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WIRELESS COMMUNICATIONS SYSTEMS

The birth of wireless communications systems can be traced back to G. Marconi's experiments to demonstrate the use of radio to contact ships in the English Channel in 1897. Since then, wireless communications systems have experienced a tremendous evolution and their use has spread throughout the world.

A brief historical review of the evolution of wireless communications can help us to understand the impact of these communications systems in our lives. A more detailed account of the history of wireless communication can be found in the article HISTORY OF WIRELESS COMMUNICATIONS.

After Marconi's experiments, the first practical communication system involving vehicles was deployed by the police of Detroit (U.S.) at the end of the 1920s. This very first mobile network supported only unidirectional links (from central station to mobile terminals) and was based on amplitude modulation (*AM*). This unidirectional system was soon improved with bidirectional capability. In 1934 more than 200 police radio systems had been adopted for public safety in the United States, all based on *AM*.

Because of the modulation characteristics, vehicle ignition noise was the factor that limited the quality of these early systems. In 1935, E. Armstrong demonstrated frequency modulation (*FM*) for the first time, and since then *FM* has become the preferred modulation technique for wireless applications.

After the great improvement in manufacturing and miniaturization achieved during World War II, the number of mobile users experienced an enormous increase and the main characteristics of wired telephony were introduced into wireless systems. Consequently these early-deployed systems began to saturate the spectrum, and the need for more efficient planning of wireless communications systems was evident.

The solution was found during the 1950s and 1960s in AT&T Bell Labs, and it received the name of cellular radiotelephony. The main idea behind the cellular theory, as it will be explained in the next section of this article, is the reuse of frequencies throughout the area of deployment of a given system. Technology was not ready to implement the cellular concept until the late 1970s.

The first commercial cellular telephone systems were that deployed by the Japanese Nippon Telephone and Telegraph (*NTT*) Company in 1979, the Nordic Mobile Telephone (*NMT*) system developed in 1981, and the U.S. Advanced Mobile Phone System (*AMPS*) implemented in 1983. The European Total Access Cellular System (*ETACS*) was deployed in 1985 and is virtually identical to *AMPS*.

These first-generation analog cellular systems were succeeded by a second generation of digital cellular systems. In Europe, the different first-generation cellular systems were incompatible with each other because of the different frequency bands and protocols being used. This fact motivated the creation of a special group within the European Conference of Postal and Telecommunications Administrations (*CEPT*) called the Special Group for Mobile Communications (*GSM*). The activities of the *GSM* group gave birth to the first pan-European digital cellular system. The *GSM* standard was first deployed in 1990 using the 900 MHz band, and it is gaining worldwide acceptance as the first digital cellular system with modern network features.

The U.S. Digital Cellular (*USDC*) was adopted in 1991 as a means of tripling the capacity of *AMPS*. In Japan, the Pacific Digital Cellular (*PDC*) or Japanese Digital Cellular (*JDC*) standard, very similar to the *USDC*, was adopted.

Since the digital revolution in wireless communications, the need of what has been called *personal communication systems* (*PCS*s) has arisen. The idea behind the name of personal communications is that anyone should be able to communicate anytime and anywhere. *PCS*s, also called personal communication networks (*PCN*s), are being deployed above 1800 MHz using, among others, extended versions of the *GSM* standard.

Both *USDC* and *GSM* are time-division multiple access (*TDMA*) systems. A cellular system based on code division multiple access (*CDMA*) was deployed by the U.S. company Qualcomm in 1993 and adopted as an interim standard (IS-95).

None of the systems described above has a worldwide projection. The idea of producing an enhanced worldwide standard led the International Telecommunications Union (*ITU*) to promote the creation of the family of standards named *IMT*-2000 (International Mobile Telecommunications for the year 2000). This family, formerly called the Future Public Land Mobile Telephone System (*FPLMTS*), will increase the amount of data that a wireless terminal is able to obtain from the network, giving rise to the possibility of multimedia wireless communications.

Many different wireless standards are converging in the *IMT*-2000 family. A classification of wireless communications systems can give the reader some insight into the similarities and differences of these existing and future systems.

Classification of Wireless Communications Systems. Wireless communications systems can be classified according to different criteria. One of the preferred schemes relies on whether the system is intended for public or private use:

Private mobile radio (*PMR*) *systems* are intended for private access and usually they are not connected to the public switched telephone network (*PSTN*). They are generally dedicated to the management of vehicle fleets and dispatching tasks. They can serve different areas, from small ones to nationwide.

Public mobile telecommunications (*PMT*) *systems* are intended to serve great areas, normally nationwide or larger. They are connected to the *PSTN* and offer similar services to those provided by wired *PSTN* terminals.

Most of the wireless communications systems used today can be considered to belong to one of these two categories. A comprehensive description is provided in the sections “Fundamentals of Wireless Communications” and “Practical Wireless Systems and Products” below.

Evolution of the Demand for Wireless Products. In the last years the wireless systems industry has grown by orders of magnitude, mainly because of circuit fabrication improvements, large-scale circuit integration, and miniaturization, making small, portable, cheap wireless equipment a reality.

In the European Union (*EU*), the cellular mobile market is growing at an annual rate of 55%. Figure 1 (1) shows, as an example, the penetration of cellular products in some European countries until November 1999. This growth suggests that more than 60 million citizens will be mobile in the year 2000, and some projections (2) indicate that 80% of the *EU* population will have some form of mobile communication terminal by 2020.

On this basis, 280 million terminals need to be designed, manufactured, and provided to the customer. Even if this projection is halved, the number of mobile customers will still approach the present number of wired telephone connections within the *EU* (2).

In the United States the number of cellular telephone users grew from 25,000 in 1984 to about 16 million in 1994, and since then, wireless services have been experiencing customer growth rates in excess of 50% per year (Fig. 2). According to the Cellular Telecommunications Industry Association (*CTIA*) (3), there were over 60 million wireless subscribers in the United States in June 1998.

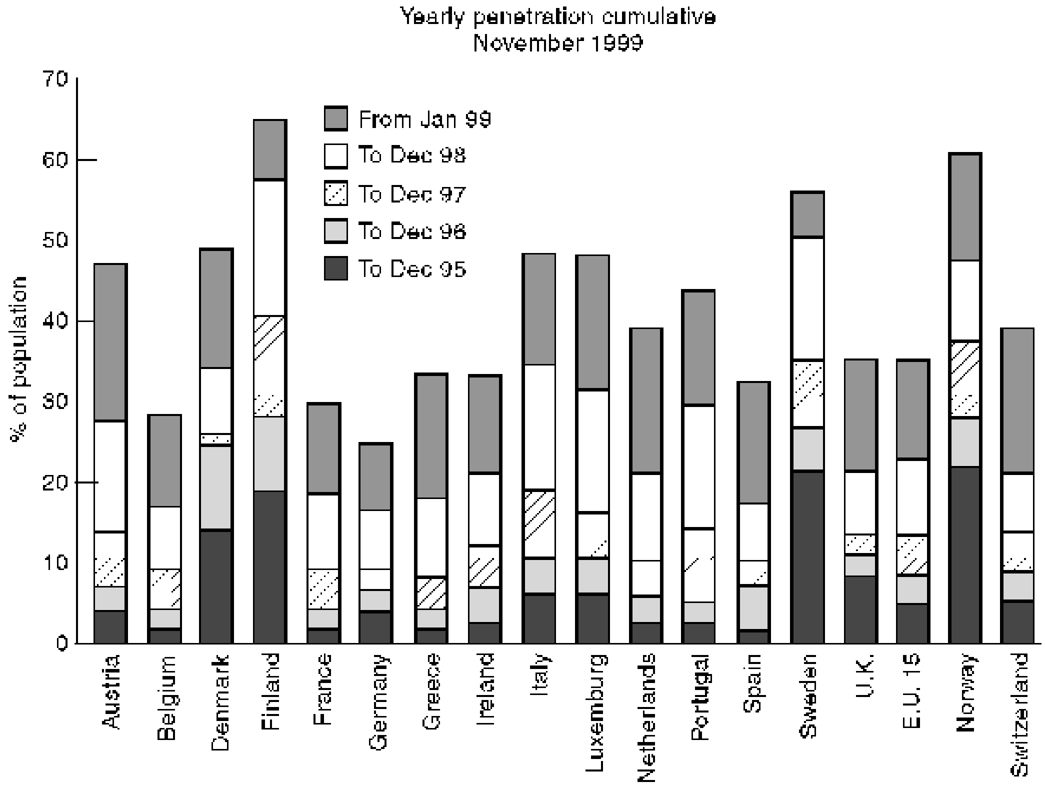

Fig. 1. European Cellular penetration until November 1999. Penetration exceeded 60% at the end of 1999 in some European countries, and projections indicate that 80% of the *EU* population by 2020 will have some form of mobile communication terminal

Fundamentals of Wireless Communications

Radio, or wireless communication, is the use of radiated electromagnetic waves that permits the transmission and reception of information over a distance without the use of wires. At the *transmitting*, or sending, end, the information to be sent (e.g., a voice signal) is imposed on a locally generated radio frequency (*RF*) signal called a *carrier*. The process of imposing the information signal on the carrier is called *modulation*. At the *receiver*, the information signal is extracted from the received signal in a process referred to as *demodulation*.

A pure, unmodulated radio carrier conveys no information and occupies only an infinitesimal amount of the spectrum. Modulation of the radio signal inevitably causes a spreading of the radio wave in frequency. Thus a radio signal conveying information occupies a range of frequencies called a *channel*. In general, the more information is sent per unit of time, the wider the channel must be.

In addition to extracting the information from the radio wave through demodulation, it is also a principal function of a receiver to accept only the information in the chosen channel and reject other information being sent simultaneously in other (e.g., adjacent) channels. The measure of the receiver's ability to reject interfering signals on other channels is referred to as its *selectivity*. Hence, two or more radio systems can use the radio spectrum in the same area at the same time as long as (a) they are separated sufficiently in frequency, so that their channels do not overlap, and (b) the receivers involved have sufficient selectivity to reject the signals on adjacent channels.

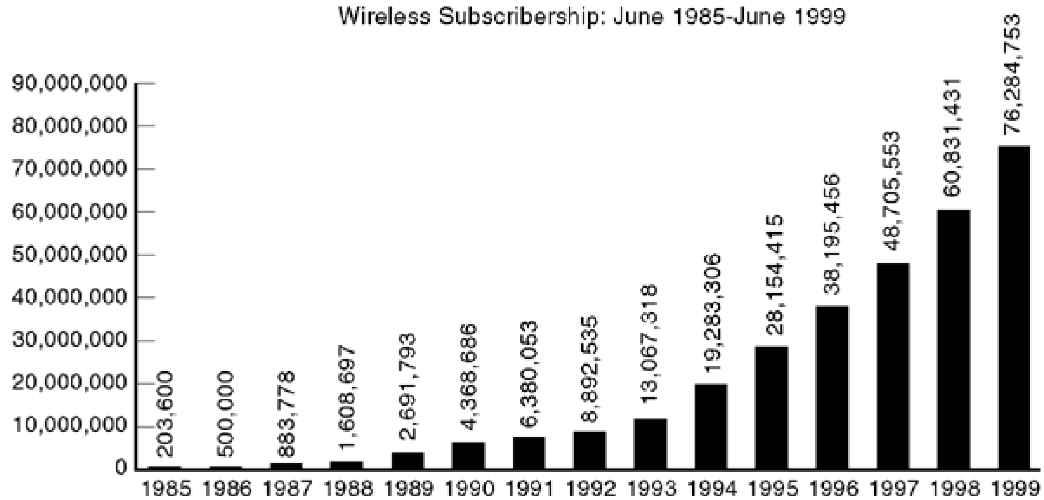

Fig. 2. Evolution of wireless subscriber number in the USA between June 1985 and June 1999 (source: Cellular Telecommunications Industry Association). Subscribers increased by 25.5 percent between June 1998 and June 1999.

In many, if not most, communication systems, it is desirable to be able to communicate in both directions at the same time. This system characteristic, which is known as full-duplex operation, is desirable because it lets one party in a voice conversation interrupt the other with a question, or one device to immediately request a retransmission of a block of information received in error during a data communications session. There are two basic ways of providing for full-duplex operation in a radio system, as will be explained in the next section.

The fundamentals of the different wireless systems and the frequency bands allocated are detailed in the following.

Cellular Systems. In the early mobile radio systems, the design objective was to achieve a large coverage area by using a single, high-powered transmitter. To avoid interference between users, the same carrier frequency could not be reused anywhere in the system. Hence, the overall system capacity was equal to the total number of channels: only a few thousand subscribers per system.

A cellular mobile communications system uses a large number of low-power wireless transmitters to create *cells*—the basic geographic service area of a wireless communications system. Variable power levels allow cells to be sized according to the subscriber density and demand within a particular region. As mobile users travel from cell to cell, their conversations are handed off between cells in order to maintain seamless service. Cells can be added to accommodate growth, creating new cells in unserved areas or overlying cells in existing areas.

Frequency Reuse. In mobile systems, users share a pool of available channels (frequencies, time slots, or codes). Let us assume that channels are associated with carrier frequencies. Since propagation losses increase with distance, the same frequency carrier, or channel, can be reused in cells at some distance away. The design process of selecting and allocating channel groups for all of the cellular base stations within a system is called *network planning*. By reusing channels in multiple cells, the system can grow without geographical limits. In addition, this approach makes possible the use of small, battery-powered portable handsets with lower *RF* transmitting power than the large vehicular mobile units used in earlier systems.

Reuse is critically dependent upon the fact that the electromagnetic field attenuation with distance (R) in the cellular bands tends to be more rapid on the earth's surface than it is in free space. As explained more deeply in the next section, measurements have shown repeatedly that typically the field intensity decays like R^{-n}, with $3 < n < 5$ (in free space $n = 2$). If we assume that propagation attenuation does not depend on the

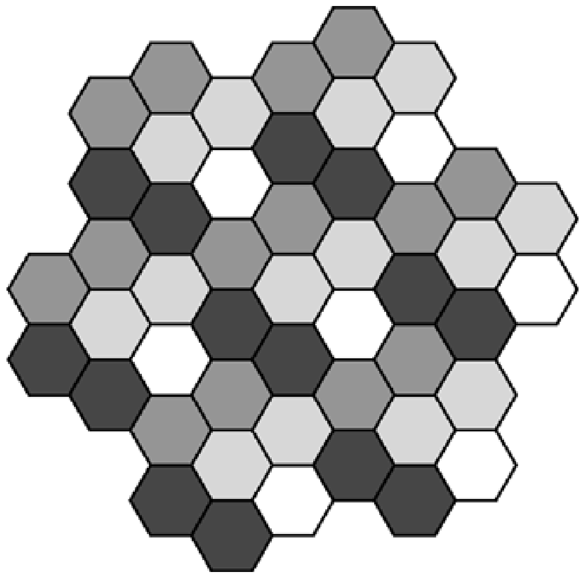

Fig. 3. Illustration of the frequency reuse concept. Cells with the same color use the same set of frequencies. In this example, the cluster size K is equal to 7, and the frequency reuse factor is 1/7.

azimuth angle, and that cell boundaries are at the equisignal points, then a planar service area is optimally covered by the classical hexagonal array of cells.

Because in the hexagonal geometry a cell has exactly six equidistant neighbors, the size and geometry of a group of cells that collectively can use a complete set of available channels are limited (4). Figure 3 shows an example in which seven sets of channels are used, one set in each colored cell. This seven-cell unit, also called a *cluster*, is then replicated over the service area.

No similarly colored cells are adjacent, and therefore there are no adjacent cells using the same channel. While real systems do not ever look like these idealized hexagonal tilings of a plane, the seven-way reuse is typical of that achieved in practice.

The pictures above assume that the cells are using omnidirectional antennas. The system capacity, measured in users per square kilometer, can be increased by antenna *sectorization*. If each site is equipped with three sets of directional antennas, with their azimuths separated by 120°, the interference is reduced, allowing smaller clusters, which means an increase in capacity.

Cell Splitting. Economic considerations make the concept of creating full systems with many small areas impractical. To overcome this difficulty, system operators developed the idea of *cell splitting*. As a service area becomes full of users, this approach is used to split it into smaller ones. In this way, urban centers can be split into as many areas as necessary in order to provide acceptable service in heavy-traffic regions, while larger, less expensive cells can be used to cover remote rural regions (see Fig. 4).

Based on the radius of the cells, there are three types of cellular networks. *Macrocells* are mainly used to cover large areas (1 km to 10 km) with low traffic (rural areas). Microcells are used in areas with high traffic density, like suburban areas; they have radii between 200 m and 1 km. Finally, *picocells*, or indoor cells, have radii between 10 m and 200 m. Today, picocell radio systems are used for wireless office communications.

Architecture of a Cellular Network. A cellular network is composed of three broad functional entities. The *mobile station* (*MS*) (normally a handset) is carried by the subscriber. The *base station* (*BS*) *subsystem* controls the radio link with the *MS*. The *network subsystem*, the main part of which is the *mobile services*

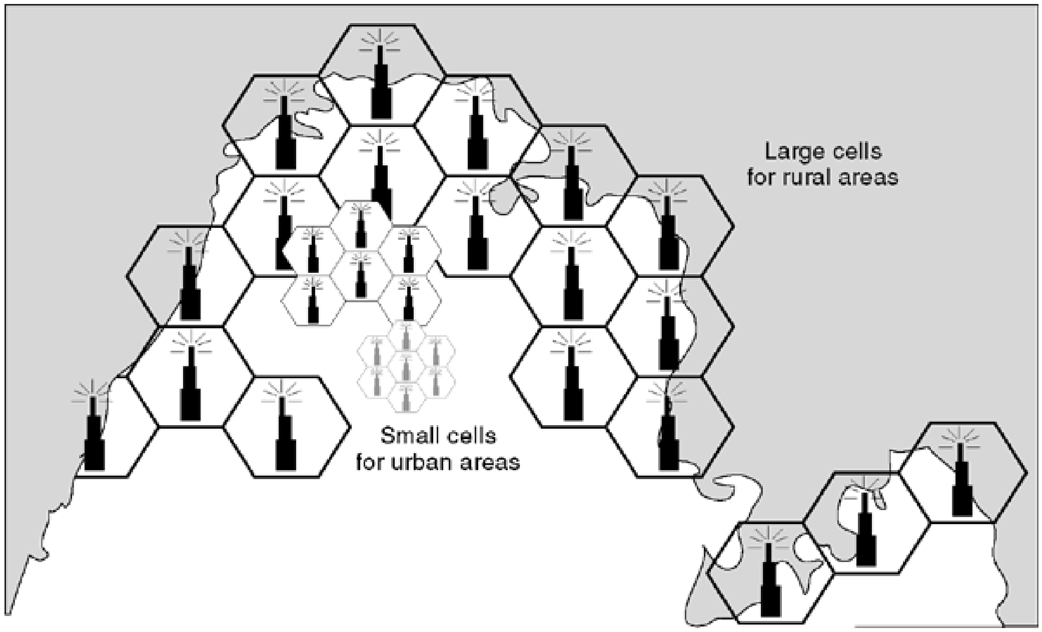

Fig. 4. Cell splitting is the process of subdividing a congested cell into smaller cells, each with its own (low-powered) base station.

switching center (*MSC*), performs the switching of calls between the mobile users, and between mobile and fixed network users. The *MSC* also handles the mobility management operations.

Handover. In a cellular network, the radio and fixed links required are not permanently allocated for the duration of a call. *Handover*, also known as handoff, is the switching of an ongoing call to a different channel or cell (see Fig. 5).

A handover is performed in three stages. The *MS* continuously gathers information on the received signal level of the *BS* it is connected with, and of all other *BS*s it can detect. This information is averaged to filter out temporary fluctuations in received signal level. The averaged data are then passed on to the decision algorithm, which decides if it will request a handover to another station. When it decides to do so, both the old *BS* and the *MS* execute handover.

Cordless Telephone Systems. Early cordless telephones operate solely as extension telephones to a dedicated base station, which is then connected to a dedicate telephone line with a specific telephone number on the *PSTN*. Recent cordless telephone systems employ the same technology as the digital cellular standards. The main difference is that cellular systems were developed for wide-area coverage, whereas the cordless standards are optimized for local coverage, with high densities of users.

The recent digital cordless standards incorporate encryption, and support high-speed data (with circuit- and packet-switched modes) as well as fax and voice communications. The cordless standards provide higher speech quality than the mobile cellular standards (*GSM*, *DCS* 1800) and comparable with that of fixed networks. Though digital cordless systems are more suited to stationary environments than to vehicles, they also provide a limited degree of mobility in a local area.

Digital cordless systems are able to handle up to 100,000 users/km^2 in an office environment. They include *dynamic channel selection* and *allocation* (*DCS* and *DCA*) algorithms that ensure that the best of the available radio channels is always used. This capability ensures that digital cordless systems can coexist with other systems in the same frequency band while providing high-quality, robust, and secure communications.

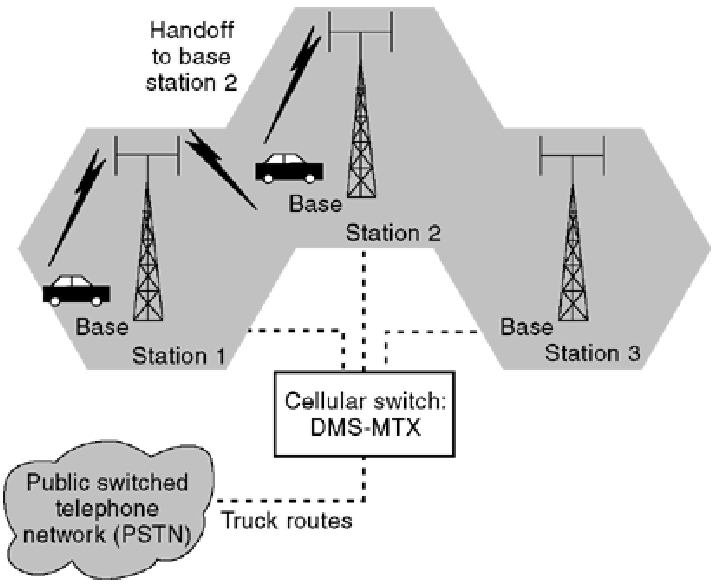

Fig. 5. Handover, also known as handoff, is a process to switch an ongoing call from one cell to the adjacent cell as the mobile user approaches the cell boundary.

Trunking Systems. Radio trunking is a two-way communication technology where a number of users share a common pool of time slots or frequencies that are dynamically allocated by the system as required. As it applies to radio, trunking is the automatic sharing of channels in a multiple repeater system. Trunking concepts are based on the presumption that individual subscribers use the system only a small percentage of the time, and a large number of users do not use the system at the same time. Typical users of trunking systems are private corporations with large fleets (ambulances, buses), government departments (fire brigades, police), and utilities (electric power lines, oil pipelines, railways). Unlike other wireless system, trunking systems can allow *direct mode* communications, that is, direct operation between mobile terminals without use of network infrastructure.

Differences between Cellular and Trunking Systems. One way of looking at wireless communications systems is to chart the degree of customization required by the users against the traffic density. In such a model, the cordless technology is positioned as the technology that offers highest density and lowest degree of customization. Next to it are cellular systems, with medium traffic density, whilst the digital trunking technology is positioned to address low to medium traffic density with a higher degree of customization.

Large Cell Sizes. This fact is also reflected in the spectrum allocation and the cell sizes. At one end, in a dense urban environment, there are digital cordless systems, with cell sizes measured in hundreds of meters and operating at frequencies up to 2 GHz. Cellular systems, which address medium to high user densites with cell sizes measured in kilometers, operate at frequencies of 900 MHz or 1800 MHz. In contrast, trunking systems address urban and suburban (low to medium) user densities, with cell sizes of tens of kilometers and operating frequencies in the lower hundreds of megahertz.

Traffic Profile. Another important factor is the traffic profile. In cellular systems, the call setup time is typically several seconds, in contrast to the fast push-to-talk operation of 300 ms for trunking systems. Also, while cellular calls are point-to-point (individual calls), a large percentage of trunking calls are group calls operating in an all-informed open-channel mode. The average duration of calls is also different: 2 min in cellular networks, compared with 20 s for a trunking call.

Decentralized Architecture. The architecture of trunking systems is optimized for the predominance of local traffic. The need for fast call setup demands that the call switching and management be devolved to the lowest possible point in the network. This requires a decentralized, distributed system architecture, comprising distributed databases and loosely coupled autonomous parts. This, in turn, leads to an inherent reliability and fault tolerance with softer failure profiles.

Paging Systems. A radio paging system is a one-way wireless messaging system that allows continuous accessibility to someone away from the wired communications network. In its most basic form, the person on the move carries a palm-sized device (the pager), which has an identification number. The calling party inputs this number, usually through the public telephone network, to the paging system, which then signals the pager to alert the called party.

The paging system makes efficient use of the radio spectrum, enabling it to provide inexpensive functions satisfying customers' demand. This has contributed to the steady expansion of the paging industry in recent years. The introduction of complementary products (e.g. pocket-sized cellular telephones, cordless telephones, and multifunctional communicators) will inevitably spur the paging industry to provide more value-added services in the future. There will be more varied applications of paging, such as the sending of email, voice mail, faxes, or other useful information to a pager, which will also take on more attractive, innovative forms.

Paging systems vary widely in their complexity and coverage area. While simple paging systems may cover a limited range of 2 km to 5 km, or may ever be confined within individual buildings, wide-area paging systems can provide worldwide coverage. Though paging receivers are simple and inexpensive, the transmission system is quite sophisticated. Wide-area paging systems consists of a network of many base-station transmitters and large radio towers that simultaneously broadcast a page from each base station (this is called *simulcasting*).

Wireless Local Area Networks. A wireless local area network (*WLAN*) is a flexible data communication system implemented as an extension to, or as an alternative for, a wired *LAN*. *WLAN*s combine data connectivity with user mobility and, through simplified configuration, enable movable *LAN*s. With wireless *LAN*s, users can access shared information without looking for a place to plug in, and network managers can set up or augment networks without installing or moving wires. Wireless *LAN*s offer the following productivity, convenience, and cost advantages over traditional wired networks:

- Mobility Wireless *LAN* systems can provide *LAN* users with access to real-time information anywhere in their organization.
- Installation Speed and Simplicity Installing a wireless system can be fast and easy and can eliminate the need to pull cable through walls and ceilings.
- Installation Flexibility Wireless technology allows the network to go where wire cannot go.
- Reduced Cost of Ownership While the initial investment required for wireless *LAN* hardware can be higher than the cost of wired *LAN* hardware, overall installation expenses and life-cycle costs can be significantly lower.
- Scalability Wireless *LAN* systems can be configured in a variety of topologies to meet the needs of specific applications and installations.

Configurations of Wireless Local Area Networks. The simplest *WLAN* configuration is an independent *LAN* that connects a set of PCs with wireless adapters. Any time two or more wireless adapters are within range of each other, they can set up an independent network. These on-demand networks typically require no administration or preconfiguration.

Access points can extend the range of ad hoc *LAN*s by acting as a repeater, effectively doubling the distance between wireless PCs.

In infrastructure *WLAN*s, multiple access points link the *WLAN* to the wired network and allow users to share network resources efficiently. The access points not only provide communication with the wired network but also mediate wireless network traffic in the immediate neighborhood. Multiple access points can provide wireless coverage for an entire building or campus.

More information about *WLAN*s can be found in the article WIRELESS NETWORKS in this encyclopedia.

Frequency Bands Allocated for the Different Wireless Services. In response to rapid growth in the wireless personal communication services, regulatory authorities have steadily increased the amount of spectrum available through successive reallocations of the resource in the higher frequency ranges. Moving higher in frequency to avoid congestion has advantages and disadvantages. Generally speaking, the *RF* devices employed within the system are more costly for higher frequencies, and such frequencies are subject to more blocking and shadowing by buildings or hills. However, the antennas involved are physically smaller, which is an important attribute for systems that seek to serve small portable units carried on one's person. At some risk of overgeneralizing, it can be said that (a) the lower frequency bands are best for economically covering wide areas (suburban and rural areas) where frequency reuse is not as important, and (b) the higher frequency bands are best for covering urban areas where high levels of frequency reuse are desired.

Technology

This section addresses basic technical issues related to the transmission in wireless communications systems. The ideas presented herein will allow the reader to understand the rationale behind the choices made for the standards described in the next section.

A Brief Review of the Propagation Characteristics in These Bands. Radio propagation exhibits inherent advantages over *line* (copper, coaxial, optical fiber, waveguide) transmission. Some of these advantages are lower cost, faster deployment, and the capability to roam. However, a radio channel is less robust than line channels. Radio propagation is not a stationary phenomenon, but the propagation parameter depends on many factors engineers cannot control, as meteorology, or any moving object in the surroundings. This channel instability makes radio transmission a much more difficult problem than line transmission and conditions the design of the equipment.

Physical phenomena that affect both attenuation and delay are free-space propagation, shadowing, and reflections. Moreover, because of multiple reflections, the path for the signal to travel from the transmitter to the receiver is not unique, but several propagation paths can occur simultaneously, producing so-called *multipath* phenomena. These affect both fixed and mobile radio links. However, channel variability is not an issue for fixed links. Propagation conditions in fixed links vary slowly or not at all.

The radio channel effects include amplitude attenuation, angular shift, time delay, and Doppler frequency shift of each of the propagation paths between transmitter and receiver. In mobile links, these four effects are time-varying and different for each path. Their sum over all these paths results in the following effects: variable signal attenuation, variable signal time delay, temporal spread, and Doppler frequency spread. In mobile communications, because of the many factors involved in the radio propagation and their unpredictable dynamics, the above propagation effects are modeled as random processes. Statistics for amplitude attenuation, time spread, and Doppler spread are briefly summarized next.

The variable attenuation suffered by the received signal is usually referred to as *fading*. The fading can be modeled as the sum of three components: large-scale path loss; medium-scale, slow-varying fading; small-scale, fast-varying fading.

The total fading is computed as the product (sum on a logarithmic scale) of the three above-mentioned factors. The large-scale path loss model accounts for the average attenuation value for a geographical area of a size in the range of several hundreds of wavelengths. Large-scale loss is modeled as a constant, and it accounts for free-space attenuation, diffraction, refraction, and reflection. When the line of sight (*LOS*) between transmitter and receiver is not obstructed by any object, the large-scale loss reduces to the free-space loss, and it can be modeled according to the following equation:

$$L_{\text{free space}} = K\left(\frac{d}{\lambda}\right)^2, \tag{1}$$

where K is a constant and λ is the signal wavelength. This attenuation is known as *free-space attenuation*.

When the propagation environment does not correspond to the free-space scenario, the large-scale loss can be modeled as

$$L_{\text{large scale}} = K\left(\frac{d}{\lambda}\right)^{\gamma} \tag{2}$$

where γ is an empirical parameter that varies from 1.8 to 6 depending on the environment (outdoor or indoor, flat or mountainous, desert or forest, etc.).

The shadowing caused by natural or man-made objects close to the receiver is the physical effect responsible for the slow-varying fading. Although this fading factor has slow dynamics, the shadowing causes the deepest variations of the received signal level. The medium scale follows a lognormal distribution, that is, the received power expressed in logarithmic units (decibels) follows a normal (Gaussian) distribution. When designing the link equipment this fading model has to be considered in one or both of the following respects. First, the transmitted power has to be increased to compensate for this variable attenuation, typically with a signaling feedback from the receiver known as *closed-loop power control*. Second, a scrambler must be inserted after the channel coder in such a way that it can be guaranteed that after descrambling at the receiver two consecutive bits do not suffer deep fading simultaneously. Thus, the channel decoder is able to correct the errors caused by the deep fade. To guarantee the efficiency of the scrambler, its symbol time length has to be longer that the fading coherence time. This parameter is studied below.

Multipath phenomena are responsible for the rapidly varying fading. When two paths arriving with close delays have the same phase, the resulting signal is stronger than that due to either path. However, when two paths arrive with a 180° phase difference, the resulting signal is smaller than that due to the stronger path, or the signal may even vanish. The fast-varying component follows a Rice or a Rayleigh distribution, depending on the existence of a *LOS*. When there is no *LOS*, the probability distribution function (*pdf*) for the received amplitude due to multipath effects follows a Rayleigh model. When there are *LOS* conditions, the amplitude *pdf* follows a Rice distribution.

Rapidly varying fading reduces the quality of the link, or in other words, it increases the bit error rate (BER). The techniques usually employed to overcome this quality degradation are forward error-correcting codes, fast transmitted-power control, and the use of space or frequency diversity. Diversity takes advantage of the fact that the probability that two signals with different carrier frequencies or antenna locations suffer a deep fading simultaneously is very small. Thus, proper combination of both frequencies or antenna locations can fight effectively against fast fading.

As mentioned before, knowledge of the temporal correlation of the fading is as important as knowledge of the amplitude distribution for correct system design. The *average fading time* is defined as the expected time the fading exceeds a given attenuation value. For Rayleigh fading, the average fading time τ may be modeled as (5)

$$\tau_{\text{f}} = \frac{\exp(\rho^2) - 1}{\sqrt{2\pi}\,\rho\, f_{\text{d}}} \tag{3}$$

where ρ is the ratio between the fading level and its root mean square (*RMS*) value, and f_d is the maximum Doppler frequency shift, calculated as

$$f_{\mathrm{d}} = \frac{V}{\lambda} = \frac{f\,V}{C} \tag{4}$$

with V the mobile speed, λ the signal wavelength, and f the carrier frequency.

Besides the variable attenuation, multipath effects cause several copies of the same signal to arrive at the receiver with different delays. The time difference between the earliest and last paths is known as the *delay spread*. Typical values for the delay spread in indoor environments are 50 ns to 250 ns for the 900 MHz band, and 10 ns to 20 ns for the 2.4 GHz band (6). Typical values in the 900 MHz band for outdoor environments are 10 μs to 25 μs (7).

If the delay spread is smaller than the reciprocal of the signal bandwidth, the receiver is not able to resolve the different paths, and the multipath phenomenon affects only the received amplitude, that is, in this case it causes fading but does not produce distortion of the received signal. Such fading is known as *flat fading*. The mechanisms available to compensate for this type of fading are channel coding, scrambling, and space and frequency diversity.

However, if the delay spread is larger than the reciprocal of the bandwidth—comparable to or larger than the symbol rate—the multipath effects cause *intersymbol interference*, that is, signal distortion. This fading is known as *frequency-selective fading*. The use of equalizers can overcome partially or totally the intersymbol interference caused by the frequency-selective fading.

Because of user and/or environment motion, each path also has its own Doppler frequency shift, which causes a frequency spread of the received signal. This effect is known as *Doppler spread*. The Doppler spread is closely related to the coherence time of the multipath channel and to the average fading time. All these quantities describe the temporal correlation of the fading.

Modulations in Wireless Systems

Digital modulation is a simple concept. The transmitter groups the digital information to send into packets of N bits. Each of the $M = 2^N$ possible values of the N-bit packet is mapped into a signal, which is properly amplified and transmitted. At the receiver end, the signal arrives corrupted by noise and interference. An optimum receiver detects what signal, among the M possible ones, was transmitted in such a way that the probability of error is minimized.

Radio transmission requires the spectrum of the M possible signals to be centered on the frequency that the antenna system is designed for, that is, it requires the use of the proper carrier frequency. Schemes of radio digital modulation can be classified into four basic groups: *ASK*, *PSK*, *FSK*, and *QAM*, corresponding to amplitude, phase, frequency, and amplitude–phase combination modulation, respectively.

AM is not advisable for mobile radio links. The large amplitude variability caused by the radio-channel fading would mask the changes of amplitude due to the modulation. Although one could envisage an automatic gain control system in the receiver that would compensate the channel attenuation, it would complicate the system unnecessarily and it would require the continuous transmission of a pilot signal to estimate the exact value of the channel attenuation. The same argument applies to amplitude–phase combination (QAM) modulation. Thus, phase or frequency modulation, where amplitude does not carry information, are advisable for mobile communications.

According to the M-level phase–shift keying (*M-PSK*) modulation scheme, pulses containing a sine wave with a constant frequency are transmitted. Each possible transmitted symbol shifts the phase of the sine wave by a different amount. To spread the phase of the M symbols uniformly, the phase shifted is a multiple of $2\pi/M$ rad. Thus the received signal, $s_i(t)$, when the ith symbol is transmitted follows the expression

$$s_i(t) = \sqrt{\frac{2E}{T}}\, p(t) \cos\left(2\pi f_c t + i\,\frac{2\pi}{M}\right), \qquad i = 0, \ldots, M-1 \tag{5}$$

where E is the average received energy per symbol, T is the symbol time length, f_c is the carrier frequency, and $p(t)$ corresponds to the pulse conformation. The symbol length, T, is related to the bit duration, T_b, as $T = T_b \log_2 M$. Therefore, the symbol rate, R, is

$$R = \frac{R_b}{\log_2 M} \tag{6}$$

The quality of any digital transmission is measured by the BER. The exact value of the BER of an optimum *M-PSK* receiver cannot be calculated analytically for an arbitrary value of M; however, the exact expressions for *BPSK* ($M = 2$) and *QPSK* ($M = 4$) are

$$\mathrm{BER}_{\mathrm{BPSK}} = \frac{1}{2}\,\mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right) \tag{7}$$

$$\begin{aligned}\mathrm{BER}_{\mathrm{QPSK}} &= \mathrm{erfc}\left(\sqrt{\frac{E}{2N_0}}\right)\left[1 - \frac{1}{4}\,\mathrm{erfc}\left(\sqrt{\frac{E}{2N_0}}\right)\right] \\ &= \mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)\left[1 - \frac{1}{4}\,\mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)\right]\end{aligned} \tag{8}$$

where N_0 is the white Gaussian noise spectral power density, E_b is the average received energy per bit, and

$$\mathrm{erfc}(x) = \frac{2}{\sqrt{\pi}} \int_x^{\infty} e^{-t^2}\, dt \tag{9}$$

The average received energy per symbol (E), the average received energy per bit (E_b), and the average received power P are related by

$$E = E_b \log_2 M = PT = PT_b \log_2 M \tag{10}$$

The bandwidth required for the transmission of an *M-PSK* signal depends on the pulse, shape $p(t)$. The minimum *M-PSK* bandwidth is

$$B_{M\text{-PSK}} = \frac{R_b}{\log_2 M} \tag{11}$$

where again R_b is the bit transmission rate.

Equation (5) can be rewritten as

$$s_i(t) = \sqrt{\frac{2E}{T}} p(t) \sin\left(i\frac{2\pi}{M}\right) \sin(2\pi f_c t) - \sqrt{\frac{2E}{T}} p(t) \cos\left(i\frac{2\pi}{M}\right) \cos(2\pi f_c t) \tag{12}$$

It can be noted in the above equation that both in-phase (cos) and quadrature (sin) components are affected by the same envelope, *p(t)*. Therefore, transitions between symbols happen at the same time in both components. When the available bandwidth is small, *p(t)* is far from rectangular, and equal (synchronous) shaping for in-phase and quadrature components causes large *RF* amplitude variation. And amplitude variation represents a handicap for nonlinear power amplifiers, which have better power efficiency.

The straightforward solution for the problem of having amplitude variation in *QPSK* has been solved by staggering the envelope of the quadrature component by a time delay of *T*/2. This modulation is know as *offset QPSK* (*OQPSK*). The received signal, $S_i(t)$, when the *i*th symbol is transmitted follows the expression

$$\begin{aligned} s_i(t) = {} & \sqrt{\frac{2E}{T}} p(t) \sin\left(i\frac{2\pi}{M}\right) \sin(2\pi f_c t) \\ & - \sqrt{\frac{2E}{T}} p\left(t - \frac{T}{2}\right) \cos\left(i\frac{2\pi}{M}\right) \cos(2\pi f_c t) \end{aligned} \tag{13}$$

Besides having lower spurious emission when it passes through a nonlinear amplifier, *OQPSK* has the same BER and bandwidth efficiency as *QPSK*.

Another means to decrease the amplitude range of a *QPSK* signal under limited bandwidth conditions is the addition of a $\pi/4$ phase to every other transmitted symbol. In this way, the received signal $S_i(t)$ is

$$s_i(t) = \sqrt{\frac{2E}{T}} p(t) \cos\left(2\pi f_c t + i\frac{2\pi}{M} + \left\{ \begin{array}{c} 0 \\ \pi/4 \end{array} \right\}\right), \qquad i = 0, \ldots, 3 \tag{14}$$

where a 0 and a $\pi/4$ phase are alternately added to the carrier.

The second type of modulation with constant amplitude is the FSK. According to the FSK modulation scheme, pulses containing a sine wave with a different frequencies are transmitted. Each possible transmitted symbol shifts the carrier frequency by a different amount. Frequency modulation has the drawback of its larger bandwidth. To overcome this problem to the maximum extent possible, frequency modulation with minimum frequency deviation has been proposed. This modulation is known as *minimum shift keying* (*MSK*). *MSK* is defined by the binary case, $M = 2$. In this case, the received signal $S_i(t)$ when the *i*th symbol is transmitted follows the expression

$$s_i(t) = \sqrt{\frac{2E}{T}} p(t) \cos\left(2\pi f_c t \pm \frac{\pi}{2T} t + \theta\right) \tag{15}$$

where the $\pm$ signs mean a $\Delta f = 1/4T$ positive frequency shift when transmitting a 1 and the same negative frequency shift when transmitting a 0. The phase term θ is added to each symbol in order to guarantee a continuous phase. For this reason *MSK* modulation is also known as continuous-phase FSK (CPFSK).

In the *GMSK* modulation, the G stands for Gaussian, and it refers to the fact that the *MSK* modulation is filtered with a Gaussian frequency response before its transmission. The Gaussian frequency response

$$H(f) = \exp\left[-\frac{\ln 2}{2}\left(\frac{f}{B}\right)^2\right] \tag{16}$$

limits the 3 dB bandwidth of the transmitted signal to B. In order to increase the bandwidth efficiency, B is adjusted to satisfy $BT_{\mathrm{b}} < 1$. Usual values for B are such that $0.2 \le BT_{\mathrm{b}} \le 0.5$.

The exact value of the BER for an optimum *GMSK* receiver is very complex, but an upper bound accurate enough for any practical propose is given by

$$\mathrm{BER}_{\mathrm{GMSK}} = \frac{1}{2}\,\mathrm{erfc}\left(\sqrt{0.68\frac{E}{N_0}}\right) \tag{17}$$

A common objective of all the digital modulation schemes described above is to maximize the spectrum efficiency, which means that they try to minimize the occupied bandwidth for a given transmission bit rate. We introduce next a set of modulation techniques that try just the opposite, to maximize the bandwidth. These modulations are known under the generic name of *spread-spectrum* techniques.

Although it might be thought that the use of such modulation is an unnecessary waste of spectrum, it can be proved that in fact spread-spectrum techniques allow a more efficient usage of the spectrum. In spread spectrum, the spectrum of transmitted signals is artificially spread by a code. This code is different for each link in the same geographical area of radio network. Although we will not go into details in this article, the receivers can demodulate each link separately, attending to its assigned code, so the other signals in the same portion of the spectrum do not interfere. As is mentioned in the next subsection, and explained in detail in the article WIRELESS NETWORKS, the possibility of discriminating links according to codes allows the multiple-access technique known as CDMA.

The spreading codes used in spread-spectrum modulation consist of a sequence of bits with random appearance. To distinguish the code bits from the information bits, the former are called *chips*. Although auto- and cross-correlation functions of the spreading codes look like those of random sequences, they are generated deterministically by shift registers with specific feedback connections. Thus the receiver can generate an exact copy of the code used by the transmitter.

From a practical perspective, there are two kinds of spread-spectrum modulations: direct sequence (*DS*) and frequency hopping (*FH*). The former takes the transmitted bits and multiplies (exclusive-OR) them by the code sequence before modulation. The chip rate of the spreading code has to be faster than the bit rate in order to spread the spectrum. The latter changes the carrier frequency according to a pattern dictated by the spreading code.

Because of practical implementation issues, *DS* uses *BPSK* or *QPSK* modulators, and *FH* is used in conjunction with bit FSK modulation. The BER in the presence of additive white Gaussian noise is not altered from the expression corresponding to *BPSK*, *QPSK*, or FSK, respectively.

Multiple Access. As explained in detail in the article WIRELESS NETWORKS, multiple-access techniques can be classified into two generic types: *scheduling and contention*. The former assigns a fixed amount of resources (time slot, frequency range, or code) to each link, so there is no possible interference among links. The later allows several links to access the same resources, but establishes, some mechanisms to minimize the probability of collision (i.e. simultaneous use of the same resources) and to detect collision if it happens.

The simplest and most straightforward multiple-access method is known as *frequency-division multiple access* (*FDMA*). With *FDMA*, the available spectrum is divided into nonoverlapping slots in the frequency dimension (domain). These frequency slots, or channels, are then put into a pool and assigned to users on either a manual or an automated basis for the duration of their particular call. For example, a 150 kHz block of spectrum could be divided into six frequency slots, each 25 kHz wide. Such an arrangement would allow six simultaneous conversations to take place, each with its own carrier within its own frequency slot.

With *time-division multiple access* (*TDMA*), the available spectrum is divided into nonoverlapping time slots in the time dimension. These time slots are then put into a pool and assigned to users for the duration of a call. To continue the example given above, in a *TDMA* system the 150 kHz of spectrum would be divided into recurring groups (frames) of six time slots, and each time slot would carry a sequence of bits representing a portion of one of six simultaneous conversations.

A third access method is known as *code-division multiple access* (CDMA). As explained above, CDMA is both a modulation and an access technique that is based upon the spread-spectrum concept. In spread-spectrum systems, multiple conversations (up to some maximum) simultaneously share the available spectrum in both the time and frequency dimensions. Hence, in a CDMA system, the available spectrum is not channelized in frequency or time as in *FDMA* and *TDMA* systems, respectively. Instead, the individual conversations are distinguished through coding; that is, at the transmitter, each conversation channel is processed with a unique spreading code that is used to distribute the signal over the available bandwidth. The receiver uses the code to accept the energy associated with it. The other signals present are each identified by a different code and simply produce background noise. In this way, many conversations can be carried on simultaneously within the same block of spectrum.

Implementation of Duplex Channels. Duplex services are those that allow users in the network to send and receive information simultaneously. Obviously, this ability requires the assignment of two links to each user. Although both links serve the same user, they can be viewed as two independent links; therefore a multiple-access technique has to be chosen.

There are two options for the duplex implementation: time-division duplex (*TDD*) and frequency-division duplex (*FDD*).

Frequency-division duplexing (*FDD*) assigns two different frequency slots per conversation—one for transmitting and one for receiving. By separating the slots sufficiently in frequency (about 5% of the nominal *RF*), filters (say in the portable radio) can be used to prevent the transmitted information from interfering with the simultaneously received information.

Time-division duplexing (*TDD*) uses the fact that it is possible to share a single radio channel in time. *TDD* is only possible with digital transmission formats and digital modulation, and is very sensitive to timing. For this reason, *TDD* has been used only recently, for indoor or small area wireless applications where the physical coverage distances (and thus the radio propagation time delay) are much smaller than the many kilometers used in conventional cellular telephone systems.

*TDD*simplifies the frequency planning when the service is not symmetric, that is, when there are different bit rates for the go and return. In the other hand, *TDD*is not advisable for long links because of the long propagation-time delays. These require large time guard gaps between reception and transmission to avoid collision between them, which results in low spectral efficiency.

Practical Wireless Systems and Products

This part is devoted to describing in detail some practical wireless communications. Descriptions follow after an overview of their frequencies of operation:

Digital trunking systems operating in the UHF band in the 400 MHz region:

- *TETRA*: 390.0125 MHz to 392.9875 MHz and 420 MHz to 425 MHz (digital 12.5 MHz to 25 kHz channels)

Cordless telephone systems:

- CT2 cordless phones: 854 MHz to 870 MHz (864.1 MHz to 868.1 MHz to be reviewed in 2002)
- (Pan-European) cordless headphones: 863 MHz to 865 MHz.
- digital European cordless telephones (*DECT*): 1880 MHz to 1900 MHz

Cellular systems operating in a band near 800 MHz or 900 MHz, and, most recently, a band near 1.8 GHz or 1.9 GHz:

- Cell phones: *GSM* (124 200 kHz *TDMA* channels): uplink 915 MHz to 960 MHz (split –45 MHz), downlink 870 MHz to 915 MHz.
- *ETACS/TACS*: 917.0125 MHz to 949.9875 MHz (25 kHz channels, 12.5 kHz offsets), to be phased out by 2005.
- Extended *GSM* (*EGSM*): 925.2 MHz to 935 MHz.
- European Railways *GSM* system: 921 MHz to 925 MHz.
- In-flight digital phones (air–ground): 1,800.30 MHz to 1804.969 MHz (164 30.303 kHz channels; ground at –130 MHz).
- *PCN* mobile phones: 1805 MHz to 1876.5 MHz (split –95 MHz: 1710 MHz to 1781.5 MHz).
- Future *UMTS* (*IMT*-2000, third-generation mobile): 1900 MHz to 2025 MHz (with 2110 MHz to 2200 MHz) (−190 MHz).

Wireless *LAN*: There are three media that can be used for transmission over wireless *LAN*s: infrared, radio frequency and microwave. The industrial, scientific, and medical (*ISM*) frequency bands are 902 MHz to 928 MHz, 2.4 GHz to 2.4853 GHz, and 5.725 GHz to 5.85 GHz.

Cellular Systems. Concepts behind cellular systems are presented in the section "Fundamentals of Wireless Communication" above. This subsection briefly presents the commercially available networks and enumerates the services they provide and the main parameters and differences between them.

During the 1970s and 1980s several analog cellular system were deployed with successful acceptance by the public: *AMPS*, *TACS*, NMT, and others. Their low spectral efficiency severely limited the network capacity. Moreover, the lack of a unified standard prevented the users from roaming through different countries.

At the beginning of the 1980s initiatives were taken to develop a new cellular standard with the following goals: digital transmission, high spectral efficiency, high network capacity, international roaming, provision of voice communications and low-bit-rate data (fax), small-size terminals, and digital signaling able to provided advanced services, among others.

The European initiative to provide a pan-European second generation was very successful, coming up with the *GSM* standard. The US initiative produced the IS-54, which provides similar services to the *GSM*. In the 1990s the United States also produced the IS-95, the first cellular system to employ spread-spectrum modulation.

GSM. Although *GSM* is now available in the frequency range of 1800 MHz, known as *DCS*-1800, *GSM*'s primary band is 935 MHz to 960 MHz for the downlink (base to mobile), and 890 MHz to 915 MHz for the uplink (mobile to base). The duplex method is *FDD* with a constant go- and return-frequency separation of 45 MHz.

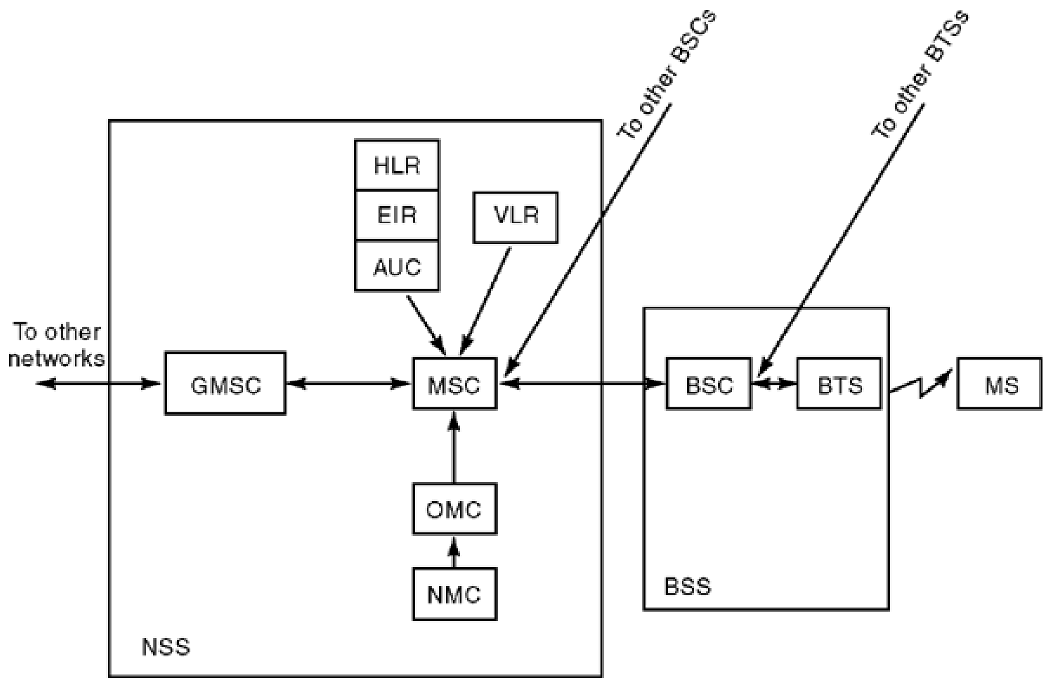

Fig. 6. *GSM* network structure. *GSM* networks consist of three main subsystems: mobile station (*MS*), base-station subsystem (*BSS*), and network switching subsystem (*NSS*).

The multiple-access technique used by *GSM* is *TDMA* plus *FDMA*. The available spectrum is divided into frequency channels separated 200 kHz. Each of this frequency channels is divided into eight time slots of duration 0.577 ms each, which makes frames of length 4.615 ms. Both traffic and signaling channels are mapped into these *TDMA–FDMA* physical channels according to an established pattern.

The transmitted raw (after channel coding) bit rate is 270.83 kbit/s for each carrier, yielding eight *TDMA* channels. The modulation is *GMSK* with a bandwidth such that $BT_b = 0.3$, with $1/T_b = 270.83$ kbit/s.

The signaling of the *GSM* is an ISDN version adapted to the mobility and radio environment. Figure 6 shows the *GSM* network structure.

GSM network is divided into three large segments: the mobile station (*MS*), base-station subsystem (*BSS*), and network switching subsystem (*NSS*).

The *BSS* consists of the base transceiver station (*BTS*) and the base-station controller (*BSC*). The *BTS* performs the physical-layer-related functions, while the *BSC* performs the higher control functions. One *BSC* may control several (up to 10) *BTS*s.

The *MSC* routes the calls towards the *BSC* closest to the *MS*. It gets the location of a particular *MS* from the home location register (HLR). The HLR is updated so it always contains the latest location of each mobile in the network. It also contains information about the user's profile. When a *MS* roams into a different network, the information in the HLR regarding its users is copied to the visitors location register (VLR) of the host network.

The equipment identity register (*EIR*) and authentication center (*AUC*) are responsible for the network security and privacy. The operation management center (*OMC*) and network management center (*NMC*) take care of resource management and maintenance. Finally, the gateway mobile switching center (*GMSC*) connects the *NSS* to other networks.

The carrier services provided by *GSM* are voice and low-speed data with a maximum bit rate of 9.6 kbit/s. Voice is digitally coded at 13 kbit/s and 6.5 kbit/s. The former is known as *full rate* (FR), and the latter as *half rate* (HR). Although HR provides worse voice quality, it doubles the capacity of the network.

Additional services provided by the *GSM* network are calling-line identification, call forwarding, call waiting, call holding, multiparty, closed user group, advice of charge, reverse charging, user-to-user signaling, and barring outcoming and ingoing calls.

IS-95. In 1993 the Telecommunication Industry Association (TIA) produced the standard for the first CDMA cellular system under the name of IS-95. The use of spread-spectrum techniques promised a larger system capacity.

IS-95 uses a voice codec with variable bit rate. The IS-95 vocoder generates a bit stream at a 9600 bit/s rate, but it detects the silence periods and reduces the bit rate during them to 1200 bit/s. This variable transmission rate further increases the network capacity.

Although they follow different spreading schemes, both downlink, and uplink channels are spread by a pseudonoise code of 1.2288 Mchip/s, such that the channel bandwidth is 1.25 MHz. IS-95 uses *FDD* duplex with 45 MHz separation between the go and return. The frequency allocation is 869 MHz to 894 MHz for the downlink, and 824 MHz to 849 MHz for the uplink.

In addition to voice service, IS-95 also provides data transmission service at 9.6 kbit/s, 14.4 kbit/s, and 28.8 kbit/s.

Cordless Systems. In contrast to cellular radio, cordless systems primarily offer an access technology rather than fully specified networks. Cordless terminals generally transmit at lower power than cellular, using microcells with a range of a few hundred meters. In high-density environments, smaller cells are used, providing higher traffic densities than those obtained by cellular standards.

Services, Features, and Standards. Voice was the only service provided by early analog cordless systems. However, cordless technology has evolved in the past years, and the cordless data market has become increasingly important as palmtop computers have emerged. The *DECT* standard has been specified to provide both voice and data services.

The potential of cordless access to permit user roaming between business and domestic applications was one of the key drivers of digital cordless standards. Also, the opportunity to offer a public cordless access, generically known as Telepoint, was recognized. Thus, modern cordless telecommunications standards provide the following applications:

Domestic Applications. The earliest application of cordless communication systems was the residential cordless telephone, with a very simple base station giving service to the home and allowing the use of the telephone throughout its small area. The domestic market was dominated until recently by analog products because of the higher price of digital products.

Business Applications. The potential business application of cordless telecommunications was one of the early drivers for the development of the technology. Although early products focused on telepoint, last years have shown a significant progress of the wireless PABX (*WPABX*).

Wireless access in the PABX environment enables the user to obtain the wide range of PABX features while roaming across a business area. In new offices, the expense and delay of deploying a PABX can be avoided because there is no need of wiring up.

ISDN as well as voice capabilities supported by the wired PABX can also be supported by *WPABX*. Also, mobility management functions must be incorporated so as to authenticate and locate users and route incoming calls.

Telepoint. *Telepoint*, also called *public access*, is a short-range radio system whereby a user with a portable handset can gain access to teleservices via a public fixed network like *PSTN*. CT2 was the first available telepoint standard. *DECT* has also been specified to support public access.

Although it is a European concept, the initial success of telepoint has been in Asia, where early systems, combined with paging, have taken off in spite of the absence of the complementary domestic and business products.

In contrast, early telepoint experiments in Europe failed to attract a significant number of users, probably because of the lack of standards and products at that time. New experiments have been launched recently (Rabbit launched by Hutchison Telecom in UK, Bibop launched by France Telecom, Birdie launched by DBP Telekom in Germany, Greenpoint launched by PTT Netherlands, and Pointer launched by Telecom Finland) that are showing greater success.

Telepoint licenses have also been issued in Canada, and trials are operational in the United States under experimental FCC licenses. However, the implementation of *PCS*s calls into question the utility of telepoint.

Wireless access. Although the preceding three applications have led to the development of cordless standards and products, a fourth application has emerged with a new potential market for cordless telecommunications: the wireless local loop (*WLL*).

Telecommunication services that used to be provided by wire from an exchange to the user's premises are alternatively using cost-effective wireless standards. This use has been encouraged by the privatization of telecommunications monopolies in Europe, leading to a number of potential service providers that do not own the wired infrastructure.

*WLL*s may be provided by different wireless technologies, with cordless standards being one likely choice because of their low cost.

The main cordless standards, providing some of the above applications, are the following:

- CT0 (Europe) and CT1 (Europe): first-generation analog systems
- CT2 (Europe): digital cordless *FDMA* standard using *TDD*
- PACS (United States), PHS (Japan), and *DECT* (Europe): latest digital cordless telecommunication systems

Since *DECT* corresponds to the latest stage in the evolution of cordless systems, it will be explained more deeply in the following sub-subsection.

Digital European Cordless Telecommunications Standard. *DECT* is based on a microcellular radio communication system that provides low-power radio (cordless) access between portable parts (*PP*s) and fixed parts (*FP*s) at ranges up to a few hundred meters. The basic technical characteristics are as follows:

- Frequency band: 1880 MHz to 1900 MHz
- Number of carriers: 10
- Carrier spacing: 1.728 MHz
- Peak transmit power: 250 mW
- Carrier multiplex: *TDMA*; 24 slots per frame
- Frame length: 10 ms
- Basic duplexing: *TDD*using two slots on the same *RF* carrier
- Gross bit rate: 1152 kbit/s
- Net channel rates: 32 kbit/s B-field (traffic) per slot; 6.4 kbit/s A-field (control and signaling) per slot

A connection is provided by transmitting bursts of data in the defined time slots. These may be used to provide simplex or duplex communications. Duplex operation uses a pair of evenly spaced slots, one for transmission and one for reception. The simplest duplex service uses a single pair of time slots to provide a 32 kbit/s digital information channel capable of carrying coded speech or other low-rate digital data. Higher data rates are achieved by using more time slots in the *TDMA* structure, and a lower data rate may be achieved by using half-slot data bursts.

DECT is able to support a number of alternative system configurations ranging from single-cell equipment (e.g. domestic *FP*s) to large multiple-cell installations (e.g. large business *WPABX*s). Direct communication between *PP*s is also supported.

The protocols are designed to support uncoordinated system installation, even where the systems coexist in the same physical location. Efficient sharing of the radio spectrum (of the physical channels) is achieved using a well-designed mechanism of dynamic channel selection.

In addition, the *DECT* protocols provide two internal mechanisms to support rapid handover of calls in progress (both intracell and intercell handover are supported). These mechanisms allow a high quality of service to be maintained where the mobility of the PP requires transparent reconnection to another FP or where a new physical channel is required in response to disturbances in the radio environment.

Detailed information about the *DECT* standard may be found in Refs. 8,9,10,11,12,13,14 to 15.

Trunking Systems. When we refer to a trunked system, or, equivalently a private mobile radio (*PMR*) system, we mean a system where all the users share a pool with all the available channels. If a user wants to make a call, the system allocates one of the idle channels.

Features. The most important features of *PMR* systems are related to the peculiarities of their users.

(1) *PMR* users, such as the emergency services (police, fire, etc.), handle incidents where calls are typically very short and speed of communication is vital. In contrast to cellular systems, where it takes several seconds to establish a call, the access to the *PMR* network takes tenths of a second.
(2) *PMR* services allow group and broadcast calls.
(3) *PMR* codecs are designed to provide good-quality speech in noisy environments.
(4) New digital *PMR* systems can provide a user "bandwidth on demand" on a dynamic basis.
(5) One powerful operational mode of *PMR* systems (not provided by cellular systems) is the so-called *direct mode* (*DMO*): the ability for radio handsets to communicate directly with each other without using the network infrastructure.

Standards. For completeness, we will summarize some of the digital technologies available.

The first one is the trans-European trunked radio (*TETRA*). It is a radio open standard based on *TDMA* defined by the European Telecommunications Standards Institute (*ETSI*). The *TETRA* standard contains high functionality for emergency services, and it is also very well suited for commercial trunked radio users.

There are other systems, all of which are proprietary and aimed at similar users and markets as *TETRA*. The major ones are iDEN and ASTRO from Motorola, Aegis and PRISM from Ericsson, and Tetrapol from Matra. However, the majority of users have by now seen the benefits of standardization and are moving away from these proprietary systems.

In response, the manufacturers with proprietary specifications have been attempting to have them adopted as standards by national or international standardization bodies. The most active has been Matra, which developed a digital system called MatraCom (MC9600), a 12.5 to 10 kHz *FDMA* system initially sold to the French gendarmerie as Rubis. Later derivatives were sold to other customers under various other names such as Acropol, Pegas, and Nexus.

In order to promote these proprietary technologies as a generic, Matra created the name Tetrapol and formed the so-called Tetrapol Forum. Also, as an active *ETSI* member, Matra has twice tried to have its technology selected by *ETSI* for the *TETRA* standard. Having failed on both occasions, the company has—this time as the Tetrapol Forum—declared its intent to have the Tetrapol specification considered for adoption as an *ETSI* deliverable under a so-called publicly available standard (PAS) procedure.

Trans-European Trunked Radio—the European Standard for Digital Private Mobile Radio. *TETRA* is a radio open standard based on *TDMA*. In contrast to *GSM*, where manufacturers and operators have determined the specifications, the users, especially emergency service users, have contributed strongly in the

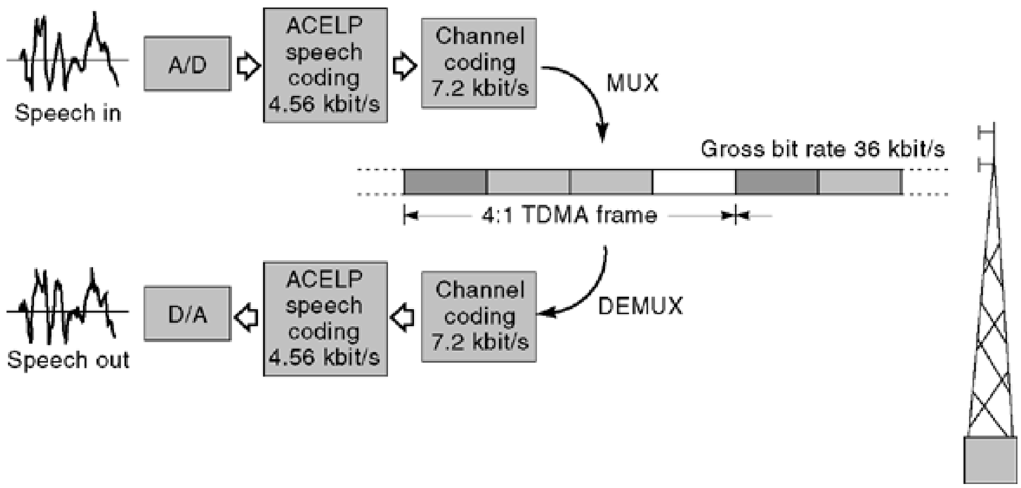

Fig. 7. Speech coding and multiplexing in *TETRA*. Digitized speech is coded using an ACELP coder, protected by introducing redundancy, and inserted in a *TDMA* frame,

creation of the standard. As a result, the *TETRA* standard contains high functionality for emergency services and is also very well suited for commercial trunked radio users.

Technology.

Physical layer.Physical layer The outline radio characteristics are as follows:

- Modulation: $\pi/4$ *DQPSK*
- Transmission rate: 36 kbit/s
- Duplex spacing: 10 MHz (45 MHz in 900 MHz band)
- *RF* carrier spacing: 25 kHz

Frequency Bands.Frequency Bands *TETRA* has been designed to work in the frequency range from VHF (150 MHz) to UHF (900 MHz). In particular, standardization bodies have released bands in the frequency range 380 MHz to 400 MHz for public-safety users.

Speech Coding.Speech Coding The *TETRA* codec was designed to provide good-quality speech in harsh environments and voice quality superior to *GSM* in conditions of poor signal. In *TETRA* systems speech is digitized and coded using an ACELP speech coder. In order to protect the transmitted information, a channel coder introduces additional redundancy. The resulting bit rate is 7.2 kbit/s. *TETRA* has the capability of using four *TDMA* slots, which allows managing four voice communications on the same carrier (see Fig. 7).

Data Transmission.Data Transmission *TETRA* can provide a user, on a dynamic basis, with up to four *TDMA* slots in a single communication channel, effectively “bandwidth on demand.” This applies to packet-mode and circuit-mode data, with data rates of up to 28 kbit/s.

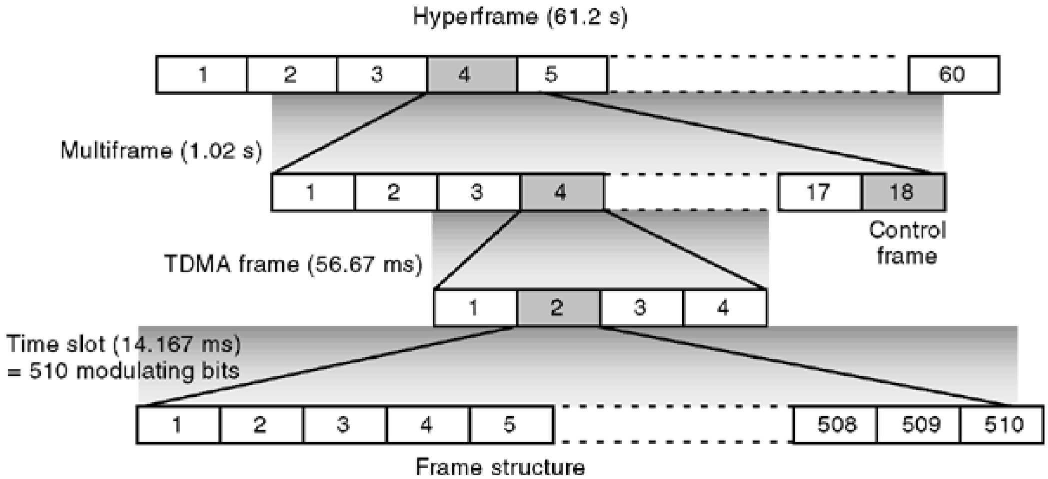

Fig. 8. Hierarchical structure of frames in *TETRA*.

TDMA Frame Structure.*TDMA* Frame Structure The *TETRA* frame structure has four slots per *TDMA* frame. This is further organized as 18 *TDMA* frames per multiframe, of which one frame per multiframe is always used for control signaling. In addition, there is a hyperframe imposed above the multiframe structure. This is for long repeat frame purposes such as encryption synchronization. Each time slot has 510 (modulating) bits duration and has the basic structure shown in Fig. 8.

Spectrum Efficiency.Spectrum Efficiency *TETRA* is one the most frequency-efficient standards for mobile communication, offering 6.25 kHz per channel, voice or data. This means that *TETRA* is currently four times more spectrum-efficient than *GSM*, and twice as efficient when the half-rate codec is implemented. In comparison with other *PMR* systems, *TETRA* can accommodate up to four times more users.

Features. In addition to the features of public cellular systems, *TETRA* offers a series of additional services. The most important are:

- Group Call and Group Communication A number of users share the same channel. This enables many users to cooperate on a task or to monitor the activities of other members of the group.
- Fast Call Setup *TETRA* makes it possible to set up calls in less than 0.3 s.
- Direct Mode Direct mode allows terminals to communicate directly with each other independently of the radio infrastructure, or in areas with no coverage.
- Mobile Terminal Used as Repeater A mobile terminal can be used as repeater and in this way extend the coverage of the *TETRA* network.
- Encryption *TETRA* offers a very high degree of encryption of voice, data, signaling, and user identity. *TETRA* defines two methods of encryption: air interface encryption and an optional end-to-end encryption for the most critical applications.
- Broadcast Transmission of messages to all users of the network.
- Priority Possibility of priority in up to eight levels. Furthermore, calls with high priority can overrule calls with low priority if no idle channels are available.

Typical users of TETRA.

(1) Public safety and public security (police, fire, ambulance, customs, etc.)
(2) Transport (airlines, ports, taxis, buses, railways, etc.)
(3) Utilities (gas, electricity, water, oil, etc.)
(4) Industry (manufacturing, plant, distribution, etc.)
(5) Nonemergency authorities (government, public health, environment protection, etc.)

Paging Systems. A radio paging system is a wireless messaging system that allows continuous accessibility to someone away from the wired communications network. In its most basic form, the person on the move carries a palm-sized device (the pager), which has an identification number. The calling party inputs this number, usually through the public telephone network, to the paging systems which then signals the pager to alert the called party.

Pager types and existing standards are described in this sub-section.

Pager types. The following describes several types of pagers that are commercially available.

- Tone only Pager alerts user; user takes predetermined action, such as calling a phone number.
- Numeric Pager Display Pager alerts user and displays numeric message; user calls phone number displayed.
- Alphanumeric and Ideographic Display Pager alerts user and displays text message; user can then take necessary action.
- Tone and Voice Pager alerts user, then delivers short (10 s to 20 s) voice message; user can then take necessary action.
- Stored Voice Pager silently alerts and stores voice message for recall at user's convenience.

Standards. Different manufacturers have developed an array of techniques to forward the required information to remote pagers. These techniques, known as encoding formats, define the means of representing the information-carrying data elements of the protocol as well as the interpretation of the overall data content.

In many cases, encoding formats send additional data, known as *error detection and correction codes*, that are capable of detecting and recovering incorrectly received data. With error correction, pager receipt reliability is improved dramatically. Both analog and digital transmission techniques are used to transfer information to pagers.

Most paging formats are manufacturer-specific and often proprietary. There are a few paging protocols that have been developed and put into the public domain so that many different manufacturers may produce compatible pagers. Among these public-domain protocols are POCSAG, the Swedish format (MBS), the radio data system (*RDS*) format, and the European Radio Message System (*ERMES*) format. All of these formats were developed in Europe.

European Radio Message Standard Format. *ERMES* is a standard that was developed by a subcommittee of the *ETSI*. The *ERMES* digital encoding format supports tone-only, numeric, and alphanumeric paging in addition to data transfer capabilities. The format operates at 6250 bits per second. *ERMES* pagers operate on multiple frequencies, scanning for the best frequency for optimum reception.

The paging format uses a modulation mechanism known as *four-level pulse-amplitude-modulated FM*. In this mechanism, two binary bits of information are transmitted simultaneously through the transmission of one of four signaling frequencies. One set of frequencies is interpreted as the two binary bits 00, another as 01, another as 10, and the last as 11. Therefore, with frequency transitions at the rate of 3125 per second, 6250 bits of information may be transferred.

Under the *ERMES* protocol, every hour is broken up into 60 cycles (cycles 0 to 59), each 1 min in duration. Each cycle is divided into five equal subsequences of 12 s each (subsequences 0 to 5). Finally, each 12 s period is divided into 16 separate batches (batches A to P). A batch contains separation partitions of information known as the synchronization partition, the system information partition, the address partition, and the message partition.

Within each batch, the address partition contains the first 18 bits (the initial address) of the unique pager number, a 35-bit address known as the *radio identity code* (*RIC*), ordered in descending order. This technique allows a pager to determine quickly if its unique address is not part of this batch so that it may return to battery-saving mode. All pagers whose addresses are larger than the initial address can return immediately to battery-saving mode. Messages are transmitted directly after the address partition. The large address field (35 bits) accommodates a global address scheme that can support hundreds of millions of pagers.

Wireless Local Area Networks. *WLAN*s constitute a development of the small-area data communication networks used to interconnect computers and peripherals, in which wired transmission is replaced by radio communications. They find their primary use inside buildings or in settings such as university campuses.

More information about *WLAN*s can be found in the article WIRELESS NETWORKS in this encyclopedia. Here, the main features and the most commonly used standard are briefly described.

Services, features, and standards. The existing service scenarios for *WLAN*s are mainly private local networks in workplaces, campuses, or public areas such as airports. Universal access points in homes and nomadic access in public places can be provided by either *WLAN* or cordless standards.

The *WLAN* market currently aims at four categories of applications: health care, factory floors, banking, and educational institutions.

Some examples of *WLAN* standards are:

- *IEEE* 802.11 (United States) Part of the *IEEE* 802 family of *LAN* standards, using either infrared (IR), direct-sequence spread spectrum (DSSS), or frequency-hopping spread spectrum (FHSS).
- HIPERLAN (Europe) A family of standards in preparation, whose first outcome has been HIPERLAN 1 (using *GMSK*); forthcoming versions will use multicarrier techniques.

IEEE 802.11, the most commonly used *WLAN* standard nowadays, is described more deeply in the following sub-subsection.

IEEE 802.11 Standard. The *IEEE* 802.11 standard for *WLAN* (16) is emerging as a mature standard presenting a well-defined technology that is being adopted by the manufacturers and accepted by users. It specifies data rates up to 2 Mbit/s using spread-spectrum technology (direct sequence or frequency hopping) in the 2.4 GHz band (extension to the 5 GHz band is in progress). Alternatively, infrared technology can be used at 850 nm to 950 nm.

Multiple access is based on *carrier sense multiple access with collision avoidance* (*CSMA/CA*), a contention-based scheme suitable for asynchronous applications. The standard also supports a contention-free prioritized point coordination function (*PFC*) mechanism for time-bounded isochronous applications.

Two network topologies are considered: infrastructure-based and ad hoc.

Advanced Technologies and Systems

Wireless communications are in continuous evolution (17). We aim to introduce here some of the technologies and systems that today are yet to be fully developed and will constitute the wireless products of the next decades.

Advanced Wireless Technologies. Some of the advanced technologies being developed for wireless communications are briefly described in the following.

Wideband Code-Division Multiple Access. Wideband CDMA (18) has emerged as the mainstream air interface solution for the third-generation networks. In Europe, Japan, Korea, and the United States wideband CDMA systems are currently being standardized.

Wideband CDMA has a bandwidth of 5 MHz or more (multiples of 5 MHz). The nominal bandwidth for all third-generation proposals is 5 MHz. There are several reasons for choosing this bandwidth. First, data rates of 144 kbit/s and 384 kbit/s, the main targets of third-generation systems, are achievable within 5 MHz bandwidth with a reasonable capacity. Even a 2 Mbit/s peak rate can be provided under limited conditions. Second, the lack of spectrum calls for reasonably small minimum spectrum allocation, especially if the system has to be deployed within the existing frequency bands occupied already by second-generation systems. Third, the 5 MHz bandwidth can resolve (separate) more paths than narrower bandwidths, increasing diversity and thus improving performance. Larger bandwidths of 10 MHz, 15 MHz, and 20 MHz have been proposed to support higher data rates more effectively.

Several wideband CDMA proposals have been made for third-generation wireless systems. They can be characterized by the following new advanced properties:

- Provision of multirate services
- Packet data
- Complex spreading
- A coherent uplink using a user-dedicated pilot
- Additional pilot channel in the downlink for beamforming
- Seamless interfrequency handover
- Fast power control in the downlink
- Optional multiuser detection

The third-generation air interface standardization for the schemes based on CDMA seems to be focusing on two main types of wideband CDMA: network-asynchronous and -synchronous. In network-asynchronous schemes the base stations are not synchronized, while in network-synchronous schemes the base stations are synchronized to each other within a few microseconds.

Orthogonal Frequency-Division Multiplexing. Orthogonal frequency-division multiplexing (OFDM) is a multicarrier modulation scheme (19) that has found many recent wireless applications due to its ability to combat impulsive noise and multipath effects and make fuller use of the available system bandwidth. It has been adopted for the European digital audio broadcasting (*DAB*) (20) and digital video terrestrial broadcasting (*DVB*-T) (21) standards, and it is under study for new wireless *LAN* generations (HIPERLAN: high-performance radio *LAN*).

In an OFDM-based system, the spectrum associated to each elemental data is a small portion of the total bandwidth B, which is divided into N subchannels. Each of them is modulated with one information symbol, and they are all multiplexed in frequency. If T represents the OFDM symbol duration, the N subcarriers are separated by $\Delta f = 1/T$ and thus placed at the frequencies

$$f_k = f_0 + k/T, \qquad k = 0, 1, \ldots, N-1 \tag{18}$$

One of the main advantages of OFDM is the possibility of an easy implementation using a fast Fourier transform (FFT) algorithm (22). Among its weaknesses, sensitivity to phase noise, frequency offsets, and nonlinear effects must be mentioned.

Combination of Multicarrier and Code-Division Multiple-Access Techniques. Some ways of combining OFDM and CDMA, aiming to obtain the advantages of both techniques, have been suggested recently. Depending on how they are combined, three different schemes have been developed: MC-CDMA, multicarrier

DS-CDMA, and MT-CDMA. They constitute different tradeoffs between transmitter–receiver complexity, spectral efficiency, and bit error rate (23).

Multiuser Detection and Blind Detection. The continued expansion of the mobile cellular industry is leading to a need for increasing density of users within any given area without any corresponding increase in the frequency allocation. This heightens the probability of users and/or services interfering with one another, implying a need for new schemes to provide sophisticated interference cancellation approaches for a more efficient utilization of spectrum and space at any time. Techniques known as *multiuser detection* (*MUD*; also called joint detection, interference cancellation, or source separation) consider multiple access interference (*MAI*) not as a simple interference to be suppressed but as a composite signal that can be processed and separated.

The optimum *MUD* algorithm (24), based on a maximum likelihood (*ML*) criterion, has a complexity growing exponentially with the number of users. For this reason, many different suboptimal schemes are being developed.

Some of these algorithms do not need the introduction of reference sequences as an aid in the detection process, in what is known as *blind detection* (25). The avoidance of overheads implies a more efficient use of the system capacity.

Smart Antennas. Smart antennas (or adaptive antennas) combine multiple antenna elements with signal-processing capability in order to optimize the radiation and/or reception pattern automatically in response to the signal environment. This ability to adapt to a changing interference environment can dramatically increase the performance characteristics and capacity of a wireless system.

An overview of smart-antenna applications can be found in Ref. 26.

Future Systems. Third-generation cellular systems and the provision of multimedia services over wireless networks represent the trends in wireless communications for the next years. Migration from actual systems to the third generation is a related topic under current study.

IMT-2000 and the Universal Mobile Telecommunications System. *IMT*-2000 is an initiative of the *ITU* to provide wireless access to the global telecommunication infrastructure through both satellite and terrestrial systems, serving fixed and mobile users in public and private networks. It is being developed on the basis of the *family-of-systems* concept, defined as a federation of systems providing *IMT*-2000 service capabilities to users of all family members in a global-roaming offering.

The radio transmission technology (*RTT*) to be used in *IMT*-2000 was standardized in November 1999. It is mainly based on wideband CDMA technology. The main objectives for the *IMT*-2000 air interface can be summarized as:

- Full coverage and mobility for 144 kbit/s, and preferably for 384 kbit/s
- Limited coverage and mobility for 2 Mbit/s
- High spectrum efficiency compared to existing systems
- High flexibility to introduce new services.

The Universal Mobile Telecommunications System (*UMTS*)is a part of the *IMT*-2000 vision of a global family of third-generation mobile communications systems, being developed by *ETSI*. The *UMTS* radio interface has been named *UTRA* (*UMTS* Terrestrial Radio Access), and it offers two possibilities: *FDD* and *TDD*, both using wideband CDMA.

Many organizations that are currently developing standards for *IMT*-2000 have agreed to cooperate in the production of technical specifications for a third-generation mobile system based on the evolved *GSM* core networks and *UTRA* (both *FDD* and *TDD*modes). The project is called the Third Generation Partnership Project and may be known by the acronym 3*GPP*.

GSM and Enhanced Data for GSM Evolution. Enhanced Data for *GSM* Evolution (EDGE), which is currently being standardized within *ETSI*, represents the final evolution of data communications within the

GSM standard. It uses a different modulation scheme to enable data throughput speeds of up to 384 kbit/s using existing *GSM* infrastructure.

Third-generation wireless systems will provide high-speed wireless access to wideband multimedia services where spectrum and licenses are made available. Today's *GSM* operators have two (nonexclusive) options for evolving their networks to third-generation wideband multimedia services: use EDGE in existing radio spectrum and in small amounts of new spectrum, or use wideband CDMA in new 2 GHz spectrum or in large amounts of existing spectrum.

Mobile Satellite Systems. Due to satellites' global coverage, mobile satellite services (*MSS*) systems will fill the gaps in areas where cellular terrestrial systems are physically or economically impractical to implement.

Satellite systems may be classified according to the orbital altitude of the satellites being used: geostationary (*GEO*) satellites at an altitude of approximately 36,000 km, low Earth orbit (*LEO*) satellites at altitudes on the order of 1000 km, medium Earth orbit (*MEO*) satellites at intermediate altitudes between *LEO* and GEO, and highly elliptical orbit satellites (*HEOS*) with elliptical orbits of widely varying altitudes.

MSS systems have been implemented and are being designed for use with any of these types of satellites. One advantage of using GEOs is that global coverage (excluding polar latitudes) can be achieved with only three satellites. *LEO*s minimize the required transmitted power and propagation delay.

More details may be found in Ref. 27.

Multimedia Wireless Communications and HIPERLAN. In response to growing market pressure for low-cost multimedia wireless communications, *ETSI* has established a new standardization project for broadband radio access networks (*BRAN*s). This project will develop new standards for a new generation of transparent radio access networks for both licensed and license-exempt applications. These networks will carry existing services such as voice and ISDN as well as providing the transport mechanisms for future services.

The *BRAN* project is developing standards for three types of network. HIPERLAN (operating at 25 Mbit/s) will provide short-range and cordless services; it can be used indoors and campus-wide, and is license-exempt. HIPERACCESS (also operating at 25 Mbit/s) works at long range (up to 5 km), may be licensed or license-exempt, and has applications for residential and small- or medium-sized business users. It will serve as a tool for operators to enable them to use broadband radio independently of any infrastructure, which will encourage new players on the field as well as the introduction of new services for which existing infrastructures are unsuitable. HIPERLINK (155 Mbit/s) is license-exempt and serves to interconnect HIPERACCESS and HIPERLAN. The HIPERLAN type 1 functional specification (20 Mbit/s for asynchronous services in 23.5 MHz bandwidth) was completed in 1996, and it is being extended to HIPERLAN type 2 (25 Mbit/s to 54 Mbit/s in 20 MHz bandwidth using OFDM). HIPERLAN type 2 will provide compatibility with ethernet, IP, and ATM infrastructures.

The project is looking at ways of complementing the *UMTS*, and discussions have begun with *ETSI* Project SMG, as well as with other forums, including the Institute of Electrical and Electronic Engineers (*IEEE*).

Migration from Second-to Third-Generation Systems. Third-generation systems will evolve from second-generation in a way sensitive to the needs of customers, operators, and manufacturers by ensuring that new systems can interwork with the radio access and fixed/mobile networks of earlier generations of equipment. Since the scope of future mobile communications encompasses multimedia, far beyond the capabilities of second-generation wireless communication systems, the objective is to progressively extend mobile communications to include multimedia and high-performance services.

This means that new systems must be inherently designed to support migration from existing systems, and harmonization between Wireless CDMA and *TDMA* access systems is under current study. A degree of harmonization can be achieved between them by adopting multiples of a common (*GSM*) carrier spacing and reference clock rate. In addition, protocol stack harmony is possible after differentiating between mode-specific and common elements.

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