H.N.Pratihari / International Journal of Engineering Research and Applications (IJERA) ISSN: 2248-9622 www.ijera.com
Vol. 2, Issue 3, May-Jun 2012, pp.1541-1549

A study of Interoperability between 3G Systems to GSM

H.N.Pratihari
Department of Electronics & Telecommunication Engineering
Orissa Engineering College, Bhubaneswar-752050

Abstract- 71% of the world's digital mobile communication subscribers use GSM. Almost all these mobile communication subscribers are using GSM but incorrectly many of these are considering that GSM systems are approaching to its end due to the advent of 3G as it is to be voice centric with low performing data capabilities. This article copes with the interoperability between 3G systems, GSM and the original GSM. Evolution of mobile telephone system, definition of GSM and GSM frequency bands are also discussed. This article also presents an overall view on an interworking architecture and represent the introduction of new '3G' technologies in GSM i.e., alignment with UMTS-The IU interface, the IP multimedia subsystem, the new flexible layer one, future radio issues, multimedia broadcast and multicast to prove GSM is a '3G' system in its own right.

1.Introduction

Global system for mobile communication (GSM) is a globally accepted standard for digital cellular communication. GSM is the name of a standardization group established in June 1982. It has two objectives:

Pan-European Roaming: Which offers compatibility throughout the European continent and interaction with the integrated service digital network (ISDN) which offers the capability to extend the single subscribe line system to a multi-service system with various services which are currently offered only through diverse telecommunication networks.

D2: system capacity was not an issue in the initial development of GSM, but due to unexpected rapid growth of cellular service, 35 revisions have been made to GSM since the first issued specification. The first commercial GSM system called D2. It was implemented in Germany in 1992.

2.Evolution of Mobile Telephone Systems

Cellular is one of the fastest growing and most demanding telecommunications applications. Today it represents a continuously increasing percentage of all new telephone subscriptions around the world. Currently there are more than 45 million cellular subscribers worldwide and nearly 50 percent of those subscribers are located in the United States. It is forecasted that cellular system using a digital technology will become the Universal method of telecommunications. By the year 2005, forecasters predict that there will be more than 100 million cellular subscribers worldwide. It has been estimated that some countries may have more mobile phones than fired phone by the year 2000. The concept of cellular service is the use of low power transmission where frequencies can be reused within a geographic area. The idea of cell based mobile radio service was formulated in the United States at Bell Labs in the early 1970s. However, the Nordic countries were first to introduce cellular services for commercial use with the introduction of the Nordic Mobile Telephone (NMT) in 1981. Cellular systems began in the United States with the release of the Advanced Mobile Phone Service (AMPS) system in 1983.

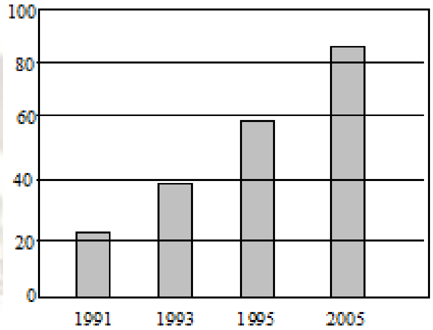

Fig. 1. Cellular Subscriber Growth Worldwide

The AMPS standard was adopted by Asia, Latin America and oceanic countries, creating the largest potential market in the world for cellular. In the early 1980s, most mobile telephone systems wee analog rather than digital, like today's newer systems. One challenge facing analog systems was the inability to handle the growing capacity needs in a cost efficient manner. As a result, digital technology was welcomed. The advantages of digital systems over analog systems include case of signaling, lower level of interference, integration of transmission and switching and increased ability to meet capacity demands.

Table-1. Charts the worldwide development of mobile telephone systems.

Year	Mobile System
1981	Nordic Mobile Telephone (NMT) 450
1983	American Mobil Phone System (AMPS)
1985	Total Access Communication System (TACS)
1986	Nordic Mobile Telephony (NMT) 900
1991	American Digital Cellular (ADC)
1991	Global System for Mobile Comm. (GSM)
1992	Digital Cellular System (DVM) 1800
1994	Personal Digital Cellular (PDC)
1995	PCS 1900 Canada
1996	PCS- United States

2.1. Definition

GSM is a digital mobile telephone system that is widely used in Europe and other parts of the world. GSM uses a variation of TDMA and is the most widely used of the three digital wireless telephone technologies (TDMA, GSM and CDMA). GSM digitizes and compresses data then sends it down a channel with two other streams of user data, each in its own time slot. It operates at either the 900MHz or 1800MHz frequency band. Since many GSM network operators have roaming agreements with foreign operators users can often continue to use their mobile phones when they travel to other countries.

American Personal Communications (APC) a subsidiary of sprint is using GSM as the technology for a broadband personal communication service (PCS). The service will ultimately have more tan 400 base stations for palms iced handsets they are being made by Ericson, Motorola & Nokia.

The handsets include a phone, a text pager and an answering of machine. GSM together with other technologies is part of an evolution of wireless mobile telecommunication that include High Speed Circuit Switched Data (HCSD), GPRS (General Packet Radio System), EDGE (Enhanced Data GSM Environment & UNTS (Universal Mobile Telecommunication Services).

2.2. GSM Frequency Bands

As GSM has grown worldwide, it has expanded to operate at three frequency bands: 900, 1800 and 1900 which is shown in figure-2.

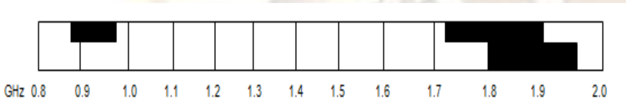

Fig. 2.GSM Frequency Bands

2.2.1 GSM 900

The original frequency band specified for GSM was 900 MHz this band. In some countries and extended version of GSM 900 can be used i.e. called E-GSM. This E-GSM provides extra network capacity, while the primary version is called P-GSM.

2.2.2 GSM 1800

In 1990, in order to increase competition between operators, the U.K. requested the start of a new version of GSM adapted to the 1800 MHz frequency band. Licenses have been issued in severed countries and networks are in full operation.

2.2.3 GSM 1900

As GSM 900 could not be used in North America due to prior allocation of the 900 MHz frequency. GSM 1900 MHz is seen as an opportunity to bridge this gap. The main differences between the American GSM 1900 standard and GSM 900 is that its supports ANSI signaling.

2.2.4 GSM 400

Ericsson and Nokia are aiming to make GSM 450 products available for the market during 2001. GSM 400 also provides NMT system operators a logical way to introduced quality digital services and seamless international roaming possibilities.

3. GSM Network/GSM Architecture

Throughout the evolution of cellular telecommunications various systems have been developed without the benefits of standardized specifications. This presented many problems directly related to compatibility, especially with the development, especially with the development of digital radio technology. The GSM standard is intended to address these problems. From 1982 to 1985 discussion were held to decide between building an analog or digital system. After multiple field tests, a digital system was adopted for GSM. The next task was to decide between a narrow or broad band solution. In May 1987, the narrow band (TDMA) Solution was change. GSM provides recommendations, not requirements. The GSM specifications define the functions and interface requirements in details but do not address the hardware. The reason for this is to limit the designers as little as possible but still it makes possible for the operators to buy equipment from different suppliers. The GSM network is divided into three major systems as shown in figure-3:-

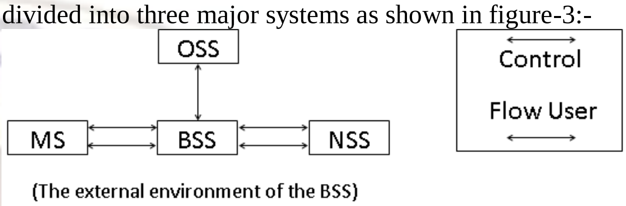

Fig. 3. Base Station Sub Systems

3.1 The Base Station Sub Systems

The BSS connects to the NS through a radio interface and also connects to the NSS. The BSS consists of a base transceiver station (BTS) located at the antennas site and a base station controller (BSS) which may control several BTSs. All radio related functions are performed in the BSS.

BSC: Base station controller it manager all the radio related functions of a GSM network. BSC provides all the control functions and physical links between the NSC and BSC. It is a high capacity that controls the radio frequency (RF) power levels in base transceiver stations. A no at BSCS may be controlled by each NSC.

BTS: It BTS consists of radio transmission and reception equipment i.e., antenna. A group of BTS are controlled by a BSC. GSM uses the open system interconnection (OSI). There are three common interface based on OSI shown in the functional architecture. A common radio interface called air interface between the NS and BTS, an interface A between the NSC and BSC and an A- b is interface between the BTS and BSC.

3.2 Network and switching sub systems

NSS in GSM uses an intelligent network (IN) and it manages the communication between GSM users and other telecommunication uses. The switching system includes the following functional unit which is shown in figure-4 with its function.

3.2.1 Home location register

The HLR is a data base used for storage and management of subscription. It stores permanent data about subscribers, including a subscriber's service profile, location information and activity status.

3.2.2. Authentication center (AUC)

A unit called the AUC which is sub division of HLR. The AUC manages the security data for subscriber authentication and encryption parameters that verify the user's identity and ensure the confidentiality of each call. The AUC protects network operators from different types of fraud found in today's cellular world.

Equipment Identity Register
It is another subdivision of HLR, which stores the data of mobile equipment. EIR prevents calls from stolen, un authorized or defective mobile stations. The AUC and EIR are implemented as stand-alone mode or as a combined node.

3.2.3 Visitor Location Register (VLR)

The VLR in a data base that contains temporary information about subscribers that is needed by the NSC in order to service the visiting subscribers. The VLR is always integrated with the NSC. When a mobile station names into new NSC area, the VLR connected to that NSC would request data about the mobile station from the HLR. Later I the mobile station make a call, the VLR will have the information needed for call setup without having to interfere the HLR each time.

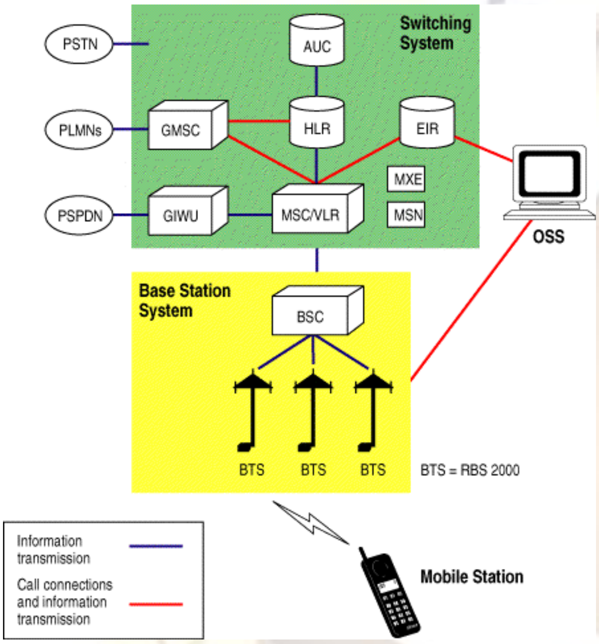

Fig. 4. Network and switching sub systems

3.3 **Network Monitoring Centers**

3.3.1 Operation and Maintenance Center (OMC)

an OMC is a computerized monitoring center which is connected to other network such as NSCs and BSC. In the OMC, staffs are presented with information about the status of the network and can monitor and control a variety of system parameters. There may be one or several OMCS within a network depending on the network size.

3.3.2 NETWORK MANAGEMENT CENTRE

Centralized control of a network is done at a network management center (NMC). Only one NMC is required for a network and this controls the subordinate ONCS. The advantage of this hierarchical approach is that staff at the NMC can concentrate on long term system wide issues, where as local personnel at each ONC can concentrate on short term. ONC and NMC functionality can be combined in the same physical network node or implemented at different locations.

3.3.2.1 The operation and support system

The operations and maintenance center (ONC) is connected to all equipment in the switching system and to the BSC. The implementation of ONC is called the operation and support system (OSS). The OSS is the functional entity from which the network operator monitors and controls the system. The purpose of OSS is to offer the customer cost effective support for centralized, regional and local operational and maintenance activities that are required for a GSM network. An important function of OSS is to provide a network overview and support the maintenance activities of different operation and maintenance organizations.

3.3.2.2 Additional Functional Elements

Other functional elements shown in fig are as follows. *Message Center (MXE)* It is a node that provides integrated voice, fax and data messaging. Specifically, the MXE handles short ménage service, cell broad cast, voice mail, fax mail, e-mail.

3.3.2.3 Mobile Service Node (MSN)

The MSN is the node that handles the mobile intelligent network (IN) services. *Gateway*

3.3.2.4 Mobile Services Switching Center (GMSC)

A gateway is a node used to interconnect two networks. The gateway is often implemented in an MSC. The MSC is then referred to as the GMSC.
In order to set up a requested call, the call is initially routed to a gateway MSC, which finds the correct HLR by knowing the directory number of the GSM subscriber.

3.3.2.5 GSM Inter working Unit (GIWU)

It consists of both hardware and software that provides an interface to various networks for data communications. Through the GIWU, users can alternate between speech and data during the same call. The GIWU hardware equipment is physically located at the MSC/VLR.

3.3.3 *MOBILE STATION (MS)*

An MS is used by a mobile subscriber to communicate with the mobile network. Several types of MSS exist, each allowing the subscriber to mare and receive calls. The range of coverage area of an MS depends on the O/P power of the MS. Different types of MS have different O/P power capabilities and consequently different ranges. For ex hand held MS have a lower output power and shorter range than car installed MS with a roof mounted antenna.
MS consists of:-

- A mobile terminal
- A subscriber identity module (SIM)

ME doesn't need to be personally assigned to one subscriber. The SIM is a subscriber module, which stores all the subscriber related information. When a subscriber's SIM is inserted into the ME of an MS that MS belongs to the subscriber and the call is delivered to that MS. The ME is not associated with a called number. It is linked to the SIM. In this case a subscriber can use any ME when the SIM is inserted in the ME.

3.3.4 GSM Network Areas

The GSM network is made up of geographic areas. As shown in figure-5, these areas include cells, locations areas (LAS), MSC/VLR service areas and public land mobile network (PLMN) area. The cell is the basic unit of a cellular system and is defined as the area of radio coverage given by one BS antenna cycle. The GSM network identifies each cell via the cell global identity (CGI) number assigned to each cell. The location area (LA) is a group of cells. It is the area in which the subscriber is paged. One or more base station controllers serve each LA. Each LA is assigned a location area identity (LAI) number.

3.3.4.1 MSC/VLR

An MSC/VLR service area represents the part of the GSM network that is covered by MSC and which is reachable as shown in figure-5.

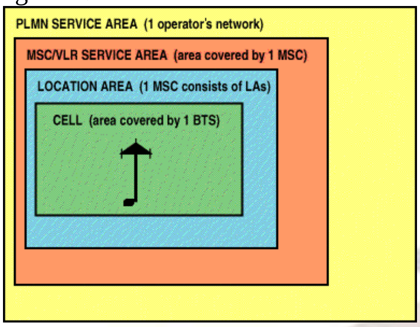

Fig. 5. *GSM Network Areas*

3.3.4.2 PLMN

The PLMN service area is an area served by one network operator and it is defined as the area in which our operator offers radio converge and access to its network.

3.3.4.3 GSM Specifications

Before looking at the GSM specifications, it is important to understand the following basic terms:

- Bandwidth—the range of a channel's limits; the broader the bandwidth, the faster data can be sent
- Bits per second (bps)—a single on-off pulse of data; eight bits are equivalent to one byte
- Frequency—the number of cycles per unit of time; frequency is measured in hertz (Hz)
- Kilo (k)—kilo is the designation for 1,000; the abbreviation kbps represents 1,000 bits per second
- Megahertz (MHz)—1,000,000 hertz (cycles per second)
- Milliseconds (ms)—one-thousandth of a second
- Watt (W)—a measure of power of a transmitter

Specifications for different personal communication services (PCS) systems vary among the different PCS networks. Listed below is a description of the specifications and characteristics for GSM.

- Frequency band—The frequency range specified for GSM is 1,850 to 1,990 MHz (mobile station to base station).
- Duplex distance—The duplex distance is 80 MHz. Duplex distance is the distance between the uplink and downlink frequencies. A channel has two frequencies, 80 MHz apart.
- Channel separation—The separation between adjacent carrier frequencies. In GSM, this is 200 kHz.
- Modulation—Modulation is the process of sending a signal by changing the characteristics of a carrier frequency. This is done in GSM via Gaussian minimum shift keying (GMSK).
- Transmission rate—GSM is a digital system with an over-the-air bit rate of 270 kbps.
- Access method—GSM utilizes the time division multiple access (TDMA) concept. TDMA is a technique in which several different calls may share the same carrier. Each call is assigned a particular time slot.
- Speech coder—GSM uses linear predictive coding (LPC). The purpose of LPC is to reduce the bit rate. The LPC provides parameters for a filter that mimics the vocal tract. The signal passes through this filter, leaving behind a residual signal. Speech is encoded at 13 kbps.

3.3.5 GSM Subscriber Services

There are two basic types of services offered through GSM: telephony (also referred to as teleservices) and data (also referred to as bearer services). Telephony services are mainly voice services that provide subscribers with the complete capability (including necessary terminal equipment) to communicate with other subscribers. Data services provide the capacity necessary to transmit appropriate data signals between two access points creating an interface to the network. In addition to normal telephony and emergency calling, the following subscriber services are supported by GSM:

- Dual-tone multifrequency (DTMF)—DTMF is a tone signaling scheme often used for various control purposes via the telephone network, such as remote control of an answering machine. GSM supports full-originating DTMF.
- Facsimile group III—GSM supports CCITT Group 3 facsimile. As standard fax machines are designed to be connected to a telephone using analog signals, a special fax converter connected to the exchange is used in the GSM system. This enables a GSM–connected fax to communicate with any analog fax in the network.
- Short message services—A convenient facility of the GSM network is the short message service. A message consisting of a maximum of 160 alphanumeric characters can be sent to or from a mobile station. This service can be viewed as an advanced form of alphanumeric paging with a number of advantages. If the subscriber's mobile unit is powered off or has left the coverage area, the message is stored and offered back to the subscriber when the mobile is powered on or has reentered the coverage area of the network. This function ensures that the message will be received.
- Cell broadcast—A variation of the short message service is the cell broadcast facility. A message of a maximum of 93 characters can be broadcast to all mobile subscribers in a certain geographic area. Typical applications include traffic congestion warnings and reports on accidents.
- Voice mail—This service is actually an answering machine within the network, which is controlled by the subscriber. Calls can be forwarded to the subscriber's voice-mail box and the subscriber checks for messages via a personal security code.
- Fax mail—With this service, the subscriber can receive fax messages at any fax machine. The messages are stored in a service center from which the subscriber via a personal security code to the desired fax number can retrieve them.

3.3.6 Supplementary Services

GSM supports a comprehensive set of supplementary services that can complement and support both telephony and data services. Supplementary services are defined by GSM and are characterized as revenue-generating features. A partial listing of supplementary services follows.

- Call forwarding—This service gives the subscriber the ability to forward incoming calls to another number if the called mobile unit is not reachable, if it is busy, if there is no reply, or if call forwarding is allowed unconditionally.
- Barring of outgoing calls—This service makes it possible for a mobile subscriber to prevent all outgoing calls.
- Barring of incoming calls—This function allows the subscriber to prevent incoming calls. The following two conditions for incoming call barring exist: baring of all incoming calls and barring of incoming calls when roaming outside the home PLMN.
- Advice of charge (AoC)—The AoC service provides the mobile subscriber with an estimate of the call charges. There are two types of AoC information: one that provides the subscriber with an estimate of the bill and one that can be used for immediate charging purposes. AoC for data calls is provided on the basis of time measurements.
- Call hold—This service enables the subscriber to interrupt an ongoing call and then subsequently reestablish the call. The call hold service is only applicable to normal telephony.
- Call waiting—This service enables the mobile subscriber to be notified of an incoming call during a conversation. The subscriber can answer, reject, or ignore the incoming call. Call waiting is applicable to all GSM telecommunications services using a circuit-switched connection.
- Multiparty service—The multiparty service enables a mobile subscriber to establish a multiparty conversation—that is, a simultaneous conversation between three and six subscribers. This service is only applicable to normal telephony.
- Calling line identification presentation/restriction—These services supply the called party with the integrated services digital network (ISDN) number of the calling party. The restriction service enables the calling party to restrict the presentation. The restriction overrides the presentation.
- Closed User Groups (CUGs)—CUGs are generally comparable to a PBX. They are a group of subscribers who are capable of only calling themselves and certain numbers.

3.4 GSM Transmission Process

Stage –1: Analog to Digital (A/D) Conversion

One of the primary functions of an MS is to convert the analog speech information into digital signal. The analog to digital (A/D) conversion process outputs a collection of bits, binary ones and zeros which represent the speech input as in figure-6.

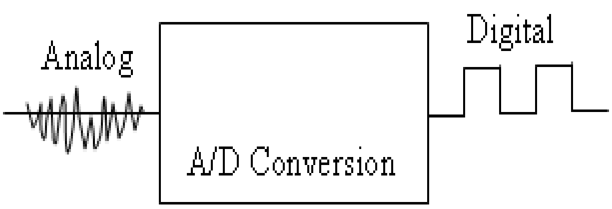

Fig. 6. A/D conversion

The A/D conversion is performed by using a process called Pulse Code Modulation (PCM). PCM involves three main steps.

- Sampling
- Quantization
- Coding

Step 1: Sampling

Sampling involves measuring the analog signal at specific time intervals as shown in figure-7.

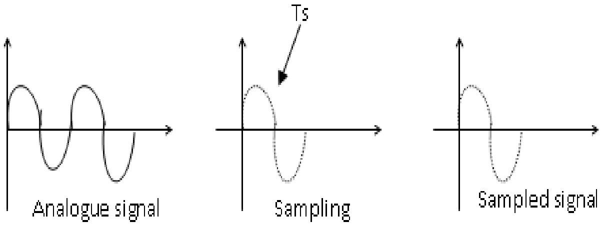

Fig. 7. Analog signal sampling

The accuracy of describing the analog signal in digital terms depends on how often analog signal is sampled. This is expressed as the sampling frequency. The sampling theory states that:

"To reproduce an analog signal without distortion the signal must be sampled with at least twice the frequency of the highest frequency component in the analog signal".

Step 2 : Quantization

The next step is to give each sample a value. For this reason, the amplitude of the signal at the time of sampling is measured and approximated to one of a finite set of values. The figure-8 below shows the principle of quantization applied to an analog signal. It can be seen that a slight error is introduced in this process when the signal is quantized or approximated. The degree of accuracy depends on the number of quantization levels used. Within common telephony, 256 levels are used while in GSM 8,192 levels are used.

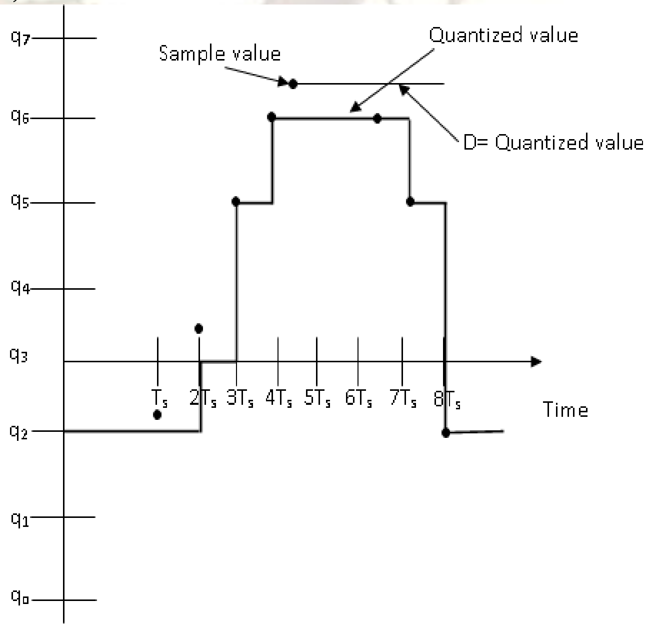

Fig. 8. Quantization Process

Step 3 : Coding:

Coding involves converting the quantized values into binary. Every value is represented by a binary code of 13 bits (2 = 8192).

Stage 2: Segmentation
The key to reducing the bit rate is to send information about the speech instead of the speech itself.
Stage 3: Speech coding
In GSM, the speech coding process analyzes speech samples and outputs parameters of what the speech consists of the tone, length of tone, pitch etc. This is then transmitted through the network to another MS, which generates the speech based on these parameters.
STAGE 4: CHANNEL CODING
Channel coding in GSM uses the 260 bits from speech coding as input to channel coding and outputs 456 encoded bits as shown in figure-9. The 260 bits are split according to their relative importance. The first block of 50 bits is sent through a block coder, which adds three parity bits that will result in 53 bits. These three bits are used to detect errors in a received message. The 53 bits from first block, the 132 bits from the second block and 4 tail bits (total = 189) are sent to a 1:2 convolution coder which outputs 378 bits. Bits added by the convolution coder enable the correction of errors when the message is received.
The bits of block 3 are not protected.

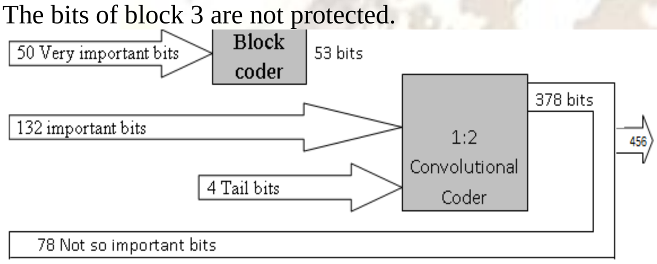

Fig. 9. Channel Coding
STAGE 5: INTERLEAVING
First level of interleaving
The channel coder provides 456 bits for every 20 ms of speech. These are interleaved, forming eight blocks of 57 bits each, as shown in the figure-10 below.

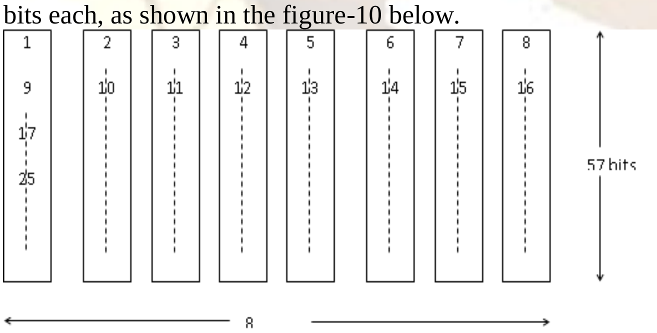

Fig. 10. Interleaving of 20 ms of encoded speech
In a normal burst there is space for two of these speech blocks of 57 bits, as can be seen in figure-11. (The remaining bits in the burst are explained later in this book). Thus, if one burst transmission is lost, there is a 25% BER for the entire 20 ms of speech (2/8 = 25%).

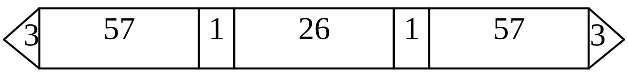

Fig. 11. Normal burst

STAGE 6: CIPHERING/ENCRYPTION
The purpose of ciphering is to encode the burst so that it cannot be interpreted by any other device than the intended receiver. The ciphering algorithm in GSM is called the A5 algorithm. It does not add bits to the burst, meaning that output to the ciphering process is the same as the input 456 bits per 20 ms.
STAGE 7: BURST FORMATTING
As previously explained, every transmission from an MS/BTS must include some extra information such as the 26 training sequence bits, 2 flag bits and 6 tail bits. The process of burst formatting is to add these bits to the basic speech/data (57 + 57 = 114 bits) being sent. Consequently this increases the burst from 114 to 148 bits, thus increasing the transmission rate on the air, but is necessary to counteract problems encountered on the radio path.
STAGE 8: MODULATION & TRANSMISSION
The bits must then be sent over the air using a carrier frequency. As previously explained, GSM uses the GMSK modulation technique.

3.5 **What is 3G?**

But nowadays many in correctly consider the system as approaching its end particularly with the advent of 3G, because it is thought to be voice center with low performing data capabilities. 3G technologies means companies developing 3G equipment envision users having the ability to receive live music, conduct interactive web session have simultaneous voice and data access with multiple parties at the same time using a single mobile handset, whether driving, walking or standing still in an office setting.
The most recent manifestation of the GSM radio access network, “GSM/EDGE Radio Access Network” (GERAN) is now considered by the International Telecommunications Union as a 3G technology in its own right. It is standardised by the same organisation as UTRAN (UMTS Radio Access Network), and has been positioned as a complementary component of the overall UMTS (Universal Mobile Telecommunications System) communications system. Originally standardised in the 1980’s, GSM has been extensively modified and enhanced, as the requirements of users and mobile communications system operators have progressed. The technical specifications were originally written and managed by the European Telecommunications Standards Institute, but this role moved in August 2000 to the 3rd Generation Partnership Project (3GPP). The 3GPP is also responsible for the global UMTS third-generation mobile communications system. Their approach has been to harmonies the wideband CDMA based UTRAN and TDMA based GERAN radio access networks into one overall third generation communications system, the UMTS.

3.3.1 INTRODUCTION OF 3G TECHNOLOGIES

In general, advancements of the original GSM system have focused on improving throughput, spectral efficiency and flexibility of the system, whilst striving to keep infrastructure upgrade costs at a minimum Continual improvements have been made to the original GSM system, and some of the major ones are:
Release 1: marked the point where GSM moved away from being a purely circuit-switched, voice centric communications system. The general packet radio system (GPRS) was introduced to GSM.

Release 2: a brand new voice coding scheme was introduced to GSM, the adaptive multi-rate voice code scheme, and support for location-based services over circuit-switched connections.

Release 3: "Enhanced data rates for GSM evolution" (or "EDGE") component introduced, bringing a new radio modulation scheme to GSM, using 8-PSK modulation to almost triple the maximum available raw data rate.

Release 4: handover of the GSM standardization process to the 3GPE and introduction of the 3GPP based standards naming convention.

Release 5: the "GERAN" nomenclature is used to designate truly third-generation radio access network based on GSM technology.

Release 6 of the GERAN specifications is currently being completed by the 3GPP.

The major advancements included in each release, and their benefit to both end user and mobile communications system operator, are summarized in table 1. Regarding commercial realization, Release 97 based products have been available for approximately two years now, whilst Release 99 based products have recently been demonstrated by a number of manufacturers, and are expected on the market during this year. The Release 5 standards were finalized in June 2002, and thus Release 5 based systems are not expected to be commercially deployed for another two or three years. The UMTS Release 6 specifications are currently being completed by the SGPP, with the specifications likely to be completed and released by the end of 2003.

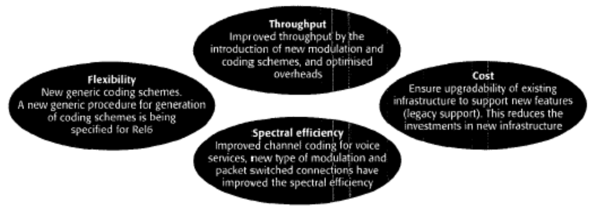

Fig. 11. advancements of the original GSM System

3.6 ALIGNMENT WITH UMTS-THE IU INTERFACE

A significant difference between UMTS and the original (pre release 5) GSM based communications system is the design of the interfaces between the radio access aspect of the system, and the core network. Here, UMTS uses the so-called "Iu interface", a collection of protocols, procedures and messages that define interaction between the radio access network and core network.

It is clearly beneficial for the entire UMTS communications system to have a similar set of services available from both the GERAN and UTRAN radio access technologies. The Iu interface is not part of the pre release 5 GSM communications network, and thus two options existed to evolve GERAN so it can provide the same services as UTRAN - the incumbent radio access to core network interfaces already in the system could be advanced to support the same services available over the Iu interface, or alternatively the Iu interface could be added to GERAN, and its' operation modified to co-operate with the procedures and protocols of the Iu interface.

The second option proved to be the most straightforward and was selected by the 3GPP as the way forward. Recently, a significant amount of design and specification work has focused on introducing the Iu interface to GERAN, with this work being finalized in the latest 3GPP release, Release 5. With the introduction of the Iu interface, GERAN can provide the same services as UTRAN, using the same interface from the core network to both radio access technologies. Furthermore, introduction of the Iu interface improves the security provision, and ensures tight integration between GERAN and UTRAN.

The new interface structure is shown in figure 2. The figure shows three interfaces between GERAN and the core network- the Iu interface, and the "A" and "Gb" interfaces, the latter two being second-generation interfaces that have been retained for backwards compatibility with second-generation networks. The intention is that third generation services will be provided across the Iu interface, whilst legacy equipment and services will be supported over the A and Gb interfaces.

3.7 THE IP MULTIMEDIA SUBSYSTEM

One of the major design ideals for future mobile communications systems is that the same Internet Protocol (IP) based multimedia and mixed media interactive communications services available on the fixed Internet be available in a mobile context, with the end user completely unaware of the access technology they are being served through.

Technically, seamless support of Internet services in a mobile communications system involves a significant number of challenges. Ideally, in order to support the very same applications and services as the fixed Internet, the mobile communications system should at least be able to support the same quality of service (QoS) mechanisms and transmission protocols as on the fixed network, and preferably, should fully adopt these same mechanisms so that such support is seamless and smooth. Straightforward in concept, such an ideal is not so simple in practice - support of k h mechanisms requires significant additions t{ the system, in order to remain reliable whilst operations such as call control and handover occur invisibly in the background.

One of the recent introductions to the overall UMTS communications system is the IP multimedia subsystem, or IMS - a new framework allowing Internet based multimedia services on UMTS, independent of the mobile communications specific functions occurring in the background. Whilst current mobile communications systems can support voice and basic packet-data based services, the IMS is designed to allow a mobile communications system to flexibly support virtually any service, from those currently existing on the fixed Internet, to future service that are unanticipated.

When considering GERAN, one of the fundamental issues regarding the support of IMS services is whether GERAN can provide a way of efficiently carrying IMS services across the GERAN radio interface. The term "bearer" is given to a mobile communications system's ability to carry traffic across the radio interface-for example, a 3G mobile communications system might be expected to

provide bearers for supporting voice traffic, streaming video traffic, and HTML-based packet data traffic. When we consider the inherent flexibility and variety of IMS services, it is clear that to efficiently support IMS services, GERAN must provide flexible radio bearers suited to carrying a wide variety of traffic, both known and unanticipated.
Such flexible and efficient support of a large variety of services is currently difficult for GERAN. Currently, the radio bearers available are pre-defined in the standards. When GERAN is expected to provide a particular service, for example circuit-switched voice communications, then the radio bearers required to support such a service have been specified in the GERAN standards. The advantage of this approach is that the configuration and parameters that define the radio bearer for this service can be carefully designed, so that appropriate QoS for the service is available, and support for the service is provided in an efficient manner. The disadvantage is that every service required of GERAN needs to be known, so that radio bearers for it can be designed and specified in the standards. Whilst this disadvantage is partially helped by the inclusion of a number of generic (GPRS, etc) radio bearers in GERAN - radio bearers designed to support a range of packet data services-it is foreseeable that such generic radio bearers may not be able to efficiently provide the appropriate QoS for a certain IMS application. The solution to this problem is to introduce new flexible radio bearers to GERAN.

3.7.1 THE NEW FLEXIBLE LAYER ONE

Generally, a radio bearer is defined by the particular physical protocol layer configuration (coding schemes, modulation types, etc) used to carry the particular traffic intended for a bearer. In saying that the radio bearers for a particular service have been specified in the GERAN standards, it is actually the physical layer configurations that are defined for a particular traffic type. Thus, flexible radio bearers are provided by a new flexible physical layer scheme, where the physical layer can be re-configured to match the quality of service (QoS) requirements of the traffic to be carried at any particular time. This new physical layer has been named the “Flexible Layer One” (FLO), and overcomes the flexibility problems encountered with the current GERAN physical layer. It allows operators to introduce new services very quickly, without the delays associated with standardization activities, and helps to “future proof” the GERAN standard. Effectively, with FLO the physical layer is completely flexible to be configured in whatever way necessary to provide an efficient radio bearer able to meet the QoS requirements of a particular service. As well as minimizing the work required in standardization at the introduction of new services, FLO has additional advantages such as easing the implementation of new services to the mobile terminal, improving physical layer performance and enhancing the traffic multiplexing capability of mobile terminals.
Like the Iu interface, FLO is a significant change to the GERAN architecture -the introduction of an entirely. New physical layer affects the entire protocol stack. In particular, the multiple access and link control protocols are affected, generally requiring the introduction of advanced (and improved) procedures to cope with FLO. The 3GPP is currently standardizing FLO, with the intention that FLO will be a major component of release 6 of the UMTS standards.

3.7.2 FUTURE RADIO ISSUES

Although a large number of features have been introduced in GERAN, the only major enhancement to the radio interface is the introduction of EDGE in release 99. This is mainly because the radio interface has so far been considered sufficient to support traditional services, and the capacity of the GERAN system has been adequate. However, with the recent deployment of GSM/EDGE in the United States and Asia-Pacific regions (areas with limited GSM spectrum allocation), the capacity available in GSM networks is now starting to be inadequate, and therefore a lot of work is currently ongoing to improve the GSM network capacity One of the most promising techniques for improving the GSM network capacity is the “single antenna interference cancellation” (SAIC) technique.
The capacity in most GSM networks today is limited by the handset performance in interference limited scenarios, which reduces the ability to deploy aggressive frequency planning, or in other words, to have tight frequency reuse. One method of improving this situation is to use stronger channel coding, thereby enabling operation in more hostile interference environments. Currently, this is the main motivation for operators to deploy and use the adaptive multi-rate voice code scheme, instead of conventional fixed rate speech codes. An alternative or supplementary method to enable operation in hostile interference environments is to introduce active downlink interference suppression or cancellation in the handset. Although interference cancellation and suppression techniques are already used (by the base station) on the uplink, these cannot be applied on the downlink because they are based on diversity reception, which due to cost and complexity issues is currently not considered in GSM mobiles. Instead, interference cancellation for single antenna mobiles is being investigated in the GERAN standardization group, targeted for inclusion in release 6 of the UMTS standards.
The lack of receive diversity in single antenna handsets makes it difficult to reuse the techniques developed and used on the uplink, and instead it has been necessary to develop completely new algorithms. So far, the SAIC algorithm development and standardization work is concentrated on cancellation of GMSK modulated interference, since voice traffic operating with GMSK modulated coding schemes is expected to remain the dominant traffic in GSM for a long while yet. Although the SAIC feasibility study is still ongoing in the GERAN standardization group, initial investigations have demonstrated considerable link level performance improvement when using SAIC, enabling SAIC mobiles to operate in’ interference limited environments where conventional mobiles would fail. It is expected that this enhanced interference robustness can be used to tighten the frequency reuse of GSM systems, significantly improving network capacity From an operator perspective,

the main advantage of SAIC is that it can be deployed in existing networks without requiring any network changes, although the strongest gains available from SAIC require timeslot synchronization and a re-planning of the network.
In addition to these features, a number of services are being considered for GERAN and the UMTS that are intended to advance the capability of the overall communications system. One of the most important GERAN relevant services to be included in the next release of the UMTS standards is the multimedia broadcast and multicast service.

3.7.3 MULTIMEDIA BROADCAST AND MULTICAST

In the wider communications community, significant interest has recently turned towards point-to-multipoint communications techniques, allowing services such as audio and video broadcasting, push services and announcements, and so on. With such interest in these services, it is clearly important similar services are available in the mobile communications context. Whilst a basic functionality for such multicast and broadcast communications is included in earlier releases of the UMTS specifications, in the "IP multicast service", and "cell broadcast service" components, these techniques are not suitable for efficient multipoint distribution of multimedia services - the cell broadcast service can only handle low data rate services, and the IP multicast service, whilst able to handle high data rate services, does not offer any resource savings over using a collection of dedicated point-to-point connections. With this in mind, the 3GPP is currently standardizing the multimedia broadcast and multicast service (MBMS) component, a new UMTS component designed to allow the provision of truly multimedia point-to-multipoint services in an efficient way MBMS will require a number of changes to the UMTS - not only will mobile terminals and other parts of the communications network need to understand the new messages and procedures used, MBMS also introduces new multicast and broadcast network server. entities to the UMTS system. Standardization of the MBMS component is currently ongoing in the 3GPp and MBMS is expected to be a part of Release *6* of the UMTS standards.

4. CONCLUSION

This article has described the interworking architecture of original GSM.
With a great deal of excitement and attention has been paid to third generation mobile communication systems in general, it is often not realized that the TDMA-based GSM communications system has continued to advance beyond its second-generation roots, and is now part of the overall UMTS third-generation mobile communications system. The most recently standardized GSM-based communications system, GERAN, can provide the same services as its wideband-CDMA based counterparts, and does so in a cost effective and evolutionary manner. Recent additions to the system ensure that it can work alongside its wideband-CDMA counterpart UTRAN, and the two together will form a seamless third generation communications system, offering the same services as the fixed network, with the underlying radio access technologies remaining invisible to the end user.
Further state of the art communications technologies that improve GERAN's flexibility, spectral efficiency, service provision and point-to-multipoint functionality are now being standardized by the 3Gp and will be part of the next UMTS standards release. With the introduction of these latest technologies, and continued consideration of other technologies of the future, the original GSM based communications system will remain a relevant and important mobile communications system for many years to come.

REFERENCES

1. S. Kapukun, J.P. Hubaux, "Secure Positioning of Wireless Devices with Application to Sensor Network", in Proceedings of the 24th Annual Conference of the IEEE Computer and Communication Socities - INFOCOM'05, 13-17 March, Miami, Florida, USA, Vol.-3, 2005, pp.-1917-1928.
2. S. Lareto, T.Mecklin, M. Opsenia, et al. "IMS Service Development API and testbed", IEEE Communication Magazine, Vol. 48, no. 4, pp.-26-32, 2010.
3. A. R. Mishra (2004), "Radio Network Planning and Optimization", Johan Wiley & Sons, Ltd, ISBN:0-470-86267-X, pp-19-54,
4. David Amzallag, Michael Livschitz, Joseph (Seffi) Naor, Danny Raz (2006), "Cell planning of 4G cellular Networks: Algorithm Technique and Results", pp-1-5.
5. Clint Smith, Daniel Collin (2002), "3G Wireless networks", Tata Macgrawhill
6. JALAL JAMAL HAMAD-AMEEN (2008), "Cell Planning in GSM Mobile", ISSN: 1109-2742, pp-393-398.
7. Jain Ajay (1987), "Cell Planning in Mobile Communication, IEEE, ICPWC'96" ,pp-190-194.
8. Rayat, Neha, "Cell Planning with Capacity Expansion in Mobile" (July 21, 2009). Available at SSRN: http://ssrn.com/abstract=1437140
9. Stuart Allen, Bachir Balloul, Steve Hurley, Simon Saunders, Roger M. Whitaker (2002), "Smart Cell Planning and Optimization for UMTS", pp-34-38.
10. Young Ha Hwang, Sung-Kee Noh, and Sang-Ha Kim (2006), "Determination of Optimal Cell Capacity for Initial Cell Planning in Wireless Cellular Networks", KIPS (ISSN: 1738-8899).