DIGITAL RADIO

This article presents an overview of digital radio communications with emphasis on wireless personal communications. Digital radio consists of two main processes: speech coding (and video coding for future systems) and modulation. In speech coding, the continuous analog voice signal is converted into a discrete digital form. In communication systems that provide digital voice services, it is necessary to encode the analog speech into a digital stream for transmission over the channel and at the receiver, to reconstruct the signal with acceptable fidelity. Modulation, on the other hand, is the process of impressing the discrete digital signal onto a radio signal, by varying some parameter or parameters in combination (usually the amplitude, frequency, or phase) of the radio signal. Digital transmission systems exist in a wide variety of forms, mainly determined by the nature of the channel over which the system operates. A process which is common to many digital transmission systems is regeneration. Typically, the received signal at the input of a repeater (receiver) arrives attenuated and dispersed by the channel and corrupted by noise. The first operation of the receiving repeater, therefore, is to preamplify and shape the weakened signal to a level and form from which a reliable threshold detection may be performed. In a wireless channel, the transmitted signal also undergoes multipath fading and shadowing due to obstructions in the transmission path. To combat the effect of multipath fading, the receiver may employ diversity reception, channel coding, and equalization.

The main objective of wireless personal communication is to allow the user access to the capabilities of the global network at any time and without regards to location and mobility. There has been a tremendous growth in the number of subscribers to wireless services in the last decade. As a result, it became very difficult to operate and maintain the quality of the original first-generation analog cellular systems. As the number of subscribers increases in these systems, call quality diminishes. To handle the increasing traffic, the cellular concept in which identical frequencies at non-interfering distances are reused, was adopted giving rise to more interference in the system. Digital radio, used extensively in second-generation and the evolving third-generation systems, considerably improves the quality of the cellular system and enhances the services available to the mobile subscribers. Some advantages of digital cellular radio include (1,2):

1. More efficient use of the limited radio-frequency (rf) spectrum to improve system capacity.
2. Improving the voice quality beyond what is possible with analog cellular systems, especially maintaining voice quality in heavy traffic conditions and use of voice activity detection to save power and increase throughput.
3. Providing support for a wider array of services and features.
4. Simplifying the task of frequency planning, operating, and maintaining the cellular system.
5. Providing a smooth transition from the analog systems to digital radio systems.

Cordless and cellular telephony have gained widespread user acceptance. Cordless telephones are low-power, low-range phones that enable the user to move around the home or office and still place and receive phone calls. The handsets typically operate within 100 m of the user's base station which is connected to the public switched telephone network (PSTN). Cordless telephony has evolved from being a simple home appliance to sophisticated systems in applications for universal low-power cordless and telepoint systems aimed at pedestrians, and cordless private branch exchange (PBX). Digital cordless telephone systems (such as CT2, DECT, PHS) are optimized for low-complexity equipment and high-quality speech in a quasi-static (with respect to user mobility) environment. They can support higher data rates and more sophisticated applications.

On the other hand, digital cellular radio, originally targeted at vehicular users in urban areas, was developed to maximize bandwidth efficiency and frequency reuse in a macro-cellular, high-speed fading environment. The first-generation cellular systems are analog systems based mostly on frequency division multiple access (FDMA) and have very limited capacity and poor to average speech quality. The second-generation cellular systems (e.g., IS-54, GSM, IS-95) are all digital systems and use more efficient multiple access techniques to share the available spectrum among the users. Although personal communications services (PCS) may be regarded as a third-generation system, its implementation uses modified versions of the cellular protocols used in the second-generation systems. While the first-generation analog and second-generation digital systems are designed to support voice communication with limited data communication capabilities, third-generation systems will focus on providing a wide variety of services which include wireless extensions of integrated services digital network (ISDN) and broadband asynchronous transfer mode (ATM). These systems will concentrate on service quality, system capacity, and terminal and personal mobility issues. They will use a variety of cell structures ranging from the conventional macrocells to microcells for urban areas, picocells for indoor applications, and supercells for satellite-based systems.

DIGITAL CELLULAR RADIO

The Cellular Concept

The continuous increase in the demand for telecommunications services and systems has resulted in spectral congestion. Thus the original one cell system with a high-power transmitter to provide good coverage in a wide service area has quickly become limited in capacity. The cellular concept in which the geographic service area is divided into small regions (called cells), each of which is served by a low-power transmitter (base station), has provided a solution to the spectral congestion problem. Adjacent cells are assigned different frequency channels so that the frequencies can be reused throughout the coverage area, leading to a considerable improvement in system capacity. Figure 1 shows a seven-cell frequency reuse pattern. Although frequency reuse increases system capacity, it also increases the amount of adjacent channel and cochannel interference present in the system. Therefore, for the efficient utilization of the radio spectrum, the frequency allocation scheme must be optimized to increase capacity and minimize interference. As the traffic grows in the coverage area, new cells and channels can be added to the system. The hexagonal cell structure is usually employed in the design of a cellular system; however, in practice the actual cell coverage area (footprint) is irregular and depends on the terrain and multipath characteristics of the radio channel (3). As such, there may be some regions within the coverage region where there is exceptionally high likelihood of deep signal fades (called *blind spots*) due to shadowing, tunnels, and other obstructions in the signal path. Blind spots can be overcome by using repeaters which receive the

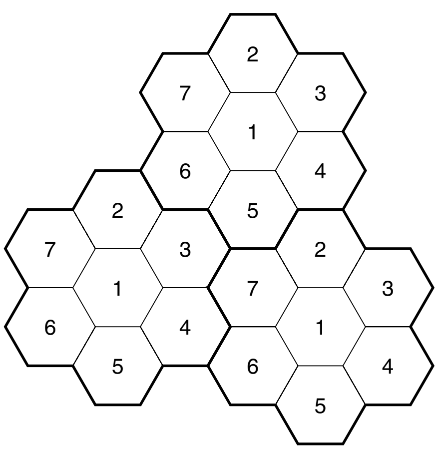

Figure 1. Frequency reuse.

signals selectively in one direction, amplify them, and then retransmit exact replicas of the signals in the required direction. Three types of repeaters may be identified, namely: broadband repeaters, frequency band selective repeaters, and channel selective repeaters (4). Signal degradation at the cell boundaries is handled by handoff operation.

Handoff

Handoff encompasses a set of functions that are supported between a mobile user and the cellular network that allows the user to move from one cell to another or one radio channel to another, within or between cells, while a call is in progress. When a mobile user is engaged in a call, it will frequently move out of the coverage area of the base station with which it is communicating, and unless the call is passed to another base station, the call will be lost. Thus, the system continuously monitors the quality of the signals received from the active mobile users. When the signal falls below a preset threshold the system checks whether another base station can receive the mobile user at a better signal level, and if so, the mobile user is commanded by a control signal to switch to the new frequency (corresponding to the new base station). Although the process of measuring the signal quality, channel allocation, and handoff may take a few seconds, there should not be any noticeable break in conversation of the mobile user. Effective and reliable handoff is essential in controlling co-channel interference, especially for microcellular systems.

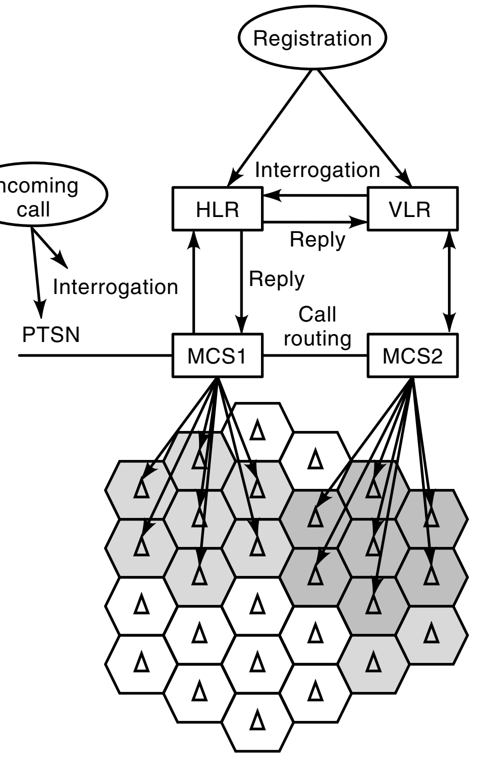

Figure 2. The cellular network.

The Cellular Network

The structure of all cellular networks is essentially similar. Being complete telephone networks, they have dedicated exchanges within the interconnected system with base stations connected to the exchanges. In practice, however, there are different ways of planning the network depending on the capacity requirement, implementation cost, and capabilities of the chosen manufacturer's equipment. As shown in Fig. 2, a typical cellular system consists of mobile stations, base stations, and a mobile switching center (MSC). Each mobile unit contains a transceiver, antenna, and control circuitry and may be hand-held or mounted in a vehicle. The communication between the mobile station and the base station is defined by a standard *air interface.* Each cell in the coverage area is served with one or more base stations which are connected to the MSC. The MSC, in turn, coordinates the activities of the base stations and connects them via microwave or fiber links to the public switched telephone network (PSTN) which forms the global telecommunications network that connects wireline telephone switching centers (called central offices) to MSCs throughout the world.

Multiple Access Schemes

Multiple access techniques are utilized to accommodate multiple users in the available channels. The multiple access scheme controls the allocation of the channel capacity to the users. The allocation scheme is chosen to maximize the spectral efficiency and minimize transmission delay in the system. Other desirable properties of the multiple access scheme include fairness of the allocation process, stability of the system, robustness with respect to equipment failure and changing conditions of other users in the system, and flexibility in allowing the integration of voice and data traffic. In addition, in a wireless mobile communications environment, the hidden terminal problem and near–far effect, the effects of multipath fading and shadowing as well as the effects of co-channel and adjacent channel interference must also be considered. The commonly used multiple access schemes in cellular systems are frequency division multiple access (FDMA), time division multiple access (TDMA), and code division multiple access (CDMA). In FDMA, the available bandwidth is divided into channels which are assigned to the users on demand and each user uses the channel for the entire duration of the transmission. Frequency guard bands are provided at the edges of the band to minimize cross-talk. Although FDMA has relatively low complexity, requires few overhead bits (for synchronization and framing), and usually does not require equalization since the effect of intersymbol interference (ISI) is minimal, it is wasteful because only a fixed number of users (channels) can be supported and when a channel is not being used it remains idle. Furthermore, FDMA requires very tight filtering to minimize adjacent channel interference and intermodulation. In TDMA systems, the usage of each frequency channel is partitioned into multiple time slots, and each user is assigned a specific frequency-time slot combination. Thus, in a given cell only a single mobile user uses the entire frequency at any given time. Although TDMA has the disadvantages of requiring synchronization (as well as overhead for guard time slots) and equalization and can also be wasteful, it permits the use of flexible bit rates and may be used for bursty transmission to save power. Another major advantage of TDMA (over FDMA) arises from the fact that by transmitting and receiving in different time slots it may be possible to eliminate the duplexer circuitry in the mobile unit, replacing

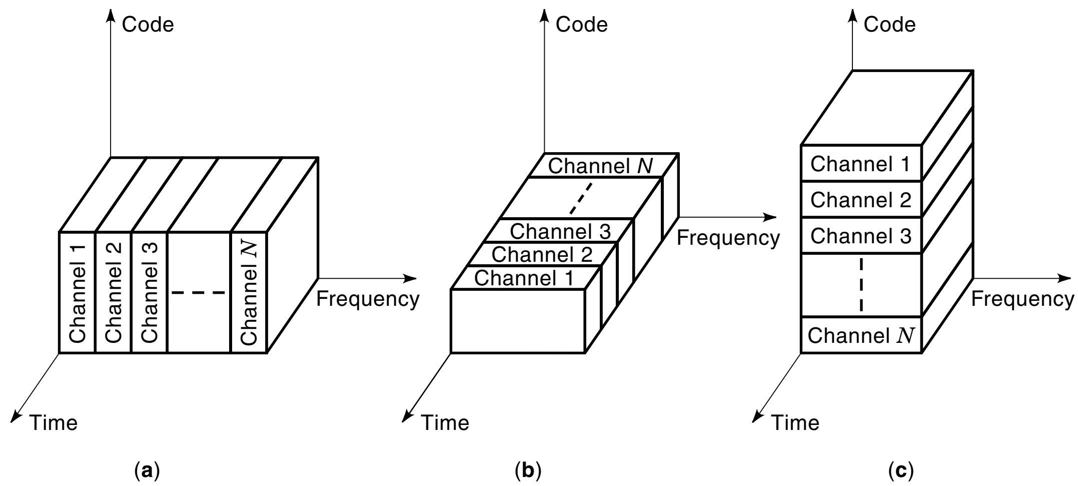

Figure 3. Multiple access techniques: (a) FDMA; (b) TDMA; (c) CDMA.

it with a fast-switching circuit which turns the transmitter and the receiver on and off at the appropriate times, thus prolonging the battery life of the handsets. Also, TDMA-based systems tend to be more flexible and more open to technological change. Thus, with improvements in speech coding algorithms, a TDMA channel is more easily reconfigurable to accept new techniques supporting higher, lower, or variable bit rates, without disrupting the frequency plan of the cellular network. With CDMA (which uses direct sequence spreading), a frequency channel is shared simultaneously by multiple users in a given cell, and the signals are distinguished by spreading them with different codes. CDMA has the advantage of offering multipath immunity and interference rejection and provides a graceful performance degradation as the number of users in the system increases. However, CDMA is susceptible to the near–far problem and requires power control (3). Figure 3 illustrates the three commonly used multiple access schemes in cellular networks.

In general, digital systems can support more users per base station per megahertz of spectrum, allowing wireless system operators to provide service in high-density areas more economically. The use of TDMA or CDMA digital architecture provides the additional advantage of sharing the radio hardware in the base station among the multiple users. It offers flexibility for mixing voice/data communication and the support of new services. A potential for further capacity increases is also possible with the use of reduced rate speech coders. Furthermore, reduced RF transmit power (increasing battery life of handsets) and the use of encryption for communication privacy, are possible. It offers a more natural integration with the evolving digital wireline network and reduced system complexity (mobile-assisted handoffs, fewer radio transceivers, etc.). While the second-generation cellular systems are based on digital transmission, some of them are designed to co-exist with their analog counterparts, while all the evolving third-generation cellular and PCS systems use digital transmission.

In wireless communications systems, it is usually desirable for the subscriber to simultaneously send information to the base station while receiving information from the base station. The process whereby the subscriber can transmit and receive information simultaneously is known as *duplexing*. The two commonly used duplexing techniques are frequency-division duplexing (FDD) and time-division duplexing (TDD). In FDD, the forward link (base-to-mobile station) and the reverse link (mobile-to-base station) transmissions are done simultaneously on different frequency channels. In this case, a device called a duplexer is used inside each subscriber unit and base station to allow the simultaneous signal transmission and reception on the duplex channel pair. In TDD systems, the same frequency band is used in both the forward and the reverse links but it is required that the transmissions in different directions occur in different time slots.

A performance measure that is commonly used to characterize a digital radio system is the spectral efficiency. The spectral efficiency of a digital radio system E_s is defined as (5)

$$E_s = \frac{\eta_c}{WCA_c} \quad (\text{erlangs/MHz/km}^2) \tag{1}$$

where η_c is the carried traffic per channel in erlang, W is the channel bandwidth in MHz, A_c is the cell area in km^2, and C is the cluster size (number of cells in a reuse cluster).

Speech Coding

In wireless systems that provide digital voice services, there is the need to encode the analog speech signal into a digital stream for transmission over the channel (air interface). At the receiver, the signal is reconstructed with acceptable fidelity. There are several major parameters to consider in choosing a speech coding scheme for wireless application. These include the transmitted bit rate (kb/s), the delivered speech quality, robustness to transmission errors, and complexity of implementation of the chosen scheme. Available speech coding techniques may be classified into three main categories, namely: waveform coding, model-based coding, and hybrid techniques. Waveform coding techniques are usually the simplest to implement and their implementation may be done in either the time-domain or the frequency-domain. At the transmitter, the analog speech is sampled, quantized, and encoded into digital stream for transmission. At the receiver, a decoder reconstructs the original speech signal. The coder–decoder combination is commonly referred to as a *codec*. Waveform speech coding techniques implemented in the time-domain include pulse code modulation (PCM), differential PCM (DPCM), delta modulation (DM), and adaptive predictive coding (APC). One form of delta modulation known as digitally variable slope delta modulation (DVSDM) is used in the second-generation UK cordless telephone system (CT2) because of its implementational simplicity and low cost, but to some extent sacrificing voice quality. When waveform coding is implemented in frequency-domain, the speech signal is filtered into contiguous, nonoverlapping sub-bands encoded independently using time-domain techniques. Examples include sub-band coding (SBC) and adaptive transform coding (ATC) schemes. In model-based speech coding techniques, signal processing algorithms are used to extract and transmit certain parameters from the analog speech waveform that correspond to the actual time-varying parameters of the speech production mechanism in the human vocal tract (modeled as an electric filter). Thus the algorithms, which are usually called *vocoders,* attempt to describe the speech production mechanism in terms of a few independent parameters used as the information-bearing signals. At the receiver, the received parameters are decoded and used to control a speech synthesizer which is an algorithmic representation of the speech generation model, thereby, approximating the original speech signal. Vocoders are medium complexity systems and operate at low bit rates. Their poor speech quality may be attributed to the oversimplified source models used and the assumption that the source and the filter are independent. The poor and synthetic quality of speech vocoders has led to the speech coding approach known as *residual excitation.* In this approach, the speech is synthesized, but a small part is transmitted as a coded waveform part and another as a vocoded part; hence the name *hybrid.* The penalty is the higher bit rate of transmission required, but now a very much improved speech quality is realized. Examples of model-based speech coding schemes that are often used in wireless applications include linear predictive coder (LPC) which usually requires some form of error-correction coding when used in wireless channels, long term predictors which include multi-pulse excitation (MPE-LPT) and regular pulse excitation (RPE-LPT), code-excited linear predictive (CELP) coder and quadrature code-excited linear predictive (QCELP) speech coding schemes, and residual-excited linear predictive coding (RELPC). Table 1 summarizes the speech coding specification for some cordless and cellular systems (3).

Table 1. Speech Coder Used in Various Digital Radio Systems

Standard	Service Type	Speech Coder Type Used	Bit Rate (kb/s)
GSM	Cellular	RPE-LTP	13
CD-900	Cellular	SBC	16
USDC (IS-54)	Cellular	VSELP	8
IS-95	Cellular	CELP	1.2, 2.4, 4.8, 9.6
IS-95 PCS	PCS	CELP	14.4
PDC	Cellular	VSELP	4.5, 6.7, 11.2
CT2	Cordless	ADPCM	32
DECT	Cordless	ADPCM	32
PHS	Cordless	ADPCM	32
DCS-1800	PCS	RPE-LTP	13
PACS	PCS	ADCM	32

Modulation Techniques

Many modern mobile communications systems use digital modulation techniques as they offer the advantages of greater noise immunity and robustness to channel impairments, easier multiplexing and integration of different types of information (e.g., voice, data, and video), and greater security over their analog counterparts. Digital modulation also allows the use of source coding, error control coding, encryption, and equalization to improve the performance of the overall system. New advances in very large scale integration (VLSI) and digital signal processing (DSP) technology have also improved the effectiveness of the digital modulation schemes used in wireless communications systems. The choice of a digital modulation scheme for use in a wireless environment is influenced by several factors. A desirable modulation scheme provides low bit error rates at low received signal-to-noise ratios, occupies a minimum bandwidth, performs well in a multipath fading environment, and is easy and cost-effective to implement. Since the existing modulation techniques do not simultaneously possess all of these qualities, trade-offs are often made in a modulation scheme for a particular wireless application. The performance of a digital modulation scheme is usually measured in terms of power efficiency and bandwidth efficiency. *Power efficiency* describes the ability of the modulation scheme to preserve the fidelity of the digital message at low power levels. It is usually expressed as the amount of power (usually given as the ratio of the signal energy per bit to noise power spectral density, E_b/N_0) required at the input of the receiver to achieve a specified probability of error. *Bandwidth efficiency* describes the ability of the modulation scheme to accommodate data within a limited bandwidth. It is defined as the ratio of the throughput data rate per hertz in a given bandwidth and directly reflects how effectively the allocated bandwidth is utilized. While power and bandwidth considerations are very important, other factors must also be considered in choosing a modulation scheme for a wireless application. For example, cellular systems are usually interference-limited and the performance of the modulation scheme in an interference environment is also important. The sensitivity of the receiver to timing jitters caused by the time-varying channel is an important consideration. For personal communications systems which serve a large number of subscribers in the service area, a modulation scheme that allows a simple yet efficient detector is desirable in order to minimize the cost and complexity of the subscriber receiver unit. Digital modulation schemes may be classified as linear or nonlinear.

In a linear modulation scheme, the transmitted signal may be expressed as

$$\begin{aligned}\phi(t) &= \mathrm{Re}\{Af(t)\exp(j\omega_c t)\} \\ &= A[f_a(t)\cos(\omega_c t) - f_b(t)\sin(\omega_c t)]\end{aligned} \tag{2}$$

where A is the amplitude, $\omega_c = 2\pi f_c$ is the angular carrier frequency, f_c is the carrier frequency, and $f(t) = f_a(t) + jf_b(t)$ is the complex envelope representation of the modulating signal. Linear modulation techniques have good bandwidth efficiency and are attractive for wireless communication systems where there is increasing demand to accommodate more and more subscribers within a limited bandwidth. However, linear modulation schemes are usually transmitted using RF amplifiers which have poor power efficiency. The use of power efficient amplifiers leads to the regeneration of filtered sidelobes causing severe adjacent channel interference. A number of techniques have been developed to handle this problem in practice. Examples of linear modulation schemes that are commonly used in practical mobile applications include binary phase shift keying (BPSK), quadrature phase shift keying (QPSK), offset QPSK, and $\pi/4$ QPSK. Coherent detection uses the carrier frequency and phase information to provide optimum detection. It is well known that when coherent detection is used at the receiver the bit error rate performance of BPSK in an additive white Gaussian noise (AWGN) channel is given by

$$P_{\mathrm{E,BPSK}}(\gamma) = Q(\sqrt{2\gamma_b}) \tag{3}$$

where

$$Q(t) = \frac{1}{\sqrt{2\pi}} \int_t^{\infty} \exp(-x^2/2)\, dx$$

$\gamma_b = \gamma^2(E_b/N_0)$ and γ is the random attenuation factor due to channel fading. The average bit error rate is then obtained by averaging Eq. (3) over the probability density function of γ^2, $p(\gamma)$. That is,

$$\int_0^{\infty} P_{\mathrm{E,BPSK}}(\gamma) p(\gamma)\, d\gamma \tag{4}$$

In a Rayleigh fading environment, we have

$$p(\gamma) = \frac{1}{\gamma} e^{-\gamma/\overline{\gamma}} \tag{5}$$

and the average bit error rate can be shown to be given by

$$\overline{P}_{\mathrm{E,BPSK}} = \frac{1}{2}\left(1 - \sqrt{\frac{\overline{\gamma}}{1+\overline{\gamma}}}\right) \tag{6}$$

where $\overline{\gamma} = E(\gamma_b) = (E_b/N_0)E(\gamma^2)$ is the average signal-to-noise ratio. In practice, the carrier phase information may not be known precisely or may be random (due to channel fluctuations). In such cases, differentially coherent detection may be employed. The probability of error for differential PSK in AWGN is

$$P_{\mathrm{E,DPSK}} = \frac{1}{2} \exp(-\gamma_b) \tag{7}$$

while the average bit error rate can be shown to be given by

$$\overline{P}_{\mathrm{E,DPSK}} = \frac{1}{2(1+\overline{\gamma})} \tag{8}$$

The bit error rate performance of QPSK is similar to that of BPSK. However, QPSK comprises two orthogonal BPSK signals and thus has the advantage of providing twice the spectral efficiency of BPSK with the same energy. As such, twice as much data can be transmitted in the same bandwidth. QPSK ideally has a constant amplitude property but occasional π-radian phase shifts momentarily cause the signal envelope of filtered QPSK to pass through zero. This causes serious problems when QPSK is used in mobile/satellite applications with nonlinear amplification because of interference from the sidelobes. OQPSK is a modified version of QPSK in which π radian phase shifts do not occur. Although OQPSK has the same signal constellation and, therefore, same bit error performance as QPSK, it is not susceptible to adjacent channel interference caused by the regeneration of sidelobes. In $\pi/4$ QPSK the maximum phase transition of 135° is a compromise between the 180° phase transition of QPSK and 90° for OQPSK. Noncoherent detection can be used to demodulate a $\pi/4$ QPSK signal and can provide better performance in a multipath fading environment than OQPSK. Thus $\pi/4$ QPSK has the same performance (bit error rate and spectral efficiency) as QPSK but has less amplitude fluctuation. $\pi/4$ QPSK has been adopted in the North American digital standard (IS-54), the Japanese digital cellular and the Trans European Trunked Radio (3).

Nonlinear modulation techniques have constant envelope so that power efficient class C amplifiers can be used without introducing degradation in the spectrum occupied by the transmitted signal but they usually occupy larger bandwidths than do linear modulation schemes. Examples of constant envelope modulation schemes that are frequently used in mobile communication applications are frequency shift keying (FSK), minimum shift keying (MSK), and Gaussian minimum shift keying (GMSK). In binary FSK, the transmitted signal of bit duration T_b may be expressed as

$$f_k(t) = \sqrt{\frac{2E_b}{T_b}} \cos\omega_k t, \quad 0 \le t \le T_b \tag{9}$$

when the binary digit k ($k = 0, 1$) is transmitted, where $\omega_0 - \omega_1 = 2n\pi/T_b$ and n is an integer. The average probability of error of the optimum coherent detector in a Raleigh fading channel corrupted by AWGN can be shown to be given by

$$\overline{P}_{\mathrm{E,CFSK}} = \frac{1}{2}\left(1 - \sqrt{\frac{\overline{\gamma}}{2+\overline{\gamma}}}\right) \tag{10}$$

while it is

$$P_{\mathrm{E,NCFSK}} = \frac{1}{2+\overline{\gamma}} \tag{11}$$

for a noncoherent detector. The phase information in FSK signal is not properly utilized at the receiver except for synchronization. MSK is a special case of *continuous phase* FSK (CPFSK) in which the peak frequency deviation is half the bit rate. Thus, MSK may be regarded as a special case of OQPSK with the rectangular pulse shaping replaced by half-sinusoidal pulse shaping. Thus, like OQPSK, MSK has a constant envelope but the phase transitions are continuous. Also, an MSK signal (like an FSK signal) can be demodulated coherently or noncoherently. Finally, GMSK may be regarded as a special case of MSK in which the sinusoidal weighting function is replaced by a Gaussian shaped pulse. GMSK also has constant envelope and excellent spectral efficiency. The average bit error rate of coherently demodulated MSK (and GMSK) in Rayleigh fading channel may be shown to be given by (3)

$$P_{\mathrm{E,MSK}} = \frac{1}{2}\left(1 - \sqrt{\frac{\eta\overline{\gamma}}{1+\eta\overline{\gamma}}}\right) \tag{12}$$

where η is a constant that depends on the product of the demodulator 3-dB bandwidth and the symbol duration (BT) and is given by

$$\eta \cong \begin{cases} 0.68, & \text{for GMSK} \quad (\text{BT} = 0.25) \\ 0.85, & \text{for MSK} \quad (\text{BT} = \infty) \end{cases} \tag{13}$$

GMSK has been adopted for use in GSM, DECT, and US cellular packet data (CDPD).

CHANNEL PROPAGATION

The performance of a wireless communications system is limited by the nature of the mobile radio channel. The transmission path between the transmitter and the receiver usually varies as a result of obstructions from buildings, mountains, and foliage and also as a result of variations in the atmosphere. Thus, electromagnetic wave propagation is usually influenced by the mechanisms of reflection, refraction, and scattering. Multiple reflections cause the transmitted signal to travel along different paths of varying lengths and attenuations, and the interactions of these waves at the receiver location causes multipath fading. Notwithstanding the multipath fading, the long term average strength of the received signal decreases as the separation between the transmitter and the receiver increases.

Channel Propagation Models

Propagation models to characterize the mobile channel can usually be classified into two groups, depending on whether they focus on predicting the average received signal strength (*large-scale* models) or the variability of the received signal (*small-scale* models) at a given distance from the transmitter. Large-scale propagation models based on measurements of actual channels indicate that the mobile radio channel may be characterized by the fourth-power loss model (6), that is:

$$P_r = \frac{P_t V}{d^\alpha} \tag{14}$$

where P_r is the average received power, P_t is the average transmitted power, d is the distance between the mobile and the base station, α is the exponent of power attenuation ($\alpha \approx 4$, for macro-cells), and V is a random variable whose decibel value can be modeled by a zero-mean Gaussian variable (i.e., V is *lognormal*) with standard deviation in the range of 6 to 12 dB. The propagation in an urban microcellular channel with a line-of-sight (LOS) may be characterized by the following dual-slope path loss model (6)

$$P_r = \frac{P_t V}{d^{\alpha_1}\left(1 + \frac{d}{d_0}\right)^{\alpha_2}} \tag{15}$$

where α_1 and α_2 are the attenuation exponents, and $d_0 = 4h_t h_r/\lambda$ with λ being the transmission wavelength, and h_t and h_r being the transmitting and receiving antenna heights, respectively. The decibel standard deviation of the lognormal random variable V is now on the order of 3 dB. Measurements in several urban environments indicate that for small transmitting and receiving antenna heights ($h_t = h_r \approx 3.7$ m), $\alpha_1 \approx 2.2$ and $\alpha_2 \approx 3.3$. For medium antenna heights (about 8.5 m), $\alpha_1 \approx 2.2$ and $\alpha_2 \approx 3.4$ and for large antenna heights, $\alpha_1 \approx 2.1$ and $\alpha_2 \approx 4.2$ (3).

On the other hand, small-scale multipath propagation causes rapid fluctuations in signal strength (fading) over small distances or time intervals. Small-scale propagation is also influenced by Doppler shifts caused by relative motion between the transmitter and the receiver, and by the time dispersion caused by the multipath propagation delays. Time dispersion due to multipath propagation causes the transmitted signal to undergo either *flat* or *frequency selective* fading. The Rayleigh and Rice probability density functions are commonly used to model envelope fluctuations in a flat fading channel when there is no direct transmission path between the transmitter and the receiver and when a LOS component is present, respectively. The Nakagami m-distribution is a more general model that has been shown to provide a better match to envelope measurements in different mobile radio environments than the Rayleigh and Rice distributions. The Rayleigh distribution is a special case of the Nakagami distribution while the Rice distribution can be approximated by the Nakagami distribution for a large range of mean signal-to-noise ratio values.

Combating Multipath Fading

Two major causes of performance degradation in wireless systems are multipath fading and shadowing. There are three ways to combat the effects of fading in these systems, namely: diversity reception, channel coding, and equalization. Diversity reception techniques are used extensively on multipath fading radio channels to reduce the effect of fading on system performance, including both fixed and mobile terminals. In diversity reception, several replicas of the transmitted signal, each carrying the same information and undergoing independent fading, are combined at the receiver. The diversity may be obtained in time, frequency, or space. There are several ways that the receiver may combine the received diversity signals for optimum performance. Three of the commonly used linear combining schemes are maximal ratio combining (MRC), equal gain combining (EGC), and selection diversity combining (SDC). *Microscopic diversity* reduces the effect of instantaneous short-term (or small-scale) fading by combining several uncorrelated signals received at the radio port using any of the combining methods. *Macroscopic diversity* mitigates the effect of long-term (or large-scale) shadowing by using several geographically distributed base stations to serve each cell. The base station with the largest average local mean signal power is usually selected (5,7). Figure 4 shows the diversity gain obtained by using MRC microscopic diversity reception in detecting BPSK signals in a Rayleigh fading channel. We observe from the figure that with a coherent detector at the receiver, in order to obtain an error rate of 10^{-3}, the receiver requires a signal-to-noise ratio (SNR) of 24 dB when there is no diversity ($L = 1$), but only 11 dB with dual-branch diversity ($L = 2$) and 4 dB with fourth order ($L = 4$) diversity. The performance of DPSK is about 3 dB inferior to that of coherent PSK. In Fig. 5, the effect of macroscopic selection diversity is shown, with the lognormal shadowing assumed to have a decibel standard deviation of 6 dB. At a bit error rate of 10^{-3} and with dual order microscopic diversity, macroscopic diversity of order three ($N = 3$) pro-

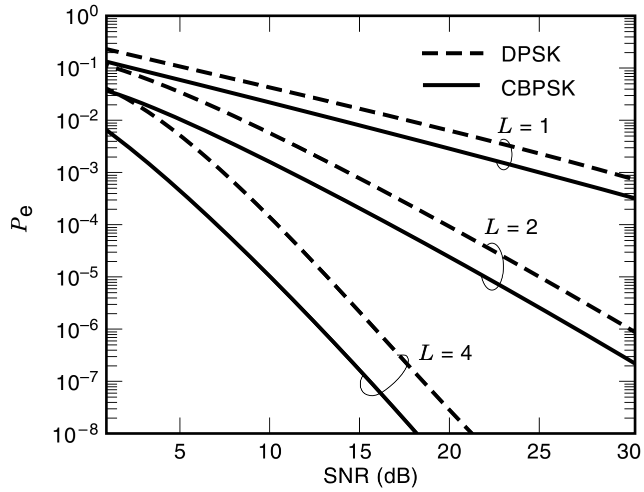

Figure 4. Average bit error rate for a BPSK system with L-branch microscopic MRC diversity in a Rayleigh fading channel.

vides about 5 dB improvement over the system with no macroscopic diversity ($N = 1$).

In channel coding schemes, extra bits (with no message) are added to the information bit stream before being transmitted over the channel. At the receiver, the added redundant bits are used to detect/correct errors that may have occurred in the bit stream. Channel error control techniques used in wireless channels may be classified into three groups, namely: error detection coding (the most commonly used error detection scheme is the cyclic redundancy check (CRC) codes because they are very easy to implement using shift registers), forward error correction (FEC) coding (using block or convolutional codes), and automatic repeat request (ARQ) transmission protocols. The amount of diversity gain introduced by channel coding depends on the minimum distance of the code. The addition (and interleaving) of redundant bits into the data bit streams in the channel coding process gives rise to time or frequency diversity and improves the resistance of the system to multipath fading.

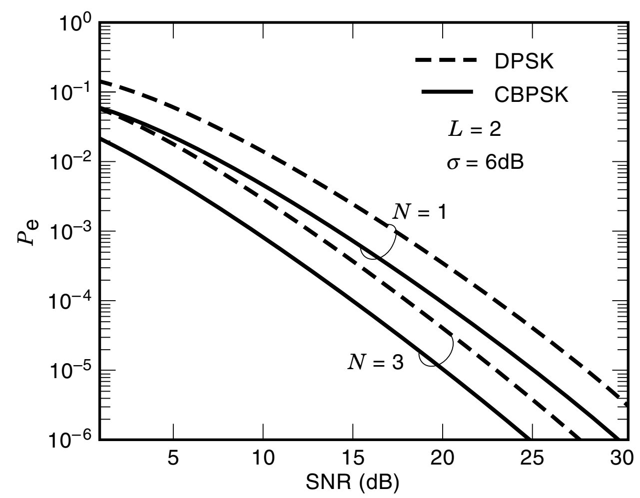

Figure 5. Average bit error rate for a BPSK system with macroscopic selection diversity (and dual-branch microscopic MRC diversity) in a shadowed Rayleigh fading channel.

The time-variant multipath channel after exhibits bursty error characteristics. By the process of interleaving, the bursty channel can be transformed into a channel having independent errors by spreading the coded data over several time slots. Interleaving is used extensively in the second-generation digital cellular systems.

A very serious problem in high data rate transmission systems is intersymbol interference (ISI) caused by frequency selective multipath fading. In this case, increasing the transmission power worsens the problem because the interference power increases. Signal processing techniques (known as equalization) may be used to minimize the effect of ISI. In wireless applications, *adaptive* equalization is used since the mobile channel is random and time varying. The operation of an equalizer usually involves the transmission of a known, fixed-length training sequence to set the parameters of the equalizer at the receiver. New algorithms, called blind equalization, which do not require training sequences are currently under research. During data transmission, the adaptive equalizer uses recursive algorithms to evaluate the channel and estimate the filter coefficients which are used to compensate for the channel distortions. Adaptive equalizers can be classified as linear or nonlinear depending on how the equalizer output is used for subsequent control (feedback) of the equalizer.

WIRELESS TRANSCEIVER STRUCTURE

The complexity of radio communication systems is increasing significantly with the application of more sophisticated multiple access and digital modulation techniques which are necessary in order to accommodate the tremendous growth in the number of subscribers of wireless communication services. Advances in wireless technology and new applications for wireless systems and services have given rise to a variety of portable voice, data, and messaging systems. The development of low-rate digital speech coding techniques and the continuous increase in the device density of integrated circuits have led to completely digital second-generation cellular systems. Also, the evolving third-generation cordless, cellular, and PCS systems are all expected to be fully digital. Digital signal processing (DSP) techniques traditionally used for speech and channel codecs are presently being used extensively for advanced digital communications transceiver design. In addition to speech and channel codecs, these techniques are also being used for detection and demodulation, equalization, frequency synthesis, and channel filtering.

Radio Receiver Principles

A considerable amount of computing resources are necessary to achieve the performance desired for personal communication systems and the required power needed to drive the constituent units of the system may be prohibitive for portable applications. Thus, the key requirements for wireless portable terminals are performance, cost, power consumption, and size. Low power consumption may be achieved through technology and system-level trade-offs. The receiver power is consumed by the RF components, baseband DSP, digital application-specific integrated circuits (ASIC), and mixed signal devices. At the system level, power consumption may be optimized by proper choice of system operations such as time-division-multiplexing and voice-activity detection. For example, many digital processors feature power-down modes that allow turning off peripheral and certain computational units. One draw back of such method, however, is that it does not always allow for fast ramp-ups. In a wireless environment, the receiver may have to process very low desired signal levels in the presence of large levels of unwanted signals. Therefore, the architecture used in the receiver front-end must satisfy requirements which include sensitivity, dynamic range, selectivity, and manufacturability. Radio receiver principles that may be used include the superheterodyne principle, digital receivers with DSP techniques, and direct conversion receivers (8).

In a direct-conversion receiver, the received rf signal is filtered in a duplexer, and passed through a low-noise amplifier followed by a bandpass filter. The output of the filter is then split and, along with two local oscillators in phase quadrature at the carrier frequency, are fed to a quadrature mixer. The outputs of the quadrature mixer are then passed through a low-pass filter followed by analog-digital (A/D) conversion to produce the inphase (I) and quadrature (Q) samples. Since the receiver processes the full RF spectrum at baseband, this architecture requires high dynamic range, high sensitivity, low noise, as well as proper amplitude and phase balancing between the I and Q branches. The main advantage of the direct-conversion architecture is its simplicity as it has a low component count. It also has a wide tuning range and high selectivity. However, a number of challenges are present in its realization. For example, a high-gain low-noise mixer is necessary to combat the $1/f$ amplifier noise at baseband as well as a technique to cancel the associated large dc-offset. The direct-conversion receiver architecture is used in a number of cellular and wireless products. For example, a radio receiver that incorporates direct-conversion into an integrated circuit in a way that avoids the need for discrete intermediate frequency filters and is particularly suitable for use in wireless devices has been proposed (9). Also, FSK paging receivers at 450 MHz and 930 MHz as well as some 900 MHz wireless LAN products are available.

Miniaturization

Wireless personal communication devices such as pagers and cellular and cordless telephones are becoming more compact and more light weight as a result of improvements in device-mounting technology and development of different kinds of devices. By using advanced very large-scale integration (VLSI) technology, the implementation of complex algorithms is economically feasible. The use of complementary metal-oxide semiconductor (CMOS) device scaling technology has facilitated the employment of denser and faster memory chips as well as digital microprocessors. Rapid advances in solid-state integrated circuit technology have fueled the growth of commercial wireless communication systems with the desire to produce high-performance, low-power, small-size, low-cost, and high-efficiency devices. The increasing use of integrated circuits in radio designs has resulted in significant improvements in the reliability and performance of the digital receivers. Rapid advances in packaging technology resulting in compact designs of wireless terminals as well as considerable drops in manufacturing costs resulting from improved procedures in assembly and testing have also increased the deployment of these devices.

Design Tools

The life cycles of many cellular and cordless products are very short. As such, in order to compete successfully, companies must turn system concepts into silicon VLSI very rapidly. High-quality computer-aided design tools are therefore very important for efficient design, simulation, and realization of the systems. The design methodology used in these tools must be chosen to allow different levels of abstraction at different points in the time scale. The algorithms used in these design tools are usually comprehensively specified with block diagrams which are specified hierarchically, with each block representing a signal processing operation. The blocks are usually parameterized so that automatic evaluation of the different simulations can be done based on the block diagrams. Depending on the digital communication system involved, multirate and variable rate processing may be supported. Two simulation approaches that are available in practical design tools are the data-flow-oriented approach (e.g., COSSAP from Synopsys) and the time-driven approach (e.g., SPW from Candence) (10). In addition to the algorithmic simulations, the architecture of the digital communication system also needs to be simulated. The simulation of the architecture may be software-based or hardware-based.

OVERVIEW OF DIGITAL RADIO SYSTEMS

Digital Cellular Systems

Digitization allows the use of TDMA and CDMA over FDMA as multiple access alternatives. The North American Digital Cellular systems have evolved into two *Interim Standards,* one based on TDMA (IS-54) and the other based on CDMA (IS-95). The Global System for Mobile communication (GSM) as well as the Japanese Pacific Digital Cellular (PDC) system (which is very similar to the IS-54 system) are also based on TDMA while the Broadband-CDMA (IS-665) system is a specialized CDMA system (3,5,11).

North American Digital Cellular Systems. The development of a digital cellular standard in North America came as a result of tremendous increase in the demand for cellular services. The capacity of the first-generation analog advanced mobile phone system (AMPS) was limited, and there was no new spectrum available to meet the increased demand. Therefore, the objective of the second-generation systems was not only to increase the capacity of the existing spectrum, but also to provide additional services. The Cellular Telecommunications Industry Association (CTIA) which consists mainly of cellular service providers and the Telecommunication Industry Association (TIA) consisting of equipment manufacturers established a technical committee to develope a digital standard. Finally, in 1989, the industry adopted the dual-mode transmission standard which is referred to as the Electronics Industry Association Interim Standard 54.

TDMA System (IS-54). This is an all-digital second generation cellular system that was designed to co-exist with and eventually replace the first-generation analog cellular system.

On the forward link, the spectrum allocation for IS-54 is 824 to 849 MHz while on the reverse link it is 869 to 894 MHz. The modulation scheme used is differential quadrature phase-shift keying (DQPSK) with $\pi/4$ radians phase shift between successive symbols, to reduce amplitude fluctuations in the signal envelope. However, being a linear modulation scheme, it has poor power efficiency resulting in larger size and weight of the handset. Each TDMA frame has 6 time-slots of 324 bits each, with a frame length of 40 ms, giving a bit rate of 48.6 kb/s. Since the channel spacing is 30 kHz, the resulting bandwidth efficiency of 1.62 bits/s/Hz is relatively high. The speech coder is VSELP operating at 7.95 kb/s and produces a speech frame every 20 ms (or 159 bits every second). Of these, the leading 77 bits of each frame are protected with error control coding and the remaining 82 bits are unprotected, resulting in 260 channel bits per frame. Thus the full-rate coder results in a transmitted data rate of 13 kb/s.

CDMA System (IS-95). This system is based on direct sequence CDMA (DS-CDMA) and was proposed by Qualcomm in 1989 and adopted in 1993. IS-95 was also designed to be compatible with the AMPS. The spectrum allocation for IS-95 on the forward link is 824 to 849 MHz while it is 869 MHz to 894 MHz on the reverse link. With an allowable bandwidth of 1.25 MHz, it uses a direct sequence spread spectrum signal with chip rate 1.228 Mb/s. The speech coder used is QCELP with variable rates (ranging from 1200 b/s to 9600 b/s) determined by the accompanying voice activity detector. Block interleaving with duration 20 ms provides time diversity while the wide bandwidth allows for frequency diversity and multipath (RAKE) diversity making the system robust to multipath fading. Different modulation and spreading techniques are employed on the forward and reverse links. On the forward link, BPSK modulation is used with QPSK spreading. For a single user, either form of modulation yields the same performance but in a multiple access environment the use of QPSK spreading randomizes the phase of the desired user relative to the other users in the system giving rise to much less phase degradation for the desired user. Although the 64×64 Hadamard matrix used may allow 64 users in a cell, only 61 Walsh codes are available since the remaining codes are reserved for the pilot, synchronization, and paging channels. Also, on the forward channel, many user signals are multiplexed and transmitted to multiple users, allowing a common pilot signal to be inserted for all the users. Therefore, coherent demodulation is possible on the forward link. On the reverse link, on the other hand, since the users operate asynchronously and are power controlled, no pilot signal is transmitted by the mobile users. Therefore, noncoherent M-ary ($M = 64$) orthogonal modulation/demodulation which is power efficient is employed on the reverse link.

IS-95 is modified in the following ways in order to support higher data rates for better speech quality at PCS frequencies:

1. On the reverse link, the convolutional code rate is changed from 1/3 to 1/2.
2. On the forward link, the convolutional code rate is changed from 1/2 to 3/4.
3. The standard QCELP speech coder is replaced by QCELP13 which also has variable rate and is designed to provide improvements in spectral quantization, voice activity detection, and pitch prediction. It operates in several modes (which includes QCELP).
4. The data rate is changed from (1200, 2400, 4800, 9600 b/s) to (1800, 3600, 7200, 14400 b/s).

Broadband CDMA System (IS-665). The wideband CDMA standard supports several bandwidths (5, 10, or 15 MHz) at PCS frequencies. The forward link is similar to that of IS-95 with a few exceptions. There is a pilot signal, a synchronization signal, and up to seven paging signals and several traffic signals are supported as options. Also, unlike IS-95 where the chip rate is 1.228 Mb/s, in IS-665, several chip rates (of 4.096, 8.192, and 12.288 Mb/s) may be used. On the reverse link, the mobile users transmit pilot signals to the base station. Therefore, coherent detection (of the QPSK modulated signal) is possible. Both the CDMA (IS-95) and the Broadband CDMA (IS-665) systems are synchronized by the Global Position Satellite (GPS) time. The speech coding scheme used is ADPCM.

European Digital Cellular–GSM (DCS 1800). The GSM standard was developed as a joint initiative by members of the Conference of European Posts and Telecommunications Administration (CEPT) with the initial objective of building a unified pan-European network, giving the subscribers a uniform service and easy roaming throughout all of Europe. The GSM technical standard makes full use of currently available technology, incorporating features such as low bit rate speech, convolutional channel coding with bit interleaving, and frequency hopping. Services supported by GSM may be classified into three types, namely: telephone services, data services, and supplementary ISDN services. The spectrum allocation for GSM at 900 MHz is categorized into the *standard* or the *extended* GSM band while the allocation for GSM at 1800 MHz is referred to as *Digital Cellular System 1800* (DCS 1800) band. The frequency assignments for these bands are as follows: for forward link, 935 to 960 MHz standard GSM, 925 to 960 MHz extended GSM, and 1805 to 1880 MHz DCS 1800; for reverse link, 890 to 915 MHz standard GSM, 880 to 915 MHz extended GSM, and 1710 to 1785 MHz DCS 1800.

With a spacing of 200 kHz, the standard GSM has 124 channels, the extended GSM has 174 channels, and DCS 1800 has 374 channels. Each GSM channel supports 8 simultaneous users using TDMA of frame length 4.615 ms. The modulation is GMSK with $BT = 0.3$ and slow frequency hopping every frame at 217 hops per second is used to provide additional protection against frequency selective fading and co-channel interference. Interleaving is also used to minimize the effect of deep fades. The speech coder is a regular pulse excited linear predictive coder (RPE-LPC) with long-term prediction with voice detection capability (voice activity detection factor of 40%) and provides a net bit rate of 13 kb/s. It operates in discontinuous transmission mode to prolong battery life. Presently, GSM networks have been deployed in over 60 countries in Europe, the Middle East, Asia, and Africa. In North America, GSM is deployed as PCS 1900.

Japanese Personal Digital Cellular. Established in 1991, the Japanese Personal Digital Cellular (PDC) system is very similar to the North American IS-54 system in terms of their operational characteristics and in the requirement that they replace an existing analog cellular system. The frequency allocation for the PDC represents the main difference between the two systems. PDC has two small frequency bands in the 800/900 and the 1400 MHz band. On the forward link, the frequency assignments are 810 to 826 MHz and 1477 to 1501 MHz while on the reverse link, they are 940 to 956 MHz and 1429 to 1453 MHz. With a channel spacing of 25 kHz to be compatible with the existing analog system, PDC uses TDMA to multiplex three slots for three users in a 20 ms frame onto a carrier. The modulation is $\pi/4$ DQPSK with a channel data rate of 422 kb/s and the VSELP voice coder uses error correction coding. Mobile-assisted handoff facilitates the use of small cells, and with the use of space diversity, reduces the required carrier-to-interference ratio. The system provides high quality services, high security, and long handset battery life.

Digital Cordless Telephony

Cordless telephones are low-power, low-range, full-duplex communication systems that use radio to extend the handset to a dedicated base station with a specific telephone number that is connected to the public switched telephone network. Cordless telephone systems provide the user with limited mobility and it is usually not possible to maintain a call if the user travels outside the coverage range of the base unit. In the first-generation cordless telephone systems, the handset typically operates with localized mobility within a very limited range (on the order of 10 m) of the base unit and is used in the home or in the office. They use analog frequency modulation and operate mainly as extension telephone to a transceiver connected to the public wired network. Because of its analog nature and limited operating range, it has limited traffic carrying capacity which in turn limits the full development potential of these systems. Second-generation cordless telephone systems are based on digital transmission format and provide wider coverage ranges, offer good speech quality, provide better security, are more resistant to interference and noise, and use compact handsets with built-in antenna (2).

Cordless Telephone—CT2. This is a second-generation cordless telephone standard introduced in Great Britain in 1989 and designed for residential and office use. It is also used to provide *telepoint services.* Telepoint is a service that is provided to cordless handset owners from cordless base stations located in public places, such as railway stations and shopping centers. This is a basic public communication service for the less migratory, more localized sector of the travelling market and does not compete directly with the wide roaming mobile cellular network. Thus, the handset purchased for residential or office use can also be used to access the telepoint service while the user is in transit between the home and the office. In CT2, speech waveforms are coded using ADPCM with a bit rate of 32 kb/s. Two-way full duplex conversation is achieved using time division duplexing (TDD). The modulation used is Gaussian filtered FSK with bandwidth-bit period product $BT = 0.3$. A Canadian enhancement of CT2 is called CT2+ and provides additional mobility management functions.

Digital European Cordless Telecommunication (DECT). DECT is a pan-European standard for cordless telephone that was designed to provide cost-effective communication to high user densities in picocells. Intended applications of DECT include residential cordless telephony, telepoint services, and cordless PBX. Although DECT is functionally closer to a cellular system that a standard cordless telephone system, the interface of DECT to the PSTN or ISDN network remains the same as for a corded telephone. DECT uses TDMA with TDD and the base station can support multiple handsets simultaneously with a single transceiver. The modulation and speech coding techniques used in DECT are similar to those in CT2.

Personal Handyphone System (PHS). PHS is a Japanese air interface standard with the design objective of providing not only service for home and office use, but also for public access capability. PHS uses TDMA and TDD, with each TDMA frame of 5 ms duration. The speech coding used is ADPCM with data rate of 32 kb/s in conjunction with CRC error detection (with no error correction) and the modulation used is $\pi/4$ DQPSK. Since PHS uses dynamic channel assignment the base stations can allocate channels based on the signal strength seen at both the base station and the portable, and handoffs are supported only at walking speeds as the system is designed for microcell/indoor PCS use.

Personal Access Communication Systems (PACS). PACS is a third-generation personal communications system designed to support voice, data, and video images for low-speed portable applications in microcell/indoor environments. The PACS interface provides wireless connectivity to a local exchange carrier (LEC) and it uses TDMA, with frequency division duplexing (FDD). The modulation used is $\pi/4$ QPSK, with coherent demodulation which provides substantially better performance than other digital cordless telephone systems with discriminator-based receivers. Two-branch polarization diversity with feedback at both the base station and the handset gives an improvement that approaches a four-branch diversity reception system. The subscriber unit uses adaptive power control to minimize battery drains during transmission and to reduce co-channel interference on the reverse link.

Paging Systems

A traditional paging system is a one-way, wireless communication device that sends brief messages (usually a numeric message, an alphanumeric message, or a voice message) to notify a subscriber of the need to call a particular telephone number or to receive further instruction from another location. There are two types of paging systems, namely: the radio common carrier (or a *subscriber* system) and the *private* paging systems. The subscriber paging system is a licensed, public paging company providing paging services to the public and the coverage area may be local, statewide, nationwide, or international. The private paging system involves a customer-owned transmission system and paging receivers for private paging use. When multiple transmitters broadcast a page (known as *simulcast*), the subscribers can roam from the home area to anywhere the paging system is networked. The traditional definition of paging has evolved from the one-way communication device to a two-way device that sends and receives data with services including customized response functions, connection to on-line information services, e-mail messaging, etc. A number of signaling standards for paging

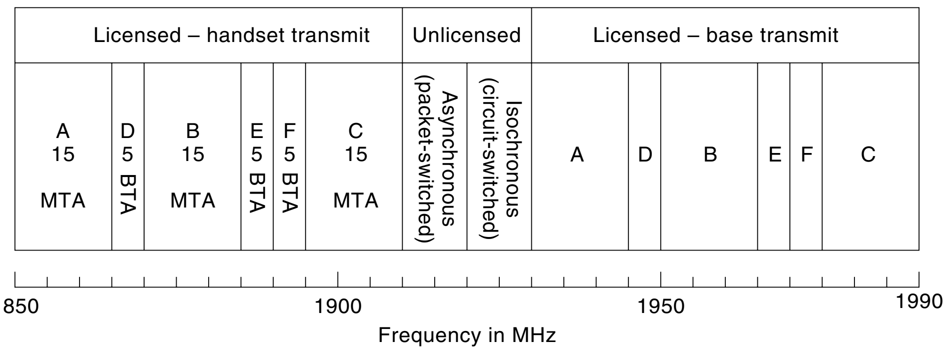

Figure 6. The US PCS frequency plan.

systems have also evolved. Examples include POCSAG which is a one-way paging system, ERMES which is the European paging system, Motorola's FLEX family of paging products, and AT&T's P-act which is a narrowband PCS paging system (12).

PERSONAL COMMUNICATION SERVICES

Wireless personal communication services (PCS) is a concept that extends wireless communications beyond the limitations of the current cellular system to provide users with the ability to communicate with anyone, anywhere, and at any time. A number of characteristics are generally associated with PCS in order to provide reliable service on demand anywhere, any time. The handsets must be portable and easy to use with a long battery life, each user must have a single personalized number that can be reached anywhere, and the system must provide an individualized feature profile that follows the user and provides service at any location. PCS must support various user mobility patterns and provide wide user roaming ability to provide for universal accessibility. Consequently, different cell sizes must be used depending on the type of application and user density. In general, four types of cells can be distinguished for PCS: the picocell (for low power indoor applications); the microcell (for low power indoor or outdoor pedestrian applications in high population density areas); the macrocell (for high power vehicular applications); and supermacro cells (for use with satellite systems). Radio systems for PCS must have a variety of operating power levels and the users should be able to use the service in diverse environments with a wide variation in the radio propagation properties. The system must allow easy integration of the wireless system with the wireline system and ubiquitous deployment of the radio systems.

The services that can be offered by PCS and cellular are identical, except that the operating frequencies are different. The subscriber is indifferent to the frequency band as long as the services are not affected. Thus, the main forces that influence the PCS and the cellular industries are similar. Some of these forces are the regulators, PCS operators, equipment vendors, subscribers, and competing products. The service area in the United States and its territories are divided into 51 major trading areas (MTA) and 493 basic trading areas (BTA) according to the Rand McNally 1992 *Commercial Atlas and Marketing Guide,* 123rd Ed. Based on FCC regulation, each area in the United States can be served by at least six PCS operators, in addition to the existing two cellular operator and one specialized mobile radio service (SMR) operators (12).

One of the most important defining elements of PCS is the FCC's allocation of 120 MHz of spectrum around the 1200 MHz frequency band for licensed operation and another 20 MHz for unlicensed operation, resulting in a total of 140 MHz for PCS. This is about three times the 50 MHz spectrum currently used by the cellular system, indicating the resolve to make PCS more widely available. Figure 6 shows the frequency allocation for PCS.

No specific technology has been mandated by the FCC for the PCS market. As such a number of competing technologies have been chosen by the major PCS carriers. Presently, seven PCS standards can be identified. These include GSM (PCS 1900), a derivative of the GSM/DCS 1800 standard; CDMA, a modification of IS-95; DAMPS; PACS, air interface for pedestrian applications; CDMA/TDMA, a composite hybrid that uses TDMA within cells and CDMA between cells; DECT-based TDMA; and broadband CDMA (12). Of these, the four most widely deployed systems at the present time are GSM, CDMA, TDMA, and PACS.

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DIGITAL RADIO BROADCASTING (DRB). See DIGITAL AUDIO BROADCASTING.

DIGITAL RADIO COMMUNICATION. See MOBILE COMMUNICATION.

DIGITAL RECORDERS. See RECORDERS.

DIGITAL RELAYS. See POWER SYSTEM RELAYING.