nyquist rate, because the samples were quantized. To reduce the error due to quantization, the quantization levels are unevenly spaced. A logarithmic scale is used that gives relatively more bits to smaller signal amplitudes and relatively fewer bits to large signal amplitudes. In this way the error is proportional to the signal amplitude.

Two versions of quantization are widely used: **μ-law**, used in North America and Japan, and **A-law**, used in Europe and the rest of the world. Both versions are specified in standard ITU G.711. An equivalent way to think about this process is to imagine that the dynamic range of the signal (or the ratio between the largest and smallest possible values) is compressed before it is (evenly) quantized, and then expanded when the analog signal is recreated. For this reason it is called

companding. It is also possible to compress the samples after they are digitized so that they require much less than 64 kbps. However, we will leave this topic for when we explore audio applications such as voice over IP.

Time Division Multiplexing

TDM based on PCM is used to carry multiple voice calls over trunks by sending a sample from each call every 125 μsec. When digital transmission began emerging as a feasible technology, ITU (then called CCITT) was unable to reach agreement on an international standard for PCM. Consequently, a variety of incompatible schemes are now in use in different countries around the world.

The method used in North America and Japan is the **T1** carrier, depicted in Fig. 2-37. (Technically speaking, the format is called DS1 and the carrier is called T1, but following widespread industry tradition, we will not make that subtle distinction here.) The T1 carrier consists of 24 voice channels multiplexed together. Each of the 24 channels, in turn, gets to insert 8 bits into the output stream.

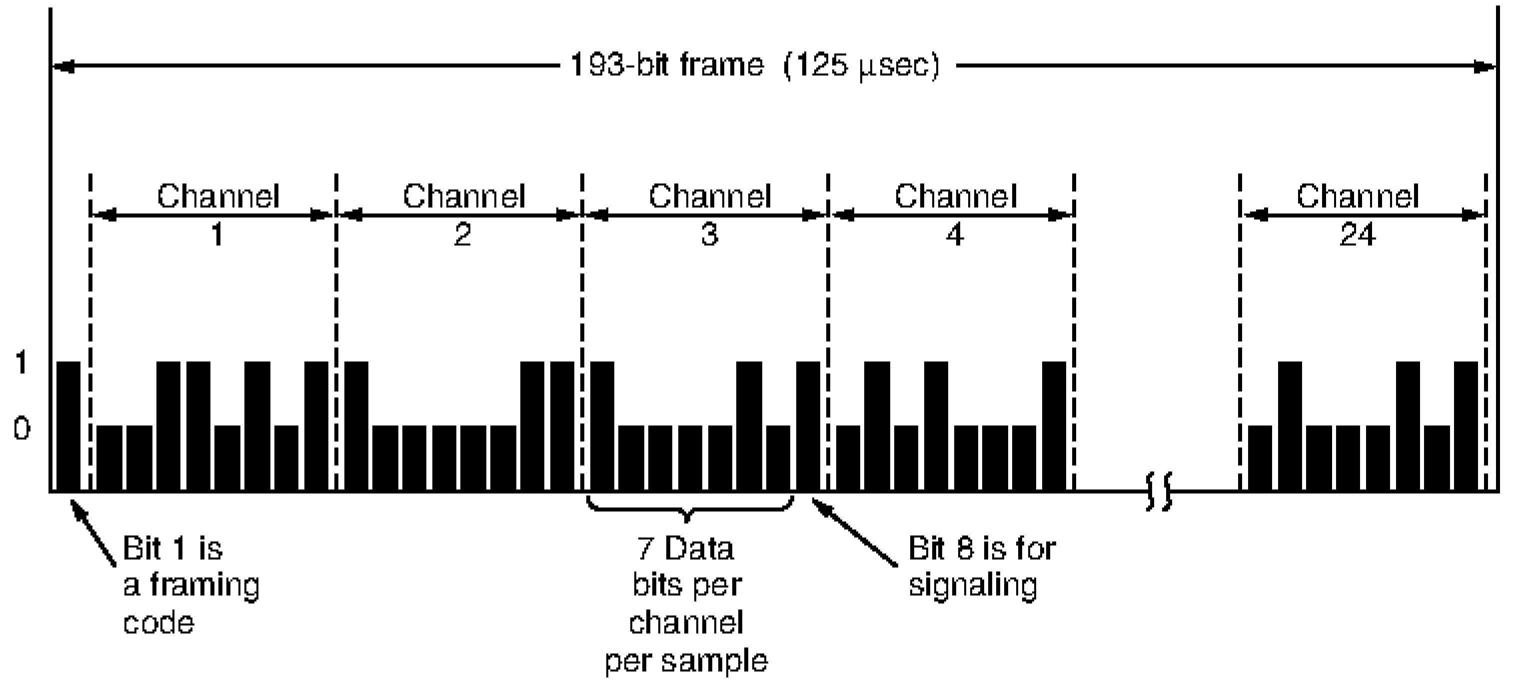


Figure 2-37. The T1 carrier (1.544 Mbps).

A frame consists of $24 \times 8 = 192$ bits plus one extra bit for control purposes, yielding 193 bits every 125 μsec. This gives a gross data rate of 1.544 Mbps, of which 8 kbps is for signaling. The 193rd bit is used for frame synchronization and signaling. In one variation, the 193rd bit is used across a group of 24 frames called an **extended superframe**. Six of the bits, in the 4th, 8th, 12th, 16th, 20th, and 24th positions, take on the alternating pattern 001011 . . . . Normally, the receiver keeps checking for this pattern to make sure that it has not lost synchronization. Six more bits are used to send an error check code to help the receiver confirm that it is synchronized. If it does get out of sync, the receiver can scan for the pattern and validate the error check code to get resynchronized. The remaining 12 bits are used for control information for operating and maintaining the network, such as performance reporting from the remote end.

The T1 format has several variations. The earlier versions sent signaling information **in-band**, meaning in the same channel as the data, by using some of the data bits. This design is one form of **channel-associated signaling**, because each channel has its own private signaling subchannel. In one arrangement, the least significant bit out of an 8-bit sample on each channel is used in every sixth frame. It has the colorful name of **robbed-bit signaling**. The idea is that a few stolen bits will not matter for voice calls. No one will hear the difference.

For data, however, it is another story. Delivering the wrong bits is unhelpful, to say the least. If older versions of T1 are used to carry data, only 7 of 8 bits, or 56 kbps can be used in each of the 24 channels. Instead, newer versions of T1 provide clear channels in which all of the bits may be used to send data. Clear channels are what businesses who lease a T1 line want when they send data across the telephone network in place of voice samples. Signaling for any voice calls is then handled **out-of-band**, meaning in a separate channel from the data. Often, the signaling is done with **common-channel signaling** in which there is a shared signaling channel. One of the 24 channels may be used for this purpose.

Outside North America and Japan, the 2.048-Mbps **E1** carrier is used instead of T1. This carrier has 32 8-bit data samples packed into the basic 125-μsec frame. Thirty of the channels are used for information and up to two are used for signaling. Each group of four frames provides 64 signaling bits, half of which are used for signaling (whether channel-associated or common-channel) and half of which are used for frame synchronization or are reserved for each country to use as it wishes.

Time division multiplexing allows multiple T1 carriers to be multiplexed into higher-order carriers. Figure 2-38 shows how this can be done. At the left we see four T1 channels being multiplexed into one T2 channel. The multiplexing at T2 and above is done bit for bit, rather than byte for byte with the 24 voice channels that make up a T1 frame. Four T1 streams at 1.544 Mbps should generate 6.176 Mbps, but T2 is actually 6.312 Mbps. The extra bits are used for framing and recovery in case the carrier slips. T1 and T3 are widely used by customers, whereas T2 and T4 are only used within the telephone system itself, so they are not well known.

At the next level, seven T2 streams are combined bitwise to form a T3 stream. Then six T3 streams are joined to form a T4 stream. At each step a small amount of overhead is added for framing and recovery in case the synchronization between sender and receiver is lost.

Just as there is little agreement on the basic carrier between the United States and the rest of the world, there is equally little agreement on how it is to be multiplexed into higher-bandwidth carriers. The U.S. scheme of stepping up by 4, 7, and 6 did not strike everyone else as the way to go, so the ITU standard calls for multiplexing four streams into one stream at each level. Also, the framing and

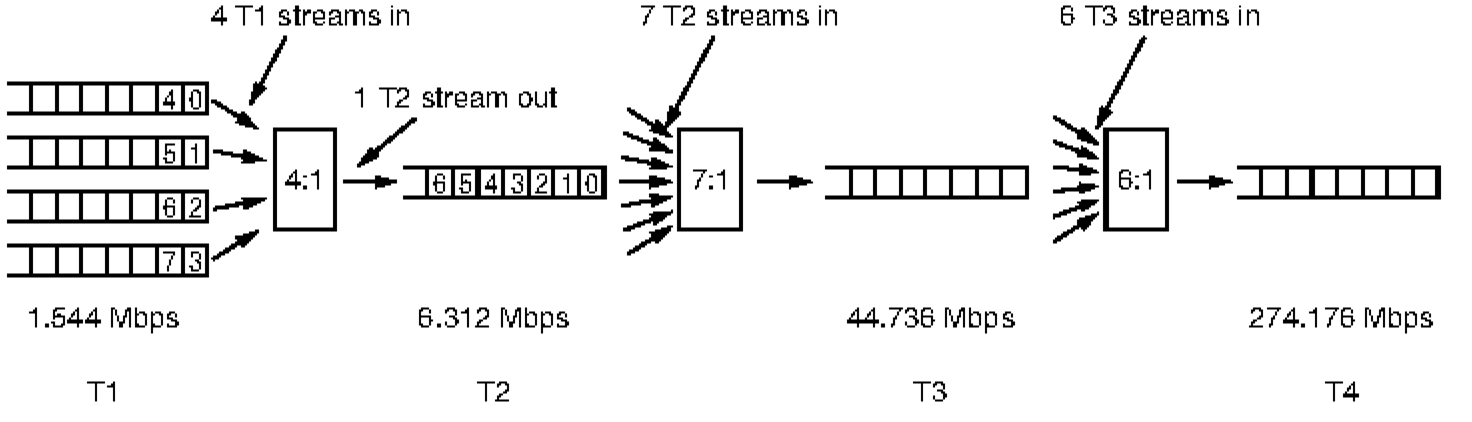


Figure 2-38. Multiplexing T1 streams into higher carriers.

recovery data are different in the U.S. and ITU standards. The ITU hierarchy for 32, 128, 512, 2048, and 8192 channels runs at speeds of 2.048, 8.848, 34.304, 139.264, and 565.148 Mbps.

SONET/SDH

In the early days of fiber optics, every telephone company had its own proprietary optical TDM system. After AT&T was broken up in 1984, local telephone companies had to connect to multiple long-distance carriers, all with different optical TDM systems, so the need for standardization became obvious. In 1985, Bellcore, the RBOC's research arm, began working on a standard, called **SONET** (**Synchronous Optical NETwork**).

Later, ITU joined the effort, which resulted in a SONET standard and a set of parallel ITU recommendations (G.707, G.708, and G.709) in 1989. The ITU recommendations are called **SDH** (**Synchronous Digital Hierarchy**) but differ from SONET only in minor ways. Virtually all the long-distance telephone traffic in the United States, and much of it elsewhere, now uses trunks running SONET in the physical layer. For additional information about SONET, see Bellamy (2000), Goralski (2002), and Shepard (2001).

The SONET design had four major goals. First and foremost, SONET had to make it possible for different carriers to interwork. Achieving this goal required defining a common signaling standard with respect to wavelength, timing, framing structure, and other issues.

Second, some means was needed to unify the U.S., European, and Japanese digital systems, all of which were based on 64-kbps PCM channels but combined them in different (and incompatible) ways.

Third, SONET had to provide a way to multiplex multiple digital channels. At the time SONET was devised, the highest-speed digital carrier actually used widely in the United States was T3, at 44.736 Mbps. T4 was defined, but not used much, and nothing was even defined above T4 speed. Part of SONET's mission was to continue the hierarchy to gigabits/sec and beyond. A standard way to multiplex slower channels into one SONET channel was also needed.

Fourth, SONET had to provide support for operations, administration, and maintenance (OAM), which are needed to manage the network. Previous systems did not do this very well.

An early decision was to make SONET a traditional TDM system, with the entire bandwidth of the fiber devoted to one channel containing time slots for the various subchannels. As such, SONET is a synchronous system. Each sender and receiver is tied to a common clock. The master clock that controls the system has an accuracy of about 1 part in 10^9. Bits on a SONET line are sent out at extremely precise intervals, controlled by the master clock.

The basic SONET frame is a block of 810 bytes put out every 125 μsec. Since SONET is synchronous, frames are emitted whether or not there are any useful data to send. Having 8000 frames/sec exactly matches the sampling rate of the PCM channels used in all digital telephony systems.

The 810-byte SONET frames are best described as a rectangle of bytes, 90 columns wide by 9 rows high. Thus, $8 \times 810 = 6480$ bits are transmitted 8000 times per second, for a gross data rate of 51.84 Mbps. This layout is the basic SONET channel, called **STS-1** (**Synchronous Transport Signal-1**). All SONET trunks are multiples of STS-1.

The first three columns of each frame are reserved for system management information, as illustrated in Fig. 2-39. In this block, the first three rows contain the section overhead; the next six contain the line overhead. The section overhead is generated and checked at the start and end of each section, whereas the line overhead is generated and checked at the start and end of each line.

A SONET transmitter sends back-to-back 810-byte frames, without gaps between them, even when there are no data (in which case it sends dummy data). From the receiver's point of view, all it sees is a continuous bit stream, so how does it know where each frame begins? The answer is that the first 2 bytes of each frame contain a fixed pattern that the receiver searches for. If it finds this pattern in the same place in a large number of consecutive frames, it assumes that it is in sync with the sender. In theory, a user could insert this pattern into the payload in a regular way, but in practice it cannot be done due to the multiplexing of multiple users into the same frame and other reasons.

The remaining 87 columns of each frame hold $87 \times 9 \times 8 \times 8000 = 50.112$ Mbps of user data. This user data could be voice samples, T1 and other carriers swallowed whole, or packets. SONET is simply a convenient container for transporting bits. The **SPE** (**Synchronous Payload Envelope**), which carries the user data does not always begin in row 1, column 4. The SPE can begin anywhere within the frame. A pointer to the first byte is contained in the first row of the line overhead. The first column of the SPE is the path overhead (i.e., the header for the end-to-end path sublayer protocol).

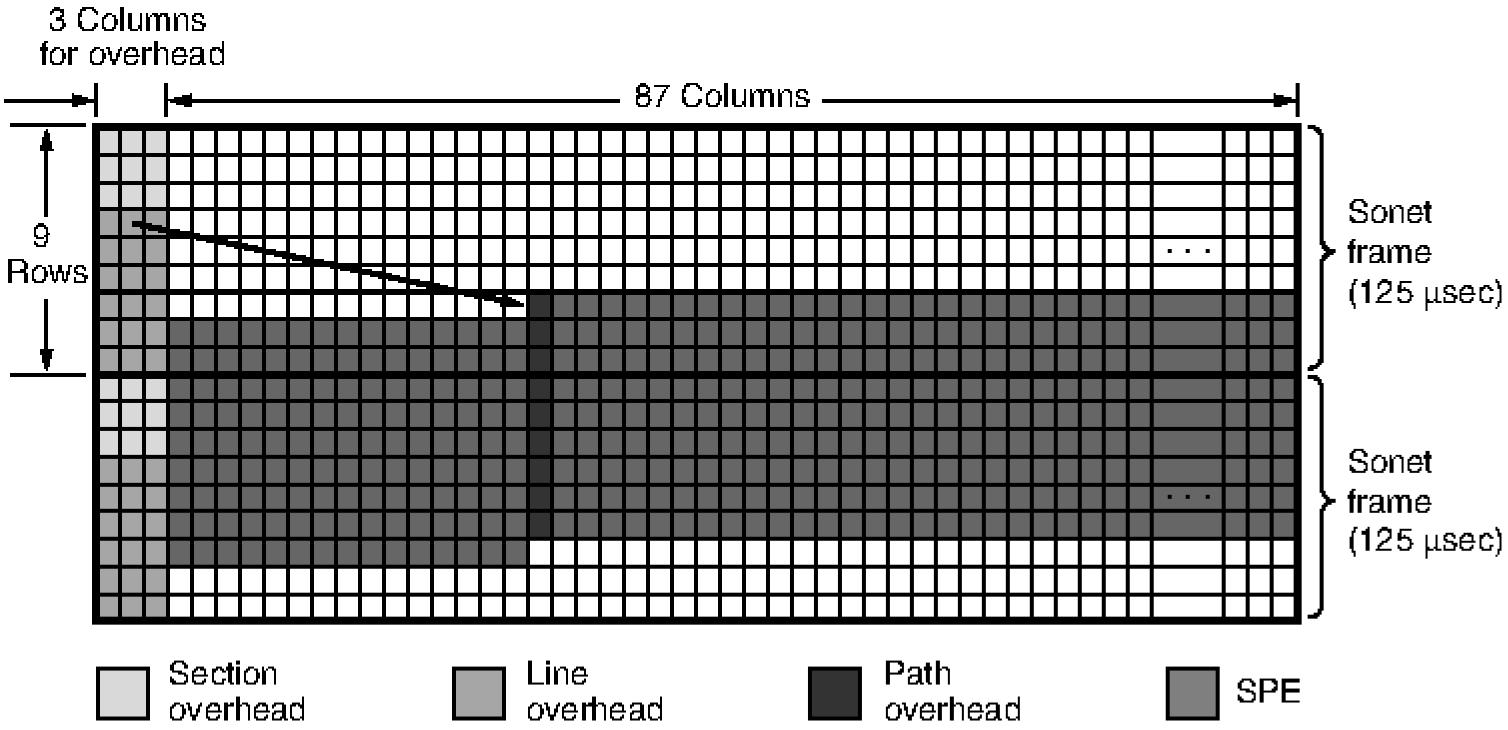


Figure 2-39. Two back-to-back SONET frames.

The ability to allow the SPE to begin anywhere within the SONET frame and even to span two frames, as shown in Fig. 2-39, gives added flexibility to the system. For example, if a payload arrives at the source while a dummy SONET frame is being constructed, it can be inserted into the current frame instead of being held until the start of the next one.

The SONET/SDH multiplexing hierarchy is shown in Fig. 2-40. Rates from STS-1 to STS-768 have been defined, ranging from roughly a T3 line to 40 Gbps. Even higher rates will surely be defined over time, with OC-3072 at 160 Gbps being the next in line if and when it becomes technologically feasible. The optical carrier corresponding to STS-*n* is called OC-*n* but is bit for bit the same except for a certain bit reordering needed for synchronization. The SDH names are different, and they start at OC-3 because ITU-based systems do not have a rate near 51.84 Mbps. We have shown the common rates, which proceed from OC-3 in multiples of four. The gross data rate includes all the overhead. The SPE data rate excludes the line and section overhead. The user data rate excludes all overhead and counts only the 87 payload columns.

As an aside, when a carrier, such as OC-3, is not multiplexed, but carries the data from only a single source, the letter *c* (for concatenated) is appended to the designation, so OC-3 indicates a 155.52-Mbps carrier consisting of three separate OC-1 carriers, but OC-3c indicates a data stream from a single source at 155.52 Mbps. The three OC-1 streams within an OC-3c stream are interleaved by column—first column 1 from stream 1, then column 1 from stream 2, then column 1 from stream 3, followed by column 2 from stream 1, and so on—leading to a frame 270 columns wide and 9 rows deep.

SONET		SDH	Data rate (Mbps)		
Electrical	Optical	Optical	Gross	SPE	User
STS-1	OC-1		51.84	50.112	49.536
STS-3	OC-3	STM-1	155.52	150.336	148.608
STS-12	OC-12	STM-4	622.08	601.344	594.432
STS-48	OC-48	STM-16	2488.32	2405.376	2377.728
STS-192	OC-192	STM-64	9953.28	9621.504	9510.912
STS-768	OC-768	STM-256	39813.12	38486.016	38043.648

Figure 2-40. SONET and SDH multiplex rates.

Wavelength Division Multiplexing

A form of frequency division multiplexing is used as well as TDM to harness the tremendous bandwidth of fiber optic channels. It is called **WDM** (**Wavelength Division Multiplexing**). The basic principle of WDM on fibers is depicted in Fig. 2-41. Here four fibers come together at an optical combiner, each with its energy present at a different wavelength. The four beams are combined onto a single shared fiber for transmission to a distant destination. At the far end, the beam is split up over as many fibers as there were on the input side. Each output fiber contains a short, specially constructed core that filters out all but one wavelength. The resulting signals can be routed to their destination or recombined in different ways for additional multiplexed transport.

There is really nothing new here. This way of operating is just frequency division multiplexing at very high frequencies, with the term WDM owing to the description of fiber optic channels by their wavelength or "color" rather than frequency. As long as each channel has its own frequency (i.e., wavelength) range and all the ranges are disjoint, they can be multiplexed together on the long-haul fiber. The only difference with electrical FDM is that an optical system using a diffraction grating is completely passive and thus highly reliable.

The reason WDM is popular is that the energy on a single channel is typically only a few gigahertz wide because that is the current limit of how fast we can convert between electrical and optical signals. By running many channels in parallel on different wavelengths, the aggregate bandwidth is increased linearly with the number of channels. Since the bandwidth of a single fiber band is about 25,000 GHz (see Fig. 2-7), there is theoretically room for 2500 10-Gbps channels even at 1 bit/Hz (and higher rates are also possible).

WDM technology has been progressing at a rate that puts computer technology to shame. WDM was invented around 1990. The first commercial systems had eight channels of 2.5 Gbps per channel. By 1998, systems with 40 channels

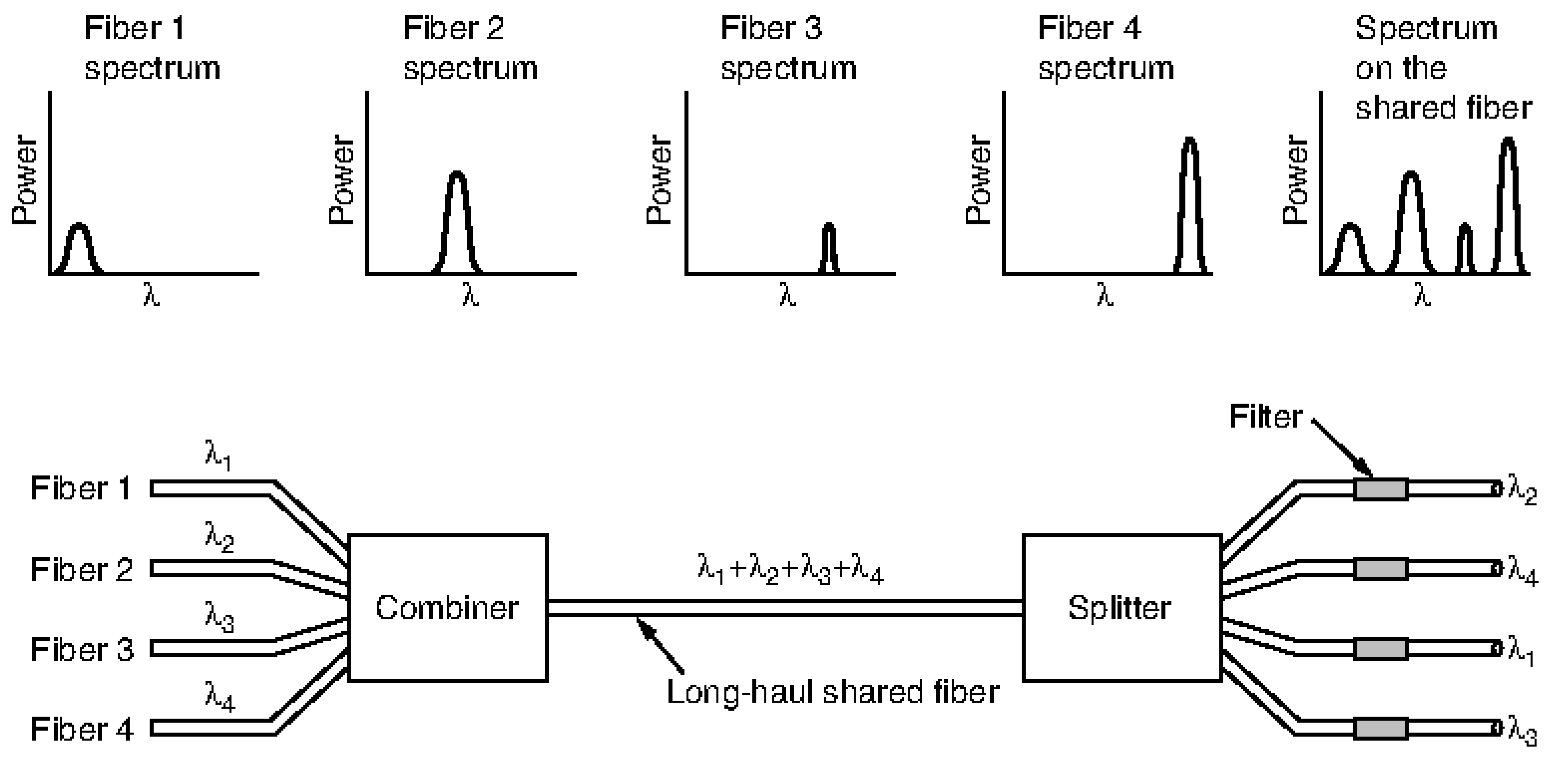


Figure 2-41. Wavelength division multiplexing.

of 2.5 Gbps were on the market. By 2006, there were products with 192 channels of 10 Gbps and 64 channels of 40 Gbps, capable of moving up to 2.56 Tbps. This bandwidth is enough to transmit 80 full-length DVD movies per second. The channels are also packed tightly on the fiber, with 200, 100, or as little as 50 GHz of separation. Technology demonstrations by companies after bragging rights have shown 10 times this capacity in the lab, but going from the lab to the field usually takes at least a few years. When the number of channels is very large and the wavelengths are spaced close together, the system is referred to as **DWDM** (**Dense WDM**).

One of the drivers of WDM technology is the development of all-optical components. Previously, every 100 km it was necessary to split up all the channels and convert each one to an electrical signal for amplification separately before reconverting them to optical signals and combining them. Nowadays, all-optical amplifiers can regenerate the entire signal once every 1000 km without the need for multiple opto-electrical conversions.

In the example of Fig. 2-41, we have a fixed-wavelength system. Bits from input fiber 1 go to output fiber 3, bits from input fiber 2 go to output fiber 1, etc. However, it is also possible to build WDM systems that are switched in the optical domain. In such a device, the output filters are tunable using Fabry-Perot or Mach-Zehnder interferometers. These devices allow the selected frequencies to be changed dynamically by a control computer. This ability provides a large amount of flexibility to provision many different wavelength paths through the telephone network from a fixed set of fibers. For more information about optical networks and WDM, see Ramaswami et al. (2009).

2.6.5 Switching

From the point of view of the average telephone engineer, the phone system is divided into two principal parts: outside plant (the local loops and trunks, since they are physically outside the switching offices) and inside plant (the switches, which are inside the switching offices). We have just looked at the outside plant. Now it is time to examine the inside plant.

Two different switching techniques are used by the network nowadays: circuit switching and packet switching. The traditional telephone system is based on circuit switching, but packet switching is beginning to make inroads with the rise of voice over IP technology. We will go into circuit switching in some detail and contrast it with packet switching. Both kinds of switching are important enough that we will come back to them when we get to the network layer.

Circuit Switching

Conceptually, when you or your computer places a telephone call, the switching equipment within the telephone system seeks out a physical path all the way from your telephone to the receiver's telephone. This technique is called **circuit switching**. It is shown schematically in Fig. 2-42(a). Each of the six rectangles represents a carrier switching office (end office, toll office, etc.). In this example, each office has three incoming lines and three outgoing lines. When a call passes through a switching office, a physical connection is (conceptually) established between the line on which the call came in and one of the output lines, as shown by the dotted lines.

In the early days of the telephone, the connection was made by the operator plugging a jumper cable into the input and output sockets. In fact, a surprising little story is associated with the invention of automatic circuit switching equipment. It was invented by a 19th-century Missouri undertaker named Almon B. Strowger. Shortly after the telephone was invented, when someone died, one of the survivors would call the town operator and say "Please connect me to an undertaker." Unfortunately for Mr. Strowger, there were two undertakers in his town, and the other one's wife was the town telephone operator. He quickly saw that either he was going to have to invent automatic telephone switching equipment or he was going to go out of business. He chose the first option. For nearly 100 years, the circuit-switching equipment used worldwide was known as **Strowger gear**. (History does not record whether the now-unemployed switchboard operator got a job as an information operator, answering questions such as "What is the phone number of an undertaker?")

The model shown in Fig. 2-42(a) is highly simplified, of course, because parts of the physical path between the two telephones may, in fact, be microwave or fiber links onto which thousands of calls are multiplexed. Nevertheless, the basic idea is valid: once a call has been set up, a dedicated path between both ends exists and will continue to exist until the call is finished.

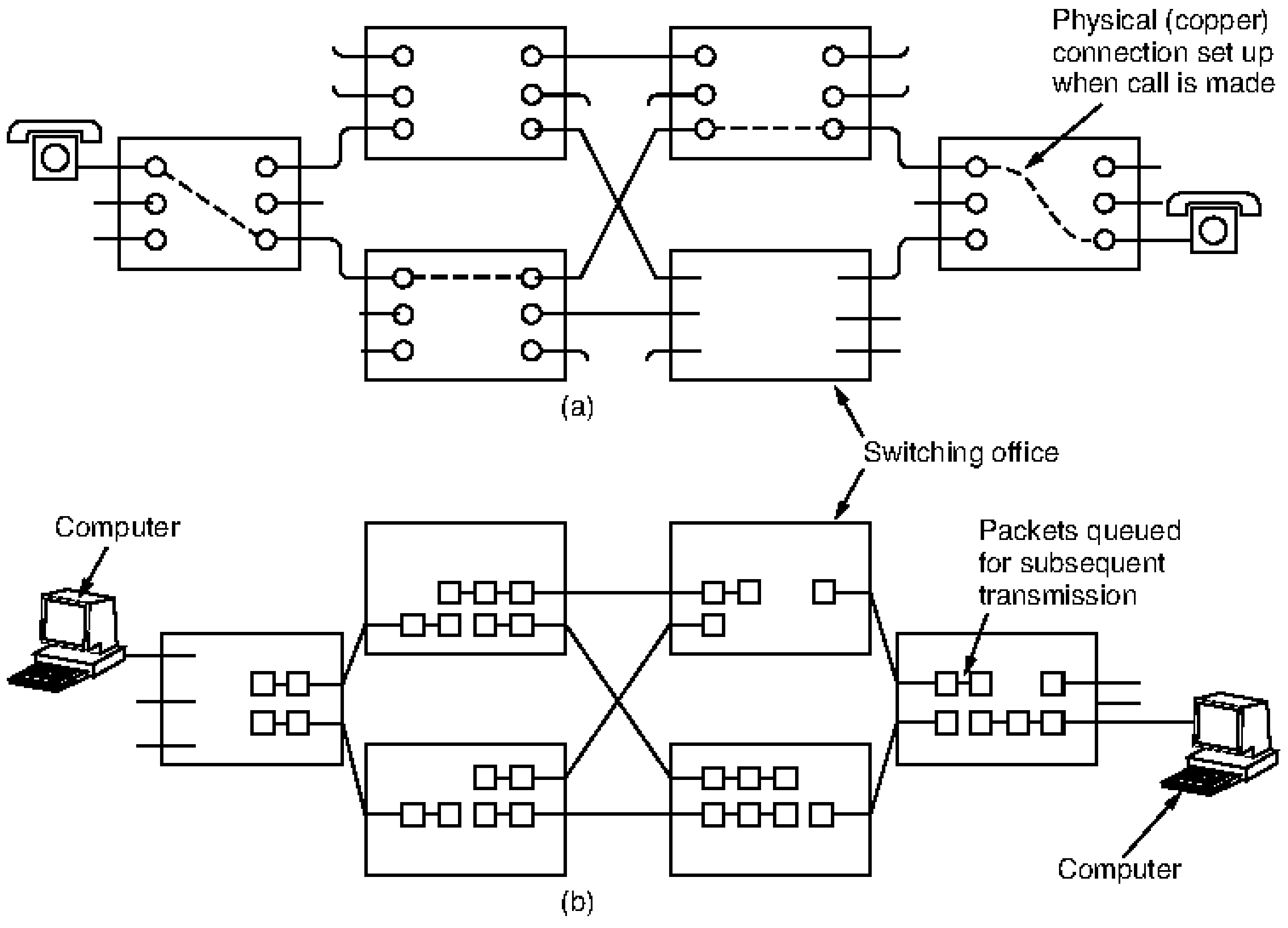


Figure 2-42. (a) Circuit switching. (b) Packet switching.

An important property of circuit switching is the need to set up an end-to-end path *before* any data can be sent. The elapsed time between the end of dialing and the start of ringing can easily be 10 sec, more on long-distance or international calls. During this time interval, the telephone system is hunting for a path, as shown in Fig. 2-43(a). Note that before data transmission can even begin, the call request signal must propagate all the way to the destination and be acknowledged. For many computer applications (e.g., point-of-sale credit verification), long setup times are undesirable.

As a consequence of the reserved path between the calling parties, once the setup has been completed, the only delay for data is the propagation time for the electromagnetic signal, about 5 msec per 1000 km. Also as a consequence of the established path, there is no danger of congestion—that is, once the call has been put through, you never get busy signals. Of course, you might get one before the connection has been established due to lack of switching or trunk capacity.

Packet Switching

The alternative to circuit switching is **packet switching**, shown in Fig. 2-42(b) and described in Chap. 1. With this technology, packets are sent as soon as they are available. There is no need to set up a dedicated path in advance, unlike

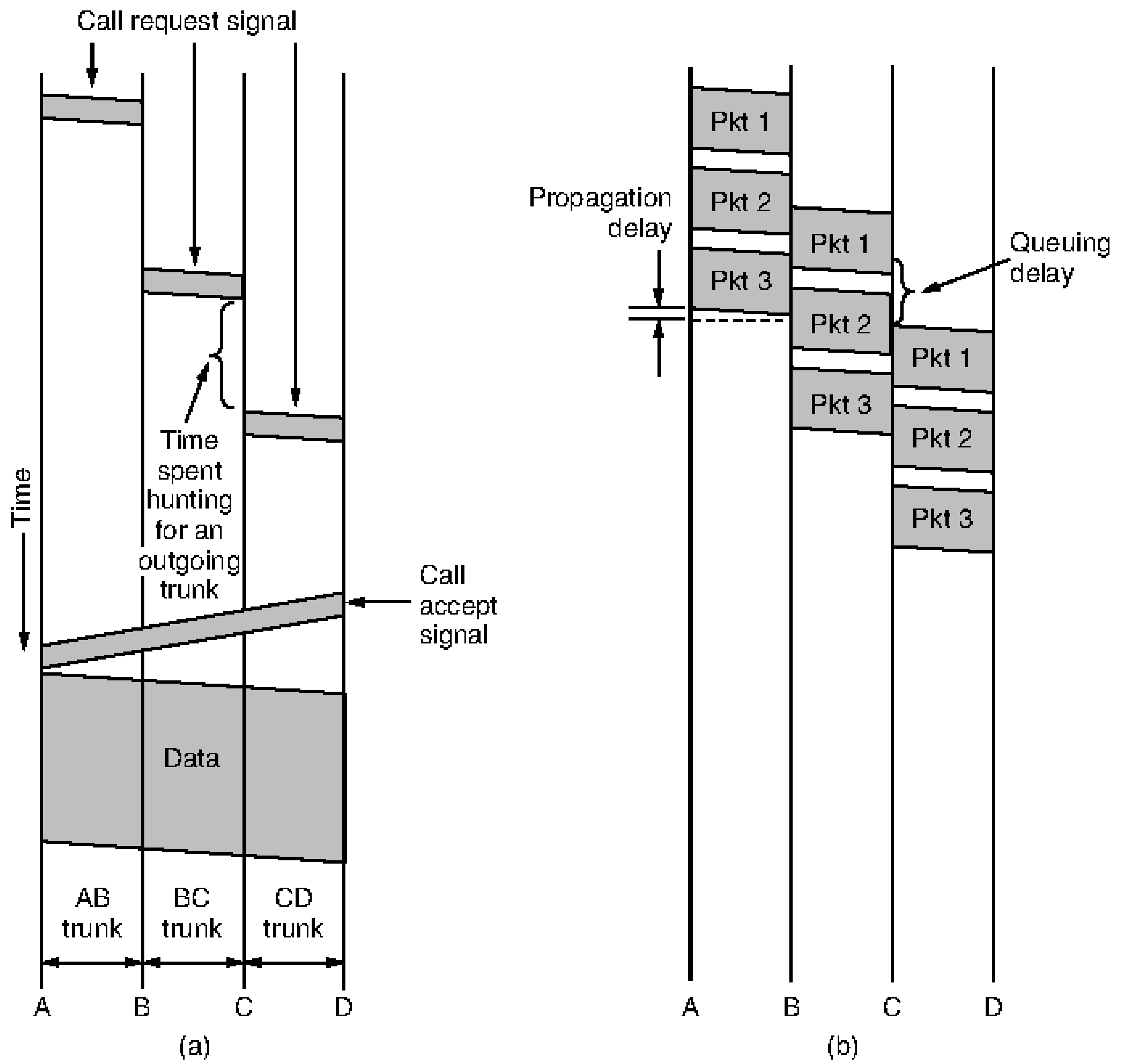


Figure 2-43. Timing of events in (a) circuit switching, (b) packet switching.

with circuit switching. It is up to routers to use store-and-forward transmission to send each packet on its way to the destination on its own. This procedure is unlike circuit switching, in which the result of the connection setup is the reservation of bandwidth all the way from the sender to the receiver. All data on the circuit follows this path. Among other properties, having all the data follow the same path means that it cannot arrive out of order. With packet switching there is no fixed path, so different packets can follow different paths, depending on network conditions at the time they are sent, and they may arrive out of order.

Packet-switching networks place a tight upper limit on the size of packets. This ensures that no user can monopolize any transmission line for very long (e.g., many milliseconds), so that packet-switched networks can handle interactive traffic. It also reduces delay since the first packet of a long message can be forwarded before the second one has fully arrived. However, the store-and-forward delay of accumulating a packet in the router's memory before it is sent on to the next router exceeds that of circuit switching. With circuit switching, the bits just flow through the wire continuously.

Packet and circuit switching also differ in other ways. Because no bandwidth is reserved with packet switching, packets may have to wait to be forwarded. This introduces **queuing delay** and congestion if many packets are sent at the same time. On the other hand, there is no danger of getting a busy signal and being unable to use the network. Thus, congestion occurs at different times with circuit switching (at setup time) and packet switching (when packets are sent).

If a circuit has been reserved for a particular user and there is no traffic, its bandwidth is wasted. It cannot be used for other traffic. Packet switching does not waste bandwidth and thus is more efficient from a system perspective. Understanding this trade-off is crucial for comprehending the difference between circuit switching and packet switching. The trade-off is between guaranteed service and wasting resources versus not guaranteeing service and not wasting resources.

Packet switching is more fault tolerant than circuit switching. In fact, that is why it was invented. If a switch goes down, all of the circuits using it are terminated and no more traffic can be sent on any of them. With packet switching, packets can be routed around dead switches.

A final difference between circuit and packet switching is the charging algorithm. With circuit switching, charging has historically been based on distance and time. For mobile phones, distance usually does not play a role, except for international calls, and time plays only a coarse role (e.g., a calling plan with 2000 free minutes costs more than one with 1000 free minutes and sometimes nights or weekends are cheap). With packet switching, connect time is not an issue, but the volume of traffic is. For home users, ISPs usually charge a flat monthly rate because it is less work for them and their customers can understand this model, but backbone carriers charge regional networks based on the volume of their traffic.

The differences are summarized in Fig. 2-44. Traditionally, telephone networks have used circuit switching to provide high-quality telephone calls, and computer networks have used packet switching for simplicity and efficiency. However, there are notable exceptions. Some older computer networks have been circuit switched under the covers (e.g., X.25) and some newer telephone networks use packet switching with voice over IP technology. This looks just like a standard telephone call on the outside to users, but inside the network packets of voice data are switched. This approach has let upstarts market cheap international calls via calling cards, though perhaps with lower call quality than the incumbents.

2.7 THE MOBILE TELEPHONE SYSTEM

The traditional telephone system, even if it someday gets multigigabit end-to-end fiber, will still not be able to satisfy a growing group of users: people on the go. People now expect to make phone calls and to use their phones to check

Item	Circuit switched	Packet switched
Call setup	Required	Not needed
Dedicated physical path	Yes	No
Each packet follows the same route	Yes	No
Packets arrive in order	Yes	No
Is a switch crash fatal	Yes	No
Bandwidth available	Fixed	Dynamic
Time of possible congestion	At setup time	On every packet
Potentially wasted bandwidth	Yes	No
Store-and-forward transmission	No	Yes
Charging	Per minute	Per packet

Figure 2-44. A comparison of circuit-switched and packet-switched networks.

email and surf the Web from airplanes, cars, swimming pools, and while jogging in the park. Consequently, there is a tremendous amount of interest in wireless telephony. In the following sections we will study this topic in some detail.

The mobile phone system is used for wide area voice and data communication. **Mobile phones** (sometimes called **cell phones**) have gone through three distinct generations, widely called **1G**, **2G**, and **3G**. The generations are:

1. Analog voice.
2. Digital voice.
3. Digital voice and data (Internet, email, etc.).

(Mobile phones should not be confused with **cordless phones** that consist of a base station and a handset sold as a set for use within the home. These are never used for networking, so we will not examine them further.)

Although most of our discussion will be about the technology of these systems, it is interesting to note how political and tiny marketing decisions can have a huge impact. The first mobile system was devised in the U.S. by AT&T and mandated for the whole country by the FCC. As a result, the entire U.S. had a single (analog) system and a mobile phone purchased in California also worked in New York. In contrast, when mobile phones came to Europe, every country devised its own system, which resulted in a fiasco.

Europe learned from its mistake and when digital came around, the government-run PTTs got together and standardized on a single system (GSM), so any European mobile phone will work anywhere in Europe. By then, the U.S. had decided that government should not be in the standardization business, so it left digital to the marketplace. This decision resulted in different equipment manufacturers producing different kinds of mobile phones. As a consequence, in the U.S.

two major—and completely incompatible—digital mobile phone systems were deployed, as well as other minor systems.

Despite an initial lead by the U.S., mobile phone ownership and usage in Europe is now far greater than in the U.S. Having a single system that works anywhere in Europe and with any provider is part of the reason, but there is more. A second area where the U.S. and Europe differed is in the humble matter of phone numbers. In the U.S., mobile phones are mixed in with regular (fixed) telephones. Thus, there is no way for a caller to see if, say, (212) 234-5678 is a fixed telephone (cheap or free call) or a mobile phone (expensive call). To keep people from getting nervous about placing calls, the telephone companies decided to make the mobile phone owner pay for incoming calls. As a consequence, many people hesitated buying a mobile phone for fear of running up a big bill by just receiving calls. In Europe, mobile phone numbers have a special area code (analogous to 800 and 900 numbers) so they are instantly recognizable. Consequently, the usual rule of "caller pays" also applies to mobile phones in Europe (except for international calls, where costs are split).

A third issue that has had a large impact on adoption is the widespread use of prepaid mobile phones in Europe (up to 75% in some areas). These can be purchased in many stores with no more formality than buying a digital camera. You pay and you go. They are preloaded with a balance of, for example, 20 or 50 euros and can be recharged (using a secret PIN code) when the balance drops to zero. As a consequence, practically every teenager and many small children in Europe have (usually prepaid) mobile phones so their parents can locate them, without the danger of the child running up a huge bill. If the mobile phone is used only occasionally, its use is essentially free since there is no monthly charge or charge for incoming calls.

2.7.1 First-Generation (1G) Mobile Phones: Analog Voice

Enough about the politics and marketing aspects of mobile phones. Now let us look at the technology, starting with the earliest system. Mobile radiotelephones were used sporadically for maritime and military communication during the early decades of the 20th century. In 1946, the first system for car-based telephones was set up in St. Louis. This system used a single large transmitter on top of a tall building and had a single channel, used for both sending and receiving. To talk, the user had to push a button that enabled the transmitter and disabled the receiver. Such systems, known as **push-to-talk systems**, were installed in several cities beginning in the late 1950s. CB radio, taxis, and police cars often use this technology.

In the 1960s, **IMTS** (**Improved Mobile Telephone System**) was installed. It, too, used a high-powered (200-watt) transmitter on top of a hill but it had two frequencies, one for sending and one for receiving, so the push-to-talk button was no longer needed. Since all communication from the mobile telephones went inbound on a different channel than the outbound signals, the mobile users could not hear each other (unlike the push-to-talk system used in taxis).

IMTS supported 23 channels spread out from 150 MHz to 450 MHz. Due to the small number of channels, users often had to wait a long time before getting a dial tone. Also, due to the large power of the hilltop transmitters, adjacent systems had to be several hundred kilometers apart to avoid interference. All in all, the limited capacity made the system impractical.

Advanced Mobile Phone System

All that changed with **AMPS (Advanced Mobile Phone System)**, invented by Bell Labs and first installed in the United States in 1982. It was also used in England, where it was called TACS, and in Japan, where it was called MCS-L1. AMPS was formally retired in 2008, but we will look at it to understand the context for the 2G and 3G systems that improved on it.

In all mobile phone systems, a geographic region is divided up into **cells**, which is why the devices are sometimes called cell phones. In AMPS, the cells are typically 10 to 20 km across; in digital systems, the cells are smaller. Each cell uses some set of frequencies not used by any of its neighbors. The key idea that gives cellular systems far more capacity than previous systems is the use of relatively small cells and the reuse of transmission frequencies in nearby (but not adjacent) cells. Whereas an IMTS system 100 km across can have only one call on each frequency, an AMPS system might have 100 10-km cells in the same area and be able to have 10 to 15 calls on each frequency, in widely separated cells. Thus, the cellular design increases the system capacity by at least an order of magnitude, more as the cells get smaller. Furthermore, smaller cells mean that less power is needed, which leads to smaller and cheaper transmitters and handsets.

The idea of frequency reuse is illustrated in Fig. 2-45(a). The cells are normally roughly circular, but they are easier to model as hexagons. In Fig. 2-45(a), the cells are all the same size. They are grouped in units of seven cells. Each letter indicates a group of frequencies. Notice that for each frequency set, there is a buffer about two cells wide where that frequency is not reused, providing for good separation and low interference.

Finding locations high in the air to place base station antennas is a major issue. This problem has led some telecommunication carriers to forge alliances with the Roman Catholic Church, since the latter owns a substantial number of exalted potential antenna sites worldwide, all conveniently under a single management.

In an area where the number of users has grown to the point that the system is overloaded, the power can be reduced and the overloaded cells split into smaller

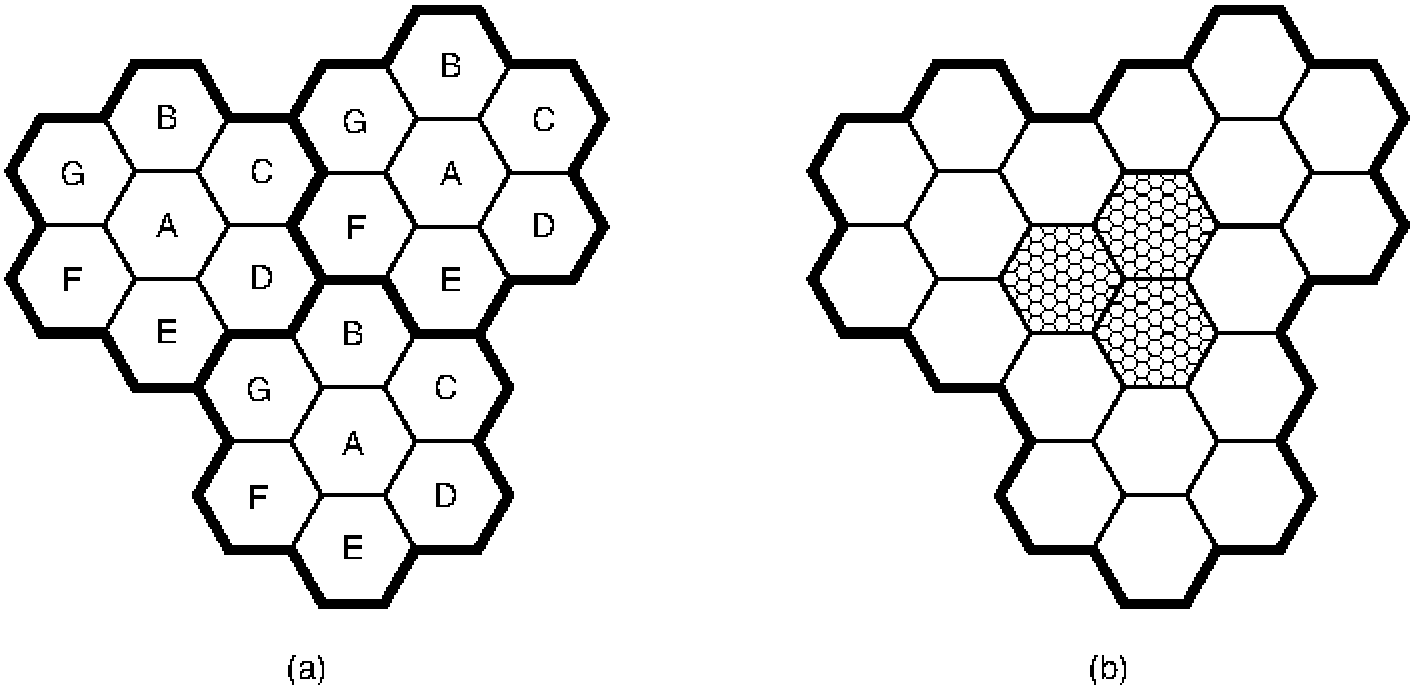


Figure 2-45. (a) Frequencies are not reused in adjacent cells. (b) To add more users, smaller cells can be used.

microcells to permit more frequency reuse, as shown in Fig. 2-45(b). Telephone companies sometimes create temporary microcells, using portable towers with satellite links at sporting events, rock concerts, and other places where large numbers of mobile users congregate for a few hours.

At the center of each cell is a base station to which all the telephones in the cell transmit. The base station consists of a computer and transmitter/receiver connected to an antenna. In a small system, all the base stations are connected to a single device called an **MSC** (**Mobile Switching Center**) or **MTSO** (**Mobile Telephone Switching Office**). In a larger one, several MSCs may be needed, all of which are connected to a second-level MSC, and so on. The MSCs are essentially end offices as in the telephone system, and are in fact connected to at least one telephone system end office. The MSCs communicate with the base stations, each other, and the PSTN using a packet-switching network.

At any instant, each mobile telephone is logically in one specific cell and under the control of that cell's base station. When a mobile telephone physically leaves a cell, its base station notices the telephone's signal fading away and asks all the surrounding base stations how much power they are getting from it. When the answers come back, the base station then transfers ownership to the cell getting the strongest signal; under most conditions that is the cell where the telephone is now located. The telephone is then informed of its new boss, and if a call is in progress, it is asked to switch to a new channel (because the old one is not reused in any of the adjacent cells). This process, called **handoff**, takes about 300 msec. Channel assignment is done by the MSC, the nerve center of the system. The base stations are really just dumb radio relays.

Channels

AMPS uses FDM to separate the channels. The system uses 832 full-duplex channels, each consisting of a pair of simplex channels. This arrangement is known as **FDD** (**Frequency Division Duplex**). The 832 simplex channels from 824 to 849 MHz are used for mobile to base station transmission, and 832 simplex channels from 869 to 894 MHz are used for base station to mobile transmission. Each of these simplex channels is 30 kHz wide.

The 832 channels are divided into four categories. Control channels (base to mobile) are used to manage the system. Paging channels (base to mobile) alert mobile users to calls for them. Access channels (bidirectional) are used for call setup and channel assignment. Finally, data channels (bidirectional) carry voice, fax, or data. Since the same frequencies cannot be reused in nearby cells and 21 channels are reserved in each cell for control, the actual number of voice channels available per cell is much smaller than 832, typically about 45.

Call Management

Each mobile telephone in AMPS has a 32-bit serial number and a 10-digit telephone number in its programmable read-only memory. The telephone number is represented as a 3-digit area code in 10 bits and a 7-digit subscriber number in 24 bits. When a phone is switched on, it scans a preprogrammed list of 21 control channels to find the most powerful signal. The phone then broadcasts its 32-bit serial number and 34-bit telephone number. Like all the control information in AMPS, this packet is sent in digital form, multiple times, and with an error-correcting code, even though the voice channels themselves are analog.

When the base station hears the announcement, it tells the MSC, which records the existence of its new customer and also informs the customer's home MSC of his current location. During normal operation, the mobile telephone reregisters about once every 15 minutes.

To make a call, a mobile user switches on the phone, enters the number to be called on the keypad, and hits the SEND button. The phone then transmits the number to be called and its own identity on the access channel. If a collision occurs there, it tries again later. When the base station gets the request, it informs the MSC. If the caller is a customer of the MSC's company (or one of its partners), the MSC looks for an idle channel for the call. If one is found, the channel number is sent back on the control channel. The mobile phone then automatically switches to the selected voice channel and waits until the called party picks up the phone.

Incoming calls work differently. To start with, all idle phones continuously listen to the paging channel to detect messages directed at them. When a call is placed to a mobile phone (either from a fixed phone or another mobile phone), a packet is sent to the callee's home MSC to find out where it is. A packet is then sent to the base station in its current cell, which sends a broadcast on the paging channel of the form "Unit 14, are you there?" The called phone responds with a "Yes" on the access channel. The base then says something like: "Unit 14, call for you on channel 3." At this point, the called phone switches to channel 3 and starts making ringing sounds (or playing some melody the owner was given as a birthday present).

2.7.2 Second-Generation (2G) Mobile Phones: Digital Voice

The first generation of mobile phones was analog; the second generation is digital. Switching to digital has several advantages. It provides capacity gains by allowing voice signals to be digitized and compressed. It improves security by allowing voice and control signals to be encrypted. This in turn deters fraud and eavesdropping, whether from intentional scanning or echoes of other calls due to RF propagation. Finally, it enables new services such as text messaging.

Just as there was no worldwide standardization during the first generation, there was also no worldwide standardization during the second, either. Several different systems were developed, and three have been widely deployed. **D-AMPS** (**Digital Advanced Mobile Phone System**) is a digital version of AMPS that coexists with AMPS and uses TDM to place multiple calls on the same frequency channel. It is described in International Standard IS-54 and its successor IS-136. **GSM** (**Global System for Mobile communications**) has emerged as the dominant system, and while it was slow to catch on in the U.S. it is now used virtually everywhere in the world. Like D-AMPS, GSM is based on a mix of FDM and TDM. **CDMA** (**Code Division Multiple Access**), described in **International Standard IS-95**, is a completely different kind of system and is based on neither FDM mor TDM. While CDMA has not become the dominant 2G system, its technology has become the basis for 3G systems.

Also, the name **PCS** (**Personal Communications Services**) is sometimes used in the marketing literature to indicate a second-generation (i.e., digital) system. Originally it meant a mobile phone using the 1900 MHz band, but that distinction is rarely made now.

We will now describe GSM, since it is the dominant 2G system. In the next section we will have more to say about CDMA when we describe 3G systems.

GSM—The Global System for Mobile Communications

GSM started life in the 1980s as an effort to produce a single European 2G standard. The task was assigned to a telecommunications group called (in French) Groupe Specialé Mobile. The first GSM systems were deployed starting in 1991 and were a quick success. It soon became clear that GSM was going to be more than a European success, with uptake stretching to countries as far away as Australia, so GSM was renamed to have a more worldwide appeal.

GSM and the other mobile phone systems we will study retain from 1G systems a design based on cells, frequency reuse across cells, and mobility with handoffs as subscribers move. It is the details that differ. Here, we will briefly discuss some of the main properties of GSM. However, the printed GSM standard is over 5000 [sic] pages long. A large fraction of this material relates to engineering aspects of the system, especially the design of receivers to handle multipath signal propagation, and synchronizing transmitters and receivers. None of this will be even mentioned here.

Fig. 2-46 shows that the GSM architecture is similar to the AMPS architecture, though the components have different names. The mobile itself is now divided into the handset and a removable chip with subscriber and account information called a **SIM card**, short for **Subscriber Identity Module**. It is the SIM card that activates the handset and contains secrets that let the mobile and the network identify each other and encrypt conversations. A SIM card can be removed and plugged into a different handset to turn that handset into your mobile as far as the network is concerned.

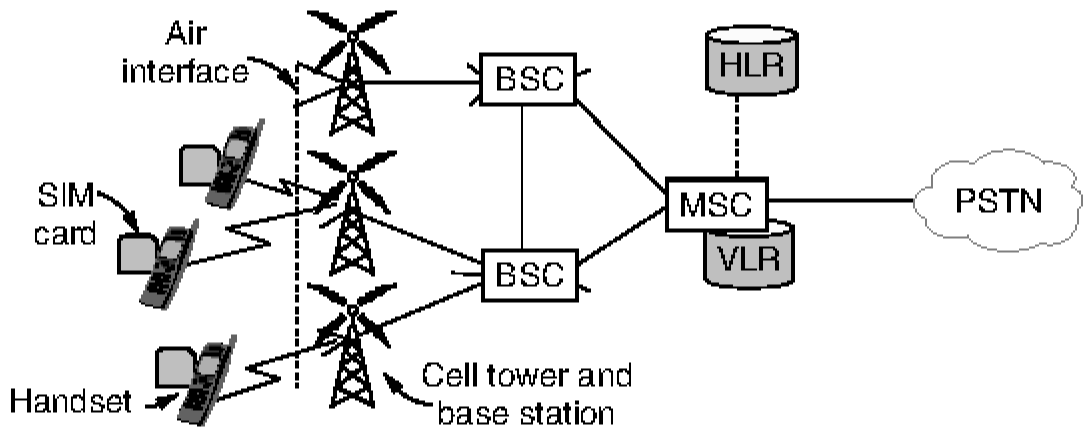


Figure 2-46. GSM mobile network architecture.

The mobile talks to cell base stations over an **air interface** that we will describe in a moment. The cell base stations are each connected to a **BSC** (**Base Station Controller**) that controls the radio resources of cells and handles handoff. The BSC in turn is connected to an MSC (as in AMPS) that routes calls and connects to the PSTN (Public Switched Telephone Network).

To be able to route calls, the MSC needs to know where mobiles can currently be found. It maintains a database of nearby mobiles that are associated with the cells it manages. This database is called the **VLR** (**Visitor Location Register**). There is also a database in the mobile network that gives the last known location of each mobile. It is called the **HLR** (**Home Location Register**). This database is used to route incoming calls to the right locations. Both databases must be kept up to date as mobiles move from cell to cell.

We will now describe the air interface in some detail. GSM runs on a range of frequencies worldwide, including 900, 1800, and 1900 MHz. More spectrum is allocated than for AMPS in order to support a much larger number of users. GSM is a frequency division duplex cellular system, like AMPS. That is, each mobile transmits on one frequency and receives on another, higher frequency (55 MHz higher for GSM versus 80 MHz higher for AMPS). However, unlike with AMPS, with GSM a single frequency pair is split by time-division multiplexing into time slots. In this way it is shared by multiple mobiles.

To handle multiple mobiles, GSM channels are much wider than the AMPS channels (200-kHz versus 30 kHz). One 200-kHz channel is shown in Fig. 2-47. A GSM system operating in the 900-MHz region has 124 pairs of simplex channels. Each simplex channel is 200 kHz wide and supports eight separate connections on it, using time division multiplexing. Each currently active station is assigned one time slot on one channel pair. Theoretically, 992 channels can be supported in each cell, but many of them are not available, to avoid frequency conflicts with neighboring cells. In Fig. 2-47, the eight shaded time slots all belong to the same connection, four of them in each direction. Transmitting and receiving does not happen in the same time slot because the GSM radios cannot transmit and receive at the same time and it takes time to switch from one to the other. If the mobile device assigned to 890.4/935.4 MHz and time slot 2 wanted to transmit to the base station, it would use the lower four shaded slots (and the ones following them in time), putting some data in each slot until all the data had been sent.

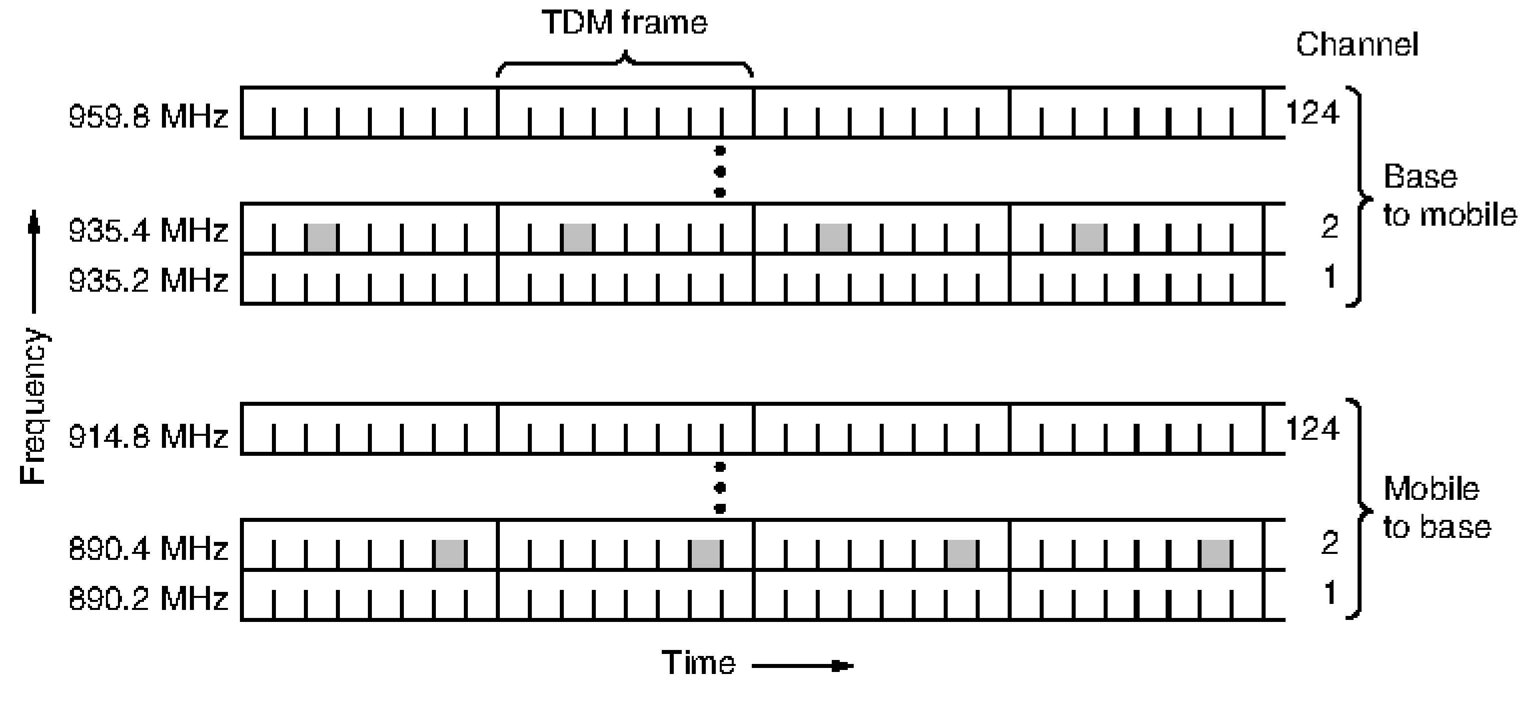


Figure 2-47. GSM uses 124 frequency channels, each of which uses an eight-slot TDM system.

The TDM slots shown in Fig. 2-47 are part of a complex framing hierarchy. Each TDM slot has a specific structure, and groups of TDM slots form multiframes, also with a specific structure. A simplified version of this hierarchy is shown in Fig. 2-48. Here we can see that each TDM slot consists of a 148-bit data frame that occupies the channel for 577 μsec (including a 30-μsec guard time after each slot). Each data frame starts and ends with three 0 bits, for frame delineation purposes. It also contains two 57-bit *Information* fields, each one having a control bit that indicates whether the following *Information* field is for voice or data. Between the *Information* fields is a 26-bit *Sync* (training) field that is used by the receiver to synchronize to the sender's frame boundaries.

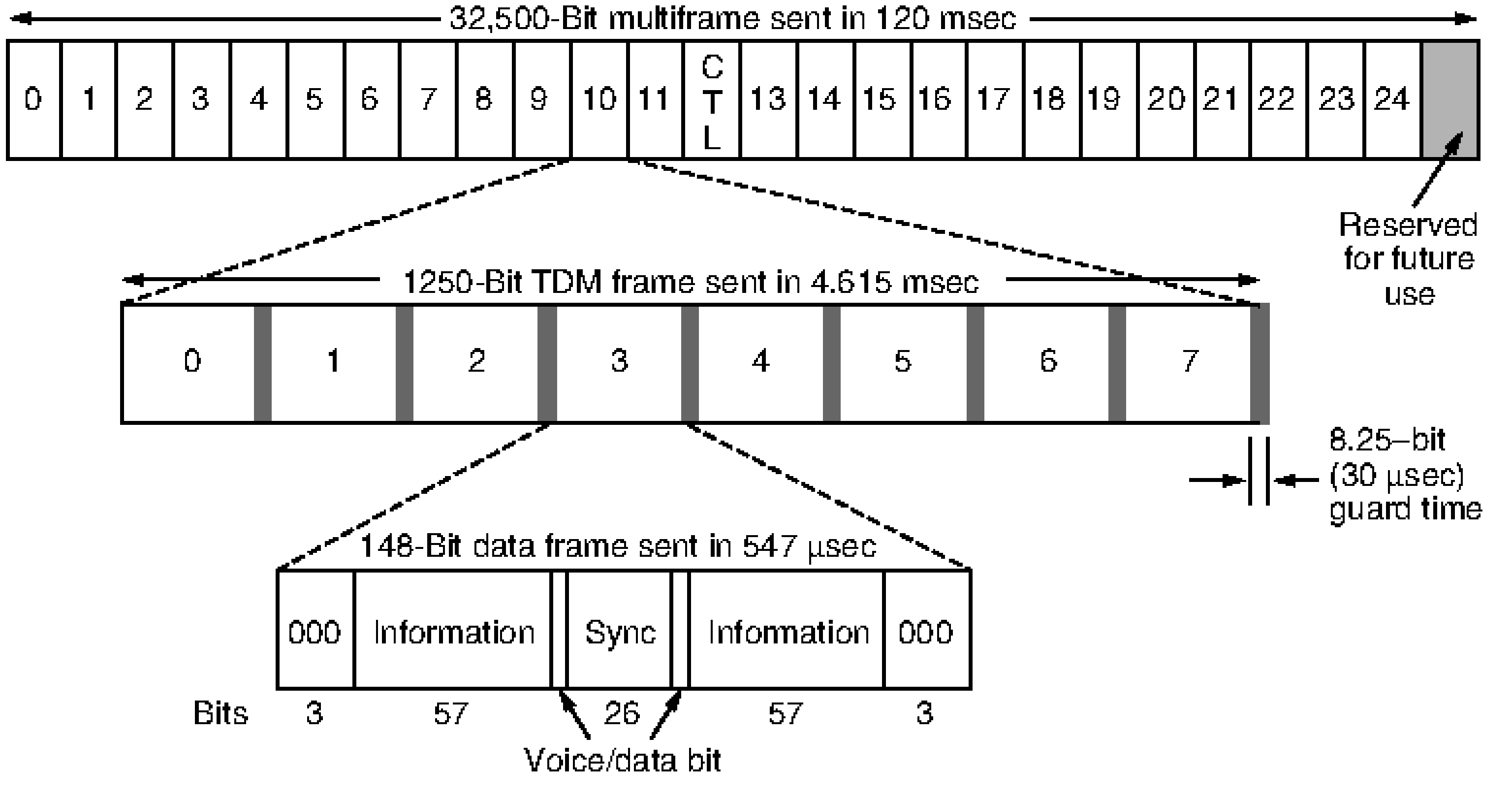


Figure 2-48. A portion of the GSM framing structure.

A data frame is transmitted in 547 μsec, but a transmitter is only allowed to send one data frame every 4.615 msec, since it is sharing the channel with seven other stations. The gross rate of each channel is 270,833 bps, divided among eight users. However, as with AMPS, the overhead eats up a large fraction of the bandwidth, ultimately leaving 24.7 kbps worth of payload per user before error correction. After error correction, 13 kbps is left for speech. While this is substantially less than 64 kbps PCM for uncompressed voice signals in the fixed telephone network, compression on the mobile device can reach these levels with little loss of quality.

As can be seen from Fig. 2-48, eight data frames make up a TDM frame and 26 TDM frames make up a 120-msec multiframe. Of the 26 TDM frames in a multiframe, slot 12 is used for control and slot 25 is reserved for future use, so only 24 are available for user traffic.

However, in addition to the 26-slot multiframe shown in Fig. 2-48, a 51-slot multiframe (not shown) is also used. Some of these slots are used to hold several control channels used to manage the system. The **broadcast control channel** is a continuous stream of output from the base station containing the base station's identity and the channel status. All mobile stations monitor their signal strength to see when they have moved into a new cell.

The **dedicated control channel** is used for location updating, registration, and call setup. In particular, each BSC maintains a database of mobile stations currently under its jurisdiction, the VLR. Information needed to maintain the VLR is sent on the dedicated control channel.

Finally, there is the **common control channel**, which is split up into three logical subchannels. The first of these subchannels is the **paging channel**, which the base station uses to announce incoming calls. Each mobile station monitors it continuously to watch for calls it should answer. The second is the **random access channel**, which allows users to request a slot on the dedicated control channel. If two requests collide, they are garbled and have to be retried later. Using the dedicated control channel slot, the station can set up a call. The assigned slot is announced on the third subchannel, the **access grant channel**.

Finally, GSM differs from AMPS in how handoff is handled. In AMPS, the MSC manages it completely without help from the mobile devices. With time slots in GSM, the mobile is neither sending nor receiving most of the time. The idle slots are an opportunity for the mobile to measure signal quality to other nearby base stations. It does so and sends this information to the BSC. The BSC can use it to determine when a mobile is leaving one cell and entering another so it can perform the handoff. This design is called **MAHO** (**Mobile Assisted HandOff**).

2.7.3 Third-Generation (3G) Mobile Phones: Digital Voice and Data

The first generation of mobile phones was analog voice, and the second generation was digital voice. The third generation of mobile phones, or **3G** as it is called, is all about digital voice *and* data.

A number of factors are driving the industry. First, data traffic already exceeds voice traffic on the fixed network and is growing exponentially, whereas voice traffic is essentially flat. Many industry experts expect data traffic to dominate voice on mobile devices as well soon. Second, the telephone, entertainment, and computer industries have all gone digital and are rapidly converging. Many people are drooling over lightweight, portable devices that act as a telephone, music and video player, email terminal, Web interface, gaming machine, and more, all with worldwide wireless connectivity to the Internet at high bandwidth.

Apple's iPhone is a good example of this kind of 3G device. With it, people get hooked on wireless data services, and AT&T wireless data volumes are rising steeply with the popularity of iPhones. The trouble is, the iPhone uses a 2.5G network (an enhanced 2G network, but not a true 3G network) and there is not enough data capacity to keep users happy. 3G mobile telephony is all about providing enough wireless bandwidth to keep these future users happy.

ITU tried to get a bit more specific about this vision starting back around 1992. It issued a blueprint for getting there called **IMT-2000**, where IMT stood for **International Mobile Telecommunications**. The basic services that the IMT-2000 network was supposed to provide to its users are:

1. High-quality voice transmission.
2. Messaging (replacing email, fax, SMS, chat, etc.).
3. Multimedia (playing music, viewing videos, films, television, etc.).
4. Internet access (Web surfing, including pages with audio and video).

Additional services might be video conferencing, telepresence, group game playing, and m-commerce (waving your telephone at the cashier to pay in a store). Furthermore, all these services are supposed to be available worldwide (with automatic connection via a satellite when no terrestrial network can be located), instantly (always on), and with quality of service guarantees.

ITU envisioned a single worldwide technology for IMT-2000, so manufacturers could build a single device that could be sold and used anywhere in the world (like CD players and computers and unlike mobile phones and televisions). Having a single technology would also make life much simpler for network operators and would encourage more people to use the services. Format wars, such as the Betamax versus VHS battle with videorecorders, are not good for business.

As it turned out, this was a bit optimistic. The number 2000 stood for three things: (1) the year it was supposed to go into service, (2) the frequency it was supposed to operate at (in MHz), and (3) the bandwidth the service should have (in kbps). It did not make it on any of the three counts. Nothing was implemented by 2000. ITU recommended that all governments reserve spectrum at 2 GHz so devices could roam seamlessly from country to country. China reserved the required bandwidth but nobody else did. Finally, it was recognized that 2 Mbps is not currently feasible for users who are *too* mobile (due to the difficulty of performing handoffs quickly enough). More realistic is 2 Mbps for stationary indoor users (which will compete head-on with ADSL), 384 kbps for people walking, and 144 kbps for connections in cars.

Despite these initial setbacks, much has been accomplished since then. Several IMT proposals were made and, after some winnowing, it came down to two main ones. The first one, **WCDMA** (**Wideband CDMA**), was proposed by Ericsson and was pushed by the European Union, which called it **UMTS** (**Universal Mobile Telecommunications System**). The other contender was **CDMA2000**, proposed by Qualcomm.

Both of these systems are more similar than different in that they are based on broadband CDMA; WCDMA uses 5-MHz channels and CDMA2000 uses 1.25-MHz channels. If the Ericsson and Qualcomm engineers were put in a room and told to come to a common design, they probably could find one fairly quickly. The trouble is that the real problem is not engineering, but politics (as usual). Europe wanted a system that interworked with GSM, whereas the U.S. wanted a system that was compatible with one already widely deployed in the U.S. (IS-95). Each side also supported its local company (Ericsson is based in Sweden; Qualcomm is in California). Finally, Ericsson and Qualcomm were involved in numerous lawsuits over their respective CDMA patents.

Worldwide, 10–15% of mobile subscribers already use 3G technologies. In North America and Europe, around a third of mobile subscribers are 3G. Japan was an early adopter and now nearly all mobile phones in Japan are 3G. These figures include the deployment of both UMTS and CDMA2000, and 3G continues to be one great cauldron of activity as the market shakes out. To add to the confusion, UMTS became a single 3G standard with multiple incompatible options, including CDMA2000. This change was an effort to unify the various camps, but it just papers over the technical differences and obscures the focus of ongoing efforts. We will use UMTS to mean WCDMA, as distinct from CDMA2000.

We will focus our discussion on the use of CDMA in cellular networks, as it is the distinguishing feature of both systems. CDMA is neither FDM nor TDM but a kind of mix in which each user sends on the same frequency band at the same time. When it was first proposed for cellular systems, the industry gave it approximately the same reaction that Columbus first got from Queen Isabella when he proposed reaching India by sailing in the wrong direction. However, through the persistence of a single company, Qualcomm, CDMA succeeded as a 2G system (IS-95) and matured to the point that it became the technical basis for 3G.

To make CDMA work in the mobile phone setting requires more than the basic CDMA technique that we described in the previous section. Specifically, we described synchronous CDMA, in which the chip sequences are exactly orthogonal. This design works when all users are synchronized on the start time of their chip sequences, as in the case of the base station transmitting to mobiles. The base station can transmit the chip sequences starting at the same time so that the signals will be orthogonal and able to be separated. However, it is difficult to synchronize the transmissions of independent mobile phones. Without care, their transmissions would arrive at the base station at different times, with no guarantee of orthogonality. To let mobiles send to the base station without synchronization, we want code sequences that are orthogonal to each other at all possible offsets, not simply when they are aligned at the start.

While it is not possible to find sequences that are exactly orthogonal for this general case, long pseudorandom sequences come close enough. They have the property that, with high probability, they have a low **cross-correlation** with each other at all offsets. This means that when one sequence is multiplied by another sequence and summed up to compute the inner product, the result will be small; it would be zero if they were orthogonal. (Intuitively, random sequences should always look different from each other. Multiplying them together should then produce a random signal, which will sum to a small result.) This lets a receiver filter unwanted transmissions out of the received signal. Also, the **auto-correlation** of