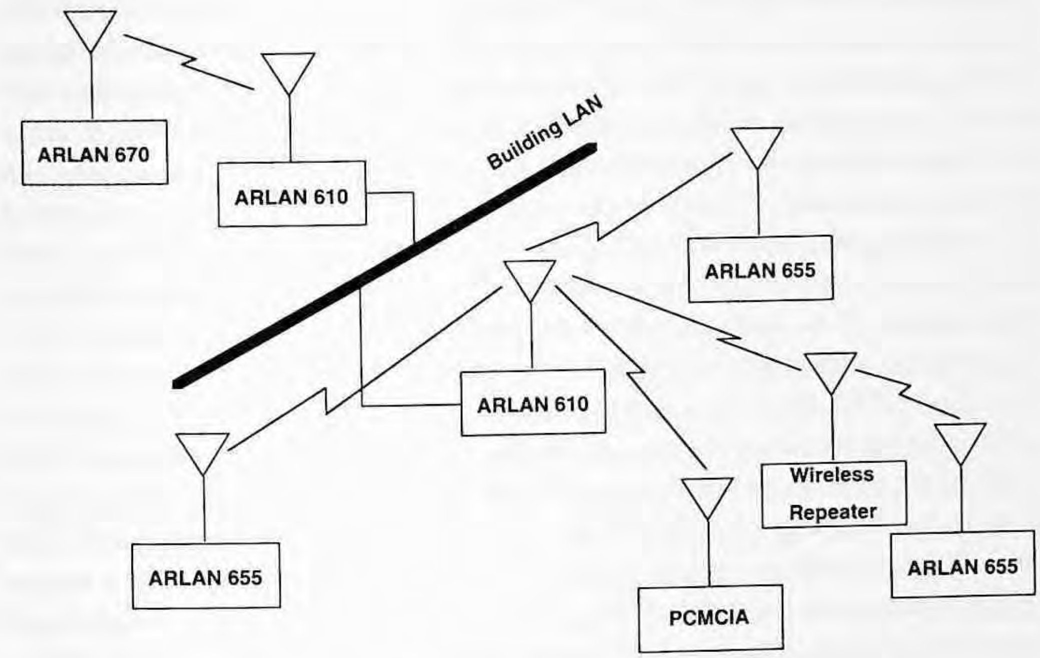

Figure 10.1 Access application for WLAN.

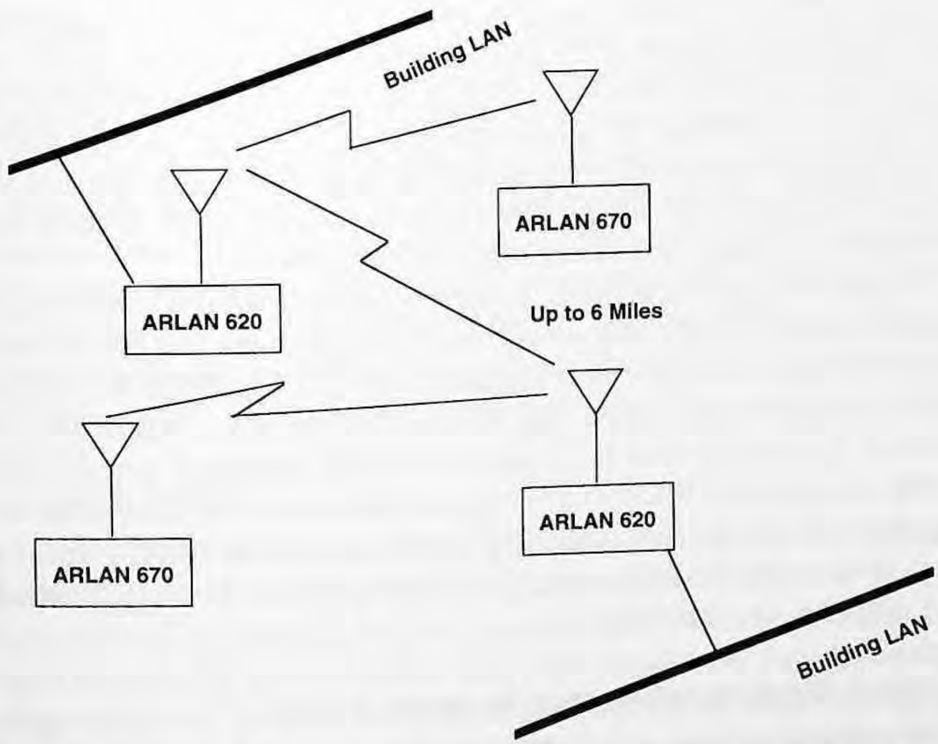

Figure 10.2 Access application for WLAN between two buildings with line of sight.

mechanism. Once it is determined that the network is free, the end station ramps up to full power and sends a preamble (a standard signaling message) to the access point. The *preamble* is a repeated bit pattern followed by a special bit sequence. It allows the access point to lock on to the signal before the data are sent. After the link is established, the end station sends address and protocol information. The header is followed by the data, which are transmitted at the on-air data rate. After the error check word is sent, the end station listens for acknowledgment from the destination. If no acknowledgment is received, the data are resent. The sequence is repeated until all the data have been sent and acknowledged.

The IEEE 802.11 committee has specified that data rates for wireless systems must be either 1 or 2 Mbps. Either the user chooses the rate or the system selects the best one according to the conditions. The on-air data rate includes message headers, retransmissions, and latency (the time between when a network station begins to seek access to a transmission channel and when that access is granted). Header overhead and retransmissions primarily affect performance of large data transfers. Whereas, latency has the greatest effect on short, bursty data transfers, because the latency involved in setting up a transmission introduces more delay than the transmission of message overhead or retransmissions. Therefore, throughput on a WLAN is lower for short messages than for longer messages. The actual throughput of an IEEE 802.11 system on an on-air data rate of 2 Mbps is about 1–1.5 Mbps for long messages and 0.5–1.0 Mbps for short messages. Throughput is also affected by the range of the system. In a typical office environment, the range of an IEEE 802.11 WLAN is 200–300 feet, which is sufficient to cover most partitioned areas and an outside rim of walled offices.

Sensitivity of the system is crucial because signal power can be affected drastically by obstacles. Sensitivity figures are the smallest amounts of received power that the radio can use. The IEEE 802.11 standard requires a sensitivity of less than –80 dBm. One issue that is not addressed by the standard is roaming capability. Roaming is made possible with overlapping WLAN cells in a configuration similar to that used for analog cellular phones. Roaming is considered to be part of the application- or driver-level technology, so vendors will be likely to resort different schemes for achieving it.

10.3.2 Wireless Information Networks Forum

The Wireless Information Networks Forum (WINForum) addresses WLAN and wireless private branch exchange (WPBX) services and focuses on spectrum etiquette to provide fair access to an unlicensed band widely used for different applications and devices. The etiquette does not preclude any common air interface standards or access technologies. It demands listen-before-talk (LBT); thus, a device may not transmit if the spectrum it will occupy is already in use within its range. The power is limited to keep the range short and allow operation in densely populated office areas. The power and connection time are related to the occupied bandwidth to equalize the interference and provide a fair access to frequency–time resources. In the view of WINForum, the asynchronous transmission used in WLAN applications is bursty, begins transmission within milliseconds, uses short bursts that contain large amounts of data, and releases the link quickly. On the other hand, the isochronous transmission, typified by voice services such as a WPBX, uses long holding time, periodic transmission, and flexible link access times that may be extended up to a second. The asynchronous subbands may range from 50 kHz to 10 MHz, whereas the isochronous subbands may be divided into 1.25-MHz segments. The two types are technically contrasting and cannot share the same spectrum.

10.3.3 High-Performance Radio Local Area Network

ETSI's subtechnical committee RES 10 has been assigned the task of developing a standard for the High-Performance Radio Local Area Network (HIPERLAN). The committee secured two bands at 5.12–5.30 GHz and 17.1–17.3 GHz for the HIPERLAN to operate at a minimum useful bit rate of 20 Mbps for point-to-point application with a range of 50 m. It is expected that, at this rate and range, a data rate of 500–1000 Mbps, comparable with fiber distributed data interface (FDDI) for a standard building floor of approximately 1000 m^2, can be achieved. RES 10 is responsible to define a radio transmission technique, including type of modulation, coding, and channel access, as well as the specific protocols.

10.3.4 ARPA

The U.S. Advanced Research Project Agency (ARPA) has sponsored WLAN projects at the University of California at Berkeley (UCB) and University of California at Los Angeles (UCLA). The UCB Infopad project is based on a coordinated network architecture with fixed coordinating nodes and DSSS (CDMA), whereas the UCLA project is for peer-to-peer networks and uses FHSS. Both APRA-sponsored projects are concentrated on the 900-MHz ISM band.

10.4 Access Methods

10.4.1 Fixed-Assignment Access Methods

In the fixed-assignment access method, a fixed allocation of channel resources (frequency or time or both) is made on a predetermined basis to a single user. The three basic access methods—FDMA, TDMA, and CDMA—are the examples of the fixed-assignment access method. In this section, we discuss only the CDMA method. With CDMA, multiple users operate simultaneously over the entire bandwidth of the time–frequency signal domain, and the signals are kept separate by their distinct user-signal codes. As we discussed in chapter 2, the number of users that can be supported simultaneously by a DS-CDMA system is

$$M = \left[\frac{G_p}{E_b/N_0}\right] \times \frac{1}{1+\beta} \times \alpha \times \frac{1}{\upsilon} \times \lambda \qquad (10.1)$$

where G_p = processing gain = B_w/R_s,
B_w = bandwidth,
R_s = symbol transmission rate,
E_b/N_0 = bit energy-to-noise ratio,
β = interference factor,
α = power control factor,
υ = voice activity factor (= 1 for data service),
λ = gain due to sector antenna.

Example 10.1

We consider a CDMA system that uses QPSK modulation and convolutional coding. The system has a bandwidth of 1.25 MHz and transmits data at 9.6 kbps. Find the number of users that can be supported by the system and the bandwidth efficiency. Assume a three-sector antenna with effective gain = 2.6, $\alpha = 0.9$, and an interference factor $\beta = 0.5$. A bit-error rate of 10^{-3} is required.

$$G_p = \frac{1.25 \times 10^6}{9.6 \times 10^3} = 130.2$$

$$P_b = 10^{-3} = \frac{1}{2}\text{erfc}\sqrt{\frac{E_b}{N_0}}$$

$$\frac{E_b}{N_0} \approx 7 \text{ dB}(5)$$

$$\therefore M = \frac{130.2}{5} \times \frac{1}{1+0.5} \times 2.6 \times 0.9 = 40.6 \approx 40$$

$$\eta_{bw} = \frac{40 \times 9.6}{1.25 \times 10^3} = 0.307 \text{ bit/Hz-sec}$$

Example 10.2

We consider a QPSK/DSSS WLAN that is designed to transmit in the 902- to 928-MHz ISM band. The symbol transmission rate is 0.5 Megasymbols/sec. An orthogonal code with 16 symbols is used. A bit-error rate of 10^{-5} is required. How many users can be supported by the WLAN? A three-sector antenna with gain = 2.6 is used. Assume an interference factor $\beta = 0.5$ to account for the interference from users in other cells, and $\alpha = 0.9$. What is the bandwidth efficiency of the system?

$$\text{Band width}, B_w = 928 - 902 = 26 \text{ MHz}$$

$$\text{Data rate}, R = R_s \log_2 16 = 0.5 \log_2 2^4 = 2 \text{ Mbps}$$

$$G_p = \frac{B_w}{R_s} = \frac{26}{0.5} = 52$$

$$P_b = 10^{-5} = \frac{1}{2}\text{erfc}\sqrt{\frac{E_b}{N_0}}$$

$$\frac{E_b}{N_0} \approx 10 \text{ dB}(10)$$

$$\therefore M = \frac{52}{10} \times \frac{1}{1 + 0.5} \times 2.6 \times 0.9 = 8.1 \approx 8$$

$$\eta_{bw} = \frac{8 \times 2}{26} = 0.62 \text{ bit/Hz-sec}$$

10.4.2 Random Access Methods

When each user has a steady flow of information to transmit (e.g., a data file transfer or a facsimile transmission), fixed-assignment access methods are useful because they use communication resources efficiently. However, when the information to be transmitted is bursty in nature, the fixed-assignment access methods result in wasting communication resources. Furthermore, in a cellular system where subscribers are charged based on a channel connection time, the fixed-assignment access methods may be expensive to transmit short messages. Random-access protocols provide flexible and efficient methods for managing a channel access to transmit short messages. The random-access methods give freedom for each user to gain access to the network whenever the user has information to send. Because of this freedom, these schemes result in contention among users accessing the network. Contention may cause collisions and may require retransmission of the information. The commonly used random-access protocols are pure ALOHA, slotted-ALOHA, and CSMA/CD. In the next section we describe briefly details of each of these protocols and provide necessary throughput expressions.

10.4.2.1 Pure ALOHA In the pure ALOHA scheme, each user transmits information whenever the user has information to send. A user sends information in packets. After sending a packet, the user waits a length of time equal to the round-trip delay for an acknowledgment (ACK) of the packet from the receiver. If no ACK is received, the packet is assumed to be lost in a collision, and it is retransmitted with a randomly selected delay to avoid repeated collisions.[1] The normalized throughput S (average packet arrival rate divided by the maximum throughput) of the pure ALOHA protocol is given as

$$S = Ge^{-2G} \tag{10.2}$$

where G = normalized offered traffic load.

From equation (10.2), note that the maximum throughput occurs at traffic load G = 50 percent and is $S = 1/(2e)$. This is about 0.184. Thus, the best channel utilization with the pure ALOHA protocol is only 18.4 percent.

10.4.2.2 Slotted-ALOHA In the slotted-ALOHA system, the transmission time is divided into time slots. Each time slot is made exactly equal to packet transmission time. Users are synchronized to the time slots, so that whenever a user has a packet to send, the packet is held and transmitted in the next time slot. With the synchronized time-slots scheme, the interval of a possible collision for any packet is reduced to one packet time from two packet times, as in the pure ALOHA scheme. The normalized throughput S for the slotted-ALOHA protocol is given as

$$S = Ge^{-G} \tag{10.3}$$

where G = normalized offered traffic load.

1. It should be noted that the protocol on CDMA access channels as implemented in TIA IS-95A is based upon the pure ALOHA approach. The mobile station randomizes its attempt for sending a message on the access channel and may retry if an acknowledgment is not received from the base station. For further details, one should reference section 6.6.3.1.1.1 of TIA IS-95A.

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The maximum throughput for the slotted-ALOHA occurs at $G = 1.0$ [equation (10.3)], and it is equal to $1/e$ or about 0.368. This implies that at the maximum throughput, 36.8 percent of the time slots carry the successfully transmitted packets.

10.4.2.3 Carrier-Sensed Multiple Access CSMA protocols have been widely used in both wired and wireless LANs. These protocols provide enhancements over the pure and slotted ALOHA-protocols. The enhancements are achieved through use of additional capability at each user station to sense the transmissions of other user stations. The carrier-sensed information is used to minimize the length of collision intervals. For carrier sensing to be effective, propagation delays must be less than packet transmission times. Two general classes of CSMA protocols are nonpersistent and p-persistent. Each of these classes can be used with the slotted or unslotted operation.

- **Non-persistent CSMA.** A user station does not sense the channel continuously while it is busy. Instead, after sensing the busy condition, it waits for a randomly selected interval of time before sensing again. The algorithm works as follows: If the channel is found to be idle, the packet is transmitted; if the channel is sensed busy, the user station backs off to reschedule the packet to a later time. After backing off, the channel is sensed again, and the algorithm is repeated again.
- ***p*-persistent CSMA.** The slot length is typically selected to be the maximum propagation delay. When a station has information to transmit, it senses the channel. If the channel is found to be idle, it transmits with probability p. With probability $q = 1 - p$, the user station postpones its action to the next slot, where it senses the channel again. If that slot is idle, the station transmits with probability p or postpones again with probability q. The procedure is repeated until either the frame has been transmitted or the channel is found to be busy. When the channel is detected busy, the station then senses the channel continuously. When it becomes free, it starts the procedure again. If the station initially senses the channel to be busy, it simply waits one slot and applies the preceding procedure.
- **1-persistent CSMA.** 1-Persistent CSMA is the simplest form of the p-persistent CSMA. It signifies the transmission strategy, which is

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to transmit with probability 1 as soon as the channel becomes idle. After sending the packet, the user station waits for an ACK. If it is not received within a specified amount of time, the user station waits for a random amount of time and then resumes listening to the channel. When channel is again found to be idle, the packet is retransmitted immediately.

For more details, refer to references [2, 21].

The throughput expressions for the CSMA protocols follow:

- **Unslotted Nonpersistent CSMA**

$$S = \frac{Ge^{-aG}}{G(1+2a)+e^{-aG}} \tag{10.4}$$

- **Slotted Nonpersistent CSMA**

$$S = \frac{aGe^{-aG}}{1-e^{-aG}+a} \tag{10.5}$$

- **Unslotted 1-Persistent CSMA**

$$S = \frac{G[1+G+aG(1+G+(aG)/2)]e^{-G(1+2a)}}{G(1+2a)-(1-e^{-aG})+(1+aG)e^{-G(1+a)}} \tag{10.6}$$

- **Slotted 1-Persistent CSMA**

$$S = \frac{Ge^{-G(1+a)}[1+a-e^{-aG}]}{(1+a)(1-e^{-aG})+ae^{-G(1+a)}} \tag{10.7}$$

where S = normalized throughput,
G = normalized offered traffic load,
$a = \tau/T_p$,
τ = propagation delay,
T_p = packet transmission time.

Example 10.3

We consider a WLAN installation in which the maximum propagation delay is 0.4 μs. The WLAN operates at a data rate of 10 Mbps, and each packet has 400 bits. Calculate the throughput with (1) an unslotted nonpersistent, (2) a slotted persistent, and (3) a slotted 1-persistent CSMA protocol.

$$T_p = \frac{400}{10} = 40\ \mu s$$

$$a = \frac{\tau}{T_p} = \frac{0.4}{40} = 0.01$$

$$G = \frac{40 \times 10^{-6} \times 10 \times 10^{6}}{400} = 1$$

- Slotted Nonpersistent:

$$S = \frac{0.01 \times 1e^{-0.01}}{1 - e^{-0.01} + 0.01} = 0.496$$

- Unslotted Nonpersistent:

$$S = \frac{1 \times e^{-0.01}}{(1 + 0.02) + e^{-0.01}} = 0.493$$

- Slotted 1-Persistent:

$$S = \frac{e^{-1.01}(1 + 0.01 - e^{-0.01})}{(1 + 0.01)(1 - e^{-0.01}) + 0.01e^{-1.01}} = 0.531$$

10.5 Error Control Schemes

Channel coding and automatic repeat request (ARQ) schemes are used to increase the performance of mobile communication systems. In the physical layer of DS-CDMA system, error detection and correction techniques such as forward error correction (FEC) schemes are used. For some of the data services, higher-layer protocols use ARQ schemes to enable retransmission of any data frames in which an error is detected. The ARQ schemes are classified as follows [21, 22].

Stop and Wait. The sender transmits the first packet numbered 0 after storing a copy of that packet. The sender then waits for an ACK numbered 0, ACK0 of that packet. If the ACK0 does not arrive before a time-out, the sender makes another copy of the first packet, also numbered 0, and transmits it. If the ACK0 arrives before a time-out, the sender discards the copy of the first packet and is ready to transmit the next packet, which it numbers 1. The sender repeats the previous steps, using number 1 instead of 0. The advantages of the Stop and Wait protocol are its simplicity and its small buffer requirements. The sender needs to keep only a copy of the packet that it last transmitted, and the receiver does not need to buffer packets at the data link layer. The main disadvantage of the Stop and Wait protocol is that it does not use the communication link efficiently.

The total time taken to transmit a packet and to prepare for transmitting the next one is

$$T = T_p + 2T_{prop} + 2T_{proc} + T_a \tag{10.8}$$

The protocol efficiency without error is

$$\eta(0) = \frac{T_p}{T} \tag{10.9}$$

where T = total time for transmitting a packet,
T_p = transmission time for a packet,
T_{prop} = propagation time of a packet or an ACK,
T_{proc} = processing time for a packet or an ACK,
T_a = transmission time for an ACK.

If p is the probability that a packet or its ACK is corrupted by transmission errors and a successful transmission of a packet and its ACK takes T seconds and occurs with probability $1 - p$, the protocol efficiency for full duplex (FD) and half duplex (HD) operation are given as

$$\eta_{FD} = \frac{(1 - p)T_p}{(1 - p)T + pT_p} \tag{10.10}$$

$$\eta_{HD} = \frac{(1 - p)T_p}{T} \tag{10.11}$$

Selective Repeat Protocol (SRP). The data link layer in the receiver delivers exactly one copy of every packet in the correct order. The data link layer in the receiver may get the packets in the wrong order from the physical layer. This occurs, for example, when transmission errors corrupt the first packet and not the second one. The second packet arrives correctly at the receiver before the first. The data link layer in the receiver uses a buffer to store the packets that arrive out of order. Once the data link layer in the receiver has a consecutive group of packets in its buffer, it can deliver them to the network layer. The sender also uses a buffer to store copies of the unacknowledged packets. The number of the packets, which can be held in the sender/receiver buffer is a design parameter. Let W be the number of packets that the sender and receiver buffers can each hold and SRP be the number of packets modulo-$2W$. The protocol efficiency without any error and with an error probability of p is given as

$$\eta(0) = \min\left\{\frac{WT_p}{T}, 1\right\} \tag{10.12}$$

For very large W, the protocol efficiency is

$$\eta(p) = 1 - p \tag{10.13}$$

where T = time-out = WTp

$$\eta(p) = \frac{2 + p(W - 1)}{2 + p(3W - 1)} \tag{10.14}$$

SRP is very efficient, but it requires buffering packets at both the sender and the receiver.

Go-Back-N (GBN). The Go-Back-N protocol allows the sender to have multiple unacknowledged packets without the receiver having to store packets. This is done by not allowing the receiver to accept packets that are out of order. When a time-out timer expires for a packet, the transmitter resends that packet and all subsequent packets. The Go-Back-N protocol improves on the efficiency of the Stop and Wait protocol but is less efficient than SRP. The protocol efficiency is given as

$$\eta_{FD} = \frac{1}{1 + \left(\frac{p}{1-p}\right)W} \tag{10.15}$$

Window-Control Operation Based on Reception Memory ARQ. In digital cellular systems, bursty errors occur by multipath fading, shadowing, and handoffs. The bit-error rate fluctuates from 10^{-1} to 10^{-6}. Therefore, the conventional ARQ schemes do not operate well in digital cellular systems. Window-control operation based on reception memory (WORM) ARQ has been suggested for control of dynamic error characteristics. It is a hybrid scheme that combines SRP with GBN. GBN protocol is chosen in the severe error condition, whereas SRP is selected in the normal error condition.

Variable Window and Frame Size GBN and SRP. Since CDMA systems have bursty error characteristics, the error control schemes should have a dynamic adaptation to bursty channel environment. The SRP and GBN with variable window and frame size have been proposed in [22] to improve error control in the CDMA systems. Table 10.3 provides the window and frame size for different bit-error rates. If the error-rate increases, the window and frame size are decreased. In the case of error-rate being small, the window and frame size are increased. The optimum threshold values of bit-error rate (BER) and window and frame sizes were obtained through computer simulation.

Table 10.3 Bit-Error Rate Versus Window and Frame Size

Bit-Error Rate (BER)	Window Size (W)	Frame Size, bits
$BER \le 10^{-4}$	32	172
$10^{-4} < BER < 10^{-3}$	8	80
$10^{-3} < BER < 10^{-2}$	4	40
$10^{-2} < BER$	2	16

In CDMA systems, the forward link consists of pilot, sync, paging, and traffic channels. System information sent on the pilot, sync, and paging channels allows each mobile station to evaluate the BER easily by measuring the ratio of the number of retransmitted frames to the number of transmitted frames over a 2-second period. Thus, the mobile station can change the window and frame size according to the BER.

Example 10.4

We consider a WLAN in which the maximum propagation delay is 4 μs. The WLAN operates at a data rate of 10 Mbps. The data and ACK packet lengths are 400 and 20 bits, respectively. The processing time for a data or ACK packet is 1 μs. If the probability p that a data packet or its ACK can be corrupted during transmission is 0.01, find the data link protocol efficiency with (1) Stop and Wait protocol, full duplex, (2) SRP with window size W = 8, and (3) Go-Back-N protocol with window size W = 8.

$$T_p = \frac{400}{10} = 40\ \mu s$$

$$T_a = \frac{20}{10} = 2\ \mu s$$

$$T_{prop} = 4\ \mu s$$

$$T_{proc} = 1\ \mu s$$

$$T = 40 + 2 \times 4 + 2 \times 1 + 2 = 52\ \mu s$$

Stop and Wait:

$$\eta = \frac{(1 - 0.01) \times 40}{52} = 0.762$$

SRP:

$$\eta = \frac{2 + 0.01(8 - 1)}{2 + 0.01(24 - 1)} = 0.954$$

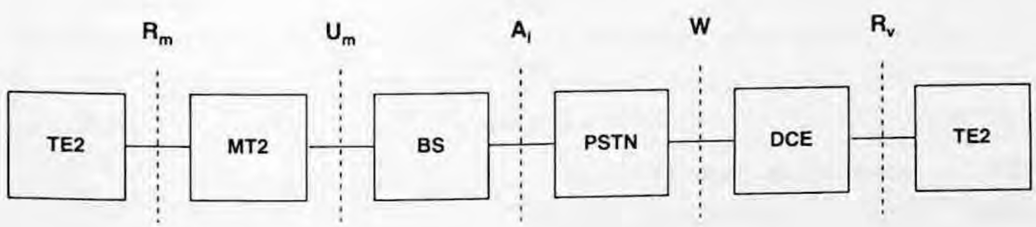
Go-Back-N:

$$\eta = \frac{1}{1 + 8\left(\frac{0.01}{1 - 0.01}\right)} = 0.925$$

10.6 The Data Services Standard for CMDA Cellular/ Personal Communications Systems

CDMA systems send data using the reference model shown in figure 10.3 as standardized in TIA IS-99 [14]. The following is the description of the reference points:

- Reference point R_m is a physical interface that connects a Terminal Equipment 2 (TE2) to an Mobile Terminal Type 2 (MT2). An MT2 provides a non-ISDN user interface.
- Reference point U_m is a physical interface that connects a Mobile Terminal Type 0 (MT0) or MT2 to a base station. U_m is the air interface. An MT0 is a self-contained data-capable mobile terminal that does not support an external interface.
- Reference point A_i is a physical interface connecting a base station to the PSTN.



TE2: Terminal Equipment 2
MT2: Mobile Termination 2
BS: Base Station
DCE: Data Circuit-terminal Equipment
PSTN: Public Switching Telephone Network

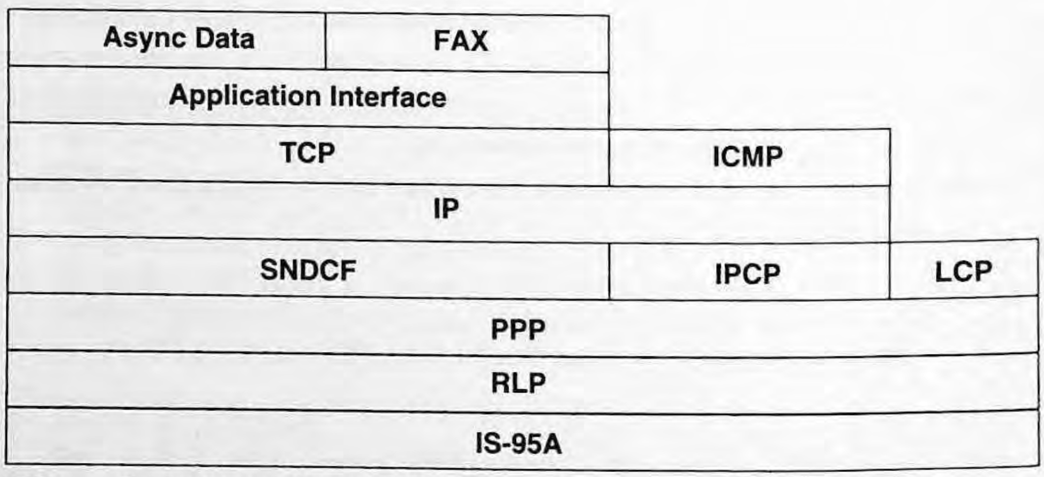
Figure 10.3 Reference model for asynchronous data transmission. (Reproduced under written permission of the copyright holder [TIA].)

- Reference point W is a physical interface that connects data circuit-terminating equipment (DCE) to the PSTN.
- Reference point R_v is a reference point between a TE2 and a Data Communications Equipment (DCE). It may be a physical interface, or it may be internal to the user equipment. CCITT V-Series modems are the examples of TE2; Group-3 FAX is an example of a DCE. TE2 is equivalent to data terminal equipment (DTE) as defined in EIA/TIA-232E [15]; similarly, MT2 is equivalent to DCE.

10.6.1 Asynchronous Data and Group-3 Facsimile

The general approach taken in TIA IS-95A [13] for data services reuses the previously specified physical layer of the IS-95A protocol stack as the physical layer. Figure 10.4 shows the air interface (U_m) protocol stack.

The current TIA standards define three primary services: asynchronous data, Group-3 facsimile, and short message service (SMS). Stan-



TCP: Transmission Control Protocol
ICMP: Internet Control Message Protocol
IP: Internet Protocol
SNDCF: Sub-Network Dependent Convergence Function
IPCP: Internet Protocol Control Protocol
LCP: Link Control Protocol
PPP: Point-to-Point Protocol
RLP: Radio Link Protocol

Figure 10.4 The U_m protocol stack.

dard activities are in progress to define packet data, synchronous data, and other primary services.

IS-95A asynchronous data has been structured as a circuit-switched service in which a dedicated path is established between the data devices for the duration of the call. It is used for connectivity through the PSTN when point-to-point communications to a PC or FAX user is required. For example, for a file transfer involving PC-to-PC communications, the asynchronous data service is the preferred cellular service mode.

The radio link protocol (RLP) employs automatic repeat request, forward error correction, and flow control. Flow control and retransmission of data blocks with errors are used to provide an improved performance in the mobile segment of the data connection at the expense of variations in throughput and delay. Typical raw channel data-error rates for cellular transmission are approximately 10^{-2}. However, an acceptable data transmission usually requires a bit-error rate of about 10^{-6}. In order to achieve this, it requires the design of efficient ARQ and error-correction codes to deal with error characteristics in the mobile environment.

The CDMA protocol stack for data and facsimile (fig. 10.4) has the following layers:

- **The Application Interface Layer** includes an application interface between the data source/destination in the mobile terminal (MT0) or terminal equipment (TE2) and the transport protocol layer. In the base station, the application interface resides between the data source/destination on the network (A_i interface) side and the transport protocol layer. The application interface provides modem control, AT command processing,[2] negotiation of air interface data compression, and data compression over the air interface (optional).
- **The Transport Layer** for CDMA asynchronous data and FAX services is based on Internet transport layer protocol known as transmission control protocol (TCP) [6]. The implementation complies with the requirements for TCP with modifications as described in IS-95 [13]. If the modified procedure is disabled, there is no maxi-

2. The AT commands were originally defined by the Hayes Microcomputer Company for their wireline modems. The command set has now been adopted by most wireline and wireless modems. The name AT is derived from the use of AT to preface all commands to the modem.

mum number of retransmission attempts during synchronization, and an established TCP connection remains open until explicitly closed by the mobile station or base station. The application interface sets the value of R_2 in the protocol. The base station follows either the procedure of the Internet control message protocol (ICMP) [7] or the preceding given procedure.

- **The Network Layer** for CDMA async data and FAX services is based on the Internet network layer protocol known as the Internet protocol (IP) [5]. The network layer includes the ICMP [6]. The implementation complies with the requirements of the IP [5] and the requirements for Internet hosts [8] with modifications as described in IS-95 [13]. The interface between the network and transport layer complies with the requirements of the ICMP [7].
- **The Subnetwork Dependent Convergence Function (SNDCF)** performs header compression on the headers of the transport and network layers. This function is negotiated using point-to-point protocol (PPP) and internet protocol control protocol (IPCP) [10]. Mobile stations support Van Jacobson TCP/IP header compression. A minimum of one compression slot is negotiated. Base stations support TCP/IP header compression compatible with that required for mobile stations. Negotiation of the parameters of header compression is carried out using IPCP. The SNDCF sublayer accepts a network layer datagram from the network layer, performs header compression as required, and passes the datagram to the PPP layer, indicating the appropriate PPP identifier. The SNDCF sublayer receives network layer datagrams with compressed or uncompressed headers from the PPP layer, decompresses the datagram header as necessary, and passes the datagram to the network layer.
- **The Data Link Layer** uses PPP [11]. The PPP link control protocol (LCP) is used for initial link establishment and for the negotiation of optional link capabilities. The data link layer uses the PPP and IPCP to negotiate IP addresses and TCP/IP header compression. The data link layer accepts network layer datagrams from the SNDCF and encapsulates them in the PPP information field. The packet is framed using the octet synchronous framing protocol, except that there is no interframe fill. No flag octets are sent between a flag octet that ends one PPP frame and the flag octet that begins the subsequent PPP frame. The framed PPP packets are passed to the RPL layer for transmission. The data link layer

accepts received octets from the RLP layer and reassembles the original PPP packets. The PPP process discards any PPP packet for which the received frame check sequence (FCS) is not equal to the computed value.

- **The Internet Protocol Control Protocol Sublayer** supports negotiation of the IP-address (type = 3) and IP-compression protocol (type = 2) parameters. IPCP negotiates a temporary IP address for the mobile station whenever a transport layer connection is actively opened. Mobile stations maintain the temporary IP address only while a transport layer connection is open or being opened; they discard the temporary IP address when the transport layer connection is closed.
- **The Link Control Protocol** layer messages with a protocol identifier of 0xC021 to the PPP layer which processes the packet according to PPP LCP. For other supported protocol identifiers, the PPP layer removes the PPP encapsulation and passes the datagram and protocol identifier to the SNDCF. For unsupported protocol identifiers, the LCP protocol-Reject is passed to the RLP layer for transmission. The mobile station supports the PPP LCP Configure-Request, Configure-ACK, Configure-NAK, Configure-Reject, Terminate-Request, Terminate-ACK, Code-Reject, and Protocol-Reject. Other LCP packet types may also be supported. The PPP LCP negotiates the following configuration options:
  - ✘ **Async control character map.** The mobile station does not require any mapping of control characters. The base station may negotiate mapping of control characters.
  - ✘ **Protocol field compression.** This option applies when the protocol number is less than 0xFF.
  - ✘ **Address and control field compression.** This option applies when the protocol number is not 0xC021.

  The mobile station may support other configuration options (e.g., maximum receive unit, authentication protocol, link quality protocol, or magic number). When an option that is not supported is received, the Configure-Reject is sent as an indication to the peer.
- **The Radio Link Protocol Layer** provides an octet stream service over the forward and reverse traffic channels and substantially reduces the error rate typically exhibited by these channels. This service is used to carry the variable-length data packets on the PPP layer. The RLP divides the PPP packets into TIA IS-95A traffic

channel frames for transmission. There is no direct relationship between PPP packets and traffic channel frames. A large packet may span multiple traffic channel frames, or a single traffic channel frame may contain all or part of several small PPP packets. The RLP is unaware of higher-layer framing; it operates on a featureless octet stream, delivering the octets in the order received from the PPP layer. For service options supporting an interface with multiplex option 1, RLP frames may be transported as primary or secondary traffic or as signaling via data burst messages. For the primary or secondary traffic, the RLP generates and supplies exactly one frame to the multiplex sublayer every 20 ms. The frame contains the service option information bits. The multiplex sublayer in the mobile station categorizes every received traffic frame and supplies the frame type and accompanying bits, if any, to the RLP layer. The frame type and frame category for primary and secondary traffic are given in tables 10.4 and 10.5. A blank frame is used for blank and burst transmission of signaling traffic.

The signaling subchannel may carry frames from multiple RLPs, with each RLP having a distinct BURST_TYPE. Each service

Table 10.4 RLP Frame with Primary Traffic

RLP Frame Type	Bits/Frame	Multiplex Option 1 Frame Categories
Full rate	171	1
Half rate	80	2, 6, 11
1/8 rate	16	4, 8, 13
Blank	0	5, 14
Erasure	0	All Others

Table 10.5 RLP Frame with Secondary Traffic

RLP Frame Type	Bits/Frame	Multiplex Option 1 Frame Categories
Rate 1	168	14
Rate 7/8	152	13
Rate 3/4	128	12
Rate 1/2	88	11
Blank	0	1–8
Erasure	0	9, 10

option defines a unique BURST_TYPE used for RLP. Each primary and secondary multiplex subchannel supports at most a single RLP layer. RLP data frames sent on one multiplex subchannel are not to be transmitted on another subchannel. RLP frames are not sent on the access and paging channels.

- **The Radio Interface** provides the physical layer between the mobile station and base station. In addition, it provides the multiplex sublayer, the radio link management, and the call control as defined in TIA IS-95A. The mobile station and the base station use service option 4 for async data services and service option 5 for Group-3 FAX services (see table 3.3). The mobile station and the base station do not transmit 1/4 rate frames when service option 4 or 5 is active. Service options 4 and 5 support an interface with multiplex option 1. RLP frames for service options 4 and 5 are transported only as primary traffic or signaling traffic. The mobile station and the base station perform service option negotiation for service options 4 and 5 as described in TIA IS-95A (sections 6.6.4.1.2 and 7.6.4.1.2). Initialization and connection in the mobile station and the base station to accept service option 4 or 5 in response to a Service Option Request Order are performed according to the specifications in TIA IS-99 (sections 3.8.4.1 and 3.8.4.2).

10.6.2 Short Message Service

The short message service [16] allows the exchange of short alphanumeric messages between a mobile station and the cellular/PCS system and between the cellular/PCS system and an external device capable of transmitting and optionally receiving short messages. The external device may be a voice telephone, a data terminal, or a short message entry system. The SMS consists of message entry features, administration features, and message transmission capabilities. These features are distributed between a cellular/PCS system and the SMS message center (MC), which together make up the SMS system. The MC may be either separate from or physically integrated into the cellular/PCS system.

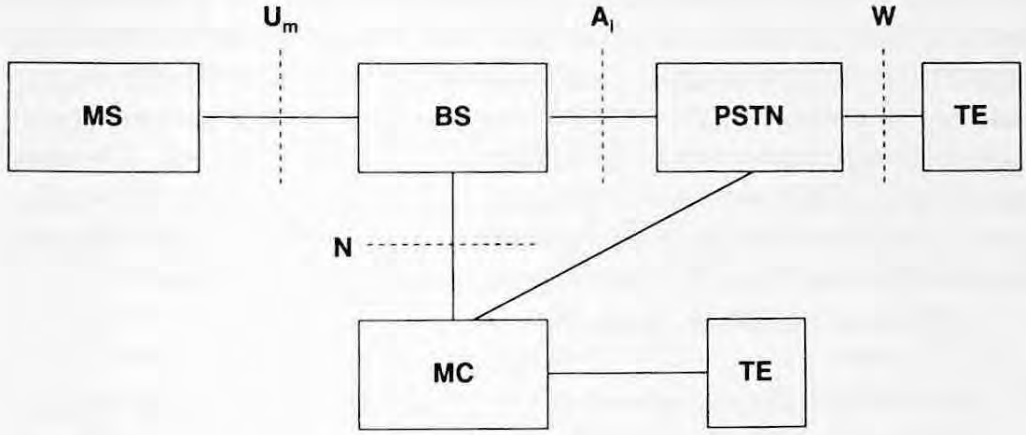
Short message entry features are provided through interfaces to the MC and the mobile station. Senders use these interfaces to enter short messages, intended destination addresses, and various delivery options. MC interfaces may include features such as audio response prompts and DTMF reception for dial-in access from voice telephones, as well as appropriate menus and message entry protocols for dial-in or dedicated data terminal access. Mobile station interfaces may include keyboard and display features to support message entry. Also, a cellular voice service subscriber can use normal voice or data features of the mobile station to call an SMS system to enter a message.

An SMS teleservice can provide the option of specifying priority level, future delivery time, message expiration interval, or one or more of a series of short, predefined messages. If supported by the teleservice, the sender can request acknowledgment that the message was received by the mobile station. An SMS recipient, after receiving a short message, can manually acknowledge the message. Optionally, the recipient can specify one of a number of predefined messages to be returned with acknowledgment to the sender.

SMS administration features include message storage, profile editing, verification of receipt, and status inquiry capabilities. The SMS transmission capabilities provide for the transmission of short messages to or from an intended mobile station and the return of acknowledgments and error messages. These messages and acknowledgments are transmitted to or from the mobile station whether it is idle or engaged in a voice or data call. The cellular service provider may offer SMS transmission to its cellular voice and data customers only or may provide an SMS only service without additional voice or data transmission capability. All available mobile stations on a CDMA paging channel can receive a broadcast message. A broadcast message is not acknowledged by the mobile station. Broadcast messaging services may be made available to mobile stations on a CDMA paging channels as well as to mobile stations during a call on a CDMA traffic channel.

Figure 10.5 shows the network reference model for SMS. The base station contains the transceiver equipment, mobile switching center, and any interworking function (IWF) required for network connection. These elements are grouped together because there is no need to distinguish them. The MC element in the model represents a generic SMS message center function. The N reference point represents one or more standardized interfaces between an SMS message center and a BS. The terminal equipment (TE) is voice or data equipment connected either directly or indirectly to the MC. It is possible for the MC to be included in or colocated with a BS. In this case, the N interface is internal to the BS.

The SMS protocol stack for the CDMA mode of operation is shown in figure 10.6. The SMS bearer service is the portion of the SMS system responsible for delivery of messages between the MC and mobile user equipment. The bearer service is provided by the SMS transport layer and SMS relay layer.



MS: Mobile Station
BS: Base Station
MC: Message Center
PSTN: Public Switching Telephone Network
TE: Terminal Equipment

Figure 10.5 Simplified SMS reference model. (Reproduced under written permission of the copyright holder [TIA].)

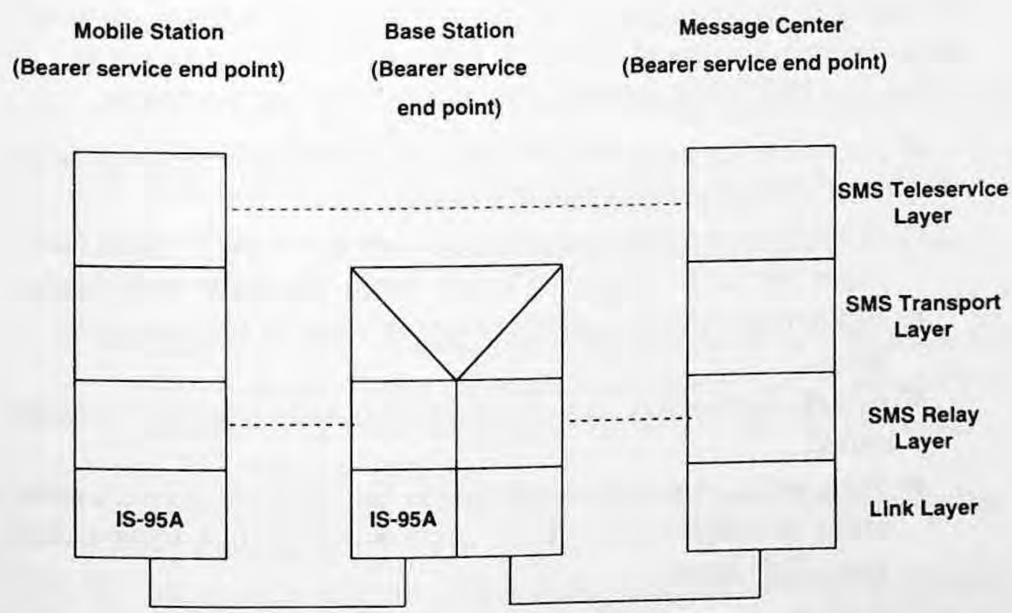


Figure 10.6 SMS protocol stack. (Reproduced under written permission of the copyright holder [TIA].)

The SMS transport layer is the highest layer of the bearer service protocol. The transport layer manages the end-to-end delivery of messages. In an entity serving as a relay point, the transport layer is responsible for receiving SMS transport layer messages from an underlying SMS relay layer, interpreting the destination address and other routing information, and forwarding the message via an underlying SMS relay layer. In entities serving as end points, the transport layer provides the interface between the SMS bearer service and SMS teleservice.

SMS uses the following layers:

- **The SMS Relay Layer** provides the interface between the transport layer and the link layer used to carry short message traffic. On the U_m interface, the SMS relay layer supports the SMS transport layer by providing the interface to the TIA IS-95A transmission protocols required to carry SMS data between CDMA mobile stations and the base stations. On the N interface, the SMS relay layer supports the SMS transport layer by providing the interface to the network protocols required to carry SMS data between the MC and TIA IS-95A base stations. The N reference point is assumed to be an intersystem network link with connectivity to the MC. Intersystem links can use a variety of public and private protocols. SMS protocols and message formats on intersystem links may differ from those used on the CDMA air interface. The N interface relay layer is responsible for formatting and parsing SMS messages as necessary when transmitting and receiving messages on the intersystem links. The SMS relay layer performs the following functions:
  - ✘ Accepting transport layer messages and delivering them to the next indicated relay point or end point.
  - ✘ Providing error indications to the transport layer when messages cannot be delivered to the next relay point or end point.
  - ✘ Receiving messages and forwarding them to the transport layer.
  - ✘ Interfacing to and controlling the link layer used for message relay.
  - ✘ Formatting messages according to the SMS standards and/or other message standards, as required by the link layer and/or peer SMS layer.
- **The SMS Transport Layer** resides in SMS bearer service end points and relay points. In a bearer service end point, the SMS

transport layer provides the means of access to the SMS system for teleservices that generate or receive SMS messages. In a bearer service relay point, the transport layer provides an interface between relay layers. The SMS transport layer uses relay layer services to originate, forward, and terminate SMS messages sent between mobile stations and MCs. It is assumed that the link layers used by the relay layers support message addressing, so that certain address parameters can be inferred by the relay layer from link layer headers and are therefore not required in transport layer messages. It is assumed that an SMS point-to-point message does not require certain address parameters because the link layers will provide this address. On the CDMA paging channel, for example, it can be assumed that the relay layer can extract the address from the ADDRESS field of the TIA IS-95A data burst message. SMS transport layers have different functions in SMS bearer service end points and relay points. In an SMS bearer service end point, the transport layer provides the following functions:

- ✘ Receiving message parameters from SMS teleservices, formatting SMS transport layer messages, and passing the message to the relay layer using the appropriate relay layer service primitives.
- ✘ Informing the relay layer when all expected acknowledgments of submitted messages have been received.
- ✘ Informing the teleservices when relay layer errors are reported.
- ✘ Receiving SMS messages from the relay layer and passing the messages to the SMS teleservice.
- ✘ Performing authentication calculations in mobile stations.

In an SMS bearer service relay point, the transport layer provides the following functions:

- ✘ Receiving SMS messages from a relay layer, reformatting the SMS transport layer message if necessary, and passing the message to another relay layer using the appropriate relay layer service primitives.
- ✘ Passing confirmations or error reports between the relay layers if requested.
- ✘ Performing authentication calculations or interfacing to the entities performing authentication calculations in the TIA IS-95A base stations.

The transport layer requires the following services from the relay layer:

- ✘ Accepting transport layer messages and delivering them to the next indicated relay point or end point.
- ✘ Returning confirmations or error reports for messages sent.
- ✘ Receiving messages and forwarding them to the transport layer with the appropriate parameters.

- **The SMS Teleservice Layer** resides in a bearer service end point and supports basic SMS functions through a standard set of subparameters of the transport layer's bearer data parameter.

  When a mobile station sends an SMS User Acknowledgment message, the teleservice layer performs the following:

  - ✘ Supplies the Destination Address parameter to the transport layer and sets the Destination Address parameter equal to the address contained in the Originating Address field of the SMS message being acknowledged.
  - ✘ Sets the MESSAGE_ID field of the Message Identifier subparameters to the value of MESSAGE_ID field in the SMS message being acknowledged.

  Broadcast Messaging Service Teleservice messages are sent using the SMS Deliver message. For more details, refer to TIA IS-637.

10.6.3 Packet Data Services for CDMA Cellular/Personal Communications Systems [17]

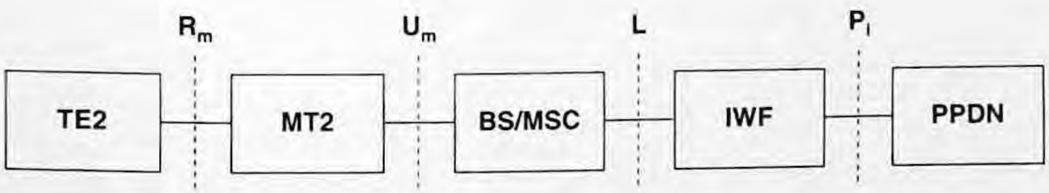
The packet data service option is specified by a mobile station at the time a packet link layer connection is opened. The packet data service option to be used may be configurable or may be fixed by the mobile station manufacturer. Service option 7 (see table 3.3) is used to request packet data service through an interworking function supporting an Internet standard PPP interface to network protocols. Service option 8 is used to request packet data service through an IWF supporting CDPD data services over a PPP interface. Even though service option 8 for packet data service does not use the same air interface as CDPD, this service options uses the same IWF supporting CDPD data services over PPP interface. Additional packet data service options may be defined in the future for the purpose of selecting other types of IWF resources or services requested.

Packet data service options provide a means of establishing and maintaining traffic channels for packet data service. When no other service option is connected, packet data service is carried as primary traffic. When another compatible service option is connected (e.g., voice or asynchronous data), packet data service can be carried as either primary or secondary traffic.

Figure 10.7 shows the reference model for packet data services. This reference model does not address intersystem and mobility issues. The reference points follow:

- **Reference Point R_m**—a physical interface to connect type 2 terminal equipment (TE2) to a mobile station (MT2).
- **Reference Point U_m**—a physical interface to connect mobile stations (type MT0 or MT2) to a base station and mobile switching center. This is the air interface.
- **Reference Point L**—A physical interface connecting a mobile switching center to an IWF.
- **Reference Point P_i**—a physical interface connecting an IWF to the public packet data network (PPDN).

The relay layers provide lower-layer communication and packet framing between the entities of the packet data service reference model. Over the R_m interface between TE2 and the MT2, the relay layer is a RS-232 [15] interface. Over the U_m interface, the relay layer is a combination of RLP and TIA IS-95A protocols. On the L interface, the relay layer uses



TE2: Terminal Equipment 2
MT2: Mobile Terminal 2
BS: Base Station
MSC: Mobile Switching Center
IWF: Interworking Function
PPDN: Public Packet Data Network

Figure 10.7 Reference model for packet data service.

the protocols defined in TIA IS-687 [18]. The two options for packet protocol stacks are shown in figures 10.8 and 10.9.

10.6.3.1 Relay Layer R_m Interface Protocol Option The relay layer R_m interface protocol option supports terminal equipment (type TE2) applications in which the end-to-end link layer supports network

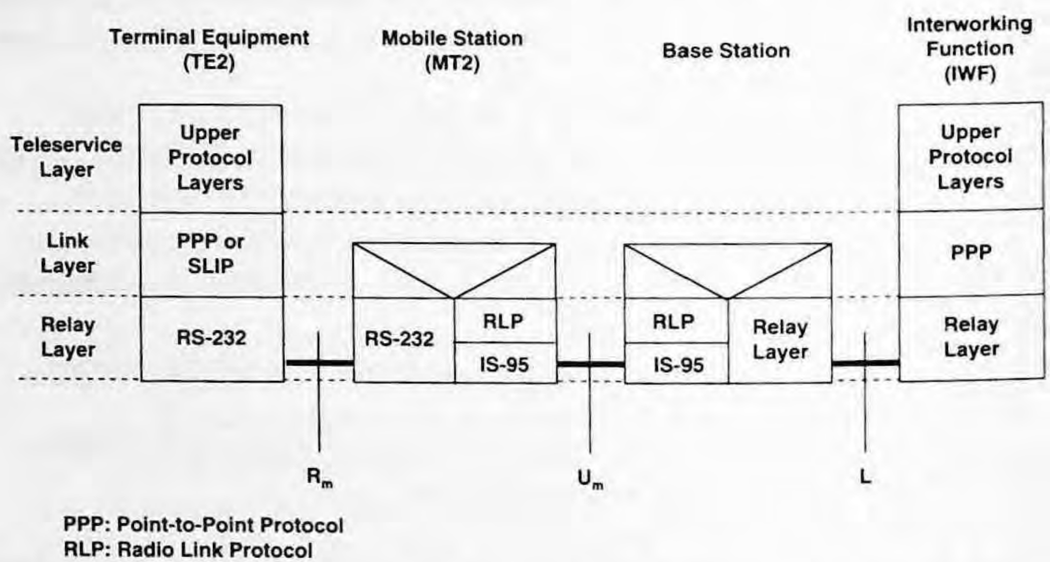


Figure 10.8 Relay layer R_m interface protocol option. (Reproduced under written permission of the copyright holder [TIA].)

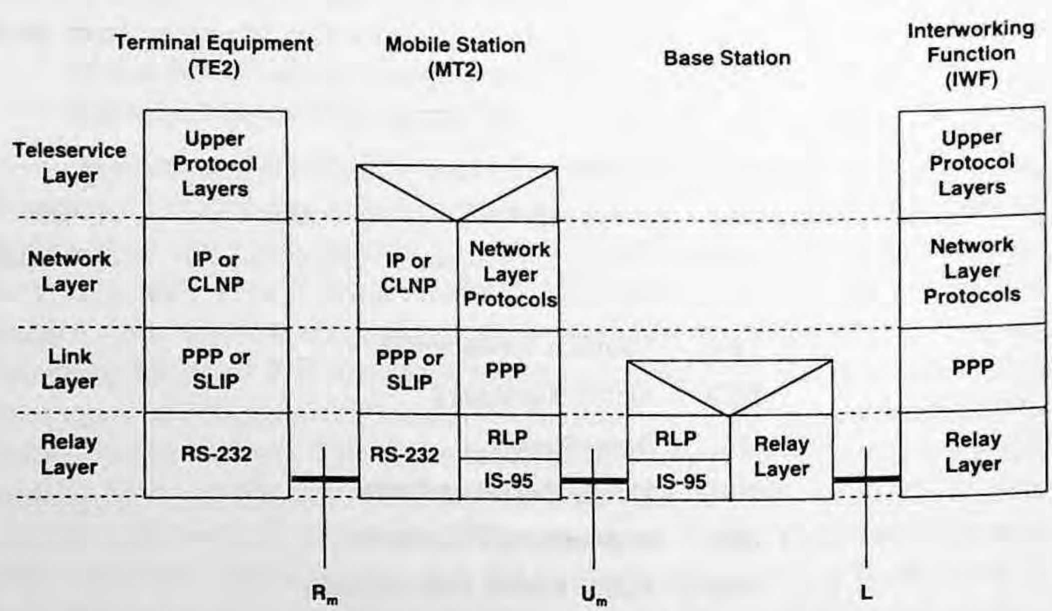


Figure 10.9 Network layer R_m interface option. (Reproduced under written permission of the copyright holder [TIA].)

and higher-layer protocols that reside entirely within the terminal equipment. With this option, the terminal equipment is responsible for all aspects of packet data service mobility management and network address management. For this R_m interface protocol option, the link layer provides the means for carrying packet data service between the TE2 and the IWF. For service options 7 and 8, the link layer is implemented using the Internet PPP. The link layer also provides protocol discrimination and supports functions such as header compression for network layer protocols.

For the R_m interface protocol option, the packet teleservice layer includes the network and higher-layer protocols used for communication between peer application entities in the TE2 and remote terminal equipment via the PPDN. For service option 7, network layer protocols may include all those protocols for which PPP indentifiers have been assigned by the Internet Assigned Number Authority. For service option 8, network layer protocols may include those protocols supported by the CDPD network.

10.6.3.2 Network Layer R_m Interface Protocol Option The network layer R_m interface protocol option supports terminal equipment (type TE2) applications, such as CDPD applications, in which the mobile station (type MT2) supports network and higher-layer protocols. With this protocol option, the mobile station is responsible for all aspects of packet mobility management and network address management, and the terminal equipment application operates as if it were locally connected to a network layer routing server. In this protocol option, independent link layers provide the means to carry packet data service data between the terminal and the mobile station and between the mobile station and the IWF. For service options 7 and 8, the link layer between the mobile station and the IWF is implemented using the Internet PPP. In the MT2-IWF link layer, the link layer also provides protocol discrimination and supports functions such as header compression for network layer protocols.

The link layer between the MT2 and TE2 is implemented using the Internet PPP. Alternatively, the SLIP protocol, as defined in RFC 1055, may be used between the MS and the terminal to support the IP network layer protocol. For this R_m interface protocol option, the network layer also provides independent services between the terminal and the MS and between the MS and the IWF. The network layer between MS and IWF includes routing protocols (e.g., IP and CLNP) and the CDPD mobile network registration protocol (MNRP). The network layer between terminal and mobile station includes the routing protocols only.

With this R_m interface protocol option, the packet teleservice layer includes the transport and higher-layer protocols used for communication between peer application layer entities in the TE2 and remote terminal equipment via PPDN. Transport layer protocols may include protocols such as TCP or TP4.

10.6.3.3 Packet Data Service States Packet data service has two states, active and inactive. Packet data service becomes active whenever selected by the user. The means for activating packet data service can be through commands on the R_m interface, a separate control on the user interface of the mobile station, or other means. If no user-accessible means of control is provided, packet data service is always active. When packet data service is active, the mobile station maintains the packet data service enabled status. The packet data service enabled status is used to control repeated packet data service origination attempts when packet data service is not available. When packet data service is enabled, the mobile station attempts to originate a packet data service with a serving IWF. When packet data service is disabled, the mobile station inhibits repeated attempts to originate packet data service.

10.6.3.4 Link Layer The IWF and the mobile station maintain a link layer connection for the purpose of transmitting and receiving packet data between the mobile station and the IWF. The link layer connection is opened when a call is made using a packet data service option. Once a link layer connection is opened, bandwidth (in the form of a traffic channel assignment) is allocated to the connection on an as-needed basis. After opening a link layer connection, the mobile station and the IWF perform link layer configuration negotiation. The mobile station and IWF may also negotiate network layer protocol control parameters and perform procedures associated with network registration and authentication.

10.6.3.5 Link Layer Connection State The link layer connection can be in any of the following states:

- **Closed.** The link layer connection is closed when IWF has no link layer connection state information about the mobile station.
- **Opened.** The link layer connection is opened when the IWF has link layer connection state information for the mobile station. The opened state has two substates:

- *Active*. An opened link layer connection is active when IWF has link layer connection state information for the mobile station, there is an L interface frame relay-switched virtual circuit for the mobile station, and the mobile station is on a traffic channel supporting a packet data service option.
- *Dormant*. An opened link layer connection is dormant when there is no L interface frame relay-switched virtual connection for the mobile station, and the mobile station is not on a traffic channel supporting a packet data service option.

The mobile station may not be aware of the state of the link layer connection. When the relay layer R_m interface protocol option is selected, the MT2 is never aware of the link layer connection state. When the network layer R_m interface protocol option is selected, the MS is aware of the link layer connection state while it is opened and active, but it may not be aware of a possible IWF-initiated link layer connection closure while it is dormant.

The mobile station and BS/MSC send packet data on the traffic channel using the RLP. RLP can be carried either as primary or secondary traffic. The mobile station and the BS/MSC support the physical layer, multiplex options, radio management, and call control protocols. When a packet data service option is connected as primary traffic, and no service option is connected for secondary traffic all secondary traffic data are discarded. For more details, refer to TIA IS-657 [17].

10.7 Summary

Because wireless data networks do not operate without interconnection to other networks, in this chapter, we presented a variety of wireless data systems including the wide area wireless data systems and high-speed WLANs and the specific systems supported by CDMA. We examined the various standards being adopted by the IEEE, the WINForum, HIPERLAN (in Europe), and ARPA for wireless LANs. Since packet networks are an important part of wireless networks, we briefly stated the characteristics of the access methods in common use and defined their throughput equations. The common packet protocols are ALOHA, slotted-ALOHA, and CSMA/CD.

We then presented the methods used to control errors for wireless data systems. We concluded by presenting the highlights of the TIA IS-99, TIA IS-637, and TIA IS-657 standards for CDMA cellular systems. CDMA supports asynchronous data, facsimile, packet data, and short message service to end points in another wireless network or to the wireline network. We examined the reference models and protocol stacks for each of these data services.

10.8 References

1. Bates, R. J., *Wireless Networked Communication*, McGraw-Hill, New York, 1994.
2. Hammond, J. L., and O'Reilly, J. P., *Performance Analysis of Local Computer Networks*, Addison-Wesley Publishing Company, Reading, MA, 1986.
3. Habab, I. M., Kavehrad, M.,and Sundberg, C.-E. W., "ALOHA with Capture Over Slow and Fast Fading Radio Channels with Coding and Diversity," *IEEE Journal of Selected Areas of Communications*, 6, 1988, pp. 79–88.
4. Pahlavan, K., and Levesque, A. H., *Wireless Information Networks*, John Wiley and Sons, New York, 1995.
5. RFC 791, "Internet Protocol."
6. RFC 793, "Transmission Control Protocol."
7. RFC 792, "Internet Control Message Protocol."
8. RFC 1122, "Requirements for Internet Hosts—Communication Layers."
9. RFC 1144, "Compressing TCP/IP Headers for Low-Speed Serial Links."
10. RFC 1332, "The PPP Internet Protocol Control Protocol (IPCP)."
11. RFC 1661, "The Point-to-Point Protocol (PPP)."
12. RFC 1700, "Assigned Numbers."
13. TIA IS-95A, "Mobile Station—Base Station Compatibility Standard for Dual-Mode Wideband Spread Spectrum Cellular System."
14. TIA IS-99, "Data Services Option Standard for Wideband Spread Spectrum Digital Cellular System."
15. TIA IS-232-E, "Interface Between DTE and DCE Employing Serial Binary Data Interchange."
16. TIA IS-637, "Short Message Services for Wideband Spread Spectrum Cellular Systems."
17. TIA IS-657, "Packet Data Services Option for Wideband Spread Spectrum Cellular System."
18. TIA IS-687, "Data Services Inter-Working Function Interface Standard for Wideband Spread Spectrum Digital Cellular System."
19. Skalar, B., *Digital Communications–Fundamentals and Applications*, Prentice Hall, Englewood Cliffs, NJ, 1988.
20. Viterbi, A. J., and Padovani, R., "Implications of Mobile Cellular CDMA," *IEEE Communication Magazines*, 30, (12), pp. 38–41, 1992.
21. Walrand, J., *Communications Networks: A First Course*, Irwin, Homewood, IL, 1991.
22. Woo, I. and Cho, D.-H., "A Study on the Performance Improvements of Error Control Schemes in Digital Cellular DS/CDMA Systems," *IEICE Trans. Communications*, E77-B, (7), July 1994.

CHAPTER 11

Management of CDMA Networks

11.1 Introduction

In the preceding chapters, we discussed the elements of a cellular and PCS system with CDMA technology to provide service to a subscriber. In this chapter, we present those elements of the CDMA system that keep the system operating on a day-to-day basis. These elements are referred to as the operations, administration, maintenance, and provisioning systems. With OAM&P systems, cellular and PCS service providers monitor the health of all network elements, add and remove equipment, test software and hardware, diagnose problems, and bill subscribers for services.

In the past, management of telecommunications networks was simple. In the days prior to the deregulation and privatization of the telephone industry, the service provider dealt with fewer issues. Generally, the competitive pressure was less than it has been after deregulation. The networks were composed of equipment from fewer vendors, thus there were fewer multivendor management issues. As service providers moved into a mixed vendors' environment, they could no longer afford different systems for each network element. The standards groups in committee T1M1 and ITU defined the interface and protocols for operations support systems under the umbrella of the Telecommunications Management Network (TMN).

Service providers are deploying PCS into the existing cellular competitive environment. Therefore, they must offer cost-effective services. Even though management functions are necessary for the smooth opera-

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tion of a system, they are a cost of doing business and do not directly improve the bottom line of a business. Therefore, service providers must manage the cost of these systems. Considerations of initial cost of the management systems and their operational costs must be carefully examined. A key for the success of personal communications systems lies in the system operator's ability to manage the new equipment and services effectively in a mixed vendors' environment. For PCS to succeed, the OAM&P costs for PCS networks and services must be lower than those of either the existing wireless or wireline networks. Automation is required to develop machine-to-machine interfaces to replace many manual functions. The need to manage mixed vendors' equipment requires some form of standardization. Finally, the need to support rapid technological evolutions demands that the interfaces be general and flexible. Furthermore, to ensure that the interfaces have sufficient consistency to allow some level of integrated management, it is necessary to develop a set of guiding principles. That set of guiding principles is provided by TMN.

11.2 Management Goals for PCS Networks

The original cellular systems were purchased from equipment vendors as complete and proprietary systems. The interfaces between network elements were proprietary to each vendor, and network elements from one vendor did not work with network elements from another vendor. Cellular systems included management systems that were tailored to the vendor's hardware and software. Both cellular and PCS networks are migrating to a mixed vendor environment with open interfaces. Thus, a service provider can purchase switching equipment from one vendor, radio equipment from another vendor, and network management equipment from a third vendor. In this new environment, specific goals for network management hardware and software are necessary. These network management goals for PCS follow.

Operation in a Mixed Vendor Environment. The network management hardware, software, and data communications must support network elements from different vendors. It should allow for the integration of equipment from different manufacturers in the same TMN, by using clearly defined interworking protocols.

Availability of Multiple Solutions. Service providers should be able to purchase network management solutions from multiple and competitive vendors to ensure the lowest cost and richest feature set for the system.

Use of Existing Resources. Cellular companies and existing wireline companies should be able to manage their PCS network with minimal additions of new data communications, computing platforms, and so on. If possible, minimal additions should be made to existing distribution networks to implement the required control or reporting system.

Support for Multiple and Interconnected Systems. When a service provider has deployed multiple systems, the different management systems should interact effectively and efficiently to deliver service by sharing a common view of the network.

Support for Sharing System and Information Among Multiple Service Providers. Multiple service providers must interact to share information on billing records, security data, subscriber profiles, and so forth, and to share call processing (e.g., intersystem handoffs). The network management systems, therefore, should allow flexible telecommunication management relationships among multiple service and network providers of PCS. Flexible management relationships include complex network and service provider arrangements consistent with individual providers that operate as separate business entities. Wireless access providers might also want management access through intermediate networks.

Support for Common Solutions Between End Users and Service Providers. Many services require a joint relationship between the service provider and the end user. End-to-end data communication is the most common example of the need for a joint relationship. When service providers and end users are operating in this mode, their management functions should be interconnected. The solutions should allow flexible telecommunication management relationships between service providers and end users.

Transparent. A wireless network management system should be as transparent as possible to the technology used in wireless network implementation.

Flexible. The network management system should be flexible to allow for evolution in wireless network functions and services.

Modular. The network management system should be modular so that irrespective of the future network size or the location of control and knowledge, the management functionality can support all management aspects.

Fail-Safe. Neither equipment failure nor operator error should render the management system and/or the wireless network inoperative.

11.3 Requirements for Management of PCNs

A personal communication network is a telecommunications network where users and the terminals are mobile rather than fixed. In addition, there are radio resources that must be managed and do not exist in a wireline network. Furthermore, the network model can be a cellular model where the radio and switching resources are owned and managed by one company. A second newer model has the wireline company operating the standard switching functions and a wireless company operating the radio-specific functions. With this new mode of operation, network management requirements for personal communication networks include standard wireline requirements and new requirements specific to personal communication networks. The new personal communication network-specific requirements are needed for the following areas.

Management of Radio Resources. PCS allows terminals to connect to the network via radio links. These links must be managed independent of the ownership of the access network and switching network. The two networks may consist of multiple network service providers or one common service provider. The multiple operator environment will require interoperable management interfaces.

Personal Mobility Management. Another important aspect of personal communication networks is mobility management. Users are no longer in a fixed location but may be anywhere in the world. Users can be addressed by their identifications (IDs) without needing to take into account their current location or status. The network automatically finds them and correctly routes the call. The network also recognizes the originator's ID and delivers stored information that augments the user's identity. This will increase the load on the management system as users manage various decision parameters about their mobility. For example, they may request different services based on the time of day or terminal busy conditions.

Terminal Mobility Management. The primary focus of personal communication networks is to deliver service via wireless terminals. These terminals may appear anywhere in the worldwide wireless network. Single or multiple users may register on a wireless terminal. The terminal associated with a user's ID is tracked, perhaps across networks, in order to deliver messages or calls to the personal ID. Terminal mobility is the ability of a terminal to access telecommunications services from different locations while in motion. The terminal management function may be integrated with the personal management function or may be separate from the personal management function. Different service providers may operate their system in a variety of modes. The management function must support all modes.

Service Mobility Management. Service mobility is the ability to use today's vertical features from remote locations or while in motion. As an example, the user has access to the messaging service anywhere, anytime. The user also has access to a global locating service. The ability to specify an event is provided via a user interface that is flexible enough to support a number of input formats and media. A user can specify addressing in a simple, consistent way no matter where he or she is.

11.4 Telecommunication Management Network and Wireless Network Management

TMN includes a logical structure, which originates from data communication networks, to provide an organized architecture for achieving the interconnection between various types of operations systems and/or telecommunication equipment types. Management information is exchanged between operations systems and equipment using an agreed upon architecture and standardized interfaces, which include protocols and messages. The Open System Interconnect management technology has been chosen as the basis for the TMN interface. The principles of TMN provide for management through the definition of a management information model (managed objects), which is operated over standardized interfaces. This model is developed through the definition of a required set of management services that are then decomposed into various service components and then decomposed into management functions. An information model represents the system and supports the management functions.

Three important aspects of the TMN that can be applied to manage wireless networks follow:

- A layered architecture that has five layers: business management layer (BML), service management layer (SML), network management layer (NML), element management layer (EML), and network element layer (NEL).
- A functional architecture that defines functional blocks [operations systems functions (OSF), mediation function (MF), work station function (WSF), network element function (NEF), and Q adapter function (QAF)] is shown in figure 11.1.

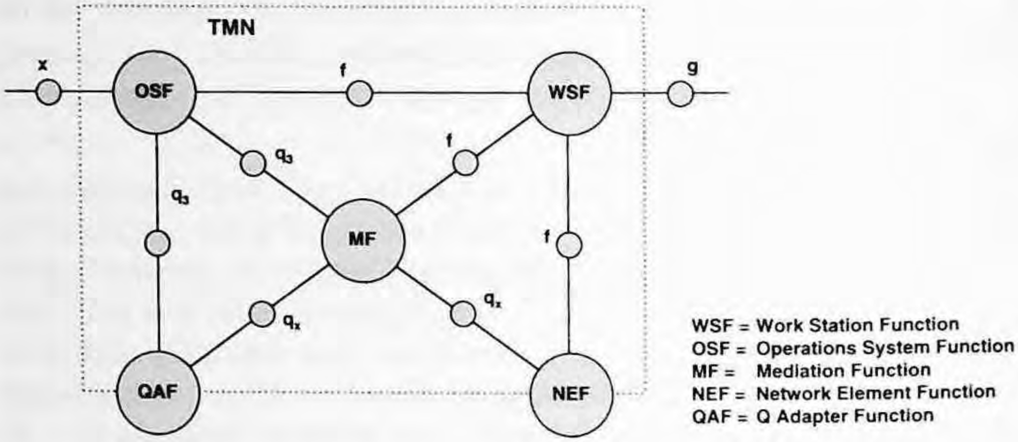

Figure 11.1 Functional architecture of TMN.

- A physical architecture that defines management roles for operations systems, communication networks, and network elements is given in figure 11.2.

TMN functions can be divided into two classes: basic functions and enhanced functions. *Basic functions* are used as building blocks to implement enhanced functions such as service management, network restoration, customer control/reconfiguration, and bandwidth management. TMN basic functions are further grouped into three categories.

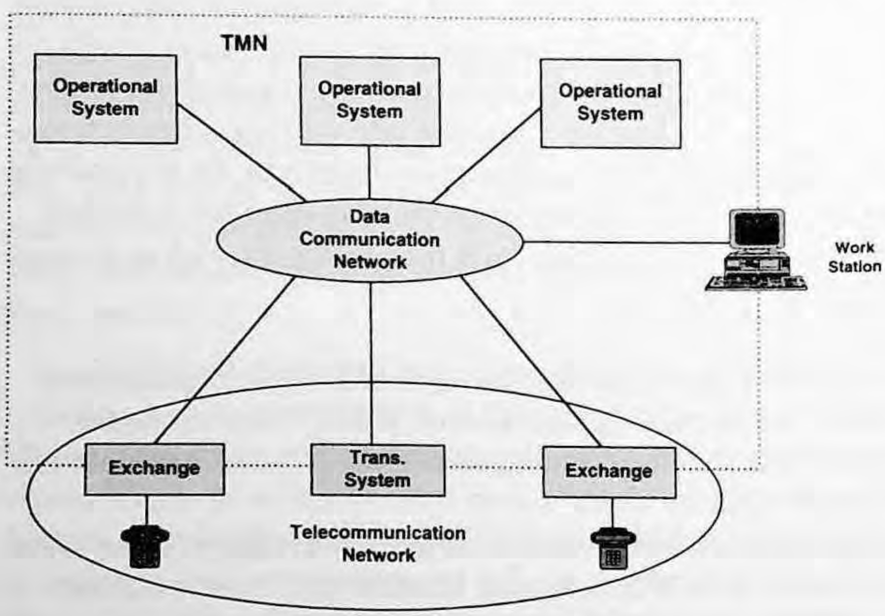

Figure 11.2 Physical architecture of TMN.

I. Management Functions

Accounting Management

Collect accounting (billing) data

Process and modify accounting records

Performance Management

Data collection

Data filtering

Trend analysis

Fault Management

Alarm surveillance (analyze, filter, correlate)

Fault localization

Testing

Security Management

Provide secure access to network element functions/capabilities

Provide secure access to TMN components (e.g., operations systems, SubNetwork Controllers (SNC), Mediation Device (MD), etc.)

Configuration Management

Provisioning

Status

Installation

Initialization

Inventory

Backup and restoration

II. Communication Functions

Operations System/Operations System Communications

Operations System/Network Element Communications

Network Element/Network Element Communications

Operations System/Workstation (WS) Communications

Network Element/WS Communications

III. Planning Functions

Network Planning, including physical resource (facility, equipment, etc.) planning, Workforce Planning.

11.4.1 Functional Architecture

The TMN architecture provides a degree of flexibility to support various and varying network topologies. The TMN functional and reference models define the key components of a management solution. The TMN models define a number of functional components that can be used by TMN designer and can be mapped onto physical TMN elements. The components of the functional architecture follow (refer to fig. 11.1).

Operations Systems Function. The OSF processes information related to telecommunications management for the purpose of monitoring, coordinating, and/or controlling telecommunication functions. The objects that are under the control of a given OSF are the components of the OSF management "domain." In certain cases, the management information may be partitioned into layers that can be hierarchically organized. In such cases, different OSFs may be responsible for the different layers that represent their respective management domains. This type of implementation is referred to as a "logically" layered architecture.

The physical implementation of OSFs must provide the alternatives of either centralizing or distributing the general functions including:

- Support application programs,
- Database functions,
- User terminal support,
- Analysis programs,
- Data formatting and reporting, and
- Analysis and decision support.

Reference Points. The relationships between components of the TMN functions are defined by reference points that allow identification of the boundaries between wholly self-contained functional units. The TMN standard interfaces follow:

- **q_3 Interface** supports the full set of OAM&P functions between operations systems and network elements and between operations systems and mediation devices. The q_3 interface may also be capable of supporting the sets of OAM&P functions between operations systems.
- **q_x Interface** supports a full set or a subset of OAM&P functions between mediation devices and network elements, between mediation devices, and between a network element with mediation function and another network element.

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- **x Interface** supports the set of operations systems to operations systems functions between TMN networks or between a TMN network and the x interface on any other type of management network.
- **f Interface** supports the set of functions for connecting workstations to operations systems, mediation devices, or network elements through a data communication network.
- **g Interface** is located outside of TMN and is between users and the workstation function.
- **m Interface** is located outside TMN and is between Q adapter functions and non-TMN managed entities.

Data Communication Function. The DCF is implemented via a data communication network (DCN). The DCN helps to connect the various TMN components with one another. The DCN is used when various functional groupings are implemented remotely from others. Each functional component contains a message communication function (MCF) to allow connection to the data communication function provided by the DCN.

Mediation Function. The MF primarily routes and/or acts on information passing between standardized interfaces. MFs can be located at network element(s) and/or operations system(s). The processes that can form mediation are classified into the following process categories:

- Communication control,
- Protocol and data conversion,
- Communication (passing) of primitive functions,
- Processes involving decision making, and
- Data storage.

Network Element Functions. NEFs communicate with the TMN for the purpose of being monitored and/or controlled. The NEF provides the telecommunications and support functions that are required by the telecommunications network being managed. The NEF includes the telecommunications functions that are the subject of management. These functions are not part of TMN but are represented to the TMN by the NEF. The part of the NEF that provides this representation in support of the TMN belongs to the TMN itself, whereas the telecommunication functions themselves are external to the TMN.

Q adapter Function. The QAF is used to connect those non-TMN entities, which are NEF-like and OSF-like to the TMN architecture. The job of the QAF is to translate messages between a TMN reference point and a non-TMN (e.g., proprietary) reference point.

Workstation Functions. WSFs are defined as the functionality that provides interaction between craft personnel and the OSFs.

11.4.2 Physical Architecture

An example of a simplified architecture for a TMN is shown in figure 11.2. TMN functions can be implemented in a variety of physical configurations. The relationship of functional blocks to TMN building blocks is given in table 11.1.

Table 11.1 Relationship of the Functional Blocks to TMN Building Blocks

Device	NEF	MF	QAF	OSF	WSF
Network Element	M	O	O	O	O
MD	—	M	O	O	O
QA	—	—	M	—	—
Operations System	—	O	O	M	O
WS	—	—	—	—	M

M = Mandatory; O = Optional

11.4.2.1 Layered Architecture

TMN uses a layered architecture; the major functionality of the five layers follow (refer to fig. 11.3).

Business Management Layer. The BML has the responsibility for the total enterprise and supports agreements between operators. This layer normally carries out goal-setting tasks rather than goal achievement but can become the focal point for action in cases where executive action is necessary. The BML is part of the overall management of the enterprise, and many interactions with other management systems are necessary.

Service Management Layer. The SML is concerned with, and responsible for, the contractual aspects of services that are being provided to customers or available to potential customers. The five principle roles of the SML are:

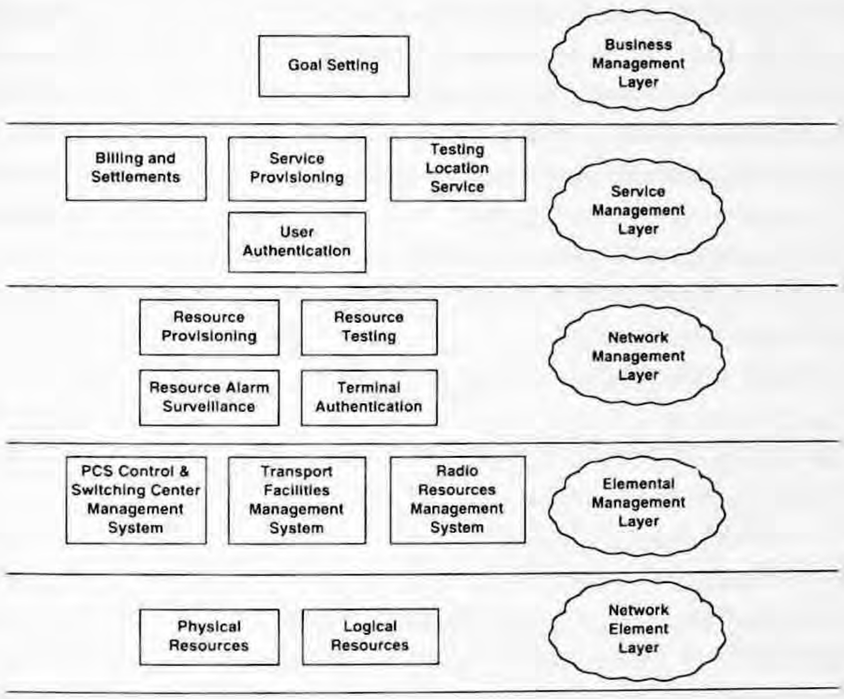

Figure 11.3 Layered architecture of TMN.

- Interfacing with customers and other administrations,
- Interacting with service providers,
- Interacting with the network management layer,
- Maintaining statistical data (e.g., quality of service), and
- Allowing interactions between services.

Network Management Layer. The NML is responsible for the management of all the network elements, as presented by the element management layer, both individually and/or as a set. This layer is not concerned with how a particular element provides service internally. Functions addressing the management of a wide geographic area are located at this layer. The whole network is completely visible, and a vendor independent view is maintained. The three principle roles of the NML are: (1) the control and coordination of the network view of all network elements within its domain, (2) the provision, cessation, or modification of network capabilities for the support of services to customers, and (3) interaction with the SML on performance, usage, availability, and so on. The NML provides the functionality to manage a network by coordinating all activities across the network and supporting the network demands made by the SML.

Element Management Layer. The EML manages each network element on an individual basis and supports abstraction of the functions provided by the network element layer. The EML has a set of element managers that are individually responsible for some subset of network elements. Each element manager has three principal roles:

- Control and coordinate of a subset of network elements,
- Provide a gateway (mediation) function to allow interactions among network elements, and
- Maintain a statistical log and other data about elements.

Network Element Layer. The NEL consists of logical and physical resources to be managed.

Information models to support the functions identified for various layers are developed in the O interface (refer to fig. 11.4) standards. These models will then be used to exchange management information across interfaces according to the particular physical architecture of the managed network. The information models for different levels of abstraction provide the required flexibility in developing implementation to support the NEL, EML, NML, SML, and BML functions.

The following five EML functions simplify management of a wireless network.

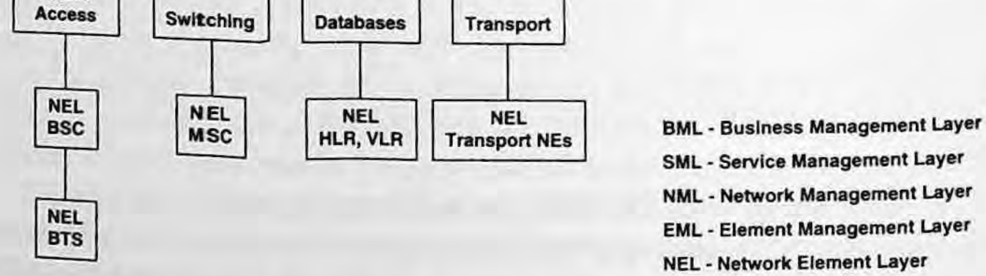

Figure 11.4 Layered architecture as applied to wireless network.

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- **Aggregated Resource Provisioning.** This function allows NML (or other EML) applications to request certain resource provisioning functions on a collection of NEL resources within its span of control. This function performs these tasks on individual resources. This may result in simplification of the provisioning process.
- **Alarm Correlation and Filtering.** In many situations, a single resource failure may be the source of failures in multiple related resources. To reduce the burden on the network level alarm surveillance tasks, the notifications of these failures may be correlated and filtered, emitting only a single alarm from the EML.
- **Software Management.** EML software management functions enable the NML applications to manage wireless software in a more generic way, while the EML provides software management for wireless network elements within its span of control.
- **Auto Resource Discovery.** The EML should be able to detect the presence of newly added resources, such as the addition of a BTS to a BSC or changes in equipment (e.g., memory and power supply).
- **Network Resource Inventory Management.** This function provides management of the resource inventory for the wireless network elements within its span of control.

11.4.3 Quality of Service

The objective of a wireless network management system is to integrate the wide range of wireless network operator activities in order to achieve coherent and seamless information exchange at a given quality of service (QoS) objective while achieving business objectives. The wireless network operator typically establishes a QoS criteria and objective in the context of the service levels to be provided to customers and with knowledge of the performance of the network infrastructure. It is necessary to compare these objectives and expectations with experience gained by monitoring the performances of services according to the level of customer complaints. The operator must also monitor the technical performance of the network in order to initiate improvements in QoS. Even though the technical performance of the network can be monitored, it does not necessarily reflect the service performance that the customer sees.

ITU-T recommendations I.350 and I.140 provide the general aspects of QoS and network performance in digital networks, including ISDN. These recommendations are applicable to a PCS network, as PCS use both digital networks and ISDN and may also provide users with digital data services in the future.

The ITU-T recommendations provide descriptions of QoS and network performance parameters. In the ITU-T Recommendation I.350, QoS aspects are restricted to the parameters that can be directly observed and measured at the point at which the service is accessed by the customer. Network performance is measured in terms of the parameters that are meaningful to the service provider. Network performance parameters are used for system design, configuration, operation, and maintenance.

11.5 Accounting Management

Accounting management refers to the usage information generation and processing function that is used to render bills to the customers. It includes the distributed function that measures usage of the network by subscribers or by the network itself (e.g., for audit purposes). It also manages call detail information generated during the associated call processing to produce formatted records containing this usage data. The formatted accounting management records include billing systems, operation systems for performance management and fraud detection, and subscribers' profiles. The primary objective of the accounting management data is to render a bill to the customer using the wireless service. The customer of services may be another network or the end user.

Accounting for resource utilization consists of three subprocesses (see fig. 11.5):

- **The Usage-Metering Process** is responsible for the creation of usage-metering records generated by the occurrence of accountable events in the system (i.e., calls originated or received and requests for supplementary services). In general, use of a service requiring multiple resources will generate several usage-metering records.
- **The Charging Process (or Rating Process)** is responsible for collecting the usage-metering records that belong to a particular service transaction and for combining them into service transaction records (i.e., call detail records). Also, pricing information is added to the service transaction records. The rating process is also responsible for logging the service transaction records.
- **The Billing Process** is responsible for collecting the service transaction records, for selecting the records that belong to a particular subscriber during a particular time-period, and for producing the bill.

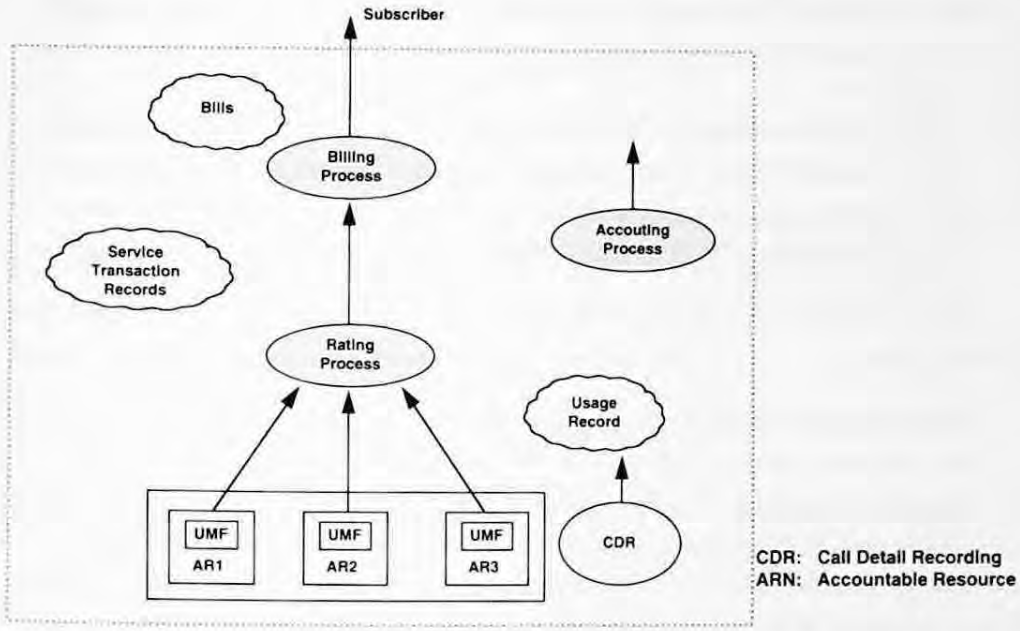

Figure 11.5 The accounting process.

11.5.1 Billing Data Management

Call and event data are required for a number of network management activities including, but not limited to, the following:

- Billing of home subscribers, either directly or via service providers, for network utilization charges;
- Settlement of accounts for traffic carried or service performed by fixed network and other operations;
- Settlement of accounts with other wireless networks for roaming traffic;
- Statistical analysis of service usage; and
- As historical evidence in dealing with customer service and billing complaints.

To support these activities, the accounting management system should support the following functions over the O interface:

- **Usage Management Functions**—usage generation, usage edit and validation of call events, usage error correction for call events, usage accumulation, in-call service request, service usage correlation, usage aggregation, usage deletion, usage distribution

- **Accounting Process Functions**—usage testing, usage surveillance, management of usage stream, administration of usage data collection
- **Control Functions**—tariff administration, tariff system change control, tariff class management, advice of charging management, data generation control, partial record generation control, data transfer control, data storage control, emergency call reporting

The PCS/cellular accounting management must also address the following items:

- **Distributed Collection of Usage Data.** Since many services will be "multinetwork services," it will be difficult for a single node (such as a switch) to generate a complete record of a call, as is done today. An example is roaming of mobile subscribers. This may involve multiple network nodes, possibly belonging to different service providers.
- **Improved Performance for Billing Collection and Report Generations.** Usage data are expected to be transmitted to a billing operations system in near-real time. There also will be concern about data concurrency and latency restrictions. These issues will pose significant performance requirements.
- **Multitude of Charging Strategies.** The usage data collected for billing must be flexible enough to support a variety of charging strategies. In some cases, the consumer will pay directly; in other cases the provider will pay. Charges for a call may be split between calling and called parties. In the case of multiple providers, the consumer may deal with only one provider (such as 900 services). Different legs of a call may receive different charging treatment.
- **Rapid Introduction of Diverse Services.** The accounting management structures must allow for the timely introduction of additional formats as services, technology, and pricing strategy evolution. When adding new formats, backward compatibility should be maintained.

11.5.2 Data Message Handling and TMN

Figure 11.6 shows the reference model for the data message handler (DMH) as defined in TIA IS-124, [9]. The DMH consists of the call detail information source (CDIS), call detail generation point (CDGP), call detail collection point (CDCP), and call detail rating point (CDRP). The functions of these components follow:

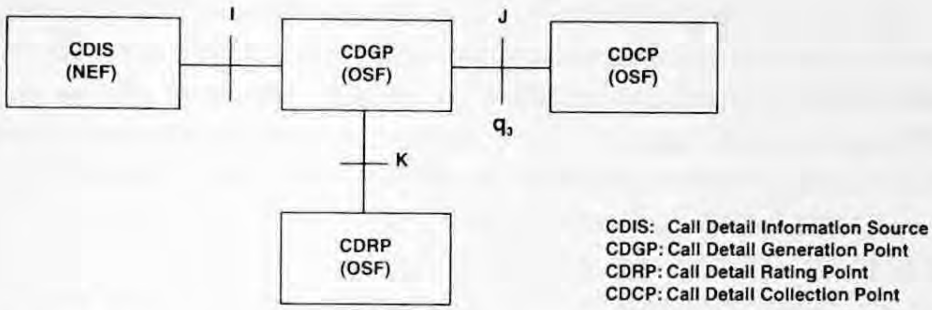

Figure 11.6 DMH reference model and mapping to TMN.

- **The CDIS** may be a PCS/cellular switching center, a radio port controller (RPC), a radio access system controller (RASC), a terminal mobility controller (TMC), a personal mobility controller (PMC), or any other source of call detail information. The CDIS provides call information without additional services or call detail information storage. In TMN, the CDIS function is associated with the NEF.
- **The CDGP** is responsible for collecting call detail information and encoding the messages for delivery to a CDCP. The CDGP may edit the collected call detail information as necessary to perform operations including reducing redundant information, discarding inconsistent records, eliminating data not required by the collection point, merging related call information, or adding rate information. The CDGP is also responsible for addressing the messages to the proper CDCP. Thus, it provides a distribution of usage data based on the service user. The CDGP stores call detail information until they are received and reconciled by the CDCP. These functions are OSF in TMN.
- **The CDCP** receives the call detail records. A CDCP may use or process call details. Examples of services are subscriber billing, intersystem charging, net settlement, and subscriber fraud detection. A CDCP may perform other record edits as necessary. The CDCP performs functions normally found in a billing system. In the context of the TMN, a CDCP performs OSF.
- **The CDRP** calculates and returns the currency information based on call detail information. In a public switched telephone network (PSTN) environment, this function is referred as tariffing. In the PSTN, the CDRP is frequently associated with the billing system and performs as an OSF.

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The DMH is intended not only for subscriber billing but also to improve wireless services management such as fraud management, credit monitoring, performance management, and network planning. The DMH requires that real-time data on the elapsed air-time and tandem-time from all switches involved in a call be sent to the control point at call completion. The control point is the anchor system switch that makes the first radio contact with the subscriber directly or through other switches during the call. At least two call records are generated for a call delivered to a roaming subscriber. The home (or originating) system has the data about the call between the originating system and the system currently serving the subscriber. The visited or serving system directly records data about the subscriber's location, use of air-time, usage of features, etc. These records are correlated to generate a consolidated bill of a call event regardless of the number of cellular/personal communications systems involved in this call. The DMH specifies five types of call records: audit records, leg records, segment records, activity report records, and event report records. These records contain information about different aspects and portions of a call such that each can be cross-referenced to the other in order to generate the complete record for further inter administration processing. In figure 10.6, a mapping of the DMH reference points I, J, and K and processing entities CDIS, CDGP, CDRP, and CDCP to possible TMN reference points and functions is also given. At present, the I and K reference points defined in the DMH are proprietary interfaces. The data transferred across the J reference point are specified based upon an open interface.

The information collected by the DMH CDGP follows:

- **Chronology**
  - ✘ *Time of Day*
  - ✘ *Time Day Offset*—offset in minutes of local civil time with respect to universal coordinated time
  - ✘ *Date*
- **Usage Measurement**
  - ✘ *Duration*—measures the duration between a beginning event and an ending event in tenths of seconds.
- **Digits**
  - ✘ *Account Code*

- ✘ *Billing Digits*—specify and identify the party to be billed under specified billing arrangements
- ✘ *Alternate Billing Digits*—identify the party to be billed under special billing arrangements
- ✘ *Called Digits*—specify and identify the called party
- ✘ *Calling Digits*—specify and identify the calling party
- ✘ *Destination Digits*—specify a telephone directory number toward which a call is routed over a network
- ✘ *Dialed Digits*
- ✘ *Interaction Digits*—specify digits dialed by the calling party in response to a voice or tone prompt
- ✘ *Routing Digits*—specify the digits used to provide a special routing for a call to access a trunk, steer a call, or specify a private network hop-off

• **Identifier**

- ✘ *Carrier Digits*
- ✘ *Billing Identification Number*—specify the number assigned at the anchor element, which was originally serving when it requested the call or was offered a call from the network
- ✘ *Electronic Serial Number*
- ✘ *Mobile Identification Number*
- ✘ *Related Billing Identification Number*
- ✘ *Report Identification Number*
- ✘ *Calling Party Indicator*
- ✘ *Calling Party Category*
- ✘ *Business Relation Identification*—specify a special billing relationship for purposes of cost allocation, revenue sharing, or similar purposes

• **Location Identifier**

- ✘ *H coordinate*
- ✘ *V coordinate*
- ✘ *Place Name*
- ✘ *North American Jurisdiction*
- ✘ *NPA-NXX*—6 digits to represent the rating point used by the serving point
- ✘ *Location Area Identifier*
- ✘ *Latitude*
- ✘ *Longitude*

- ✘ *Resolution*—specify the resolution of latitude and longitude measurements in feet

- **Network Resources**
  - ✘ *Switch Number*
  - ✘ *System Identification*
  - ✘ *Feature Bridge Identification*
  - ✘ *Trunk Group Identification*
  - ✘ *Trunk Number*
  - ✘ *Transceiver Number*
  - ✘ *CDMA Station Classmark*
  - ✘ *FDMA Station Classmark*
  - ✘ *TDMA Station Classmark*
- **Features and Services**
  - ✘ *Bearer Indicator*
  - ✘ *Feature Bridge Indicator*
  - ✘ *Feature Indicator*
  - ✘ *Feature Operation*
  - ✘ *Services Used Indicator*
- **Charging and Rating**
  - ✘ *Billing Indicator*
  - ✘ *Charge Amount*
  - ✘ *Charge Indicator*
  - ✘ *Charge Tax*
  - ✘ *Charge Tax Indicator*
  - ✘ *Charge Units*
  - ✘ *Multiple Rating Period Indicator*
  - ✘ *Rate Period Indicator*
  - ✘ *Toll Tariff Indicator*
- **Event Indicator**
  - ✘ *Authorization Type*
  - ✘ *Authorization Count*
  - ✘ *Event Indicator*
  - ✘ *Feature Result Indicator*
  - ✘ *Release Indicator*
  - ✘ *Redirection Indicator*
  - ✘ *Priority Access Indicator*

- **Traffic Measurement (see section 11.9.5)**
- **Fraud Indicator**
    - ✘ *Credit Limit*
    - ✘ *Call Count*
    - ✘ *Number of Calls/Hour*
    - ✘ *Call Patterns*

11.6 Security Management

Security management contains a set of functions that control and administer the integrity, confidentiality, and continuity of telecommunication services against security threats. This set of functions support the application of security policies and audit trails by controlling security services and mechanisms, distributing security-related information, and reporting security-related events.

Security management does not address the passing of security-related information in protocols and services that call up specific security services (e.g., in parameters of connection requests).

11.6.1 Information-Gathering Mechanisms

It is desirable to record the occurrence of various security events. Depending on the type of information, frequency of occurrence, and importance of the event, one of several mechanisms may be used to record the occurrence:

- Scanners to collect and periodically report measurement information on high-frequency or low-importance events;
- Counters, allowing for the definition of threshold crossing and notification severity;
- Security alarms for high-importance and/or infrequent events.

11.6.2 Audit Trail Mechanism

Some security events that occur during the life of a system may need to be reviewed immediately, and immediate actions may need to be taken. For other events, it may be useful to review the history in order to identify patterns of failures or abuse. It is recommended that these data be maintained in a log that holds security audit records. This log may be kept either at the agent or at the manager side.

11.6.3 Security Alarm Reporting Mechanisms

The manager needs to be alerted whenever an event indicating a potential breach in security of the wireless network is detected. This detection may be reported by an alarm notification. The security alarm report should identify the cause of the security alarm, its perceived severity, and the event that caused it. All requirements relative to storage and forwarding of alarm notifications identified under fault management are assumed to apply to security alarms as well.

11.7 CONFIGURATION MANAGEMENT

Configuration management contains a set of functions that are used to control, identify, and collect data from/to the network elements. A wireless network operator needs to have an overview of the entire network including but not limited to the following:

- Hardware, firmware, software, and combinations that are compatible with each other.
- Hardware, firmware, and software that are currently used in each network element.
- Frequency plan for the wireless network and the frequency allocation for each cell.
- Coverage plan and the coverage area of each base station.

Configuration management is concerned with monitoring and control that relate to the operation of units in the system. It deals with initial installation, growth, or removal of system components. Configuration management supports the following management functions.

Creating Network Elements and Resources. The creation of a network element is used initially to set up a wireless network or to extend an already existing network. The action of creation includes a combination of installation, initialization, and introduction of the newly installed equipment to the network and to the operations system that will control it. The creation can affect equipment, software, and data. Whenever a wireless network or parts of it are installed, the created network elements are required to be:

- Physically installed and tested and then initialized with a possible default configuration;

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- Logically installed by means of introduction to the network possibly involving changes to existing network element configuration;
- Put into service.

The management system must support mechanisms for user-friendly identification of these elements and resources. It should be possible to associate information such as resource name, location, description, and version with logical or physical elements.

The sequence of physical and logical installation may vary depending on the specific wireless network operator strategy. In case the logical creation occurs before the physical creation, no related alarms should be reported to the operator.

Deleting Network Elements and Resources. If a network is found to be overequipped, the operator may wish to reduce the equipment or to reuse the spare equipment elsewhere. This situation can occur when an operator overestimates the traffic in one area and underestimates the load in a different area. The deletion of a network element requires:

- Taking the affected network elements out of service;
- Logically removing it from the network (possibly involving changes to other network element configurations, for example, the neighbor cell description);
- If necessary, physical dismantling the equipment;
- Returning other affected network elements to service.

The sequence of logical and physical removal will not matter if the affected network elements are taken out of the service prior to their removal. This will help to protect the network from error situations.

Conditioning Network Elements and Network Resources. When a network element is to be modified, the following actions should be performed:

- Logical removal,
- Required modification, and
- Logical reinstallation.

This sequence is recommended to protect the network against fault situations that may occur during the modification process. The result of conditioning should be able to be determined by the operator by using the appropriate mechanisms.

A modification to data that has a controlling effect on some of the resources could affect the resource throughput or its capability to originate new traffic during the modification period. This should be evaluated for particular modifications because the capacity of the network element can be reduced without affecting the ongoing traffic. The forecast of a modification on capacity, throughput, and current activity of a resource will help the operator of a network to decide when the modification should be carried out.

The data characterizing a wireless network will not all be subject to the same rate of change or need to be modified using the same mechanism. Changes to the logical configuration may also need to be applied across multiple network elements.

A major aspect of configuration management is the ability of the operator to monitor the current configuration of the network. This is necessary to determine the operational state of the network and to determine the consistency of information among various network elements. The monitoring capability requires the information request function, the information report function, and the response/report control function.

- **Information Request Function.** The network operator should monitor the network in order to support its operation. The operations system should be able to collect information as needed from various network elements. The operations system should be able to request information for any single attribute defined in a management information base. Also, the operations system should be able to collect a large amount of data in a single request by providing an appropriate scope and filter in the request. On receipt of a valid request, the addressed network element must respond with the current values of the specified data elements. This response should be immediate if so requested by the operations system. However, in the cases where very large amounts of data are concerned and where the operations system and the network element support the capabilities, the operations system may request the network element to store the data in a file and transfer it using a file transfer mechanism.
- **Information Report Function.** A network element should have the capability of reporting information autonomously. This will be done when some information on the state or operation of the system

has changed. For appropriate events in the system, a network element must be able to identify the notification as an alarm and be able to indicate the severity and cause of the condition in the report. Notifications may be logged locally. Logged notifications may be requested to be transferred using the defined file transfer mechanism.

- **Response/Report Control Function.** For responses to information requests and for information reports, it should be possible for the network operator to specify where and when the information should be sent. The operations system and network element should have the capability to configure the response/reporting to meet the following requirements:

    - ✘ Information forwarding must be able to be enabled and disabled,
    - ✘ Information must be forwarded to the operations system as soon as it is available,
    - ✘ Information must be able to be directed to any of the various operations systems,
    - ✘ Information must be able to be logged locally by the network element and optionally by the operations system, and
    - ✘ Information must be retrieved from logs using appropriate filtering specifications.

11.8 Fault Management

Fault management consists of a set of functions that enable the detection, isolation, and collection of an abnormal operation of a network element reported to an operations system. Fault management uses surveillance data collected at one or more network elements. The collection and reporting of the surveillance data are controlled by a set of generic functions. Fault management functions for operations system/network element interfaces deal with maintaining and examining error logs, reporting error conditions, and localizing and tracing faults by conducting diagnostic tests. The fault management supports the following management service components:

- **Alarm Surveillance.** This service deals with managing information in a centralized environment about service affecting performance degradation. Alarm functions are used to monitor or interrogate network elements about events or conditions.

- **Fault Localization or Identification.** This service requires that the management system have capabilities for determining which unit is at fault. The repair or replacement of the faulty unit can then be done to restore normal operation of the system. Such repair or replacement may be accomplished automatically by the agent, by the operations system, or manually by craft. The first step in this process is to identify the faulty unit as part of an alarm notification or test result. The alarm notifications and test results should contain identification of the repairable/replaceable unit whenever possible. Even though it is not possible to identify a specific repairable/replaceable unit, a list of potential faulty units should be provided so that additional diagnostic tests may be carried out to further localize the fault. For alarm notifications, the first localization information is provided by the identification of the object instance reporting the alarm. Second, the alarm "Probable Cause" and, optionally, "Specific Problem" values will provide the wireless network provider or manufacturer with the information that will help to localize the fault to a specific replaceable/repairable unit.
- **Fault Restoration, Correction, or Recovery.** The correction of faults should be either through repair of equipment, replacement of equipment, software recovery, logic, or cessation of abnormal conditions. If a network element uses an automatic recovery action, the management system should support the capability of the network element to notify the operations system of the changes that were made. Automatic recovery mechanisms should take system optimization into account. The operator should be able to shut down and lock out a resource before taking any fault management action. The operations system sends a request to the network element indicating the network element resources to be shut down. The network element locks out the indicated resource immediately, if it is not active. If the specified resource is active, then the lock-out is delayed until all activities are ended or until a lock-out is requested by the operations system. The fault management system should also support the establishment and selection of previous versions of software and databases (e.g., backup, and fall-back) in order to support recovery actions.
- **Testing.** Periodic scheduled and on-demand tests are used to detect faults in the system. On-demand tests are also used to assist in localizing a fault to one or more repairable/replaceable units. On-demand tests are also conducted to verify a replaceable unit before

placing it into service. In-service tests may be used to gather information related to a particular wireless subscriber's activities within a network. Such data might include information related to successful and failed registrations, call attempts, handoffs, and the like. Records of these activities or events may need to be correlated across network boundaries in order to diagnose problems related to delivery of the service. Testing of wireless components should be automated to the extent that human interface in the testing process is minimized. The testing capabilities should support the running of a single test or multiple tests in parallel. Control capabilities are required to simulate the same type of test to be run with different control parameters on the same or on different system units. Once started, tests must run to completion. For controllable tests, it should be possible to monitor the operation, to suspend and resume the operation, as well as to terminate the tests either gracefully or abruptly. General requirements for tests and test management can be found in ITU X.737 and ITU X.745 specifications.

- **System Monitoring and Fault Detection.** The fault management of the network element requires capabilities to support the recognition and reporting of abnormal conditions and events. Detection also addresses activities such as trend analysis, performance analysis, and periodic testing. Fault management requires that the operations systems have a consistent view of the current state and configuration of the system that is being managed. The management system should be able to request current information about the state and configuration of the system in the form of solicited reports. The local OAM&P activities should be reported to the operations system as soon as practical. Whenever possible, a network element should generate a single notification for a single fault. A network element should support the local storage of fault information. This stored information should be accessible and erasable from the operations system. Network elements should support capabilities to allow the request and reporting of current alarm status information. This should indicate units with alarms outstanding and their severity. Fault management should have the capability to report alarms to identify failures in equipment (hardware and/or software), databases, or environmental conditions. Information about faults should be stored in the operations system for statistical verification/estimation of equipment reliability. The alarm-reporting functions are described in ITU X.733. They fall into four general areas: the generation of an event notification by a managed resource; the forwarding of that event notification to a management application; storage (temporary) of the event record; and alarm status monitoring capability.

11.9 Performance Management

Performance management functions deal with a continuous process of data collection about the grade of service, traffic flow, and utilizations of the network elements. It does not affect the service provided to the wireless customer. Performance monitoring is designed to measure the overall service quality to detect service deterioration due to faults or to planning and provisioning errors. Performance monitoring may also be designed to detect characteristic signal patterns before signal quality has dropped below an acceptable level. The performance monitoring is sensitive to the sampling scheme of the monitored signal and/or choice of signal parameters calculated from the raw signal data.

Wireless network system behavior requires that performance data be collected and recorded by their network elements according to the schedule established by the operations system. This aspect of the management environment is referred to as performance management. The purpose of the *performance management* is to collect data that can be used to evaluate the operation of the system to verify that it is within the defined QoS limits and to localize potential problems as early as possible. Data are required to be produced by the network elements to support the following areas of performance evaluation:

- Traffic levels within the wireless network, including the level of both user traffic and signaling traffic;
- Verification of network configuration;
- Resource access measurements;
- Quality of service (e.g., delays during call setup); and
- Resource availability.

The production of the measurement data by the network elements should be administered by the operations system. Phases of administration of performance measurements are

- The management of the performance measurement collection process;

- The generation of performance measurement results;
- The local storage of measurement results in the network element;
- The transfer of measurement results from network element to an operations system; and
- The storage, preparation, and presentation of results to the craft.

11.9.1 Requirements on Types of Data

Typical requirements for the types of performance data to be produced by the network elements of a wireless network are discussed next.

Traffic Measurement Data. Traffic measurement data provide the information from which the planning and operation of the network can be carried out. They include

- Traffic load on radio interface (signaling and user traffic), where measured values may include pages per location area per hour, busy hour call attempts, and handoffs per hour;
- Usage of resources within the network nodes; and
- User activation and use of supplementary services.

Network Configuration Evaluation Data. Once a network plan or changes to a network plan have been implemented, it is important to be able to evaluate the effectiveness of the plan or planned changes. The measurements required to support this activity will indicate the traffic levels with particular relevance to the way the traffic uses the network.

Resource Access Data. For accurate evaluation of resource access, each count must be produced for regular time intervals across the network or for a comparable part of the network.

Quality of Service Data. QoS data indicate the wireless network performance expected to be experienced by the user.

Resource Availability Data. The availability performance is dependent on the defined objectives (i.e., the availability performance activities carried out during the different phases of the life cycle of the system) and on the physical and administrative conditions.

11.9.2 Measurement Administration Requirements

Measurement administration functions allow the system operator to manage measurement data collection and forwarding to an operations system.

Measurement Job Administration. The processes to accumulate the data and assemble them for collection and/or inspection should be scheduled for the period or periods for which collection of data should be performed. The administration of measurement job consists of the following actions:

- Create/delete a measurement job;
- Modify the characteristics of a measurement job;
- Define measurement job scheduling;
- Report and route results (to one or more operations systems);
- Suspend/resume an active measurement job; and
- Retrieve information related to measurement jobs.

Measurement Result Collection Method. The measurement data can be collected in each network element of the wireless network in a number of ways:

- Cumulative incremental counters triggered by the occurrence of the measured event;
- Status inspection (i.e., a mechanism for high-frequency sampling of internal counters at predefined rates);
- Gauges (i.e., high value mark, low value mark); and
- Discrete event registration, where data related to a particular event are captured.

Local Storage of Results at the Network Element. It should be possible for the network element to retain measurement data it has produced until they are retrieved by the operations system. These data should be retained at the network element as an explicit request from the operations system. The storage capacity and the duration for which the data should be retained at the network element will depend upon the network design.

Measurement Result Transfer. The results of the measurement job can be forwarded to the operations system when available or be stored in the network element and retrieved by the operations system when required. The measurement result can be retrieved from the network element by the operations system on request. The network element should return the current value of the measurement job with any related information but should not affect the scheduled execution of this or any other measurement jobs actively reporting that same data item.

11.9.3 Requirement on Measurement Definition

The measurements defined for the wireless network are collected at the network elements. Each network element plays its own role in the provision of the wireless telephony service and has a different perspective on the performance of the network. The measurement definition should contain a description of the intended result of measurement in terms of what is being measured.

The definition of a measurement should accurately reflect which types of events are to be included in the collection of data. If a general event description can be characterized by several subtypes, then the measurement definition should be precise as to which subtypes are included or specifically excluded from the measurement.

In a multivendor network, it is important to ensure that measurement data produced by a network element from one supplier is equivalent to the measurement data being produced by the equivalent network element from another supplier. This is particularly important when analyzing data across the whole network.

In a complex wireless network, it is easy to generate large amounts of performance data. It is essential that all data are recognizable with respect to each request.

11.9.4 Measurement Job Requirements

The measurement schedule specifies the time frame during which the measurement job will be active. The system should support a job start time of up to at least 90 days from the job creation date. If no start time is specified, the measurement job should become active immediately. The measurement job remains active until the stop time (if supplied in the schedule) is reached. If no job stop time is specified, the measurement job should run indefinitely and can be stopped only by manual intervention. The time frame defined by the measurement schedule may contain one or more recording intervals. These recording intervals may repeat on a daily and/or weekly basis and specify the time periods during which the measurement data are collected with the network element.

The granularity period is the time between the initiation of two successive gatherings of measurement data. Required values for the granularity period are 5, 15, 30, and 60 minutes. The minimum granularity period is 5 minutes in most cases, but for some measurements it may be necessary to collect data for a larger granularity period. The granularity period should be synchronized on the full hour and its value not changed during the lifetime of the job.

Scheduled measurement reports are generated at the end of each granularity period. All reports generated by a particular measurement job should have the same layout and contain the information requested by the system operator. The information may contain:

- An identification of the measurement job that generated the report;
- An identification of the involved measurement type(s) and measured network resource;
- A time stamp, referring to the end of the granularity period;
- The result value and indication of the validity for each measurement type; and
- An indication that the scan is not complete and the reason why the scan could not be completed.

Scheduled measurement reports generated at the end of each granularity period, if the measurement job is not suspended, can be transferred to the operations system in two ways:

- The reports are automatically forwarded to the operations system at the end of the granularity period (e.g., immediate notifications).
- The reports are stored locally in the network element, where they can be retrieved when required.

11.9.5 Performance Measurement Areas

The areas of performance measurement follow:

- **Traffic and Signaling Data Collection in Network Elements,** (e.g., the base station, mobile switching center, and home location register).
- **Quality of Service Data Collection.** QoS data are collected for dropped calls, connection establishment, connection quality for mobile services, connection quality for data services, call/event record integrity, and quantity per network management. For QoS, the following measurements should be taken:
  - ✘ Delay to provide a new service;
  - ✘ Delay to change subscriber data;
  - ✘ Operator complaints about user type, service type, call type, and the like;

- ✘ Delay to re-establish a service; and
- ✘ Customer service promptness.

- **Availability Measurements.**
- **Network Performance Measurements.** The following measurements should be used:
  - ✘ Percentage overflow (%OFL) (count used whenever the cell site cannot allocate a traffic channel for setting up a call),
  - ✘ Answer seizure ratio (ASR),
  - ✘ Answer bid ratio (ABR),
  - ✘ Mean holding time per seizure, and
  - ✘ Abnormal termination ratio.
- **Call Destinations.** The MSC should be able to monitor call destinations as country codes, area codes, exchange codes, or any combination of them. The entities that should be measured on call destinations for network management purposes are
  - ✘ Attempts per destination per hour,
  - ✘ Seizures per destination per hour,
  - ✘ Answers per destination per hour, and
  - ✘ Count of calls affected by network management control (controls on destination).
- **Radio Interface Traffic Measurement.** The air interface is the most traffic-sensitive area in the mobile network. It is important that the maximum feasible number of measurements on the radio interface be taken. The following measurements are recommended:
  - ✘ Amount of signaling leading to unsuccessful events,
  - ✘ Congestion probability on mobile traffic channels,
  - ✘ Lost call probability due to handoff failure,
  - ✘ Call re-establishment probability, and
  - ✘ Loads on the air interface.
- **Processing Throughput Measurements.** The level of processing throughput in terms of busy hour call attempts or transaction rates versus delay times should be monitored on the MSC, BS, HLR, and SS7 signaling links to allow for overload criteria to be used with appropriate network management actions to be taken. Throughput is very important in the mobile network due to extra processing required when compared to the fixed networks because of handoff, location updates, authentication, and the like.

- **Handoff Statistics.** Handoffs add to the processing and signaling load of the mobile system. From the network management viewpoint, the following data should be collected:
  - ✘ Handoff counts per call;
  - ✘ Success rate of handoff attempts; and
  - ✘ Reasons for failure and retries.

  In collecting the handoff data, the following should be recorded:
  - ✘ The handoff statistics should be qualified with time of day, cells or location areas, or similar measurements, as selected by the mobile operator;
  - ✘ Statistics concerning the handoff parameters, in order to be able to modify and tune the current handoff algorithm later on;
  - ✘ Statistics about the causes of the handoff and their corresponding success rates;
  - ✘ Intra-BS handoffs;
  - ✘ Inter-BS (intra-MSC) handoffs;
  - ✘ Inter-MSC handoffs; and
  - ✘ Inter-mobile networks handoffs.
- **Connection Establishment and Retention.** The following data should be taken to evaluate this measurement:
  - ✘ Percentile of ineffective calls:
    - (a) Mobile originated due to congestion in mobile network, congestion in attached/adjacent network, excessive delay in mobile network, excessive delay in adjacent network, premature release by calling MS during call setup, and called PSTN number not answering.
    - (b) Mobile terminated due to congestion in mobile network, excessive delay in mobile network, premature release by calling network during call setup, called MS busy, called MS not switched on, called MS no answer, called MS deregistration, called to barred/unobtainable MS, and failed paging to MS.
  - ✘ Mean time of mobile network connection establishment for any service including mobile subscriber to MSC, mobile subscriber to gateway MSC, MSC to mobile subscriber, or gateway MSC to mobile subscriber.

  - ✘ Post-dialing delays for national calls for both same mobile network operator and more than one mobile network operator case and international calls per destination country and mobile network operators involved.
  - ✘ Mean time of service interruptions due to faults.
  - ✘ Mean time between failures causing interruptions.

- **Connection Quality Measurements.** To evaluate connection quality, the following measurements should be taken:
  - ✘ Probability of unintelligibility;
  - ✘ Duration of interruption to call;
  - ✘ Bit-error rate;
  - ✘ Throughput for nontransparent services;
  - ✘ Coverage for nontransparent services;
  - ✘ Coverage for transparent services; and
  - ✘ Error-free seconds.

- **Billing integrity measurements.**
  - ✘ Number of billing errors (percent of bills sent);
  - ✘ Percentile of justified complaints about billing; and
  - ✘ Percentile of calls with tariffing errors.

11.10 Summary

In this chapter, we discussed the management goals for PCS networks and provided the necessary requirements for PCS network management. We then discussed important features of the Telecommunications Management Network and outlined the five management functions: accounting management, security management, configuration management, fault management, and performance management. We also discussed the layered architecture for network management and presented the functional and physical architecture for TMN. We concluded the chapter by outlining the requirements in the five functional areas of management for a wireless network.

11.11 References

1. ANSI T1.210, "Operations, Administration, Maintenance and Provisioning (OAM&P)—Principles of Functions, Architecture and Protocols for Telecommunication Management Network (TMN) Interfaces."

2. ANSI T1.215, "Operations, Administration, Maintenance and Provisioning (OAM&P)—Fault Management Messages for Interfaces between Operations Systems and Network Elements."
3. ANSI T1.227, "Operations, Administration, Maintenance and Provisioning (OAM&P)—Extension to Generic Network Model for Interfaces between Operations Systems across Jurisdictional Boundaries to Support Fault Management–Trouble Administration."
4. "Draft ANSI T1.XXX—1996, American National Standard for Telecommunications—Operation, Administration, Maintenance, and Provisioning (OAM&P)—Performance Management Functional Area Services for Interfaces between Operations Systems and Network Elements."
5. "Draft ANSI T1.XXX—1996, American National Standard for Telecommunications—Operation, Administration, Maintenance, and Provisioning (OAM&P)—Technical Report on PCS Accounting Management Guidelines," T1M1.5/95-011R4.
6. ITU-T Rec. M.3010, "Principles for a Telecommunication Management Network (TMN)," Draft 950630, ITU—Telecommunications Standardization Sector, 1995.
7. CCITT Rec. M.3400, TMN Management Functions, ITU—Telecommunications Standardization Sector, 1992 (under revision, December 1996).
8. Draft CTIA, "Requirements for Wireless Network OAM&P Standards," CTIA OAM&P SG/95.11.28.
9. EIA/TIA IS-124, "Cellular Radio Telecommunications Intersystem Non-signaling Data Message Handler (DMH)."
10. Garg, V. K., and Wilkes, J. E., *Wireless and Personal Communications System,* Prentice Hall, Upper Saddle River, NJ, 1996.

C H A P T E R 12

Interconnection Between Systems

12.1 Introduction

CDMA systems do not exist in a vacuum. They must coexist with other wireless systems that share the same frequency spectrum, and they must be interconnected to other wireless and wireline voice and data networks. In this chapter, we will examine the interworking issues between different systems in sufficient detail to examine various dual-mode technologies being built and proposed. We will then discuss the dual-mode cellular/PCS CDMA and AMPS systems and phones that are being proposed by TIA committees TR-45 and TR-46. Although there is no active standards work in the area of dual-mode digital systems, we will identify the issues that need to be resolved to construct the systems and phones. The key to service interoperability is a seamless connection between the wireless and wireline networks. The work on intelligent networks for the wireline network has been extended to the wireless arena; therefore, we will conclude the chapter with a discussion of Wireless Intelligent Networking being planned by the international standards community.

12.2 Interworking Issues

The interworking between a wireless telephone and the worldwide wireline network is a topic that is being actively worked throughout the world. Early mobile telephone users often had to dial special access codes or use dialing sequences that were different from the wireline network. While never optimum, it was acceptable since mobile telephony was a

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premium service with a low number of users. As wireless telephony becomes ubiquitous, service providers and telephone stores can no longer afford the special training and support necessary to teach new users how to use the phone. Therefore, new wireless phones must operate identically to the wireline network.[1] In this section, we will discuss the issues related to obtaining seamless operation between the wireline and wireless networks. We will discuss only the issues here. A more complete treatment of issues and potential solutions can be found in the book *Wireless and Personal Communications* [3] and by actively following the activities of the standards bodies in the United States, Europe, and the world.

The wireline telephone network is primarily based on analog telephones and analog local loops using dual-tone multifrequency (DTMF) signaling between the subscriber and the central office and digital switching with digital signaling between central offices. Recently, business and residential subscribers are switching to Integrated Service Digital Network (ISDN) with digital telephones and digital signaling. Neither of these two methods currently uses wireless communications. Although most cellular telephones use analog frequency modulation (FM), the signaling is digital since DTMF tones do not have high reliability when sent over a fading radio channel. While there are clearly digital wireless phones, most do not support an ISDN channel with two 64-kbps bearer channels and a 16-kbps signaling channel. Thus, the voice and signaling protocols used in a wireless system must be converted into the protocols used on the wireline network. The base station or the mobile switching center will perform the conversion, called *interworking*.

Low-speed data are transmitted over the telephone network today using voice band modems that range from 1200 to 28,800 baud. Some of the uses for this data are

- Accessing electronic mail,
- Accessing remote computers,
- Transferring files,
- Using facsimile transmissions, and
- Performing transaction services (e.g., credit card validation).

1. It is unlikely that new wireless phones will generate a "dial tone" when off-hook and process dialed digits using the same procedures as the wireline network of today. However, new wireline digital services and the proliferation of advanced calling features may result in the two services moving to a common mode of operation.

Transmission at higher rates is typically used for video conferencing, mainframe-to-mainframe communications, and other uses that would not initially be carried over a wireless mobile link.

Except for those PCS supporting pulse code modulation or adaptive differential pulse code modulation (e.g., W-CDMA), the speech-coding systems for transmission of voice have been optimized for the voice transmission and have not been optimized for the transmission of non-voice signals such as voice band modems. Therefore, both the mobile station and the cellular/personal communications system must have interworking capabilities to provide the wide range of services currently using voice band modems. Over the air interface, the data must be transmitted digitally since that is the only option available. If the air interface supports a data rate higher than the basic data rate of the voice band modem, then interworking is possible. Interworking is not possible if the required data rate is higher than the needed data rate. For example, 28.8-kbps data service is not possible over a 16-kbps data link. Those systems supporting PCM or ADPCM can interwork with voice band modems without a data interworking platform. However, error performance may suffer because of the high error rate of the radio channel.

The various telephone administrations and telephone companies around the world have adopted a variety of national dialing plans for reaching telephones in their nations. They have assigned a (nationally) unique number to each telephone in their administration. In North America, there is a uniform 10-digit dialing plan. In other parts of the world, the numbering plan permits numbers of variable lengths from 6 to 15 digits. Even though the differences between numbering plans can easily be handled by the wireline telecommunications network, it creates complications for roaming mobile telephones. A telephone number that is unique in one nation may not be unique worldwide. Thus a mobile telephone registering in a wireless system in other than its home country may identify itself to the network with a number that duplicates a number from another country. The various standards bodies are solving this problem by migrating all mobile telephone numbers to a 15-digit International Mobile Station Identification (IMSI) number. The lack of a unique identity will exist until all current mobile telephones numbers are upgraded to the IMSI (See chapter 4).

In chapter 7, we studied the call flows for a variety of telephony functions. These call flows, while specific to CDMA, are part of a general set of call-processing functions based on the North American cellular/PCS network. They are part of a set of requirements for services [2], intersystem communications [3], base station-to-mobile switching center communications [4], and an air interface [5, 6]. The overall system is based on ISDN and SS7 signaling with overlays for mobile communications. The overlay is called a mobile application part (MAP). We have examined the IS-41-based MAP, but there is a second MAP used in Europe and migrating to North America that is based on the global system for mobile communication (GSM) system and called the GSM MAP. The overall call processing is done somewhat differently between the MAPs, and the security mechanisms are different. Significant work remains to provide interworking between these two MAPs.

A critical feature for wireless systems where the user roams to other systems is the ability to bill for calls placed or received on a visited system. A critical component to this is to support a common billing record system across all systems. In North America, the TIA IS-124 standard provides the common billing format. In other areas of the world, work is under progress to generate a common billing standard.

The final area where interworking is important is support for multiple air interfaces. The dominant interfaces in North America will be AMPS, CDMA, TDMA, and GSM. Methods must be developed for interworking between these and any other interfaces (e.g., PACS). The next two sections are devoted to these issues.

12.3 Dual-Mode Digital/AMPS Systems and Phones

In the United States, a decision by the Federal Communications Commission to require one nationwide standard, and thus support roaming between any systems in the country, resulted in the rapid deployment of cellular systems. In Europe, several nations built their own systems that were incompatible with systems in other European nations. Similarly, Japan built its own cellular system. Thus, throughout the world, there are at least five different, incompatible, first-generation cellular standards. Each of these standards depends on frequency modulation of analog signals for speech transmission, out-of-band signaling for call setup, and in-band signaling to send control information between a mobile station and the rest of the network during a call.

When the GSM system was designed for Europe, there was no push for a dual-mode (analog/digital) telephone since the analog phone would work only in one country. Thus, there was a push for a new Europe-wide digital system that was rapidly deployed. Unlike the incompatible analog systems, customers of the new system would gain roaming capabilities throughout Europe.

In North America, the EIA IS-553 cellular standard (now TIA IS-91) is fully deployed, and customers enjoy roaming throughout Mexico, the United States, and Canada. Any new digital system would not be fully deployed in the early years and, therefore, would not be competitively viable. The North American standards permit coexistence with the first-generation standard, the advanced mobile phone system (AMPS), and add a digital voice transmission capability for new digital equipment. Channels in a given geographic area may be assigned to either digital or analog transmission, whereas in another area the two signals may share channels. The North American TDMA Standard, IS-54/IS-136, enhances, rather than replaces, the analog cellular technology. Similarly for CDMA, a dual-mode design is possible; however, as we discussed in chapter 9, for the first CDMA channel, nine analog channels at each base station must be devoted to CDMA with an overall improvement in spectrum efficiency. Additional CDMA channels are obtained by removing six analog channels from service from each base station.

When a dual-mode phone and system is introduced, the designers must solve the following problems:

- Spectrum sharing between the analog and digital systems,
- Phone initialization,
- Handoffs from analog to digital and from digital to analog,
- Voice privacy, and
- Compatible call control between analog and digital operation.

Chapter 9 discusses the spectrum sharing between analog AMPS and CDMA. We describe the growth process from a purely analog system to a mixed analog/digital system to a pure CDMA digital system.

The IS-95 standard defines the operations of the mobile station during initialization. The MS must chose the frequency band to use, A or B band, and the modulation to use, analog or CDMA system. The frequency band options can be chosen independently of the modulation options. The preferences are stored in the MS when service is activated and often can be changed by the user of the phone. Some typical preferences (as defined by the standards) are

- System A only,
- System B only,
- System A preferred (if No Service can choose System B),
- System B preferred (if No Service can choose System A),

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- CDMA only,
- Analog only,
- CDMA preferred (if No Service can choose Analog operation), and
- Analog preferred (if No Service can choose CDMA operation).

Other preferences that are often designed into the MS are

- Home only,
- No service on a list of SIDs, and
- Chose alternate telephone number if roaming and alternate telephone number has same SID as the roaming system.

All these options are for a cellular dual-mode MS. When a dual-mode, dual-frequency cellular/PCS AMPS/CDMA phone is standardized, PCS frequencies will be added to the CDMA options.

When CDMA service is available, the dual-mode MS would normally start a call on a CDMA channel. The analog channels will be reserved for analog-only MSs. As the MS moves from one area of the system to another, it will be handed off between base stations. At some point, it may move outside the coverage area of the CDMA base stations. When that event happens, the system will process a hard handoff to an analog cellular channel. The handoff to the analog channel has several consequences. The call will continue as an analog call until it is ended. The standards do not support the handoff back to a CDMA channel. The message to inform the MS of a new CDMA channel, using the analog blank-and-burst data transmission, is excessive and would result in a long handoff outage. Also, unless other means are provided, the user has lost voice privacy since analog FM signals are easily intercepted on older police/fire scanners or older TV sets that cover channels 70–83.

The IS-95 standard supports analog and CDMA operation as separate functions once the MS is initialized. A handoff to the analog operation is allowed to change the phone from one mode to the other. Since the two modes are independent, it would be possible for the call-processing functions to be different in the two modes. However, cellular and personal communications systems designed for use in North America use a common intersystem communications functionality defined in the IS-41 protocol. Thus, the call-processing functions will be the same for either digital or analog operation.

Other dual-mode analog/digital systems are either defined or being defined by the standards bodies. An AMPS/TDMA MS is defined in IS-54 and IS-136 and provides similar functionality to the AMPS/CDMA MS.

The European GSM system has been modified for use in North America and will see deployment in many cities. Therefore, a dual-mode AMPS/GSM MS will also be defined. Unfortunately, the operation of the MS may not be as seamless as a dual-mode AMPS/CDMA MS since the GSM MAP is different from the IS-41 MAP and telephony services may operate differently in different systems (AMPS and GSM).

12.4 Dual-Mode Digital Systems

In this section, we will examine the operation of a dual-mode digital system. No dual-mode systems currently exist, so we will limit our discussion to the problems that must be solved to build them. We will first examine the issues of the mobile station and then examine the system issues.

Two key problems need to be solved for a dual- (or multiple) mode phone: initial system selection and handoffs. When a phone is first turned on, it goes through a sequence of events that initializes the phone and registers it with a system. Although we can force a set of preferences similar to those described in the previous section, a multimode phone needs to have a set of procedures that will enable it to try frequencies and protocols for two or more air interfaces. Dual digital mode MSs may require multiple receivers and transmitters and extensive software to control them. The start-up sequence may be lengthy because the phone does not know what system it is in when first powered up. The user of the phone may help the phone by giving information about location but, in general, the phone should do its own analysis. Users of a dual mode system roaming into new areas may see start-up delays of several tens of seconds or more before service is available. After the phone has registered with a system, calls can be placed or received.

After a call is established, it may be necessary to handoff the call to another radio system. All air interfaces support handoff; and the issues of handoff between radio systems (using the same air interface) connected to the same or a different PCS switch are well understood and have been discussed in chapter 7.

Handoffs between systems essentially require transmission and signaling facilities between the switches of each system. Other than the extra facilities, the handoff between two radio systems on two different systems (using the same air interface) does not differ from the handoff between radio systems on the same PCS or cellular system.

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However, when the air interfaces differ between the two radio systems, then the problem is more complicated. All the digital systems assume that the data receiver can quickly regain synchronization when a call is handed off to another base station. In particular, when CDMA is used, the receiver in the MS receives both radio systems simultaneously and uses signals from both to produce a composite signal. When the handoff occurs between two different air interfaces, new handoff messages will be necessary so that the MS can be informed about the characteristics of the new systems. A second receiver may also be necessary so that the MS can be listening to the new radio system before the handoff occurs. Otherwise, the handoff delay may be excessive and calls may be dropped.

If the handoff algorithm differs (MS initiated, anchor switch initiated, or anchor radio system initiated), then new handoff procedures and messages need to be developed and tested.

Clearly extensive modifications to the MS and the network will be needed to support handoffs between different air interfaces in different systems.

For good performance on MS initialization, each system in an area may need to broadcast information about all systems in the area. Thus, when an MS enters a new area, if it can receive any transmissions from any system, it can find its preferred system. System information messages that inform MSs of other systems do not currently exist except for limited case of public/private systems of the same modulation type. All air interface protocols need upgraded capabilities on the system information channel to support multimode digital terminals.

The alternative would require that the MS try multiple sets of frequencies and protocols until it starts decoding information, which will result in a lengthy start-up process and time.

For dual-mode CDMA/GSM MSs, an additional problem results and is similar to the dual-mode AMPS/GSM problem. CDMA supports the IS-41 MAP which is different from the GSM MAP. Thus, a dual-mode CDMA/GSM MS will operate differently depending on which system it is being used on. This may make the user unwilling to place calls in areas where CDMA service is not available and defeats the purpose of the dual-mode phone.

12.5 Wireless Intelligent Networks

For the wireline network, BellCore has defined a set of protocols called intelligent network (IN) [1] that enable a rich set of new telephony capabilities to be generated without additional software development in the central switch. As IN has grown in popularity for the wireline network, the wireless network is also embracing the concept. IN improves the ability to locate and efficiently direct calls to roaming MSs and provides other advanced features. In this section, we will examine the work of the international standards bodies and TR-45/TR-46 in the United States to add IN to the wireless systems.

The International Telecommunications Union (ITU) has adapted IN for use in wireless networks. Question 8 of Study Group 11 has defined a communications control plane (fig. 12.1) and a radio resource control plane (fig. 12.2) using IN concepts.

On the network side of the communications control plane, the following functional elements are defined:

- The **Bearer Control Function (BCF)** provides those bearer functions needed to process handoffs. A conference bridge to support soft handoffs is a common example.
- The **Bearer Control Function for the Radio Bearer (BCFr)** provides the functions necessary to select bearer functions and radio resources. It also detects and responds to pages from the network and performs handoff processing. Some example bearer func-

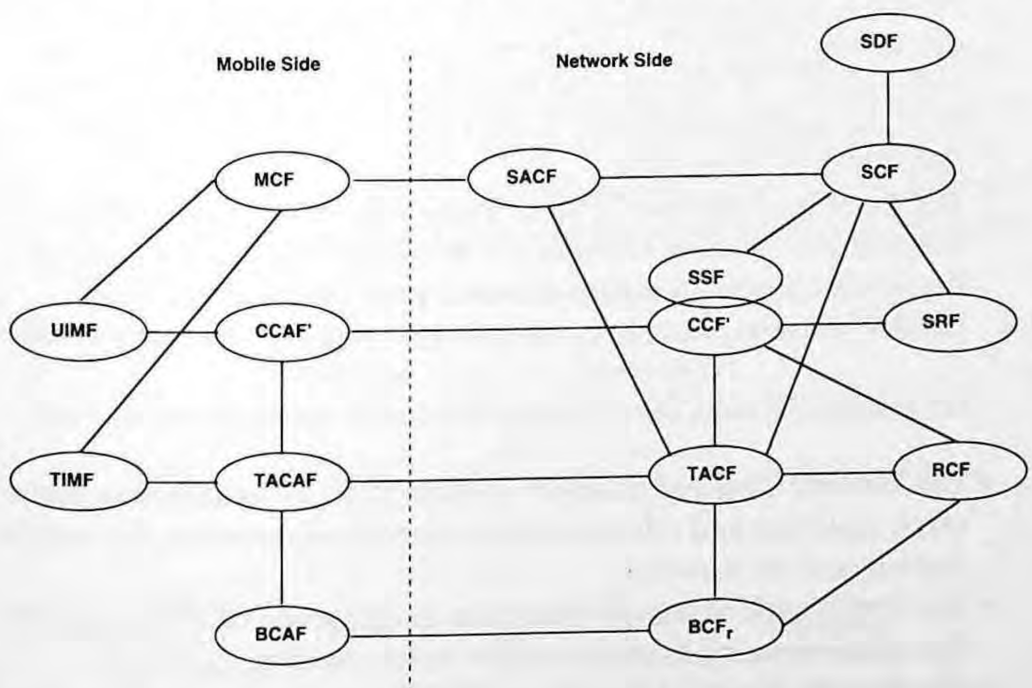

Figure 12.1 Communications control plane.

tions are PCM voice, ADPCM voice, packet data, and circuit-switched data.

- The **Call Control Function-Enhanced (CCF')** provides the call and connection control in the network. Some examples include establishing, maintaining, and releasing call instances requested by the CCAF', providing IN triggers to the service switching function (SSF), and controling bearer connection elements in the network.
- The **Service Access Control Function (SACF)** provides the network side of mobility management functions. Some examples are registration and location of the mobile station.
- The **Service Control Function (SCF)** contains the service and mobility control logic and call processing to support the functions of a mobile terminal.
- The **Service Data Function (SDF)** provides data storage and data access in support of mobility management and security data for the network.
- The **Specialized Resource Function (SRF)** provides the specialized functions needed to support execution of IN services. Some examples are dialed-digit receivers, conference bridges, and announcement generators.
- The **Service Switching Function (SSF)** provides the functions required for interaction between the CCF' and the SCF. It supports extensions of the CCF' logic to recognize IN triggers and interact with the SCF. It manages the signaling between the CCF' and the SCF and modifies functions in the CCF' to process IN services under control of the SCF.
- The **Terminal Access Control Function (TACF)** provides control of the connection between the mobile station and the network. It provides paging of mobile stations, page response handling, handoff decision, completion, and trigger access to IN functionality.

On the mobile side, the following functional elements are defined:

- The **Bearer Control Agent Function (BACF)** establishes, maintains, modifies, and releases bearer connections between the mobile station and the network.
- The **Call Control Agent Function-Enhanced (CCAF')** supports the call-processing functions of the mobile station.
- The **Mobile Control Function (MCF)** supports the mobility management functions of the mobile station.

- The **Terminal Access Control Agent Function (TACAF)** provides the functions necessary to select bearer functions and radio resources. It also detects and responds to pages from the network and performs handoff processing.
- The **Terminal Identification Management Function (TIMF)** stores the terminal-related security information. It provides terminal identification to other functional elements and provides the terminal authentication and cryptographic calculations.
- The **User Identification Management Function (UIMF)** provides user-related security information similar to the TIMF.

Both the TIMF and the UIMF can be stored in either the mobile station or a separate security module often implemented in a smart card.

The radio resource control plane (fig. 12.2) is in charge of assigning and supervising radio resources. Four function entities (two on the mobile side and two on the network side) perform the functions of the radio access subsystem:

- The **Radio Resource Control (RRC)** provides functionality in the network to select radio resources (e.g., channels and spreading codes), make handoff decisions, control the RF power of the mobile station, and provide system information broadcasting.
- The **Radio Frequency Transmission and Reception (RFTR)** provides the network side of the radio channel. It provides the radio channel encryption and decryption (if used), channel quality estimation (data-error rates for digital channels), sets the RF power of the mobile station, and detects accesses of the system by the mobile station.

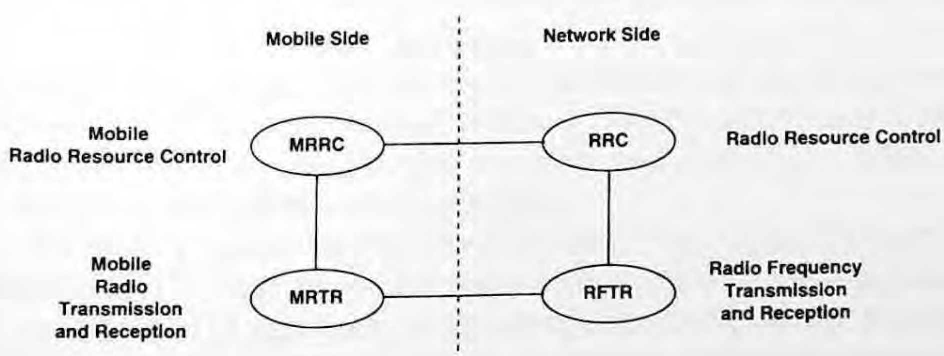

Figure 12.2 Radio resource control plane.

- The **Mobile Radio Resource Control (MRRC)** processes the mobile side of the radio resource selection. It provides base station selection during start-up, mobile-assisted handoff control, and system access control.
- The **Mobile Radio Transmission and Reception (MRTR)** provides the mobile side of the radio channel and performs functions similar to the RFTR.

The two control planes interact to provide services to the mobile station and the network.

The TR-46 PCS network reference model working group has generated a simplified version of the ITU model and has explicitly shown the operations functions on the model. The IN function reference model for PCS (fig. 12.3) has many of the functional elements with the same element name as the ITU model. The differences follow:

- The **Radio Terminal Function (RTF)** contains all functionality of the mobile side of the reference model.
- The **Radio Access Control Function (RACF)** is similar to the SACF and provides mobility management functions.

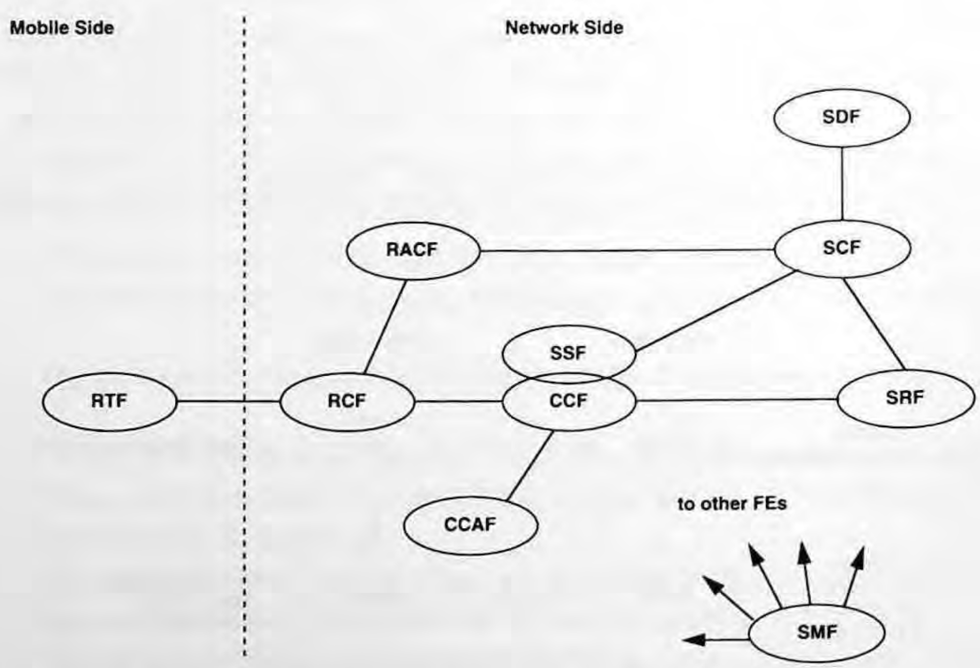

Figure 12.3 IN functional reference model for PCS.

- The **Radio Control Functions (RCF)** provide the capabilities of the TACF, the BCFr, and the BCF in the ITU model. It provides the radio ports and the radio port controller capabilities in the PCS network.
- The **Call Control Agent Function (CCAF)** provides access to the wireless network by wireline users.
- The **Service Management Function (SMF)** provides the network management functions for each functional element.

With both the ITU and the TR-46 reference models, the functionality to support all the features and capabilities for a wireless network can be partitioned into different functional elements and still meet the variety of national and worldwide standards. Therefore, no exact partitioning of the functions of the network reference models described in chapter 4 can be made. We encourage you to examine the standards and implementations of the various manufacturers for example partitioning.

12.6 Summary

In this chapter, we discussed the interconnection of a wireless network to the worldwide wireline telecommunications network. We showed that, for both voice and data, the wireless network must perform conversions to enable calls to proceed onto the wireline network. We demonstrated that wireless telephones roaming to different areas of the world present issues of numbering and billing that do not occur on a wireline network. The use of a standard 15-digit International Mobile Station Identification will solve the numbering problem. Only the support of a standard billing format that can be quickly sent between different areas of the world will solve the billing problem.

We then described the operation of the dual-mode AMPS/CDMA MS as it provides service to users on either the analog or digital system. Since there are no dual-mode digital systems, we examined the problems that must be solved by the design of the systems and phones but await the standards bodies to offer the solutions. Clearly custom solutions will be provided before the standards are written and adopted, but these custom solutions may not work in all systems.

Finally we examined the migration of the intelligent network concepts from the wireline network to the wireless network. As with many of the issues raised in this book, we encourage you to follow the worldwide standards activities to obtain the latest plans to solve the issues discussed in this chapter.

12.7 References

1. BellCore, "Advanced Intelligent Network (AIN) Release 1 Switching System Generic Requirements," TA-NWT-001123, Issue 1, May 1991.
2. ITU Study Group 11, "Version 1.1.0 of Draft New Recommendation Q.FNA, Network Functional Model for FPLMTS," Document Q8/TYO-50, September 15, 1995.
3. Garg, V. K., and Wilkes, J. E., *Wireless and Personal Communications Systems*, Prentice Hall, Upper Saddle River, NJ, 1996.
4. TIA Interim Standard, IS-41C, "Cellular Radio Telecommunications Intersystem Operations."
5. TIA Interim Standard, IS-54B, "Cellular System Dual-Mode Mobile Station–Base Station Compatibility Standard."
6. TIA Interim Standard, IS-91, "Cellular System Mobile Station–Base Station Compatibility Standard."
7. TIA Interim Standard, IS-95, "Mobile Station–Base Station Compatibility Standard for Dual-Mode Wideband Spread Spectrum Cellular System."
8. TIA Interim Standard, IS-136, "Cellular System Dual-Mode Mobile Station–Base Station Compatibility Standard."
9. TR-46, "Personal Communications Services Network Reference Model for 1800 MHz," PN-3436 Baseline Text Version 1, June 26, 1995.
10. EIA/TIA IS-124, "Cellular Radio Telecommunications Intersystem Nonsignaling Data Communications (DMH)."

CHAPTER 13

Evolution of CDMA Technology for Wireless Communications

13.1 Introduction

In previous chapters, we discussed current standards for CDMA technology and their application to wireless communications. Presently, these standards are being implemented by manufacturers and being deployed by service providers. The wireless industry is investigating methods to improve and to reduce costs so that the mobile subscriber will experience better service at a reduced cost.

This chapter examines several areas that address these goals. First, service providers are seeking ways to reduce administrative costs by streamlining the activation procedure for new mobile subscribers. This chapter discusses over-the-air service provisioning which supports this capability. Second, advances in digital technology will make it possible to improve the quality of speech coders at a given digital rate. This chapter discusses the enhanced variable rate codec, which provides better performance than the current standardized speech processor. Third, the wireless industry is studying improvements to transmission schemes over the air interface. The resulting improvements will increase channel capacity. To mobile subscribers, this translates to a better service at a lower cost. This chapter discusses interference cancellation, multiple beam adaptive antenna arrays, and improvements of the handoff algorithm, which are three separate approaches for achieving the capacity improvement objective.

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13.2 Over-the-Air Service Provisioning

Most mobile subscribers currently must obtain and initialize service through an authorized dealer rather than directly from the service provider.[1] From the point of view of the service provider, this procedure has several disadvantages:

- The service provider pays a commission to the authorized dealer. This is an expense to the service provider.
- The service provider does not have total control during this process; consequently, there is opportunity for fraud to occur.
- The mobile subscriber is inconvenienced in that the subscriber must visit the authorized dealer. Several hours may be required to complete this operation, and, furthermore, service may not be immediately activated.

Thus, the wireless industry is investigating ways to simplify the activation procedure and to reduce associated expenses. Standards body TIA TR-45 is currently developing a feature called over-the-air service provisioning (OTASP) to address this need. OTASP allows the service provider to configure a mobile station directly and to provision the mobile during a call initiated by the mobile subscriber. To initiate service, the mobile subscriber makes a mobile-originated call by dialing an activation code. The call is established between the mobile subscriber and the Customer Service Center (CSC), which is operated by the service provider. OTASP assumes that a conversation between the mobile subscriber and a CSC attendant is established so that information (e.g., credit card information and service options) can be exchanged. However, the exact procedure is determined by the service provider and may vary among service providers. As an example, the service provider may wish to reauthenticate the mobile before credit card information is exchanged. This allows the credit card information to be encypted using signaling message encryption.

OTASP defines procedures for the following operations:

1. Enable uploading the mobile's indicator data parameter block from the mobile to the CSC.

1. Some service providers have their own stores, but the cost of operating them is high.

2. Download the A-key to the mobile.[2]
3. Program the mobile's indicator data parameter block.
4. Reauthenticate the mobile.

The mobile's indicator data parameter block contains information such as the station class mark (SCM), SID/NID list (as discussed in chapter 3), and access overload class (ACCOLC). The indicator data parameter block consists of a subset of the mobile's number assignment module (NAM). Also, OTASP allows the service provider to download the mobile's A-key. Finally, the mobile station may be reauthenticated so that either voice privacy or signaling message encryption or both can be activated during the remainder of the call to complete the transaction with a high degree of security. The A-key, which is a 64-bit quantity, is needed to support authentication, voice privacy, and signaling message encryption.

Without OTASP, either the mobile subscriber or the authorized dealer must enter the A-key into the mobile station or the mobile station must be preprogrammed with the A-key. None of these choices is desirable. This operation is very important since manually entering the A-key has a significant probability of error and fraud. However, the A-key cannot feasibly be sent to the mobile over the air interface without a sufficient degree of encryption. It is imperative that the A-key be known at only the authentication center (AC) and the mobile station. At intermediate points between the AC and the mobile station, the A-key must be securely encrypted. The wireless industry has selected the Diffie-Hellman key exchange procedure (DHKEP) to address this concern. The DHKEP is summarized by the following steps:

1. The AC chooses a 160-bit random number a, such that $4 \le a \le 2^{160} - 1$.
2. The AC chooses a 160-bit random number g, such that $1 < g \le 2^{160} - 1$.
3. The AC chooses a large prime number p having a 512-bit representation, such that $2^{511} < p \le 2^{512} - 1$.
4. The AC sends g and p to the mobile station. Of course, this information traverses part of the network and is transmitted over the air interface. However, there is no attempt to encrypt g and p.

2. The A-key is a 64-bit pattern stored in the mobile station that is used to generate/update the mobile station's Shared Secret Data. Refer to section 2.3.12 (Authentication, Encryption of Signaling Information/User Data) of TIA IS-95A for further information.

5. The mobile station chooses a 160-bit random number b, such that $4 \le b \le 2^{160} - 1$.
6. The mobile station calculates $M = g^b \bmod p$.
7. The mobile station sends M to the AC over the air interface.
8. The AC calculates $B = g^a \bmod p$.
9. The AC sends B to the mobile station over the air interface.
10. Both the mobile station and the AC calculate $X = g^{ab} \bmod p$, where $X = M^a \bmod p$ (as calculated by the AC) and $X = B^b \bmod p$ (as calculated by the mobile station). Both the mobile station and the AC can derive the temporary A-key by truncating X to the 64 least significant bits.
11. Both the mobile station and the AC store the temporary A-key. The temporary A-key is considered the A-key only after the CSC instructs the mobile station to commit its temporary memory to semipermanent memory.

Once the temporary A-key is created at the mobile station and at the AC, the temporary shared secret data (SSD)[3] can be created. OTASP has defined a procedure so that the reauthentication can be invoked during the call. Without this new procedure, authentication requires the initiation of a new call, which means that the original call would be ended. This, of course, would present a serious complexity to the transaction. Once reauthentication is completed, voice privacy and/or signaling message encryption can be invoked. At this time, user-sensitive information can be transmitted over the air interface with a high degree of security. Once the transaction has been completed, the CSC initiates a *data commit order,* which transfers any downloaded data, the temporary A-key, and temporary SSD to semipermanent memory. At that time, the mobile station and the AC consider the programmed data as permanent.

13.3 Improvement of Speech Coders

In order to quantize the quality of service, the wireless industry measures the speech coder's performance by the mean opinion score (MOS). The MOS is determined by statistically combining the opinions of a group of human listeners. An MOS has a range from 1 to 5, where 5 means excellent voice quality, and 1 means unsatisfactory voice quality.

3. The shared secret data is a 128-bit pattern that is needed for the authentication and encryption process. Refer to section 2.3.12 of TIA IS-95A and TIA PN-3569 for further information.

Toll quality has an MOS score of 4, signifying good voice quality. With analog operation, a service provider typically wishes to provide service corresponding to an MOS score of 4. The existing mobile subscriber will expect this grade of service. Thus, the grade of service provided by CDMA systems will be gauged by this quality of service.

It is anticipated that by mid 1996, standards body TIA TR-45.5 will complete the specifications for an improved variable rate speech coding algorithm for the 9.6-kbps physical layer (i.e., rate set 1) (refer to chapter 5). This is the same physical layer that the TIA IS-96A variable rate speech coder uses. Since the new speech coder is an improvement with respect to the existing 8-kbps speech coder,[4] the Telecommunications Industry Association calls the new algorithm the enhanced variable rate codec (EVRC) algorithm. This algorithm is applicable both to IS-95A mobiles (800 MHz) and to ANSI J-STD-008 mobiles (1.8–2.0 GHz). Like the TIA IS-96A algorithm, the EVRC algorithm is based upon the code-excited linear prediction (CELP) algorithm (see fig. 13.1).

The EVRC speech coder uses the relaxed code-excited linear prediction (RCELP) algorithm, and thus it does not match the original residual signal but rather a time-warped version of the original residual that conforms to a simplified pitch contour. This approach reduces the number of bits per frame that are dedicated to pitch representation. This allows additional bits to be dedicated to stochastic excitation and to channel impairment protection. The EVRC algorithm categorizes speech into full rate (8.55 kbps), half rate (4 kbps), and eighth rate (0.8 kbps) frames that

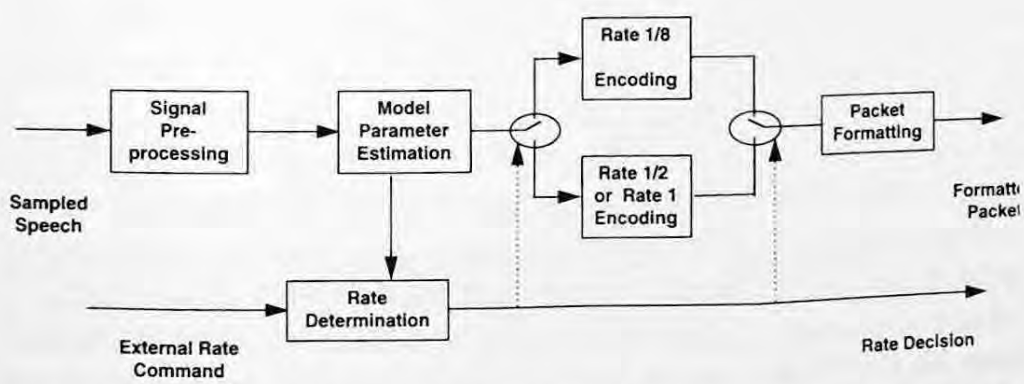

Figure 13.1 Functional diagram of the EVRC algorithm. (Reproduced under written permission of the copyright holder [TIA].)

4. Actually, the maximum data rate is 8.55 kbps, but the wireless industry references the rate as being equal to 8 kbps.

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are formed every 20 ms. The TIA IS-96A algorithm categorizes speech into these rates as well as quarter rate frames.

The EVRC algorithm offers a significant performance improvement over the IS-96A speech coder (see table 13.1). These measurements do not include the effects of acoustical noise (e.g., car noise). The improvement ranges from an MOS increase of approximately 0.66 for an error-free CDMA channel to 0.95 for a channel having a frame error rate[5] of 3 percent. At a frame error rate of 1 percent, the speech coder performance of the TIA IS-96A coder is 3.17 (fair), while that of the EVRC coder is 3.83 (near toll quality). The performance at a frame error rate of 1 percent is important since channel capacity of a CDMA system is often calculated with the assumption that power control maintains the frame error rate within 1 percent. Table 13.1 also shows the performance of the CDMA Development Group's[6] 13-kbps (CDG-13 kbps) speech coder, which offers high voice quality but results in a decrease in channel capacity of approximately 40 percent. The MOS improvement of the CDG-13 kbps speech coder over the EVRC speech coder is approximately 0.12. The improvement at the frame error rate of 1 percent is not nearly as great as the improvement when transitioning from the TIA IS-96A speech coder to the EVRC speech coder. Given these observations, it is expected that the EVRC algorithm will be the basis of the speech coders in the future.

Table 13.1 Speech coder performance (MOS) as function of frame error rate

Frame Error Rate, %	CDG-13 kbps	EVRC	IS-96A
0	4.00	3.95	3.29
1	3.95	3.83	3.17
2	3.88	3.66	2.77
3	3.67	3.50	2.55

A CDMA system will need to contend with different type of speech coders (i.e., TIA IS-96A, CDG-13 kbps, and EVRC speech coders). A CDMA system may or may not support all types of these speech coders; however, the CDMA system must be able to negotiate the type of speech coder with the mobile station. As mentioned in chapter 3, this is done

5. The frame error rate (FER) is defined as the number of bad frames per unit of time.

6. The CDMA Development Group is a consortium of companies that developed this 13-kbps speech coding algorithm as a proprietary service option.

though service negotiation. The TIA IS-96A, CDG-13 kbps, and EVRC speech coders are assigned service option values of 1; 32,768; and 3, respectively (refer to table 3.3).

With greater penetration of personal communication services, there will be a greater importance of mobile-to-mobile calls. If both mobile stations use the same speech coder, then it is possible to send the digital output from one mobile station to the other. If this is not done, then degradation due to tandem operations of the coders occurs since two additional conversions are necessary (see fig. 13.2).

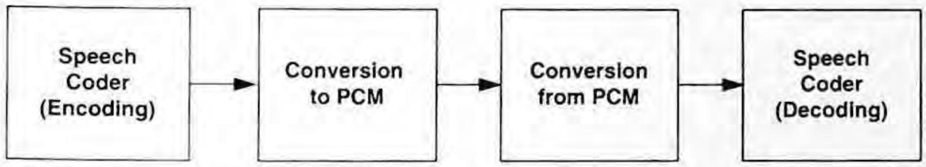

Figure 13.2 Tandem operation of speech coders.

The wireless industry is discussing this topic; however, at this time there are no definite plans to standardize a solution. At first glance, this issue seems to be a straightforward problem; however, it is not. First, there is no assurance that the speech coders at the originating and the terminating ends are the same. If they are not, then the speech coders cannot talk directly with each other. Second, a dual-mode mobile may change call modes from CDMA to analog during a call. Third, call supplementary services such as three-way calling may conference a non-CDMA party to the call. This possibility vastly increases the complexity of the conference circuit.

13.4 Interference Cancellation

CDMA technology provides a significant gain in call capacity with respect to analog technology. So one might ask, "Why should the wireless industry be concerned with capacity improvements?" There are at least three reasons that motivate the wireless industry:

- If the number of mobile subscribers continue to grow at even a portion of the current rate, the assigned radio spectrum will be depleted in the future.
- New services, such as wideband data and video, are much more spectrum-intensive than the voice service.

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- There is the economy of numbers. As the number of mobile subscribers being served by a wireless system increases, the cost per subscriber decreases.

One obvious way of increasing the capacity is to develop a reduced-rate speech coder. As discussed in the previous section, the EVRC speech coder will be introduced in the near future (probably within several years). This speech coder has a maximum data rate of 8.55 kbps, while the average data rate is approximately 4.4 kbps. In the near future, it is doubtful that variable rate speech coding technology will make a major breakthrough to reduce the average data rate appreciably, even though the voice quality will be improved at the given data rate.

Current research is investigating the practicality of canceling the interference attributed to other mobile subscribers being served by a given base station [1]. This process is a form of multiuser detection. Without such a process, interference from other mobile users translates into an increased level of noise associated with the user's CDMA channel. The base station simultaneously processes all users that are being served by that base station. Thus, this procedure has one input and multiple outputs, each output corresponding to one user. (See fig. 13.3.) The matched filter bank is implemented by multiplying the received signal by the user's spreading function $s_i(t)$. The output of the matched filter bank can be expressed in matrix notation as

$$[y] = [R] \bullet [W] \bullet [x] + [z] \tag{13.1}$$

where $[R]$ and $[W]$ are $K \times K$ matrices,
$[y]$ is the $K \times 1$ vector representing the processed output signal,
$[x]$ is the $K \times 1$ vector representing the transmitted signal, and
$[z]$ is the $K \times 1$ vector representing the received noise.

The matrix R is given by:

$$R_{ij} = \int_0^T s_i(t) \times s_j(t)dt \tag{13.2}$$

The matrix $[W]$ is a diagonal matrix with each diagonal element representing the channel attenuation of the corresponding mobile's signal path. The output of the matched filter bank is processed through another matrix filter that represents the inverse of $[R]$, (i.e., $[R]^{-1}$). The decision for the i^{th} mobile's signal is calculated by the step function:

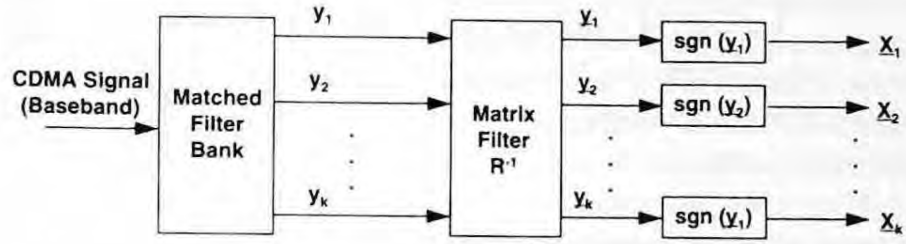

Figure 13.3 Multiuser detection.

$$\underline{x}_i = \text{sgn}(\underline{y}_i) \tag{13.3}$$

This approach eliminates multiple-access interference, but it is affected by the channel noise. With the approach in figure 13.3, it is assumed that the receiver uses synchronous detection and that each mobile's signal travels along a single path. The effects of the second restriction can be reduced by incorporating a RAKE receiver as is discussed in chapter 2. The first restriction exists with a CDMA system (TIA IS-95A), since the pilot signal is not included in the reverse traffic channel. However, with W-CDMA, the pilot is transmitted on the reverse path, and thus synchronous detection is possible. Without synchronous detection, we must use noncoherent multiuser detection. More details are presented in [1].

Processing is limited to canceling interference that is associated with that base station. Otherwise, each base station would be required to send an inordinate amount of data about its associated mobile users to the other base stations. Multiuser detection is applicable only for increasing the capacity on the reverse radio link and not on the forward radio link. As discussed in chapter 9, the capacity on the reverse radio link is usually less than on the forward radio link. However, for bilateral services, any improvement of the capacity on the reverse radio channel exceeding that on the forward radio channel is not justifiable. The upper bound for improving the capability on the reverse radio channel with multiuser detection is $(1 + N_I)/N_I$, where N_I is the mean interference from neighboring cells as discussed in chapter 9. For $N_I = 0.57$, which corresponds to the support of three-way soft handoffs, the upper bound equals approximately 2.75. This is the maximum gain that can be achieved; however, in practice the gain is less. An optimal multiuser detector is too complex; rather, we must accept the reduced performance of a suboptimal realization. Also, the call configuration of a base station is dynamic (i.e., new calls are set up by the base station and existing calls are handed into and out of the cell). Thus, the detector must dynamically adjust its coefficients. The wireless industry is currently exploring the utility of this approach to determine if moderate benefits can be achieved within practical bounds.

13.5 Multiple Beam Adaptive Antenna Array

An *adaptive antenna array* is defined as one that modifies its radiation pattern, frequency response, or other parameters, by means of internal feedback control while the antenna system is operating. The basic operation is described in terms of a reduction in sensitivity in certain angular position, toward a source of interference. Adaptive antennas adjust their directional beam patterns to maximize the signal-to-interference ratio at the output of the receiver. The adaptive array contains a number of antenna elements, not necessarily identical, coupled together via some form of amplitude control and phase shifting mechanism to form a single output (refer to fig. 13.4). The amplitude and phase control involves a set of complex weights. By suitable choice of weights, the array will accept a wanted signal from direction θ_1 and steer null toward interface source located at θ_k, for $k \neq 1$. The weighting mechanism can be optimized to steer

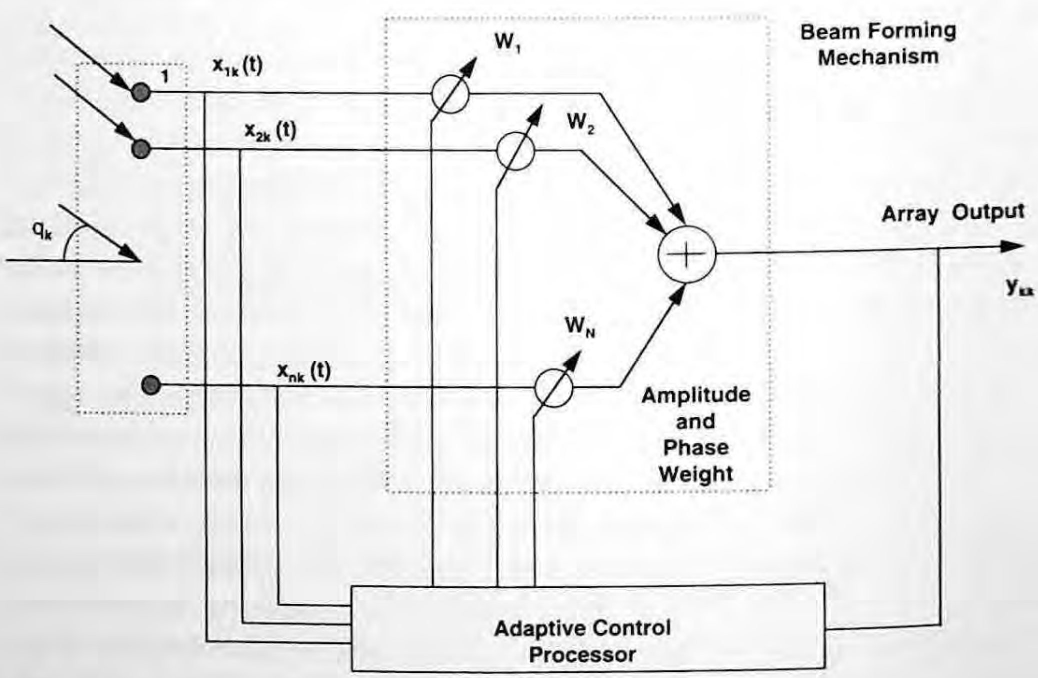

Figure 13.4 An adaptive antenna array.

beams (a radiation pattern maxima of finite width) in a specific direction, or directions. An N element array has $N - 1$ degrees-of-freedom yielding up to $N - 1$ independent pattern nulls. By controlling the weights via a feedback loop designed to maximize the signal-to-interference ratio at the array output, the system can be treated as an *adaptive spatial filter.*

The antenna elements can be arranged in various geometries, with uniform line, circular, and planar arrays being very common. The circular array geometry provides complete coverage from a central base station as beams can be steered through 360 degrees. The antenna elements are typically sited $\lambda/2$ apart, where λ represents the wavelength of the received signal. Spacing greater than $\lambda/2$ improves the spatial resolution of the array, but the formation of grating lobes (secondary maxima) can also result. These are regarded as undesirable.

Each mobile is tracked in azimuth by a narrow beam for both mobile-to-base station and base station-to-mobile transmission. The directive nature of the beams ensures that in a given system the mean interference power experienced by any one user, due to other active mobiles, would be much less than that experienced using conventional wide coverage base station antennas. Since CDMA-based cellular/PCS networks are designed to be interference limited, the adaptive antenna would considerably increase the potential user capacity.

Since the base station antenna could track any mobile or group of mobiles within its coverage area, the receiving array is capable of resolving the angular distribution of the users as they appear at the base station. The base station is then in position to form an optimal set of beams, confining the energy directed at a given mobile within a finite volume. The antenna system dynamically assigns a single narrow beam to illuminate a single mobile and broad beams to the numerous groupings of mobiles (fig. 13.5). By constraining the energy transmitted toward the mobiles, there are directions in which little or no signal is radiated. This phenomenon gives rise to the reduction in the probability of interference occurring in neighboring cells, and thereby increases the capacity of the network. A comparison made between the conventional and adaptive array antenna schemes has shown that a marked improvement in capacity of the network can be achieved. An idealized eight-beam antenna could provide a threefold increase in network capacity when compared with existing schemes such as cell splitting. The overall cost of the system is less, since fewer base stations are required for an equivalent user capacity. Also, unlike the technique of cell splitting and cell sectorization,

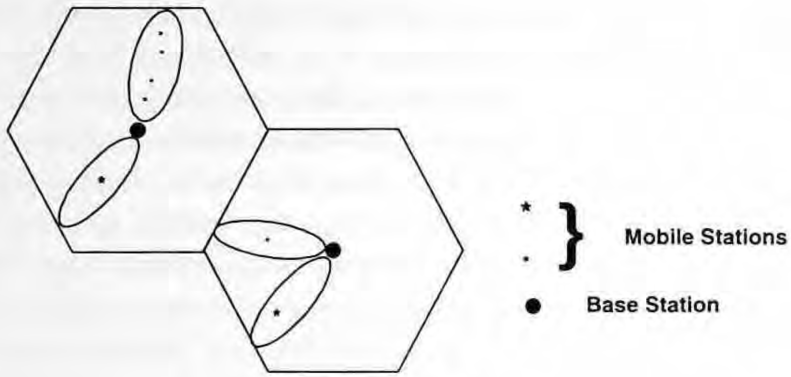

Figure 13.5 Optimal beam forming.

the multiple-beam adaptive array antenna would not impair the trunking efficiency of the network.

It should be noted that this approach can be applied to improving the channel capacity on both the forward radio link and the reverse radio link, while interference cancellation, as discussed in the previous section, is limited to improving the channel capacity only on the reverse radio link.

13.6 Improvements of Handoff Algorithms

Handoffs may be initiated either by the mobile station or by the base station.[7] In the first case, the mobile station sends a *pilot strength measurement* message to the serving base station, indicating that the received signal strength of the pilot in the active set is below T_DROP or a pilot in the candidate set is above T_ADD. The base station may initiate a handoff, but it is not obligated to do so. This type of handoff is classified as a *mobile-assisted handoff*. Alternatively, measurements at the base station and at neighboring base stations, at the request of the serving base station, can initiate a handoff. This type of handoff is categorized as a *base-station-directed handoff*. Both types of handoffs are supported by current CDMA standards. A mobile-assisted handoff can be extended to a *mobile-directed handoff* in which the mobile station determines that a handoff to a selected base station will occur. The mobile-directed handoff has the potential benefit of reducing the handoff interval, but this approach has

7. In this discussion, the term *base station* is really the Base Transceiver System (BTS) as discussed in chapter 4.

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several associated issues. First, the mobile station is determining its decision only on the forward radio link. The performance on the reverse radio link may not match that of the forward radio link at a given time. Also, the wireless infrastructure transfers handoff control to the mobile stations. This approach is vulnerable to malfunctioning mobile stations. Currently, mobile-directed handoffs are not supported by CDMA standards. A CDMA system will often combine information provided by the mobile station with information measured by the base station in determining handoff actions.

The handoff performance can often be improved by partitioning base stations into microcells and macrocells [3]. A macrocell has a large coverage area when compared with that of a microcell. In general, macrocells can overlay microcells or can abut microcells. A mobile station can be assigned to a microcell rather than a macrocell if the mobile station is traversing the cell at a slow enough velocity. If microcells and macrocells share the same frequency spectrum, then a higher transmitted power level at the microcells is necessary to compensate for interference from the macrocells. This induces a coverage hole in the macrocell in which the microcell is situated. Thus, overlaying microcells on macrocells supporting the same frequency spectrum has serious practical implications.

If the overlaid microcells use a different frequency spectrum than the macrocells, this is not an issue. This approach, however, introduces another problem. With current CDMA standards, the mobile station can monitor signals on the serving frequency. If a handoff from a microcell to a macrocell or from a macrocell to a microcell is necessary, the mobile station is not able to "assist" since it cannot measure the pilot's strength of the target base station. In this situation, the base station can either instruct neighboring base stations to measure and to report the signal strength of the mobile station, or it can use a priori information about the system configuration to determine the target base station. The first approach requires that neighboring base stations be equipped with CDMA location radios. The second approach assumes that the probability of choosing the correct target base station is high. Thus, the second approach restricts the number of system configurations that are applicable.

These issues are moot if the mobile station can measure the strength of pilots on a frequency that is different from that of the serving frequency. This capability is being discussed in CDMA standards bodies.

13.7 Summary

This chapter discussed several areas that are being investigated by the industry in order to continue the evolution of CDMA technology for wireless communications. These areas include over-the-air service provisioning to streamline the activation of mobile stations, the improvement of standardized speech-coding algorithms, and the improvement of the CDMA channel capacity. The last area includes interference cancellation, multiple beam adaptive antenna arrays, and the improvement of the handoff algorithm. These approaches have technical merit, and the wireless industry is cautiously optimistic about the corresponding benefits in the future.

However, before we conclude this chapter, it is important that we do not lose track of the reasons that we adopt new technologies. The reason is not to challenge the engineering community. The real reasons are to solve problems and satisfy the needs of the customer and to provide better service at the same or lower cost. As an example, capacity improvement is not a goal itself but rather a means to a goal (i.e., a reduced price to the customer). The future of the wireless industry is dependent upon holding true to this philosophy.

13.8 References

1. Duel-Hallen, A., Holtzman, J., and Zvonar, Z., "Multiuser Detection for CDMA Systems," *IEEE Personal Communications*, April 1993.
2. Naguib, A. F., Paulraj, A., and Kailath, T., "Capacity Improvement with Base–Station Antenna Arrays in Cellular CDMA", *IEEE Trans., Veh., Technol., VT-43*, 1994, pp. 691–698.
3. Pollini, G. P., "Trends in Handover Design," *IEEE Communications Magazine*, March 1996.
4. Swales, S. C., Beach, M. A., Edwards, D. J., and McGeehan, J. P., "The Performance Enhancement of Multibeam Adaptive Base–Station Antennas for Cellular Land Mobile Radio Systems," *IEEE Trans., Veh., Technol., VT-39*, 1990, pp. 56–66.
5. TIA PN-3646, "Enhanced Variable Rate Codec (EVRC), Service Option 3—Draft," December 1995.
6. TIA PN-3569, "Over-the-Air Service Provisioning of Mobile Stations in WBSS Systems—Ballot Text," September 1, 1995.
7. TIA TR45.5, Contribution TR45.5.1.1/95.07.17.03, "EVRC Host and Listening Laboratories—Draft Interim Report," July 1995.

APPENDIX A

Traffic Tables

This appendix[1] provides traffic tables for a variety of blocking probabilities and channels. The blocked calls cleared (Erlang B) call model is used. With the Erlang B model, we assume that when traffic arrives in the system, it either is served, with probability from the table, or is lost to the system. A customer attempting to place a call will therefore either see a call completion or be blocked and abandon the call. This assumption is acceptable for low blocking probabilities. In some cases, the call will be placed again after a short period of time. If too many calls reappear in the system after a short delay, the Erlang B model will no longer hold.

In Tables A.8 and A.9, where the number of channels is high (greater than 250 channels), linear interpolation between two table values is possible. We provide the deltas for one additional channel to assist in the interpolation.

1. The data in the tables was supplied by V. H. MacDonald.

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Table A.1 Offered Loads (in Erlangs) for Various Blocking Objectives: According to the Erlang B Model— System Capacity for 1–20 Channels

Trunks	0.01	0.015	0.02	0.03	0.05	0.07	0.1	0.2	0.5
P(B)=									
1	0.010	0.015	0.020	0.031	0.053	0.075	0.111	0.250	1.000
2	0.153	0.190	0.223	0.282	0.381	0.471	0.595	1.000	2.732
3	0.455	0.536	0.603	0.715	0.899	1.057	1.271	1.930	4.591
4	0.870	0.992	1.092	1.259	1.526	1.748	2.045	2.944	6.501
5	1.361	1.524	1.657	1.877	2.219	2.504	2.881	4.010	8.437
6	1.913	2.114	2.277	2.544	2.961	3.305	3.758	5.108	10.389
7	2.503	2.743	2.936	3.250	3.738	4.139	4.666	6.229	12.351
8	3.129	3.405	3.627	3.987	4.543	4.999	5.597	7.369	14.318
9	3.783	4.095	4.345	4.748	5.370	5.879	6.546	8.521	16.293
10	4.462	4.808	5.084	5.529	6.216	6.776	7.511	9.684	18.271
11	5.160	5.539	5.842	6.328	7.076	7.687	8.487	10.857	20.253
12	5.876	6.287	6.615	7.141	7.950	8.610	9.477	12.036	22.237
13	6.607	7.049	7.402	7.967	8.835	9.543	10.472	13.222	24.223
14	7.352	7.824	8.200	8.803	9.730	10.485	11.475	14.412	26.211
15	8.108	8.610	9.010	9.650	10.633	11.437	12.485	15.608	28.200
16	8.875	9.406	9.828	10.505	11.544	12.393	13.501	16.807	30.190
17	9.652	10.211	10.656	11.368	12.465	13.355	14.523	18.010	32.181
18	10.450	11.024	11.491	12.245	13.389	14.323	15.549	19.215	34.173
19	11.241	11.854	12.341	13.120	14.318	15.296	16.580	20.424	36.166
20	12.041	12.680	13.188	14.002	15.252	16.273	17.614	21.635	38.159

Table A.2 Offered Loads (In Erlangs) for Various Blocking Objectives: According to the Erlang B Model—System Capacity for 20–39 Channels

Trunks	0.005	0.01	0.015	0.02	0.03	0.05	0.07	0.1
P(B)=								
20	11.092	12.041	12.680	13.188	14.002	15.252	16.273	17.614
21	11.860	12.848	13.514	14.042	14.890	16.191	17.255	18.652
22	12.635	13.660	14.352	14.902	15.782	17.134	18.240	19.693
23	13.429	14.479	15.196	15.766	16.679	18.082	19.229	20.737
24	14.214	15.303	16.046	16.636	17.581	19.033	20.221	21.784
25	15.007	16.132	16.900	17.509	18.486	19.987	21.216	22.834
26	15.804	16.966	17.758	18.387	19.395	20.945	22.214	23.885
27	16.607	17.804	18.621	19.269	20.308	21.905	23.214	24.939
28	17.414	18.646	19.487	20.154	21.224	22.869	24.217	25.995
29	18.226	19.493	20.357	21.043	22.143	23.835	25.222	27.053
30	19.041	20.343	21.230	21.935	23.065	24.803	26.229	28.113
31	19.861	21.196	22.107	22.830	23.989	25.774	27.239	29.174
32	20.685	22.053	22.987	23.728	24.917	26.747	28.250	30.237
33	21.512	22.913	23.869	24.629	25.846	27.722	29.263	31.302
34	22.342	23.776	24.755	25.532	26.778	28.699	30.277	32.367
35	23.175	24.642	25.643	26.438	27.712	29.678	31.294	33.435
36	24.012	25.511	26.534	27.346	28.649	30.658	32.312	34.503
37	24.852	26.382	27.427	28.256	29.587	31.641	33.331	35.572
38	25.694	27.256	28.322	29.168	30.527	32.624	34.351	36.643
39	26.539	28.132	29.219	30.083	31.469	33.610	35.373	37.715

Table A.3 Offered Loads (In Erlangs) for Various Blocking Objectives: According to the Erlang B Model—System Capacity for 40–60 Channels

Trunks	0.005	0.01	0.015	0.02	0.03	0.05	0.07	0.1
P(B)=								
40	27.387	29.011	30.119	30.999	32.413	34.597	36.397	38.788
41	28.237	29.891	31.021	31.918	33.359	35.585	37.421	39.861
42	29.089	30.774	31.924	32.838	34.306	36.575	38.447	40.936
43	29.944	31.659	32.830	33.760	35.255	37.565	39.473	42.012
44	30.801	32.546	33.737	34.683	36.205	38.558	40.501	43.088
45	31.660	33.435	34.646	35.609	37.156	39.551	41.530	44.165
46	32.521	34.325	35.556	36.535	38.109	40.545	42.559	45.243
47	33.385	35.217	36.468	37.463	39.063	41.541	43.590	46.322
48	34.250	36.111	37.382	38.393	40.019	42.537	44.621	47.401
49	35.116	37.007	38.297	39.324	40.976	43.535	45.654	48.481
50	35.985	37.904	39.214	40.257	41.934	44.534	46.687	49.562
51	36.856	38.802	40.132	41.190	42.893	45.533	47.721	50.644
52	37.728	39.702	41.052	42.125	43.853	46.533	48.756	51.726
53	38.601	40.604	41.972	43.061	44.814	47.535	49.791	52.808
54	39.477	41.507	42.894	43.999	45.777	48.537	50.827	53.891
55	40.354	42.411	43.817	44.937	46.740	49.540	51.864	54.975
56	41.232	43.317	44.742	45.877	47.704	50.544	52.902	56.059
57	42.112	44.224	45.667	46.817	48.669	51.548	53.940	57.144
58	42.993	45.132	46.594	47.759	49.636	52.553	54.979	58.229
59	43.875	46.041	47.522	48.701	50.603	53.559	56.018	59.315
60	44.759	46.951	48.451	49.645	51.570	54.566	57.058	60.401

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Table A.4 Offered Loads (In Erlangs) for Various Blocking Objectives: According to the Erlang B Model—System Capacity from 61–80 Channels

Trunks	0.005	0.01	0.015	0.02	0.03	0.05	0.07	0.1
P(B)=								
61	45.644	47.863	49.381	50.590	52.539	55.573	58.099	61.488
62	46.531	48.776	50.311	51.535	53.509	56.581	59.140	62.575
63	47.418	49.689	51.243	52.482	54.479	57.590	60.181	63.663
64	48.307	50.604	52.176	53.429	55.450	58.599	61.224	64.750
65	49.197	51.520	53.110	54.377	56.422	59.609	62.266	65.839
66	50.088	52.437	54.044	55.326	57.395	60.620	63.309	66.927
67	50.980	53.355	54.980	56.276	58.368	61.631	64.353	68.016
68	51.874	54.273	55.916	57.226	59.342	62.642	65.397	69.106
69	52.768	55.193	56.853	58.178	60.316	63.654	66.442	70.196
70	53.663	56.113	57.791	59.130	61.292	64.667	67.487	71.286
71	54.560	57.035	58.730	60.083	62.268	65.680	68.532	72.376
72	55.457	57.957	59.670	61.036	63.244	66.694	69.578	73.467
73	56.356	58.880	60.610	61.991	64.222	67.708	70.624	74.558
74	57.255	59.804	61.551	62.945	65.199	68.723	71.671	75.649
75	58.155	60.729	62.493	63.901	66.178	69.738	72.718	76.741
76	59.056	61.654	63.435	64.857	67.157	70.753	73.765	77.833
77	59.958	62.581	64.379	65.814	68.136	71.769	74.813	78.925
78	60.861	63.508	65.322	66.772	69.116	72.786	75.861	80.018
79	61.765	64.435	66.267	67.730	70.097	73.803	76.909	81.110
80	62.669	65.364	67.212	68.689	71.078	74.820	77.958	82.203

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Table A.5 Offered Loads (In Erlangs) for Various Blocking Objectives: According to the Erlang B Model—System Capacity for 81–100 Channels

Trunks	0.005	0.01	0.015	0.02	0.03	0.05	0.07	0.1
P(B)=								
81	63.574	66.293	68.158	69.648	72.059	75.838	79.007	83.297
82	64.481	67.223	69.104	70.608	73.042	76.856	80.057	84.390
83	65.387	68.153	70.051	71.568	74.024	77.874	81.107	85.484
84	66.295	69.085	70.999	72.529	75.007	78.893	82.157	86.578
85	67.204	70.016	71.947	73.491	75.991	79.912	83.207	87.672
86	68.113	70.949	72.896	74.453	76.975	80.932	84.258	88.767
87	69.023	71.882	73.846	75.416	77.959	81.952	85.309	89.861
88	69.933	72.816	74.796	76.379	78.944	82.972	86.360	90.956
89	70.844	73.750	75.746	77.342	79.929	83.993	87.411	92.051
90	71.756	74.685	76.697	78.306	80.915	85.014	88.463	93.146
91	72.669	75.621	77.649	79.271	81.901	86.035	89.515	94.242
92	73.582	76.557	78.601	80.236	82.888	87.057	90.568	95.338
93	74.496	77.493	79.553	81.202	83.875	88.079	91.620	96.434
94	75.411	78.431	80.506	82.167	84.862	89.101	92.673	97.530
95	76.326	79.368	81.460	83.134	85.850	90.123	93.726	98.626
96	77.242	80.307	82.414	84.101	86.838	91.146	94.779	99.722
97	78.158	81.245	83.368	85.068	87.827	92.169	95.833	100.819
98	79.075	82.185	84.323	86.036	88.815	93.193	96.887	101.916
99	79.993	83.125	85.279	87.004	89.805	94.217	97.941	103.013
100	80.911	84.065	86.235	87.972	90.794	95.240	98.995	104.110

Table A.6 Offered Loads (in Erlangs) for Various Blocking Objectives: According to the Erlang B Model—System Capacity for 105–200 Channels

Trunks	0.005	0.01	0.015	0.02	0.03	0.05	0.07	0.1
P(B)=								
105	85.518	88.822	91.030	92.823	95.747	100.371	104.270	109.598
110	90.147	93.506	95.827	97.687	100.713	105.496	109.550	115.090
115	94.768	98.238	100.631	102.552	105.680	110.632	114.833	120.585
120	99.402	102.977	105.444	107.426	110.655	115.772	120.121	126.083
125	104.047	107.725	110.265	112.307	115.636	120.918	125.413	131.583
130	108.702	112.482	115.094	117.195	120.622	126.068	130.708	137.087
135	113.366	117.247	119.930	122.089	125.615	131.222	136.007	142.593
140	118.039	122.019	124.773	126.990	130.612	136.380	141.309	148.101
145	122.720	126.798	129.622	131.896	135.614	141.542	146.613	153.611
150	127.410	131.584	134.477	136.807	140.621	146.707	151.920	159.122
155	132.106	136.377	139.337	141.724	145.632	151.875	157.230	164.636
160	136.810	141.175	144.203	146.645	150.647	157.047	162.542	170.152
165	141.520	145.979	149.074	151.571	155.665	162.221	167.856	175.668
170	146.237	150.788	153.949	156.501	160.688	167.398	173.173	181.187
175	150.959	155.602	158.829	161.435	165.713	172.577	178.491	186.706
180	155.687	160.422	163.713	166.373	170.742	177.759	183.811	192.227
185	160.421	165.246	168.602	171.315	175.774	182.943	189.133	197.750
190	165.160	170.074	173.494	176.260	180.809	188.129	194.456	203.273
195	169.905	174.906	178.390	181.209	185.847	193.318	199.781	208.797
200	174.653	179.743	183.289	186.161	190.887	198.508	205.108	214.323

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Table A.7 Offered Loads (In Erlangs) for Various Blocking Objectives: According to the Erlang B Model—System Capacity for 205–245 Channels

Trunks	0.005	0.01	0.015	0.02	0.03	0.05	0.07	0.1
P(B)=								
205	179.407	184.584	188.192	191.116	195.930	203.700	210.436	219.849
210	184.165	189.428	193.099	196.073	200.976	208.894	215.765	225.376
215	188.927	194.276	198.008	201.034	206.023	214.089	221.096	230.904
220	193.694	199.127	202.920	205.997	211.073	219.287	226.427	236.433
225	198.464	203.981	207.836	210.963	216.125	224.485	231.760	241.963
230	203.238	208.839	212.754	215.932	221.180	229.686	237.094	247.494
235	208.016	213.700	217.675	220.902	226.236	234.887	242.430	253.025
240	212.797	218.564	222.598	225.876	231.294	240.090	247.766	258.557
245	217.582	223.430	227.524	230.851	236.354	245.295	253.103	264.089

Table A.8 Offered Loads (in Erlangs) for Various Blocking Objectives: According to the Erlang B Model— System Capacity for 250–600 Channels

Trunks / P(B)=	0.005	0.01	0.015	0.02	0.03	0.05	0.07	0.1
250	222.370	228.300	232.452	235.828	241.415	250.500	258.441	269.622
Δ	0.961	0.977	0.988	0.998	1.015	1.042	1.069	1.107
300	270.410	277.144	281.853	285.707	292.142	302.617	311.866	324.961
Δ	0.966	0.980	0.991	1.001	1.017	1.044	1.070	1.108
350	318.698	326.155	331.424	335.738	342.995	354.836	365.359	380.384
Δ	0.969	0.984	0.994	1.005	1.018	1.045	1.071	1.109
400	367.163	375.334	381.128	385.963	393.895	407.096	418.890	435.813
Δ	0.972	0.989	0.998	1.004	1.020	1.046	1.071	1.109
450	415.779	424.774	431.022	436.178	444.877	459.408	472.456	491.263
Δ	0.975	0.987	0.997	1.006	1.021	1.047	1.072	1.109
500	464.518	474.130	480.890	486.480	495.919	511.759	526.049	546.730
Δ	0.977	0.989	0.999	1.007	1.022	1.048	1.072	1.110
550	513.361	523.600	530.843	536.846	547.012	564.142	579.663	602.208
Δ	0.979	0.991	1.000	1.008	1.023	1.048	1.073	1.110
600	562.292	573.142	580.859	587.267	598.145	616.552	633.295	657.697

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Table A.9 Offered Loads (in Erlangs) for Various Blocking Objectives: According to the Erlang B Model—System Capacity for 600–1050 Channels

Trunks P(B)=	0.005	0.01	0.015	0.02	0.03	0.05	0.07	0.1
600	562.292	573.142	580.859	587.267	598.145	616.552	633.295	657.697
Δ	0.983	0.992	1.001	1.009	1.023	1.049	1.073	1.110
650	611.418	622.748	630.927	637.732	649.313	668.982	686.941	713.193
Δ	0.981	0.993	1.002	1.010	1.024	1.049	1.073	1.110
700	660.462	672.410	681.042	688.238	700.511	721.432	740.598	768.697
Δ	0.982	0.994	1.003	1.011	1.024	1.049	1.073	1.110
750	709.586	722.119	731.196	738.777	751.735	773.896	794.266	824.206
Δ	0.984	0.995	1.004	1.011	1.025	1.050	1.074	1.110
800	758.762	771.872	781.386	789.346	802.981	826.375	847.943	879.719
Δ	0.985	0.996	1.004	1.012	1.025	1.050	1.074	1.110
850	807.987	821.662	831.608	839.942	854.247	878.865	901.627	935.236
Δ	0.985	0.996	1.005	1.012	1.026	1.050	1.074	1.110
900	857.256	871.487	881.857	890.561	905.530	931.365	955.317	990.757
Δ	0.986	0.997	1.005	1.013	1.026	1.050	1.074	1.110
950	906.565	921.343	932.132	941.202	956.829	983.875	1009.013	1046.281
Δ	0.987	0.998	1.006	1.013	1.026	1.050	1.074	1.111
1000	955.910	971.226	982.430	991.862	1008.142	1036.393	1062.715	1101.808
Δ	0.988	0.998	1.006	1.014	1.027	1.050	1.074	1.111
1050	1005.289	1021.136	1032.748	1042.539	1059.468	1088.918	1116.420	1157.337

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List of Abbreviations and Acronyms

A

ABR	Answer Bid Ratio
AbS	Analysis by Synthesis
AC	Authentication Center
ACM	Address Complete Message
ADPCM	Adaptive Differential Pulse Code Modulation
AMPS	Advanced Mobile Phone System
APC	Adaptive Predictive Coding
APC	Automatic Power Control
ARLAN	Advanced Radio LAN
ARPA	Advanced Research Project Agency
ARQ	Automatic Repeat Request
ATIS	Alliance for Telecommunications Industry Solutions
ASR	Answer Seizure Ratio
AWGN	Additive White Gaussian Noise

B

BER	Bit-Error Rate
BHCA	Busy Hour Call Attempts
BML	Business Management Layer
BPSK	Binary Phase-Shift Keying
BS	Base Station
BSAP	Base Station Application Part
BSC	Base Station Controller
BSMAP	Base Station Management Application Part
BTS	Base Transceiver System

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C

CDCP	Call Detail Collection Point
CDGP	Call Detail Generation Point
CDIS	Call Detail Information Source
CDMA	Code-Division Multiple Access
CDRP	Call Detail Rating Point
CELP	Code-Excited Linear Prediction
CGI	Call Global Identification
CI	Call Identity
CM	Channel Modem
CNIP	Calling Number Identification Presentation
CNIR	Calling Number Identification Restriction
CRC	Cyclic Redundancy Check
CSMA/CA	Carrier-Sensed Multiple Access with Collision Avoidance

D

DCE	Data Communication Equipment
DCN	Data Communications Network
DMH	Data Message Handler
DN	Directory Number
DSSS	Direct Sequence Spread Spectrum
DTAP	Direct Transfer Application Part
DTE	Data Terminal Equipment
DTMF	Dual-Tone Multifrequency

E

EIA	Electronic Industry Association
EIR	Equipment Identity Register
EIRP	Effective Isotropic Radiated Power
EML	Element Management Layer
ERP	Effective Radiated Power
ESN	Electronic Serial Number
EVRC	Enhanced Variable Rate Codec

F

FAF	Floor Attenuation Factor
FCC	Federal Communications Commission
FCS	Frame Check Sequence
FD	Full Duplex
FDD	Frequency-Division Duplex
FDMA	Frequency-Division Multiple Access

FEC	Forward Error Correction
FHSS	Frequency-Hopping Spread Spectrum
FM	Frequency Modulation

G

GBN	Go-Back-N
GFSK	Gaussian Frequency Shift Keying
GHz	gigahertz
GMSK	Gaussian Minimum Shift Keying
GOS	Grade of Service
GPS	Global Positioning System
GSM	Global System of Mobile Communications

H

HAAT	Height Above Average Terrain
HD	Half Duplex
HIPERLAN	High-Performance Radio Local Area Network
HLR	Home Location Register

I

IAM	Initial Address Message
ICMP	Internet Control Message Protocol
IDs	Identifications
IEEE	Institute of Electrical and Electronic Engineers
IF	Intermediate Frequency
IMSI	International Mobile Station Identification
IN	Intelligent Network
IP	Internet Protocol
IPCP	Internet Protocol Control Protocol
ISDN	Integrated Services Digital Network
ISM	Industrial Scientific Medical
ISO	International Standard Organization
IWF	InterWorking Function

K

kbps	kilobit per second
kHz	kilohertz

L

LAC	Local Area Code
LAN	Local Area Network

LAC	Linear Amplifier Circuit
LBT	Listen-Before-Talk
LCP	Link Control Protocol
LOS	Line of Sight
LPC	Linear Predictive Coding
LSP	Line Spectral Pairs

M

MAC	Media Access
MAN	Metropolitan Area Network
MAP	Mobile Application Part
MC	Message Center
MCC	Mobile Country Code
Mcps	million chips per second
MF	Multifrequency
MF	Mediation Function
MHz	megahertz
MIN	Mobile Identification Number
MM	Mobility Management
MMAP	Mobility Management Application Part
MNC	Mobile Network Code
MNRP	Mobile Network Registration Protocol
MOS	Mean Opinion Score
MS	Mobile Station
MSC	Mobile Switching Center
MTP	Message Transfer Part

N

NAM	Number Assignment Module
NE	Network Element
NEF	Network Element Function
NEL	Network Element Layer
NID	Network Identification
NML	Network Management Layer

O

OA&M	Operation, Administration, & Maintenance
OAM&P	Operation, Administration, Maintenance, & Provisioning
OQPSK	Offset Quadrature Phase-Shift Keying
OS	Operations System
OSF	Operations Systems Function

OSI	Open System Interconnect
OTASP	Over-The-Air Service Provisioning

P

PBX	Private Branch Exchange
PCM	Pulse Code Modulation
PCMCIA	Personal Computer Memory Card International Association
PCN	Personal Communications Network
PCS	Personal Communications Services
PCSAP	PCS Application Part
PDA	Personal Digital Assistants
PIN	Personal Identification Number
PLMN	Public Land Mobile Network
PMC	Personal Mobility Controller
PN	Pseudorandom Noise
PPDN	Public Packet Data Network
PPP	Point-to-Point Protocol
PSPDN	Public Switched Packet Data Network
PSTN	Public Switched Telephone Network

Q

QAF	Q Adapter Function
QCELP	QUALCOMM Code-Excited Linear Prediction
QoS	Quality of Service
QPSK	Quadrature Phase-Shift Keying

R

RASC	Radio Access System Controller
RELP	Residual-Excited Linear Prediction
RLP	Radio Link Protocol
RPC	Radio Port Controller

S

SCCP	Signaling Connection Control Part
SCI	Synchronized Capsule Indicator
SCM	Station Class Mark
SID	System Identification
SML	Service Management Layer
SMRS	Specialized Mobile Radio Services
SMS	Short Message Service

SNDCF	SubNetwork Dependent Convergence Function
SOF	Start Of Frame
SOM	Start Of Message
SOS	Start Of Slot
SRP	Selective Repeat Request
SS	Spread Spectrum
SSD	Shared Secret Data
SS7	Signaling System 7

T

TA	Terminal Adapter
TCP	Transmission Control Protocol
TDMA	Time-Division Multiple Access
TE	Terminal Equipment
TH	Time Hopped
TIA	Telecommunications Industry Association
TMA	Telesystems Microcellular Architecture
TMC	Terminal Mobility Controller
TMN	Telecommunications Management Network
TMSI	Temporary Mobile Station Identification
T-R	Transmitter Receiver

U

UCB	University of California at Berkeley
UCLA	University of California at Los Angeles

V

VLR	Visitor Location Register

W

WAN	Wide Area Network
W-CDMA	Wideband CDMA
WIN	Wireless Intelligent Network
WLAN	Wireless Local Area Network
WORM	Window-control Operation based on Reception Memory
WPBX	Wireless Private Branch Exchange
WS	Work Station
WSF	Work Station Function

X

XC	Transcoder

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COMMUNICATIONS
Mobile & Wireless

Applications of CDMA in Wireless/Personal Communications

Vijay K. Garg • Kenneth Smolik • Joseph E. Wilkes

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