INFORMATION THEORY OF MULTIACCESS COMMUNICATIONS

The basic communications resources available to users are frequency and time. The efficient allocation of these communications resources lies in the domain of communications multiple access. The term "multiple access" means the remote sharing of a communications resource (e.g., satellite). The term *multiple access* is often confused with the term *multiplexing*. Multiplexing indicates the local sharing of a communications resource (e.g., a circuit board). Normally, for multiplexing, the resource allocation is normally assigned a priori. This article focuses on the theory of multiple access. High level description of various multiple access techniques and a comparison among them will be given.

For multiple access, there are three basic techniques for distributing the communications resources: frequency-division multiple access (FDMA), time-division multiple access (TDMA), and code-division multiple access (CDMA). For FDMA, one specifies the subbands of frequency to be allocated to users. For TDMA, periodically recurring time slots are identified and then allocated to users. This technique allows users to access the resource at fixed or random times, depending on the systems. For CDMA, full channel bandwidth is utilized simultaneously with the time resource. In addition, two other techniques for multiple access are also available, namely, space-division multiple access (SDMA) and polarization-division multiple access (PDMA). SDMA, also referred to as multibeam frequency reuse multiple access, uses spot beam antennas to separate radio signals by pointing them in different directions, which allows for reuse of the same frequency band. PDMA, or dual polarization frequency reuse, employs orthogonal polarization to separate signals, which also allows for reuse of the same frequency band.

The three basic multiple access schemes are implemented with various multiuser access algorithms to form fixed-assignment or random-access schemes. In a fixed-assignment access scheme, a fixed allocation of communication resources, frequency or time, or both, is made on a predetermined basis to a single user. The random-access scheme allows the users to access communications resources randomly. When the random-access algorithm exercises some control over the access method to improve the efficiency of the uncontrolled random access methods, the result is referred to as the controlled random access technique.

This article describes the underlying theory behind the multiple access techniques and their applications in satellite and cellular systems. Both fixed- and random-access techniques will be described with their associated applications. Since the article is intended for readers who are unfamiliar with this field, only high level descriptions with minimum technical details are presented.

FIXED-ASSIGNMENT MULTIPLE ACCESS

As mentioned earlier, multiple access techniques are required for multiple users to efficiently share remote communications resources. There are two major categories of multiple access methods: fixed assignment and random access. This section describes the three basic approaches for fixed-assignment multiple access: FDMA, TDMA, and CDMA. For completeness, brief descriptions of SDMA and PDMA are also presented.

Frequency-Division Multiple Access

The frequency-division multiple access (FDMA) technique is derived based on the frequency-division multiplexing (FDM) method. The FDM method involves mixing (or heterodyning) the signals at the same frequency band with fixed frequencies from local oscillators to different frequency bands and then combining the resulting multiple signals (at different frequency bands) for transmission as a single signal with a wider bandwidth (1). Figure 1 shows the FDM scheme (1, Fig. 9.3, page 480). Note that "guard bands" between the frequency assignments are provided as buffer zones to mitigate the adjacent channel interference. For a fixed-assignment FDMA system, a user is assigned to a fixed subchannel for transmission, and the subchannel is retained until released by the assigned user. The receiver terminal has precise knowledge of the transmission subchannel, and a filter is used to extract the designated signal out of the received composite signal.

The advantages of the fixed-assignment FDMA are (2):

- Channel capacity increases as information bit rate decreases. To reduce the information bit rate one can use an efficient modulation method such as M-ary phase-shift keying (PSK) (3) or the continuous phase modulation (CPM) technique (4).
- Implementation is simple due to technological advances.

The disadvantages associated with a fixed-assignment FDMA are:

- It needs to back-off the transmitter high power amplifier (HPA) from saturation point to avoid intermodulation caused by AM–AM and AM–PM distortions (1,5). This means it is power inefficient.
- FDMA is involved with narrowband technology, which also involves narrowband filters that may not be realizable in very large scale integrated (VLSI) digital circuits. This means higher cost for terminals even under volume production conditions.
- It is inflexible due to limited fixed bit rate per channel.

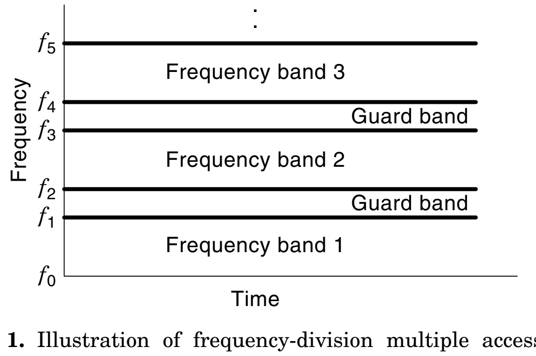

Figure 1. Illustration of frequency-division multiple access technique.

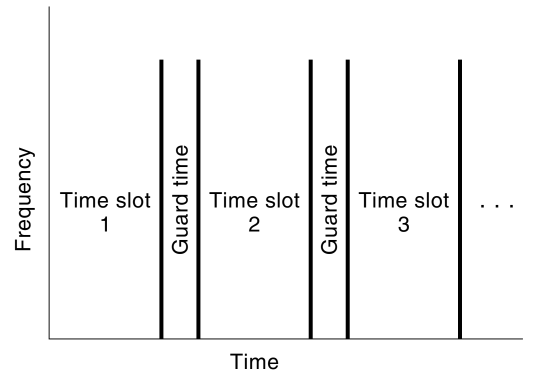

Figure 2. Illustration of time-division multiple access technique.

Time-Division Multiple Access

Time-division multiple access (TDMA) uses the full spectrum occupancy that is allocated to the system for a short duration of time called the time slot, as shown in Fig. 2 (1, Fig. 9.7, p. 484). Note that the guard band is provided here for crosstalk avoidance. TDMA employs the time-division multiplexing method in which the system time is divided into multiple time slots used by different users. Several time slots make up a frame. Each slot is made up of a preamble plus information bits addressed to various terminal users as shown in Fig. 3 (1, Fig. 9.9, p. 485). In a fixed-assignment TDMA system, a transmit controller assigns different users to different time slots, and the assigned time slot is retained by that user until the user releases it. At the receiving end, a user terminal synchronizes to the TDMA signal frame and extracts the time

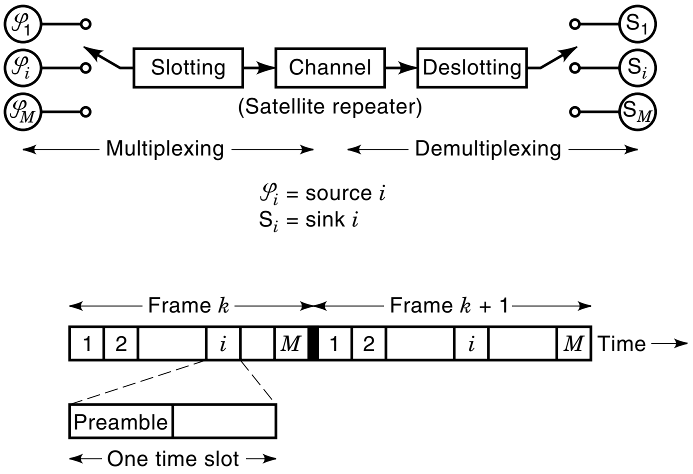

Figure 3. Illustration of fixed-assignment time-division multiple access technique.

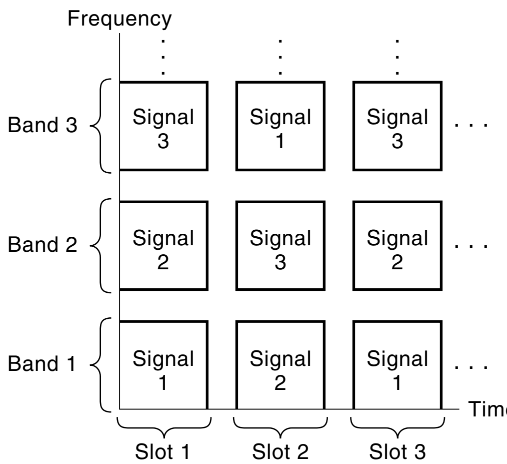

Figure 4. Illustration of code-division multiplexing.

slot assigned to that user. Figure 3 illustrates the demultiplexing procedure for a fixed-assignment TDMA system.

The advantages of a fixed-assignment TDMA include:

- When used with a constant modulation scheme, the transmitter HPA can operate at saturation. This means it is power efficient.
- It is flexible due to variable bit rates allowed for users.
- VLSI technology can be used for low cost in volume production.
- TDMA utilizes bandwidth more efficiently because no frequency guard band is required between the channels.

The disadvantages associated with fixed-assignment TDMA are (2):

- TDMA requires higher peak power than FDMA. This may cause significant drawback for mobile applications due to the shortening of battery life.
- Complicated signal processing is used in the detection and synchronization with a time slot.

Code-Division Multiple Access

Code-division multiple access (CDMA) is a hybrid combination of FDMA and TDMA (1,6). Figure 4 illustrates this concept (1, Fig. 9.14, p. 491). For CDMA, the system operates simultaneously over the entire system frequency bandwidth and system time. In CDMA systems the users are kept separate by assigning each of them a distinct user-signal code. The design of these codes is usually based on spread-spectrum (SS) signaling to provide sufficient degrees of freedom to separate different users in both time and frequency domains (although the use of SS does not imply CDMA). SS technique can be classified into two categories, namely, direct-sequence SS (DS-SS) and frequency-hopping SS (FH-SS) (7). Hence, CDMA can also categorize into DS-CDMA and FH-CDMA. In CDMA systems a hybrid combination of DS and FH for CDMA is also allowed. In the following, brief descriptions of the DS-CDMA and FH-CDMA are given.

Direct-Sequence CDMA. In DS-CDMA systems each of N users is preassigned its own code, $PN_i(t)$, where $i = 1, 2, 3, \ldots, N$. The user codes are selected such that they are approximately orthogonal to each other. This means that the cross-correlation of two different codes is approximately zero; that is

$$\int_0^{T_c} PN_i(t) PN_j(t)\, dt \approx \begin{cases} 1, & \text{for } i = j \\ 0, & \text{for } i \neq j \end{cases} \tag{1}$$

where T_C denotes the time duration of the code and usually is referred to as the chip duration. Since the assigned codes are orthogonal to each other, they can be spread over the entire spectrum of the communication resource simultaneously.

The modulated signal for user 1 is denoted as

$$S_1(t) = A_1(t) \cos[\omega_0 t + \phi_1(t)] \tag{2}$$

where $A_1(t)$, ω_o, and $\phi_1(t)$ are the amplitude, angular frequency, and phase, respectively, of the signal specified for user 1. Note that the modulated waveform presented in Eq. (2) is expressed in general form, without any restriction placed on modulation type. Then the spread signal is obtained by multiplying signal $S_1(t)$ with the code $PN_1(t)$, and the resultant signal, $S_1(t)PN_1(t)$, is then transmitted over the channel. Figure 5 shows a simplified block diagram for a typical CDMA system (1, Fig. 10.25, p. 572). Here the bandwidth of the code $PN_1(t)$ is much larger than the unspread signal $S_1(t)$. If one denotes the code rate for $PN_1(t)$ as R_c and the signal data rate

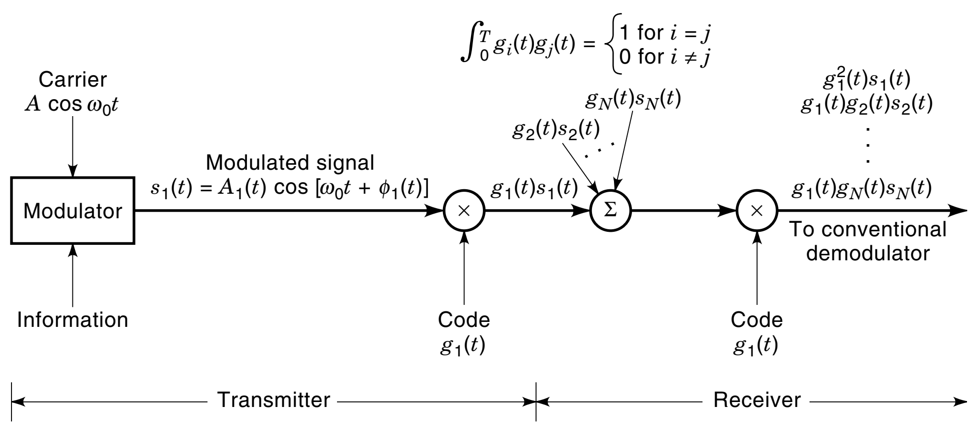

Figure 5. Illustration of code-division multiple access technique.

as R_s, then the processing gain G_p of the system is given by (1,7)

$$G_p\,(\mathrm{dB}) = 10\log\left(\frac{R_c}{R_s}\right) \tag{3}$$

The processing gain provides an indication of how well the signal $S_1(t)$ is being protected from interfering signals (intentional or unintentional). The larger the value of G_p, the better the protection the code can provide.

The spread signal $S_1(t)PN_1(t)$ is received in the presence of other spread signals, $S_2(t)PN_2(t)$, $S_3(t)PN_3(t)$, . . ., $S_N(t)PN_N(t)$. Assuming that the noise at the receiver is zero and the signal delays are negligible, we can write the received signal $R(t)$ as

$$R(t) = S_1(t)PN_1(t) + \sum_{i=2}^{N} S_i(t)PN_i(t) \tag{4}$$

Here we will also assume that the receiver is configured to receive messages from user 1 so that the second term shown in Eq. (4) is an interference signal. To recover the signal $S_1(t)$, the received signal $R(t)$ is despread by multiplying $R(t)$ with the code $PN_1(t)$ stored at the receiver,

$$R(t)PN_1(t) = S_1(t) + \sum_{i=2}^{N} S_i(t)PN_i(t)PN_1(t) \tag{5}$$

Here we have used the property $PN_1^2(t) = 1$. If we chose the code to have the orthogonal property, that is, the codes are chosen to satisfy the condition expressed in Eq. (1), then it can be shown that the undesired signal expressed in the second term of Eq. (5) is negligible (7,8). Since the codes are not perfectly orthogonal, the second term of Eq. (5) is negligible for a limited number of users. The performance degradation caused by the crosscorrelation in the second term sets the maximum number of simultaneous users. A rule of thumb for determining the maximum number of users N appears to be that (7)

$$N \approx \frac{10^{G_p(\mathrm{dB})/10}}{10} \tag{6}$$

While the code design is of paramount importance, of potential greater importance in DS-CDMA is the so-called near–far problem (7,9,10). Since the N users are usually geographically separated, a receiver is trying to detect the ith user, which is much farther than the jth user. Therefore, if each user transmits with equal power, the power received by the jth user would be much stronger than that received by the ith user. This particular problem is often so severe that DS-CDMA systems will not work without appropriate power control algorithms.

Advantages associated with DS-CDMA include:

- Multiple users can share the communication resources, both frequency and time, simultaneously.
- Communication privacy is possible due to assigned codes being known only to the users.
- There is an inherent robustness against mobile channel degradations such as fading and multipath (7–10).
- There is greater resistance to interference effects in a frequency reuse situation.
- More flexibility is possible because there is no requirement on time and frequency coordination among the various transmitters.

The disadvantages of DS-CDMA are:

- It requires power control algorithms due to the near–far problem.
- Timing alignments must be within a fraction of a coded sequence chip.
- Performance degradation increases as the number of users increases.

Frequency-Hopping CDMA. An alternative to DS-CDMA is FH-CDMA (1,7). In FH-CDMA systems each user is assigned a specific hopping pattern, and if all hopping patterns assigned are orthogonal, the near–far problem will be solved (except for possible spectral spillover from a specified frequency slot into adjacent slots). In practice, the codes assigned for these hopping patterns are not truly orthogonal; thus, interference will result when more than one signal uses the same frequency at a given instant of time. A simplified block diagram for a typical FH-CDMA modulator is shown in Fig. 6 (1, Fig, 9.15, p. 492).

FH-CDMA can be classified as fast FH-CDMA or slow FH-CDMA. Fast FH systems use a single frequency hop for each transmitted symbol. This means that, for fast FH systems, the hopping rate equals or exceeds the information symbol rate. On the other hand, slow FH systems transmit two or more symbols in the time interval between frequency hops.

The advantages associated with FH-CDMA include:

- Multiple users can share the communication resources, both frequency and time, simultaneously.
- Communication privacy is possible due to assigned codes being known only to the users.
- There is an inherent robustness against mobile channel degradations such as fading and multipath (7–10).

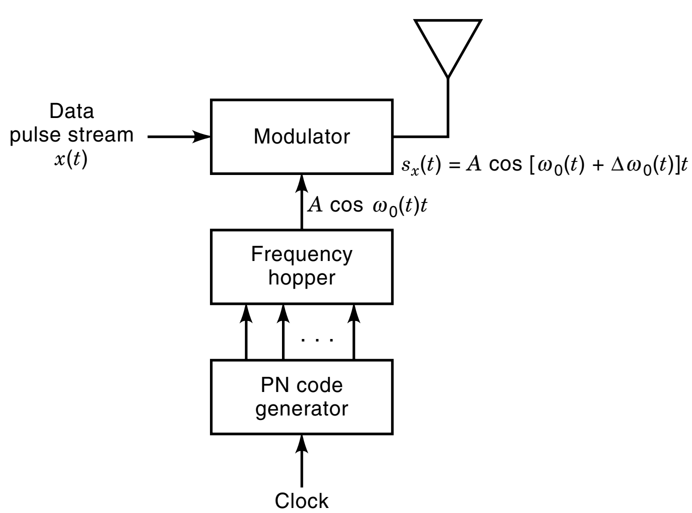

Figure 6. Illustration of code-division multiple access frequency hopping.

- There is an inherent robustness against interference.
- The near–far problem does not exist.
- Network implementation for FH-CDMA is simpler than DS-CDMA systems because the required timing alignments must be within a fraction of a hop duration as compared to a fraction of a chip duration.
- It performs best when a limited number of signals are sent in the presence of nonhopped signals.

The disadvantages are:

- Performance degradation is possible due to spectral spillover from a specified frequency slot into adjacent slots.
- Frequency synthesizer can be very costly.
- As the hopping rate increases the reliability decreases and synchronization becomes more difficult.

Space-Division Multiple Access

For wireless applications, space-division multiple access (SDMA) can be classified into cell-based and beamforming-based SDMA. The difference between the two approaches can best be illustrated in Fig. 7 (11, Fig. 1.1, p. 4) for cell-based SDMA and Fig. 8 (11, Fig. 1.2, p. 5) for beamforming-based SDMA.

A primitive form of SDMA exists when frequency carriers are reused in different cells separated by a special distance to reduce the level of co-channel interference. The larger the number of cells the higher the level of frequency reuse and thus the higher capacity that can be attained. This has resulted in cell-based SDMA, which has been predominant for quite a long time.

In a frequency reuse system, the term *radio capacity* is used to measure the traffic capacity, and is defined as

$$C_{\mathrm{r}} = \frac{M}{K \cdot S} \tag{7}$$

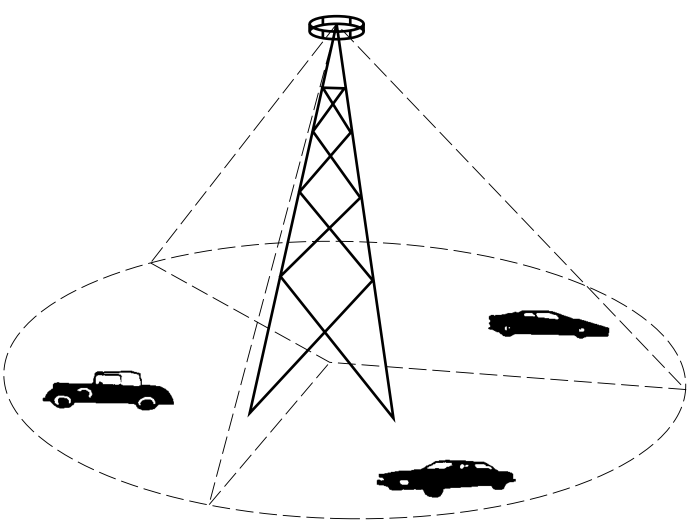

Figure 7. Illustration of the cell-based space-division multiple access. A different set of carrier frequencies is used in each of the sectors. These frequencies are used in other sectors of other cell sites. The frequency reuse pattern is selected to minimize the interference.

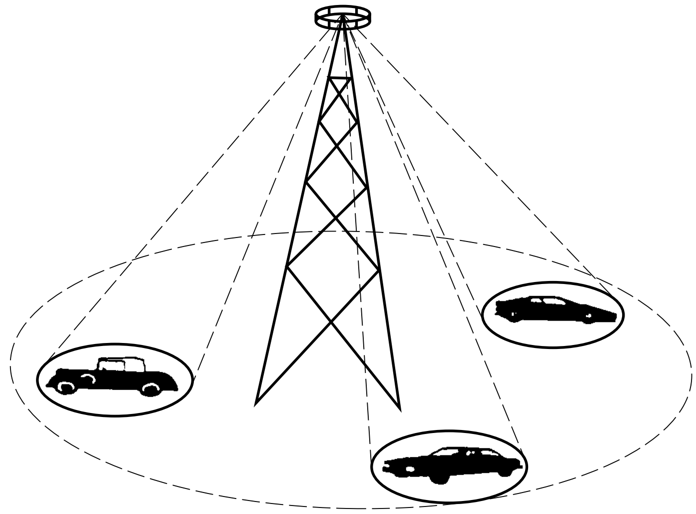

Figure 8. Illustration of beamforming-based space-division multiple access.

where M is the total number of frequency channels, K is the cell reuse factor and S is the number of sectors in a cell. K can be expressed as

$$K = \frac{1}{3}\left(\frac{D}{R}\right)^2 \tag{8}$$

where D is the distance between two co-channel cells and R is the cell radius. The corresponding average signal-to-interferer ratio (SIR) can be calculated for different types of sectoring systems, including adaptive beamforming with several beams in beamforming-based SDMA.

The system benefits of beamforming-based SDMA include:

- Improvement of multipath fading problems since narrower beams are used and the implicit optimal diversity combining performed by the beamformer
- More flexible coverage of each base station to match the local propagation conditions

Table 1 lists the capacity and SIR for several SDMA configurations (12).

Adaptive beamforming algorithms require a certain reference signal in the optimization process. If the reference signal is not explicit in the received data, blind adaptive beamforming (BAF) can be used instead. For digital communication signals, one can vary certain signal properties such as constant modulus applicable to FSK or PSK signals to result in the

Table 1. Radio Capacity and Signal-to-Noise Ratio for Different Cells

	K	S	Capacity (Channels/Cell)	SIR (dB)
Omnicells	7	1	$M/7$	18
120° sectorial cells	7	3	$M/21$	24.5
60° sectorial cells	4	6	$M/24$	26
60° sectorial beams	7	6	$3M/7$	20
N adaptive beams	7	1	$MN/7$	18

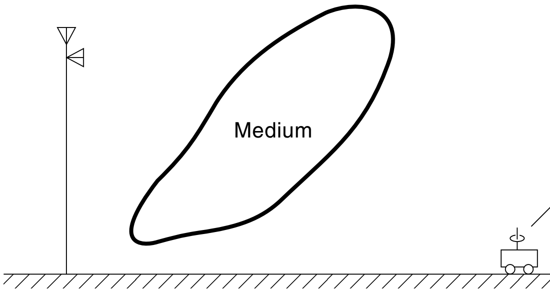

Figure 9. Illustration of horizontal and vertical polarization diversity signals.

constant modulus adaptive beamforming algorithm (13), or the cyclostationary properties of bauded signals to suggest the spectral self-coherence restoral (SCORE) algorithm (14).

Another method (15) that can be considered blind adaptive beamforming is based on decision-directed equalization to combat intersymbol interference (ISI) in digital communications. Using this concept, a BAF demodulates the beamformer output and uses it to make a decision in favor of a particular value in the known alphabet of the transmit sequence. A reference signal is then generated based on the modulated output of this decided demodulated beamformer output.

Polarization-Division Multiple Access

Signals transmitted in either horizontal or vertical electric field are uncorrelated at both the mobile and base station's receiver. Suppose that a received vertically polarized signal is

$$\Gamma_{11} = \sum_{i-1}^{N} a_i e^{j\psi_i} e^{-j\beta Vt\cos\phi_i} \tag{9}$$

and the received horizontally polarized signal is

$$\Gamma_{22} = \sum_{i=1}^{N} a_i' e^{j\psi_i'} e^{-j\beta Vt\cos\phi_i} \tag{10}$$

where a_i and ψ_i are the amplitude and phase, respectively, for each wave path and a_i' and ψ_i' are their counterparts in Eq. (9), V is the vehicle velocity, and ϕ_i is the angle of arrival of the ith wave. Although these two polarized waves arrived at the receiver from the same number of incoming waves, it is not difficult to see that Γ_{11} and Γ_{22} are uncorrelated because of their different amplitudes and phases. Thus, a PDMA system can be illustrated as in Fig. 9 (16, Fig. 9-6, p. 281). In this system, the base station can be two vertical and horizontal dipoles and the antenna at the mobile can be a pair of whip and loop antennas.

RANDOM-ASSIGNMENT MULTIPLE ACCESS

Fixed-assignment multiple access is most efficient when each user has a steady flow of information for transmission. However, this method becomes very inefficient when the information to be transmitted is intermittent or bursty in nature. As an example, for mobile cellular systems, where the subscribers pay for service as a function of channel connection time, fixed-assignment access can be very expensive for transmitting short messages. In this case, the random-access methods are more flexible and efficient than the fixed-access methods. This section discusses the three basic random-access schemes, namely, pure ALOHA, modified ALOHA (slotted and reservation), and carrier-sense multiple access with collision detection.

Pure ALOHA

Pure ALOHA (P-ALOHA), or basic ALOHA, was developed at the University of Hawaii in 1971 with the goal of connecting several university computers by the use of random-access protocol (17). The system concept is very simple and has been summarized by Sklar (1). The algorithm is listed below for future comparison with the enhanced version, the so-called slotted ALOHA.

- *Mode 1: Transmission Mode.* Users transmit at any time they desire, encoding their transmissions with an error detection code.
- *Mode 2: Listening Mode.* After a message transmission, a user listens for an acknowledgment (ACK) from the receiver. Transmissions from other users will sometimes overlap in time, causing reception errors in the data in each of the contending messages. We say the messages have collided. In such cases, the errors are detected, and the users receive a negative acknowledgment (NACK).
- *Mode 3: Retransmission Mode.* When a NACK is received, the messages are simply retransmitted. Of course, if the colliding users were retransmitted immediately, they would collide again. Therefore, the users retransmit after a random delay.
- *Mode 4: Timeout Mode.* If after a transmission, the user does not receive either an ACK or NACK within a specified time, the user retransmits the message.

Figure 10 shows the concept of the pure ALOHA algorithm (6, Fig. 11.15, p. 465).

Modified ALOHA

In order to improve the pure ALOHA algorithm, the slotted (18) and reservation ALOHA algorithms (19) have been proposed. Based on the summary described in Sklar (1), a brief description of these algorithms will be given here.

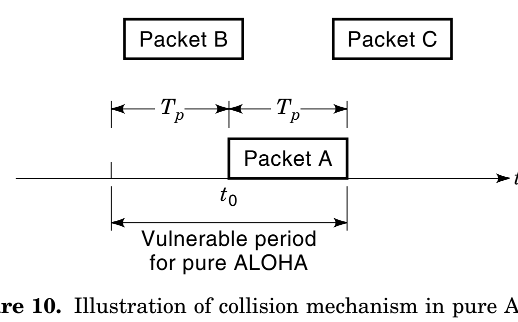

Figure 10. Illustration of collision mechanism in pure ALOHA.

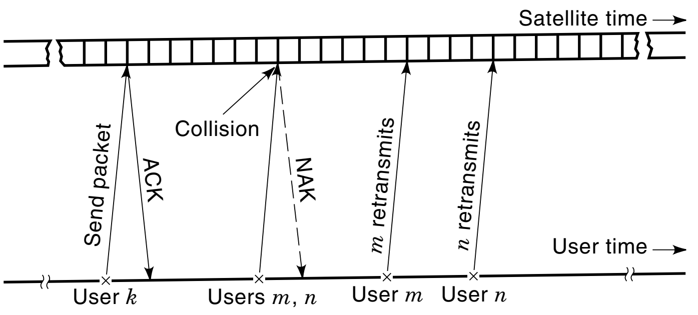

Figure 11. Illustration of slotted ALOHA.

Slotted ALOHA. The operation of the slotted ALOHA (S-ALOHA) is illustrated in Fig. 11 (1, Fig. 9.21, p. 501). A sequence of synchronization pulses is broadcast to all users for coordination among the users. Messages are sent through data packets with constant length between the synchronization pulses and can be started only at the beginning of a time slot. This modification reduces the rate of collisions by half, since only a packet transmitted in the same slot can interfere with one another (1). In S-ALOHA systems the users retransmit after a random delay of an integer number of slot times when a NACK occurs.

Reservation ALOHA. Significant improvement can be achieved with the reservation ALOHA (R-ALOHA) scheme. This scheme has two modes, namely, an unreserved mode and reserved mode.

The unreserved or quiescent mode, mode has three stages:

- A time frame is formed and divided into several reservation subslots.
- Users employ these small subslots to reserve message slots.
- After requesting a reservation, the users listen for an ACK and slot assignment.

The reserved mode has four stages:

- The time frame is divided into $M + 1$ slots whenever a reservation is made.
- The first M slots are used for message transmissions.
- The last slot is subdivided into subslots to be used for reservation/requests.
- Users send message packets only in their assigned portion of the M slots.

Figure 12 shows an example of the R-ALOHA system (1, Fig. 9.22, p. 503). In this example, the users seek to reserve 3 slots with $M = 5$ slots and $V = 6$ subslots. Compared with S-ALOHA, R-ALOHA is very efficient for high traffic intensity.

Carrier-Sense Multiple Access

To improve the previous algorithms and to make efficient use of the communications resources, the user terminal listens to the channel before attempting to transmit a packet. This protocol is called listen-before-talk and usually is referred to as carrier-sense multiple access (CSMA) protocol (12). This algorithm is widely used in both wired and wireless local area networks (LANs), where the transmission delays are low. There are several modified versions of CSMA, namely, CSMA with busy-tone signaling, CSMA with collision detection, and CSMA with collision avoidance. In addition, there is another modified version of CSMA, called data-sense multiple access (DSMA), which has been developed and adopted for use in wireless packet data networks such as cellular digital packet data (CDPD).

This section describes the three basic CSMA schemes, namely, 1-persistent CSMA, nonpersistent CSMA, and p-persistent CSMA. Modified versions of CSMA will also be described briefly.

1-Persistent Carrier-Sense Multiple Access. 1-Persistent carrier-sense multiple access (1-P CSMA) is the simplest form of CSMA. In the basic form, 1-P CSMA is unslotted. The "1-persistent" signifies the strategy in which the message is sent with probability 1 as soon as the channel is available. After sending the packet, the user station waits for an ACK, and if none is received in a specified amount of time, the user will wait for a random amount of time and then resume listening to the channel. When the channel is sensed idle, the packet is retransmitted immediately. In unslotted form, the system does not require synchronization between the user stations and all transmissions are synchronized to the time slots. In contrast with the unslotted form, the slotted 1-P CSMA requires synchronization among all user stations and all transmissions, whether initial transmissions or retransmissions, are synchronized to the time slots (1,6).

Nonpersistent Carrier-Sense Multiple Access. The main difference between the 1-P CSMA and nonpersistent carrier-

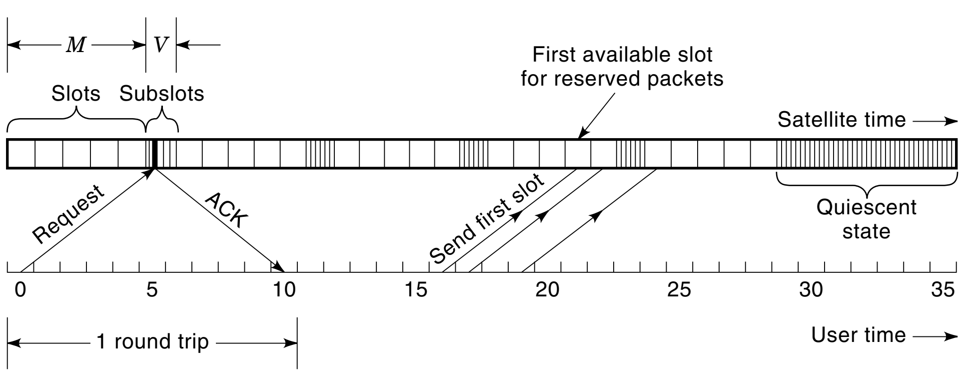

Figure 12. Illustration of reservation ALOHA. Station seeks to reserve 3 slots (M = 5 slots, V = 6 subslots).

sense multiple access (NP CSMA) is that a user station does not sense the channel continuously while it is busy. Instead, after sensing the busy condition, the NP CSMA system waits a randomly selected interval of time before sensing again. This random waiting time associated with NP CSMA could eliminate most of the collisions that would result from multiple users transmitting simultaneously upon sensing the transition from the busy to idle condition.

p-Persistent Carrier-Sense Multiple Access. The p-persistent carrier-sense multiple access (pP CSMA) is a generalization of the 1-P CSMA scheme, which is applicable to slotted channels. In this scheme, the slotted length is chosen to be the maximum propagation delay. In this system, a message is sent from a station with probability p when the channel is sensed to be idle. With probability $q = 1 - p$ the station defers action to the next slot, where the station senses the channel again. If the next slot is idle, the station transmits with probability p or defers with probability q. This procedure is repeated until either the whole frame has been transmitted or the channel is sensed to be busy. If the channel is busy, the station monitors the channel continuously until it becomes free; then it starts the above procedure again (6).

Carrier-Sense Multiple Access with Busy-Tone Signaling. In wireless networks, the user terminals are not always within the range and line-of-sight of each other, and when this situation occurs, it is referred to as "hidden terminal problem." This problem can be solved by using the carrier-sense multiple access with busy-tone signal (CSMA/BTS) technique (6). This technique divides the system bandwidth into two channels: a message channel and a busy-tone channel. The scheme works as follows. Whenever the central station senses signal energy on the message channel, it transmits a simple busy-tone signal on the busy-tone channel, and this tone is detectable by all the user stations. With this technique, a user station first senses the channel by detecting the busy-tone signal to determine if the network is busy. The procedure the user station then follows in transmission of the message depends on the particular version of CSMA being used in the network, and any of the CSMA techniques described earlier can be chosen.

Carrier-Sense Multiple Access with Collision Detection. The carrier-sense multiple access with collision detection (CSMA/CD) technique, also referred to as the "listen-while-talk" (LWT) technique, can be used with 1-P CSMA, NP CSMA, or pP CSMA, each with a slotted or unslotted version (6). In the operation of CSMA/CD, if the channel is detected to be idle or busy, a user station first sends the message (in the form of data packets) using the procedure dictated by the selected protocol in use. While sending the packets, the user station keeps monitoring the transmission; it stops transmission, aborting the collided packets and sending out a jamming signal, as soon as it detects a collision. The retransmission back-off procedure is initiated immediately after detecting a collision. The purpose of the jamming signal is to force consensus among users as to the state of the network, in that it ensures that all other stations know of the collision and go into back-off condition. Design of a proper back-off algorithm to ensure stable operation of the network is an important topic for communications design engineers.

Carrier-Sense Multiple Access with Collision Avoidance. The carrier-sense multiple access with collision avoidance (CSMA/CA) technique is widely used in many WLANs. The specific collision avoidance strategy for CSMA/CA is different from one manufacturer to another. In one system, CSMA/CA is referred to as CSMA with an exponential back-off strategy and an acknowledgment scheme. Note that the exponential back-off strategy is referred to as a collision avoidance mechanism. Other systems can employ R-ALOHA as the collision avoidance strategy.

Data-Sense Multiple Access. Digital or data sense multiplex access (DSMA) is commonly used in full-duplex wireless data communication networks such as CDPD and trans-European trunked radio (TETRA) (6). In these systems, communications from the mobile to base (also referred to as reverse channel or uplink) and from base to mobile (also referred to as forward channel or downlink) are performed on different frequency channels using different access techniques. The downlink uses TDMA, while the uplink uses DSMA. Interleaved among other signals broadcast on the downlink, the base station transmits a busy–idle bit in each time frame to report the status of the uplink channel. A mobile terminal will check this flag bit before transmission. If this bit indicates idle channel, the terminal proceeds to send its packet in the following time slot. As soon as the transmission starts, the base station switches the flag bit to busy state until the transmission from the mobile terminal is completed.

Polling Technique

The polling technique is a form of "control" random-assignment multiple access. In systems using this technique one station is used as a controller that periodically polls all the other stations to determine if they have data to transmit (17). Note that in R-ALOHA the control is distributed among all user terminals, while the polling technique utilizes centralized control.

Based on Refs. 6 and 20, a brief description of this technique is given here. Usually the controller station in the system is given a polling, instructing the order in which the terminals are poled. If the polled station has something to transmit, it starts transmission. If not, a "negative reply" or "no reply" is detected by the controller, which then polls the next terminal in the sequence. This technique is efficient only if (1) the round-trip propagation delay is small (due to constant exchange of control messages between the controller and terminals), (2) the overhead due to the polling message is low, and (3) the user population is not large and bursty.

Token Passing

This technique is another form of controlled random-assignment multiple access and it has been used widely in wired local area networks (LANs) for connecting computers. However, this scheme is not very popular in wireless networks. In this system a ring or loop topology is used. Figure 13 illustrates a typical token-ring network (1, Fig. 9.42, p. 529). As shown in Fig. 13, in the token-ring network, messages are passed from station to station along unidirectional links, until they return to the original station. This scheme passes the access privilege sequentially from station to station around the ring. Any station with data to send may, upon receiving

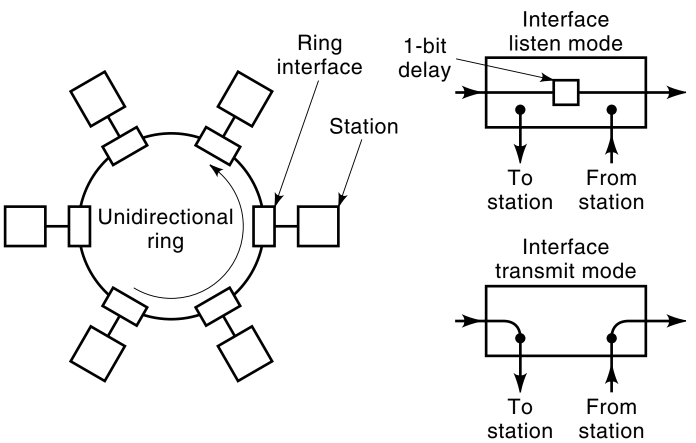

Figure 13. Illustration of the token-ring network.

the token, remove the token from the ring, send its message, and then pass on the control token.

PERFORMANCE COMPARISON OF MULTIPLE ACCESS TECHNIQUES

This section compares various standard mutiple access techniques such as FDMA, TDMA, pure ALOHA, slotted ALOHA, unslotted/slotted 1-P CSMA, and unslotted/slotted NP CSMA.

FDMA versus TDMA

Comparison between FDMA and TDMA schemes is not straightforward. It involves several issues such as bit error rate (BER) performance, throughput performance, system delay, and implementation. Some of these issues have greater or lesser performance depending on the type of system in which the access method is to be employed. In this section we briefly describe some of the major issues of comparison between FDMA and CDMA.

Bit Rate Capability. If one neglects all overhead elements such as guard bands in FDMA and guard time in TDMA, then the data rate capability for both systems is identical. The effective data rate is given by

$$R = \frac{Mb}{T} \tag{11}$$

where M is the number of disjoint channels and b is the number of data bits transmitted over T seconds.

Message Packet Delay. The message packet delay is defined as the packet waiting time before transmission plus packet transmission time. If we let M be the number of users generating data at a constant uniform rate of R/M bits/s, and use FDMA and TDMA systems that each transmit a packet of N bits every T seconds, then one can show that the average packet delays for FDMA and TDMA, respectively, are given by (6)

$$\text{Delay}_{\text{FDMA}} = T \tag{12}$$

$$\text{Delay}_{\text{TDMA}} = \text{Delay}_{\text{FDMA}} - \frac{T}{2}\left(1 - \frac{1}{M}\right) \tag{13}$$

Therefore, based on Eq. (13), TDMA is superior to FDMA with respect to the average delay packet when there are two or more users. It is interesting to note that for larger numbers of users the average delay packet of TDMA is half that of FDMA.

Spurious Narrowband Interference. An FDMA system outperforms a TDMA system in the presence of spurious narrowband interference. In an FDMA system, where the format is a single user per channel per carrier, the narrowband interference can impair the performance of only one user channel. On the other hand, in a TDMA system, a narrowband interference signal can cause performance degradation to all user channels in the TDMA data stream.

Pure ALOHA versus Slotted ALOHA

If one defines the throughput S as the number of successfully delivered packets per packet transmission time T_p, and G is the offered traffic load in packets per packet time, then the throughputs for P-ALOHA and S-ALOHA, respectively, are given by (1,6)

$$S_{\text{P-ALOHA}} = Ge^{-2G} \tag{14}$$

$$S_{\text{S-ALOHA}} = Ge^{-G} \tag{15}$$

The maximum throughput S occurs at

$$S_{\text{P-ALOHA}}(\max) = \frac{1}{2e} = 0.18 \tag{16}$$

$$S_{\text{S-ALOHA}}(\max) = \frac{1}{e} = 0.37 \tag{17}$$

at the values of G = 0.5 and 1, for P-ALOHA and S-ALOHA, respectively. This means that for a P-ALOHA channel, only 18% of the communication resource can be utilized. Comparing Eqs. (16) and (17), we see that, for S-ALOHA, there is an improvement of two times the P-ALOHA. A plot of P-ALOHA (or ALOHA) and S-ALOHA is shown in Fig. 14 (6, Fig. 11.19, p. 473).

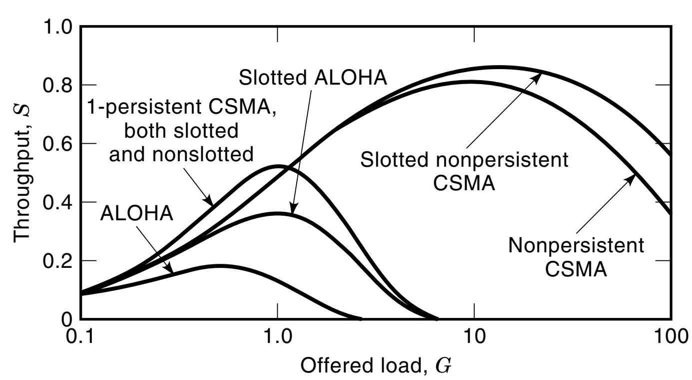

Figure 14. Throughput performance comparison of multiple access techniques.

1-P CSMA versus NP CSMA

Again, using the same definition for the throughput, and letting a be the normalized propagation delay, we have

$$a = \frac{\tau}{T_\mathrm{p}} \tag{18}$$

The parameter a described here corresponds to the time interval, which is normalized to the packet duration, during which a transmitted packet can suffer a collision in the CSMA schemes. Note that practical values of a on the order of 0.01 are usually of interest. The throughput for unslotted 1-P CSMA is found to be (6)

$$S_{\mathrm{Unslot-1P}} = \frac{G\left[1 + G + aG\left(1 + G + \dfrac{aG}{2}\right)\right]e^{-G(1+2a)}}{G(1+2a) - (1 - e^{-aG}) + (1 + aG)e^{-G(1+a)}} \tag{19}$$

For slotted 1-P CSMA,

$$S_{\mathrm{Slot-1P}} = \frac{G[1 + a - e^{aG}]e^{-G(1+a)}}{(1+a)(1-e^{-aG}) + ae^{-G(1+a)}} \tag{20}$$

For unslotted NP-CSMA,

$$S_{\mathrm{Unslot-NP}} = \frac{Ge^{-aG}}{G(1+2a) + e^{-aG}} \tag{21}$$

For slotted NP-CSMA,

$$S_{\mathrm{Slot-NP}} = \frac{aGe^{-aG}}{1 + a - e^{-aG}} \tag{22}$$

The plots of Eqs. (19), (20), (21), and (22), for $a = 0.01$, are shown in Fig. 14 (6, Fig. 11.19, p. 473). This figure shows that, for low levels of offered traffic, the persistent protocols provide the best throughput, but for higher load levels, the nonpersistent protocols are by far the best. The figure also shows that the slotted NP-CSMA protocol has a peak throughput almost twice that of 1-P CSMA schemes.

APPLICATIONS OF RANDOM-ACCESS TECHNIQUES IN CELLULAR TELEPHONY

The objective for earlier mobile radio systems was to achieve a large coverage area by using a high-powered transmitter with antenna on a tall tower to extend the receiving area. The extensive coverage from this approach has also resulted in limited user capacity capability, since increasing frequency reuse would certainly increase interference for the users of the system. At the same time, government regulatory agencies are not able to allocate frequency bands in a timely manner to keep up with demand for wireless services. It is therefore necessary to construct a mobile radio system to achieve both high capacity and large coverage area with the constraint of a crowded radio spectrum.

Cellular Communications Concept

The cellular communications concept was developed to provide the solution for spectral congestion and user capacity by

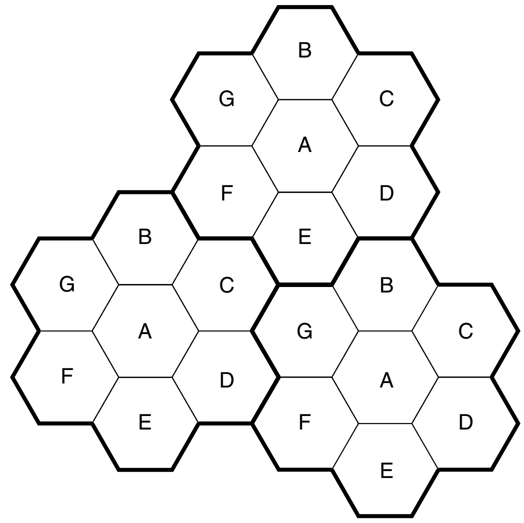

Figure 15. Illustration of the cellular frequency reuse concept. Cells with the same letter use the same set of frequencies. A cell cluster is outlined in bold and replicated over the coverage area.

replacing the single high power transmitter representing a large cell with several small low powered transmitters as in small cells. Figure 15 (21, Fig. 2.1, p. 27) illustrates the arrangement of the smaller cells to achieve frequency reuse in the allocated frequency band where cells labeled with the same letter use the same group of channels. The hexagonal shape of each cell serves to model the conceptual and idealistic boundary of each cell in terms of coverage and would be much more irregular in a real environment due to differing propagation effects and practical consideration in base station placement.

Cellular Telephone System Terminology. Figure 16 (22, Fig. 1.5, p. 15) shows a basic cellular telephone system consisting of *mobile stations, base stations,* and a *mobile switching service center* (MSC), sometimes called a *mobile telephone switching office* (MTSO). The function of the MSC is to provide connectivity to all *mobile units* to the *public switched telephone network* (PSTN) in a cellular system. Each mobile unit communi-

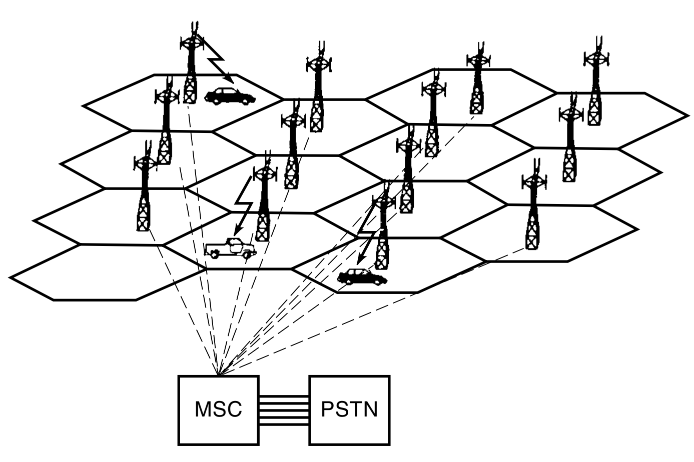

Figure 16. Illustration of a cellular system. The towers represent base stations, which provide radio access between mobile users and the mobile switching center.

cates with the base and may be handed off to any other base stations during the call.

The mobile unit handset contains a *transceiver,* antenna, and control unit, whereas a base station consists of several transmitters and receivers to handle simultaneous full duplex calls. The base station typically consists of a tower, with multiple antennas for receiving transmitting RF signals, and associated electronics at the base. The communication lines between the base station and the MSC can be regular telephone and point-to-point microwave links. The typical MSC handles the routing, billing, and system maintenance functions of calls to and from the mobile units, and multiple MSCs can be used together by a wireless operator.

The communication between the base station and mobile units is defined by a standard common air interface (CAI). The CAI typically specifies the communication parameters, such as multiple access methods and modulation type, and the use of four different channels for data transmission. From the base station to the mobile unit, the forward voice channel (FVC) is used for voice transmission and the forward control channel (FCC) is used for initiating and controlling mobile calls. From the mobile unit to the base station, the reverse voice channel (RVC) and the reverse control channel (RCC) accomplish the same functionality as the forward channel, only in the other direction to ensure full duplex communications.

All cellular systems provide *roaming* service for a cellular subscriber who uses the mobile unit in a service area other than the one area subscribed to by the mobile user. The registration of a roamer is accomplished by the MSC using the FCC to ask for all mobile units, which are not registered to report their MIN, and ESN reported over the RCC. This information is then used for validation as well as billing purposes.

The Process of a Cellular Call. When a mobile unit is first powered up, it scans for a group of forward control channels to find the strongest available one to lock on and changes to another channel when the signal level drops below a specified level. The control channels are standardized over a geographic area. The standard ensures that the mobile unit will be using the same control channel when ready to make a phone call.

Upon initiating a phone call on the reverse control channel using the subscriber's telephone number (*mobile identification number* or MIN), *electronic serial number* (ESN), called telephone number, and other control information, the base station relays this information to the MSC, which validates the request and makes the connection to the called party through the PTSN or through another MSC in the case of a called mobile unit. Once the appropriate full duplex voice channels are allocated, the connection is established as a phone call.

For a call to a mobile from a PSTN phone, the MSC dispatches the request to all base stations in the cellular system. Then the base stations, using a paging message, broadcast the called telephone number (or MIN) over the forward control channel. When the mobile unit receives the paging message, it responds to the base station by identifying itself over the reverse control channel. The base station relays this information to the MSC, which then sets up the appropriate voice channels and connection for the call.

Overview of Cellular Systems

Since the world's first cellular system was implemented by Nippon Telephone and Telegraph (NTT) and deployed in Japan in 1979, many other systems have been developed in other countries. Tables 2–4 list the cellular systems in three major geographical areas of the world—North America, Europe, and Japan.

In the United States, the Advance Mobile Phone System (AMPS) was introduced in 1983 by AT&T as the first major analog cellular system based in FDMA technology. By 1991, the TIA (Telecommunication Industry Standard) IS-54B digital standard was developed to allow US cellular operators to ease the transition of analog cellular phone to an all digital system using TDMA. To increase capacity in large AMPS markets, Motorola developed the narrowband AMPS (N-AMPS) that essentially provides three users in the 30 kHz bandwidth AMPS standard and thus reduces voice quality. By 1993, a cellular system based on CDMA was developed by Qualcomm Inc. and standardized as TIA IS-95. At the same time as IS-95, cellular digital packet data (CDPD) was introduced as the first data packet switching service that uses a full 30 kHz AMPS channel on a shared basis and utilizes slotted CSMA/CD as the channel access method. The auction of the 1900 MHz PCS band by the US government in 1995 opens the market for other competing cellular standards, such as the popular European GSM standard, which is implemented in the DCS-1900 standard.

In the United Kingdom, the E-TACS (Extended European Total Access Cellular System) was developed in 1985 and is virtually identical to the US AMPS system except for the smaller voice channel bandwidth. The Nordic Mobile Telephone (NMT) system in the 450 MHz and 900 MHz bands was developed in 1981 and 1986 using FDMA technology and was deployed in the Scandinavian countries. In Germany, a cellular standard called C-450 was introduced in 1985. Because of the need to standardize over these different cellular systems in Europe, the GSM (Global System for Mobile) was first deployed in 1991 in a new 900 MHz band dedicated as the cellular frequency band throughout Europe.

In Japan, JTACS and NTACS (Narrowband and Japanese Total Access Communications System) are analog cellular systems similar to AMPS and NAMPS. The Pacific Digital Cellular (PDC) standard provides digital cellular coverage using a system similar to North America's IS-54.

Major Cellular Systems

Currently, only a few of the cellular standards have survived or been developed into major systems around the world in terms of the number of users. These major systems are briefly described in this section.

AMPS and ETACS. In AMPS and ETACS, the FCC (forward control channel) continuously transmits control messages data at 10 kbit/s (8 kbit/s for ETACS) using binary FSK with a spectral efficiency of 0.33 bit/s/Hz. When a voice call is in progress, three in-band SATs (supervisory signal tones) at 5970 Hz, 6000 Hz, or 6030 Hz serve to provide a handshake between the mobile unit and base station. Other control signals are bursty signaling tone (ST) on the RVC to indicate end of call, and blank-and-burst transmission in the voice

Table 2. Cellular Standards in North America

Standard	Year of Introduction	Multiple Access Technique	Frequency Band (MHz), Reverse/Forward	Data/Control Parameters	Channel Bandwidth (kHz)
AMPS	1983	FDMA	824–849/869–894	FM/10 kbps FSK	30
IS-54	1991	TDMA	824–849/869–894	48.6 kbps π/4DQPSK/ 10 kbps FSK	30
NAMPS	1992	FDMA	824–849/869–894	FM/10 kbps FSK	10
CDPD	1993	FH/Packet	824–894	GMSK (BT = 0.5) 19.2 kbps	30
IS-95	1993	CDMA	824–894, 1.8–2.0 GHz	QPSK/BPSK	1.25
DCS-1900 (GSM)	1994	TDMA	1.85–1.99 GHz	GMSK	200

band having a duration less than 100 ms so as not to affect voice quality.

Prior to frequency modulation, voice signals are processed using a compander, a pre-emphasis filter, a deviation limiter, and a postdeviation limiter filter. These steps are taken to accommodate a large speech dynamic range, to prevent spurious emission, and to minimize interference with the in-band SAT signal. The channel coding on the forward and reverse control channels is BCH(40, 28) on FCC and BCH(48,36) on RCC. The line code used is Manchester.

IS-54. The analog AMPS system was not designed to support the demand for large capacity in large cities. Cellular systems using digital modulation techniques potentially offer large improvements in capacity and system performance. The IS-54 standard, also known as the USDC (US Digital Cellular) was set up to share the same frequencies, the frequency reuse plan, and base stations as AMPS so that both base stations and subscriber units can be provided with both AMPS and USDC channels within the same equipment. This way, US cellular carriers would be able to gradually replace analog phone and base stations with digital ones.

To maintain compatibility with AMPS phones, USDC forward and reverse control channels use exactly the same signaling techniques as AMPS while USDC voice channels use $\pi/4$ DQPSK at a rate of 48.6 kbit/s and spectral efficiency of 1.62 bit/s/Hz.

The numbers of USDC control channels are doubled from AMPS to provide flexibility to service offerings such as paging. There are three types of supervisory channels: the coded digital verification color code (CDVCC), whose function is similar to the SAT in AMPS, and the slow associated control channel (SACCH) and fast associated control channel (FACCH), which carry various control messages to effect power control and call processing.

The USDC voice channel occupies the 30 kHz bandwidth in each of the forward and reverse links and uses a TDMA scheme with six time slots to support a maximum of three users. For full-rate speech, each user is assigned two time slots in an equally spaced fashion as compared to one slot per user for half-rate speech.

The speech coder used in IS-54 is called the vector sum excited linear predictive (VSELP) code and is based on a code book that determines how to quantize the residual excitation signal. The VSELP coder has an output bit rate of 7950 bps and can produce a speech frame every 20 ms. The 159 bits of speech within a speech frame are divided into two classes according to their perceptual importance. Class 1 of 77 bits, being more important, are error protected using a rate $\frac{1}{2}$ convolutional code of constraint length $K = 6$, in addition to using a 7 bit CRC error detection code on the 12 most significant bits. Before transmission, the encoded speech data are interleaved over two time slots with the speech data from adjacent frames. For demodulation, differential detection may be performed at IF or base band, and equalization is needed based on training pattern imbedded in the data.

The IS-136 standard (formerly known as IS-54 Rev. C), recently introduced, is an improved version of IS-54. This standard comes with the addition of a DQPSK digital control channel to the existing FSK control channel, a greatly improved digital speech coder, new cellular features, and protocol additions to allow greater mobility management and better cellular service. The IS-136 protocol can be used in both the 800 MHz cellular band and the 1900 MHz PCS.

Global Mobile System. Global Mobile System or GSM utilizes two bands of 25 MHz set aside for system use in all

Table 3. Cellular Standards in Europe

Standard	Year of Introduction	Multiple Access Technique	Frequency Band (MHz), Reverse/Forward	Data/Control Parameters	Channel Bandwidth (kHz)
NMT-450	1981	FDMA	453–457.5/463–467.5	FM/10 kbps FSK	25
E-TACS (UK)	1985	FDMA	872–905/917–950	FM/10 kbps FSK	25
C-450 (Germany, Portugal)	1985	FDMA	450–455.74/460–465.74	FM	20/10
NMT-900	1986	FDMA	890–915/935–960	FM/10 kbps FSK	12.5
GSM	1990	TDMA and slow FH	890–915/935–960	GMSK (BT = 0.3)	200
DCS-1800	1993	TDMA	1710–1785/1805–1880	GMSK	200

Table 4. Cellular Standards in Japan

Standard	Year of Introduction	Multiple Access Technique	Frequency Band (MHz), Reverse/Forward	Data/Control Parameters	Channel Bandwidth (kHz)
NTT	1979	FDMA	400/800	FM	25
JTACS	1988	FDMA	860–925	FM/10 kbps FSK	25
PDC	1993	TDMA	810–830 1429–1453/940–960 1477–1501	$\pi/4$ DQPSK/10 kbps FSK	25
NTACS	1993	FDMA	843–925	FM	12.5

member countries. The multiaccess method is a combination of TDMA and slow FH. The use of FH combined with interleaving is for mitigation of fading caused by multipath transmission or interference effects. Frequency hopping is carried out on a frame-by-frame basis, and as many as 64 different channels may be used before the hopping sequence is repeated.

The available forward and reverse frequency bands are divided into 200 kHz wide channels. There are two types of GSM channels—traffic channels (TCH), carrying digitally encoded user speech or data, and control channels (CCH), carrying signaling and synchronizing commands between the base stations and subscriber units. There are three main control channels in the GSM system—*broadcast channel* (BCH), *common control channel* (CCCH), and *dedicated control channel* (DCCH). Each control channel consists of several logical channels distributed in time to provide the necessary GSM control function (22).

Each TDMA frame has 8 time slots for up to eight users with an aggregate bit rate of up to 24.7 kbit/s per user. The modulation used is 0.3 GMSK. The following full-rate speech and data channels are supported:

- Full-rate speech channel (TCH/FS) carries the user speech digitized at the raw rate of 13 kbit/s. With GSM channel coding applied, the full-rate speech channel is sent at 22.8 kbit/s.
- Full-rate data channel for 9600 bit/s (TCH/F9.6) carries raw user data sent at 9600 bit/s. With GSM forward error correction coding, this 9600 bit/s data is sent at 22.8 kbit/s.
- Full-rate data channel for 4800 bit/s (TCH/F4.8) carries raw user data sent at 4800 bit/s. With GSM forward error correction coding, this 4800 bit/s data is sent at 22.8 kbit/s.
- Full-rate data channel for 2400 bit/s (TCH/F2.4) carries raw user data sent at 2400 bit/s. With GSM forward error correction coding, this 2400 bit/s data is sent at 22.8 kbit/s.
- Half-rate speech channel (TCH/HS) carries the user speech digitized at half the rate of full-rate speech. With GSM channel coding applied, the full-rate speech channel is sent at 11.4 kbit/s.
- Half-rate data channel for 4800 bit/s (TCH/H4.8) carries raw user data sent at 4800 bit/s. With GSM forward error correction coding, this 4800 bit/s data is sent at 11.4 kbit/s.
- Half-rate data channel for 2400 bit/s (TCH/H2.4) carries raw user data sent at 2400 bit/s. With GSM forward error correction coding, this 2400 bit/s data is sent at 1.4 kbit/s.

The GSM speech code is based on the residually excited linear predictive (RELP) coding, which is enhanced by including a long-term predictor. The GSM coder takes advantage of the fact that, in a normal conversation, a person speaks less than 40% of the time on average. By incorporating a voice activity detector (VAD), the GSM system operates in a discontinuous transmission mode, thus providing longer subscriber battery life and reduced radio interference when the transmitter is not active during the speech silent period. Channel coding for speech and control channels is based on a rate $\frac{1}{2}$ convolutional encoder with constraint length $K = 5$, whereas channel coding for data channels is based on a modified CCITT V.110 modem standard.

Security is built into GSM by ciphering the contents of the data block with encryption keys known only to the base station and the subscriber unit and is further enhanced by changing encryption algorithm from call to call.

IS-95. Similar to IS-54, TIA IS-95 is designed to be compatible with the existing US analog system, where base stations and mobile units can work in the dual mode operation. Since this is a direct-sequence CDMA system, the need for frequency planning within a region is virtually eliminated.

Specification for IS-95 reverse link operation is in the 824–849 MHz band and forward link operation is in the 869–894 MHz band. The maximum user data rate is 9600 bps and is spread to a channel chip rate of 1.2288 Mchips/s using a combination of techniques. Each mobile subscriber is assigned a different spreading sequence to provide perfect signal separation from other users.

Unlike other cellular standards, the user data rate but not the channel chip rate changes in real-time depending on the voice activity and network requirement. On the forward link, the base station simultaneously transmits user data for (or broadcast to) all mobile subscribers by using a different spreading code (Walsh functions) for each subscriber. A pilot code is also broadcast at a high power level to allow all mobiles to perform coherent carrier detection while estimating the channel condition. On the reverse link, all mobiles would respond asynchronously and have a constant signal level due to power control exercised by the base station to avoid the "near–far problem" arising from different received power levels.

The user data stream on the reverse link is first convolutionally coded with a $\frac{1}{3}$ rate code. After interleaving, each block of six encoded symbols is mapped to one of the 64 orthogonal Walsh functions to provide 64-ary orthogonal signaling. A final fourfold spreading, giving a data rate of 1.2288 Mchips/s, is achieved by user specific codes having periods of $2^{42} - 1$ chips and base station specific codes having period of 2^{15}. For the forward traffic channels, Table 5 summarizes the modulation parameters for different data rates.

Note that Walsh functions are used for different purposes on the forward and reverse channels. On the forward channels, Walsh functions are used for spreading to indicate a particular user channel, whereas on the reverse channel, Walsh functions are used for data modulation.

The speech encoder exploits gaps and pauses to reduce the output from 9600 bps to 1200 bps during the silent period. Rates lower than 9600 bps are repeated to achieve a constant coded rate of 19,200 symbols per second for all possible information data rates.

At both the base station and subscriber unit, RAKE receivers (1) are used to combine the delayed replica of the transmitted signal and therefore reduce the degree of fading. In IS-95, a three finger RAKE receiver is used at the base station.

Cellular Digital Packet Data. There are a number of wide-area packet-switched data services being offered over a dedicated network using the specialized mobile radio (SMR) frequency band near 800/900 MHz, for example, ARDIS (Advanced Radio Data Information Service) and RAM Mobile Data System. However, CDPD is the packet-switched network that uses the existing analog cellular network such as AMPS. CDPD occupies the voice channel on a secondary, non-interfering basis by utilizing the unused airtime between channel assignment by the MSC, which is estimated to be 30% of the time. CDPD supports broadcast, dispatch, electronic mail, and field monitoring applications. Figure 17 (2, Fig. 14.2, p. 361) illustrates a typical CDPD network.

The CDPD network has three interfaces: the *air link interface* (A-Interface), the *external interface* (E-Interface) *for external network interface,* and the *inter-service provider interface* (I-Interface) for cooperating CDPD service providers. The mobile subscribers (M-ES) are able to connect through the *mobile data base stations* (MDBS) to the Internet via the *intermediate systems* (MD-IS and IS), which act as servers and routers for the subscribers. Through the I-Interface, CDPD can carry either Internet protocol (IP) or OSI connectionless protocol traffic.

In CDPD, the forward channel serves as a beacon and transmits data from the PSTN side of the network while the reverse channel serves as the access channel and links all the mobile subscribers to the CDPD network. Collisions result when many mobile subscribers attempt to access the channel simultaneously and are resolved by slotted DSMA/CD.

Table 5. Summary of Forward Traffic Channel Modulation Parameters

Parameter	Data Rate (bps)			
User data rate	9600	4800	2400	1200
Coding data rate	1/2	1/2	1/2	1/2
Data repetition period	1	2	4	8
Baseband coded data rate	19,200	19,200	19,200	19,200
PN chips/coded data bit	64	64	64	64
PN chip rate (Mcps)	1.2288	1.2288	1.2288	1.2288
PN chips/bit	128	256	512	1024

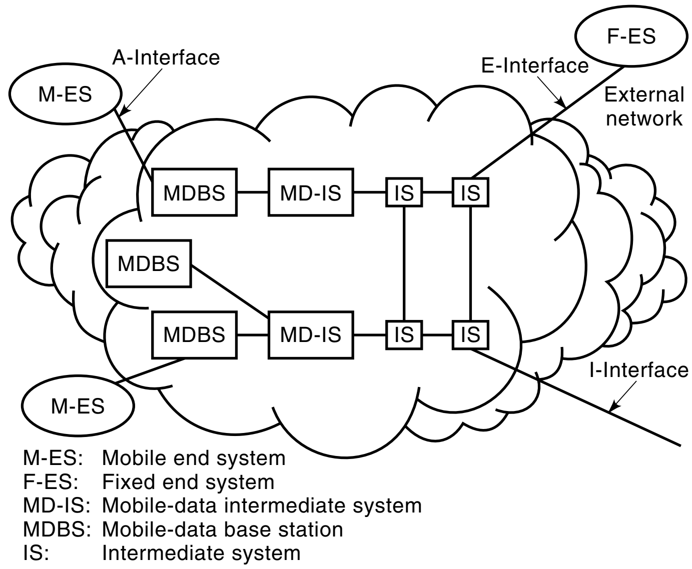

Figure 17. Cellular digital packet data network.

At the physical layer, CDPD transmissions are carried out using fixed-length blocks. The channel coding used is Reed–Salomon (63,47) block code with 6 bit symbols. For each packet, 282 bits are encoded into 378 bit blocks and provide correction for up to eight symbols. At the OSI layer 2, the mobile data link protocol (MDLP) is used to convey information between the data link layer across the common air interface. The MDLP also provides logical data link connection, sequence control, error detection, and flow control. The radio resource management protocol (RRMP) is a layer 3 function used for the management of radio resources, base station identification and configuration, channel hopping, and handoffs.

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