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Enhancements in Second Generation Systems

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26.1 Introduction

Present digital **cellular** and **cordless** systems were optimized for voice services. At the very best, they can provide low and medium bit rate information services. The development of enhanced versions of these radio interfaces is motivated by the following:

- Provision of additional services, including high bit rate circuit-switched and packet-switched services that meet the short-term needs of mobile multimedia services.
- Improvements in the radio coverage of existing cellular systems. An extension to allow cordless coverage in the home is also being considered.
- Even more efficient utilization of the available frequency spectrum, which is a valuable but limited resource.

This chapter deals mainly with the air interface of second generation systems. The subject of enhancement in system performance is addressed from two directions. On the one hand, system features like adaptive multirate coders, packet transmission, which have been explicitly included in the standard, are discussed in detail. On the other hand, methods of equipment design or network design possible with the new or already existing features of the air interface are presented.

26.2 Overview of Second Generation Systems

Initially the need of a pan-European system to replace a large variety of disparate analog cellular systems was the major motivating factor behind the creation of the Global System for Mobile communications (GSM). In North America and Japan, where unique analog systems existed, the need to standardize respectively IS-54, IS-95, and Personal Digital Cellular (PDC) for digital cellular applications arose from the lack of spectrum to serve the high traffic density areas [3]. Additionally, some of the second generation systems like Digital European Cordless Telecommunications (DECT) and Personal Handy Phone Systems (PHS) are the result of a need to offer wireless services in residential and office environments with low cost subscriber equipment [16].

The physical layer characteristics of all these systems offer robust radio links paired with good spectral efficiency. The network related functionalities have been designed to offer secure communication to authenticated users even when roaming between various networks based on the same system. Table 26.1 provides the essential characteristics of the second generation systems as initially designed in the late eighties and early nineties. Since then several additions have been made to those standards. Enhancements of air interface as well as network subsystem functionalities have been incorporated.

26.3 Capacity Enhancement

The **capacity** of a mobile network can be defined as the Erlangs throughput by a cell, a cluster of cells, or by a portion of a network. For a given radio interface, the achievable capacity is a function of the robustness of the physical layer, the effectiveness of the medium access control (MAC) layer and the multiple access technique. Moreover, it is strongly dependent on the radio spectrum available for network planning.
If we define:

BW	Available radio spectrum
Wc	Spectrum occupied by a single radio carrier
Nc	Number of circuits handled by a single carrier (e.g., number of time slots per frame in a TDMA system)
Cs	Permissible cluster size which guarantees a good quality for a vast majority of active calls (like 90 to 95%)

The number of circuits per cell is given by (BW / Wc) $\times$ (Nc / Cs).

The Erlang capacity can be derived from this expression after consideration of the signalling overhead required by the pilot channel, signalling and traffic channel overheads for handover, and the specified call blocking rate.

This definition of capacity is applicable in case of TDMA/FDMA air interfaces and when the networks are designed using "deterministic frequency allocation." A slightly different approach to the problem is necessary in systems like DECT which use dynamic channel selection and for DS-CDMA systems. IS-95 networks pretend to use a cluster size (Cs) of one. However, the number of good quality circuits in a cell is a function of the available noninterfered spreading codes.
Capacity enhancement can be obtained through:

TABLE 26.1 Air Interface Characteristics of Second Generation Systems

	CELLULAR				CORDLESS	
Standard	GSM	IS-54	IS-95	PDC	DECT	PHS
Frequency band (MHz)	Europe	USA	USA	Japan	Europe	Japan
Uplink	890–915 (1710–1785)	824–849	824–849	940–956 (1429–1441)	1880–1900	1895–1907
Downlink	935–960 (1805–1880)	869–894	869–894	810–826 (1477–1489)		
Duplex spacing (MHz)	45 (95)	45	45	130 (48)	—	—
Carrier spacing (kHz)	200	30	1250	25	1728	300
Number of radio channels in the frequency band	124 (374)	832	20	640 (480)	10	77
Multiple access	TDMA	TDMA	CDMA	TDMA	TDMA	TDMA
Duplex mode	FDD	FDD	FDD	FDD	TDD	TDD
Number of channels per carrier	8 (half rate: 16)	3 (half rate: 6)	128	3 (half rate: 6)	12	4
Modulation	GMSK	$\Pi/4$ DQPSK	QPSK BPSK	$\Pi/4$ DQPSK	GFSK	$\Pi/4$ DQPSK
Carrier bit rate (kb/s)	270.8	48.6	1288	42	1152	384
Speech coder (full rate)	RPE-LTP	VSELP	QCELP	VSELP	ADPCM	ADPCM
Net bit rate (kb/s)	13	7.95	(var.rate: 8, 4, 2, 0.8)	6.7	32	32
Channel coder for speech channels	1/2 rate convolutional + CRC	1/2 rate convolutional + CRC	1/2 (downlink), 1/3 (uplink) convolutional + CRC	1/2 rate convolutional + CRC	no	no
Gross bit rate speech+channel coding (kb/s)	22.8	13	var. rate 19.2, 9.6, 4.8, 2.4	11.2	—	—
Frame size (ms)	4.6	40	20	20	10	5
MS transmission power (W)	Peak / Aver. 8 / 1 2 / 0.25 (1) / (0.125)	Peak / Aver. 9 / 3 4.8 / 1.6 1.8 / 0.6	0.6	Peak / Aver. 2 / 0.66	Peak / Aver. 0.25 / 0.01	Peak / Aver. 0.08 / 0.01
Power control MS	Y	Y	Y	Y	N	Y
Power control BS	Y	Y	Y	Y	N	Y
Operational C/I(dB)	9	16	6	17	21	26
Equalizer	needed	needed	Rake receiver	option	option	no
Handover	Y	Y	Soft handoff	Y	Y	Y

- An increase in number of radio carriers and/or the number of traffic channels (e.g., voice circuits)
- Improved interference management techniques
- Novel cellular configurations.

26.3.1 Capacity Enhancement Through Increase in Number of Carriers and/or Voice Circuits

This family of relatively simple solutions can be subdivided in three categories related to the availability of spectrum in the same band, or in a different band where the same radio interface can be utilized or the situation in which the number of voice circuits can be increased by the introduction of multiple speech codecs.

Increased Spectrum Availability

In such a fortunate situation, extra carriers can be added to the already installed base stations (BS). The existing Cs is maintained and the increase in capacity is according to the added carriers/circuits. An additional gain in Erlang capacity is available due to increased trunking efficiency.

Supplementary Spectrum Availability (in Different Frequency Bands)

Most of the second generation systems were implemented in the 800 and 900 MHz frequency bands. Their application has been extended to the 1800 and 1900 MHz bands too (e.g., GSM 1800 and PCS1900). Carriers from both the bands can be used at the same BSs—either according to a common or independent frequency reuse scheme. Increase in traffic throughput can be maximized by using dual band mobile stations (MSs) and by providing adequate handover mechanisms between the carriers of two bands. Also, due to difference in propagation for carriers from two widely separated bands, certain adjustments related to coverage planning might be required when the carriers are co-sited.

Multiple Speech Codecs for Increased Number of Voice Circuits

Originally, only full rate (FR) speech codecs were used in second generation systems. Hence, a one-to-one correspondence between the physical channels on the air interface and the available voice circuits was established. If the number of installed carriers in a cell is kept unchanged, the introduction of half rate (HR) codecs will double the number of voice circuits in the cell and a more than two-fold increase in Erlang capacity in the cell will be achievable. A similar possibility is offered by the adaptive multirate codec (AMR). The output bit rates of speech codec and channel codec can be adapted in order to minimize the carrier occupancy time necessary to offer a predetermined call quality. The statistical multiplexing gain thus obtained can be exploited to enhance the traffic throughput.

Such capacity enhancement methods can be implemented only if corresponding MSs or MSs with multiple codec capabilities are commercialized. Moreover, every cell site in a network will have to maintain a certain number of FR voice circuits necessary to offer downwards compatibility. Table 26.2 provides an applicability matrix related to the above mentioned capacity enhancement methods.

TABLE 26.2 Applicability Matrix of Methods for Capacity Enhancements Based on Increased Spectrum Availability

Second Generation System	Method for Capacity Enhancement		
	Additional Spectrum in Same Band	Additional Spectrum in Another Band	Multiple Speech Codecs
GSM Family	Available	Under trial in real networks	Under trial in real networks
IS-54/IS-136	Available	Applicable	Applicable
PDC	Available	Applicable	Applicable
IS-95 Family using DS-CDMA	Some of these solutions are applicable. However, their implementation shall adversely effect the functioning of soft hand over which is an essential feature of DS-CDMA networks.		
DECT	Available	Under Consideration	Under Consideration
PHS	Available	Not Applicable	Available

26.3.2 Improved Interference Management Techniques

Initially, the design of a cellular network is based on some simplifying assumptions like uniformly distributed subscriber density and traffic patterns or homogeneous propagation conditions. Usually, such design results in a worst case value of Cs. The situation can be improved through better managemnt of cochannel interference by implementing one or a combination of the following:

- Slow Frequency Hopping (SFH)
- Voice Activity Detection (VAD) and Discontinuous Transmission (DTx)
- Transmit Power Control (PC)
- Antenna beam forming

Slow Frequency Hopping

Every call is "spread" over all the carrier frequencies available in the cell. To avoid intracell interference, orthogonal (random or pseudo random) frequency hopping laws are used for different calls. Also, the worst-case intercell interference situation can last only one hop and an averaging out of cochannel interference in the network occurs. Statistically, the distribution of carrier-to-cochannel interference ratio in the network is more compact with frequency hopping than without it, and this can be exploited to reduce the cluster size Cs and increase capacity.

Voice Activity Detection (VAD) and Discontinuous Transmission (DTx)

Collection of statistics related to telephone conversations has demonstrated that the duty cycle of voice sources is around 40%. With DTx, the radio signal is transmitted according to the activity of voice source. An overall reduction in the averaged cochannel interference experienced by the calls can thus be observed. This offers the possibility of implementing smaller Cs. Also, saving energy with DTx in the up-link (UL) results in a prolonged autonomy for the MS.

Transmit Power Control

Usually, the full available transmit power is necessary for the initial access and for a short duration after call establishment. For the rest of the call, both DL and UL transmit powers can be reduced to a level necessary to maintain a good link quality. The overall improvement in radio interference in the network thus obtained is helpful for the reduction of cluster size. Like VAD/DTx, the DL transmit power control is not permitted for pilot carriers if the mobile assisted handover is implemented in the system.

Antenna Beamforming

Interference related to every call can be individually managed by "dynamic cell sectorization." Actually, the base station receiver captures the up-link signal in a narrow antenna beam dynamically "placed" around the MS. Similarly, the down-link signal is transmitted in a beam focused towards the MS. This sort of spatial filtering of cochannel interference is useful for implementing very compact frequency reuse schemes. Generally, both the down-link and up-link beamforming capabilities are placed at the BS where an antenna array and signal processing algorithms related to direction finding, signal source separation, and beam synthesis have to be implemented. Table 26.3 provides an applicability matrix for the second generation systems.

TABLE 26.3 Applicability Matrix of Methods for Capacity Enhancements Based on Improved Interference Management

Second Generation System	Method for Capacity Enhancement			
	Slow Frequency Hopping	VAD/DTx	Power Control	Antenna Beam Forming
GSM Family	AAP	AAP	AAP	APP
IS-95 (DS-CDMA)	Not Applicable	Essential requirements for satisfactory system operation and not capacity enhancement features		APP
IS-54/IS-136	ANP	ANP	AAP	APP
PDC	ANP	ANP	AAP	APP
DECT	ANP	ANP	ANP	APP
PHS	ANP	ANP	ANP	APP

Note: AAP Applicable and already provided by the standard.
ANP Applicable but not explicitly provided by the standard.
APP Applicable depending on BS equipment design.

Capacity enhancements of 200% or more have been reported [10, 13] through the implementation of combinations of SFH, VAD/DTx, and PC in GSM networks. With SFH and antenna beam forming, Cs of three is achievable for GSM networks [1]. Since VAD/DTx and PC cannot be applied to the pilot carriers, most of the operational networks deploy a dual cluster scheme where Cs for pilot carriers is slightly higher than Cs for traffic carriers. Moreover, some other issues related to pilot channels/carriers have somewhat impeded the introduction of antenna beam forming in GSM or IS-95 (DS-CDMA) networks for cellular applications. However, in wireless local loop applications, substantial capacity gains have been reported for IS-95 [12].

26.3.3 Novel Cellular Configurations

The traffic capacity throughput by a regular grid of homogeneous cells can be increased by cell splitting. Theoretically speaking, if the cell size is divided by two by adding base stations in the middle of existing ones and the current frequency reuse scheme is maintained the achievable traffic capacity is multiplied by four. However, a reduction in cell size beyond a limit leads to excessive overlap between cells due to increased difficulty of coverage prediction. Moreover, since the dwelling time of MS in a cell is reduced, the average number of handovers per call increases. Conventional cellular organizations can no longer meet the requirements for good quality of service due to the failure of handover mechanisms. For high capacity coverage, the following novel cellular configurations have been suggested/implemented:

- Microcells and the associated hierarchical network organization
- Concentric cells
- Frequency reuse of one through generalized slow frequency hopping.

Microcells and Associated Hierarchical Network Organization

Hot spot coverage in dense urban areas is realized in the form of isolated islands of microcells. This is implemented through micro-BS antennas placed below the roof tops of the surrounding buildings. Each antenna radiates very low power. Each island is covered by an umbrella cell which is a part of a continuous macrocellular network over a wider area. Traffic throughput is optimized by intelligent spectrum management performed either off-line or on-line. A set of carriers is assigned to the macrocellular layer organized in a conventional manner. The remaining carriers are repeatedly used in the islands of microcells. Cell selection parameters in the MS and the call admission control algorithms (e.g., Forced Directed Retry) in the base station controllers are adjusted such that a maximum of traffic in the hot spot is taken by the microcells. The umbrella cells are dimensioned to handle the spill-over traffic. This can be the fast moving MS which could generate too many handovers if kept with the microcells. An MS which experiences a sudden degradation of link budget with respect to its serving microcell and/or in the absence of a good target microcell can be temporarily handed over to the umbrella cell.

Despite its difficulty related to spectrum management, this technique has proved to be quite popular for the densification of parts of existing networks based on second generation systems using TDMA/FDMA. [6, 11] provide analysis and guidelines for spectrum management for optimized efficiency in hierarchical networks.

Concentric Cells

A concentric cell coverage is implemented by splitting the available traffic carriers in two groups. One group transmits at full power required to cover the complete cell and the other group transmits at a lower level thus providing the coverage of an inner zone concentric with the original cell. The pilot carrier is transmitted at full power. The localized transmission in the inner zone creates a lower level of interference for other cells. Hence, a smaller cluster size can be used for the frequencies of the inner zone leading to traffic capacity enhancement. Call admission control is designed to keep the MSs near the BS on the carriers of inner zone. Simple intracell handover mechanisms are used to ensure call continuity for the MSs moving across the boundary of the inner zone. Analysis and simulations have shown that optimized capacity enhancement is achieved if the inner zone is limited to 40% of the total cell area. Field trials of concentric cell networks have demonstrated 35% higher spectrum efficiency as compared to single cell networks.

Reuse of One Through Generalized Frequency Hopping and Fractional Loading

Micro- or picocellular networks with TDMA/FDMA systems can be deployed with frequency “reuse of one.” Every base station has the capability of using all the available carriers by slow frequency hopping. The allocation of frequency hopping laws for MSs in clusters of adjacent cells is managed by a centralized control.

During the steady state of operation in a loaded network, only a fraction of the total available bandwidth is in active use in every cell (fractional loading). The level of interference for active calls, for unused circuits and the availability of noninterfered frequency hopping laws is constantly monitored by the network. New calls in a cell are accepted according to the availability of interference-free circuits. In case of unevenly distributed traffic load, an average circuit occupancy in the network is maintained at a level necessary to keep the interference level for active calls below a predetermined threshold. Extreme situations where the same circuit and/or same frequency hopping law is used in two adjacent cells are remedied by intracell handover of one of the two calls. Very high capacity enhancement has been demonstrated in operational GSM networks by using this technique.

Table 26.4 provides the applicability matrix for novel cellular configurations.

TABLE 26.4 Applicability Matrix of Methods for Capacity Enhancements Based on Novel Cellular Configurations

Second Generation System	Method for Capacity Enhancement		
	Micro-cells and Hierarchical N/W	Concentric Cells	Reuse of One with GSFH
GSM Family	AAP	AAP	AAP
IS-54/IS-136	AAP	AAP	ANP
PDC	AAP	AAP	ANP
IS-95 Family using DS-CDMA	Not applicable due to the resulting near–far problem and the power control complexity		Inherent due to DS-CDMA
DECT	AAP	AAP	ANP. Reuse of one possible with DCS
PHS	AAP	AAP	ANP

Note: AAP Applicable and already provided by the standard.
ANP Applicable but not explicitly provided by the standard.
APP Applicable depending on BS equipment design.

26.4 Quality Enhancement

26.4.1 Quality Aspects and Definitions

The quality of service in a telecommunications network can simply be defined as the average performance perceived by the end user in setting up and maintaining a communication. However, its assessment is complex since it is influenced by many parameters, especially in digital wireless systems. In these systems the quality is primarily based on the end-to-end bit error rate and on the continuity of radio links between the two ends. Interference-free radio coverage with sufficient desired signal strength needs to be provided to achieve the above. Moreover, communication continuity has to be ensured between coverage areas with high traffic density (microcells) and low/medium traffic density (macrocells). Three main quality aspects will be distinguished in the following, namely the call handling quality, the communication quality, and the coverage quality.

For speech transmission, the communication quality strongly depends on the intrinsic performance of the speech coder, and its evaluation normally requires intensive listening tests. When it is comparable to the quality achieved on modern wire-line telephone networks, it is called "toll quality." But the **speech quality** is also influenced by other parameters linked to the communication characteristics like radio channel impairments (bit error rate), transmission delay, echo, background noise and tandeming (i.e., when several coding/decoding operations exist in the link).

For data transmission, the communication quality can be more easily quantified based on bit error rate and transmission delay. In synchronous circuit mode, delay is fixed and bit error rate depends on radio channel quality. In packet mode, bit error rate can be kept low thanks to retransmission mechanisms, but average delay increases and throughput decreases as the radio channel degrades.

The **coverage quality** is the percentage of the served area where a communication can be established. It is determined by the acceptable path loss of the radio link and by the propagation characteristics in the area. The radio link budget generally includes some margin depending on the type of terrain (for shadowing effects) and on operator's requirements (for indoor penetration). A coverage quality of 90% is a typical value for cellular networks.

The call handling quality mainly depends on the capacity of the mobile network. When the user is under coverage of the network, the call set-up performance is measured by the blocking rate which depends on the network load. Since in cellular networks mobility of the users is high, capacity requirements are less predictible than in fixed networks which results in a higher blocking rate (typically 1%). Requirements are even more stringent in cordless systems.

A last call-handling quality attribute specific to mobile networks is the success rate in maintaining the communication for mobile users. The handover procedure triggered in cellular networks when the user moves from one cell to another implies the establishment of a new link with some risk to lose the call. The performance is here given by the handover success rate. It must be noted that even if successful, a handover generally results in a short transmission break which degrades the communication quality.

In fact, capacity and quality are dual parameters and a compromise is needed when designing a mobile network. The call handling quality is strongly linked to the correct dimensioning of the network capacity. On the other hand, offering a high capacity in a mobile network implies an intensive use of the available radio spectrum and hence a high average interference level, which may in turn degrade the communication quality. In the following, some major enhancements in speech quality and coverage quality standardized or implemented in second generation systems are reviewed.

26.4.2 Speech Quality Enhancements

In digital cellular systems the first generation of speech coders (full rate) were standardized in the late 1980s to provide good communication quality at medium bit rate (6.7 to 13 kb/s). Operation at a fixed bit rate matched to the intended communication channel was the most important requirement in creating these standards. The main characteristics of the full rate speech coders standardized for second generation cellular systems are listed in Table 26.1 together with other air interface parameters.

With the recent advances in speech coding techniques, low delay toll quality at 8 kb/s and near-toll quality below 6 kb/s are now available [2]. This has enabled the standardization of half rate codecs in cellular systems, though with an increased complexity of implementation. In GSM/DCS, IS-54, and PDC this evolution was anticipated at an early stage, so the need for a half rate speech channel was taken into account in the design of the TDMA frame structure. Half rate codecs provide a doubling of the systems capacity (in terms of number of voice circuits per cell site) while maintaining a speech quality comparable to that available from related full rate codecs. A 5.6 kb/s VSELP coder for GSM and a 3.45 kb/s PSI-CELP coder for PDC were standardized in 1993. However, they have not been widely introduced up to now because of their slightly lower quality in some conditions (e.g., background noise, tandeming) compared to their full rate counterpart.

On the contrary, operators have pushed towards speech quality enhancements, and a new generation of speech coders has emerged. In 1996 ETSI standardized the GSM EFR (enhanced full rate) coder, adopting without any competitive selection process the US1 coder defined for PCS1900 in the U.S. The EFR coder has the remarkable feature of keeping the same channel coding as for the full rate channel, hence simplifying its implementation in the infrastructure. The ACELP coder (using an algebraic codebook with some benefits in computational efficiency like in the the 8 kb/s G.729 ITU standard) has a net bit rate of 12.2 kb/s which leaves room for additional protection (CRC).

Similarly the IS-641 coder has been standardized for IS-54 cellular systems. Its structure is very similar to that of G.729 but with a frame size of 20 ms and a bit rate of 7.4 kb/s. Also a 13 kb/s QCELP coder has been standardized for IS-95. These new coders provide near toll quality in ideal transmission conditions.

Concerning cordless systems, both DECT and PHS use 32 kb/s ADPCM and, therefore, provide toll quality in normal conditions thanks to their relatively high operational C/I. For these systems the concern is rather to introduce lower rate coders with equivalent quality in some kind of half rate mode, allowing for capacity increase, but such coders have not been standardized yet.

With the emergence of variable bit rate (VBR) techniques, some further developments in speech coding are now taking place to satisfy the requirements of cellular operators for toll quality speech with better robustness to radio channel impairments, combined with the capacity increase achievable with half rate operation. One VBR approach is to adapt the bit rate according to the source requirements, taking advantage of silence and stationary segments in the speech signal. Another VBR approach is to adapt the bit rate according to the radio channel quality, either by varying the bit rate allocation between speech and channel coding within a constant gross bit rate (for better quality) or by reducing the gross bit rate in good transmission conditions (for higher capacity). Adaptation to the source characteristics is fast (on a frame-by-frame basis) and adaptation to the channel quality is slower (maximum, a few times per second). In all cases, the underlying idea is that current coders are dimensioned for worst-case operation so that the system quality and capacity can be increased by exploiting the large variations over time of the bit rate requirements (for a given quality).

VBR capabilities with source driven adaptation were introduced since the beginning in IS-95. In CDMA systems the resulting average bit rate reduction directly translates into increased capacity. The initial IS-96B QCELP coder supports four bit rates (8, 4, 2, and 0.8 kb/s) with a scalable CELP architecture, and bit rate adjustment is performed based on adaptive energy thresholds. In practice the two extreme rates are used most frequently. The achievable speech quality was estimated to be lower than that of the IS-54 VSELP coder. Subsequently an enhanced variable rate coder (EVR) was standardized as IS-127. It is based on the relaxed CELP (RCELP) coding technique and supports three rates (8.5, 4, and 0.8 kbit/s).

In Europe, ETSI has launched in 1997 the standardization of an Adaptive Multirate coder (AMR) for GSM systems. Its output bit rate is continuously adapted to radio channel conditions and traffic load. The objective is to find the best compromise between speech quality and capacity by selecting an optimum combination of channel mode and codec mode. The AMR codec can operate in two channel modes, full rate and half rate. For each channel mode there are several possible codec modes (typically three) with different bit rate allocations between speech and channel coding. An adaptation algorithm tracks the variations in speech quality using specific metrics and decides upon the changes in codec mode (up to several times per second). Changes in codec mode is detected by the receiver either via in-band signalling or by automatic mode identification. As shown in Fig. 26.1, the multiplicity of codec modes gives significant performance improvement over any of the corresponding fixed rate codecs.

AMR codec will also allow handovers between half rate and full rate channels using intracell handover mechanisms. The AMR codec will be selected by October 1998 and the full standard will be available in mid-1999. A potential extension is wideband coding, which could be added later to AMR as an option. The wideband option would extend the audio bandwidth from the current 300–3400 Hz to 50–5000 Hz or even 50–7000 Hz.

Speech quality enhancements in cellular networks do not concern only the speech coder itself. First, it is well known that tandeming (i.e., several cascaded coding/decoding operations) can be a

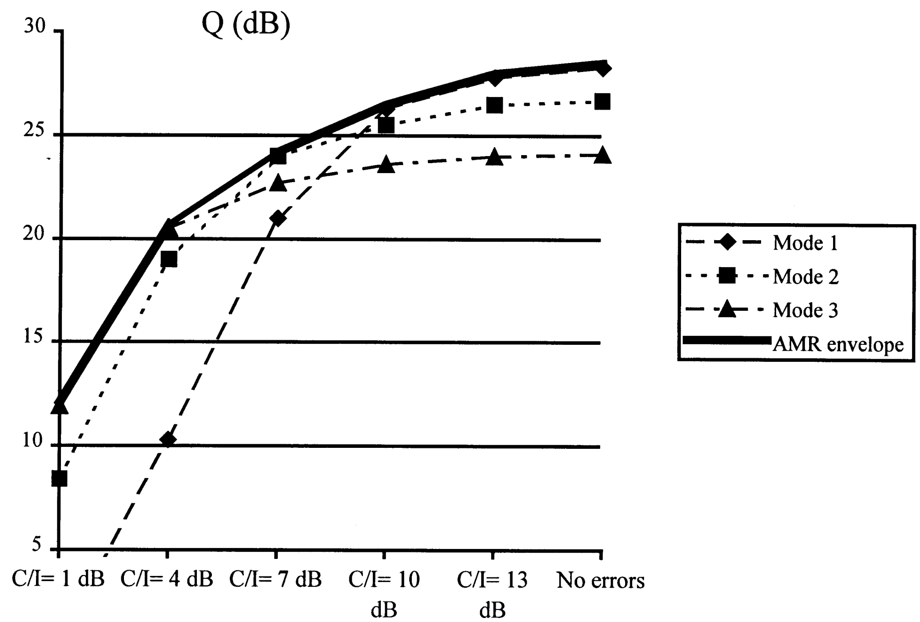

FIGURE 26.1: AMR codec quality as a function of C/I. (Typical results derived from the first AMR selection tests. Quality is given on the equivalent MNRU scale in dB).

source of significant degradation of speech quality. In the case of mobile-to-mobile calls there is no need to decode the speech signal in the network (assuming both mobiles use the same speech codec). Therefore, a tandem-free operation (TFO) will soon be introduced for GSM using in-band signalling between peer transcoders. The benefit of TFO, however, will largely depend on the percentage of intra-GSM mobile-to-mobile calls.

In CDMA systems like IS-95, the soft handoff feature enables a smooth transition from one cell to another. On the contrary in TDMA systems, handovers produce short breaks in the communication which locally degrade the speech quality in spite of the speech extrapolation mechanism used in the decoder. In GSM the duration of the transmission break can be as long as 160 ms due to the necessary time alignment of the mobile in the new cell. The speech interruption can be reduced by 40 ms in a synchronized network, but in practice local synchronization is only offered for co-sited cells. The same performance can be more easily achieved with the presynchronized handover where the mobile receives an indication of the distance to the base station together with the handover command. Some improvement can also be obtained on the uplink by switching at the right point in time and on the downlink by broadcasting the speech information to both the old and the new cell. As a result of all these improvements, the interruption time can be reduced down to 60 ms.

Some advances have also been made concerning the robustness to radio channel impairments. Unequal error protection (UEP) is now used in most codecs designed for cellular systems and it is often based on rate-compatible punctured convolutional codes. UEP enables the adjustment of the protection rate as a function of the sensitivity of bits output by the speech coder. Besides, most linear predictive coders use a speech extrapolation procedure, replacing potentially corrupted information in the current frame with more reliable information from a previously received frame. More sophisticated error concealment techniques, using both reliability information at the output of the channel decoder and a statistical source model (*a priori* knowledge), have been reported to provide up to 3 dB improvement in E_b/N_0 under adverse channel conditions [5].

Robustness to background noise is another topic of interest. Speech quality improvements in medium- to low-bit rate coders have been obtained with coding algorithms optimized for speech signals. Such algorithms may produce poor results in the presence of background noise. Therefore, optional noise cancellation processing has been introduced in recent standards. This can be performed either in the time domain (Kalman filtering) as in JDC half rate or in the frequency domain (spectral subtraction) as in IS-127 EVR.

26.4.3 Coverage Quality Enhancements

Various second generation systems provide the possibility of implementing mechanisms like slow frequency hopping (as in GSM), antenna diversity (also called microdiversity), macrodiversity or multisite transmission, and dynamic channel selection (as in DECT). Such mechanisms are useful to alleviate the effects of radio transmission phenomena like shadowing, fading, and cochannel interference. They are, therefore, particularly interesting to enhance the coverage quality in interference-limited or strong multipath environments (e.g., urban areas). Optional equalizers have been defined for cordless systems (DECT and PHS) with the introduction of a suitable training sequence ("prolonged preamble" in DECT), allowing for channel estimation in the presence of longer delay spread and thus providing increased coverage and/or quality.

Operators also have to face network planning issues linked to natural or artificial obstacles. Various solutions have been designed for the cases where the coverage cannot be efficiently ensured with regular base stations. Radiating cables (leaky feeders) are typically used for the coverage of tunnels. Radio repeaters have been standardized for GSM and for DECT (Wireless Relay Station, WRS) and are useful to fill coverage holes and to provide outdoor-to-indoor or outdoor-to-underground coverage extensions. Microcells (and microbase stations) may also be used as "gap fillers."

The use of smart antennas at cell sites helps to extend the cell range. In rural environments where traffic is low, the required number of cell sites is minimized by using antenna arrays and signal processing algorithms to ensure high sensitivity reception at the base station. Algorithms that implement either n-fold receive diversity or mobile station direction finding followed by beamforming for mobile station tracking have been shown to perform well in such radio channels. Adaptive beamforming techniques with an M-element antenna array generally provide a directivity gain of M, e.g., 9 dB with $M = 8$ (plus some diversity gain depending on channel type). It results in a significant increase of the uplink range (typically by a factor > 1.5 with $M = 8$), but an increased range in the downlink is also needed to get an effective reduction of the number of cell sites. An increased downlink range can be achieved using adaptive beamforming (but with a much higher complexity compared to the uplink-only implementation), a multibeam antenna (i.e., a phased array doing fixed beamforming), or an increased transmit power of the base station. However, the success of smart antenna techniques for range extension applications in second generation systems has been slowed down by their complexity of implementation and by operational constraints (multiple feeders, large antenna panels).

26.5 High Bit Rate Data Transmission

26.5.1 Circuit Mode Techniques

All second generation wireless systems support circuit mode data services with basic rates typically ranging from 9.6 kb/s (in cellular systems) to 32 kb/s (in cordless systems) for a single physical radio resource. With the growing needs for higher rates, new services have been developed based on multiple allocation or grouping of physical resource.

In GSM, HSCSD (High Speed Circuit Switched Data) enables multiple Full Rate Traffic Channels (TCH/F) to be allocated to a call so that a mobile subscriber can use n times the transmission capacity of a single TCH/F channel (Fig. 26.2). The n full rate channels over which the user data stream is split are handled completely independently in the physical layer and for layer 1 error control. The HSCSD channel resulting from the logical combination of n TCH/F channels is controlled as a single radio link during cellular operations such as handover. At the A interface, calls will be limited to a single 64 kb/s circuit. Thus HSCSD will support transparent (up to 64 kb/s) and nontransparent modes (up to $4 \times 9.6 = 38.4$ kb/s and, later, $4 \times 14.4 = 57.6$ kb/s). The initial allocation can be changed during a call if required by the user and authorized by the network. Initially the network allocates an appropriate HSCSD connection according to the requested user bit rate over the air interface. Both symmetric and asymmetric configurations for bidirectional HSCSD operation are authorized. The required TCH/F channels are allocated over consecutive or nonconsecutive timeslots.

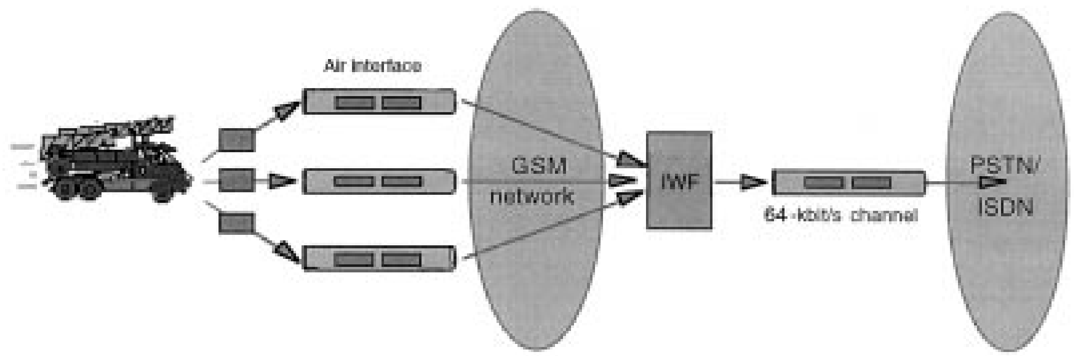

FIGURE 26.2: Simplified GSM network configuration for HSCSD.

Similar multislot schemes are envisaged or standardized for other TDMA systems. In IS-54 and PDC, where radio channels are relatively narrowband, no more than three time slots can be used per carrier and the achievable data rate is therefore limited to, say, 32 kb/s. On the contrary, in DECT up to 12 time slots can be used at 32 kb/s each, yielding a maximum data rate of 384 kb/s. Moreover, the TDD access mode of DECT allows asymmetric time slot allocation between uplink and downlink, thus enabling even higher data rates in one direction.

26.5.2 Packet Mode Techniques

There is a growing interest for packet data services in second generation wireless systems to support data applications with intermittent and bursty transmission requirements like the Internet, with a better usage of available radio resources, thanks to the multiplexing of data from several mobile users on the same physical channel. Cellular Digital Packet Data (CDPD) has been defined in the U.S. as a radio access overlay for AMPS or D-AMPS (IS-54) systems, allowing packet data transmission on available radio channels. However, CDPD is optimized for short data transmission and the bit rate is limited to 19.2 kb/s. A CDMA packet data standard has also been defined (IS-657) which supports

CDPD and Internet protocols with a similar bit rate limitation but allowing use of the same backhaul as for voice traffic.

In Europe, ETSI has almost completed the standardization of GPRS (General Packet Radio Service) for GSM. A GPRS subscriber will be able to send and receive in an end-to-end packet transfer mode. Both point-to-point and point-to-multipoint modes are defined. A GPRS network coexists with a GSM PLMN as an autonomous network. In fact, the Serving GPRS Support Node (SGSN) interfaces with the GSM Base Station Controller (BSC), an MSC and a Gateway GPRS Service Node (GGSN). In turn, the GGSN interfaces with the GGSNs of other GPRS networks and with public Packet Data Networks (PDN). Typically, GPRS traffic can be set up through the common control channels of GSM, which are accessed in slotted ALOHA mode. The layer 2 protocol data units, which are about 2 kbytes in length, are segmented and transmitted over the air interface using one of the four possible channel coding schemes. The system is highly scaleable as it allows from one mobile using 8 radio time slots up to 16 mobiles per time slot, with separate allocation in up- and downlink. The resulting peak data rate per user ranges from 9 kb/s up to 170 kb/s. Time slot concatenation and variable channel coding to maximize the user information bit rate are envisaged for future implementations. This is indicated by the mobile station, which provides information concerning the desire to initiate in-call modifications and the channel coding schemes that can be used during the call set up phase. It is expected that use of the GPRS service will initially be limited and traffic growth will depend on the introduction of GPRS capable subscriber terminals. Easy scalability of the GPRS backbone (e.g., by introducing parallel GGSNs) is an essential feature of the system architecture (Fig. 26.3).

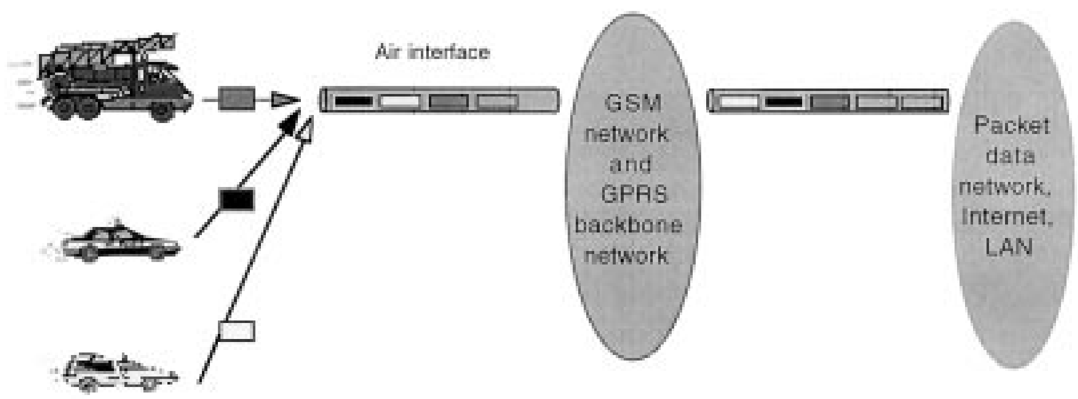

FIGURE 26.3: Simplified view of the GPRS architecture.

26.5.3 New Modulation Schemes

New modulation schemes are being studied as an option in several second generation wireless standards. The aim is to offer higher rate data services equivalent or close to the 2 Mb/s objective of the forthcoming third generation standards. Multilevel modulations (i.e., several bits per modulated symbol) represent a straightforward means to increase the carrier bit rate. However, it represents a significant change in the air interface characteristics, and the increased bit rate is achieved at the expense of a higher operational signal-to-noise plus interference ratio, which is not compatible with large cell dimensions. Therefore, the new high bit rate data services are mainly targetting urban areas, and the effective bit rate allocated to data users will depend on the system load.

Such a new air interface option is being standardized for GSM under the name of EDGE (Enhanced Data rates for GSM Evolution). The selected modulation scheme is 8-PSK, suitable coding schemes are under study, whereas the other air interface parameters (carrier spacing, TDMA frame structure,...) are kept unchanged. Reusing HSCSD (for circuit data) and GPRS (for packet data) protocols and service capabilities, EDGE will provide similar ECSD and EGPRS services but with a three-fold increase of the user bit rate. The higher level modulation requires better radio link performances, typically a loss of 3 to 4 dB in sensitivity and a C/I increased by 6 to 7 dB. Operation will also be restricted to environments with limited time dispersion and limited mobile speed. Nevertheless, EGPRS will roughly double the mean throughput compared to GPRS (for the same average transmitted power). EDGE will also increase the maximum achievable data rate in a GSM system to 553.6 kb/s in multislot (unprotected) operation. Six different protection schemes are foreseen in EGPRS using convolutional coding with a rate ranging from 1/3 to 1 and corresponding to user rates between 22.8 and 69.2 kb/s per time slot. This is in addition to the four coding schemes already defined for GPRS. An intelligent link adaptation algorithm will dynamically select the most appropriate modulation and coding schemes, i.e., those yielding the highest throughput for a given channel quality. The first phase of EDGE standardization should be completed by end 1999. It should be noted that a similar EDGE option is being studied for IS-54/IS-136 (and their PCS derivatives). Initially, the 30 kHz channel spacing will be maintained and then extension to a 200 kHz channel will be provided in order to offer a convergence with its GSM counterpart.

A higher bit rate option is also under standardization for DECT. Here it is seen as an essential requirement to maintain backward compatibility with existing equipment so the new multilevel modulation will only affect the payload part of the bursts, keeping the control and signalling parts unchanged. This ensures that equipment with basic modulation and equipment with a higher rate option can efficiently share a common base station infrastructure. Only 4-level and 8-level modulations are considered and the symbol length, carrier spacing, and slot structure remain unchanged. The requirements on transmitter modulation accuracy need to be more stringent for 4- and 8-level modulation than for the current 2-level scheme. An increased accuracy can provide for coherent demodulation, whereby some (or most) of the sensitivity and C/I loss when using the multilevel mode can be regained. In combination with other new air interface features like forward error correction and double slots (with reduced overhead), the new modulation scheme will provide a wide range of data rates up to 2 Mb/s. For instance using (Π/4-DQPSK modulation (a possible/suitable choice), an unprotected connection with two double slots in each direction gives a data rate of 384 kb/s. Asymmetric connections with a maximum of 11 double slots in one direction will also be supported.

26.6 Conclusion

Since their introduction in the early 1990s, most of the second generation systems have been enjoying exponential growth. With more than 100 million subscribers acquired worldwide in less than ten years of lifetime, the systems based on the GSM family of standards have demonstrated the most spectacular development. Despite a more regional implementation of other second generation systems, each one of those can boast a multimillion subscriber base in mobile or fixed wireless networks.

A variety of service requirements of third generation mobile communication systems are being already met by the upcoming enhancements of second generation systems. Two important trends are reflected by this:

- The introduction of third generation systems like Universal Mobile Telecommunication

System (UMTS) or International Mobile Telecommunication-2000 (IMT-2000) might be delayed to a point in time where the evolutionary capabilities of second generation systems have been exhausted.

- The deployment of networks based on third generation systems will be progressive. Any new radio interface will be imposed worldwide if and only if it provides substantial advantages as compared to the present systems. Another essential requirement is the capability of downward compatibility to second generation systems.

Defining Terms

Capacity: In a mobile network it can be defined as the Erlangs throughput by a cell, a cluster of cells, or by a portion of a network. For a given radio interface, the achievable capacity is a function of the robustness of the physical layer, the effectiveness of the medium access control (MAC) layer and the multiple access technique. Moreover, it is strongly dependent on the radio spectrum available for network planning.

Cellular: Refers to public land mobile radio networks for generally wide area (e.g., national) coverage, to be used with medium- or high-power vehicular mobiles or portable stations and for providing mobile access to the Public Switched Telephone Network (PSTN). The network implementation exhibits a cellular architecture which enables frequency reuse in nonadjacent cells.

Cordless: These are systems to be used with simple low power portable stations operating within a short range of a base station and providing access to fixed public or private networks. There are three main applications, namely, residential (at home, for Plain Old Telephone Service, POTS), public-access (in public places and crowded areas, also called Telepoint), and Wireless Private Automatic Branch eXchange (WPABX, providing cordless access in the office environment), plus emerging applications like radio access for local loop.

Coverage quality: It is the percentage of the served area where a communication can be established. It is determined by the acceptable path loss of the radio link and by the propagation characteristics in the area. The radio link budget generally includes some margin depending on the type of terrain (for shadowing effects) and on operator's requirements (for indoor penetration). A coverage quality of 90% is a typical value for cellular networks.

Speech quality: It strongly depends on the intrinsic performance of the speech coder and its evaluation normally requires intensive listening tests. When it is comparable to the quality achieved on modern wire-line telephone networks, it is called "toll quality." In wireless systems it is also influenced by other parameters linked to the communication characteristics like radio channel impairments (bit error rate), transmission delay, echo, background noise, and tandeming (i.e., when several coding/decoding operations are involved in the link).

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Further Information

European standards (GSM, CT2, DECT, TETRA) are published by ETSI Secretariat, 06921 Sophia Antipolis Cedex, France.

U.S. standards (IS-54, IS-95, APCO) are published by Electronic Industries Association, Engineering Department, 2001 Eye Street, N.W., Washington D.C. 20006, U.S.A.

Japanese standards (PDC, PHS) are published by RCR (Research and Development Center for Radio Systems), 1-5-16, Toranomon, Minato-ku, Tokyo 105, Japan.