CHAPTER 12

Evolution of TDMA-Based 2G Systems to 3G Systems

12.1 Introduction

Third-generation (3G) wireless systems will offer access to services anywhere from a single terminal; the old boundaries between telephony, information, and entertainment services will disappear. Mobility will be built into many of the services currently considered as fixed, especially in such areas as high-speed access to the Internet, entertainment, information, and electronic commerce (e-commerce) services. The distinction between the range of services offered via wireline or wireless will become less and less clear and, as the evolution toward 3G mobile services speeds up, these distinctions will disappear within a decade.

Applications for 3G wireless networks will range from simple voice-only communications to simultaneous video, data, voice, and other multimedia applications. One of the main benefits of 3G is that it will allow a broad range of wireless services to be provided efficiently to many users.

Packet-based Internet protocol (IP) technology will be at the core of the 3G services (refer to Chapters 13 and 14 for details). Users will have continuous access to online information. E-mail messages will arrive at hand-held terminals nearly instantaneously and business users will be able to stay permanently connected to company intranets. Wireless users will be able to make video conference calls to the office and surf the Internet simultaneously, or play computer games interactively with friends in other locations. Figure 12.1 shows the bit rate requirement for various services.

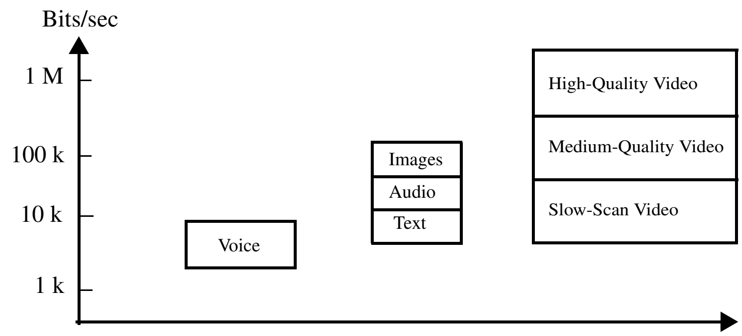

Figure 12.1 User Data Requirements

In 1997, the TIA/EIA IS-136 community, through the Universal Wireless Communications Consortium (UWCC) and Telecommunications Industry Association (TIA) TR 45.3, adopted a three-part strategy for evolving its IS-136 time division multiple access (TDMA)-based networks to 3G wireless networks in order to satisfy IMT-2000 requirements. The strategy consists of

- Enhancing the voice and data capabilities of the existing 30 kHz carrier (IS-136+)
- Adding a 200-kHz carrier (enhanced data rates for GSM evolution [EDGE]) for high-speed data (384 kbps) in high-mobility applications
- Introducing a 1.6-MHz carrier for very high-speed data (2 Mbps) in low-mobility applications (W-TDMA, FMA1 without spreading)

The highlight of this strategy was the global convergence of IS-136 TDMA with Global System for Mobile (GSM) Communications TDMA through the evolution of the 200-kHz GSM carrier for supporting high-speed data applications (384 kbps) while also improving the 30-kHz carrier for voice and mid-speed data applications.

In this chapter, we first discuss enhancements to IS-136 and then focus on general packet radio service (GPRS) for providing packet data services in IS-136 and GSM systems. We then concentrate on EDGE, which will be used for converging the IS-136 and GSM systems to offer IMT-2000-based 3G services.

12.2 IS-136+

The goal of the IS-136 voice services program is to develop a higher-quality voice service, focusing on enhancements to voice quality under fading channels, high background noise, tandeming, and music conditions. At the beginning of the IS-136+ program, it was recognized that the largest opportunity for improvement in faded channels existed in the downlink for two reasons:

1. Downlink was perceived as the limiting link in urban areas.
2. Uplink enhancements such as interference cancellation had recently been shown to provide large uplink gains.

Enhancements to IS-36 include:

- An improved channel coding (CC2) and interleaving
- An improved vocoder US1/enhanced full-rate (EFR)
- Improved modulation scheme $\pi/4$ -DQPSK and 8-PSK

The details of these enhancements are discussed below.

With the definition of a new time slot format, improved channel encoding (CC2), and interleaving options, the robust voice service mode can achieve an additional 4 dB faded channel improvement over the existing IS-641 vocoder (CC1). One of the major difference between CC1 and CC2 is that in CC2 certain fields are eliminated from the downlink (base station to mobile station) slot structure to free 18 bits for use as additional channel coding. The CC2 convolutional encoder uses a tail bit and a higher-constraint length code ($K = 7$ instead of $K = 6$ in CC1) to achieve channel coding gain over CC1. CC2 also supports a 3-slot interleaving mode for improved time diversity over the conventional 2-slot interleaving mode used in CC1.

The detailed CC2 downlink slot structure is given in Figure 12.2.

28	142	12	136	1	1	4
SYNC	Data	CDVCC	Data	F	RSVD	PRAMP

Figure 12.2 CC2 Downlink Slot Format in IS-136+

In the CC2 downlink, a 28-bit SYNC field is used by the receiver for synchronization purposes, a 142-bit and a 136-bit data field together form the total 278-bit data field, a 12-bit coded digital verification color code (CDVCC) field is used to minimize channel interference, a 1-bit fast power control (F) field is used for a faster version of uplink power control, and a 1-bit reserved (RSVD) field and 4-bit power ramp (PRAMP) field allow time for changes in downlink output power. The total number of bits in one slot is 324. The major difference between CC2 and CC1 time slot structures is that the slow associated control channel (SACCH) and coded digital control channel locator (CDL) fields are not used in CC2. The removal of the SACCH has little impact since all messages can also be sent via a fast associated control channel (FACCH) message, which replaces the voice information with signaling data. Although the FACCH replaces voice, it has been found that if the FACCH messages are sent either between talk spurts or spaced far enough in time, they are unnoticeable. Table 12.1 gives bit allocation for IS-641A algebraic code excited linear prediction (ACELP) vocoder. CC2 with $K = 7$ provides a 2-dB improvement in frame error rate (FER) over CC1 with $K = 6$ at 10 Hz Doppler shift, where K is the constraint length in channel coding.

The 1-slot format (#1) (refer to Table 12.2) has no interleaving delay, but requires 6 dB more link margin than the conventional 2-slot format to support a 1 percent Class Ia FER at 10 Hz Doppler shift. To ensure adequate voice quality, the 1-slot format is best used for indoor applications and environments in which signal-to-interference (S/I) and signal-to-noise (S/N) are relatively high (20 dB). The 3-slot format (#3) is the extra robustness mode, which, in conjunction with CC2, provides about 3.7 dB improvement in downlink performance. To minimize extra delay, 3-slot interleaving is limited to only one link at a time. This ensures that for mobile-to-mobile calls, the increase in delay over 2-slot format will be limited to 20 ms. Notice application of format #4 adds additional time diversity with 3-slot interleaving to the space diversity common in existing base stations. The additional time diversity gain is about 0.5 dB less at 1 percent FER that seen on the downlink in format #3 [1,2].

Table 12.1 Bit Allocation for IS-641-A ACELP Vocoder

Information	Number of Bits per Frame
LP filter coefficients	26
Adaptive excitation	26
Fixed or algebraic excitation	68
Gains	28
Total bits	148
Rate	7.4 kbps

Table 12.2 Interleaving Options for CC2

Format	Uplink Interleaving	Downlink Interleaving
#1	1-slot	1-slot
#2	2-slot	2-slot
#3	2-slot	3-slot
#4	3-slot	2-slot

12.2.1 US1/EFR Vocoder

The US1 vocoder is identical to the GSM enhanced full-rate (EFR) vocoder, which is also used by North American GSM1900 operators. The US1 vocoder operates at 12.2 kbps, and under high S/I and S/N offers a high-quality voice service. The US1 vocoder is identical to the IS-641 vocoder in basic structure. Both vocoders are based on ACELP, with the major difference being that the US1 vocoder employs more bits to represent the various speech parameters (see Table 12.3). Since 244 bits are generated in every 20-ms speech frame, the resulting output bit rate is 12.2 kbps.

Figures 12.3 and 12.4 show channel coding for the downlink and uplink used with US1 vocoder. The coded and interleaved bits are combined (see Figure 12.3) to form a series of 3-bit sequences or triads. For the downlink, there are 399 bits (133 triads), whereas for the uplink there are 372 bits (124 triads). For the downlink, the bits are combined such that every Class Ia bit (coded and interleaved) has one Class II bit to form the first 89 triads. The remaining 44 triads are formed from Class Ib bits.

For the uplink, the first 86 triads include two Class Ia bits and one Class II bit. The next three triads are composed of two Class Ib and one Class II bit, whereas the remaining 35 triads are formed from Class Ib bits only.

Table 12.3 Bit Allocation for US1 ACELP Vocoder

Information	Number of Bits per Frame
LP filter coefficients	38
Adaptive excitation	30
Fixed or algebraic excitation	140
Gains	36
Total bits	244
Rate	12.2 kbps

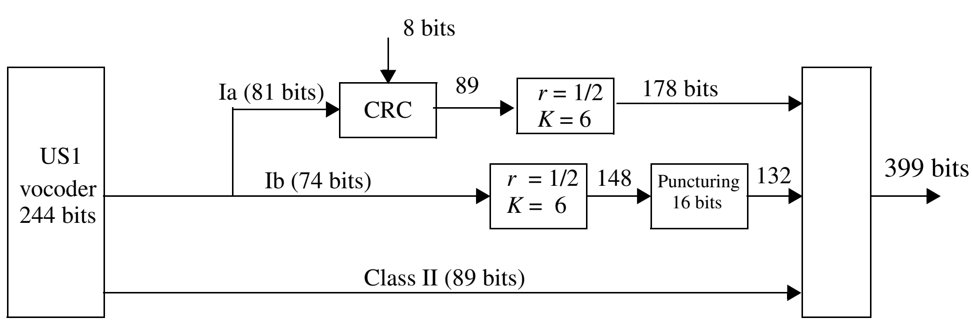

Figure 12.3 Channel Coding for Downlink with US1 Vocoder

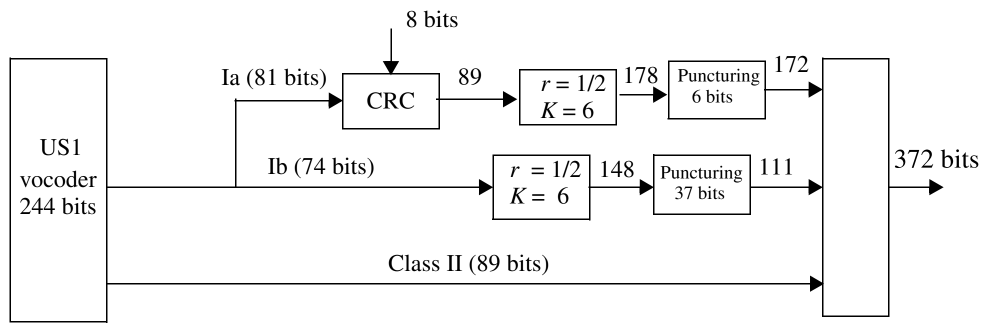

Figure 12.4 Channel Coding for Uplink with US1 Vocoder

The triads are reordered to provide additional interleaving gain [1], and then intraslot interleaving is applied. As with CC2, there are three primary interleaving options: 1-slot, 2-slot, and 3-slot. In 1-slot interleaving, there is no intraslot interleaving and current triads are simply transmitted in the current slot. In 2-slot interleaving, certain triads (from the current triad vector) are transmitted in the current slot, and remaining triads in the next slot. Thus, the current slot contains triads from the current and previous triad vector. In 3-slot interleaving, the concept is extended to include another set of triads in each transmitted slot. The current slot contains triads from the current triad vector, the previous triad vector, and the one before the previous triad vector. To minimize delay in mobile-to-mobile calls, the 3-slot option cannot be used simultaneously on both the uplink and downlink.

The selection of US1 for use in IS-136 is the initial step for the convergence of speech coding technologies between IS-136 and GSM-based systems. US1 has been deployed by both GSM1900 operators in North America and GSM900/1800 operators in Europe, Asia, and Africa.

TIA TR 45.3 committee modified the original GSM 200 kHz channel coding used with GMSK modulation by optimizing for the 30-kHz channel used in IS-136.

The goal of the European Telecommunications Standards Institute (ETSI) adaptive multirate (AMR) coder program was to develop a robust full- and half-rate solution to provide significant improvement in voice quality at low S/I and S/N. The AMR design selected by ETSI incorporates multiple submodes for use in full- or half-rate modes that are determined by the channel quality. The defined submodes, speech coder source rates, and channel coding rates for the ETSI AMR full-rate and half-rate are listed in Tables 12.4 and 12.5. The speech coder source rates common to the AMR design, US1/GSM EFR, and IS-136 full-rate speech coder (IS-641) are in parentheses. The full-rate AMR has submodes that incorporate bit-exact versions of both the 12.2-kbps US1/GSM EFR and 7.4-kbps IS-641 full-rate speech coders. Table 12.6 gives the mean opinion score (MOS) results for speech coded with background noise. Table 12.7 compares the delay in milliseconds for IS-641, GSM-EFR, and G.728 vocoders.

Table 12.4 ETSI AMR Full-Rate Submodes

Submode	Speech Coder Rate (kbps)	Channel Coding Rate (kbps)
1	12.2 (US1/GSM EFR)	10.6
2	10.2	12.6
3	7.95	14.85
4	7.4 (IS-641)	15.4
5	6.7	16.1
6	5.9	16.9
7	5.15	17.65
8	4.75	18.05

Table 12.5 ETSI AMR Half-Rate Submodes

Submode	Speech Coder Rate (kbps)	Channel Coding Rate (kbps)
1	7.95	3.45
2	7.4 (IS-641)	4.0
3	6.7	4.7
4	5.9	5.5
5	5.15	6.25
6	4.75	6.65

Table 12.6 Mean Opinion Score (MOS) for Speech Coded with and without Background Noise

Condition	Original	IS-641	G.728	GSM-EFR
Clean speech	4.34	4.09	4.23	4.26
Clean speech*		3.62	3.99	4.13
15 dB babble	3.75	3.49	3.81	3.70
15 dB babble*		3.08	3.69	3.47
20 dB car noise	3.72	3.61	3.64	3.75
20 dB car noise*		3.11	3.58	3.48
15 dB office noise	3.70	3.40	3.51	3.58
15 dB office noise*		2.75	3.55	3.31
15 dB music	3.99	3.82	3.98	3.99
15 dB music*		3.16	3.92	3.85

* With background noise

Table 12.7 Delay (ms) for Systems Based on IS-641, GSM-EFR, and G.728

Delay Cause	IS-641	G.728	GSM-EFR
Look-ahead	5	0	0
Frame size	20	20	0.625
Processing	16	16	0.5
Bitstream buffer	0	0	19.375
Transmission	26.6	6.6	6.6
Delay	67.6	42.6	27.1

12.2.2 Modulation

Differential Quadrature Phase Shift Keying (DQPSK). The modulation scheme used for the CC2 is $\pi/4$ -DQPSK. In $\pi/4$ -DQPSK, every two bits of information are encoded into one modulator symbol. The actual information is differentially encoded in the phase change from one symbol to the next. The most significant bit is the first bit in the input stream, and Gray code mapping is used to minimize the probability of bit error. With $\pi/4$ -DQPSK modulation, the 324 input bits per slot are translated into 162 modulator output symbols. The IS-136 symbol is 24.3 kHz, and thus the raw or gross instantaneous bit rate is 48.6 kbps. The gross rate for a full-rate user is 16.2 kbps, which consists of 7.4 kbps of speech, 6.5 kbps of channel coding, and 2.3 kbps for the remaining field within the time slot.

The CC1 and CC2 provide a spectral efficiency of 1.62 b/s/Hz based on 48.6 kbps gross bit rate and an effective channel bandwidth of 30 kHz.

8-PSK. The design goal of the 8-PSK modulation was to maintain the 24.3-kHz symbol rate so that the existing transmit and receive filters (square-root raised cosine with roll-off of 0.55) could be used and the existing 30 kHz channel bandwidth is maintained. The slot length of 6.67 ms and 162 symbols per slot were also to be maintained. Thus, with 3 bits per symbol, 8-PSK modulation supports 486 bits per slot, which is consistent with the slot structures (see Figure 12.5). The instantaneous gross bit rate is 72.9 kbps, which is 50 percent more than that supported by $\pi/4$ -DQPSK. For full-rate voice users using the US1 vocoder with 8-PSK modulation, the effective gross rate is 24.3 kbps. For downlink, this 24.3 kbps consists of 12.2 kbps of speech bits, 7.55 kbps of channel coding, and 4.75 kbps for the remaining field within the slot. For the uplink, the speech bit rate is the same, but the channel coding contribution is 6.0 kbps, and the remaining fields contribution is 6.1 kbps. 8-PSK modulation provides a spectral efficiency of 2.43 b/s/Hz, a 50 percent increase over $\pi/4$ -DQPSK.

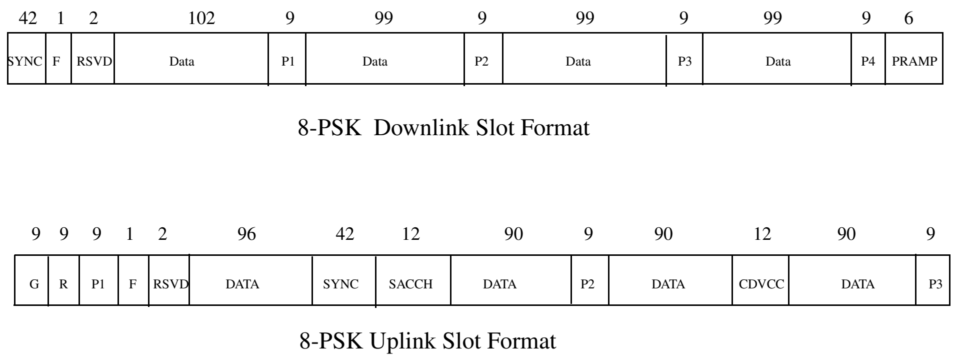

Figure 12.5 8-PSK Slot Structure

Figure 12.6 shows a pictorial representation of 8-PSK bit-to-symbol mapping.

The detailed downlink and uplink slot structures for the US1 vocoder with 8-PSK modulation are shown in Figure 12.5. One of the major difference between this slot structure and those of CC1 and CC2 slot formats is the inclusion of pilot fields, which are included to facilitate coherent detection at the receiver by developing an estimate of the channel condition.

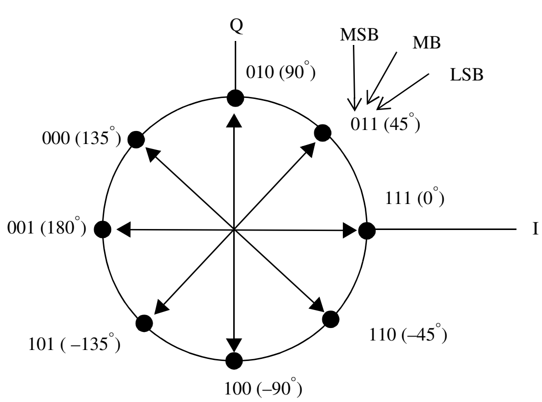

Figure 12.6 8-PSK Bit-to-Symbol Mapping

Given that the earlier versions of IS-136 allowed mobile stations to detect and utilize information (SYNC) from adjacent slots on the same RF carrier, the SYNC in 8-PSK slot format remains modulated at $\pi/4$ -DQPSK. Thus, within a given time slot there are both

$\pi/4$ -DQPSK and 8-PSK modulations. This produces a potential problem in decoding 8-PSK since it needs a phase reference. This could be provided by adding a reference symbol of known phase (e.g., zero phase) after the differentially encoded SYNC. This will waste 3 bits that could otherwise be used for data, and will place the importance of the phase reference for the entire slot on the reliable detection of only one symbol.

An alternate approach was used to allow channel sharing 8-PSK time slots with $\pi/4$ -DQPSK slots. The SYNC remains differentially encoded from the preceding symbol in the previous time slot, and all of its values remain the same as currently defined. After differentially encoding the SYNC from the immediately proceeding symbol, a constant phase-shift equal to the absolute phase of the last symbol of the immediately proceeding time slot is added to all of the symbols following the SYNC. In this manner, the phase-shift of the receiver drives from the SYNC signal can also be removed from the data fields in order to obtain the absolute phases.

12.3 GSM Evolution for Data

From a radio access perspective, adding 3G capabilities to 2G systems mainly means supporting higher bit rates. Possible scenarios depend on spectrum availability for the network service provider. Depending on the spectrum situation, two different migration paths must be supported:

- Reframing of existing spectrum bands
- New or modified spectrum bands

Two 3G radio access schemes have been identified to support the different spectrum scenarios:

- Enhanced data rates for GSM evolution (EDGE) with high-level modulation in 200 kHz TDMA channel is based on plug-in transceiver equipment, thereby allowing the migration of existing bands in small spectrum segments.
- Universal mobile telecommunications services (UMTS) is a new radio access network based on 5-MHz wideband CDMA (W-CDMA) and optimized for efficient support of 3G services. UMTS can be used in both new and existing spectra.

From a network point of view, 3G capabilities implies the addition of packet-switched (PS) services, Internet access, and IP connectivity. With this approach, the existing mobile networks will reuse the elements of mobility support, user authentication/service handling, and circuit-switched (CS) services. PS services and IP connectivity will then be added to provide a mobile multimedia core network by evolving existing mobile network.

In Europe, GSM is moving to develop and enhance cutting-edge, customer-focused solutions to meet the challenges of the new millennium and 3G mobile services. When GSM was first designed, no one could have predicted the dramatic growth of the Internet and the rising demand for multimedia services. These developments have brought about new challenges to the world of GSM. For GSM operators, the emphasis is now rapidly changing from instigating and driving the development of technology and fundamentally enabling mobile data transmission toward improving speed, quality, simplicity, coverage, and reliability in terms of tools and services that will boost mass market application.

People increasingly demand access to information and services wherever they are and whenever they want. GSM should provide that connectivity. Internet access, Web browsing, and the whole range of mobile multimedia capability is the major driver for development of higher data speed technologies.

Current data traffic on most GSM networks is modest, less than 3 percent of total GSM traffic. But with the new initiatives coming to fruition during the course of the next two to three years, exponential growth in data traffic is forecast. Messaging-based applications may reach a penetration of up to 25 percent in developed markets by the year 2001, and 70 percent by 2003. GSM data transmission using high-speed circuit-switched data (HSCSD) and GPRS may reach a penetration of 10 percent by 2001 and 25 percent by 2003.

Today's GSM operators will have two nonexclusive options for evolving their networks to 3G wideband multimedia operation: (1) they can use GPRS and EDGE (discussed below) in the existing radio spectrum and in small amounts of the new spectrum, or (2) they can use wideband CDMA (W-CDMA) in the new 2-GHz bands, or in large amounts of the existing spectrum. Both approaches offer a high degree of investment flexibility because roll-out can proceed in line with market demand and extensive reuse of existing network equipment and radio sites.

In the new 2-GHz bands, 3G capabilities will be delivered using a new wideband radio interface that will offer much higher user data rates than are available today—384 kbps in the wide area and up to 2 Mbps in local areas. Of equal importance for such services will be the high-speed packet switching provided by GPRS and its connection to public and private IP networks.

Even without the new wideband spectrum, GSM and Digital-Advanced Mobile Phone System (D-AMPS) (IS-136) operators will be able to use existing radio bands to deliver 3G services by evolving current networks and deploying GPRS and EDGE technologies. In the early years of 3G service deployment, a large proportion of wireless traffic will still be voice-only and low-rate data. So whatever the ultimate capabilities of 3G networks, efficient and profitable ways of delivering more basic wireless services will still be needed.

The significance of EDGE for today's GSM operators is that it will increase data rates up to 384 kbps and potentially even higher in good quality radio environment, using current GSM spectrum and carrier structures more efficiently. EDGE will both complement and be an alternative to new W-CDMA coverage. EDGE will also have the effect of unifying the GSM, D-AMPS, and W-CDMA services through the use of dual-mode terminals.

12.3.1 High-Speed Circuit Switched Data (HSCSD) in GSM

HSCSD [3,4] is a feature that enables the coallocation of multiple full-rate traffic channels (TCH/F) of GSM into a HSCSD configuration. The aim of HSCSD is to provide a mixture of services with different air interface user rates by a single physical layer structure. The available capacity of a HSCSD configuration is several times the capacity of a TCH/F, leading to a significant enhancement in the air interface data transfer capability.

Ushering faster data rates into the mainstream is the new speed of 14.4 kbps per time slot and HSCSD protocols that approach wire-line access rates of up to 57.6 kbps by using multiple 14.4 kbps time slots. The increase from the current baseline 9.6 kbps to 14.4 kbps is due to a nominal reduction in the error-correction overhead of the GSM radio link protocol (RLP), allowing the use of a higher data rate. Implementation of v.4.2 bits compression could double the throughput.

For operators, migration to HSCSD brings data into the mainstream, enabled in many cases by relatively standard software upgrades to base station (BS) and mobile switching center (MSC) equipment. Flexible air interface resource allocation allows the network to dynamically assign resources related to the air interface usage according to network operator's strategy, and the end-user's request for a change in the air interface resource allocation based on data transfer needs. The provision of the asymmetric air interface connection allows simple mobile equipment (Type 1) to receive data at higher rates than would otherwise be possible with a symmetric connection.

For end-users, HSCSD enables the roll-out of mainstream high-end segment services that enable faster Web browsing, file downloads, mobile video-conference and navigation, vertical applications, telematics, and bandwidth-secure mobile LAN access. Value-added service providers (VASP) will also be able to offer guaranteed quality of service and cost-efficient mass-market applications, such as direct IP where users make circuit-switched data calls straight into a GSM network router connected to the Internet. To the end-user, the VASP or the operator is equivalent of an Internet service provider (ISP) that offers a fast secure dial-up IP service at cheaper mobile-to-mobile rates.

HSCSD is provided within the existing mobility management. Roaming is also possible. The throughput for an HSCSD connection remains constant for the duration of the call, except for interruption of transmission during handoff. The handoff is simultaneous for all time slots making up an HSCSD connection. End-users wanting to use HSCSD have to subscribe general bearer services. Supplementary services applicable to the general bearer services can be used simultaneously with HSCSD.

Firmware on most current GSM PC cards will have to be upgraded. The reduced RLP layer also means that a stronger signal strength will be necessary. Multiple time slot usage will probably only be efficiently available in off-peak times, increasing overall off-peak idle capacity usage.

12.3.2 General Packet Radio Service (GPRS) in GSM

The next phase in the high-speed road map will be the evolution of current short message services (SMS), such as smart messaging and unstructured supplementary service data (USSD), toward the new GPRS [5,6], a packet data service using TCP/IP and X.25 to offer speeds up to 115 kbps. GPRS has been standardized to optimally support a wide range of applications ranging from very frequent transmission of medium to large data volume and infrequent transmission of large data volume. Services of GPRS have been developed to reduce connection setup time and allow an optimum usage of radio resources. GPRS provides a packet data service for GSM where time slots on the air interface can be assigned to GPRS over which packet data from several mobile stations is multiplexed.

A similar evolution strategy, also adopting GPRS, has been developed for D-AMPS (IS-136). For operators planning to offer wideband multimedia services, the move to GPRS packet-based data bearer service is significant; it is a relatively small step compared with building a totally new 3G IMT-2000 network. Use of the GPRS network architecture for IS 136+ packet data service enables data subscription roaming with GSM networks around the globe that support GPRS and its evolution. The IS-136+ packet data service standard is known as GPRS-136. GPRS-136 provides the same capabilities as GSM GPRS. The user can access either X.25 or IP-based data networks.

GPRS provides a core network platform for current GSM operators not only to expand the wireless data market in preparation for the introduction of 3G services, but also a platform on which to build IMT-2000 frequencies should they acquire them.

GPRS enhances GSM data services significantly by providing end-to-end packet-switched data connections. This is particularly efficient in Internet/intranet traffic, where short bursts of intense data communications activity are interspersed with relatively long periods of inactivity. Because there is no real end-to-end connection to be established, setting up a GPRS call is almost instantaneous and users can be continuously online. Users have the additional benefit of paying for the actual data transmitted, rather than for connection time.

Because GPRS does not require any dedicated end-to-end connection, it only uses network resources and bandwidth when data is actually being transmitted. This means that a given amount of radio bandwidth can be shared efficiently and simultaneously among many users.

The implementation of GPRS has a limited impact on the GSM core network. It simply requires the addition of new packet data switching and gateway nodes, and an upgrade to existing nodes to provide a routing path for packet data between the wireless terminal and a gateway node. The gateway node provides interworking with external packet data networks for access to Internet, intranets, and databases.

A GPRS architecture for GSM is shown in Figure 12.7. GPRS will support all widely used data communications protocols, including IP, so it will be possible to connect with any data source from anywhere in the world using a GPRS mobile terminal. GPRS will support applications ranging from low-speed short messages to high-speed corporate LAN communications. However, one of the key benefits of GPRS—that it is connected through the existing GSM air interface modulation scheme—is also a limitation, restricting its potential for delivering data rates higher than 115 kbps. To build even higher rate data capabilities into GSM, a new modulation scheme is needed.

GPRS can be implemented in the existing GSM systems. It requires only minor changes in an existing GSM network. The base station subsystem (BSS) consists of base station controller (BSC) and packet control unit (PCU). The PCU supports all GPRS protocols for communication over the air interface. Its function is to set up, supervise, and disconnect packet-switched calls. PCU supports cell change, radio resource configuration, and channel assignment. The base transceiver station (BTS) is a relay station without protocol functions. It performs modulation and demodulation.

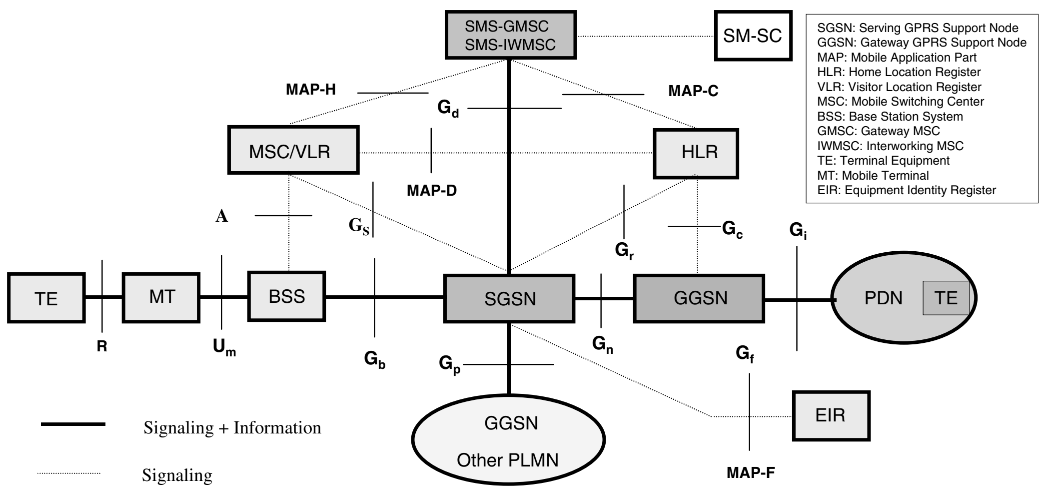

Figure 12.7 A GPRS Architecture in GSM

The GPRS standard introduces two new nodes, the serving GPRS support node (SGSN) and the gateway GPRS support node (GGSN). The home location register (HLR) is enhanced with GPRS subscriber data and routing information. Two types of services are provided by GPRS:

- Point-to-point (PTP)
- Point-to-multipoint (PTM)

Independent packet routing and transfer within the public land mobile network (PLMN) is supported by a new logical network node called the GPRS support node (GSN). The GGSN acts as a logical interface to external packet data networks. Within the GPRS networks, protocol data units (PDUs) are encapsulated at the originating GSN and decapsulated at the destination GSN. In between the GSNs, IP is used as the backbone to transfer PDUs. This whole process is referred to as *tunneling* in GPRS. The GGSN also maintains routing information used to tunnel the PDUs to the SGSN that is currently serving the mobile. All GPRS user-related data required by the SGSN to perform the routing and data transfer functionality is stored within the HLR. In GPRS, a user may have multiple data sessions in operation at one time. These sessions are called packet data protocol (PDP) contexts. The number of PDP contexts that are open for a user is limited only by the user's subscription and any operational constraints of the network. The main goal of the GPRS-136 architecture is to integrate IS-136 and GSM GPRS as much as possible with minimum changes to both technologies. In order to provide subscription roaming between GPRS-136 and GSM GPRS networks, a separate functional GSM GPRS HLR is incorporated into the architecture in addition to the IS-41 HLR.

The ETSI has specified GPRS as an overlay to the existing GSM network to provide packet data services. In order to operate a GPRS service over a GSM network, new functionality has been introduced into existing GSM network elements (NEs) and new NEs are integrated into the existing service provider GSM network.

The BSS of GSM is upgraded to support GPRS over the air interface. The BSS works with the GPRS backbone system (GBS) to provide GPRS service in a manner similar to its interaction with the switching subsystem for the circuit-switched services. The GBS manages the GPRS sessions set up between the mobile terminal and the network by providing functions such as admission control, mobility management (MM), and service management (SM). Subscriber and equipment information is shared between GPRS and the switched functions of GSM by the use of a common HLR and coordination of data between the visitor location register (VLR) and the GPRS support nodes of the GBS. The GBS is composed of two new NEs, the SGSN, and the GGSN.

The SGSN serves the mobile and performs security and access control functions. The SGSN is connected to BSS via frame-relay. The SGSN provides packet routing, mobility management, authentication, and ciphering to and from all GPRS subscribers located in the SGSN service area. A GPRS subscriber may be served by any SGSN in the network, depending on location. The traffic is routed from the SGSN to the BSC and to the mobile terminal via a BTS. At GPRS attach, the SGSN establishes a mobility management context containing information about mobility and security for the mobile. At packet data protocol (PDP) context activation, the SGSN establishes a PDP context which is used for routing purposes with the GGSN that GPRS subscriber uses. The SGSN may send in some cases location information to the MSC/VLR and receive paging requests.

The GGSN provides the gateway to external IP network, handling security and accounting functions as well as dynamic allocation of IP addresses. The GGSN contains routing information for the attached GPRS users. The routing information is used to tunnel PDUs to the mobile's current point of attachment, SGSN. The GGSN may be connected with the HLR via optional interface G_c. The GGSN is the first point of public data network (PDN) interconnection with a GSM PLMN supporting GPRS. From the external IP network's point of view, the GGSN is a host that owns all IP addresses of all subscribers served by the GPRS network.

The point-to-multipoint service center (PTM-SC) handles PTM traffic between the GPRS backbone and the HLR. The nodes will be connected by an IP backbone network. The SGSN and GGSN functions may be combined in the same physical node or separated, even residing in different mobile networks.

A special interface (G_s) is provided between MSC/VLR and SGSN to coordinate signaling for mobile terminals that can handle both circuit-switched and packet-switched data.

The HLR contains GPRS subscription data and routing information, and can be accessible from the SGSN. For the roaming mobiles, the HLR may reside in a different PLMN than the current SGSN. The HLR also maps each subscriber to one or more GGSNs.

The objective of the GPRS design is to maximize the use of existing GSM infrastructure while minimizing the changes required within GSM. The GSN contains most of the necessary capabilities to support packet transmission over GSM. The critical part in the GPRS network is the mobile-to-GSN (MS-SGSN) link, which includes the MS-BTS, BTS-

BSC, BSC-SGSN, and the SGSN-GGSN link. In particular, the U_m interface including the radio channel is the bottleneck of the GPRS network due to spectrum and channel speed/quality limitations. Since multiple traffic types of varying priorities will be supported by the GPRS network, quality of service criteria as well as resource management is required for performance evaluation.

The BSC will require new capabilities for controlling the packet channels, new hardware in the form of a PCU and new software for GPRS mobility management and paging. The BSC will also have a new traffic and signaling interface from SGSN.

The BTS will have new protocols supporting packet data for the air interface, together with new slot and channel resource allocation functions. The utilization of resources will be optimized through dynamic sharing between the two traffic types, handled by the BSC.

MS-SGSN Link. The logical link control (LLC) layer is responsible for providing a link between the mobile station (MS) and the SGSN. It governs the transport of GPRS signaling and traffic information from the MS to the SGSN. GPRS supports three service access points (SAPs) entities: the layer 3 management, subnet dependent convergence, and short message service (SMS). On the MS-BSS link, the radio link control (RLC), the media access control (MAC), and GSM RF protocols are supported (see Figure 12.8).

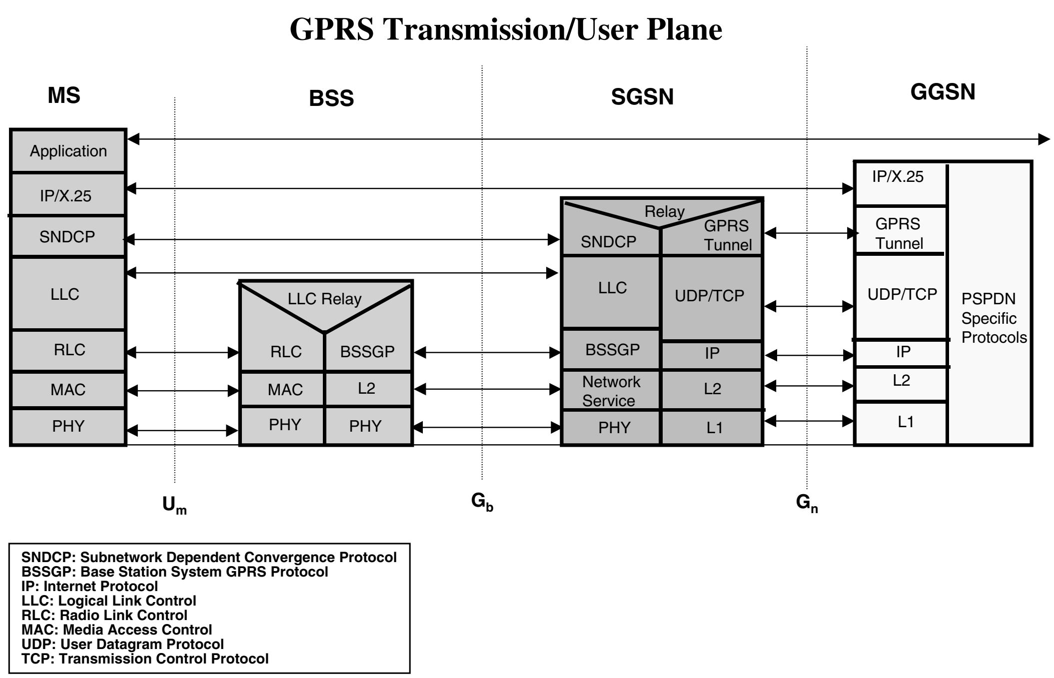

Figure 12.8 Protocol Stack in GPRS

The main drawback in implementing GPRS on an existing GSM infrastructure is that the GSM network is optimized for voice transmission (i.e., the GSM channel quality is designed for voice, which can tolerate errors at a predefined level). It is therefore expected that GPRS could have varied transmission performance in different network or coverage areas. To overcome this problem, GPRS supports multiple coding rates at the physical layer.

GPRS could share radio resources with GSM circuit-switched (CS) service. This is governed by a dynamic resource sharing based on the capacity on demand criteria. GPRS channel is allocated only if an active GPRS terminal exists in the network. Once resources are allocated to GPRS, at least one channel will serve as the *master* channel to carry all necessary signaling and control information for the operation of GPRS. All other channels will serve as *slave channels* and are only used to carry user and signaling information. If no master channel exists, all GPRS users will use the GSM common control channel (CCCH) and inform the network to allocate GPRS resources.

A physical channel dedicated to GPRS is called a packet data channel (PDCH). It is mapped into one of the physical channels allocated to GPRS. A PDCH can be used either as a packet common control channel (PCCCH) (see Figure 12.9), a packet broadcast control channel (PBCCH), or a packet traffic channel (PTCH).

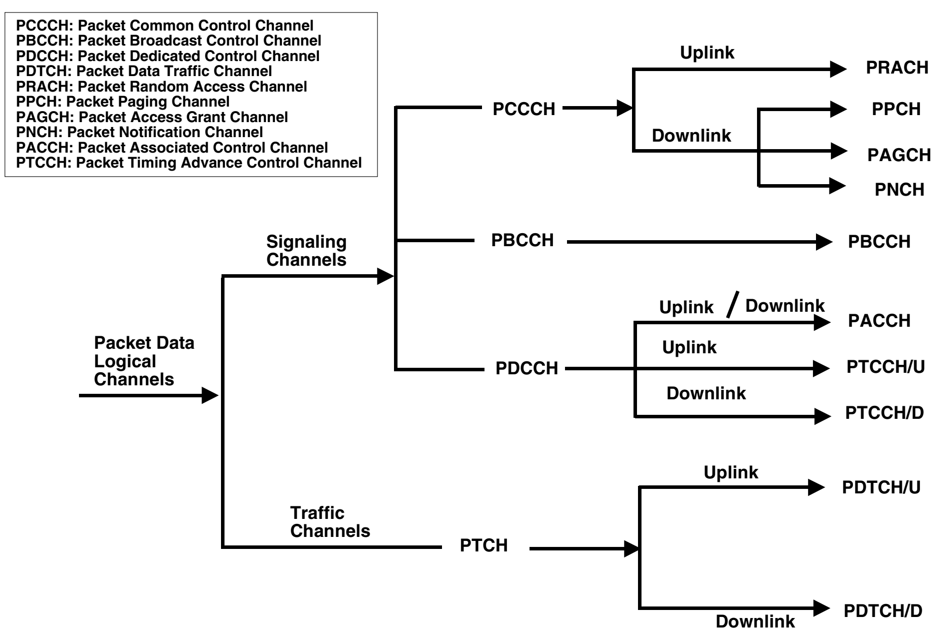

Figure 12.9 GPRS Logical Channels

The PCCCH consists of the following (see Figure 12.9):

- Packet random access channel (PRACH)—uplink
- Packet access grant channel (PAGCH)—downlink
- Packet notification channel (PNCH)— downlink
- On the other hand, the PTCH can either be:
  - Packet data traffic channel (PDTCH)
  - Packet associated control channel (PACCH)

For a given traffic characteristic, GPRS logical channels require a combination of PCCCH and PTCH. Fundamental questions such as how many PDTCHs can be supported by a single PCCCH are required in the dimensioning of GPRS.

RLC/MAC Layer. The multiframe structure of the packet data channel (PDCH) in which GPRS RLC messages are transmitted is composed of 52 TDMA frames organized into RLC blocks of four bursts resulting into 12 blocks per multiframe plus four idle frames located in the 13th, 26th, 39th, and 52nd position (see Figure 12.10).

B_0 consists of frames 1, 2, 3, and 4; B_1 consists of frames 5, 6, 7, 8, and so on. It is important that the mapping of logical channels onto the radio blocks is by means of an ordered set of blocks (B_0, B_6, B_9, B_1, B_7, B_4, B_{10}, B_2, B_8, B_5, B_{11}, B_3). The advantage of ordering the blocks is mainly to spread the locations of the control channels in each time slot reducing, the average waiting time for the users to transmit signaling packets. Secondly, it provides an interleaving of the GPRS multiframe.

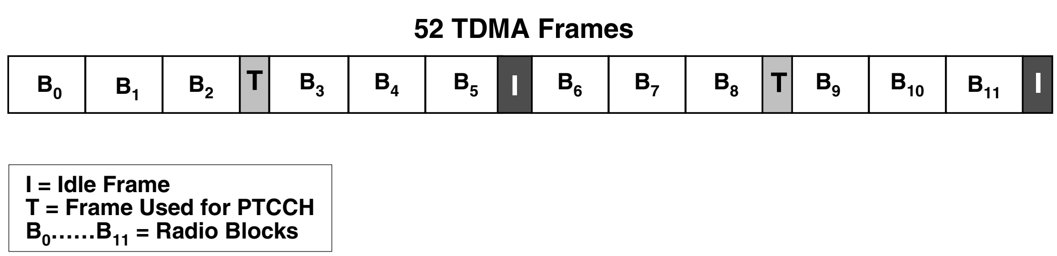

Figure 12.10 Mapping of Logical Channels to Physical Channels

GPRS uses a reservation protocol at the media access control layer. Users that have packets ready to send request a channel via the PRACHs. The random access burst consists of only one TDMA frame of duration enough to transmit an 11-bit signaling message. Only the PDCHs carrying PCCCHs contain PRACHs. The blocks used as PRACHs are indicated by an uplink state flag [uplink state flag (USF) = free] by the downlink pair channel.

Alternatively, the first K blocks following the ordered set of blocks can be assigned to PRACH permanently. The access burst is transmitted in one of the four bursts assigned as PRACH. Any packet channel request is returned by a packet immediate assignment on the PRACHs whose locations are broadcasted by PBCCH. Optionally, a packet resource request for additional channels is initiated and returned by a packet resource assignment. The persistence of random access is maintained by the traffic load and user class with a back-off algorithm for unsuccessful attempts. In the channel assignment, one or more PTCHs (time slot) will be allocated to a particular user. A user reserves a specific number of blocks on the assigned PTCH as indicated by the uplink state flag (USF). It is possible to accommodate more than one user per PTCH. User signaling is also transmitted on the same PTCH using the PAGCH, whose usage depends on the user's needs.

The performance of the media access control layer depends on the logical arrangement of the GPRS channels (e.g., allocation of random access channels, access grant channels, broadcast channels) for a given set of traffic statistics. This is determined by the amount of resources allocated for control and signaling as compared with the data traffic. A degree of flexibility is also achieved with logical channels as the traffic varies. The arrangement of logical channels is determined through the PBCCH.

LLC Layer. The LLC layer is responsible for providing a reliable link between the mobile and the SGSN. It is based on the high-level data link control (HDLC) and link access procedure on the D-channel (link access procedure on the D-channel [LAPD]) protocols. It is designed to support variable length transmission in a point-to-point or multipoint topology. It includes layer functions such as sequence control, flow control, error detection, ciphering, and recovery, as well as the provision of one or more logical link connections between two layer 3 entities. A logical link is identified by a data link control identifier (DLCI), which consists of a service access point identifier (SAPI) and terminal equipment identity (TEI) mapped on the LLC frame format. Depending on the status of the logical link, it supports an unacknowledged or an acknowledged information transfer. The former does not support error recovery mechanisms. The acknowledged information transfer supports error and flow control. This operation only applies to point-to-point operations. The LLC frame consists of an address field (1 or 5 octets), control field (2 or 6 octets), a length indicator field (2 octets maximum), information fields (1,500 octets maximum), and frame check sequence of 3 octets. Four types of control field formats are allowed; these include the supervisory functions (S format), the control functions (U), and acknowledged and unacknowledged information transfer (I and UI).

In the performance evaluation, the objective is to determine delay during the exchange of commands and responses involved in various operations supported by the LLC in relation to the transfer of an LLC PDU. The LLC commands and responses are exchanged between two layer 3 entities in conjunction with a service primitive invoke by the mobile or the SGSN.

Data Packet Routing in GPRS Network. Here, we discuss data packet routing for the mobile-originated and mobile-terminated data call scenarios [7]. In the case of mobile-originated data routing, the mobile gets an IP packet from an application and requests a channel reservation. The mobile transmits data in the reserved time slots. The packet-switched public data network (PSPDN) PDU is encapsulated into a subnetwork dependent convergence (SNDC) protocol unit that is sent via the LLC protocol over the air interface to the SGSN currently serving the mobile (see Figure 12.11).

For mobile-terminated data routing, we have two cases: routing to a home GPRS network, and routing to a visited GPRS network. In the first case, a user sends a data packet to a mobile. The packet goes through the local area network (LAN) via a router out on the GPRS context for the mobile. If the mobile is in GPRS idle state, the packet is rejected. If the mobile is in standby or active mode, the GGSN routes the packet in an encapsulated format to SGSN.

In the second case, the home GPRS network sends the data packet over the interoperator backbone network to the visiting GPRS network. The visiting GPRS network routes the packet to the appropriate SGSN (see Figure 12.11).

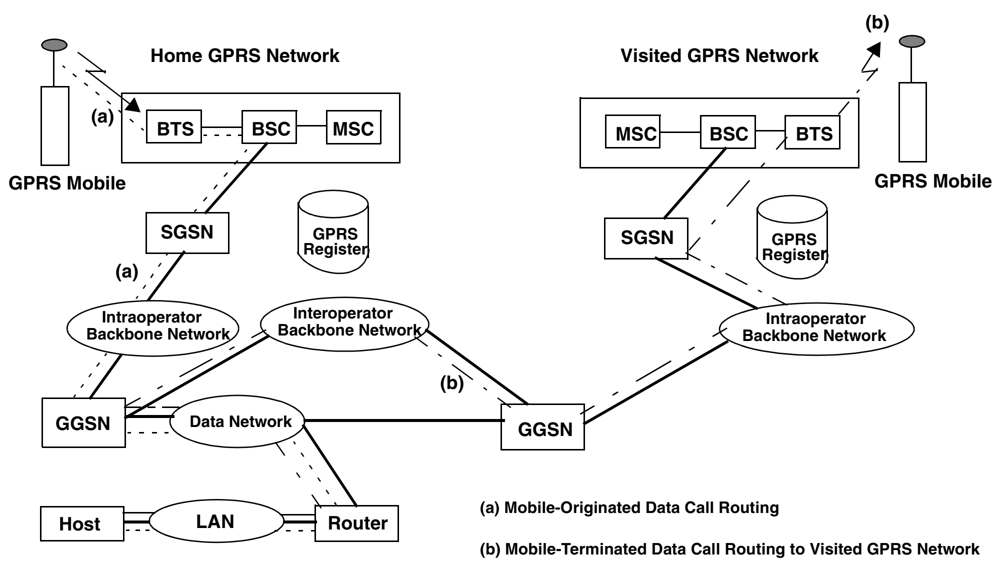

Figure 12.11 Data Call Routing in GPRS Network

Point-to-point and point-to-multipoint applications of GPRS are as follows:

- Point-to-point
  - messaging (e.g., e-mail)
  - remote access to corporate networks
  - access to the Internet
  - credit card validation (point-of-sale)
  - utility meter readings

  - road toll applications
  - automatic train control
- Point-to-multipoint
  - PTM multicast (send to all)
  - news
  - traffic information
  - weather forecasts
  - financial updates
  - PTM group call (send to some)
  - taxi fleet management
  - conferencing

GPRS will provide a service for bursty and bulky data transfer; radio resources on demand; shared use of physical radio resources; existing GSM functionality; mobile applications for the mass application market; volume-dependent charging; and integrated services, operation, and management.

12.3.3 Enhanced Data Rates for GSM Evolution (EDGE)

EDGE provides an evolutionary path that enables existing 2G systems (GSM, IS-136) to deliver 3G services in existing spectrum bands. The advantages of EDGE include fast availability, reuse of existing GSM, IS-136, and PDC infrastructure, as well as support for gradual introduction of 3G capabilities.

EDGE reuses the GSM carrier bandwidth and time slot structure. EDGE can be seen as a generic air interface for efficiently providing high bit rates, facilitating an evolution of existing 2G systems toward 3G systems.

EDGE (2.5G system) [7,8] was designed to enhance user bandwidth through GPRS. This is achieved through the use of higher-level modulation schemes. Although EDGE reuses the GSM carrier bandwidth and time slot structure, the technique is by no means restricted to GSM systems; it can be used as a generic air interface for efficient provision of higher bit rates in other TDMA systems. In the Universal Wireless Communications Consortium (UWCC), the 136 high-speed (136 HS) radio interface was proposed as a means of satisfying the requirements for an IMT-2000 RTT. EDGE was adopted by UWCC in 1998 as the outdoor component of 136 HS to provide 384-kbps data service.

The standardization effort for EDGE has two phases. In the first phase the emphasis has been placed on enhanced GPRS (EGPRS) and enhanced CSD (ECSD). The second phase is being defined with improvements for multimedia and real-time services as possible work items.

EDGE is primarily a radio interface improvement, but it can also be viewed as a system concept that allows GSM and IS-136 networks to offer a set of new services. EDGE has been designed to improve S/I by using link quality control. *Link quality control* adapts the protection of the data to the channel quality so that an optimal bit rate is achieved for all channel qualities.

The EDGE air interface is designed to facilitate higher bit rates than those currently achievable in existing 2G systems. The modulation scheme based on 8-PSK is used to increase the gross bit rate. GMSK modulation as defined in GSM is also part of the EDGE system. The symbol rate is 271 kbps for both GMSK and 8-PSK, leading to gross bit rates per time slot of 22.8 kbps and 69.2 kbps, respectively. The 8-PSK pulse shape is linearized GMSK to allow 8-PSK to fit into the GSM spectrum mask. The 8-PSK burst format is similar to GSM (see Figure 12.12).

3	58	26	58	3	8.25

$\longleftarrow$ 0.577 ms $\longrightarrow$

Figure 12.12 Burst Format for EDGE

In order to achieve a higher gross rate, a new modulation scheme, quaternary offset quadrature amplitude modulation (QOQAM), has been proposed for EDGE, since it can provide higher data rates and good spectral efficiency. An offset modulation scheme is proposed because it gives smaller amplitude variation than 16-QAM, which can be beneficial when using nonlinear amplifiers. EDGE will coexist with GSM in the existing frequency plan and will provide *link adaptation* (i.e., modulation and coding are adapted for channel conditions).

12.3.4 Radio Protocol Design

The radio protocol strategy in EDGE is to reuse the protocols of GSM/GPRS whenever possible, thus minimizing the need for new protocol implementation. EDGE enhances both GSM circuit-switched (HSCSD) and packet-switched (GPRS) mode operation. EDGE includes one packet-switched (PS) and one circuit-switched (CS) mode, EGPRS and ECSD, respectively.

Enhanced GPRS (EGPRS). The EDGE radio link control (RLC) protocol is somewhat different from the corresponding GPRS protocol. The main changes are related to improvements in the link quality control scheme.

A link adaptation scheme regularly estimates the link quality and subsequently selects the most appropriate modulation and coding scheme for transmission to maximize the user bit rate. The link adaptation scheme offers mechanisms for choosing the best modulation and coding alternative for the radio link. In GPRS, only the coding schemes can be changed between two consecutive link layer control (LLC) frames. In the EGPRS, even the modulation can be changed. Different coding and modulation schemes enable adjustment for the robustness of the transmission according to the environment.

Another way to handle link quality variations is *incremental redundancy*. In this scheme, information is first sent with very little coding, yielding a high bit rate if decoding is immediately successful. If decoding is not successful, additional coded bits (redundancy) are sent until decoding succeeds. The more coding that has to be sent, the lower the resulting bit rate and the higher the delay.

EGPRS will support a combined link adaptation and incremental redundancy schemes. In this case, the initial code rate of the incremental redundancy scheme is based on measurements of the link quality. Benefits of this approach are the robustness and high throughput of the incremental redundancy operation in combination with lower delays and lower memory requirements enabled by the adaptive initial code rate.

In EGPRS, the different initial code rates are obtained by puncturing a different number of bits from a common convolutional code ($r = 1/3$). The resulting coding schemes are given in Table 12.8. Incremental redundancy operation is enabled by puncturing a different set of bits each time a block is retransmitted, whereby the code rate is gradually decreased toward 1/3 for every new transmission of the block. The selection of the initial modulation and code rate is based on regular measurements of link quality.

Actual performance of modulation and the coding scheme together with channel characteristics form the basis for link adaptation. Channel characteristics are needed to estimate the effects of a switch to another modulation and coding combination; these include an estimated S/I ratio, but also time dispersion and fading characteristics (that affect the efficiency of interleaving).

In the case of GSM, EDGE with the existing GSM radio bands will offer wireless multimedia IP-based applications at the rate of 384 kbps with a bit rate of 48 kbps per time slot, and up to 69.2 kbps per time slot under good radio conditions.

EGPRS offers eight additional coding schemes. EGPRS users will have eight modulation and coding schemes available, compared with four for GPRS. Besides changes in the physical layer, modifications in the protocol structure are also needed. The lower layers of the user data plane designed for GPRS are the physical, radio link control (RLC)/media access control (MAC), and link layer control (LLC) layers. With EDGE functionality, the LLC layer will not require any modifications; however, the RLC/MAC layer has to be modified to accommodate features for efficient multiplexing and link adaptation procedures to support the essentially new physical layers in the EDGE.

Enhanced CSD (ECSD). In this case, the objective is to keep the existing GSM CS data protocols as intact as possible. In order to provide higher data rates, multislot solutions as found in ECSD are provided in EDGE. This has no impact on link or system performance.

A data frame is interleaved over 22 frames as in GSM, and three new 8-PSK channel coding schemes are defined along with the four already existing for GSM. The radio interface rate varies from 3.6 to 38.8 kbps per time slot (see Table 12.9).

Fast introduction of EGPRS/ECSD services is possible by reusing the existing transcoder rate adaptation unit (TRAU) formats and 16 kbps channel structure on the A-bis interface. Since data above 14.4 kbps cannot be rate adapted to fit into one 14.4-kbps TRAU frame, TRAU frames on several 16 kbps channels will be used to meet the increased capacity requirement. In this case, a BTS is required to handle a higher number of 16-kbps A-bis channels than time slots used on the radio interface. The benefit of using the current TRAU formats is that the introduction of new channel coding does not have any impact on the A-bis transmission, but it makes possible to hide the new coding from the TRAU unit. On the other hand, some additional complexity is introduced in the BTS owing to modified data frame handling.

Instead of reusing the current A-bis transmission formats for EDGE, new TRAU formats and rate adaptation optimized for increased capacity can be specified. The physical layer can be dimensioned statically for the maximum user rate specified for particular EDGE service or more dynamic reservation of A-bis transmission resources can be applied. The A-bis resources can even be released and reserved dynamically during the call, if the link adaptation is applied.

The channel coding schemes defined for EDGE in PS transmission are listed in Table 12.8. The schemes for EDGE in CS transmission are listed in Table 12.9.

Table 12.8 Channel Coding Scheme in EDGE (PS Transmission)

Coding Scheme	Gross Bit Rate (kbps)	Code Rate	Modulation	Radio Interface Rate per Time Slot (kbps)	Radio Interface Rate on 8 Time Slots (kbps)
CS-1	22.8	0.49	GMSK	11.2	89.6
CS-2	22.8	0.63	GMSK	14.5	116.0
CS-3	22.8	0.73	GMSK	16.7	133.6
CS-4	22.8	1.0	GMSK	22.8	182.4
PCS-1	69.2	0.329	8-PSK	22.8	182.4
PCS-2	69.2	0.496	8-PSK	34.3	274.4
PCS-3	69.2	0.596	8-PSK	41.25	330.0
PCS-4	69.2	0.746	8-PSK	51.60	412.8
PCS-5	69.2	0.829	8-PSK	57.35	458.8
PCS-6	69.2	1.000	8-PSK	69.20	553.6

Table 12.9 Channel Coding Scheme in EDGE (CS Transmission)

Channel Name	Code Rate	Modulation	Radio Interface Rate per Time Slot (kbps)
TCH/F2.4	0.16	GMSK	3.6
TCH/F4.8	0.26	GMSK	6.0
TCH/F9.6	0.53	GMSK	12.0
TCH/F14.4	0.64	GMSK	14.5
ECSD TCS-1 (NT +T)	0.42	8-PSK	29.0
ECSD TCS-2 (T)	0.46	8-PSK	32.0
ECSD TCS-3 (NT)	0.56	8-PSK	38.8

12.3.5 Services Offered by EDGE

PS Services. The GPRS architecture provides IP connectivity from mobile station to an external fixed IP network. For each service, a quality of service (QoS) profile is defined. The QoS parameters include priority, reliability, delay, and maximum and mean bit rate. A specified combination of these parameters defines a service, and different services can be selected to suit the needs of different applications.

CS Services. The current GSM standard supports both transparent and nontransparent services. Eight transparent services are defined, offering constant bit rates in the range of 9.6 to 64 kbps.

A nontransparent service uses radio link protocol (RLP) to ensure virtually error-free data delivery. For this case, there are eight services offering maximum user bit rates from 4.8 to 57.6 kbps. The actual user bit rate may vary according to channel quality and the resulting rate of transmission.

The introduction of EDGE implies no change of service definitions. The bit rates are the same, but the way services are realized in terms of channel coding is different. For example, a 57.6 kbps nontransparent service can be realized with coding scheme ECSD TCS-1 and two time slots, while the same service requires four time slots with standard GSM using coding scheme TCH/F14.4.

Thus, EDGE CS transmission makes the high-bit-rate services available with fewer time slots, which is advantageous from a terminal implementation perspective. Additionally, more users can be accepted since each user needs fewer time slots, which increases the capacity of the system.

Asymmetric Services Due to Terminal Implementation. ETSI has standardized two mobile classes: one that requires only GMSK transmission in uplink and 8-PSK in the downlink, and one that requires 8-PSK in both links. For the first class, the uplink bit rate will be limited to that of GSM/GPRS, while the EDGE bit rate is still provided in the downlink. Since most services are expected to require higher bit rates in the downlink than in the uplink, this is a way of providing attractive services with a low complexity mobile station. Similarly, the number of time slots available in uplink and downlink need not be the same. However, transparent services will be symmetrical.

12.3.6 EDGE Implementation

EDGE makes use of the existing GSM infrastructure in a highly efficient manner: radio network planning will not be greatly affected, since it will be possible to reuse many existing BTS sites. GPRS packet-switching nodes will be unaffected, because they function independently of the user bit rates. Any modifications to the switching nodes will be limited to software upgrades. There is also a smooth evolutionary path defined for terminals to ensure that EDGE-capable terminals will be small and competitively priced.

EDGE-capable channels will be equally suitable for standard GSM services, and no special EDGE, GPRS, or GSM services will be needed. From an operator viewpoint this allows seamless introduction of new EDGE services—perhaps starting with the deployment of EDGE in the service hot spots and gradually expanding coverage as demand dictates. The roll-out of EDGE-capable BSS hardware can become part of the ordinary expansion and capacity enhancement of the network. The wideband data capabilities offered by EDGE will allow a step-by-step evolution to IMT-2000, probably through a staged deployment of the new 3G air interface on the existing core GSM network. Keeping GSM as the core network for the provision of 3G wireless services has additional commercial benefits. It protects the investment of existing operators, it helps to ensure the widest possible customer base from the outset, and it fosters supplier competition through the continuous evolution of systems.

GSM operators who win licences in new 2-GHz bands will be able to introduce IMT-2000 wideband coverage in areas where early demand is likely to be greatest. Dual-mode EDGE/IMT-2000 mobile terminals will allow full roaming and handoff from one system to the other, with mapping of services between the two systems. EDGE will contribute to the commercial success of 3G system in the vital early phases by ensuring that IMT-2000 subscribers will be able to enjoy roaming and interworking globally.

Compared with establishing a total 3G system, building on an existing GSM infrastructure will be relatively fast and inexpensive. The intermediate move to GPRS and later to EDGE will make the transition to 3G easier.

While GPRS and EDGE will require new functionality in the GSM network, with new types of connections to external packet data networks, they are essentially extensions of GSM. Moving to a GSM/IMT-2000 core network will likewise be a further extension of this network.

EDGE provides GSM operators—whether or not they get a new 3G licence—with a commercially attractive solution to develop the market for wideband multimedia services. Familiar interfaces such as the Internet, volume-based charging, and a progressive increase in available user data rates will remove some of the barriers to large-scale application of wireless data services. The way forward to 3G services will be a staged evolution from today's GSM data services through GPRS and EDGE.

Increased user data rates over the radio interface will require redesign of the physical transmission methods, frame formats, and signaling protocols in different network interfaces. The extent of modification needed will depend on the user data rate requirement, i.e., whether the support of higher data is required or merely a more efficient usage of the radio time slot to support current data services is needed.

Several alternatives to cover the increased radio interface data rates on the A-bis interface for EGPRS and ECSD can be envisioned. The existing physical structure can be reused as much as possible or new transmission method optimized for EDGE can be specified.

Table 12.10 provides a comparison of GSM data services.

Table 12.10 Comparison of GSM Data Services

Service Type	Data Unit	Max. Sustained User Data Rate	Technology	Resources Used
Short message service (SMS)	Single 140 octet packet	9 bps	Simplex circuit	SDCCH or SACCH
Circuit-switched data	30 octet frames	9,600 bps	Duplex circuits	TCH
HSCSD	192 octet frames	115 kbps	Duplex circuits	1–8 TCH
GPRS	1,600 octet frames	171 kbps	Virtual circuit/ packet switching	PDCH (1–8 TCH)
EDGE		384 kbps	Virtual circuit/ packet switching	1–8 TCH

Note: SDCCH: stand-alone dedicated control channel; SACCH: slow associated control channel; TCH: traffic channel; PDCH: packet data channel (all refer to GSM logical channels).

12.4 Upgrade to UMTS (W-CDMA) in the Core GSM

A primary assumption for UMTS is that it will be based on an evolved GSM core network [7–11]. This will provide backward compatibility with GSM in terms of network protocols and interfaces (MAP, ISUP, etc.). The core network will support both GSM and UMTS/IMT-2000 services, including handover and roaming between the two (see Figure 12.13) [12]. The proposed W-CDMA-based UMTS Terrestrial Radio Access Network (UTRAN) will be connected to the GSM-UMTS core network using a new multivendor interface (I_u). The transport protocol within the new radio network and to the core network will be ATM.

There will be a clear separation between the services provided by UTRAN and the actual channels used to carry these services. All radio network functions (such as resource control) will be handled within the radio access network, and clearly separated from the service and subscription functions in the UMTS core network (UCN). The GSM-UMTS network, shown in Figure 12.14, will consist of three main parts:

- GSM-UMTS core network
- UMTS terrestrial radio access network (UTRAN)
- GSM base station subsystem (BSS)

Like the GSM-GPRS core network, the GSM-UMTS core network will have two different parts: a circuit-switched MSC and a packet-switched GRPS support node (GSN). The core network access point for GSM circuit-switched connections is the GSM MSC, and for packet-switched connection it is the SGSN.

GSM-defined services (up to and including GSM Phase 2+) will be supported in the usual GSM manner. The GSM-UMTS core network will implement supplementary services according to GSM principles (HLR-MSC/VLR). New services beyond Phase 2+ will be created using new service capabilities. These service capabilities may be seen as building blocks for application development. These include:

- Bearers defined by QoS
- Mobile station execution environment (MExE)
- Telephony value-added services (TeleVAS)
- Subscriber identity module (SIM) Toolkit
- Location services
- Open interfaces (APIs) to mobile network functions
- Downloadable application software
- Intelligent network/customized applications for mobile enhanced logic (IN/CAMEL) and service nodes

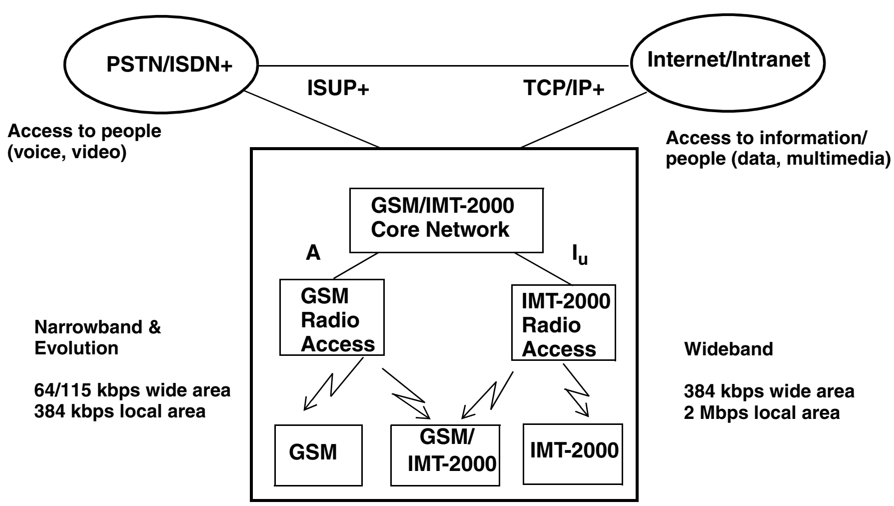

Figure 12.13 Evolution to UMTS/IMTS-2000 in a GSM Environment

In addition to new services provided by the GSM-UMTS network itself, many new services and applications will be realized using a client/server approach, with the server residing on service LANs outside the GSM-UMTS core network (see Figure 12.14). For such services, the core network will simply act as a transparent bearer. This approach is in line with current standardization activities, and will be important from a service continuity point of view. The core network will ultimately be used for the transfer of data between the end points, the client, and the server.

Intelligent network (IN) techniques are one way to provide seamless interworking across GSM-UMTS network. CAMEL already provides the basis for GSM/IN interworking. The IN infrastructure may be shared by fixed and mobile networks, and can support fixed/mobile service integration, as needed by IMT-2000. The inherent support for third-party service providers in IN means such providers could offer all or part of the integrated services. This role of IN is already apparent in services such as virtual private networks (VPN), regional subscriptions, and One Number, which are available as network-independent and customer-driven services.

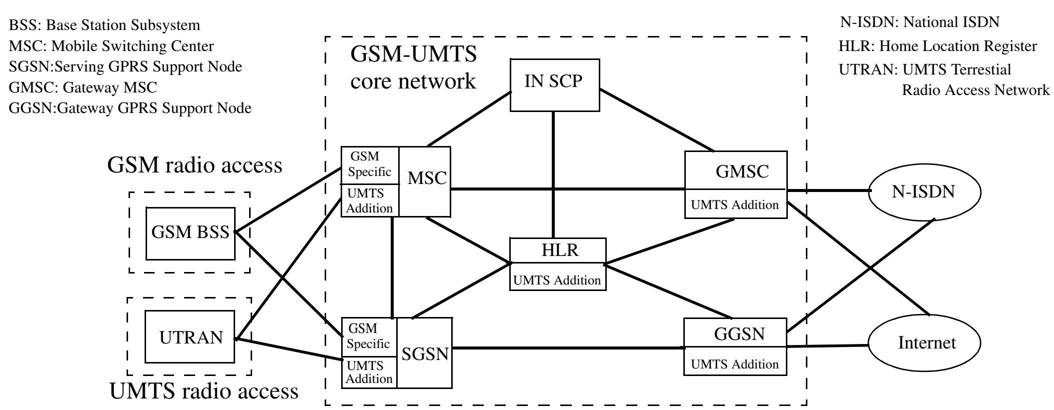

Figure 12.14 General GSM-UMTS Network Architecture

Service nodes and IN can play a complementary role. IN is suitable for subscription control and group services where high service penetration in a very wide area with frequent service invocation is more important than sophistication. Service nodes are better for providing differentiated user interfaces, e.g., personal call and messaging services that use advanced in-band processing and span several access networks.

To make the most of the new radio access network's capabilities and to cater to the large increase in data traffic volume, it is likely that ATM will be used as the transport protocol within UTRAN and toward the GSM-UMTS core network. The combination of an ATM cell-based transport network, W-CDMA's use of variable-rate speech coding with improved channel coding, and an increased volume of packet data traffic over the air interface will mean a saving of about 50 percent in transmission costs, compared with equivalent current solutions. ATM, with the newly standardized AAL2 adaptation layer, provides an efficient transport protocol, optimized for delay-sensitive speech services and packet data services. Statistical multiplexing in ATM provides maximum utilization of existing and new transmission infrastructure throughout the entire network.

In the complex multiservice, multivendor, multiprovider environment of 3G wireless services, network management will be a critical issue. The growth of packet data traffic will require new ways of charging for services and new billing systems to support them. There will continue to be a growing demand for better customer care and cost reductions in managing mobile networks, driven by the need to

- Provide sophisticated personal communications services
- Expand the customer base beyond the business user base
- Separate the service provider and network operator roles
- Provide "one-stop" billing for a range of services

New operations and management functions will be needed to support new services and network functionality. Standardization of interfaces will be critical, especially for alignment with current management interfaces in the GSM-UMTS core network. Management information will need to be part of standard traffic interfaces.

With the right service strategy and network planning, GSM operators will be able to capitalize on the wideband multimedia market through a staged evolution of their core networks with the addition of new radio access technology as it becomes available.

12.5 Summary

In this chapter, we examined the evolution of TDMA-based 2G networks to 3G networks to provide multimedia services up to 2 Mbps in the local area and up to 384 kbps in the wide area with a UMTS W-CDMA air interface. We discussed GPRS, the new packet-based data bearer service for GSM and IS-136, and building a core network capable of delivering GPRS service to meet the requirements of IMT-2000. We also discussed the use of EDGE (a 2.5-G system) to offer wireless multimedia IP-based applications of speeds up to 384 kbps. We concluded the chapter by outlining the evolutionary path of a GSM core network to UMTS to provide backward compatibility in terms of network protocol and interfaces, and to support both GSM and UMTS/IMT-2000 services with handoff and roaming between two systems.

12.6 References

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