# MULTIPLE ACCESS MOBILE COMMUNICATIONS

Multiplexing and multiple access refer to the sharing of a physical medium, often the radio spectrum, among different signals or users. When all signals access the medium through a common access point and can easily be coordinated, this is usually referred to as multiplexing. When the signals access the medium from different physical locations, this is usually referred to as multiple access. The key shared resources are time and radio frequency.

There are several possible physical configurations for the multiple access channel or system. Figure 1(a) shows a centralized network where every user communicates with a central access node (or base station) and vice versa. Signals are multiplexed in the forward direction from the access node to the mobile terminal, and a multiple access strategy is used in the return direction. The access node could provide a connection to a wired network such as the public switched telephone network (PSTN) or it could be a private dispatch office. When there are a number of access points in the system, then the access nodes become an important resource that must be shared. This introduces a third dimension to the multiple access problem, in addition to time and frequency. Figure 1(b) shows a decentralized network where each user can communicate directly with the other users. In this case, a multiple access strategy, often the same one, is required for both transmitting and receiving. In the following, we will concentrate mainly on centralized networks, although many of the ideas carry over to decentralized networks.

There are a number of similarities between the multiple access issues for fixed and mobile wireless systems. The main difference with wireless mobile systems is the time-varying nature of the communications channel. In mobile communications, multipath fading is the time-varying amplitude of the received signal resulting from constructive and destructive interference that is caused by receiving the same signal from multiple reflected paths. Shadowing is signal attenuation or

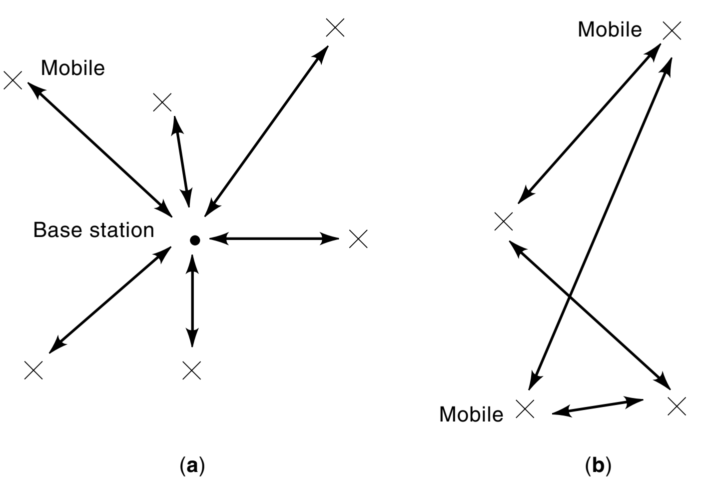


**Figure 1.** The (a) centralized and (b) decentralized networks are two examples of mobile multiple access channels.

blockage due to terrain, buildings, or vegetation. Both are important propagation effects. These propagation effects make single user communications more difficult and result in significant power differentials between users. Consequently, there is greater potential for harmful interference. A significant part of the design of a successful multiple access strategy is controlling the mutual interference between users to an acceptable level.

The major multiple access issues that are unique to mobile applications are

1. *Multipath Fading and Shadowing.* These problems are usually addressed through the modulation, coding, and antenna strategy. The solutions are usually some form of diversity—sending signals by multiple paths—and are often a combination of time, frequency, and space diversity. Time diversity is spreading information in time, usually through use of a forward error correction code and interleaving. Frequency diversity can be obtained by sending the same signal at multiple frequencies or spreading the signal over a large bandwidth. Space diversity is obtained by sending the same information over multiple radio links.
2. *The Near-Far Problem.* Received signal strength naturally decreases as a function of the distance between the transmitter and receiver. In a mobile environment, the rate of decrease is faster because of shadowing, and it implies potentially large differences in the received signal strength of signals from transmitters at different distances. The traditional solution is frequency separation (guard bands) between adjacent channel users and spatial separation between cochannel users. Recent systems have resorted to dynamic power control techniques to relax these separation requirements.
3. *Synchronization and Tracking.* Spectral efficiency, the average number of user transmissions per unit spectrum on a system-wide basis, can often be improved if there is some form of synchronization between users. In a mobile environment, acquiring and maintaining synchronization is a challenging and dynamic problem.
4. *Handover and Paging.* When there are multiple access points in a system, it is often necessary during a mobile communication session to switch between one access point and another. This handover requires that an access node keep sufficient resources to handle calls that are in progress at other access points, and which could potentially be handed over. Paging refers to the problem of locating a user to receive a call with a minimum of radio resources, when there are multiple access points in a system. These are important secondary issues in a cellular system with mobile users.

The object of an efficient multiple access strategy is to maximize the number of users that can simultaneously access the system. In a digital system, this *spectral efficiency* is often measured in bits/sec/Hz/user. Because the key constraint limiting the number of users is mutual interference, a design goal is to minimize interference between users, or in a sense, to make users as orthogonal as possible.

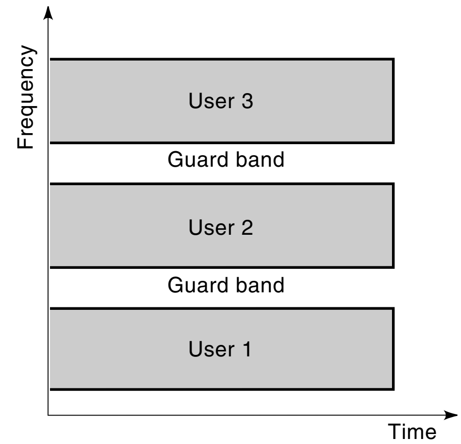


**Figure 2.** With FDMA, each user is given a dedicated frequency band in the time–frequency plane.

## BASIC TECHNIQUES

### Frequency Division Multiple Access

The traditional multiple access strategy is frequency division multiple access (FDMA). With this approach, the time-frequency plane is divided into a number of channels as illustrated in Fig. 2. Each user is assigned a distinct channel and is thus frequency "orthogonal" to the other users. This orthogonality is not perfect. Implementation limitations mean that the *out-of-band transmissions* of any user are nonzero. Causes of these out-of-band transmissions can be poor frequency control, poor transmit filtering, and amplifier nonlinearities. There is usually a requirement to limit these out-of-band transmissions to a specified level. This level will depend on the application and is often determined by the potential power differences between adjacent channel users as received at the base station. In mobile radio applications where one user can be significantly closer to the base station than another, these power differences can be as much as 80 dB or more. This is referred to as the *near–far problem.* This *adjacent channel interference* determines how closely the channels can be spaced, and ultimately the spectral efficiency. FDMA has the advantage that it is a relatively simple system. If all transmitters meet the out-of-band transmission limits then coordination of the users is simply a matter of assigning frequencies. In the simplest scenario, these frequencies are assigned on a fixed basis, but in many systems they are assigned on a dynamic basis when the user requires service. In the latter case, in addition to the traffic channels shown in Fig. 2, the system will need signaling channels to handle channel requests and channel assignments.

### Time Division Multiple Access

With time division multiple access (TDMA), the time-frequency plane is divided into time slots as shown in Fig. 3. Each user is assigned different slots, often on a periodic basis, during which transmissions are allowed. This approach reduces the spectrum wasted because of the *guard bands* required with FDMA, since frequency errors are typically a smaller fraction of the transmitted bandwidth. However, it introduces the need for time synchronization between users.

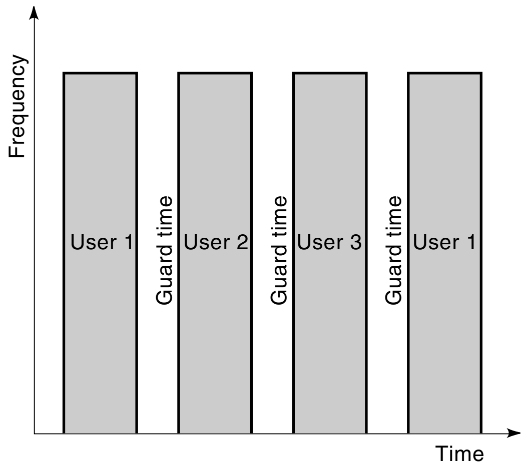


**Figure 3.** With TDMA, each user is given dedicated, often periodic, time slots in the time–frequency plane.

In a centralized network, coarse time synchronization can usually be obtained from the forward link, either explicitly from a transmitted timing reference or implicitly from the multiplexed signal. However, fine time synchronization is often necessary to compensate for the different distances of the users from the access node. This fine synchronization is often done through a feedback loop between the access node and each mobile unit. The use of global timing sources such as the Global Positioning Satellite (GPS) system can sometimes simplify this. Synchronization is never perfect in practice, and some guard times must be left to allow for timing errors between user transmissions. The spectral efficiency of the system depends on the ratio between these guard times and the length of a time slot (transmission burst). In mobile applications, timing errors are compounded because the terminal's distance to the base station varies as the terminal moves. Thus, a feedback mechanism must be implemented to track the timing and insure synchronization is maintained. A disadvantage of TDMA, relative to FDMA, is that it requires higher peak powers from the transmitter, as the instantaneous data rate is higher to achieve the same average throughput.

### Code Division Multiple Access

With code division multiple access (CDMA) users are assigned unique codes to access the time-frequency plane, which produces low correlation with the signals of other users, that is, minimal average interference. There are two major variants: frequency-hopped spread spectrum (FHSS) and direct sequence spread spectrum (DSSS). These techniques were originally developed for military communications because of their low probability of interception, but, they have since found commercial application. With FHSS, the conceptual approach is to divide the time-frequency plane into time and frequency slots as shown in Fig. 4(a). Each user is given a distinct pseudorandom sequence that defines which time and frequency slots to use. This sequence is known by the receiving terminal but not necessarily by other users. Given a large number of frequency slots and short time slots, the probability of users colliding is low, depending upon the number of users, and the effects of collisions can often be compensated by error correction coding. With DSSS, each terminal uses a distinct modulating waveform derived from a pseudorandom sequence of bits. These modulating waveforms are approximately orthogonal, that is, they have low cross correlation with each other. These modulating waveforms or spreading codes span the allocated frequency band, as shown in Fig. 4(b). A conventional DSSS receiver correlates the received signal, which is the sum of all user signals, with the modulating waveform of the desired terminal. The desired signal produces a strong correlation, while the other signals produce weak correlations. The interference caused by other terminals because of their imperfect orthogonality can often be approximated as Gaussian noise. As with FDMA, there is a serious near–far problem with CDMA, and for mobile radio applications *power control* is often a requirement. This implies a feedback loop between the base station and the mobile to dynamically adjust mobile transmit power to obtain an acceptable level at the base station receiver.

In their simplest form, the capacity of FHSS and DSSS systems are typically quite low, compared to FDMA and TDMA, without some coordination between users. In CDMA, this coordination takes the form of power control. While FHSS is relatively insensitive to power level variations, capacity can be significantly improved with some synchroniza-

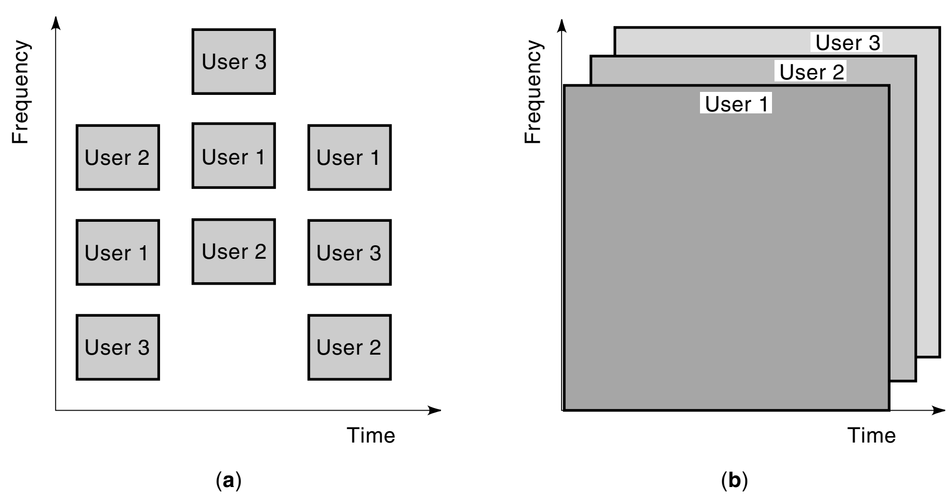


**Figure 4.** With CDMA, users are not dedicated time and frequency slots in the time–frequency plane. (a) With FHSS, users are independently assigned random time and frequency slots and packet collisions are possible. (b) With DSSS, users are assigned modulating waveforms that span the time–frequency plane, but have low cross-correlations with other users.

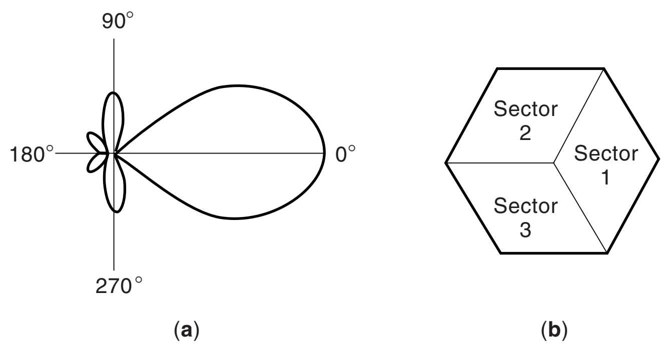


**Figure 5.** SDMA uses directional antennas to subdivide the service area: (a) polar plot of antenna gain versus azimuth angle, and (b) division of service area into sectors with 120° antennas located at center of service area.

tion between users. In practice, both approaches rely heavily on forward error correction (FEC) coding to reduce the effect of multiple access interference.

### Spatial Division Multiple Access

Spatial division multiple access (SDMA) is a technique that can be overlaid on any of the previous time and frequency sharing techniques to allow sharing in space. This is one of the primary techniques for allowing frequency reuse. In its simplest form, systems are allowed to reuse the same frequency by physically separating the service areas where a particular frequency is used, so that the natural attenuation of signal strength with distance insures that interference is reduced to minimal levels. An example of this is commercial AM and FM broadcast radio, where frequencies are only reassigned with separations of hundreds of kilometers. It is possible to more fully exploit the spatial dimension in a wireless communication system by equipping the access points of a wireless network with directional antennas. A directional antenna, with a gain versus azimuthal angle characteristic such as shown in Fig. 5(a), can increase the base station range and improve coverage. Due to the directivity of the antenna, interference may be reduced, resulting in improved performance. The capacity gains that result depend upon the modulation technique. With narrowband modulations such as used with FDMA, the interference is often not reduced enough to allow reuse of the same frequencies in adjacent antenna beams. Thus, with FDMA and a three-sector system, as shown in Fig. 5(b), there would be improved performance but no increase in capacity. With greater sectorization, one can reuse frequencies and increase capacity. With the wideband modulations typical of CDMA, one can achieve nearly full spectral reuse with as few as three sectors. A second key to maximizing frequency reuse is to limit the transmitted power to the minimum required. With mobile transmitters this implies that power control is an important factor in all multiple access strategies, and not just with CDMA.

## SUMMARY

In practice, a combination of techniques is often used. Two common combinations are FDMA/TDMA/SDMA and FDMA/CDMA/SDMA. The choice of technique is a tradeoff between economics and difficulties associated with the side issues. The economics depend on system capacity and expected fiscal return. Potential side issues include growth options, available spectrum, signaling demands, integration of different services having potentially different data rates, and performance requirements. The capacity or spectral efficiency of a wireless system depends on the combination of the modulation and coding strategy of the individual user and the choice of multiple access strategy. The choice of modulation and coding strategy will depend on the service parameters, such as data rates, tolerable delay, tolerable outages, performance requirements and complexity. Often what is spectrally efficiency from a single-user viewpoint is not spectrally efficient from a system level.

## MOBILE RADIO SPECTRUM

Since the late 1980s, there has been a huge growth in wireless telecommunications, and a significant portion of this is mobile. However, the amount of useable radio spectrum is limited and regulated. Internationally, the radio spectrum is regulated by the International Telecommunications Union (ITU) based in Geneva, Switzerland. The recommendations of this international body are enforced by local authorities. The limited spectral resources has led to significant competition and auctions for spectral licenses in some countries. Some frequency bands allocated for mobile radio and the corresponding user population are listed in Table 1.

The majority of current mobile radio systems are in frequency bands lower than 2 GHz, primarily because of limitations in mobile terminal technology. Although these limitations are disappearing, the demand for spectrum is still outstripping the supply, which emphasizes the importance of spectrally-efficient multiple access strategies. The dominant application to this point has been voice, until quite recently, using analog transmission techniques such as frequency modulation and single-sideband amplitude modulation. However, digital techniques (1) permit greater spectral efficiency; (2) allow applications such as fax and data; and (3) promise mobile multimedia for the future.

**Table 1. Some Mobile Radio Bands and Example Applications**

| Band | Some Applications |
|---|---|
| 118–136 MHz | Aeronautical safety radio |
| 150–174 MHz | Public safety radio |
| 450–470 MHz | European FM cellular telephony |
| 825–870 MHz | North American FM cellular and first generation digital |
| 902–928 MHz | ISM band for low-power unlicensed spread-spectrum users (North America) |
| 890–960 MHz | GSM cellular telephony |
| 1452–1492 MHz | Digital audio broadcasting |
| 1525–1559 MHz | Mobile satellite downlink |
| 1610–1660 MHz | Mobile satellite uplink |
| 1930–1980 MHz | PCS cellular telephones |
| 1980–2010 MHz | Mobile satellite telephony downlink |
| 2170–2200 MHz | Mobile satellite telephony uplink |
| 2400–2480 MHz | ISM band for low-power unlicensed spread-spectrum users (North America) |

## GENERALIZED MULTIPLE ACCESS

Proakis (1) shows that a real valued signal $s(t)$ with frequency content concentrated in a band of frequencies near a frequency $f_k$ can be expressed as

$$s_k(t) = a_k(t)\cos[2\pi f_k + \theta_k(t)] \tag{1}$$

where $a_k(t)$ represents the amplitude and $\theta_k(t)$ represents the phase. We use the subscript $k$ to indicate that this is the signal of the $k$th terminal in a multiple access system with $K$ users. The received multiple access signal can then be represented as the sum of Eq. (1) over all users

$$r(t) = \sum_{k=1}^{K} w_k s_k(t-\tau_k) + n(t) \tag{2}$$

where $\tau_k$ are the relative delays, and $w_k$ are the propagation losses of the different users, respectively. The factor $w_k$ can also include a relative phase rotation for each user. The term $n(t)$ represents additive white Gaussian noise. The delays and propagation losses are position dependent, and thus for a mobile user may also be time dependent. Equation (2) represents what is referred to as a flat fading, or frequency-independent, model of a mobile channel (2). Depending upon the bandwidth of the signal and nature of channel, other models may be appropriate (2).

Although multiple-access questions apply to the transmission of both analog and digital information, in this article we will focus on digital systems. To emphasize this, we will often represent each user's signal as $s_k(\mathbf{b}_k,t)$ where $\mathbf{b}_k$ represents the information bits in the message of the $k$th user. A message could represent a data packet, for example, or a segment of digitally encoded speech. For digital signals, a linear modulation strategy (1) is often used and the $k$th user's signal can then be expressed as

$$s_k(b_k,t) = \sum_{i=1}^{N} b_k(i)p_k(t-iT)\cos(2\pi f_k t + \phi_k) \tag{3}$$

where $T$ is the symbol period, $N$ is the number of transmitted symbols, $\mathbf{b}_k = \{b_k(1), \ldots b_k(N)\}$ are the data symbols, and $p_k(t)$ is the time domain representation of the pulse shaping or filtering applied to the data.

For an FDMA scheme, the modulation schemes of the users are typically identical, for example, with linear modulation the pulse shaping $p_k(t)$ is the same for all users, and only the center frequencies, $f_k = f_{\min} + (k-1)\Delta f$, differ where $\Delta f$ is the frequency spacing of channels. A critical issue for FDMA is guard bands required to limit adjacent channel interference (ACI). This depends upon the modulation and filtering strategies and nonlinearities present in the transmit chain. It also depends upon the frequency accuracy of the transmitter oscillator and Doppler-induced frequency errors due to terminal motion. The Doppler shift $f_D$ is given by $f_D = f_{RF}v/c$ where $f_{RF}$ is the radio frequency, $v$ is the speed of the terminal in the direction of the receiver, and $c$ is the speed of light. This frequency shift, $f_D$, can be greater than a kilohertz for an aircraft terminal operating at 1.5 GHz.

For a TDMA strategy, user modulation schemes are also typically identical, except that each user is only allowed to transmit during a preassigned time interval. Unlike FDMA, the center frequency is identical for all users $f_k = f_j$ all $k$ and $j$. With mobile users, a feedback mechanism must be included to track delay variations and maintain timing sync. The delay is given by $\tau_k = r_k/c$, where $r_k$ is the distance between the transmitter and the receiver.

For a CDMA strategy, all users have a common frequency band, and the modulation depends upon whether DSSS or FHSS is used. With DSSS, a linear modulation is often used and Simon et al. (3) show that the pulse shaping in Eq. (3) for a DSSS binary phase-shift keyed (BPSK) signal can be expressed as

$$p_k(t-iT) = \sum_{j=1}^{P} a_k(j)p_c(t-jT_c-iT) \tag{4}$$

where $p_c(t)$ corresponds to a rectangular pulse of width $T_c$, $P$ is the number of these pulses (chips) per symbol, and $\{a_k\}$ is the spreading sequence. The spreading sequence is usually synchronous with the bit sequence $\{\mathbf{b}_k\}$, that is, it repeats every symbol period, but it is sometimes overlaid with a further randomizing code that has a much longer period. The bandwidth of the resulting signal is approximately $1/T_c$. Considerable research has been invested in determining the optimum spreading or pseudo-noise (PN) sequence. Maximal-length sequences and Gold codes (3) are two common examples.

Taking a basic modulation technique and changing the carrier frequency in some pseudorandom manner is the frequency-hopping approach to generating a spread spectrum signal. The modulation technique often used is $M$-ary frequency-shift keying ($M$-FSK) (1). When binary frequency-shift keying (2-FSK) with FHSS, Simon et al. (3) show that the transmitted signal can be presented as

$$\begin{gathered} s_k(b_k(i),t) = \cos[2\pi(f_c + f_{k,i} + b_k(i)\Delta f)t + \phi_k] \\ iT \le t < (i+1)T \end{gathered} \tag{5}$$

where $f_{k,i}$ is a sequence of randomly chosen frequencies, and $\Delta f$ is the modulation frequency. Each user uses a different sequence of hop frequencies $\{f_{k,i}\}$ that range over the allocated bandwidth and are known at the receiver. Frequency-hopping strategies are classified into fast- and slow-hopping, depending upon how long the transmitter dwells on a particular hop frequency. With the former, only one or a fraction of one symbol is transmitted each dwell time. With the latter, multiple symbols are transmitted at each frequency. Slow-hopping strategies may use FSK but are often combined with a TDMA strategy and use $M$-ary phase-shift keyed (PSK) modulation. This combination provides a degree of frequency diversity that is not normally available with a TDMA strategy.

### Conventional Detectors

Whalen (4) shows that the optimum linear single-user receiver for an additive white Gaussian noise channel is a bank of correlators matched to all possible transmitted sequences for the $k$th user. That is, for the $k$th user, one computes the correlation values

$$L_j = \int r(t)s_k(\mathbf{b}_k^j,t)\,dt \tag{6}$$

for all possible data sequences $\{\mathbf{b}_k^j : j = 1 \ldots 2^N\}$ of the $k$th user, assuming binary modulation, and chooses the one with the largest $L_j$ value. This optimum receiver has a complexity that is exponential in the sequence length. In practice, when specialized to a particular modulation, the complexity of the receiver can often be reduced to a linear function of the sequence length. Proakis (1) describes specific modulation techniques and the corresponding receivers. There it is shown that, for linear modulation, see Eq. (3), a sufficient statistic for optimum detection of an individual data symbol is given by

$$y_k(i) = \int_{-\infty}^{\infty} r(t) p_k(t - iT - \tau_k) \cos(2\pi f_c t + \phi_k)\, dt \qquad (7)$$

That is, the optimum receiver for an additive white Gaussian noise channel filters the received signal with a filter matched to the transmitted pulse shape. This receiver can be applied to all the multiple access systems described thus far. Implicit in Eq. (7) is the assumption that the modulating waveform of each user is known at the receiver, and that the receiver locks to the signaling interval and phase of each active user. In a mobile multiuser environment, constructing this type of a coherent receiver is a challenging problem that is not dealt with here. For binary signaling, a conventional detector without FEC coding simply takes the sign of the bits at the output of the matched filter, that is, $b_k(i) = 1$ if $y_k(i) > 0$ and $-1$ otherwise. In practice, most current receivers are an approximation of the optimum receiver. This is what we will assume in the remainder. With this type of receiver, the interference due to another user can be represented as

$$I_j(i) = \int_{-\infty}^{\infty} w_j s_j p_k(t - iT - \tau_k) \quad \cos(2\pi f_c t + \phi_k)\, dt \qquad (8)$$

or its frequency domain equivalent. This interference is classified as cochannel or adjacent channel interference, depending upon whether most of the spectrum of $s_j(t)$ overlaps that of $s_k(t)$ or not. The sum of the interference from all the other users is often referred to as the multiple access interference (MAI).

### Multiplexing for the Forward Link

In the forward direction, base station to mobile, there is the opportunity to coordinate and synchronize users to minimize the multiple access interference. That is, one can provide access to the channel on a contention-free basis. There are a number of ways to multiplex several users onto a single channel.

**Time-Division Multiplexing.** In packet switched networks, a common method is time-division multiplexing (TDM) of the packets for each user. That is, the bits of each packet are transmitted sequentially over the same channel with no need for guard times between the packets. Depending upon the regularity of the users, each packet may include addressing information, and each terminal monitors all packets to determine those that are addressed to it, as shown in Fig. 6. If packets are regular or periodic, then individual addressing may be forgone in favor of a look-up table that is broadcast at less frequent intervals. This approach does not require synchronization or guard time overhead to be associated with each packet because it is a continuous data stream. Some regular synchronization information is required, however, to speed initial acquisition and aid reacquisition, should the mobile terminal lose the signal.

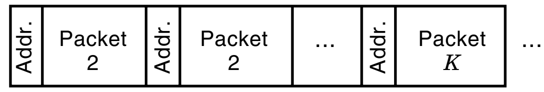


**Figure 6.** Time-division multiplexing of different user packets eliminates the need for guard times.

**Frequency-Division Multiplexing.** For analog channels, frequency-division multiplexing (FDM) is a common traditional approach, where each mobile is assigned a unique forward frequency for the duration of the call. This has the advantage of simplicity, and that the mobile can use the dedicated channel for just about any application that fits in the defined channel bandwidth. For this reason it has also been used in many digital and analog transmission systems.

**Orthogonal Frequency-Division Multiplexing.** An enhancement to basic FDM is orthogonal frequency-division multiplexing (OFDM). Weinstein (5) shows that digitally modulated carriers can be significantly overlapped without harmful interference, as long as the carrier spacing was equivalent to the symbol period, eliminating the need for guard bands. That is, in a system with $K$ carriers, data is transmitted $K$ symbols at a time, which can be represented as

$$s(t) = \sum_{k=-K/2+1}^{K/2} \mathbf{b}_k(i) e^{-2\pi j(f_c + k f_u)t} \quad iT_u \le t < (i+1)T_u \qquad (9)$$

where $f_u = 1/T_u$. This corresponds to $K$ different users or some other combination of data from fewer users. There are several advantages to this approach. The first is that the modulator and demodulator can be implemented as a Discrete Fourier Transform (DFT) when pulse-shaping is rectangular. With this approach a set of $K$ data, often referred to as frequency-domain symbols, are transformed by the inverse DFT (6) to form a set of time domain symbols $B_n(i)$ to be transmitted over the channel (7),

$$B_n(i) = \sum_{k=-k_2+1}^{K_2} \mathbf{b}_k(i) e^{j2\pi in/K} \quad n = 0, \ldots, K-1 \qquad (10)$$

and these samples are transmitted sequentially in the interval $iT_u \le t < (i + 1)T_u$ on a carrier of frequency $f_c$. This corresponds to a sampled version of Eq. (9) with $K$ samples per symbol period. The demodulator is implemented using a DFT. This has a fast implementation when $K$ is a power of 2, or a product of prime powers, that is known as the Fast Fourier Transform (FFT). From a transmission viewpoint, the multicarrier approach is advantageous for higher data rates in a fading environment where there may be frequency-selective fading. With the multicarrier rates, the effective symbol rate is reduced by a factor equivalent to the number of carriers. Since any time dispersion—multipath where the relative delays of the different paths are significant relative to the symbol interval—will cause intersymbol interference (ISI) (1), a short guard time is usually added to each symbol period to avoid ISI due to multipath. If the resulting guard time is longer than the expected time dispersion of the channel (2), there is not a need for an equalizer other than for channel gain and phase compensation. A disadvantage of OFDM is that it can be sensitive to frequency errors. A second disadvantage is that the transmitted signal is not a constant envelope and requires a linear transmitter. This is often not a concern for transmissions from a base station.

**Code-Division Multiplexing.** An alternative approach to multiplexing the data from several users is code division multiplexing (CDM). Because one can make the sequences of the different users synchronous at the transmitter, it is possible to choose perfectly orthogonal spreading codes for the different users. When the spreading factor is a power of 2, a common choice for the spreading codes are the rows of the corresponding Hadamard matrix, which is given recursively by

$$H_1 = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} H_2 = \begin{bmatrix} H_1 & H_1 \\ H_1 & -H_1 \end{bmatrix} \ldots H_p = \begin{bmatrix} H_{p-1} & H_{p-1} \\ H_{p-1} & -H_{p-1} \end{bmatrix} \tag{11}$$

The rows of these matrices are often referred to as Walsh functions ($H_p$ provides $2^p$ Walsh functions), and they would be used as the $\{a_k(j), j = 1 \ . \ . \ . \ 2^p\}$ in Eq. (5). Similar to OFDM, this approach spreads each user over the available bandwidth and has potential frequency diversity advantages in a frequency selective fading environment. Time dispersion of the signal due to multipath is a potential problem with wideband signals, such as, non-flat fading, and Proakis (1) shows that a RAKE receiver can be employed to recover the energy in time-dispersed multipath channels.

### Multiple Access for the Return Link

The number of issues on the return link is greater than on the forward link, and they are difficult to treat in isolation. In the following, we present a number of multiple access protocols and show how they deal with these difficulties. Packet switched networks, cellular systems, and spread spectrum systems have been selected as examples, but the techniques presented can be applied to a much wider variety of systems.

**Packet Switched Networks.** The problem that packet switched networks attempt to solve is the sharing of packets of digital information between a number of different users who share a common channel. The classic sample of this is the ARPANET network (8), which consisted of a number of research institutions that were linked by a common satellite channel. One characteristic that is used advantageously in this system is that, because of the delay when transmitting over a geostationary satellite (approximately 240 ms), one can listen to one's own transmission if the packet is sufficiently short. This gave rise to a number of packet-switching protocols that require increasing degrees of cooperation between the mobile units.

***Pure ALOHA.*** The first such protocol has come to be known as pure ALOHA, in which users transmit any time they desire. If, after one propagation delay, they hear their successful transmission, then they assume no collision with a packet from another user has occurred. Otherwise, a collision is assumed and the packet must be retransmitted. The pure ALOHA strategy is a form of uncoordinated TDMA. Let the packet transmission period be $T_p$ and let $\eta$ denote the channel throughput or efficiency (average number of successful transmissions per transmission period $T_p$). Collisions between packets (cochannel interference) is the main source of degradation here. If the total channel traffic $G$, the average number of packets (initial plus retransmitted) offered per transmission period $T_p$, comes from an infinite population of users each with an independent Poisson distribution, then Kleinrock (8) shows

$$\eta = Ge^{-2G} \tag{12}$$

The maximum efficiency is $1/2e \approx 0.184$ which occurs at $G = 1/2$.

***Slotted ALOHA.*** The second technique is known as slotted ALOHA, and is a more coordinated form of TDMA, where time is segmented into slots matching the packet length $T_p$ (plus some guard time) and all users are required to confine their transmissions to slots. This confines a collision between packets to a slot and results in increased efficiency. Then, for $K$ statistically equivalent users with total offered traffic, Kleinrock (8) shows that

$$\eta = G\left(1 - \frac{G}{K}\right)^{K-1} \xrightarrow[K\to\infty]{} Ge^{-G} \tag{13}$$

Thus, the slotted protocol has double the maximum efficiency of the pure ALOHA system. To compute the average packet delay, let $R$ represent the delay (in slots) before a user knows whether a transmission was successful. When a collision is detected, the packet is retransmitted, at random, in one of the $M$ subsequent slots. Then, the average packet delay in slots is approximately (8)

$$T \cong e^G\left[R + 1 + \frac{M-1}{2}\right] - \frac{M-1}{2} \tag{14}$$

when $M > 10$. The delay increases exponentially as the loading on the channel increases. There is a fundamental tradeoff between throughput and delay for this strategy and it is often operated at levels much below the maximum throughput in order to have reasonable delays. This is also necessary to maintain system stability, for if the system is operated too close to the optimum level, statistical variation of the traffic can lead to excessive collisions with offered traffic and delays approaching infinity.

***Carrier Sense Multiple Access (CSMA).*** In terrestrial systems, the propagation delay is often so short that one cannot listen to one's own transmission. However, short propagation delay does allow one to listen to determine if the channel is occupied before transmitting, which gives rise to carrier sense multiple access (CSMA). With this protocol, if the channel is in use, the terminal postpones its transmission until the channel is sensed to be idle. If, through lack of positive acknowledgment, the mobile determines its transmission was unsuccessful, then it reschedules the retransmission according to a randomly distributed transmission delay, and repeats the protocol. If all packets are of the same length with packet transmission time $P$, and the one-way propagation delay $d/2$ is identical for all source-destination pairs, then the throughput is given by (8)

$$\eta = \frac{Ge^{-aG}}{G(1+2a)+e^{-aG}} \tag{15}$$

where $a = d/(2P)$. There are many variations on this basic approach. A common approach is $p$-persistent CSMA where a mobile transmits only with probability $p$, if it has a packet ready and detects the channel as idle. These latter approaches offer the possibility of greater throughputs with lower delays (8). Both persistent and nonpersistent approaches can be used with slotted or unslotted formats.

***Packet Reservation Multiple Access (PRMA).*** PRMA is a packet protocol for the return link of a centralized network based on the reservation ALOHA protocol. Reservation Aloha is a slotted ALOHA system with both reserved and unreserved slots, together with a reservation system that assigns slots on a dynamic basis. PRMA adds the cyclical frame structure of TDMA to reservation ALOHA in a manner that allows each TDMA slot to carry either voice or data, where voice is given priority. With PRMA, time is divided into frames of a fixed length, and each frame is divided into a number of slots. The frame length typically equals the frame period of the speech encoding algorithm being used in each terminal. The slot sizes are designed to handle one speech frame (voice packet) at the given transmission rate. It is assumed that the base station can organize the forward traffic on a contention free basis. It is assumed that the network is small physically so that propagation delays are very small, and it is possible to acknowledge a burst in the same time slot that it is transmitted, or at least within one time slot. All terminals are assumed to use voice detection algorithms and to transmit only when voice is present. Each terminal keeps track of those frames that are reserved. The first packet of a voice spurt is transmitted in any unreserved slot, much like the slotted ALOHA protocol. If the voice packet transmission is successful, it results in that slot being reserved for that user in all future frames. A reservation is canceled by not transmitting during a reserved slot. If the voice packet transmission is unsuccessful, retransmission is tried in the next unreserved slot with probability $q$. Voice packets are only kept for up to one frame period, at which point they are discarded. Nonperiodic data packets are integrated into the system by simply using the slotted ALOHA protocol. If successful, they do not result in a periodic reservation, and if unsuccessful, the data packet is retransmitted with probability $p$ in the next unreserved slot. If the data packets are not delay sensitive, they need not be discarded as they age. Goodman et al. (9) show that one can achieve quite high channel efficiencies that takes advantage of voice activation with acceptable dropped packet rates (voice quality) with PRMA. The one limitation is that PRMA is only applicable to local area systems, since it requires immediate acknowledgments.

**Cellular Networks.** In cellular systems, frequencies are often assigned in a hexagonal pattern as shown in Fig. 7 with users in each cell communicating with a base station at the

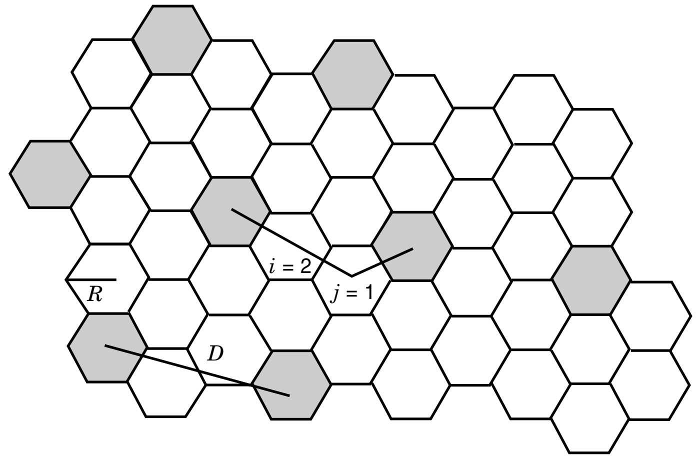


**Figure 7.** The one-in-seven hexagonal pattern with a cell radius $R$ and reuse distance $D$ is one frequency reuse scheme for cellular systems.

centre of the cell. The term cellular is usually applied to terrestrial systems, but similar considerations apply to multibeam satellites. For non-geostationary satellite systems, there is an added difficulty in that the cells move with respect to the earth as the satellite moves.

A hexagonal cell shape is often used because of its close approximation to the circle and its ease of analysis. With FDMA strategies, frequencies are not reused in each cell due to excessive co-channel. With a hexagon geometry, the reuse pattern can be defined relative to a given reference cell (10): *Move i cells along any chain of hexagons, turn counterclockwise 60 degrees; move j cells along the chain that lies on this new heading,* as shown in Fig. 6. The $j$th cell and the reference cell are cochannel cells. With this hexagonal geometry, the cells form natural clusters around the reference cell in the centre and each of its cochannels cells. The number of cells per cluster is given by (10)

$$N = i^2 + ij + j^2 \tag{16}$$

The ratio of $D$, the distance between the centres of nearest neighbouring cochannel cells, to $R$, the cell radius, is the normalized reuse distance, and is given by

$$\frac{D}{R} = \sqrt{3N} \tag{17}$$

From Eq. (16) this allows reuse factors of one-in-three, one-in-four, one-in-seven, one-in-nine, one-in-twelve, and so on. In terrestrial systems, cochannel isolation is determined by propagation losses, and consequently the reuse distance is a function of the propagation loss. For satellite systems, the reuse distance is a function of spotbeam isolation. In the above expression, the reuse distance $D$ is a function of both the cell radius and $N$ the number of cells per cluster.

For terrestrial propagation, the mean propagation loss between a transmitter and a receiver can be approximated by (2)

$$P_r = \frac{P_o}{r^n} \tag{18}$$

where $P_r$ is the received power, $P_o$ is the power at a reference distance, $r$ is the transmitter-receiver separation, and $n$ is a parameter that can range from two to five depending upon the propagation environment. The value of $n = 2$ corresponds to free space loss, while $n = 5$ approximates a dense urban environment. The reference power also depends upon a number of factors such as the height and gain of the transmitting and receiving antennas. For users with similar modulation, Rappaport (11) shows that the mean carrier-to-interference ratio can then be approximated by

$$\frac{C}{I} = \frac{r_d^{-n_d}}{\sum_{k \neq d} r_k^{-n_k}} \tag{19}$$

where $d$ corresponds to the desired user, and $k \neq d$ corresponds to interfering cochannel users. Assuming that the six closest interferers in a cellular system cause almost all of the interference, and that these interferers are at the centre of their cells while the desired user is at the edge of its cell, then one has

$$\frac{R^{-n}}{6D^{-n}} \geq \left(\frac{C}{I}\right)_{\min} \tag{20}$$

where $n$ is the common propagation loss, and $(C/I)_{\min}$ is the minimum tolerable carrier interference ratio for the system. Typical values of the latter are 18 dB for an analog FM signal and 12 dB for a narrowband digital system, but exact values depend on the modulation and coding strategy and quality of service required. It follows from Eqs. (17) and (20) that the frequency reuse factor is lower bounded by

$$N \geq \frac{1}{3}\left[6\left(\frac{C}{I}\right)_{\min}\right]^{2/n} \tag{21}$$

The resulting efficiency of an FDMA cellular strategy is given by

$$\eta = \frac{R_k}{N(B + B_g)} \tag{22}$$

where $R_k$ is the information of the $k$th user, $B$ is the channel bandwidth, $B_g$ is the guard band required between channels to reduce adjacent channel interference to acceptable levels, and $1/N$ is the frequency reuse factor required to reduce cochannel interference to acceptable levels.

**Spread-Spectrum Systems.** ***DSSS Systems.*** In a DSSS system the cochannel interference given by Eq. (8) is treated as equivalent to Gaussian noise of the equivalent power with a flat power spectral density $I_0$. The exception to this rule is when the channels are synchronized or partly synchronized, and are using orthogonal spreading codes. In the latter case, the multiple access noise is zero or close to it, assuming an ideal implementation.

In the simplest case, all users within a cell are assumed to be power controlled such that they are received at similar level at the base station and all cells are equally loaded. Under these conditions, Viterbi (12) shows that the intracell interference—interference from users in the same cell—density can be written as

$$I_i = (K-1)\frac{P_r}{W} \tag{23}$$

where $P_r$ is the received power of each user, and $W$ is the bandwidth over which each user is spread. In a multicell scenario, one must also consider the intercell interference—interference from users in other cells. In terrestrial systems, the intercell interference is determined by the propagation losses; in a satellite system it is determined by the spotbeam rolloff characteristics. Viterbi represents the intercell interference as a factor $f$ times the intracell interference. The total noise that the desired signal must contend with is

$$N_0 + I_0 = N_0 + (1+f)(K-1)\frac{E_b R_k}{W} \tag{24}$$

where $N_0$ is the thermal noise density, $P_r = E_b R_k$, $E_b$ is the received energy per bit, and $R_k$ is the bit rate. The interference constraint faced in this system is that the signal-to-noise ratio at the receiver

$$\frac{E_b}{N_0 + I_0} \leq \left(\frac{E_b}{N_0}\right)_{\min} \tag{25}$$

For this system, the frequency reuse is given by $KR/W$, so approximating $(K - 1)$ by $K$ in Eq. (24) and substituting the result in Eq. (25) one obtains the upper bound

$$\eta = \frac{KR_k}{W} \leq \frac{1}{1+f}\left[\frac{1}{(E_b/N_0)_{\min}} - \frac{1}{E_b/N_0}\right] \tag{26}$$

This efficiency can be increased by considering voice activation and sector reuse. In a typical telephone conversation each user speaks approximately 40% of the time, so a signal that is transmitted only when voice is active reduces interference by a factor, $G_V \approx 2.5$, on average. In CDMA, unlike FDMA, frequency bands can be reused in adjacent antenna sectors. In practice, sectored antennas covering 120° in azimuth are often used in high-traffic areas so the corresponding gain is spectral reuse is $G_A \approx 3$. In some terrestrial situations, the background noise can be negligible relative to the multiple access noise. Considering all these factors, Viterbi (12) shows that Eq. (26) can be approximated by

$$\eta \approx \frac{G_A G_V}{1+f}\frac{1}{(E_b/N_0)_{\min}} \tag{27}$$

For a cellular telephony values for $\eta$ approaching one have been suggested (12).

The spreading codes in DSSS can be implemented as a combination of pseudorandom sequences and forward error correction coding. One of the advantages of CDMA is that forward error correction coding can be introduced to reduce the required $E_b/(N_0 + I_0)$ and thus increase system capacity with no compromises other than an increase in detector complexity. In practice, to minimize transmit power, extend battery power, and minimize health concerns with handheld transmitters, operation is typically at low $E_b/N_0$ as well. As an example, to achieve a bit error rate of $10^{-3}$ that is a typical requirement for vocoded speech, in an additive white Gaussian noise channel with a rate 1/2 constraint length 7 convolutional code requires an $(E_b/N_0)_{\min}$ of approximately 3 dB (1).

A potential advantage of DSSS CDMA is high efficiency in cellular systems. From a modulation viewpoint, it has a nearly constant envelope, and with a RAKE receiver, one can take advantage of frequency diversity. The disadvantages are the power control requirements and that large bandwidths are required for even low-traffic areas. In addition, for higher data rates, the available spectrum is often insufficient to allow significant spreading.

***FHSS Systems.*** The probability of collisions between hops of different users is what determines the performance in a FHSS system. Let $\mu$ represent the probability that there is a collision between any two or more users. For $K$ independently hopping users using $M$-FSK, where $N_t$ is the total number of frequency bins,

$$\mu = \frac{K}{N_t} M \tag{28}$$

By itself, this would imply a relatively low spectral efficiency for reasonable performance. However, frequency-hopping systems usually include FEC encoding. Reed–Solomon encoding (13) is a common choice because it can be used to map $M$-FSK symbols directly into code symbols, and because it can very effectively use side information about whether a hop was jammed or not. A common frequency-hopping detector is an FFT followed by an energy detector, which will often indicate the presence of more than one tone when there is a collision on a hop. If all collided hops are known, a rate 1/2 Reed–Solomon code could effectively reduce a symbol error approaching 50% to negligible levels. In particular, the probability of a code word error with a $(n, k)$ Reed–Solomon code for $M$-ary symbols, and knowing which symbols are jammed, is given by (13)

$$P_e = \sum_{j=n-k+1}^{n} \binom{n}{j} \mu^j (1-\mu)^{n-j} \tag{29}$$

If the jammed symbols are not known then the lower bound on the summation changes to $(n = k)/2 + 1$. Frequency-hopping systems have the advantage that they do not require power control. The disadvantage is that even with FEC encoding, the spectral efficiency is low.

### Other Multiple Access Techniques

***Spatial Division Multiple Access.*** The increasing demand for capacity in wireless systems traditionally translates into a demand for bandwidth. However, limited bandwidth has led to the consideration of other ways of increasing spectral efficiency, such as more efficient use of the spatial dimension by employing antenna arrays at the base stations. The primary benefit is a reduction of the multiple access interference. Let $P_r(\mathbf{x})$ be the received power with an omnidirectional antenna of an interferer at position $\mathbf{x}$, and let $G(\phi)$ be the gain of the directional station antenna as a function of azimuth angle $\phi$. Then the average MAI with a directional antenna is given by

$$\text{MAI} = \int_R G(\phi) P_r(\mathbf{x}) p(\mathbf{x})\, d\mathbf{x} \tag{30}$$

where $R$ is the service area, and $p(\mathbf{x})$ is the probability distribution of interferers over the service area. With 120° sectors, out-of-sector users are significantly attenuated, but generally not enough to allow frequency reuse in an FDMA system. This improves performance because of reduced interference and provides better coverage because of higher gain antennas, but there is no effective capacity gain. With greater sectorization, one can achieve some frequency reuse with narrowband modulations. With CDMA, one can achieve significant spectral reuse even with three sectors.

## EXAMPLE MULTIPLE ACCESS SYSTEMS

There are a number of existing systems that illustrate the principles defined in the previous section.

### A TDMA Cellular Telephony System

A widespread TDMA standard that is used for cellular telephony is the Global System for Mobile communications (GSM) (14). The channel transmission format consists of a superframe that is divided into frames, which are subdivided into slots that are assigned to users. A superframe of 6.12 s is divided into 1326 TDMA frames of 4.615 ms each. Each frame has eight time slots of 0.577 ms for 148 bits, plus a guard time equivalent to 8.25 bits. The resulting TDMA burst rate is 271 kbps, and there are eight full-rate users per channel. There are 125 duplex channels paired between the 890 to 915 MHz return link band and the 935 MHz to 960 MHz forward link band.

This system uses constant-envelope partial-response Gaussian Minimum Shift-Keyed (GMSK) modulation with a channel spacing of 200 kHz. The controlled GMSK-induced ISI and the uncontrolled channel-induced ISI are removed by a channel equalizer at the receiver. The traffic channels use a $r = \frac{1}{2}$, $k = 5$ convolutional code, but some control channels have greater error protection through the use of block codes.

Since the GSM standard allows frequency hopping, each physical channel corresponds to a sequence of RF channels and time slots. Logical channels, listed in Table 2, are assigned to the physical channels in either a fixed or dynamic manner. The various control channels are defined in Table 2; BCCH, FCH, SCH, and CCH are transmitted on a single RF channel from each base station. These channels allow the terminal to acquire the system [first in frequency (FCH), then in time (SCH)] and then determine the current system configuration (BCCH). The remaining control channel (CCH) is used to notify the user of an incoming call or provide a channel assignment. The traffic channels (TCH) carry the voice/data, and each terminal is assigned one slot (and frequency) in 24

**Table 2. The Logical Channels into Which the Physical Frequency-Hopped TDMA Channels Are Divided**

| | |
|---|---|
| BCCH | broadcast channel |
| FCH | frequency channel |
| SCH | synchronization channel |
| CCH | access grant/paging channels |
| TCH | traffic channel |
| SACCH | slow associated control channel |
| FACCH | fast associated control channel |
| RACH | random access channel |

out of every 26 frames (in the half-rate mode each mobile is assigned one slot in 12 out of every 26 frames), while the remaining two frames are used for SACCH. During a call the base station continually monitors the mobile's timing error and received power levels, and sends any corrections via the SACCH. The FACCH can carry the same information as the SACCH but is only used when there is a need for heavy duty signaling, such as in a cell handover. The FACCH obtains capacity by stealing frames from the TCH when required.

In the return direction, one also has the TCH plus the random-access channel (RACH). The format of the RACH differs in that the slots are 235 ms long in order to accommodate initial timing errors of the users, and uses the slotted ALOHA protocol. Frequency reuse is similar to that described for FDMA system, with frequencies reused in cells of sufficient distance. Typical frequency reuse numbers are 21, 12, and 9 (in sectored systems).

**A CDMA Cellular Telephony Standard.** A widespread CDMA-DSSS standard that is used for cellular telephony applications is IS-95 (15). To aid synchronization to the spreading sequence, each base station emits a unmodulated pilot PN sequence in the forward link to identify itself. The period of the PN sequence is $2^{15}$ chips with a chip rate of 1.22288 MHz. Different offsets of the same PN sequence are used to identify different base stations. In all, 512 possible offsets (52 $\mu$s) are allowed. The terminal first acquires the strongest pilot sequence, which identifies the closest base station. It also allows the terminal to immediately acquire the sync channel that is a synchronized but modified (Walsh spread) version of the pilot sequence. The sync channel provides synchronization information to allow the mobile to listen to the paging channel for that base station and submit an access request (on the return link access channel). This allows call setup and then, knowing the appropriate spreading codes, forward and return traffic channels are initiated. All spreading codes are approximately synchronized to the forward link pilot signal as this reduces the search range in the receiver acquisition process. The pilot signal, sync channel, paging channel, and forward traffic all share the same 1.25 MHz frequency band. Similarly all access request channels and reverse traffic share the same return 1.25 MHz frequency band. The access request channel is an ALOHA channel with capture. That is, the receiver may be able to correctly demodulate two colliding bursts as long as their timing is not identical.

Each base station is allowed to use the same pair of forward and return 1.25 MHz frequency bands. This allows the mobile to initiate handovers between base stations based on the received signal strength of their pilot signals. The return link is in the band from 825 MHz to 850 MHz, and the forward link is in the band from 870 MHz to 895 MHz.

In the forward direction, the primary data rate for this system is 9.6 kbps with submultiples of this rate also implemented by reducing the transmission duty cycle proportionately. The data is rate 1/2 encoded with a constraint length 9 convolutional code to achieve a coded data rate of 19.2 kbps. The coded data is block interleaved to provide time diversity against fast fading. The data is then triply spread. The initial spreading is by a long PN code that has a period of $2^{42} - 1$ chips and has a chip rate of 1.2288 MHz. This long PN code is specific to the user. It is then spread by a Walsh sequence, also with a chip rate of 1.2288 MHz, and with 64 chips per coded bit. There is a final spreading by the PN same code as used for the pilot signal for that base station, which is also at a chip rate of 1.2288 MHz. This final spreading is combined with a quaternary PSK (QPSK) modulator, the output of which is filtered before transmitting. The nominal transmit bandwidth is 1.25 MHz.

In the return direction, the data rates are similar, but encoding and modulation differ. The data is rate 1/3 encoded with a constraint length 9 convolutional code to achieve a coded data rate of 28.8 kbps. The data is block interleaved to provide time diversity against fast fading. Then, six code symbols are modulated as one of 64 modulation symbols (Walsh functions) with an orthogonal modulator (1). This results in a Walsh chip rate of 307.2 kHz. The resulting signal is then spread by the long PN code for that user, each Walsh chip being spread by four PN chips. The signal is further spread into both the inphase and quadrature channels by the pilot sequence corresponding to the forward link. The resulting signal is offset-QPSK modulated and filtered.

It is the multiple access noise from the mobile's cell and the surrounding cells that limits the capacity of this system. To minimize the multiple access noise, all mobile units include both open-loop and closed-loop power control. This has the secondary benefit of extending battery life for handheld terminals. The system is designed such that ideally about 60% to 65% of the multiple access interference is intracell, approximately 36% comes from the surrounding six cells, and less than 4% comes from more remote cells (7).

### An OFDM Digital Audio Broadcasting System

One digital audio broadcasting standard (16) for mobile, portable, and fixed receivers uses OFDM for broadcasting data and music, and so on, from a number of sources. There are four possible modes of operation in system. All four modes provide a channel rate of 3.072 megabits per second, but are implemented with different numbers of carriers and symbol rates. This allows the system to tradeoff robustness and coverage versus complexity. The nominal channel bandwidth in all three cases is 1.536 MHz.

The Mode 1 transmission frame consists of 76 consecutive OFDM symbols, excluding the null period. The length of a mode 1 frame is $T_f = 96$ ms. Each OFDM symbol consists of a set of 1536 equally spaced carriers, with carrier spacing equal to $1/T_u$, and where $T_u = 1$ ms. The data is convolutional encoded with puncturing to allow unequal error protection. The mother code is a rate 1/4 constraint length 7 convolutional code with interleaving to provide time diversity. The data is differentially encoded QPSK modulated before being placed on a carrier with rectangular pulse shaping. Each Mode 1 frame consists of:

1. A synchronization block consisting of the first two OFDM symbols, the null symbol, and the phase reference symbol. The null symbol is a transmission-off period of $T_{\text{null}} = 1.297$ ms followed by a phase reference symbol that constitutes a reference for differential modulation of the next OFDM symbol. The phase reference symbol and all subsequent symbols are of length $T_s = T_u + \Delta = 1.246$ ms. A null period of length $\Delta = 0.246$ ms is inserted at the end of each symbol to reduce ISI resulting from delay spread of the channel. As a result

the receiver can be implemented without the need for an equalizer.

2. A fast information channel (FIC) made of four blocks of 256 bits carrying the information necessary to interpret the configuration of the main service channel (MSC). This information is rate 1/3 code, split into three blocks of 3072 channel bits, and transmitted on the first three OFDM symbols following the synch block.
3. An MSC is made up of a sequence of four common interleaved frames (CIF). Each CIF consists of a data field of 55296 bits including coding, which may be subdivided to form subchannels. The minimum subchannel throughput is 8 kbps and can be allocated to handle either packet or stream data. The four CIFs are divided into 72 blocks of 3072 bits and transmitted on the last 72 OFDM symbols in a frame.

## INTERFERENCE CANCELLATION

The theoretical maximum capacity of a multiple access channel in additive white Gaussian noise is derived in (17). Achieving the capacity requires forward error correction coding and a method for extracting the interference caused by cochannel users from the desired signal. The latter is a serious problem. In mobile applications, the problem is even more challenging because of nonuniform propagation characteristics and cellular reuse strategies. The conventional receiver treats interference as equivalent to thermal noise. The approach of more advanced receivers is to recognize that there is some coherence to the interference and to attempt to process it in a manner that reduces its effect.

These interference reduction/cancellation techniques form the leading edge of research in multiple access systems. In the following, we will briefly discuss a number of these techniques. We classify these techniques into three broad categories: (1) minimization techniques, (2) compensation techniques, and (3) multiuser detection techniques.

### Minimization Techniques

Minimization techniques are those that attempt to minimize the interference before it gets into the receiver. They tend to apply more to narrowband than to wideband modulations.

**Intelligent Antennas.** The simplest approach used in both terrestrial and mobile satellite systems is to use directional antennas at base station or satellite. This can reduce both intracell and intercell interference seen at a receiver. The simplest approach to this is to use sectored antennas as was discussed earlier under SDMA. This approach is passive in the sense that the antenna configuration is independent of the mobile terminal. A more advanced approach to this is using so-called intelligent antennas. Intelligent antennas take an active receiver role. With a phased array antenna, one can electronically optimize the antenna for each user, maximizing the gain in the direction of the desired signal and possibly positioning an antenna null in the direction of the strongest interferer (18). Furthermore, in a multipath environment research is continuing into designing the antenna beam to coherently capture as much multipath energy as possible. The latter is much easier to do in the return link than in the forward.

A second antenna-related approach that has been suggested for mobile satellite applications is to use signals with orthogonal polarizations. This is possible in satellite applications because there is a strong direct path, and it is suggested that this could possibly double capacity. However, there are some unanswered questions in this area, as reflected paths often reverse their polarization, which may cause problems for mobile applications.

**Dynamic Channel Assignment.** Dynamic channel assignment is an interference avoidance technique. With this approach a base station continually monitors and/or probes channels during a call to determine the local interference characteristics (19). If the interference reaches a level that is degrading to the call, the call is automatic switched to a new frequency channel that has better characteristics. This approach not only provides protection against time-varying interference, but it also provides a form of switched frequency diversity that will provide some alleviation against multipath. For research in this area, see Ref. 19 and the references therein.

**New Multiple Access Techniques.** A considerable amount of research is about future services to be provided to mobile terminals. The general consensus appears to be that the future terminal will be a form of multimedia terminal. Although the multimedia services that will be demanded is a subject of debate, it appears clear that significantly higher data rates than used at present will be required. Thus the problem is providing reliable high data rate service, such as 64 kbps to 2 Mbps, over a mobile fading channel. Considerable research (20) has been done on multicarrier (MC) techniques such as MC-OFDM and MC-CDMA because of their potential to provide higher data rates in limited bandwidths and yet, because of the problems of the multicarrier approach, provide some frequency diversity that allows protection against multipath. It also allows implementation without the need for a high-speed equalizer capable of tracking channel variations.

### Compensation Techniques

Compensation techniques attempt to minimize the effect of the interference once it gets into the receiver. These techniques rely only on minimal a priori knowledge of the interfering signal. They tend to apply to wideband desired signals, although the interference can be narrow or wideband.

**Narrowband Interference.** With wideband modulation schemes such as DSSS there is the opportunity to excise narrowband interferers. The motivation of these approaches often comes from military applications where the desired signal is a DSSS signal and it is intentionally being jammed by a CW signal. However, the problem can also occur in commercial environments where a wideband system is being operated close to a narrowband system, or in some cases may be overlaid on a narrowband system. The object of these approaches is to reduce the spectral density of the interference to the level of the desired signal. The simplest technique is a notch filter that simply filters out the interferer. The degradation to the desired signal caused by the notch filter is approximately proportional to the fraction of the bandwidth removed. More advanced approaches recognize that interference-rejection techniques need to be adaptive because of the dynamic nature of interference and the channel. Hence, adaptive filters or equalizers based on the LMS algorithm or Kalman filtering techniques are often considered. Nonlinear filters have also been shown to be advantageous for impulsive noise. For an extensive survey of these techniques see Laster and Reed (21) and the references therein.

**Wideband Interference.** When one or more DSSS signals interfere with the desired signal, the interference can be viewed as colored noise, and equalization techniques can be considered. The techniques range from symbol-spaced to fractionally spaced to chip-rate based equalizers. An example of these techniques is the minimum mean square error (MMSE) multiuser detector (22), which produces a linear filter $c(t)$, sampled at the chip rate, that minimizes the MSE between the received signal and desired signal. These techniques can be formulated with knowledge of the other user parameters but, in practice, they can be implemented without this knowledge using simple gradient search techniques. The simplest practical approach requires a training sequence to perform the initial adaptation before it proceeds to a decision-directed mode. See Ref. 21 and the references therein.

**Multiuser Detection.** Multiuser detection techniques are equivalent to detecting all the users and subtracting their effect from the desired user. Hence, these techniques require as much knowledge about the interference as they do about the desired signal. These techniques originally addressed DSSS-CDMA systems (23), but they have been extended to FHSS systems and narrowband systems.

**Optimum.** In Eq. (6) we described the optimum single-user detector as a bank of correlators with a correlator for every possible transmitted sequence $b_k$. In a completely analogous fashion, if we let $S(\mathbf{b}, t)$ represent the corresponding $K$-user signal, then the optimum multiuser detector is a bank of correlators with a correlator for every possible transmitted $K$-user sequence $\mathbf{b}$. For binary transmission, this means $2^{NK}$ correlators (of length $N$ symbols), which is clearly beyond reason. Verdu (24) showed that for asynchronous DSSS systems this complexity could be reduced to a complexity of $\mathcal{O}(2^K)$ per symbol using a variation of the Viterbi algorithm. In most CDMA applications, this is still not practical and has motivated the search for simpler but possibly suboptimal multiuser detectors.

**Decorrelator.** Lupas and Verdu (25) showed that, with a DSSS system with rectangular pulses has the discrete time equivalent model

$$\begin{aligned}\boldsymbol{y}(i) =& H(+1)W^{1/2}\boldsymbol{b}(i-1) + H(0)W^{1/2}\boldsymbol{b}(i)\\ &+ H(-1)W^{1/2}\boldsymbol{b}(i+1) + \boldsymbol{n}(i)\end{aligned} \tag{31}$$

where $\boldsymbol{y}(i)$ is the $K$-vector of outputs from a bank of single user matched filters, optimally sampled at the symbol rate. The matrices $H(i)$ are the crosscorrelation between the modulating waveforms

$$[H(i)]_{kl} = \int_{-\infty}^{\infty} s_k(t - iT - \tau_k)s_l(t-\tau_l)\,dt \tag{32}$$

the vector $\boldsymbol{n}(i)$ is a set of zero-mean correlated noise samples with $\mathrm{E}[\boldsymbol{n}(i)\boldsymbol{n}(j)^T] = \sigma^2 H(j)\delta(i-j)$, and $W$ is the diagonal matrix with non-zero elements $\{w_k\}$, the propagation losses. With synchronous users $H(1)$ and $H(-1)$ are zero and the discrete time model reduces to

$$\boldsymbol{y}(i) = W^{1/2}H(0)\boldsymbol{b}(i) + \boldsymbol{n}(i) \tag{33}$$

and the decorrelating detector estimates the data as

$$\hat{\boldsymbol{b}}(i) = sign\{H(0)^{-1}\boldsymbol{y}(i)\} \tag{34}$$

In the asynchronous case, the asymptotic decorrelator is the discrete time matrix filter (25)

$$D(z) = [H(-1)z^{-1} + H(0) + H(1)z]^{-1} \tag{35}$$

that is applied to the vector sequence $[\boldsymbol{y}(i)]$ before detection. The decorrelating approach works well if the inverse operation is well-conditioned, otherwise it can result in noise enhancement that can significantly degrade the performance of some users.

**Multistage Detector for a Direct Sequence System.** Varanasi et al. (26) describe a multistage detector for asynchronous CDMA that uses a conventional detector for the first stage but in the $n$th stage use the decisions of the $(n-1)$st stage to cancel MAI present in the received signal. The performance of this feedback approach depends on the relative powers of the users. A more reliable multistage detector is obtained if the conventional detector in the first stage is replaced by a decorrelator.

**Multistage Detection for Frequency-Hopped Systems.** There are various ways to address frequencies in a frequency-hopped system. One method is to perform a one-to-one mapping of the frequencies to the elements of a Galois field (13). A hopping sequence corresponds to a sequence of frequencies (addresses) that are represented as elements of a Galois field. Similarly, the data can be represented as elements of the same Galois field. The actual transmit frequency corresponds to the Galois-field sum of the two quantities. For this scheme, a simple iterative interference cancellation has been proposed by Fiebig (27). Effectively, the scheme assumes diversity transmission, with the same symbol transmitted on each of $L$ hops, often referred to as fast frequency hopping. In Ref. 27, $L$ is the log of the alphabet size, but this does not appear to be a requirement. For a particular symbol at the receiver, one despreads (dehops) the received signal for each of users, respectively. One then searches for any unambiguous decisions for individual users. Those unambiguous decisions are "erased" from the dehopped representations of other users, and further unambiguous decisions are searched for. This is continued until no further progress is made. The majority of the gain appears to be made with three iterations. This approach provides large improvements in the capacity over a system that uses a conventional detector and does so with minimal processing and no coding.

**Decision Feedback Equalizers (DFE).** With a DSSS-CDMA system, the MAI can be modeled as equivalent to $K$-dimensional ISI. Consequently, equalizer approaches used for single user channels (1) can be extended to multiuser channels. In Ref. 28 Duel-Hallen et al. describe a multiuser decision feedback equalizer (DFE), characterized by two matrix transformations: a feedforward filter and a feedback filter. In addition to equalization, these multiuser decision feedback detectors employ successive cancellation using decisions made in order of decreasing user strength. The performance of the DFE is similar to the decorrelator for the strongest user, and gradually approaches the single-user bound as a user's power decreases relative to the power of other users.

**Multiuser Decoding.** A limitation of many of the previously mentioned multiuser detectors is that their performance degrades at low SNR, or if there is high correlation between the users. This implies a degradation in performance if the data is FEC encoded, which is often the case in mobile channels. Encoded systems typically operate at lower SNRs, and consequently initial decisions on channel bits tend to be unreliable and insufficient to bootstrap an iterative scheme. It also implies a degradation if these techniques are used for narrowband modulations. The same behaviour occurs if users are highly correlated, that is, they are not spread as in a CDMA system, but are more narrowband as in an FDMA system. Theory still claims detection should be possible (17), but it is only recently that there have been clues about how to do it practically.

In (29) an iterative technique is proposed for multiuser decoding, where each user employs FSK modulation. This can be viewed in some ways as an extension of Fiebig's work (27) for frequency-hopped systems to the case where FEC encoding is employed with a soft decoding algorithm. In (30) another approach based on iterative decoding is applied to users whose correlations approach 1. This can be applied to users using PSK modulation with very little isolation between frequency reuses, and has the potential of closely approaching the theoretical capacity of the multiple access channel.

## BIBLIOGRAPHY


1. J. G. Proakis, *Digital Communications,* 2nd ed., New York: McGraw-Hill, 1989.
2. D. Parsons, *The Mobile Radio Propagation Channel,* New York: Wiley, 1992.
3. M. K. Simon et al., *Spread Spectrum Communications,* Rockville, MD: Computer Science Press, 1985, vols. 1, 2, 3.
4. A. D. Whalen, *Detection of Signals in Noise,* New York: Academic Press, 1971.
5. S. B. Weinstein, Data transmission by frequency-division multiplexing using the discrete Fourier transform, *IEEE Trans. Commun. Technol.,* **19**: 628–634, 1971.
6. A. V. Oppenheim and R. W. Schafer, *Digital Signal Processing,* Englewood Cliffs, NJ: Prentice-Hall, 1975.
7. J. D. Gibson (ed.), *Handbook on Communications,* Boca Raton, FL: CRC Press, 1997.
8. L. Kleinrock, *Queueing Systems,* New York: Wiley, 1976, vols. 1, 2.
9. D. J. Goodman et al., Packet reservation multiple access for local wireless communications, *IEEE Trans. Commun.,* **37**: 885–890, 1989.
10. V. H. MacDonald, The Cellular Concept, *BSTJ,* **58** (1): 15–41, 1979.
11. T. S. Rappaport, *Wireless Communications,* Englewood Cliffs, NJ: Prentice-Hall, 1996.
12. A. J. Viterbi, *CDMA—Principles of Spread Spectrum Communications,* Reading, MA: Addison-Wesley, 1995.
13. R. E. Blahut, *Theory and Practice of Error-Control Codes,* Reading, MA: Addison-Wesley, 1983.
14. S M. Redl, M. K. Weber, and M. W. Oliphant, *An Introduction to GSM,* Boston: Artech House, 1995.
15. TIA/EIA Interim Standard, Mobile station—base station compatibility standard for dual-mode wideband spread spectrum cellular system, *TIA/EIA/IS-95-A,* May 1995.
16. European Telecommunication Standard, Radio broadcast systems; Digital audio broadcasting (DAB) to mobile, portable and fixed receivers, *ETS 300 401,* June 1996.
17. T. M. Cover and J. A. Thomas, *Elements of Information Theory,* New York: Wiley, 1991.
18. B. Ottersten, Array processing for wireless communications, *Proc. 8th IEEE Workshop Stat. Sig. Array Process.,* Corfu, Greece, 1996, pp. 466–473.
19. I. Katzela and M. Naghshineh, Channel assignment schemes for cellular mobile telecommunication systems: a comprehensive survey, *IEEE Personal Commun.,* **2** (6): 10–31, 1996.
20. K. Fazel and G. P. Fettweis (eds.), *Multi-carrier Spread-spectrum,* Dordrecht, The Netherlands: Kluwer Academic, 1997.
21. J. D. Laster and J. H. Reed, Interference rejection in digital wireless communications, *IEEE Sig. Process.,* **14** (3): 37–62, 1997.
22. U. Madhow and M. Honig, MMSE interference suppression for direct-sequence spread spectrum CDMA, *IEEE Trans. Commun.,* **42**: 3178–3188, 1994.
23. A. Duel-Hallen, J. Holtzman, and Z. Zvonar, Multiuser detection for CDMA systems, *IEEE Personal Commun.,* **1** (4): 46–58, 1995.
24. S. Verdú, Minimum probability of error for asynchronous Gaussian multiple-access channels, *IEEE Trans. Inf. Theory,* **32**: 85–96, 1986.
25. R. Lupas and S. Verdu, Near-far resistance of multiuser detectors in asynchronous channels, *IEEE Trans. Commun.,* **38**: 496–508, 1990.
26. M. K. Varanasi and B. Aazhang, Multistage detection in asynchronous code division multiple-access communications, *IEEE Trans. Commun.,* **38**: 509–519, 1990.
27. U.-C. G. Fiebig, Iterative interference cancellation for FFH/MFSK MA systems, *IEE Proc. Commun.,* **143** (6): 380–388, 1996.
28. A. Duel-Hallen, A family of multiuser decision-feedback detectors for asynchronous code-division multiple-access channels, *IEEE Trans. Commun.,* **43**: 421–434, 1995.
29. A. J. Grant and C. Schlegel, Collision-type multiple-user communications, *IEEE Trans. Inf. Theory,* **43**: 1725–1736, 1997.
30. M. Moher, An iterative multiuser decoder for near-capacity communications, *IEEE Trans. Commun.,* **46** (7): 1998.


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