NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

GSM RADIO INTERFACE

Graduation Project
EE-400

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ABSTRACT

GSM happens to be one of the most popular cellular radio systems implemented in the world today. Now GSM has gathered more than 22 million subscribers only in Europe. It provides a wide range of services and facilities, both voice and data, that are compatible with the existing fixed Public Switched telephone Networks (PSTN), Public Switched Data Networks (PSDN), Public Land Mobile Networks (PLMN) and Integrated Service Digital Networks (ISDN).

It's main function is to give compatibility considered mobile subscriber the access to any mobile subscriber in any country, which operates the system, and provides facilities for automatic roaming, locating and updating the mobile subscriber's status.

GSM as a modern telecommunication system is a complex object. It's implementation and operation is not an easy task, neither easy it's description.

GSM Radio Interface is the interface between the mobile stations and the fixed infrastructure. It is one of the most important interfaces in the GSM system. The specification of the radio interface has then as important influence on the spectrum efficiency.

INTRODUCTION

In this project Global Systems for Mobile is introduced and specifically GSM Radio Interface has been considered. This project consists of five chapters.

Chapter 1 is an introduction to GSM, history and architecture of GSM, working principle of GSM, its communication specifications, coding and modulations.

Chapter 2 is mainly focused on Radio Transmission and GSM Transmission Systems.

Chapter 3 is a brief description of Digital Radio Systems, containing information's about advantages of radio transmission, GSM Pan European Digital Cellular Radio, Aspects of The GSM Networks, Equalization and Multipath Propagation, Digital Modulation, Technical Difference Between GPRS and EGPICS, Coding Schemes, Packet Handling, Addressing Window, Measurement Accuracy, Interleaving, EGPRS Link Controlling Function, Link Adaptation, Incremental Reducing, Impact of EGPRS, GSM/GPRS Networks, Standardization, GERAN System Architecture.

Chapter 4 is a short description of Encryptions, Limitations of Security, The Functions of The Services, Anonymity including Authentications and User Data Signalling Protection, Implementation and Roaming and World Wide Use of The Algorithms.

Chapter 5 presents a short introduction of Power Control, Frequency Hopping, Measurement Campaign, Statistic Trials, Specific Trials, Measurements Results, Bit Error Rate and Interfacing BCCH

1. INTRODUCTION

1.1 History of G.S.M

During the early 1980s, analog cellular telephone systems were experiencing rapid growth in Europe, particularly in Scandinavia and the United Kingdom, but also in France and Germany. Each country developed its own system, which was incompatible with everyone else's in equipment and operation. This was an undesirable situation, because not only was the mobile equipment limited to operation within national boundaries, which in a unified Europe were increasingly unimportant, but there was a very limited market for each type of equipment, so economies of scale, and the subsequent savings, could not be realized.

In 1981 a joint Franco German study was initiated to develop a common approach which, it was hoped, would become a standard for Europe. Soon after, in 1982 a proposal from Nordic Telecom and Netherlands PTT to the CEPT (Conference of European Post and Telecommunications) to develop a new digital cellular standard that would cope with the ever burgeoning demands on European mobile networks. Then a study group formed called the Global System for Mobile (GSM) to study and develop a pan-European public land mobile system. The proposed system had to meet certain criteria:

- Good subjective speech quality
- Low terminal and service cost
- Support for international roaming
- Ability to support handheld terminals
- Support for range of new services and facilities
- Spectral efficiency
- ISDN compatibility

In 1989, GSM responsibility was transferred to the European Telecommunication Standards Institute (ETSI), and phase I of the GSM specifications was published in 1990. Commercial service was started in mid-1991, and by 1993 there were 36 GSM networks in 22 countries. Although standardized in Europe, GSM is not only a European standard. Over

200 GSM networks (including DCS1800 and PCS1900) are operational in 110 countries around the world. In the beginning of 1994, there were 1.3 million subscribers worldwide, which had grown to more than 55 million by October 1997. With North America making a delayed entry into the GSM field with a derivative of GSM called PCS1900, GSM systems exist on every continent, and the acronym GSM now aptly stands for Global System for Mobile communications.

The developers of GSM chose an unproven (at the time) digital system, as opposed to the then-standard analog cellular systems like AMPS in the United States and TACS in the United Kingdom. They had faith that advancements in compression algorithms and digital signal processors would allow the fulfillment of the original criteria and the continual improvement of the system in terms of quality and cost. The over 8000 pages of GSM recommendations try to allow flexibility and competitive innovation among suppliers, but provide enough standardization to guarantee proper interworking between the components of the system. This is done by providing functional and interface descriptions for each of the functional entities defined in the system.

The original French name was later changed to Global System for Mobile Communications, but the original GSM acronym stuck.

1.2 G.S.M System Architecture

A GSM network is composed of several functional entities, whose functions and interfaces are defined. The GSM network can be divided into three broad parts. The Mobile Station is carried by the subscriber, the Base Station Subsystem controls the radio link with the Mobile Station. The Network Subsystem, the main part of which is the Mobile services Switching Centre, performs the switching of calls between the mobile and other fixed or mobile network users, as well as management of mobile services, such as authentication. With the Operations and Maintenance centre, which oversees the proper operation and setup of the network. The Mobile Station and the Base Station Subsystem communicate

across the Um interface, also known as the air interface or radio link. The Base Station Subsystem communicates with the Mobile service Switching Center across the A interface.

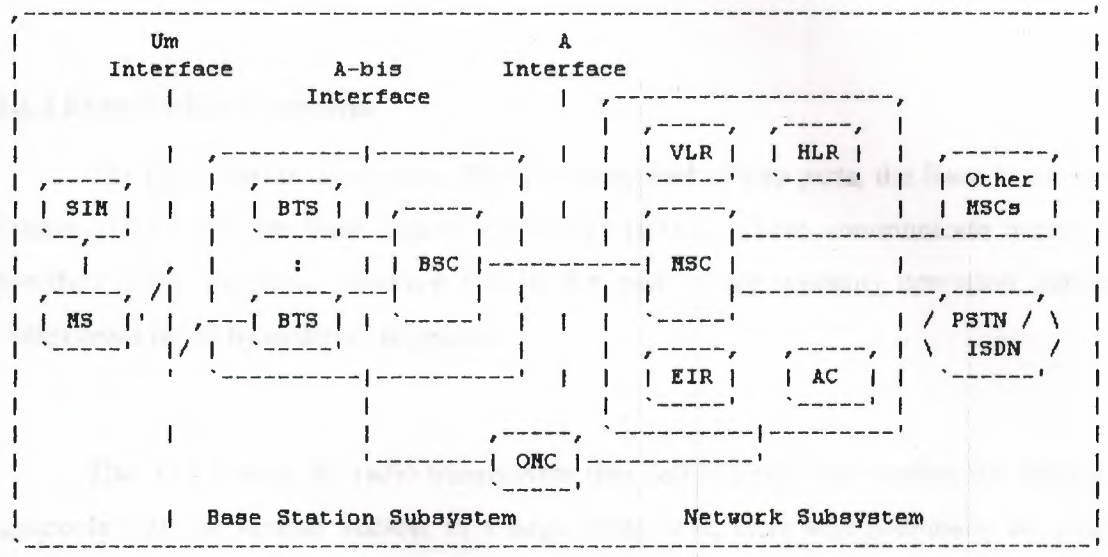

Figure 1.1 GSM Architecture Illustrated

1.2.1 Mobile Station

The mobile station (MS) consists of the physical equipment, such as the radio transceiver, display and digital signal processors, and a smart card called the Subscriber Identity Module (SIM). The SIM provides personal mobility, so that the user can have access to all subscribed services irrespective of both the location of the terminal and the use of a specific terminal. By inserting the SIM card into another GSM cellular phone, the user is able to receive calls at that phone, make calls from that phone, or receive other subscribed services.

The mobile equipment is uniquely identified by the International Mobile Equipment Identity (IMEI). The SIM card contains the International Mobile Subscriber Identity

(IMSI), identifying the subscriber, a secret key for authentication, and other user information. The IMEI and the IMSI are independent, thereby providing personal mobility. The SIM card may be protected against unauthorized use by a password or personal identity number.

1.2.2 Base Station Subsystem

The Base Station Subsystem (BSS) is composed of two parts, the Base Transceiver Station (BTS) and the Base Station Controller (BSC). These communicate across the specified Abis interface, allowing (as in the rest of the system) operation between components made by different suppliers.

The BTS houses the radio transceivers that define a cell and handles the radio link protocols with the Mobile Station. In a large urban area, there will potentially be a large number of BTSs deployed. The requirements for a BTS are ruggedness, reliability, portability, and minimum cost. BTS is responsible for providing layers 1 and 2 of the radio interface, that is, an error-corrected data path. Each BTS has at least one of its radio channels assigned to carry control signals in addition to traffic.

The BSC manages the radio resources for one or more BTSs. It is responsible for the management of the radio resource within a region. Its main functions are to allocate and control traffic channels, control frequency hopping, undertake handovers (except to cells outside its region) and provide radio performance measurements. Once the mobile has accessed, and synchronized with, a BTS the BSC will allocate it a dedicated bi-directional signaling channel and will set up a route to the Mobile services Switching Center (MSC). The BSC also translates the 13 KBPS voice channel used over the radio link to the standard 64 KBPS channel used by the Public Switched Telephone Network or ISDN.

1.2.3 Network Subsystem

The central component of the Network Subsystem is the Mobile services Switching Center (MSC). It acts like a normal switching node of the PSTN or ISDN, and in addition

provides all the functionality needed to handle a mobile subscriber, such as registration, authentication, location updating, handovers, and call routing to a roaming subscriber. These services are provided in conjunction with several functional entities, which together form the Network Subsystem. The MSC provides the connection to the public fixed network (PSTN or ISDN), and signaling between functional entities uses the ITUT Signaling System Number 7 (SS7), used in ISDN and widely used in current public networks.

The Home Location Register (HLR) and Visitor Location Register (VLR), together with the MSC, provide the call routing and (possibly international) roaming capabilities of GSM. The HLR contains all the administrative information of each subscriber registered in the corresponding GSM network, along with the current location of the mobile. It also contains a unique authentication key and associated challenge/response generators. The current location of the mobile is in the form of a Mobile Station Roaming Number (MSRN), which is a regular ISDN number used to route a call to the MSC where the mobile is currently located. There is logically one HLR per GSM network, although it may be implemented as a distributed database.

The VLR contains selected administrative information from the HLR, necessary for call control and provision of the subscribed services, for each mobile currently located in the geographical area controlled by the VLR. Although each functional entity can be implemented as an independent unit, most manufacturers of switching equipment implement one VLR together with one MSC, so that the geographical area controlled by the MSC corresponds to that controlled by the VLR, simplifying the signaling required. Note that the MSC contains no information about particular mobile stations - this information is stored in the location registers.

The other two registers are used for authentication and security purposes. The Equipment Identity Register (EIR) is a database that contains a list of all valid mobile equipment on the network, where each mobile station is identified by its International

Mobile Equipment Identity (IMEI). An IMEI is marked as invalid if it has been reported stolen or is not type approved. The Authentication Center is a protected database that stores a copy of the secret key stored in each subscriber's SIM card, which is used for authentication and ciphering of the radio channel.

1.2.4 Operation and Management

A network having complexity like G.S.M service must be managed and maintained. This is the function of the Operations and Maintenance Center (OMC). GSM leaves decisions on O&M to the individual operators, but general guidelines are given which reinforce the move towards an overlaid telecommunications management network. In this approach five separate management functions are identified as:

- Operations and Performance Management
- Maintenance
- System Change Control
- Security Management
- Administration and Commercial Functionality

1.3 How It Works

1.3.1 Make Call

When the mobile user initiates a call, his equipment will search for a local base station, i.e. The BSS. Once the mobile has accessed, and synchronized with, a BTS the BSC will allocate it a dedicated bi-directional signaling channel and will set up a route to the Mobile services Switching Center (MSC).

1.3.2 Call initialization

When a mobile requests access to the system it has to supply its IMEI (International Mobile Equipment Identity). This is a unique number, which will allow the system to initiate a process to confirm that the subscriber is allowed to access it. This process is called

authentication. Before it can do this, however, it has to find where the subscriber is based. Every subscriber is allocated to a home network, associated with an MSC within that network. This is achieved by making an entry in the Home Location Register (HLR), which contains information about the services the subscriber is allowed.

Whenever a mobile is switched on and at intervals thereafter, it will register with the system; this allows its location in the network to be established and its location area to be updated in the HLR. A location area is a geographically defined group of cells. On first registering, the local MSC will use the IMSI to interrogate the subscriber's HLR and will add the subscriber data to its associated Visitor Location Register (VLR). The VLR now contains the address of the subscriber's HLR and the authentication request is routed back through the HLR to the subscriber's Authentication Center (AC). This generates a challenge/response pair, which is used by the local network to challenge the mobile. In addition, some operators also plan to check the mobile equipment against an Equipment Identity Register (EIR), in order to control stolen, fraudulent or faulty equipment.

1.3.3 Authentication

The authentication process is very powerful and is based on advanced cryptographic principles. It especially protects the network operators from fraudulent use of their services. It does not however protect the user from eavesdropping. The Time Division Multiple Access (TDMA) nature of GSM coupled with its frequency hopping facility will make it very difficult for an eavesdropper to lock onto the correct signal however and thus there is a much higher degree of inherent security in the system than is found in today's analogue systems. Nevertheless for users who need assurance of a secure transmission, GSM offers encryption over the air interface. This is based on a public key encryption principle and provides very high security.

1.3.4 Call Set-up

Once the network accepts the user and his equipment, the mobile must define the type of service it requires (voice, data, supplementary services etc.) and the destination number. At this point a traffic channel with the relevant capacity will be allocated and the MSC will route the call to the destination. Note that the network may delay assigning the traffic channel until the connection is made with the called number. This is known as off-air call set-up, and it can reduce the radio channel occupancy of any one call thus increasing the system traffic capacity.

1.3.5 Handover

GSM employs mobile-assisted handover. In this technique the mobile continuously monitors other base stations in its vicinity, measuring signal strength and error rate. These measurements are combined into a single function and the identities of the best six base stations are transmitted back to the system. The network can then decide when to initiate handover. The use of bit error rate, in addition to signal strength, adds considerably to the ability of the network to make informed handover decisions and is another example of the advantage of digital transmission over analogue. The BSC can initiate and execute handover if both BTS's are under its own control. In this instance the BSC can be considered as the manager of a specific group of radio frequencies for a geographic region and can control that resource to maximize its utilization. Alternatively, and whenever handover must take place to a cell outside the control of the BSC, the MSC controls and executes handover.

1.4 Communication Specification

1.4.1 Frequency Allocation

Throughout Europe GSM has been allocated a specific 50 MHz of spectrum divided into transmit and receive bands. The International Telecommunication Union (ITU), which manages the international allocation of radio spectrum (among other functions) allocated

the bands 890-915 MHz for the uplink (mobile station to base station) and 935-960 MHz for the downlink (base station to mobile station) for mobile networks in Europe.

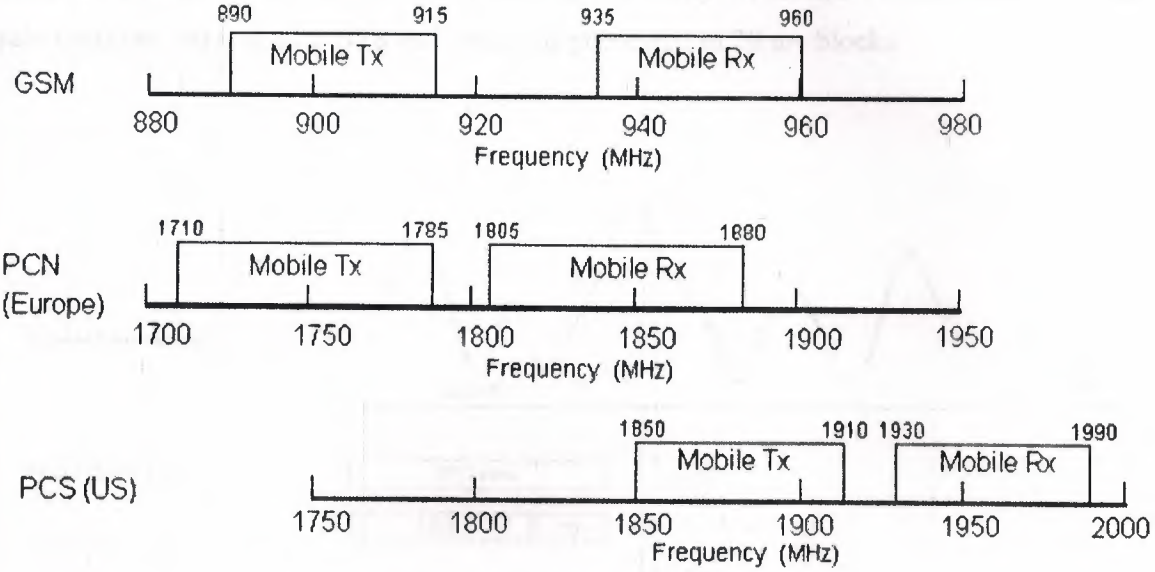

Figure 1.2 The Frequency Allocation of GSM

Since radio spectrum is a limited resource shared by all users, a method must be devised to divide up the bandwidth among as many users as possible. The method chose by GSM is a combination of Time and Frequency Division Multiple Access (TDMA/FDMA). The FDMA part involves the division by frequency of the total 25 MHz bandwidth into 124 carrier frequencies of 200 kHz bandwidth. One or more carrier frequencies are then assigned to each base station. Each of these carrier frequencies is then divided in time, using a TDMA scheme, into eight time slots. One time slot is used for transmission by the mobile and one for reception. They are separated in time so that the mobile unit does not receive and transmit at the same time, a fact that simplifies the electronics.

1.4.2 Speech Coding

GSM is a digital system, so speech signals, inherently analog, have to be digitized. The spectral efficiency target set for the GSM system demands a speech codec, which can

provide toll quality speech at 16 KBPS or less. The solution adopted is based on a Residually Excited Linear Predictive coder (RELP) enhanced by the inclusion of a long-term predictor (LTP). This improves the speech quality by removing the structure from the vowel sounds prior to coding the residual data. It has the effect of removing the coarseness often associated with linear predictive coding, especially on female voices. The basic data rate from the coder is 13 KBPS and speech is processed in 20 ms blocks.

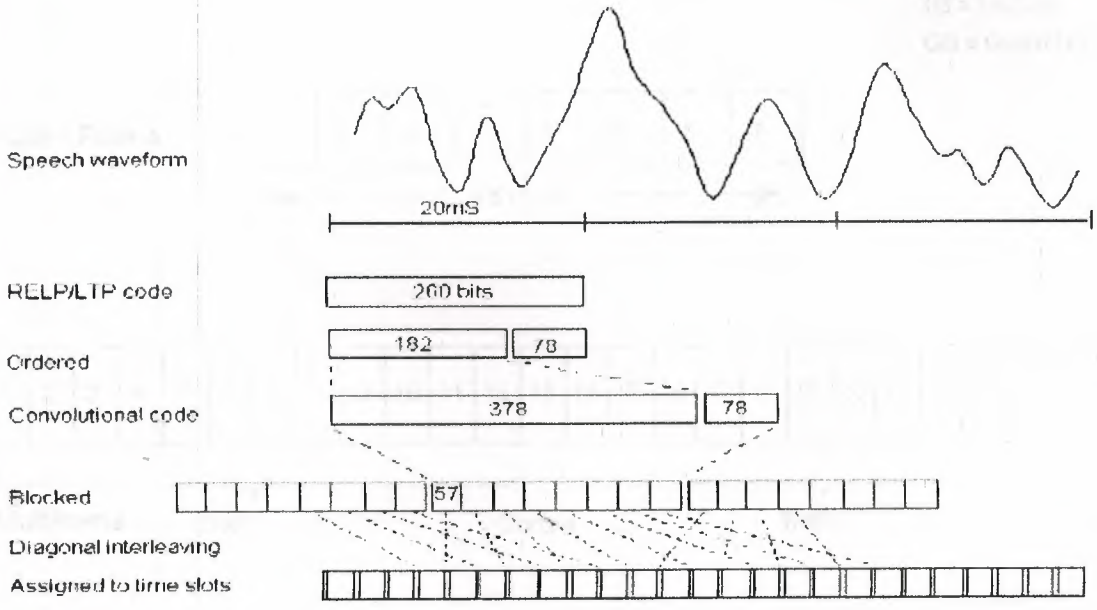

Figure 1.3 Speech Coding

The resulting code is split into two parts, the most critical bits being put first. This first part has a half-rate convolution code applied to it and when recombined with the second part the total block length is 456 bits. As we will see later this block length can be fitted into four time slots, but in practice it is spread over eight. This process is called diagonal interleaving and it allows the convolution code more chance to recover if a sequence of TDMA frames is badly corrupted during radio transmission.

1.4.3 Data/Channel Structure

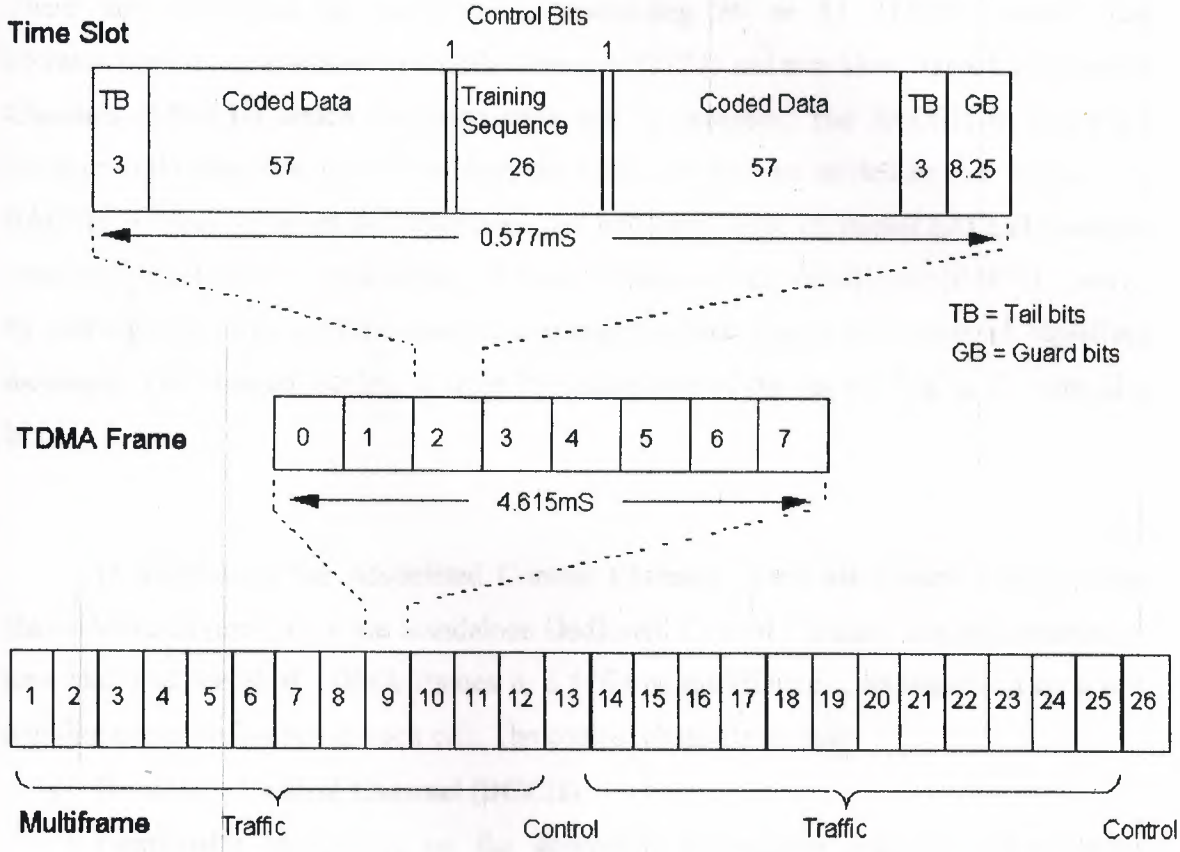

Figure 1.4 Structure of Data Channel

The structure of the most common timeslot burst is shown as above. A total of 156.25 bits is transmitted in 0.577 milliseconds, giving a gross bit rate of 270.833 KBPS. There are three other types of burst structure for frame and carrier synchronization and frequency correction. The 26bit training sequence is used for equalization, as described below. The 8.25 bit guard time allows for some propagation time delay in the arrival of bursts.

Each group of eight time slots is called a TDMA frame, which is transmitted every 4.615 ms. TDMA frames are further grouped into multiframes to carry control signals. There are two types of multiframe , containing 26 or 51 TDMA frames. The 26frame multiframe contains 24 Traffic Channels (TCH) and two Slow Associated Control Channels (SACCH) which supervise each call in progress. The SACCH in frame 12 contains eight channels, one for each of the eight connections carried by the TCHs. The SACCH in frame 25 is not currently used, but will carry eight additional SACCH channels when half rate traffic is implemented. A Fast Associated Control Channel (FACCH) works by stealing slots from a traffic channel to transmit power control and handover signalling messages. The channel stealing is done by setting one of the control bits in the time slot burst.

In addition to the Associated Control Channels, there are several other control channels which (except for the Standalone Dedicated Control Channel) are implemented in time slot 0 of specified TDMA frames in a 51frame multiframe , implemented on a non hopping carrier frequency in each cell. The control channels include:

- **Broadcast Control Channel (BCCH):**
  Continually broadcasts, on the downlink, information including base station identity, frequency allocations, and frequency hopping sequences.
- **Standalone Dedicated Control Channel (SDCCH):**
  Used for registration, authentication, call setup, and location updating.
  Implemented on a time slot, together with its SACCH, selected by the system operator.
- **Common Control Channel (CCCH):**
  Comprised of three control channels used during call origination and call paging.
- **Random Access Channel (RACH):**
  A slotted Aloha channel to request access to the network
- **Paging Channel (PCH):**
  Used to alert the mobile station of incoming call.

- **Access Grant Channel (AGCH):**
  Used to allocate an SDCCH to a mobile for signaling, following a request on the RACH.

1.5 Channel Coding and Modulation

Due to natural or man made electromagnetic interference, the encoded speech or data transmitted over the radio interface must be protected as much as is practical. The GSM system uses convolution encoding and block interleaving to achieve this protection. The exact algorithms used differ for speech and for different data rates. The method used for speech blocks will be described below.

Recall that the speech codec produces a 260-bit block for every 20 ms speech sample. From subjective testing, it was found that some bits of this block were more important for perceived speech quality than others. The bits are thus divided into three classes:

- Class I a 50 bits - most sensitive to bit errors
- Class I b 132 bits - moderately sensitive to bit errors
- Class II 78 bits - least sensitive to bit errors

Class Ia bits have a 3 bit Cyclic Redundancy Code added for error detection. If an error is detected, the frame is judged too damaged to be comprehensible and it is discarded. It is replaced by a slightly attenuated version of the previous correctly received frame. These 53 bits, together with the 132 Class Ib bits and a 4-bit tail sequence (a total of 189 bits), are input into a 1/2 rate convolution encoder of constraint length 4. Each input bit is encoded as two output bits, based on a combination of the previous 4 input bits. The convolution encoder thus outputs 378 bits, to which are added the 78 remaining Class II bits, which are unprotected. Thus every 20 ms speech sample is encoded as 456 bits, giving a bit rate of 22.8 KBPS.

To further protect against the burst errors common to the radio interface, each sample is diagonally interleaved. The 456 bits output by the convolution encoder are divided into 8 blocks of 57 bits, and these blocks are transmitted in eight consecutive timeslot bursts. Since each timeslot burst can carry two 57-bit blocks, each burst carries traffic from two different speech samples.

Recall that each timeslot burst is transmitted at a gross bit rate of 270.833 KBPS. This digital signal is modulated onto the analog carrier frequency, which has a bandwidth of 200 kHz, using Gaussian filtered Minimum Shift Keying (GMSK). GMSK was selected over other modulation schemes as a compromise between spectral efficiency, complexity of the transmitter, and limited spurious emissions. The complexity of the transmitter is related to power consumption, which should be minimized for the mobile station. The spurious radio emissions, outside of the allotted bandwidth, must be strictly controlled so as to limit adjacent channel interference, and allow for the coexistence of GSM and the older analog systems (at least for the time being).

2. RADIO TRANSMISSION

2.1 Introduction

GSM is a digital wireless communication system for mobile phones. It was developed in the early 1980's to eliminate certain problems with the existing cellular networks:

- Analog networks could not handle the growing capacity of cellular networks, especially in an economic way
- The existing digital networks were not compatible with each other

It was hoped that by creating one digital standard, the problems of analog systems would be overcome and users could roam from network to network without changing their equipment.

In addition to providing a solution for the above two problems, GSM supported many other useful features such as security, authentication and the SIM card, which makes it possible to switch phones without the need of reconfigure the phone.

In 1982 the European Commission reserved the 900 MHz spectrum in all member countries for GSM, thus setting the stage for interoperability across Europe. From 1982 to 1990 the specifications for GSM have been decided upon and in 1990 the final specifications were chosen. After 2 years, the first GSM network was operable in 1992 in Finland, and since then GSM has been a huge success.

Since the original GSM specifications in 1990, the GSM standard has been upgraded twice. In 1991 a new European GSM standard was developed to operate at 1800 MHz (usually called DCS1800). The new standard provided more bandwidth and less power requirements than the previous standard. Furthermore, in 1995 a North American GSM standard at 1900 MHz was established.

2.2 The GSM Transmission System

To transform the speech that enters the headset into the radio waves that are transmitted to the base station, GSM performs the following operations.

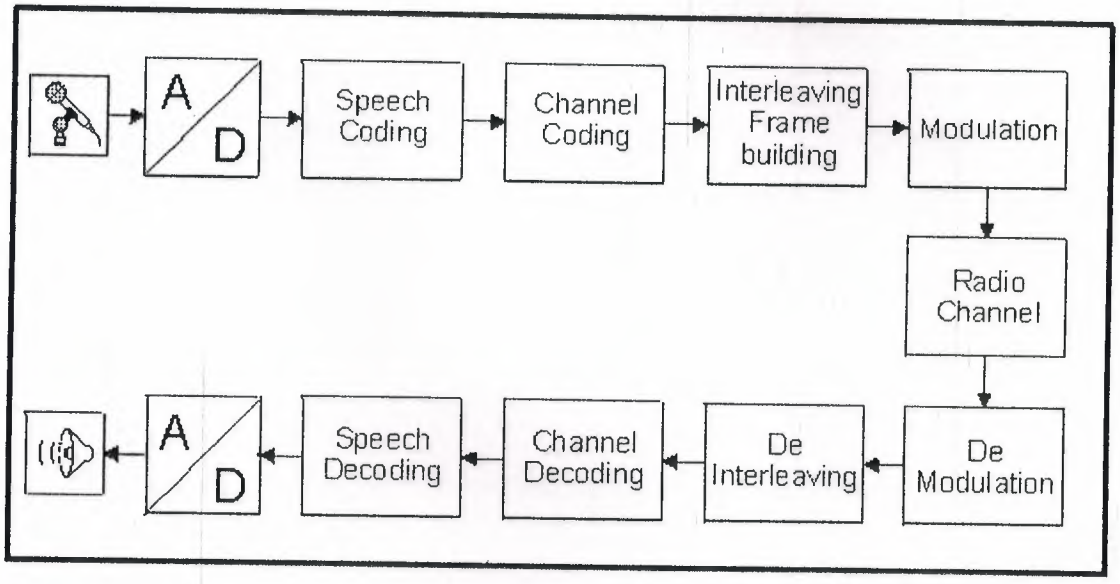

Figure 2.1 Transmission Systems in GSM

GSM is a digital communications standard, but voice is analog, and therefore it must be converted to a digital bit stream. GSM uses Pulse Coded Modulation to digitize voice, and then uses a speech codec to remove the redundancy in the signal.

Once the voice signal has been coded into a digital bit stream, extra bits are added to it so that the receiver can recognize and correct errors in the stream, which could have occurred during transmission. GSM uses a technique called convolution coding. Interleaving is the processes of rearranging the bits. It allows the error correction

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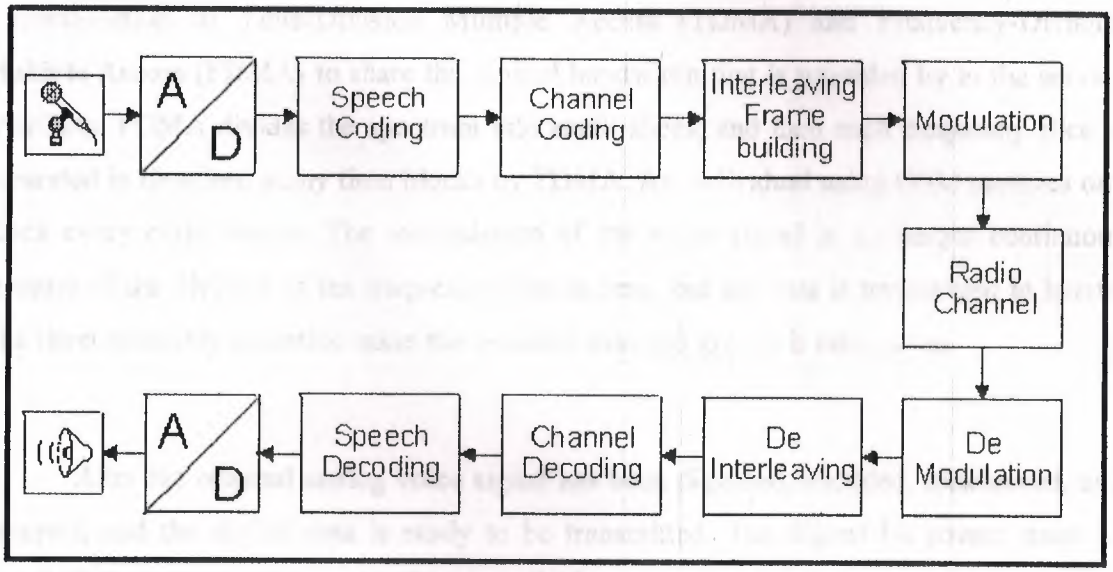

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algorithms to correct more of the errors that could have occurred during transmission. By interleaving the code, there is less possibility that a whole burst of bits is lost.

GSM allows many users to use their cell phones at the same time. The standard uses a combination of Time-Division Multiple Access (TDMA) and Frequency-Division Multiple Access (FDMA) to share the limited bandwidth that is provided by to the service providers. FDMA divides the spectrum into small slices, and then each frequency slice is separated in time into many time blocks by TDMA. An individual using GSM receives one block every eight blocks. The transmission of the voice signal is no longer continuous because of the division of the frequency slice in time, but the data is transmitted in bursts. The burst assembly operation takes the encoded data and groups it into bursts.

After the original analog voice signal has been digitized, encoded, interleaved, and grouped, and the digital data is ready to be transmitted. The digital bit stream must be encoded in a pulse and transmitted over radio frequencies. Modulation changes the '1' and '0's in a digital representation to another representation that is more suitable for transmission over airwaves.

3. DIGITAL RADIO SYSTEMS

3.1 The Advantages of Digital Transmission

Only a limited number of valid (predetermined) signal levels are transmitted. This permits:

- Waveform regeneration (for point-to-point, or line systems) - hence no cumulative distortion with distance.
- Satisfactory operation with lower carrier/noise levels than for analogue systems
- Readily permits Time Division Multiplexing (TDM) which has a greater transmission economy than Frequency Division Multiplexing used for analogue systems (filter guard-bands are replaced by time-slot guard periods)
- Greater large scale economy (VLSI)

3.1.1 Digital Radio Systems

Digital modulation used in communication systems provides several, benefits over analogue methods

Increased capacity - some analysts say 5 to 10 times more efficient than analogue hence greater capacity. Due to:

Multiple carriers per channel voice activity detection demand assignment

Increased security/privacy - digital speech can be easily encrypted **Additional services** (ISDN) - fax, data, etc.,

Reduced risk of fraud - as digital systems are more secure

Common system: - international standards allow international roaming

Spectrum - digital signals are less susceptible to interference from nearby cells than analogue; thus cell suites (clusters) may be closer - better frequency reuse by a factor of between two and four.

Traffic - improvements in call capacity over analogue networks means that there is less network delay and hence fewer channel busy conditions.

Greater compatibility with digital networks (especially ISDN), intelligent network Compatibility

Data - enrol, correction (FEC and ARQ) facilitates data transmission

Quality - slightly better radio performance - resistance to noise and cross talk. Digital systems can mask problems in the transmission channel such as deep fades and multi-path. Error correction and regeneration.

Flexibility

- Software reconfigurable.
- Smart card technology
- Generic data packets for all types of service

3.2 GSM Pan European Digital Cellular Radio

Originally known as Group Special Mobile, now known as Global System for Mobile communications. GSM was initially set up under the CEPT but was developed under the control of ETSI. A fundamental requirement of GSM was that it provided pan European roaming. A second requirement was that it provides a more efficient use of the spectrum than the first generation cellular systems (analogue).

GSM uses the following frequency bands:

Mobile Transmit 905 - 915 MHz and 872 - 888 (additional requested)

Base Transmit 935 - 960 MHz and 917 - 933.

Channel Bandwidth = 200 kHz - (compare with 25 kHz for TACS) but eight stations share each channel on a time division multiplex basis.

One or several Base Station Systems are connected to the Mobile Switching Centre (MSC)

The MSC is similar to an ISDN exchange and is the interconnection point to the ISDN/PSTN.

Mobiles are classified as **follows:**

Class 1	20 W peak (for vehicle use)
Class 2	8 W peak "
Class 3	5 W peak (hand/portable)
Class 4	2 W peak "
Class 5	0.85 W peak (small handsets)

As with TACS the actual mobile transmit power is controlled by the base station

Base Station Transmitter Power varies between 7 W and 45 W (100 W ERP)

Additional features of the GSM system:

ISDN capability with full ISDN numbering, but the data rate is limited to 9.6 kbits/s (not 64 kbits/s) so Fax messages are group 3 not group 4 and the Bit Error Rate may slow fax operation.

The Mobile Station also contains a removable Subscriber Identity Module (SIM) which is used to identify and validate the user, as well as storing some of the user's operating parameters such as his number. The use of the SIM means that a subscriber may use various different handsets at different times, with all calls being billed to his account. Certain limited facilities should be available from a mobile without the SIM fitted, e.g. access to emergency services.

The GSM system is designed for a maximum range (maximum cell size) of 35 km. The performance of all types of cellular radio systems is primarily restricted by co-channel interference. Digital modulation can tolerate a much higher level of co-channel interference than analogue systems and still produce an acceptable voice quality - this results in an increase in spectral efficiency.

The radio transmission method chosen for GSM is a Narrow-Band Time Division Multiple Access structure. Eight stations share a 200 kHz bandwidth radio channel. The mobile's transmitter and receiver are allocated different carrier frequencies (45 MHz separation) and differing time slots, therefore the transmitter and receiver do not operate simultaneously (as with TACS) and diplexers are not needed.. (Diplexers are very high performance filters that protect the receiver input from being swamped by the output of the transmitter). Diplexers are needed for the base stations, as they have to transmit and receive simultaneously.

The further the mobile is from the base station the longer the signals take to arrive. TDMA requires accurate synchronisation of the transmissions from the mobiles in order to ensure that signals do not overlap at the base station. The transmission times are calculated

at the base station and synchronisation information is sent to the mobile. Without this procedure it would be necessary to allow relatively long guard times.

Speech is coded into a data stream of 13 kbits/s (compare with 64 kbits/s for PCM). An option allows for half of the 13 kbits/s coding rate, which will double the spectrum efficiency and the system capacity.

The coded speech is divided into 20 ms blocks of 260 bits each, and the first 182 bits of each block are extended to 189 by adding parity and spare bits. These are then encoded for error detection and correction purposes by means of a convolutional code which adds a further 189 bits, and then recombined with the remaining 78 bits to give a total block length of 456 bits. The bits which are error protected are those for which accuracy is most critical for the voice coder.

The 456 bit length blocks are then divided into eight sub-blocks of 57 bits which are interleaved with sub-blocks from adjacent blocks - in order to protect against bursts of errors. This is to help combat the effects of any burst errors and deep fades. Pairs of sub-blocks are combined with the 26 bit equaliser training sequence, and two control bits plus three header and three tail bits, into a time slot of 148 bits. Guard times equivalent to 8.25 bits are allowed between frames. Time slots from each of the eight mobiles are combined sequentially into a TDMA frame of 4.615 ms (each of the 8 time slots last for 576.92 microsecs). Twenty-four of the TDMA frames are combined with two supervisory and control frames which have a channel associated structure and contain the signalling information for eight mobiles. The overall bit rate is 270 kbits/s.

3.3 Aspects of the GSM Network

When a mobile is switched on three actions take place: it registers with the network, is authenticated and its location is updated. Cell reselection takes place automatically as it moves across cell boundaries and the network continuously tracks the mobile at all times.

Thus the network is able to route incoming calls quickly and efficiently, and allow or deny outgoing calls, whilst calls in progress can continue without interruption as the mobile crosses cell boundaries.

The GSM radio interface uses both Frequency Division Multiple Access (FDMA) - as does TACS, and Time Division Multiple Access (TDMA) as well as a Slow Frequency Hopping (SFH) scheme. Each radio channel has a bandwidth of 200 kHz and each cell is allocated a number of radio channels from the GSM bands according to the traffic requirements. The basic frequency bands initially allocated to GSM are two sub-bands of 25 MHz each:

890 - 915 MHz (Mobile Transmit) and, 935 - 960 MHz (Base Transmit).

Note that 45 MHz separates the bands, as with TACS.

In addition there are other bands, which were initially allocated for analogue cellular (TACS) which have been earmarked for GSM expansion (872 - 888 - mobile Tx and 917 - 933 - Base Tx).

The modulation scheme use by GSM results in a spectrum which is somewhat greater than the 200 kHz per channel, allocated. So careful planning is required in order to minimise adjacent channel interference. Thus there are 125 channels, each 200 kHz wide, operated on a Frequency Division Multiple Access duplex basis. Each channel is multiplexed on a Time Division Multiple Access basis, with 8 different time slots per radio channel. Each channel time slot provides for individual 13 kbit/s speech (or soon two 6.5 kbit/s speech channels) or data at 9.6 kbit/s. Thus each channel may be used "simultaneously" by eight mobiles operating with a full channel system, or 16 mobiles operating with a half channel system.

Currently, the two 25 MHz bands allocated for GSM will support 1000 mobiles. This is similar to the case for TACS. In TACS, each 25 kHz wide channel was allocated to an individual mobile. Hence 1000 mobiles would occupy 25 MHz. Therefore, one might ask how is GSM able to use the spectrum more efficiently than TACS? The answer lies with the fact that the GSM system is able to operate with a much lower Carrier to Interference (C/I) ratio than TACS. Typical figures are 10 dB for GSM compared with 18 dB for TACS. This 8 dB difference represents a factor of 6 or 7 times. The result is that GSM is able to use a much smaller frequency re-use distance and thus smaller cells may be used which results in a much higher system capacity.

3.4 Equalisation and Multi-Path Propagation

One of the effects of multi-path is that delayed signals combine to give a composite signal at the antenna of the mobile receiver and the amplitude and phase of the composite signal undergoes wide and rapid fluctuations. The delay may be serious enough to cause serious inter-symbol interference, which can limit the maximum usable bit rate.

In the GSM system the long excess delays can only be tolerated by using an **equaliser** in the receiver. A known bit pattern of 26 bits (called a **training sequence**) is transmitted at regular intervals. The equaliser compares the received bit pattern with the training sequence and adjusts the parameters of a digital filter so as to compensate for the effects of multipath. Because with a fast moving vehicle the geometry of the path can change rapidly, the training sequence is included on every time slot.

3.5 Digital Modulation.

3.5.1 Frequency Shift Keying:

Carrier is shifted to specific frequencies, one for each symbol

(e.g. TACS: logic I = fc + 6.4 kHz, logic 0 = fc-6.4 kHz)

a) Advantages:

- Constant amplitude
- No need to adjust the demodulation threshold as fading occurs - as would be the case for amplitude based modulation systems

b) Minimum Shift Keying

- A form of FSK which uses continuous phase
- Frequencies are spaced so that during the symbol interval one symbol goes through 180 degrees more phase shift than the other
- The result is a maximum phase difference at the end of each symbol interval with a minimum of change in frequency
- Phase continuity is maintained at signalling transitions
- Advantage: a compact spectrum

c) GMSK modulation

The GSM specifications require the out-of-band radiated power in the adjacent channels to be between 40 and 70 dBs below that of the desired channel. To satisfy this requirement, it is necessary to bandlimit the spectrum of the R-F output signal. This cannot be achieved easily at the final RF stage in multi-channel transceivers. In order to comply with the spectrum requirements narrow band digital modulation schemes must be used. Amongst these schemes, Minimum Shift Keying, (covered above) has proved to provide a better modulation for use on bandlimited and non-linear channels. It has the following properties:

- Constant amplitude envelope
- Relative narrow bandwidth
- Coherent/non-coherent detection capabilities

- A good bit-error-rate performance

However, MSK does not completely satisfy the out-of-band radiation requirements. To overcome this limitation a pre-modulation filter is used. A filter with a Gaussian response is suitable for this role and the modified form of modulation is termed Gaussian MSK (GMSK). The filter is included between the NRZ data stream from the voice processor and the MSK modulator stage.

The Gaussian filter should have the following properties:

- Narrow bandwidth and sharp cut-off (to suppress high frequency components)
- Low overshoot response (to protect against excessive instantaneous frequency deviation)
- The same phase differences per symbol interval as MSK
- The same phase continuity during signal transitions as MSK.

GMSK is a form of FM. The carrier is therefore at a constant amplitude during transmission, allowing the use of low cost, efficient transmitters (class C bias).

3.5.2 Frequencies, Bursts, Time and Slots

There are 125 frequencies available in the main GSM bands and these frequencies are distributed over the different cells in a way that minimises interference and matches traffic distribution. All communications over the radio/air interface consist of packetised bursts of data which have to fit into a defined time and frequency window called a slot. A slot is 577 μs long and 200 kHz wide.

The basic TDMA frame consists of eight successive slots (16 in the case of the half rate voice coding system) therefore any given frequency may be used to carry traffic to, or from, eight different mobiles. The time slots are numbered TNO to TN7.

There are two different types of channels:

- Dedicated channels, which carry communications between a particular mobile and the base station, and are used only when the mobile is actively engaged in a call
- Common channels, which are used by all mobiles both during the idle state and during calls, and are predominantly used for control purposes.

The common channels are always carried on a fixed frequency for a given cell known as the

Beacon frequency. Time slot zero (TNO) on the beacon frequency includes synchronisation information to help the mobiles lock onto the system. The unique presence of the synchronisation signals defines and identifies TNO. Additional common channel capacity may be provided in other TNs on the same frequency.

Table 3.1 Slot used for common channels, other slots are used for dedicated channels

TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0

This white paper is intended for global system for mobile communications (GSM) operators interested in understanding Ericsson's view on how enhanced data for global evolution (EDGE) can play an important role in the evolution toward wideband code division multiple access (WCDMA). EDGE can be introduced in two ways: (1) as a packet-switched enhancement for general packet radio service (GPRS), known as enhanced GPRS or EGPRS, and (2) as a circuit-switched data enhancement called enhanced circuit-switched data (ECSD). This white paper, however, will only discuss the packet-switched enhancement, EGPRS. The purpose of this white paper is to describe EDGE technology and how it leverages existing GSM systems and complements WCDMA for further growth. The benefits described here are based on Ericsson's vision of one seamless network for GSM and WCDMA. The white paper is based on Ericsson's current experience with operators' deployment processes, our past experience with technology transitions and our expertise with all major wireless standards, including GPRS, EDGE and WCDMA.

3.6 Executive Summary

EDGE is the next step in the evolution of GSM and IS-136. The objective of the new technology is to increase data transmission rates and spectrum efficiency and to facilitate new applications and increased capacity for mobile use. With the introduction of EDGE in GSM phase 2+, existing services such as GPRS and high-speed circuit switched data (HSCSD) are enhanced by offering a new physical layer. The services themselves are not modified. EDGE is introduced within existing specifications and descriptions rather than by creating new ones. This white paper focuses on the packet-switched enhancement for GPRS, called EGPRS. GPRS allows data rates of 115 kbps and, theoretically, of up to 160 kbps on the physical layer. EGPRS is capable of offering data rates of 384 kbps and, theoretically, of up to 473.6 kbps. A new modulation technique and error-tolerant transmission methods, combined with improved link adaptation mechanisms, make these EGPRS rates possible. This is the key to increased spectrum efficiency and enhanced applications, such as wireless Internet access, e-mail and file transfers. GPRS/EGPRS will be one of the pacesetters in the overall wireless technology evolution in conjunction with WCDMA. Higher transmission rates for specific radio resources enhance capacity by enabling more traffic for both circuit- and packet-switched services. As the Third-generation Partnership Project (3GPP) continues standardization toward the GSM/EDGE radio access network (GERAN), GERAN will be able to offer the same services as WCDMA by connecting to the same core network. This is done in parallel with means to increase the spectral efficiency. The goal is to boost system capacity, both for real-time and best effort services, and to compete effectively with other third-generation radio access networks such as WCDMA and cdma2000.

3.7 Technical Differences between GPRS and EGPRS

Regarded as a subsystem within the GSM standard, GPRS has introduced packet-switched data into GSM networks. Many new protocols and new nodes have been introduced to make this possible. EDGE is a method to increase the data rates on the radio link for GSM. Basically, EDGE only introduces a new modulation technique and new

channel coding that can be used to transmit both packet-switched and circuit-switched voice and data services. EDGE is therefore an add-on to GPRS and cannot work alone. GPRS has a greater impact on the GSM system than EDGE has. By adding the new modulation and coding to GPRS and by making adjustments to the radio link protocols, EGPRS offers significantly higher throughput and capacity. EGPRS introduces changes to GPRS only on the base station system part of the network. GPRS and EGPRS have different protocols and different behavior on the base station system side. However, on the core network side, GPRS and EGPRS share the same packet-handling protocols and, therefore, behave in the same way. Reuse of the existing GPRS core infrastructure (serving GRPS support node/gateway GPRS support node) emphasizes the fact that EGPRS is only an "add-on" to the base station system and is therefore much easier to introduce than GPRS. GPRS Protocol

In addition to enhancing the throughput for each data user, EDGE also increases capacity. With EDGE, the same time slot can support more users. This decreases the number of radio resources required to support the same traffic, thus freeing up capacity for more data or voice services. EDGE makes it easier for circuit-switched and packet-switched traffic to coexist while making more efficient use of the same radio resources. Thus in tightly planned networks with limited spectrum, EDGE may also be seen as a capacity booster for the data traffic.

3.7.1 EDGE Technology

EDGE leverages the knowledge gained through use of the existing GPRS standard to deliver significant technical improvements. GPRS and EDGE: A Comparison of technical data. (Legend 8PSK, 8-phase shift keying; GMSK, Gaussian minimum shift keying

Figure compares the basic technical data of GPRS and EDGE. Although GPRS and EDGE share the same symbol rate, the modulation bit rate differs. EDGE can transmit three

times as many bits as GPRS during the same period of time. This is the main reason for the higher EDGE bit rates. The differences between the radio and user data rates are the result of whether or not the packet headers are taken into consideration. These different ways of calculating throughput often cause misunderstanding within the industry about actual throughput figures for GPRS and EGPRS. The data rate of 384 kbps is often used in relation to EDGE. The International Telecommunications Union (ITU) has defined 384 kbps as the data rate limit required for a service to fulfill the International Mobile Telecommunications-2000 (IMT-2000) standard in a pedestrian environment. This 384 kbps data rate corresponds to 48 kbps per time slot, assuming an eight-time slot terminal.

3.7.2 EDGE Modulation Technique

The modulation type that is used in GSM is the Gaussian minimum shift keying (GMSK), which is a kind of phase modulation. This can be visualized in an I/Q diagram that shows the real (I) and imaginary (Q) components of the transmitted signal (Figure 3). Transmitting a zero bit or one bit is then represented by changing the phase by increments of + _ p. Every symbol that is transmitted represents one bit; that is, each shift in the phase represents one bit. Figure 3. I/Q diagram showing EDGE modulation benefits. To achieve higher bit rates per time slot than those available in GSM/GPRS, the modulation method requires change. EDGE is specified to reuse the channel structure, channel width, channel coding and the existing mechanisms and functionality of GPRS and HSCSD. The modulation standard selected for EDGE, 8-phase shift keying (8PSK), fulfills all of those requirements. 8PSK modulation has the same qualities in terms of generating interference on adjacent channels as GMSK. This makes it possible to integrate EDGE channels into an existing frequency plan and to assign new EDGE channels in the same way as standard GSM channels. The 8PSK modulation method is a linear method in which three consecutive bits are mapped onto one symbol in the I/Q plane. The symbol rate, or the GPRS: GMSK Modulation GPRS: 8PSK Modulation

GPRS EDGE

- Modulation GMSK 8-PSK/GMSK
- Symbol rate 270 ksym/s 270 ksym/s
- Modulation bit rate **270** kb/s **810** kb/s
- Radio data rate per time slot 22,8 kb/s 69,2 kb/s
- User data rate per time slot **20** kb/s (CS4) **59,2** kb/s (MCS9)
- User data rate (8 time slots) 160 kb/s 473,6 kb/s (182,4 kb/s) (553,6 kb/s)

Time remains the same as for GMSK, but each symbol now represents three bits instead of one. The total data rate is therefore increased by a factor of three. The distance between the different symbols is shorter using 8PSK modulation than when using GMSK. Shorter distances increase the risk for misinterpretation of the symbols because it is more difficult for the radio receiver to detect which symbol it has received. Under good radio conditions, this does not matter. Under poor radio conditions, however, it does. The "extra" bits will be used to add more error correcting coding, and the correct information can be recovered. Only under very poor radio environments is GMSK more efficient. Therefore the EDGE coding schemes are a mixture of both GMSK and 8PSK.

3.8 Coding Schemes

For GPRS, four different coding schemes, designated CS1 through CS4, are defined. Each has different amounts of error-correcting coding that is optimized for different radio environments. For EGPRS, nine modulation coding schemes, designated MCS1 through MCS9, are introduced. These fulfill the same task as the GPRS coding schemes. The lower four EGPRS coding schemes (MSC1 to MSC4) use GMSK, whereas the upper five (MSC5 to MSC9) use 8PSK modulation. Figure 2.1 shows both GPRS and EGPRS coding schemes, along with their maximum throughputs. GPRS user throughput reaches saturation at a maximum of 20 kbps with CS4, whereas the EGPRS bit rate continues to increase as the radio quality increases, until throughput reaches saturation at 59.2 kbps. Both GPRS CS1 to CS4 and EGPRS MCS1 to MCS4 use GMSK modulation with slightly different throughput performances. This is due to differences in the header

size (and payload size) of the EGPRS packets. This makes it possible to resegment EGPRS packets. A packet sent with a higher coding scheme (less error correction) that is not properly received, can be retransmitted with a lower coding scheme (more error correction) if the new radio environment requires it. This resegmenting (retransmitting with another coding scheme) requires changes in the payload sizes of the radio blocks, which is why EGPRS and GPRS do not have the same performance for the GMSK modulated coding schemes. Re-segmentation is not possible with GPRS.

3.9 Packet Handling

Another improvement that has been made to the EGPRS standard is the ability to retransmit a packet that has not been decoded properly with a more robust coding scheme. For GPRS, re-segmentation is not possible. Once packets have been sent, they must be retransmitted using the original coding scheme even if the radio environment has changed. This has a significant impact on the throughput, as the algorithm decides the level of confidence with which the link adaptation (LA) must work.

Below is an example of packet transfer and retransmission for GPRS.

A. The GPRS terminal receives data from the network on the downlink. Due to a GPRS measurement report that was previously received, the link adaptation algorithm in the base station controller decides to send the next radio blocks (e.g., numbers 1 to 4) with CS3. During the transmission of these packages, the carrier-to-interference ratio (C/I) decreases dramatically, changing the radio environment. After the packets have been transmitted, the network polls for a new measurement report, including the acknowledged/unacknowledged bitmap that tells the network which radio blocks were received correctly.

B. The GPRS handset replies with a packet downlink acknowledged/unacknowledged message containing the information about the link quality and the bitmap. In this scenario, it is assumed that packets 2 and 3 were sent erroneously.

C. Based on the new link quality information, the GPRS link adaptation algorithm will adapt the coding scheme to the new radio environment using CS1 for the new packets 5 and 6. However, because GPRS cannot resegment the old packets, packets 2 and 3 must be retransmitted using CS3, although there is a significant risk that these packets still may not be decoded correctly. As a result, the link adaptation for GPRS requires careful selection of the coding scheme in order to avoid retransmissions as much as possible. With EGPRS, resegmentation is possible. Packets sent with little error protection can be retransmitted with more error protection, if required by the new radio environment. The rapidly changing radio environment has a much smaller effect on the problem of choosing the wrong coding scheme for the next sequence of radio blocks because resegmentation is possible. Therefore, the EGPRS link-controlling algorithm can be very aggressive when selecting the modulation coding schemes.

3.10 Addressing Window

Before a sequence of coded radio link control packets or radio blocks can be transmitted over the Um (radio) interface, the transmitter must address the packets with an identification number. This information is then included in the header of every packet. The packets in GPRS are numbered from 1 to 128. After transmission of a sequence of packets (e.g., 10 packets), the transmitter asks the receiver to verify the correctness of the packets received in the form of an acknowledged/unacknowledged report. This report informs the transmitter which packet or packets were not successfully decoded and must be retransmitted. Since the number of packets is limited to 128 and the addressing window is 64, the packet sending process can run out of addresses after 64 packets. If an erroneously decoded packet must be retransmitted, it may have the same number as a new packet in the queue. If so, the protocol between the terminal and the network stalls, and all the packets belonging to the same low-layer capability frame must be retransmitted. In EGPRS, the addressing numbers have been increased to 2048 and the window has been increased to

1024 in order to minimize the risk for stalling. This, in turn, minimizes the risk for re-transmitting low-layer capability frames and prevents decreased throughput

3.11 Measurement Accuracy

As in the GSM environment, GPRS measures the radio environment by analyzing the channel for carrier strength, bit error rate, etc. Performing these measurements takes time for a mobile station, which is of no concern in the speech world as the same coding is used all the time. In a packet-switched environment, it is essential to analyze the radio link quickly in order to adapt the coding toward the new environment. The channel analysis procedure that is used for GPRS makes the selection of the right coding scheme difficult since measurements for interference are performed only during idle bursts. As a result, measurements can only be performed twice during a 240-millisecond period. For EGPRS, the standard does not rely on the same "slow" measurement mechanism. Measurements are taken on each and every burst within the equalizer of the terminal, resulting in an estimate of the bit error probability (BEP). Estimated for every burst, the BEP is a reflection of the current C/I, the time is persion of the signal and the velocity of the terminal. The variation of the BEP value over several bursts will also provide additional information regarding velocity and frequency hopping.

A very accurate estimation of the BEP is then possible to achieve.

A mean BEP is calculated per radio block (four bursts) as well as the variation (standard deviation of the BEP estimation divided by the mean BEP) over the four bursts. These results are then filtered for all radio blocks sent within the measurement period.

This results in highly accurate measurements even during short measurement periods. Short measurement periods, in turn, enable quick reaction to changes in the radio

environment. It is therefore possible to achieve a better and more flexible link adaptation for EGPRS.

3.12 Interleaving

To increase the performance of the higher coding schemes in EGPRS (MCS7 to MCS9) even at low C/I, the interleaving procedure has been changed within the EGPRS standard.

When frequency hopping is used, the radio environment is changing on a per-burst level. Because a radio block is interleaved and transmitted over four bursts for GPRS, each burst may experience a completely different interference environment. If just one of the four bursts is not properly received, the entire radio block will not be properly decoded and will have to be retransmitted. In the case of CS4 for GPRS, hardly any error protection is used at all. With EGPRS, the standard handles the higher coding scheme differently than GPRS to combat this problem. MCS7, MCS8 and MCS9 actually transmit two radio blocks over the four bursts, and the interleaving occurs over two bursts instead of four. This reduces the number of bursts that must be retransmitted should errors occur. The likelihood of receiving two consecutive error-free bursts is higher than receiving four consecutive error-free bursts. This means that the higher coding schemes for EDGE have a better robustness with regard to frequency hopping.

3.13 EGPRS Link Controlling Function

To achieve the highest possible throughput over the radio link,

Retransmission of lost Block necessary Retransmission of second half only Figure 7. Interleaving. (Legend: CS, coding scheme; EGPRS, enhanced GPRS; MCS, modulation-coding scheme)

EGPRS uses a combination of two functionalities: link adaptation and incremental redundancy. Compared to a pure link adaptation solution, this combination of mechanisms significantly improves performance.

3.14 Link Adaptation

Link adaptation uses the radio link quality, measured either by the mobile station in a downlink transfer or by the base station in an uplink transfer, to select the most appropriate modulation coding scheme for transmission of the next sequence of packets. For an uplink packet transfer, the network informs the mobile station which coding scheme to use for transmission of the next sequence of packets. The modulation-coding scheme can be changed for each radio block (four bursts), but a change is usually initiated by new quality estimates. The practical adaptation rate is therefore decided by the measurement interval. There are three families: A, B and C. Within each family, there is a relationship between the payload sizes, which makes resegmentation for retransmissions possible.

3.15 Incremental Redundancy

Incremental redundancy initially uses a coding scheme, such as MCS9, with very little error protection and without consideration for the actual radio link quality. When information is received incorrectly, additional coding is transmitted and then soft combined in the receiver with the previously received information. Soft combining increases the probability of decoding the information. This procedure will be repeated until the information is successfully decoded. This means that information about the radio link is not necessary to support incremental redundancy. For the mobile stations, incremental redundancy support is mandatory in the standard.

3.16 Impact of EGPRS on Existing

3.16.1 GSM/GPRS Networks

Due to the minor differences between GPRS and EGPRS, the impact of EGPRS on the existing GSM/GPRS network is limited to the base station system. The base station is affected by the new transceiver unit capable of handling EDGE modulation as well as new software that enables the new protocol for packets over the radio interface in both the base station and base station controller. The core network does not require any adaptations. Due to this simple upgrade, a network capable of EDGE can be deployed with limited investments and within a short time frame. Use same MCS for retransmissions MCS9 becomes more robust than MCS5 for similar bit rate

3.16.2 Standardization

a) Background

Standardization of the first releases of the third generation cellular systems that comply with ITU/IMT-2000 requirements has now been finalized with European Telecommunications Standards Institute (ETSI/3GPP) Release 99. Two such major systems are Universal Mobile Telecommunications System (UMTS) and GSM/EDGE.

b) Fulfilling the EDGE Standardization

EDGE standardization can be divided in three areas:

- Standardization of the physical layer changes (definition of the modulation and coding schemes)
- The protocol changes for ECSD and
- EGPRS.

c) EDGE standard and references

The EDGE base station system work item provides a platform to employ new modulation techniques, whereas the EDGE network support subsystem work item defines the network changes to facilitate the physical layer. According to the work item descriptions, EDGE will provide two phases: Phase 1: Single- and multi-slot packet-switched services and single and multi-slot circuit switched services. Phase 2: Real-time services employing the new modulation techniques that are not included in Phase 1. Phase 1 has been completed with 3GPP Release 99. Phase 2 is ongoing in the 3GPP standardization, and its scope has been extended to cover the alignment with WCDMA and the provisioning of Internet protocol (IP) multimedia. This concept, currently standardized in 3GPP, is known as GERAN.

d) Requirements on EDGE

From the beginning, the standardization of EDGE was restricted to the physical layer and to the introduction of a new modulation scheme. Since EDGE was intended as an evolution of the existing GSM radio access technology, the requirements were set accordingly:

- EDGE- and non-EDGE-capable mobile stations should be able to share one and the same time slot.
- EDGE- and non-EDGE-capable transceivers should be deployable in the same spectrum.
- A partial introduction of EDGE should be possible. To ease implementation of new terminals while taking into account the asymmetrical characteristic of most services currently available, it was also decided that two classes of terminals should be supported by the EDGE standard:
- A terminal that provides 8PSK capability in the downlink only, and
- A terminal that provides 8PSK in the uplink and downlink.

e) Service Aspects

The introduction of EGPRS enables bit rates that are approximately three times higher than standard GPRS bit rates. Within the EDGE work item, this was simply handled by reusing the GPRS quality of service (QoS) profiles and extending the parameter range to reflect the higher bit rates, or in other words, introducing higher throughput values.

f) Architecture

EGPRS does not bring about any direct architecture impacts (see GSM 03.60). The packet control unit may still be placed either in the base station, the base station controller or the GPRS support node, and the central control unit is always placed in the base station. However, since the radio link control automatic repeat request function on the network side is located in the packet control unit, any delay introduced between the PCU and the radio interface will directly affect the radio link control acknowledged/ unacknowledged round-trip times. This, in turn, results in a higher risk of stalling the radio link control protocol. To mitigate this risk and to allow the operator to optimize network behavior, the maximum radio link control automatic repeat request window size has been extended for EGPRS.

g) User plane protocols

The transmission plane protocol structure for GPRS is shown in **Figure 11**. The protocols that are influenced by the introduction of EDGE are shaded. The protocols closest to the physical layer (the radio link control and mobile allocation channel) are most affected by EDGE (see GSM 04.60). There also are some minor modifications to the base station system GPRS protocol. Apart from these changes, the rest of the protocol stack remains intact after the introduction of EDGE.

h) Control plane protocols and channels

The introduction of EGPRS also has an impact on these control plane layers: mobility management and radio resource management. There is no impact on session

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GSM RADIO INTERFACE

Graduation Project
EE-400

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Finally, I want to thank my family ,especially My Parents, because without there support, love and encouragement, it was impossible for me to achieve my current position, and what I am today or what I will be tomorrow I will be because of them so all my love goes to them with every happiness of life.

ABSTRACT

GSM happens to be one of the most popular cellular radio systems implemented in the world today. Now GSM has gathered more than 22 million subscribers only in Europe. It provides a wide range of services and facilities, both voice and data, that are compatible with the existing fixed Public Switched telephone Networks (PSTN), Public Switched Data Networks (PSDN), Public Land Mobile Networks (PLMN) and Integrated Service Digital Networks (ISDN).

It's main function is to give compatibility considered mobile subscriber the access to any mobile subscriber in any country, which operates the system, and provides facilities for automatic roaming, locating and updating the mobile subscriber's status.

GSM as a modern telecommunication system is a complex object. It's implementation and operation is not an easy task, neither easy it's description.

GSM Radio Interface is the interface between the mobile stations and the fixed infrastructure. It is one of the most important interfaces in the GSM system. The specification of the radio interface has then as important influence on the spectrum efficiency.

INTRODUCTION

In this project Global Systems for Mobile is introduced and specifically GSM Radio Interface has been considered. This project consists of five chapters.

Chapter 1 is an introduction to GSM, history and architecture of GSM, working principle of GSM, its communication specifications, coding and modulations.

Chapter 2 is mainly focused on Radio Transmission and GSM Transmission Systems.

Chapter 3 is a brief description of Digital Radio Systems, containing information's about advantages of radio transmission, GSM Pan European Digital Cellular Radio, Aspects of The GSM Networks, Equalization and Multipath Propagation, Digital Modulation, Technical Difference Between GPRS and EGPICS, Coding Schemes, Packet Handling, Addressing Window, Measurement Accuracy, Interleaving, EGPRS Link Controlling Function, Link Adaptation, Incremental Reducing, Impact of EGPRS, GSM/GPRS Networks, Standardization, GERAN System Architecture.

Chapter 4 is a short description of Encryptions, Limitations of Security, The Functions of The Services, Anonymity including Authentications and User Data Signalling Protection, Implementation and Roaming and World Wide Use of The Algorithms.

Chapter 5 presents a short introduction of Power Control, Frequency Hopping, Measurement Campaign, Statistic Trials, Specific Trials, Measurements Results, Bit Error Rate and Interfacing BCCH

1. INTRODUCTION

1.1 History of G.S.M

During the early 1980s, analog cellular telephone systems were experiencing rapid growth in Europe, particularly in Scandinavia and the United Kingdom, but also in France and Germany. Each country developed its own system, which was incompatible with everyone else's in equipment and operation. This was an undesirable situation, because not only was the mobile equipment limited to operation within national boundaries, which in a unified Europe were increasingly unimportant, but there was a very limited market for each type of equipment, so economies of scale, and the subsequent savings, could not be realized.

In 1981 a joint Franco German study was initiated to develop a common approach which, it was hoped, would become a standard for Europe. Soon after, in 1982 a proposal from Nordic Telecom and Netherlands PTT to the CEPT (Conference of European Post and Telecommunications) to develop a new digital cellular standard that would cope with the ever burgeoning demands on European mobile networks. Then a study group formed called the Global System for Mobile (GSM) to study and develop a pan-European public land mobile system. The proposed system had to meet certain criteria:

- Good subjective speech quality
- Low terminal and service cost
- Support for international roaming
- Ability to support handheld terminals
- Support for range of new services and facilities
- Spectral efficiency
- ISDN compatibility

In 1989, GSM responsibility was transferred to the European Telecommunication Standards Institute (ETSI), and phase I of the GSM specifications was published in 1990. Commercial service was started in mid-1991, and by 1993 there were 36 GSM networks in 22 countries. Although standardized in Europe, GSM is not only a European standard. Over

200 GSM networks (including DCS1800 and PCS1900) are operational in 110 countries around the world. In the beginning of 1994, there were 1.3 million subscribers worldwide, which had grown to more than 55 million by October 1997. With North America making a delayed entry into the GSM field with a derivative of GSM called PCS1900, GSM systems exist on every continent, and the acronym GSM now aptly stands for Global System for Mobile communications.

The developers of GSM chose an unproven (at the time) digital system, as opposed to the then-standard analog cellular systems like AMPS in the United States and TACS in the United Kingdom. They had faith that advancements in compression algorithms and digital signal processors would allow the fulfillment of the original criteria and the continual improvement of the system in terms of quality and cost. The over 8000 pages of GSM recommendations try to allow flexibility and competitive innovation among suppliers, but provide enough standardization to guarantee proper interworking between the components of the system. This is done by providing functional and interface descriptions for each of the functional entities defined in the system.

The original French name was later changed to Global System for Mobile Communications, but the original GSM acronym stuck.

1.2 G.S.M System Architecture

A GSM network is composed of several functional entities, whose functions and interfaces are defined. The GSM network can be divided into three broad parts. The Mobile Station is carried by the subscriber, the Base Station Subsystem controls the radio link with the Mobile Station. The Network Subsystem, the main part of which is the Mobile services Switching Centre, performs the switching of calls between the mobile and other fixed or mobile network users, as well as management of mobile services, such as authentication. With the Operations and Maintenance centre, which oversees the proper operation and setup of the network. The Mobile Station and the Base Station Subsystem communicate

across the Um interface, also known as the air interface or radio link. The Base Station Subsystem communicates with the Mobile service Switching Center across the A interface.

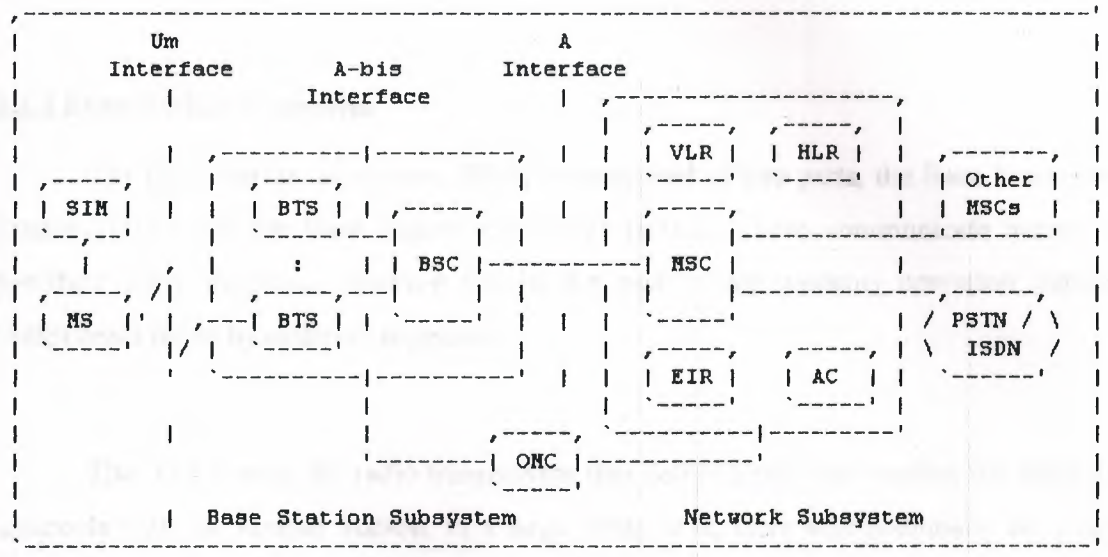

Figure 1.1 GSM Architecture Illustrated

1.2.1 Mobile Station

The mobile station (MS) consists of the physical equipment, such as the radio transceiver, display and digital signal processors, and a smart card called the Subscriber Identity Module (SIM). The SIM provides personal mobility, so that the user can have access to all subscribed services irrespective of both the location of the terminal and the use of a specific terminal. By inserting the SIM card into another GSM cellular phone, the user is able to receive calls at that phone, make calls from that phone, or receive other subscribed services.

The mobile equipment is uniquely identified by the International Mobile Equipment Identity (IMEI). The SIM card contains the International Mobile Subscriber Identity

(IMSI), identifying the subscriber, a secret key for authentication, and other user information. The IMEI and the IMSI are independent, thereby providing personal mobility. The SIM card may be protected against unauthorized use by a password or personal identity number.

1.2.2 Base Station Subsystem

The Base Station Subsystem (BSS) is composed of two parts, the Base Transceiver Station (BTS) and the Base Station Controller (BSC). These communicate across the specified Abis interface, allowing (as in the rest of the system) operation between components made by different suppliers.

The BTS houses the radio transceivers that define a cell and handles the radio link protocols with the Mobile Station. In a large urban area, there will potentially be a large number of BTSs deployed. The requirements for a BTS are ruggedness, reliability, portability, and minimum cost. BTS is responsible for providing layers 1 and 2 of the radio interface, that is, an error-corrected data path. Each BTS has at least one of its radio channels assigned to carry control signals in addition to traffic.

The BSC manages the radio resources for one or more BTSs. It is responsible for the management of the radio resource within a region. Its main functions are to allocate and control traffic channels, control frequency hopping, undertake handovers (except to cells outside its region) and provide radio performance measurements. Once the mobile has accessed, and synchronized with, a BTS the BSC will allocate it a dedicated bi-directional signaling channel and will set up a route to the Mobile services Switching Center (MSC). The BSC also translates the 13 KBPS voice channel used over the radio link to the standard 64 KBPS channel used by the Public Switched Telephone Network or ISDN.

1.2.3 Network Subsystem

The central component of the Network Subsystem is the Mobile services Switching Center (MSC). It acts like a normal switching node of the PSTN or ISDN, and in addition

provides all the functionality needed to handle a mobile subscriber, such as registration, authentication, location updating, handovers, and call routing to a roaming subscriber. These services are provided in conjunction with several functional entities, which together form the Network Subsystem. The MSC provides the connection to the public fixed network (PSTN or ISDN), and signaling between functional entities uses the ITUT Signaling System Number 7 (SS7), used in ISDN and widely used in current public networks.

The Home Location Register (HLR) and Visitor Location Register (VLR), together with the MSC, provide the call routing and (possibly international) roaming capabilities of GSM. The HLR contains all the administrative information of each subscriber registered in the corresponding GSM network, along with the current location of the mobile. It also contains a unique authentication key and associated challenge/response generators. The current location of the mobile is in the form of a Mobile Station Roaming Number (MSRN), which is a regular ISDN number used to route a call to the MSC where the mobile is currently located. There is logically one HLR per GSM network, although it may be implemented as a distributed database.

The VLR contains selected administrative information from the HLR, necessary for call control and provision of the subscribed services, for each mobile currently located in the geographical area controlled by the VLR. Although each functional entity can be implemented as an independent unit, most manufacturers of switching equipment implement one VLR together with one MSC, so that the geographical area controlled by the MSC corresponds to that controlled by the VLR, simplifying the signaling required. Note that the MSC contains no information about particular mobile stations - this information is stored in the location registers.

The other two registers are used for authentication and security purposes. The Equipment Identity Register (EIR) is a database that contains a list of all valid mobile equipment on the network, where each mobile station is identified by its International

Mobile Equipment Identity (IMEI). An IMEI is marked as invalid if it has been reported stolen or is not type approved. The Authentication Center is a protected database that stores a copy of the secret key stored in each subscriber's SIM card, which is used for authentication and ciphering of the radio channel.

1.2.4 Operation and Management

A network having complexity like G.S.M service must be managed and maintained. This is the function of the Operations and Maintenance Center (OMC). GSM leaves decisions on O&M to the individual operators, but general guidelines are given which reinforce the move towards an overlaid telecommunications management network. In this approach five separate management functions are identified as:

- Operations and Performance Management
- Maintenance
- System Change Control
- Security Management
- Administration and Commercial Functionality

1.3 How It Works

1.3.1 Make Call

When the mobile user initiates a call, his equipment will search for a local base station, i.e. The BSS. Once the mobile has accessed, and synchronized with, a BTS the BSC will allocate it a dedicated bi-directional signaling channel and will set up a route to the Mobile services Switching Center (MSC).

1.3.2 Call initialization

When a mobile requests access to the system it has to supply its IMEI (International Mobile Equipment Identity). This is a unique number, which will allow the system to initiate a process to confirm that the subscriber is allowed to access it. This process is called

authentication. Before it can do this, however, it has to find where the subscriber is based. Every subscriber is allocated to a home network, associated with an MSC within that network. This is achieved by making an entry in the Home Location Register (HLR), which contains information about the services the subscriber is allowed.

Whenever a mobile is switched on and at intervals thereafter, it will register with the system; this allows its location in the network to be established and its location area to be updated in the HLR. A location area is a geographically defined group of cells. On first registering, the local MSC will use the IMSI to interrogate the subscriber's HLR and will add the subscriber data to its associated Visitor Location Register (VLR). The VLR now contains the address of the subscriber's HLR and the authentication request is routed back through the HLR to the subscriber's Authentication Center (AC). This generates a challenge/response pair, which is used by the local network to challenge the mobile. In addition, some operators also plan to check the mobile equipment against an Equipment Identity Register (EIR), in order to control stolen, fraudulent or faulty equipment.

1.3.3 Authentication

The authentication process is very powerful and is based on advanced cryptographic principles. It especially protects the network operators from fraudulent use of their services. It does not however protect the user from eavesdropping. The Time Division Multiple Access (TDMA) nature of GSM coupled with its frequency hopping facility will make it very difficult for an eavesdropper to lock onto the correct signal however and thus there is a much higher degree of inherent security in the system than is found in today's analogue systems. Nevertheless for users who need assurance of a secure transmission, GSM offers encryption over the air interface. This is based on a public key encryption principle and provides very high security.

1.3.4 Call Set-up

Once the network accepts the user and his equipment, the mobile must define the type of service it requires (voice, data, supplementary services etc.) and the destination number. At this point a traffic channel with the relevant capacity will be allocated and the MSC will route the call to the destination. Note that the network may delay assigning the traffic channel until the connection is made with the called number. This is known as off-air call set-up, and it can reduce the radio channel occupancy of any one call thus increasing the system traffic capacity.

1.3.5 Handover

GSM employs mobile-assisted handover. In this technique the mobile continuously monitors other base stations in its vicinity, measuring signal strength and error rate. These measurements are combined into a single function and the identities of the best six base stations are transmitted back to the system. The network can then decide when to initiate handover. The use of bit error rate, in addition to signal strength, adds considerably to the ability of the network to make informed handover decisions and is another example of the advantage of digital transmission over analogue. The BSC can initiate and execute handover if both BTS's are under its own control. In this instance the BSC can be considered as the manager of a specific group of radio frequencies for a geographic region and can control that resource to maximize its utilization. Alternatively, and whenever handover must take place to a cell outside the control of the BSC, the MSC controls and executes handover.

1.4 Communication Specification

1.4.1 Frequency Allocation

Throughout Europe GSM has been allocated a specific 50 MHz of spectrum divided into transmit and receive bands. The International Telecommunication Union (ITU), which manages the international allocation of radio spectrum (among other functions) allocated

the bands 890-915 MHz for the uplink (mobile station to base station) and 935-960 MHz for the downlink (base station to mobile station) for mobile networks in Europe.

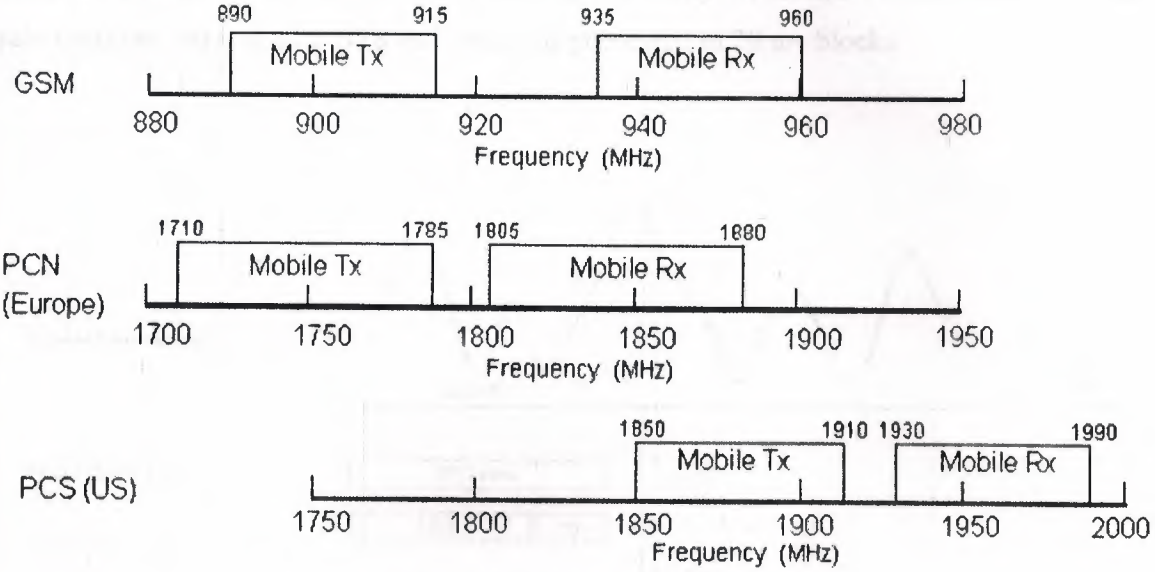

Figure 1.2 The Frequency Allocation of GSM

Since radio spectrum is a limited resource shared by all users, a method must be devised to divide up the bandwidth among as many users as possible. The method chose by GSM is a combination of Time and Frequency Division Multiple Access (TDMA/FDMA). The FDMA part involves the division by frequency of the total 25 MHz bandwidth into 124 carrier frequencies of 200 kHz bandwidth. One or more carrier frequencies are then assigned to each base station. Each of these carrier frequencies is then divided in time, using a TDMA scheme, into eight time slots. One time slot is used for transmission by the mobile and one for reception. They are separated in time so that the mobile unit does not receive and transmit at the same time, a fact that simplifies the electronics.

1.4.2 Speech Coding

GSM is a digital system, so speech signals, inherently analog, have to be digitized. The spectral efficiency target set for the GSM system demands a speech codec, which can

provide toll quality speech at 16 KBPS or less. The solution adopted is based on a Residually Excited Linear Predictive coder (RELP) enhanced by the inclusion of a long-term predictor (LTP). This improves the speech quality by removing the structure from the vowel sounds prior to coding the residual data. It has the effect of removing the coarseness often associated with linear predictive coding, especially on female voices. The basic data rate from the coder is 13 KBPS and speech is processed in 20 ms blocks.

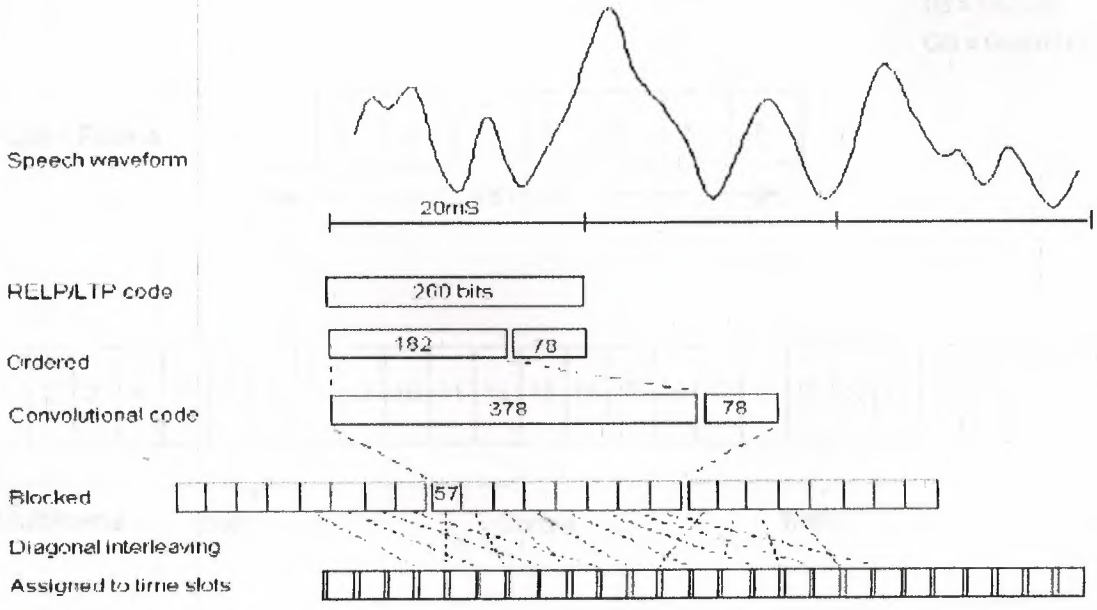

Figure 1.3 Speech Coding

The resulting code is split into two parts, the most critical bits being put first. This first part has a half-rate convolution code applied to it and when recombined with the second part the total block length is 456 bits. As we will see later this block length can be fitted into four time slots, but in practice it is spread over eight. This process is called diagonal interleaving and it allows the convolution code more chance to recover if a sequence of TDMA frames is badly corrupted during radio transmission.

1.4.3 Data/Channel Structure

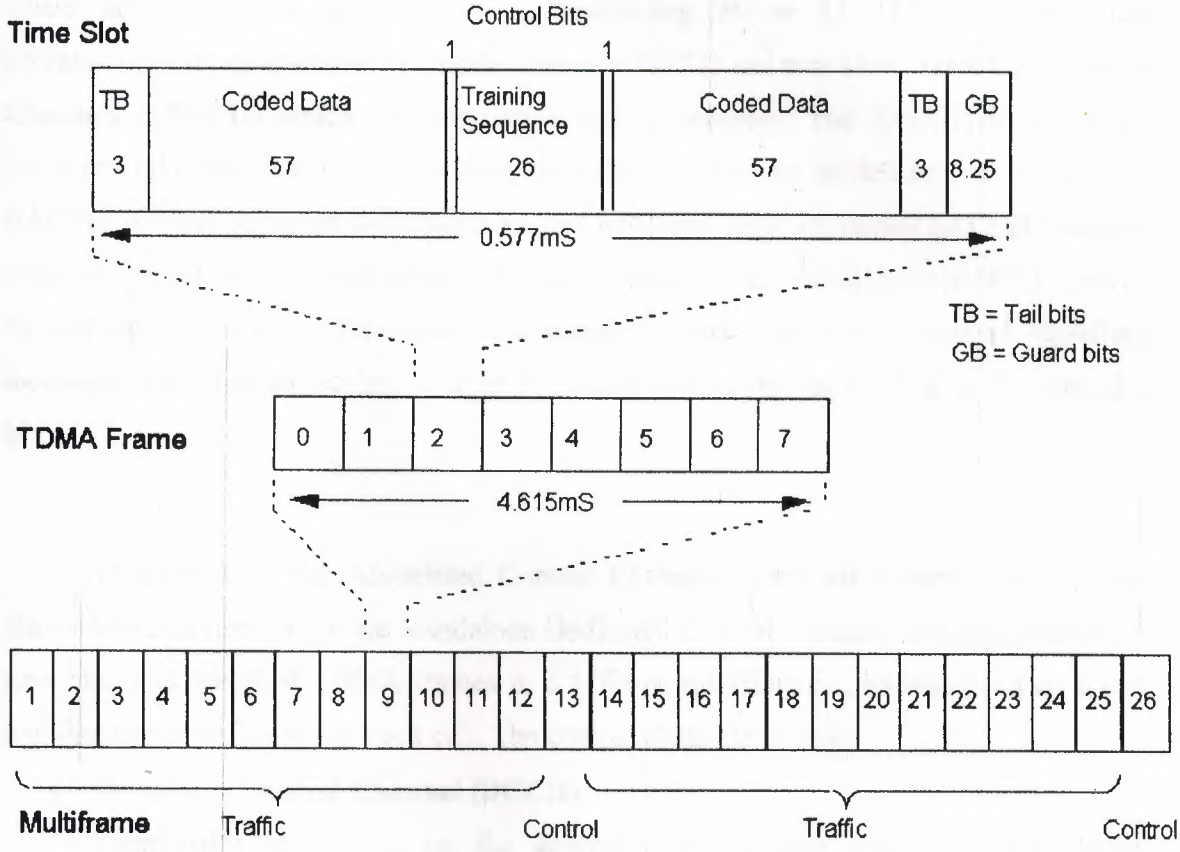

Figure 1.4 Structure of Data Channel

The structure of the most common timeslot burst is shown as above. A total of 156.25 bits is transmitted in 0.577 milliseconds, giving a gross bit rate of 270.833 KBPS. There are three other types of burst structure for frame and carrier synchronization and frequency correction. The 26bit training sequence is used for equalization, as described below. The 8.25 bit guard time allows for some propagation time delay in the arrival of bursts.

Each group of eight time slots is called a TDMA frame, which is transmitted every 4.615 ms. TDMA frames are further grouped into multiframes to carry control signals. There are two types of multiframe , containing 26 or 51 TDMA frames. The 26frame multiframe contains 24 Traffic Channels (TCH) and two Slow Associated Control Channels (SACCH) which supervise each call in progress. The SACCH in frame 12 contains eight channels, one for each of the eight connections carried by the TCHs. The SACCH in frame 25 is not currently used, but will carry eight additional SACCH channels when half rate traffic is implemented. A Fast Associated Control Channel (FACCH) works by stealing slots from a traffic channel to transmit power control and handover signalling messages. The channel stealing is done by setting one of the control bits in the time slot burst.

In addition to the Associated Control Channels, there are several other control channels which (except for the Standalone Dedicated Control Channel) are implemented in time slot 0 of specified TDMA frames in a 51frame multiframe , implemented on a non hopping carrier frequency in each cell. The control channels include:

- **Broadcast Control Channel (BCCH):**
  Continually broadcasts, on the downlink, information including base station identity, frequency allocations, and frequency hopping sequences.
- **Standalone Dedicated Control Channel (SDCCH):**
  Used for registration, authentication, call setup, and location updating.
  Implemented on a time slot, together with its SACCH, selected by the system operator.
- **Common Control Channel (CCCH):**
  Comprised of three control channels used during call origination and call paging.
- **Random Access Channel (RACH):**
  A slotted Aloha channel to request access to the network
- **Paging Channel (PCH):**
  Used to alert the mobile station of incoming call.

- **Access Grant Channel (AGCH):**
  Used to allocate an SDCCH to a mobile for signaling, following a request on the RACH.

1.5 Channel Coding and Modulation

Due to natural or man made electromagnetic interference, the encoded speech or data transmitted over the radio interface must be protected as much as is practical. The GSM system uses convolution encoding and block interleaving to achieve this protection. The exact algorithms used differ for speech and for different data rates. The method used for speech blocks will be described below.

Recall that the speech codec produces a 260-bit block for every 20 ms speech sample. From subjective testing, it was found that some bits of this block were more important for perceived speech quality than others. The bits are thus divided into three classes:

- Class I a 50 bits - most sensitive to bit errors
- Class I b 132 bits - moderately sensitive to bit errors
- Class II 78 bits - least sensitive to bit errors

Class Ia bits have a 3 bit Cyclic Redundancy Code added for error detection. If an error is detected, the frame is judged too damaged to be comprehensible and it is discarded. It is replaced by a slightly attenuated version of the previous correctly received frame. These 53 bits, together with the 132 Class Ib bits and a 4-bit tail sequence (a total of 189 bits), are input into a 1/2 rate convolution encoder of constraint length 4. Each input bit is encoded as two output bits, based on a combination of the previous 4 input bits. The convolution encoder thus outputs 378 bits, to which are added the 78 remaining Class II bits, which are unprotected. Thus every 20 ms speech sample is encoded as 456 bits, giving a bit rate of 22.8 KBPS.

To further protect against the burst errors common to the radio interface, each sample is diagonally interleaved. The 456 bits output by the convolution encoder are divided into 8 blocks of 57 bits, and these blocks are transmitted in eight consecutive timeslot bursts. Since each timeslot burst can carry two 57-bit blocks, each burst carries traffic from two different speech samples.

Recall that each timeslot burst is transmitted at a gross bit rate of 270.833 KBPS. This digital signal is modulated onto the analog carrier frequency, which has a bandwidth of 200 kHz, using Gaussian filtered Minimum Shift Keying (GMSK). GMSK was selected over other modulation schemes as a compromise between spectral efficiency, complexity of the transmitter, and limited spurious emissions. The complexity of the transmitter is related to power consumption, which should be minimized for the mobile station. The spurious radio emissions, outside of the allotted bandwidth, must be strictly controlled so as to limit adjacent channel interference, and allow for the coexistence of GSM and the older analog systems (at least for the time being).

2. RADIO TRANSMISSION

2.1 Introduction

GSM is a digital wireless communication system for mobile phones. It was developed in the early 1980's to eliminate certain problems with the existing cellular networks:

- Analog networks could not handle the growing capacity of cellular networks, especially in an economic way
- The existing digital networks were not compatible with each other

It was hoped that by creating one digital standard, the problems of analog systems would be overcome and users could roam from network to network without changing their equipment.

In addition to providing a solution for the above two problems, GSM supported many other useful features such as security, authentication and the SIM card, which makes it possible to switch phones without the need of reconfigure the phone.

In 1982 the European Commission reserved the 900 MHz spectrum in all member countries for GSM, thus setting the stage for interoperability across Europe. From 1982 to 1990 the specifications for GSM have been decided upon and in 1990 the final specifications were chosen. After 2 years, the first GSM network was operable in 1992 in Finland, and since then GSM has been a huge success.

Since the original GSM specifications in 1990, the GSM standard has been upgraded twice. In 1991 a new European GSM standard was developed to operate at 1800 MHz (usually called DCS1800). The new standard provided more bandwidth and less power requirements than the previous standard. Furthermore, in 1995 a North American GSM standard at 1900 MHz was established.

2.2 The GSM Transmission System

To transform the speech that enters the headset into the radio waves that are transmitted to the base station, GSM performs the following operations.

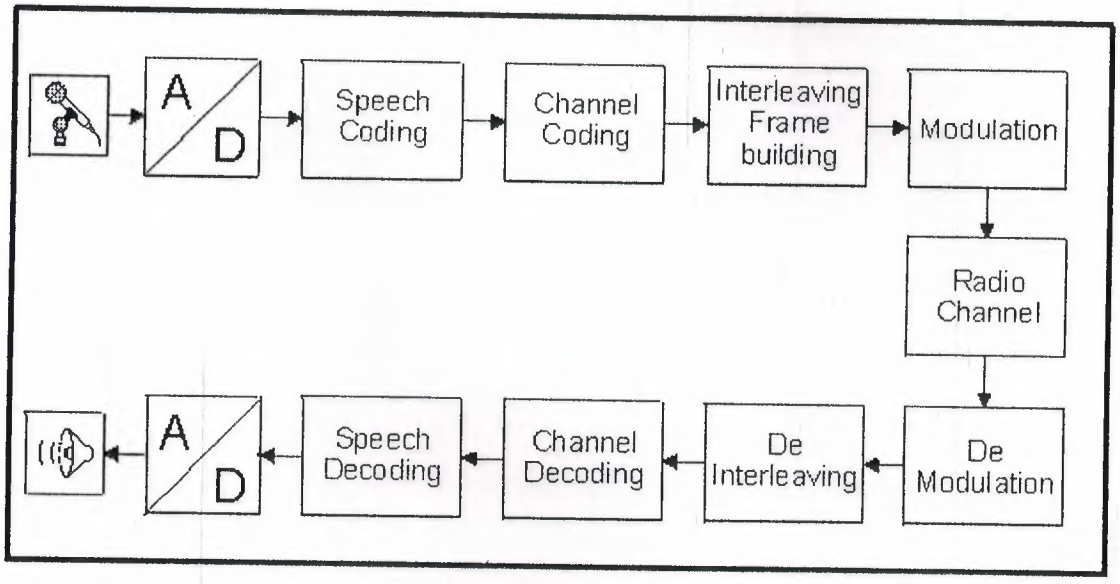

Figure 2.1 Transmission Systems in GSM

GSM is a digital communications standard, but voice is analog, and therefore it must be converted to a digital bit stream. GSM uses Pulse Coded Modulation to digitize voice, and then uses a speech codec to remove the redundancy in the signal.

Once the voice signal has been coded into a digital bit stream, extra bits are added to it so that the receiver can recognize and correct errors in the stream, which could have occurred during transmission. GSM uses a technique called convolution coding. Interleaving is the processes of rearranging the bits. It allows the error correction

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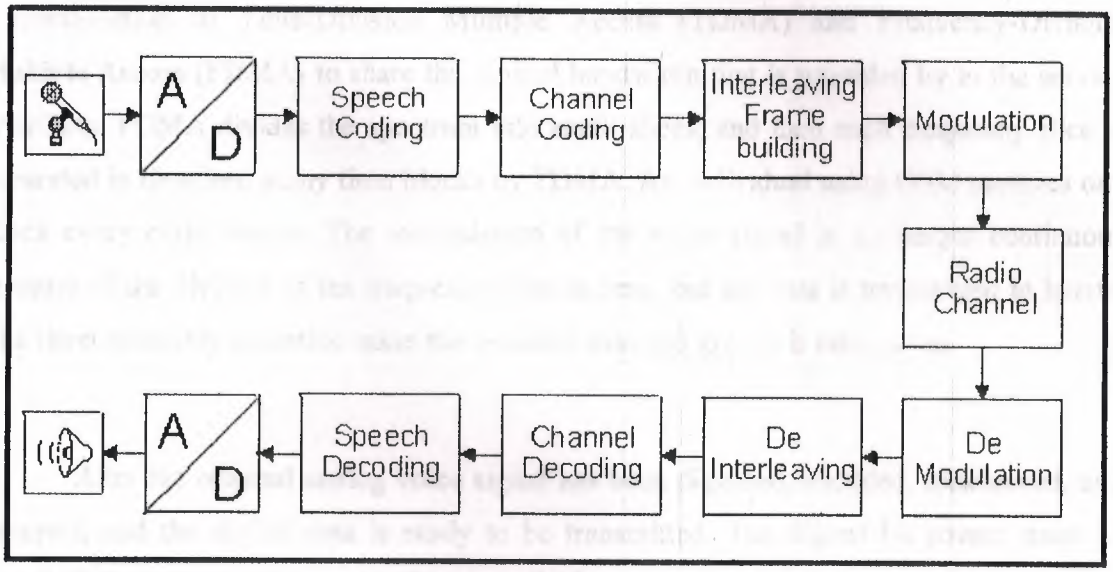

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algorithms to correct more of the errors that could have occurred during transmission. By interleaving the code, there is less possibility that a whole burst of bits is lost.

GSM allows many users to use their cell phones at the same time. The standard uses a combination of Time-Division Multiple Access (TDMA) and Frequency-Division Multiple Access (FDMA) to share the limited bandwidth that is provided by to the service providers. FDMA divides the spectrum into small slices, and then each frequency slice is separated in time into many time blocks by TDMA. An individual using GSM receives one block every eight blocks. The transmission of the voice signal is no longer continuous because of the division of the frequency slice in time, but the data is transmitted in bursts. The burst assembly operation takes the encoded data and groups it into bursts.

After the original analog voice signal has been digitized, encoded, interleaved, and grouped, and the digital data is ready to be transmitted. The digital bit stream must be encoded in a pulse and transmitted over radio frequencies. Modulation changes the '1' and '0's in a digital representation to another representation that is more suitable for transmission over airwaves.

3. DIGITAL RADIO SYSTEMS

3.1 The Advantages of Digital Transmission

Only a limited number of valid (predetermined) signal levels are transmitted. This permits:

- Waveform regeneration (for point-to-point, or line systems) - hence no cumulative distortion with distance.
- Satisfactory operation with lower carrier/noise levels than for analogue systems
- Readily permits Time Division Multiplexing (TDM) which has a greater transmission economy than Frequency Division Multiplexing used for analogue systems (filter guard-bands are replaced by time-slot guard periods)
- Greater large scale economy (VLSI)

3.1.1 Digital Radio Systems

Digital modulation used in communication systems provides several, benefits over analogue methods

Increased capacity - some analysts say 5 to 10 times more efficient than analogue hence greater capacity. Due to:

Multiple carriers per channel voice activity detection demand assignment

Increased security/privacy - digital speech can be easily encrypted **Additional services** (ISDN) - fax, data, etc.,

Reduced risk of fraud - as digital systems are more secure

Common system: - international standards allow international roaming

Spectrum - digital signals are less susceptible to interference from nearby cells than analogue; thus cell suites (clusters) may be closer - better frequency reuse by a factor of between two and four.

Traffic - improvements in call capacity over analogue networks means that there is less network delay and hence fewer channel busy conditions.

Greater compatibility with digital networks (especially ISDN), intelligent network Compatibility

Data - enrol, correction (FEC and ARQ) facilitates data transmission

Quality - slightly better radio performance - resistance to noise and cross talk. Digital systems can mask problems in the transmission channel such as deep fades and multi-path. Error correction and regeneration.

Flexibility

- Software reconfigurable.
- Smart card technology
- Generic data packets for all types of service

3.2 GSM Pan European Digital Cellular Radio

Originally known as Group Special Mobile, now known as Global System for Mobile communications. GSM was initially set up under the CEPT but was developed under the control of ETSI. A fundamental requirement of GSM was that it provided pan European roaming. A second requirement was that it provides a more efficient use of the spectrum than the first generation cellular systems (analogue).

GSM uses the following frequency bands:

Mobile Transmit 905 - 915 MHz and 872 - 888 (additional requested)

Base Transmit 935 - 960 MHz and 917 - 933.

Channel Bandwidth = 200 kHz - (compare with 25 kHz for TACS) but eight stations share each channel on a time division multiplex basis.

One or several Base Station Systems are connected to the Mobile Switching Centre (MSC)

The MSC is similar to an ISDN exchange and is the interconnection point to the ISDN/PSTN.

Mobiles are classified as **follows:**

Class 1	20 W peak (for vehicle use)
Class 2	8 W peak "
Class 3	5 W peak (hand/portable)
Class 4	2 W peak "
Class 5	0.85 W peak (small handsets)

As with TACS the actual mobile transmit power is controlled by the base station

Base Station Transmitter Power varies between 7 W and 45 W (100 W ERP)

Additional features of the GSM system:

ISDN capability with full ISDN numbering, but the data rate is limited to 9.6 kbits/s (not 64 kbits/s) so Fax messages are group 3 not group 4 and the Bit Error Rate may slow fax operation.

The Mobile Station also contains a removable Subscriber Identity Module (SIM) which is used to identify and validate the user, as well as storing some of the user's operating parameters such as his number. The use of the SIM means that a subscriber may use various different handsets at different times, with all calls being billed to his account. Certain limited facilities should be available from a mobile without the SIM fitted, e.g. access to emergency services.

The GSM system is designed for a maximum range (maximum cell size) of 35 km. The performance of all types of cellular radio systems is primarily restricted by co-channel interference. Digital modulation can tolerate a much higher level of co-channel interference than analogue systems and still produce an acceptable voice quality - this results in an increase in spectral efficiency.

The radio transmission method chosen for GSM is a Narrow-Band Time Division Multiple Access structure. Eight stations share a 200 kHz bandwidth radio channel. The mobile's transmitter and receiver are allocated different carrier frequencies (45 MHz separation) and differing time slots, therefore the transmitter and receiver do not operate simultaneously (as with TACS) and diplexers are not needed.. (Diplexers are very high performance filters that protect the receiver input from being swamped by the output of the transmitter). Diplexers are needed for the base stations, as they have to transmit and receive simultaneously.

The further the mobile is from the base station the longer the signals take to arrive. TDMA requires accurate synchronisation of the transmissions from the mobiles in order to ensure that signals do not overlap at the base station. The transmission times are calculated

at the base station and synchronisation information is sent to the mobile. Without this procedure it would be necessary to allow relatively long guard times.

Speech is coded into a data stream of 13 kbits/s (compare with 64 kbits/s for PCM). An option allows for half of the 13 kbits/s coding rate, which will double the spectrum efficiency and the system capacity.

The coded speech is divided into 20 ms blocks of 260 bits each, and the first 182 bits of each block are extended to 189 by adding parity and spare bits. These are then encoded for error detection and correction purposes by means of a convolutional code which adds a further 189 bits, and then recombined with the remaining 78 bits to give a total block length of 456 bits. The bits which are error protected are those for which accuracy is most critical for the voice coder.

The 456 bit length blocks are then divided into eight sub-blocks of 57 bits which are interleaved with sub-blocks from adjacent blocks - in order to protect against bursts of errors. This is to help combat the effects of any burst errors and deep fades. Pairs of sub-blocks are combined with the 26 bit equaliser training sequence, and two control bits plus three header and three tail bits, into a time slot of 148 bits. Guard times equivalent to 8.25 bits are allowed between frames. Time slots from each of the eight mobiles are combined sequentially into a TDMA frame of 4.615 ms (each of the 8 time slots last for 576.92 microsecs). Twenty-four of the TDMA frames are combined with two supervisory and control frames which have a channel associated structure and contain the signalling information for eight mobiles. The overall bit rate is 270 kbits/s.

3.3 Aspects of the GSM Network

When a mobile is switched on three actions take place: it registers with the network, is authenticated and its location is updated. Cell reselection takes place automatically as it moves across cell boundaries and the network continuously tracks the mobile at all times.

Thus the network is able to route incoming calls quickly and efficiently, and allow or deny outgoing calls, whilst calls in progress can continue without interruption as the mobile crosses cell boundaries.

The GSM radio interface uses both Frequency Division Multiple Access (FDMA) - as does TACS, and Time Division Multiple Access (TDMA) as well as a Slow Frequency Hopping (SFH) scheme. Each radio channel has a bandwidth of 200 kHz and each cell is allocated a number of radio channels from the GSM bands according to the traffic requirements. The basic frequency bands initially allocated to GSM are two sub-bands of 25 MHz each:

890 - 915 MHz (Mobile Transmit) and, 935 - 960 MHz (Base Transmit).

Note that 45 MHz separates the bands, as with TACS.

In addition there are other bands, which were initially allocated for analogue cellular (TACS) which have been earmarked for GSM expansion (872 - 888 - mobile Tx and 917 - 933 - Base Tx).

The modulation scheme use by GSM results in a spectrum which is somewhat greater than the 200 kHz per channel, allocated. So careful planning is required in order to minimise adjacent channel interference. Thus there are 125 channels, each 200 kHz wide, operated on a Frequency Division Multiple Access duplex basis. Each channel is multiplexed on a Time Division Multiple Access basis, with 8 different time slots per radio channel. Each channel time slot provides for individual 13 kbit/s speech (or soon two 6.5 kbit/s speech channels) or data at 9.6 kbit/s. Thus each channel may be used "simultaneously" by eight mobiles operating with a full channel system, or 16 mobiles operating with a half channel system.

Currently, the two 25 MHz bands allocated for GSM will support 1000 mobiles. This is similar to the case for TACS. In TACS, each 25 kHz wide channel was allocated to an individual mobile. Hence 1000 mobiles would occupy 25 MHz. Therefore, one might ask how is GSM able to use the spectrum more efficiently than TACS? The answer lies with the fact that the GSM system is able to operate with a much lower Carrier to Interference (C/I) ratio than TACS. Typical figures are 10 dB for GSM compared with 18 dB for TACS. This 8 dB difference represents a factor of 6 or 7 times. The result is that GSM is able to use a much smaller frequency re-use distance and thus smaller cells may be used which results in a much higher system capacity.

3.4 Equalisation and Multi-Path Propagation

One of the effects of multi-path is that delayed signals combine to give a composite signal at the antenna of the mobile receiver and the amplitude and phase of the composite signal undergoes wide and rapid fluctuations. The delay may be serious enough to cause serious inter-symbol interference, which can limit the maximum usable bit rate.

In the GSM system the long excess delays can only be tolerated by using an **equaliser** in the receiver. A known bit pattern of 26 bits (called a **training sequence**) is transmitted at regular intervals. The equaliser compares the received bit pattern with the training sequence and adjusts the parameters of a digital filter so as to compensate for the effects of multipath. Because with a fast moving vehicle the geometry of the path can change rapidly, the training sequence is included on every time slot.

3.5 Digital Modulation.

3.5.1 Frequency Shift Keying:

Carrier is shifted to specific frequencies, one for each symbol

(e.g. TACS: logic I = fc + 6.4 kHz, logic 0 = fc-6.4 kHz)

a) Advantages:

- Constant amplitude
- No need to adjust the demodulation threshold as fading occurs - as would be the case for amplitude based modulation systems

b) Minimum Shift Keying

- A form of FSK which uses continuous phase
- Frequencies are spaced so that during the symbol interval one symbol goes through 180 degrees more phase shift than the other
- The result is a maximum phase difference at the end of each symbol interval with a minimum of change in frequency
- Phase continuity is maintained at signalling transitions
- Advantage: a compact spectrum

c) GMSK modulation

The GSM specifications require the out-of-band radiated power in the adjacent channels to be between 40 and 70 dBs below that of the desired channel. To satisfy this requirement, it is necessary to bandlimit the spectrum of the R-F output signal. This cannot be achieved easily at the final RF stage in multi-channel transceivers. In order to comply with the spectrum requirements narrow band digital modulation schemes must be used. Amongst these schemes, Minimum Shift Keying, (covered above) has proved to provide a better modulation for use on bandlimited and non-linear channels. It has the following properties:

- Constant amplitude envelope
- Relative narrow bandwidth
- Coherent/non-coherent detection capabilities

- A good bit-error-rate performance

However, MSK does not completely satisfy the out-of-band radiation requirements. To overcome this limitation a pre-modulation filter is used. A filter with a Gaussian response is suitable for this role and the modified form of modulation is termed Gaussian MSK (GMSK). The filter is included between the NRZ data stream from the voice processor and the MSK modulator stage.

The Gaussian filter should have the following properties:

- Narrow bandwidth and sharp cut-off (to suppress high frequency components)
- Low overshoot response (to protect against excessive instantaneous frequency deviation)
- The same phase differences per symbol interval as MSK
- The same phase continuity during signal transitions as MSK.

GMSK is a form of FM. The carrier is therefore at a constant amplitude during transmission, allowing the use of low cost, efficient transmitters (class C bias).

3.5.2 Frequencies, Bursts, Time and Slots

There are 125 frequencies available in the main GSM bands and these frequencies are distributed over the different cells in a way that minimises interference and matches traffic distribution. All communications over the radio/air interface consist of packetised bursts of data which have to fit into a defined time and frequency window called a slot. A slot is 577 μs long and 200 kHz wide.

The basic TDMA frame consists of eight successive slots (16 in the case of the half rate voice coding system) therefore any given frequency may be used to carry traffic to, or from, eight different mobiles. The time slots are numbered TNO to TN7.

There are two different types of channels:

- Dedicated channels, which carry communications between a particular mobile and the base station, and are used only when the mobile is actively engaged in a call
- Common channels, which are used by all mobiles both during the idle state and during calls, and are predominantly used for control purposes.

The common channels are always carried on a fixed frequency for a given cell known as the

Beacon frequency. Time slot zero (TNO) on the beacon frequency includes synchronisation information to help the mobiles lock onto the system. The unique presence of the synchronisation signals defines and identifies TNO. Additional common channel capacity may be provided in other TNs on the same frequency.

Table 3.1 Slot used for common channels, other slots are used for dedicated channels

TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0
TN0	TN1	TN2	TN3	TN4	TN5	TN6	TN7	TN0

This white paper is intended for global system for mobile communications (GSM) operators interested in understanding Ericsson's view on how enhanced data for global evolution (EDGE) can play an important role in the evolution toward wideband code division multiple access (WCDMA). EDGE can be introduced in two ways: (1) as a packet-switched enhancement for general packet radio service (GPRS), known as enhanced GPRS or EGPRS, and (2) as a circuit-switched data enhancement called enhanced circuit-switched data (ECSD). This white paper, however, will only discuss the packet-switched enhancement, EGPRS. The purpose of this white paper is to describe EDGE technology and how it leverages existing GSM systems and complements WCDMA for further growth. The benefits described here are based on Ericsson's vision of one seamless network for GSM and WCDMA. The white paper is based on Ericsson's current experience with operators' deployment processes, our past experience with technology transitions and our expertise with all major wireless standards, including GPRS, EDGE and WCDMA.

3.6 Executive Summary

EDGE is the next step in the evolution of GSM and IS-136. The objective of the new technology is to increase data transmission rates and spectrum efficiency and to facilitate new applications and increased capacity for mobile use. With the introduction of EDGE in GSM phase 2+, existing services such as GPRS and high-speed circuit switched data (HSCSD) are enhanced by offering a new physical layer. The services themselves are not modified. EDGE is introduced within existing specifications and descriptions rather than by creating new ones. This white paper focuses on the packet-switched enhancement for GPRS, called EGPRS. GPRS allows data rates of 115 kbps and, theoretically, of up to 160 kbps on the physical layer. EGPRS is capable of offering data rates of 384 kbps and, theoretically, of up to 473.6 kbps. A new modulation technique and error-tolerant transmission methods, combined with improved link adaptation mechanisms, make these EGPRS rates possible. This is the key to increased spectrum efficiency and enhanced applications, such as wireless Internet access, e-mail and file transfers. GPRS/EGPRS will be one of the pacesetters in the overall wireless technology evolution in conjunction with WCDMA. Higher transmission rates for specific radio resources enhance capacity by enabling more traffic for both circuit- and packet-switched services. As the Third-generation Partnership Project (3GPP) continues standardization toward the GSM/EDGE radio access network (GERAN), GERAN will be able to offer the same services as WCDMA by connecting to the same core network. This is done in parallel with means to increase the spectral efficiency. The goal is to boost system capacity, both for real-time and best effort services, and to compete effectively with other third-generation radio access networks such as WCDMA and cdma2000.

3.7 Technical Differences between GPRS and EGPRS

Regarded as a subsystem within the GSM standard, GPRS has introduced packet-switched data into GSM networks. Many new protocols and new nodes have been introduced to make this possible. EDGE is a method to increase the data rates on the radio link for GSM. Basically, EDGE only introduces a new modulation technique and new

channel coding that can be used to transmit both packet-switched and circuit-switched voice and data services. EDGE is therefore an add-on to GPRS and cannot work alone. GPRS has a greater impact on the GSM system than EDGE has. By adding the new modulation and coding to GPRS and by making adjustments to the radio link protocols, EGPRS offers significantly higher throughput and capacity. EGPRS introduces changes to GPRS only on the base station system part of the network. GPRS and EGPRS have different protocols and different behavior on the base station system side. However, on the core network side, GPRS and EGPRS share the same packet-handling protocols and, therefore, behave in the same way. Reuse of the existing GPRS core infrastructure (serving GRPS support node/gateway GPRS support node) emphasizes the fact that EGPRS is only an "add-on" to the base station system and is therefore much easier to introduce than GPRS. GPRS Protocol

In addition to enhancing the throughput for each data user, EDGE also increases capacity. With EDGE, the same time slot can support more users. This decreases the number of radio resources required to support the same traffic, thus freeing up capacity for more data or voice services. EDGE makes it easier for circuit-switched and packet-switched traffic to coexist while making more efficient use of the same radio resources. Thus in tightly planned networks with limited spectrum, EDGE may also be seen as a capacity booster for the data traffic.

3.7.1 EDGE Technology

EDGE leverages the knowledge gained through use of the existing GPRS standard to deliver significant technical improvements. GPRS and EDGE: A Comparison of technical data. (Legend 8PSK, 8-phase shift keying; GMSK, Gaussian minimum shift keying

Figure compares the basic technical data of GPRS and EDGE. Although GPRS and EDGE share the same symbol rate, the modulation bit rate differs. EDGE can transmit three

times as many bits as GPRS during the same period of time. This is the main reason for the higher EDGE bit rates. The differences between the radio and user data rates are the result of whether or not the packet headers are taken into consideration. These different ways of calculating throughput often cause misunderstanding within the industry about actual throughput figures for GPRS and EGPRS. The data rate of 384 kbps is often used in relation to EDGE. The International Telecommunications Union (ITU) has defined 384 kbps as the data rate limit required for a service to fulfill the International Mobile Telecommunications-2000 (IMT-2000) standard in a pedestrian environment. This 384 kbps data rate corresponds to 48 kbps per time slot, assuming an eight-time slot terminal.

3.7.2 EDGE Modulation Technique

The modulation type that is used in GSM is the Gaussian minimum shift keying (GMSK), which is a kind of phase modulation. This can be visualized in an I/Q diagram that shows the real (I) and imaginary (Q) components of the transmitted signal (Figure 3). Transmitting a zero bit or one bit is then represented by changing the phase by increments of + _ p. Every symbol that is transmitted represents one bit; that is, each shift in the phase represents one bit. Figure 3. I/Q diagram showing EDGE modulation benefits. To achieve higher bit rates per time slot than those available in GSM/GPRS, the modulation method requires change. EDGE is specified to reuse the channel structure, channel width, channel coding and the existing mechanisms and functionality of GPRS and HSCSD. The modulation standard selected for EDGE, 8-phase shift keying (8PSK), fulfills all of those requirements. 8PSK modulation has the same qualities in terms of generating interference on adjacent channels as GMSK. This makes it possible to integrate EDGE channels into an existing frequency plan and to assign new EDGE channels in the same way as standard GSM channels. The 8PSK modulation method is a linear method in which three consecutive bits are mapped onto one symbol in the I/Q plane. The symbol rate, or the GPRS: GMSK Modulation GPRS: 8PSK Modulation

GPRS EDGE

- Modulation GMSK 8-PSK/GMSK
- Symbol rate 270 ksym/s 270 ksym/s
- Modulation bit rate **270** kb/s **810** kb/s
- Radio data rate per time slot 22,8 kb/s 69,2 kb/s
- User data rate per time slot **20** kb/s (CS4) **59,2** kb/s (MCS9)
- User data rate (8 time slots) 160 kb/s 473,6 kb/s (182,4 kb/s) (553,6 kb/s)

Time remains the same as for GMSK, but each symbol now represents three bits instead of one. The total data rate is therefore increased by a factor of three. The distance between the different symbols is shorter using 8PSK modulation than when using GMSK. Shorter distances increase the risk for misinterpretation of the symbols because it is more difficult for the radio receiver to detect which symbol it has received. Under good radio conditions, this does not matter. Under poor radio conditions, however, it does. The "extra" bits will be used to add more error correcting coding, and the correct information can be recovered. Only under very poor radio environments is GMSK more efficient. Therefore the EDGE coding schemes are a mixture of both GMSK and 8PSK.

3.8 Coding Schemes

For GPRS, four different coding schemes, designated CS1 through CS4, are defined. Each has different amounts of error-correcting coding that is optimized for different radio environments. For EGPRS, nine modulation coding schemes, designated MCS1 through MCS9, are introduced. These fulfill the same task as the GPRS coding schemes. The lower four EGPRS coding schemes (MSC1 to MSC4) use GMSK, whereas the upper five (MSC5 to MSC9) use 8PSK modulation. Figure 2.1 shows both GPRS and EGPRS coding schemes, along with their maximum throughputs. GPRS user throughput reaches saturation at a maximum of 20 kbps with CS4, whereas the EGPRS bit rate continues to increase as the radio quality increases, until throughput reaches saturation at 59.2 kbps. Both GPRS CS1 to CS4 and EGPRS MCS1 to MCS4 use GMSK modulation with slightly different throughput performances. This is due to differences in the header

size (and payload size) of the EGPRS packets. This makes it possible to resegment EGPRS packets. A packet sent with a higher coding scheme (less error correction) that is not properly received, can be retransmitted with a lower coding scheme (more error correction) if the new radio environment requires it. This resegmenting (retransmitting with another coding scheme) requires changes in the payload sizes of the radio blocks, which is why EGPRS and GPRS do not have the same performance for the GMSK modulated coding schemes. Re-segmentation is not possible with GPRS.

3.9 Packet Handling

Another improvement that has been made to the EGPRS standard is the ability to retransmit a packet that has not been decoded properly with a more robust coding scheme. For GPRS, re-segmentation is not possible. Once packets have been sent, they must be retransmitted using the original coding scheme even if the radio environment has changed. This has a significant impact on the throughput, as the algorithm decides the level of confidence with which the link adaptation (LA) must work.

Below is an example of packet transfer and retransmission for GPRS.

A. The GPRS terminal receives data from the network on the downlink. Due to a GPRS measurement report that was previously received, the link adaptation algorithm in the base station controller decides to send the next radio blocks (e.g., numbers 1 to 4) with CS3. During the transmission of these packages, the carrier-to-interference ratio (C/I) decreases dramatically, changing the radio environment. After the packets have been transmitted, the network polls for a new measurement report, including the acknowledged/unacknowledged bitmap that tells the network which radio blocks were received correctly.

B. The GPRS handset replies with a packet downlink acknowledged/unacknowledged message containing the information about the link quality and the bitmap. In this scenario, it is assumed that packets 2 and 3 were sent erroneously.

C. Based on the new link quality information, the GPRS link adaptation algorithm will adapt the coding scheme to the new radio environment using CS1 for the new packets 5 and 6. However, because GPRS cannot resegment the old packets, packets 2 and 3 must be retransmitted using CS3, although there is a significant risk that these packets still may not be decoded correctly. As a result, the link adaptation for GPRS requires careful selection of the coding scheme in order to avoid retransmissions as much as possible. With EGPRS, resegmentation is possible. Packets sent with little error protection can be retransmitted with more error protection, if required by the new radio environment. The rapidly changing radio environment has a much smaller effect on the problem of choosing the wrong coding scheme for the next sequence of radio blocks because resegmentation is possible. Therefore, the EGPRS link-controlling algorithm can be very aggressive when selecting the modulation coding schemes.

3.10 Addressing Window

Before a sequence of coded radio link control packets or radio blocks can be transmitted over the Um (radio) interface, the transmitter must address the packets with an identification number. This information is then included in the header of every packet. The packets in GPRS are numbered from 1 to 128. After transmission of a sequence of packets (e.g., 10 packets), the transmitter asks the receiver to verify the correctness of the packets received in the form of an acknowledged/unacknowledged report. This report informs the transmitter which packet or packets were not successfully decoded and must be retransmitted. Since the number of packets is limited to 128 and the addressing window is 64, the packet sending process can run out of addresses after 64 packets. If an erroneously decoded packet must be retransmitted, it may have the same number as a new packet in the queue. If so, the protocol between the terminal and the network stalls, and all the packets belonging to the same low-layer capability frame must be retransmitted. In EGPRS, the addressing numbers have been increased to 2048 and the window has been increased to

1024 in order to minimize the risk for stalling. This, in turn, minimizes the risk for re-transmitting low-layer capability frames and prevents decreased throughput

3.11 Measurement Accuracy

As in the GSM environment, GPRS measures the radio environment by analyzing the channel for carrier strength, bit error rate, etc. Performing these measurements takes time for a mobile station, which is of no concern in the speech world as the same coding is used all the time. In a packet-switched environment, it is essential to analyze the radio link quickly in order to adapt the coding toward the new environment. The channel analysis procedure that is used for GPRS makes the selection of the right coding scheme difficult since measurements for interference are performed only during idle bursts. As a result, measurements can only be performed twice during a 240-millisecond period. For EGPRS, the standard does not rely on the same "slow" measurement mechanism. Measurements are taken on each and every burst within the equalizer of the terminal, resulting in an estimate of the bit error probability (BEP). Estimated for every burst, the BEP is a reflection of the current C/I, the time is persion of the signal and the velocity of the terminal. The variation of the BEP value over several bursts will also provide additional information regarding velocity and frequency hopping.

A very accurate estimation of the BEP is then possible to achieve.

A mean BEP is calculated per radio block (four bursts) as well as the variation (standard deviation of the BEP estimation divided by the mean BEP) over the four bursts. These results are then filtered for all radio blocks sent within the measurement period.

This results in highly accurate measurements even during short measurement periods. Short measurement periods, in turn, enable quick reaction to changes in the radio

environment. It is therefore possible to achieve a better and more flexible link adaptation for EGPRS.

3.12 Interleaving

To increase the performance of the higher coding schemes in EGPRS (MCS7 to MCS9) even at low C/I, the interleaving procedure has been changed within the EGPRS standard.

When frequency hopping is used, the radio environment is changing on a per-burst level. Because a radio block is interleaved and transmitted over four bursts for GPRS, each burst may experience a completely different interference environment. If just one of the four bursts is not properly received, the entire radio block will not be properly decoded and will have to be retransmitted. In the case of CS4 for GPRS, hardly any error protection is used at all. With EGPRS, the standard handles the higher coding scheme differently than GPRS to combat this problem. MCS7, MCS8 and MCS9 actually transmit two radio blocks over the four bursts, and the interleaving occurs over two bursts instead of four. This reduces the number of bursts that must be retransmitted should errors occur. The likelihood of receiving two consecutive error-free bursts is higher than receiving four consecutive error-free bursts. This means that the higher coding schemes for EDGE have a better robustness with regard to frequency hopping.

3.13 EGPRS Link Controlling Function

To achieve the highest possible throughput over the radio link,

Retransmission of lost Block necessary Retransmission of second half only Figure 7. Interleaving. (Legend: CS, coding scheme; EGPRS, enhanced GPRS; MCS, modulation-coding scheme)

EGPRS uses a combination of two functionalities: link adaptation and incremental redundancy. Compared to a pure link adaptation solution, this combination of mechanisms significantly improves performance.

3.14 Link Adaptation

Link adaptation uses the radio link quality, measured either by the mobile station in a downlink transfer or by the base station in an uplink transfer, to select the most appropriate modulation coding scheme for transmission of the next sequence of packets. For an uplink packet transfer, the network informs the mobile station which coding scheme to use for transmission of the next sequence of packets. The modulation-coding scheme can be changed for each radio block (four bursts), but a change is usually initiated by new quality estimates. The practical adaptation rate is therefore decided by the measurement interval. There are three families: A, B and C. Within each family, there is a relationship between the payload sizes, which makes resegmentation for retransmissions possible.

3.15 Incremental Redundancy

Incremental redundancy initially uses a coding scheme, such as MCS9, with very little error protection and without consideration for the actual radio link quality. When information is received incorrectly, additional coding is transmitted and then soft combined in the receiver with the previously received information. Soft combining increases the probability of decoding the information. This procedure will be repeated until the information is successfully decoded. This means that information about the radio link is not necessary to support incremental redundancy. For the mobile stations, incremental redundancy support is mandatory in the standard.

3.16 Impact of EGPRS on Existing

3.16.1 GSM/GPRS Networks

Due to the minor differences between GPRS and EGPRS, the impact of EGPRS on the existing GSM/GPRS network is limited to the base station system. The base station is affected by the new transceiver unit capable of handling EDGE modulation as well as new software that enables the new protocol for packets over the radio interface in both the base station and base station controller. The core network does not require any adaptations. Due to this simple upgrade, a network capable of EDGE can be deployed with limited investments and within a short time frame. Use same MCS for retransmissions MCS9 becomes more robust than MCS5 for similar bit rate

3.16.2 Standardization

a) Background

Standardization of the first releases of the third generation cellular systems that comply with ITU/IMT-2000 requirements has now been finalized with European Telecommunications Standards Institute (ETSI/3GPP) Release 99. Two such major systems are Universal Mobile Telecommunications System (UMTS) and GSM/EDGE.

b) Fulfilling the EDGE Standardization

EDGE standardization can be divided in three areas:

- Standardization of the physical layer changes (definition of the modulation and coding schemes)
- The protocol changes for ECSD and
- EGPRS.

c) EDGE standard and references

The EDGE base station system work item provides a platform to employ new modulation techniques, whereas the EDGE network support subsystem work item defines the network changes to facilitate the physical layer. According to the work item descriptions, EDGE will provide two phases: Phase 1: Single- and multi-slot packet-switched services and single and multi-slot circuit switched services. Phase 2: Real-time services employing the new modulation techniques that are not included in Phase 1. Phase 1 has been completed with 3GPP Release 99. Phase 2 is ongoing in the 3GPP standardization, and its scope has been extended to cover the alignment with WCDMA and the provisioning of Internet protocol (IP) multimedia. This concept, currently standardized in 3GPP, is known as GERAN.

d) Requirements on EDGE

From the beginning, the standardization of EDGE was restricted to the physical layer and to the introduction of a new modulation scheme. Since EDGE was intended as an evolution of the existing GSM radio access technology, the requirements were set accordingly:

- EDGE- and non-EDGE-capable mobile stations should be able to share one and the same time slot.
- EDGE- and non-EDGE-capable transceivers should be deployable in the same spectrum.
- A partial introduction of EDGE should be possible. To ease implementation of new terminals while taking into account the asymmetrical characteristic of most services currently available, it was also decided that two classes of terminals should be supported by the EDGE standard:
- A terminal that provides 8PSK capability in the downlink only, and
- A terminal that provides 8PSK in the uplink and downlink.

e) Service Aspects

The introduction of EGPRS enables bit rates that are approximately three times higher than standard GPRS bit rates. Within the EDGE work item, this was simply handled by reusing the GPRS quality of service (QoS) profiles and extending the parameter range to reflect the higher bit rates, or in other words, introducing higher throughput values.

f) Architecture

EGPRS does not bring about any direct architecture impacts (see GSM 03.60). The packet control unit may still be placed either in the base station, the base station controller or the GPRS support node, and the central control unit is always placed in the base station. However, since the radio link control automatic repeat request function on the network side is located in the packet control unit, any delay introduced between the PCU and the radio interface will directly affect the radio link control acknowledged/ unacknowledged round-trip times. This, in turn, results in a higher risk of stalling the radio link control protocol. To mitigate this risk and to allow the operator to optimize network behavior, the maximum radio link control automatic repeat request window size has been extended for EGPRS.

g) User plane protocols

The transmission plane protocol structure for GPRS is shown in **Figure 11**. The protocols that are influenced by the introduction of EDGE are shaded. The protocols closest to the physical layer (the radio link control and mobile allocation channel) are most affected by EDGE (see GSM 04.60). There also are some minor modifications to the base station system GPRS protocol. Apart from these changes, the rest of the protocol stack remains intact after the introduction of EDGE.

h) Control plane protocols and channels

The introduction of EGPRS also has an impact on these control plane layers: mobility management and radio resource management. There is no impact on session

management. The mobility management modifications are related to introducing information on EGPRS capabilities in the mobile station radio access capabilities information element. These capabilities include the EGPRS multi-slot class, the EDGE modulation capability and the 8PSK power class. On the radio resource management layer, support for setting up and maintaining EGPRS temporary block flows is introduced as opposed to standard GPRS temporary block flows. Signaling supporting the radio link control, link quality control and measurement procedures is also introduced (see GSM 03.64, 04.18, 04.60).

The next evolutionary step for the GSM/EDGE cellular system includes a definition of enhancements that will lead to increased alignment with UMTS/UTRAN (UMTS terrestrial radio access network), further evolving GSM toward third-generation wireless systems. These enhancements are currently being specified for GERAN in the coming releases of the 3GPP standard. Based on EDGE high-speed transmission techniques combined with enhancements to the GPRS radio link interface, GERAN will provide support for conversational and streaming service classes as defined for WCDMA. By doing so, a new range of applications, including IP multimedia applications, can be adequately supported (see 3GPP TS 43.051). The next step for the GSM/EDGE evolution focuses on support for conversational and streaming service classes, the so-called real-time services. A driver for such evolution is the paradigm shift within the telecommunications world from circuit- to packet-switched communications. This trend is valid not only for traditional data services, such as e-mail and web browsing, but also for real-time services, such as videoconferencing and voice over IP. Both the second-generation packet-switched core network defined for GPRS and the current GSM/EDGE radio access network require modification to support real-time services. A part of the solution is to adopt the same IU interface to the third-generation WCDMA/GPRS core network as UTRAN. This simplifies the alignment of the services that will be provided in WCDMA and also enables the connection to the same third-generation core network. The current evolution of GSM/EDGE that covers all of the above aspects is called GERAN in the 3GPP standardization. In summary, the two main objectives for GERAN are:

- Alignment with WCDMA services primarily related to providing conversational and streaming service classes, and
- The possibility to interface the WCDMA core network over the same IU interface as WCDMA/UTRAN.
- In addition, GERAN will include performance enhancements for existing services.

3.17 GERAN System Architecture

Support of packet-based real-time services and adoption of the WCDMA QoS architecture requires changes to the second-generation GPRS core network. Instead of introducing these changes, another attractive solution is to connect GERAN to the third-generation WCDMA/GPRS core network, which supports both real-time services and the WCDMA QoS architecture. This allows for one common core network for UTRAN and GERAN connected over a common interface (see 3GPP TS 43.051). To connect to the third-generation WCDMA/GPRS core network, GERAN will use the IU interface (3GPP. Release 5 of the specification), as shown in Figure 12.

The IU interface is composed of two parts: the IU-ps, which connects to the packet-switched domain of the core network, and IU-cs, which connects to the circuit-switched domain of the core network. interfaces remain intact to support Release 99 terminals, making GERAN fully backward compatible with regard to terminal support for GSM, GPRS, ECSD and EGPRS. The main reason for supporting A and Gb interfaces for backward compatibility and not using the IU-ps interface for Release 99 terminals is that the functional split between the radio access network and the core network differs substantially between Iu and A/Gb. The radio interface between the mobile station and GERAN, called the Um interface, is based on the radio link interface of Release 99. However, several enhancements are being specified on different radio link protocol layers in order to provide adequate radio bearers for real-time services. Examples of the enhancements include support for cell reselection for packet-switched domain, separation

of user and control planes, and transparent modes in radio link protocol layers. It will still be possible to multiplex packet data traffic to and from terminals operating either in IU mode or Gb mode on the same time slot Release 99 of the ETSI standard has shown efficient support for services without strict delay requirements, such as Internet access, file downloads, e-commerce and e-mail. With the standards in 3GPP's Release 5, GERAN will provide a complete range of third generation wireless services. This includes support for all QoS classes specified for WCDMA, including the conversational service class with its real-time requirements. Furthermore, by interfacing to the third generation WCDMA core network over the IU interface that is common with UTRAN, a high level of WCDMA alignment is achieved.

EGPRS introduces a new modulation technique, along with improvements to the radio protocol, that allows operators to use existing frequency spectrums (800, 900, 1800 and 1900 MHz) more effectively. The simple improvements of the existing GSM/GPRS protocols make EDGE a cost-effective, easy-to implement add-on. Software upgrades in the base station system enable use of the new protocol; new transceiver units in the base station enable use of the new modulation technique. EDGE triples the capacity of GPRS. This capacity boost improves the performance of existing applications and enables new services such as multimedia services. It also enables each transceiver to carry more voice and/or data traffic. EDGE enables new applications at higher data rates. This will attract new subscribers and increase an operator's customer base. Providing the best and most attractive services will also increase customer loyalty.

4. ENCRYPTION

The objective of security for GSM system is to make the system as secure as the public switched telephone network. The use of radio at the transmission media allows a number of potential threats from eavesdropping the transmissions. It was soon apparent in the threat analysis that the weakest part of the system was the radio path, as this can be easily intercepted.

The GSM MoU Group produces guidance on these areas of operator interaction for members. The technical features for security are only a small part of the security requirements, the greatest threat is from simpler attacks such as disclosure of the encryption keys, insecure billing systems or corruption! A balance is required to ensure that these security processes meet these requirements.

At the same time a judgment must be made of the cost and effectiveness of the security measures.

4.1 Limitations of Security

Existing cellular systems have a number of potential weaknesses that were considered in the security requirements for GSM.

The security for GSM has to be appropriate for the system operator and customer:

- The operators of the system wish to ensure that they could issue bills to the right people, and that the services cannot be compromised.
- The customer requires some privacy against traffic being overheard.
- The countermeasures are designed:

- To make the radio path as secure as the fixed network, which implies anonymity and confidentiality to protect against eavesdropping;
- To have strong authentication, to protect the operator against billing fraud;
- To prevent operators from compromising each others' security, whether inadvertently or because of competitive pressures.
- The security processes must not:
- Significantly add to the delay of the initial call set up or subsequent communication;
- Increase the bandwidth of the channel,
- Allow for increased error rates, or error propagation;
- Add excessive complexity to the rest of the system,
- Must be cost effective.

The designs of an operator's GSM system must take into account the environment and have secure procedures such as:

- The generation and distribution of keys,
- Exchange of information between operators,
- The confidentiality of the algorithms.

4.2 Descriptions of the Functions of the Services

The security services provided by GSM are:

- **Anonymity** So that it is not easy to identify the user of the system.
- **Authentication** So the operator knows who is using the system for billing purposes.
- **Signaling Protection** So that sensitive information on the signaling channel, such as telephone numbers, is protected over the radio path.
- **User Data Protection** So that user data passing over the radio path is protected.

4.3 Anonymity

Anonymity is provided by using temporary identifiers. When a user first switches on his radio set, the real identity is used, and a temporary identifier is then issued. From then on the temporary identifier is used. Only by tracking the user is it possible to determine the temporary identity being used.

4.3.1 Authentication

Authentication is used to identify the user (or holder of a Smart Card) to the network operator. It uses a technique that can be described as a "Challenge and Response", based on encryption.

Authentication is performed by a challenge and response mechanism. A random challenge is issued to the mobile, the mobile encrypts the challenge using the authentication algorithm (A3) and the key assigned to the mobile, and sends a response back. The operator can check that, given the key of the mobile, the response to the challenge is correct.

Eavesdropping the radio channel reveals no useful information, as the next time a new random challenge will be used. Authentication can be provided using this process. A random number is generated by the network and sent to the mobile. The mobile use the Random number R as the input (Plaintext) to the encryption, and, using a secret key unique to the mobile Ki, transforms this into a response Signed RE Sponse (SRES) (Cipher text) which is sent back to the network.

The network can check that the mobile really has the secret key by performing the same SRES process and comparing the responses with what it receives from the mobile.

networks. These algorithms can all be built using a few thousand transistors, and usually takes a small area of a chip within the mobile.

4.5 World-wide Use of the Algorithms

There are now three different possibilities for GSM, unencrypted, and use of the A5/1 algorithm or the A5/2 algorithm to secure the data. This arose because the GSM standard was designed for Western Europe, and export regulations did not allow the use of the original technology outside Europe. The uses of the algorithms in the network operator's infrastructure are controlled by the GSM Memorandum of Understanding Group (MoU) according to the formula below:

- The present A5/1 algorithm can be used by countries which are members of CEPT.
- The algorithm A5/2 is intended for any operators in countries that do not fall into the above category.

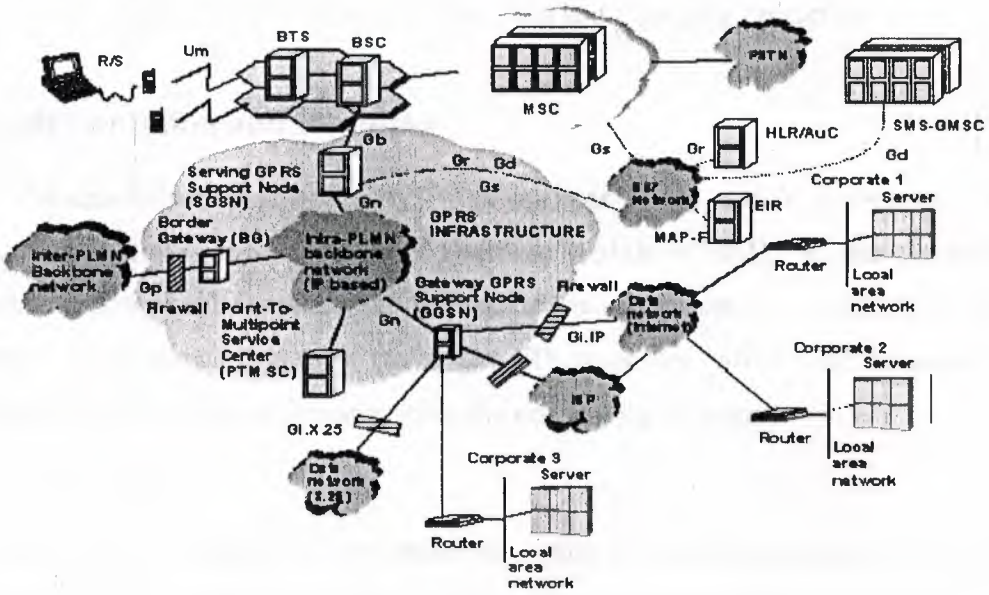

Figure 4.2 World-wide Use of the Algorithms

4.3.2 User Data and Signaling Protection

The response is then passed through an algorithm A8 by both the mobile and the network to derive the key Kc used for encrypting the signaling and messages to provide privacy(A5seriesalgorithms)

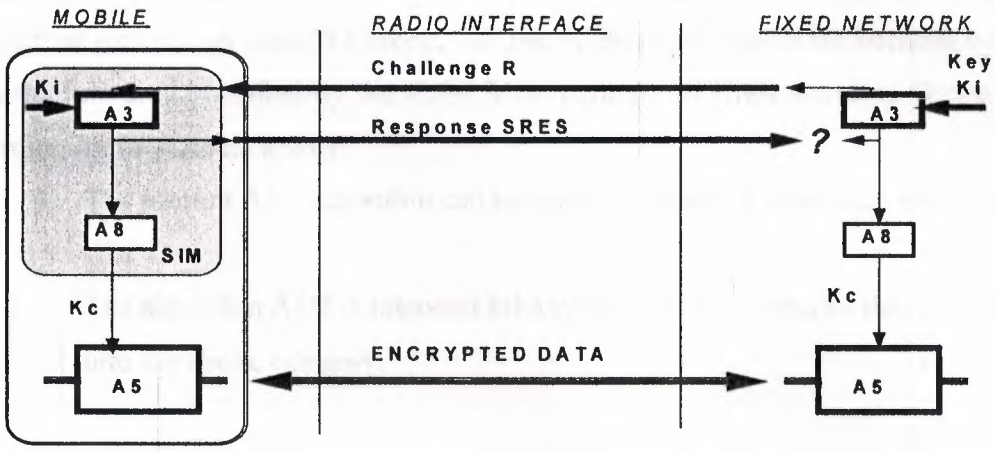

Figure 4.1 Illustration of User Data and Signaling Protection

4.4 Implementation and Roaming

The authentication algorithm A3 is an operator option, and is implemented within the smart card (known as the Subscriber Interface Module or SIM). So that the operators may inter-work without revealing the authentication algorithms and mobile keys (Ki) to each other, GSM allows triplets of challenges (R), responses (SRES) and communication keys (Kc) to be sent between operators over the connecting networks.

The A5 series algorithms are contained within the mobile equipment, as they have to be sufficiently fast and are therefore hardware. There are two defined algorithms used in GSM known as A5/1 and A5/2. The enhanced Phase 1 specifications developed by ETSI allows for inter-working between mobiles containing A5/1, A5/2 and unencrypted

networks. These algorithms can all be built using a few thousand transistors, and usually takes a small area of a chip within the mobile.

4.5 World-wide Use of the Algorithms

There are now three different possibilities for GSM, unencrypted, and use of the A5/1 algorithm or the A5/2 algorithm to secure the data. This arose because the GSM standard was designed for Western Europe, and export regulations did not allow the use of the original technology outside Europe. The use of the algorithms in the network operator's infrastructure are controlled by the GSM Memorandum of Understanding Group (MoU) according to the formula below:

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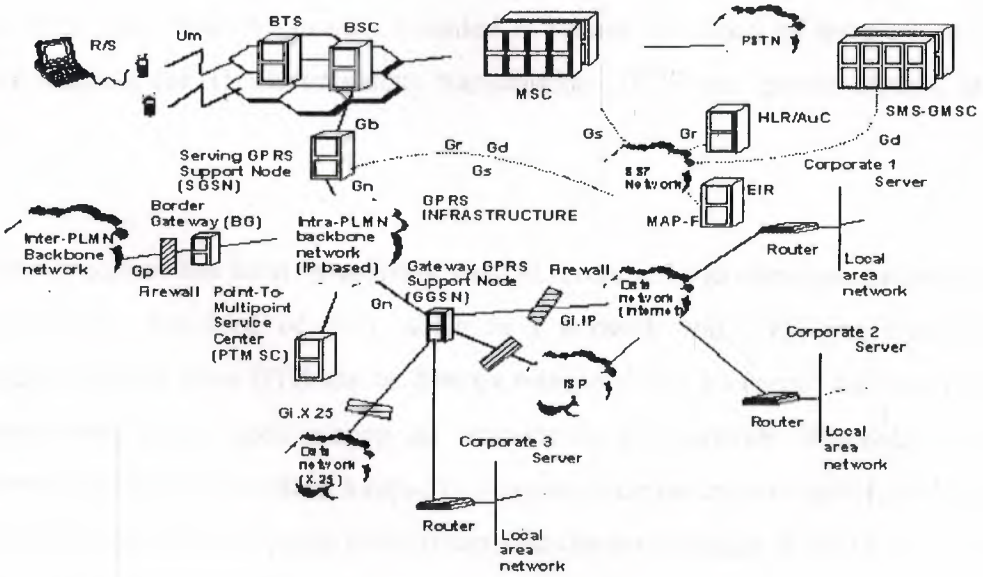


Figure 4.2 World-wide Use of the Algorithms

5. POWER CONTROL

5.1 Introduction

Cellular systems are limited by interference. In these systems, multiple co-Channel interference, though controlled, is a normal situation, and is the main factor to determine the service area. The goal is to allow the higher interference level in order to reuse the available frequencies within the smallest area. As quality of service depends on the carrier/interference ratio (CIR) more than on the signal/noise ratio, there is a certain trade-off between quality and capacity that can be tolerated by the system.

Mobile markets have the larger growing rates among telecom markets. For this growing rate to continue, higher levels of capacity and quality are needed. This means that all possible techniques must be used to improve such features in order to reach a progressive enhancement of the radio and network performance. Current implementation of GSM has some powerful mechanism intended to reduce the effect of interference: slow frequency hopping (SFH), discontinuous transmission (DTX) and power control, among others.

Power control has been extensively studied recently. Discontinuous transmission is closely related to the used of SFH, since in a network with SFH implemented the interference reduction from DTX can be directly translated into a capacity increase because the improvement is averaged among all mobiles in the network. Without SFH, the improvement in CIR will not mean a capacity increase since the mobile stations will benefit differently. Therefore, in this paper we will focus on the performance of the slow frequency hopping as a tool to reduce the interferences in a cellular system.

Most of existing papers about SFH and DTX present software simulations, but very few refer to measurements made with a statistical approach. Therefore, knowledge from real experiences about the behaviour of these features is needed, specially regarding:

- Relationship between the bit error rate (RX_QUAL) and the frame erasure rate (audio quality).
- Changes in measured signal levels and possible changes in the uplink and downlink power budgets.
- Changes in the hand off rates due to the bit error rate increase.
- Changes in the dropped call rate.
- Frequency planning criteria about number of frequencies in the hopping sequence, maximum co-channel and adjacent interference levels allowed by the system, frequency reuse scheme, hopping sequence separation, etc.
- Limitations, real implementation behaviour, etc.

5.2 Frequency Hopping

Slow Frequency Hopping consists in changing the frequency of the channel in every transmitted burst (217 hopes per second) providing frequency diversity and interference averaging. This allows randomising the risk of interference and improving the behaviour of the channel (for the selective fadings). The frequency hopping can be classified in two categories: base band hopping and synthesiser hopping. Synthesiser hopping uses only one transmitter for all burst belonging to a specific connection and the base band hopping uses as many transmitters as frequencies in the hopping sequence. In this work, we have analysed the former because is more efficient and flexible. If we consider the way of changing the frequency, the hopping can be also cyclic or random. The latter is the one studied in this work.

There are many factors affecting the performance of the SFH:

- Number of hopping frequencies. The higher number of hopping frequencies the better system performance as it improves frequency diversity. However, using more than 8

hopping frequencies does not provide a significant improvement due to the GSM interleaving period of 8 burst.

- Hopping frequencies separation. The larger the frequency separation between the hopping frequencies, the better system performance as the effects of propagation becomes more uncorrelated.
- System load. It has a direct influence in the SFH performance. Low system load means lower interference probability in each hopping frequency and therefore, more benefits from SFH.
- Frequency plan. When SFH is activated, a conventional frequency reuse scheme based on a worst case interference situation is spectrally inefficient. Results show the limitations of different frequency plans introducing this additional parameter in the analysis, proposing a reuse scheme. Tighter reuse schemes can be achieved with the use of the SFH providing more capacity.

5.3 Measurement Campaign

In Measurement Campaign we showed a number of tests carried out in the city of Palma de Mallorca (Spain) which was selected as test area, consisting of 7 trisectorial sites serving a total of 21 cells. 19 cells were equipped with 2 carriers and 2 cells with one carrier. Synthesiser random hopping was used, (Reference no.7. Page no.60)

The frequency subsets used for the BCCH carriers frequency plan and the traffic carriers frequency plan are different. Due to the problematic configuration of the city of Palma de Mallorca 22 carriers are used to implement the BCCH frequency plan, so, in a total of 39 carriers assigned to Telefónica Móviles for its GSM network, we have only 17 free frequencies for the traffic carriers frequency plan.

The traffic slots of the BCCH carriers were blocked for all the offered traffic to use the traffic carriers, (the only ones that implement the SFH) having then, a higher traffic load in the hopping system.

The traffic load conditions in which the trials were developed were homogeneous with differences below 10% in the traffic load of the involved cells in the different days of trials. The average traffic per cell per day was 23, 9 E. This means a very high traffic condition for a network with only one carrier (The BCCH carrier was blocked for the traffic).

Two different kinds of trials were performed: statistics trials, to test the performance of SFH under different reuse patterns in a statistical approach using commercial traffic; and specific trials to test some special configurations.

5.4 Statistics Trials

Comparison between different possible reuse pattern were made: 4/12 with no SFH, 1/3 with 4 frequencies in the hopping sequences, 1/3 with 6 frequencies, 2/6 with 3 frequencies (2/6 it is not a theoretical cluster size but was the one used) . These reuse patterns are restricted to the number of free frequencies for the traffic carriers.

In these statistics trials, the same routes around the 7 stations affected by the trials were measured several times and the statistics of these stations were analysed.

5.5 Specific Trials

Specific trials consisted in measuring different routes around two stations with special configuration to test:

- The effect of different frequency separation in the mean received signal.
- Test a neighbouring cell with a BCCH co-channel or adjacent to one frequency included in the hopping sequence, using a different number of hopping frequencies.

5.6 Measurements Results

5.6.1 Statistics trials

The effect of SFH will be analysed for each considered parameter.

5.6.2 Dropped Call Rate

The drooped calls rate for the busy hour is shown in the graphic We can notice that in the case of SFH and 1/3 cluster with 4 carriers in the hopping sequence (FH-1/3-4C), the number of dropped calls is the same (2.6%) as in the case of no hopping, using the same total number of frequencies (12). If we use more carriers (18) like in the 1/3 (6C) and 2/6 (3C) the number of dropped calls is reduced (2.36% and 2.4%).

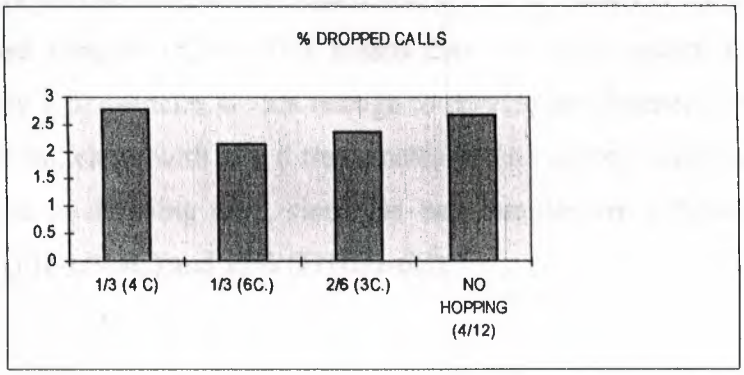


Figure 5.1 Dropped call rate for different cases

5.6.3 Number of Hand-Overs

The total number of handovers per day in the 21 cells of these trials is increased if the SFH is used. As shown in the picture, using the configuration FH-1/3-4C, the number of handover experiments an increase of 29% referred to the case of not using SFH. The increases for the cases FH-1/3-6C and FH-2/6-3C are 27% and 23.4%.

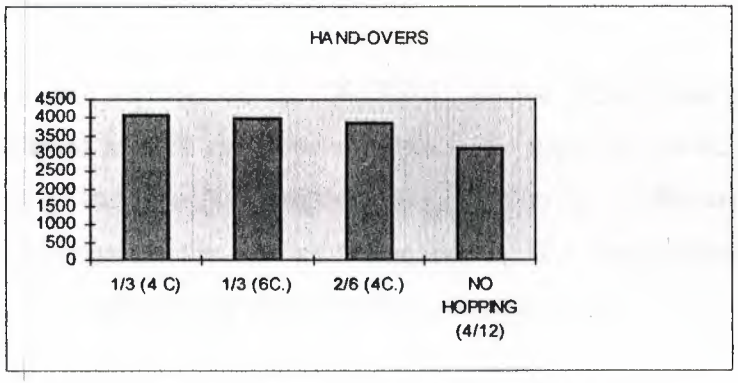


Figure 5.2 Number of hand-overs

5.6.4 Audio Quality

To analyse the audio quality we can use the graphic bellow, showing the average audio quality for the uplink and downlink. Good samples are 90-100 %, acceptable 80% and less than 80% are bad samples. We can notice that in the case FH-2/6-3C, there is a big percentage of bad samples (52%). This means that this reuse pattern is not valid. The reason is that only 3 frequencies are not enough to provide interference diversity. If we use the 1/3 reuse pattern, either with 4 or 6 frequencies in the hopping sequence, the results are better than in the no hopping case, since the bad samples are reduced from 15% (no hopping) to 7% (FH-1/3-4C) and 12% (FH-1/3-6C).

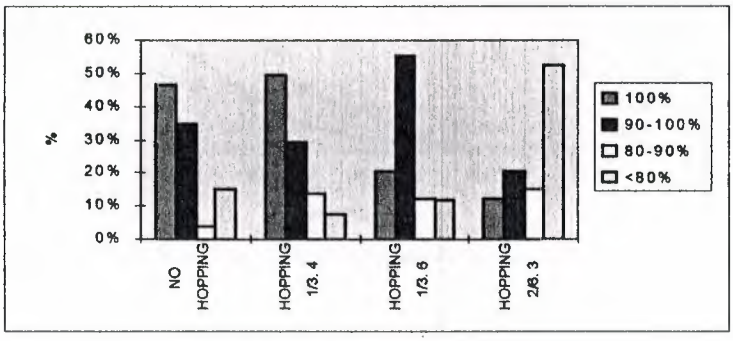
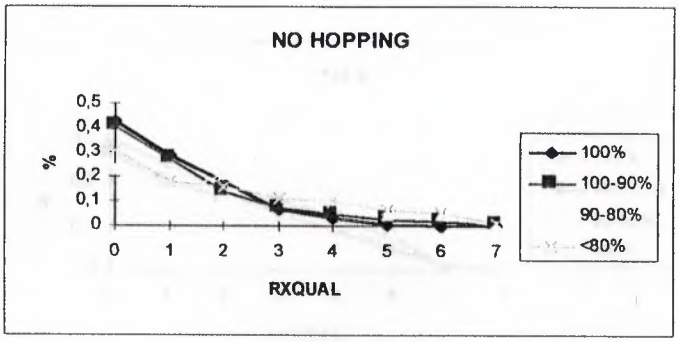
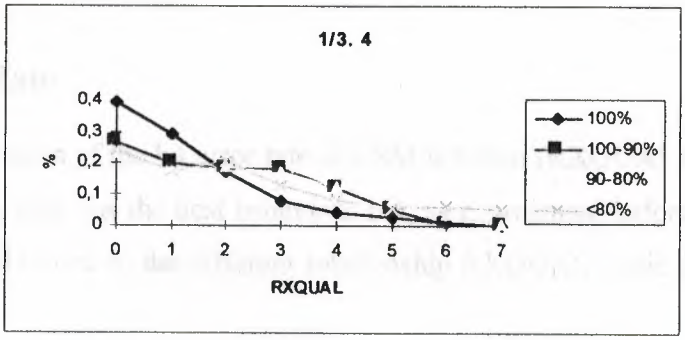


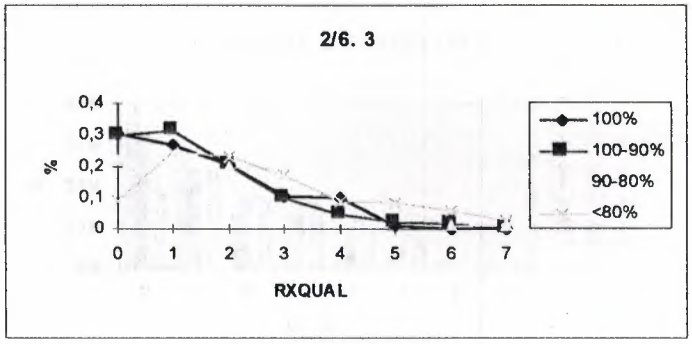
Figure 5.3 Audio quality

The fact of having best results with 4 frequencies than with 6, despite of using less frequencies, is due to not to have enough free frequencies (18) for the pattern 1/3-6C.

In figure 4 the distribution of RX-QUAL against audio quality is shown. The distribution is different in each case. For example in the case FH-1/3-4C. For good audio quality the number of bad RX-QUAL reports (<4) is higher than in the case of no hopping. This tendency is stronger in the case of 6 frequencies. If 3 frequencies are used in the hopping sequence, the behaviour is similar to the no hopping case.







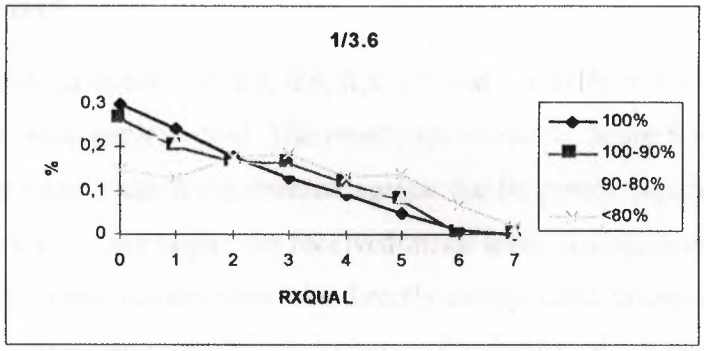


Figure 5.4 Distribution of RXQUAL against audio quality

5.7 Bit Error Rate

The distribution of the bit error rate in GSM notation (RXQUAL) is shown in figure 5. The no hopping case has the best behaviour but, as commented before, it does not mean the best audio quality due to the different relationship RXQUAL-audio quality when using SFH.

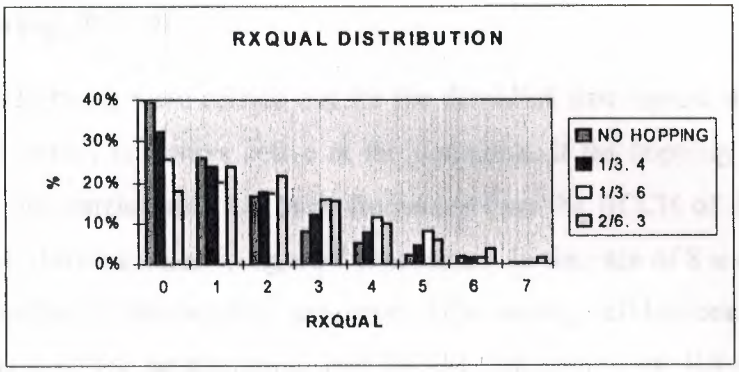


Figure 5.5 RXQual distribution

5.8 Specific Trials

In these trial separations of 0.4, 0.6, 0.8, 1.0 and 1.4 MHz of the frequencies of the hopping sequence were implemented. The results are shown in figure 6 where the received mean level for the same route is represented against the frequency separation. The broader the frequency separation, the higher the received mean level. An improvement of 6 dB can be expected, though these results cannot be directly extrapolated because they are strongly dependant on the propagation conditions (coherent bandwidth of the channel). However a separation of 1 MHz is near the optimum (broader separations have not a considerable improvement). The differences between uplink and downlink are due to the fact that the routes are different for each link.

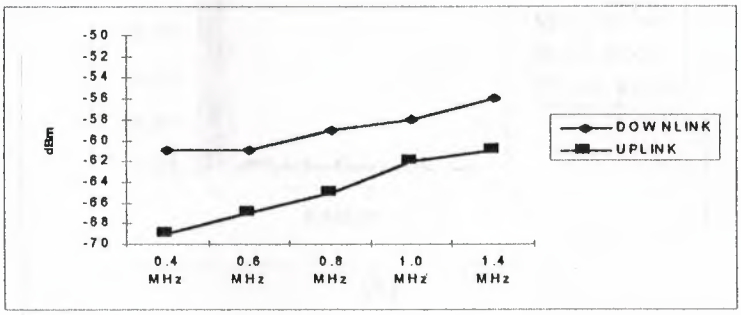
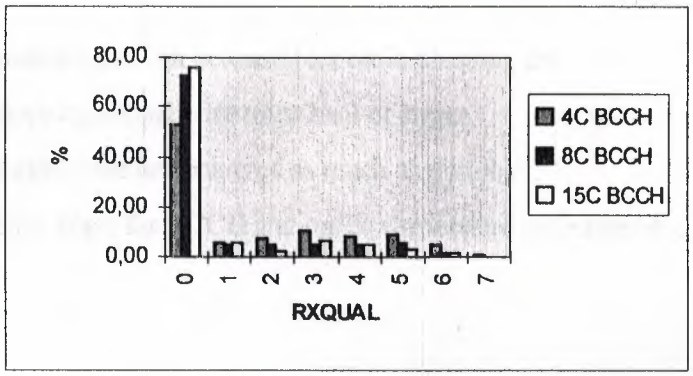


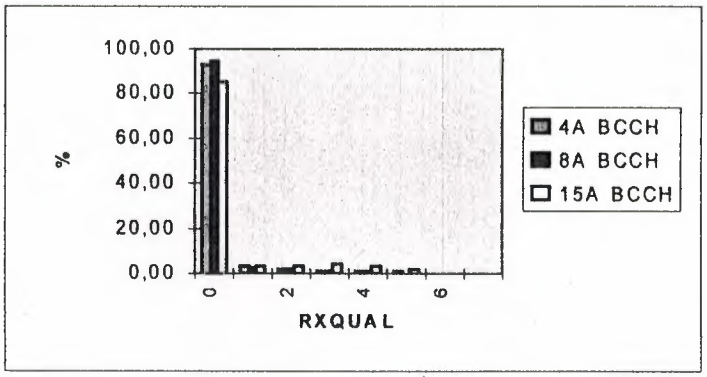
Figure 5.6: Received mean level against frequency separation

5.9 Interfering BCCH

All measurements were carried out for the downlink that represents the worst case since the BCCH carrier is always active in the downlink. If the hopping sequence of the serving cell has one carrier with the same frequency than the BCCH of the neighbouring cell, a RX-QUAL distribution as in figure 7 is achieved. In the case of 8 and 15 frequencies the quality is excellent. If the hopping sequence of the serving cell has one carrier being an adjacent frequency of the neighbouring cell BCCH, the results for RX-QUAL are very good.



(a)



(b)

Figure 5.7 a) RXQUAL distribution with co channel BCCH

b) RXQUAL distribution with adjacent BCCH

CONCLUSION

Use of SFH in a GSM network has been tested. The introduction of this feature provides a significant improvement on system performance regarding:

- Dropped call reduction.
- Increase of the received mean level.
- Possibility of using tighter schemes (like 1/3) providing higher capacity.
- No degradation of audio quality.

Some conclusions that can be useful for radio planning are:

- The number of hopping frequencies must be 4 of larger.
- Hopping frequencies must be separated as much as possible.
- Different frequency plans for BCCH and traffic carriers are recommended.

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