MULTICASTING OF ADAPTIVE MULTISESSIONS ON GPRS PLATFORMS

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Abstract

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With the fast development of new services such as video streaming, and the oncoming leap to the fourth generation of mobile communications a new challenge has arisen. How can the limited radio and core network resources for cellular mobile systems be utilized to provide the required resources for new services?

In a time when the end-users mobility becomes more and more important it brings an additional significance and demand of research and development in this area.

This thesis is focused on multicast technology as a way to optimize the available network utilization and improve the performance of cellular networks. Global System for Mobile Communications (GSM) and General Packet Radio Service (GPRS) system structure and technology are described in detail. Information for different types of multicast protocols are provided and implementations of GPRS platforms are discussed.

We have demonstrated the positive influence of multicast implementation on GPRS platforms by theoretical arguments and simulation results. We have provided the evidence of the contribution that multicast technology will enhance the network performance, which can be utilized to justify its further development and implementation on existing networks.

Acknowledgements

I would like to express my genuine gratitude to my supervisor Dr. A.K.Elhakeem, for his valuable guidance and support during my research. His outstanding knowledge has encouraged me to further expand my acquaintance in new fields and furthermore bring me a greater understanding of new concepts and technologies which will contribute to my further work.

My deep gratitude is due to my family for their patience and understanding during the hard periods of my multiple studies.

I would like to thank all of my friends who helped motivate me to achieve my goals.

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List of Acronyms

3GPP	Third Generation Partnership Project
AC	Authentication Center
ACCH	Associated Control Channel
AGCH	Access Grant Channel
ARFCN	Absolute Radio Frequency Channel Number
BCCH	Broadcast Control Channel
BCH	Broadcast Channel
BGMP	Border Gateway Multicast Protocol
BM-SC	Broadcast/Multicast Service Center
BSC	Base Station Controller
BSS	Base Station System
BSSGP	Base Station Subsystem GPRS Protocol
BTS	Base Transceiver Station
CBT	Core-Based Tree
CCCH	Common control channel
CCH	Control Channel
CCU	Channel Codec Unit
CN	Core Network
DCCH	Dedicated Control Channel
DVMRP	Distance Vector Multicast Routing Protocol
EIR	Equipment Identity Register
ETSI	European Telecommunications Standards Institute
FACCH	Fast Associated Control Channel
FCCH	Frequency Correction Channel
FDMA	Frequency Division Multiple Access
GGSN	Gateway GPRS Support Node
GMSK	Gaussian Minimum Shift Keying
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communication
GTP	GPRS Tunneling Protocol
H-DVMRP	Hierarchical DVMRP
HLR	Home Location Registe
H-PIM	Hierarchical PIM
IGMP	Internet Group Management Protocol
IMGI	International Mobile Group Identity
LAI	Local Area Identity
LLC	Logical Link Control

MAC/RLC	Medium Access Control/Radio Link Control
MANET	Mobile Ad hoc Network
MBMS	Mobile Broadcast/Multicast Service
MM	Mobility Management
MMARP	Multicast MANet Routing Protocol
MOSPF	Multicast Extension to Open Shortest Path First
MPE_LTP	Multi-Pulse Excited Long Term Prediction Codec
MSC	Mobile-Service Switching Center
N-PDU	Network Protocol Data Units
NSAPI	Network Service Access Point Identifier
OCBT	Ordered Core Based Tree
PCH	Paging Channel
PCU	Packet Control Unit
PDP	Packet Data Protocol
PIM	Protocol Independent Multicas
PIM-DM	PIM Dens Mode
PIM-SM	PIM Sparse Mode
PLMN	Public Land Mobile Network
P-TMSI	Packet - Temporary Mobile Subscriber Identity
RACH	Random Access Channel
RAI	Routing Area Identity
RAN	Radio Access Network
RELP	Residual Excited Linear Predictive
REP_LTP	Regular Pulse Excited Long Term Prediction
RF	Radio Frequency
RFCH	Radio Frequency Channel
RIP	Routing Information Protocol
RP	Rendezvous Pints
SACCH	Slow Associated Control Channel
SCH	Synchronization Channel
SDCCH	Standalone Dedicated Control Channel
SGSN	Serving GPRS Support Node
SIR	Signal-to-Interference Ratio
SNDCP	Subnetwork Dependent Convergence Protocol
STP	Spanning Tree Protocol
TCH	Traffic Channel
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TLLI	Temporary Logical Link Identity
TRAU	Transcoding and Rate Adaptation Unit
VLR	Visitor Location Register

Chapter 1

Introduction

1.1 Motivation and Scope

Every newly developed technology faces the challenge to justify it's potential and become alive in a real working environment. Combining two of the leading new technologies, GPRS and multicasting in the field of mobile communications makes this challenge even more exigent. At a time when radio and core network resources are utilized at maximum, multicasting has the potential to improve overall resource utilization and as result to provide more channels for voice or data transmission. This also significantly improves the quality of the service and allows applications with high bandwidth demands such as streaming video and video conferencing to be widely promoted.

In this thesis we will focus on studying the impact of multicast implementation on GPRS platforms. From the analyses of the output results from the system simulator we will determine the multicast influence on a GPRS network for different network loads, number of users and multicast density modes. We have studied the impact of packet delays and packet overflow which is in close relation to the Quality of Service provided by the system.

1.2 Thesis Organization

In chapter 2 we introduce the basic concepts of GSM networks, the elements of the network structure, and radio resource management. Furthermore we explain the basic concepts of the GPRS networks as based on the GSM architecture and as step to the next generation of mobile communications.

In chapter 3 were discuss the benefits of multicast implementation on mobile networks in particular GPRS networks. Different type of multicast protocols were introduced. GPRS structure modification and basic element functionality for multicasting was discussed. This chapter also includes Mobility Management and Session Management information and flow charts.

In chapter 4 the simulation algorithms and different input parameters are defined. Based on the simulation outputs analyses we will determine the effects of multicast implementation on GPRS platforms. Conclusions were made in regards to network resource utilization, delays and packet overflow.

In chapter 5 we summarize the thesis contribution, further areas of research were suggested.

Chapter 2

Overview of GSM and GPRS systems

2.1 Introduction to mobile networks GSN and GPRS

In the present climate of increasing demand of modern telecommunications to be available wherever and whenever they are needed, mobile service providers and suppliers of telecommunication equipment face new challenges. In order for this demand to be met at an international level the first European Telecommunications Standards Institute (ETSI) and later Third Generation Partnership Project (3GPP) have specified the GSM (Global System for Mobile Communication).

The basic requirements that every cellular radio system must fulfill are:

- Good spectrum efficiency
- High traffic performance
- Cost effectiveness

2.1.1 Spectrum efficiency

To achieve optimum spectrum efficiency we have to consider two major requirements. The radio channel spacing (the width of a frequency band assigned to a radio channel), should be large enough to guarantee good voice/data transmission quality. Enough bandwidth should be provided to carry the data together with error control (forward error correction), synchronization and control information. On the other side we are looking to minimize the bandwidth and to make the media available for more customers and free network resources for new applications which are becoming more and more attractive to mobile subscribers.

An excellent way of improving spectrum efficiency is provided by Time Division Multiplexing (TDM). The technique is based on more then one signal or bit streams to be transferred apparently simultaneously as sub-channels in one communication channel which results in effective utilization of the limited radio channels.

2.1.2 Traffic performance

For better channel utilization on the cellular mobile radio networks a customization should be carefully built based on the traffic volumes (connection traffic).

Traffic performance development (capacity expanding) is not possible based on an increase of the number of radio frequency channels since this number is immutably defined in the 3GPP/GSM standards. A basic approach to enhancing traffic performance is to re-use the same radio frequency channels several times at appropriately large distances from each other. This has to reflect the design of the base stations antennas to limit the coverage area to within the boundaries of a cell. The frequency reuse has to be selected in a way that the distance between the cells with the same frequency utilization must be large enough to reduce co-channel faults to a level which remains acceptable to the system. [1]

The traffic performance therefore depends on an arrangement of the cell structures which is optimized for the traffic volume. Fig. 2.1 shows an example for cellular system with frequency reuse factor 1/7 corresponding to 7 radio cells per cluster using the principle of cell reuse. A hexagonal geometry figures are used to represent the cell radio coverage known as footprint, as it closely approximate the coverage of omnidirectional antenna. The number of radio cells in a cluster caters for the local traffic volume which is to be expected over a defined area. In an area of small cells a larger number of subscribers per unit of area can be serviced since more frequent cell reuse is possible.

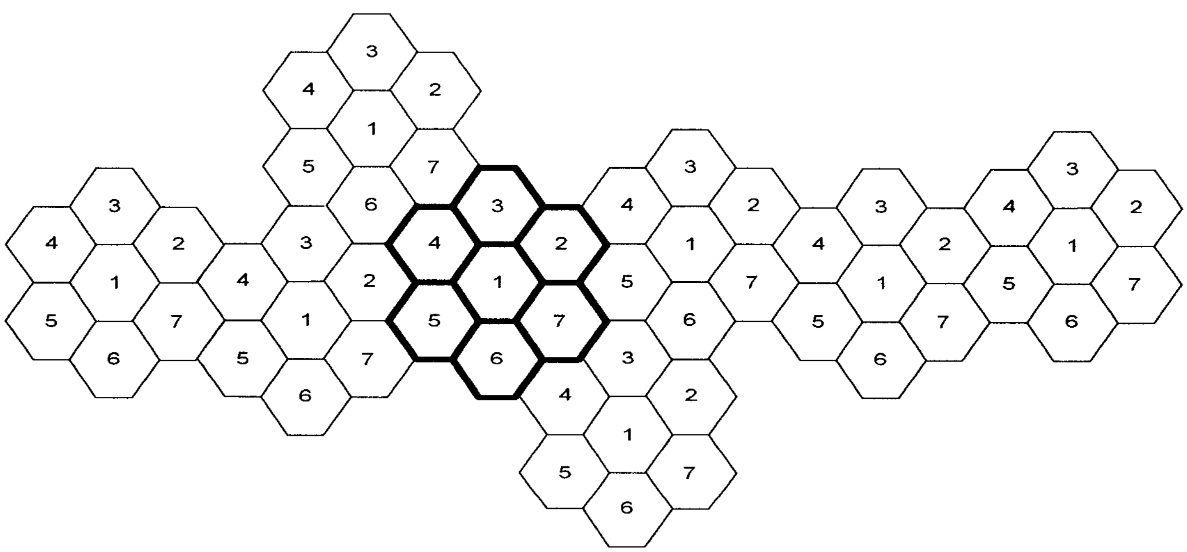

Figure 2. 1: Cellular frequency reuse concept. Cluster size equal to seven.

Other techniques to improve the coverage and the capacity of a cellular system are cell splitting, sectoring and coverage zone approaches.

Cell splitting is based on increasing the numbers of base stations which leads to an increase of the number of times that channels are reused and this results in an increase in system capacity.

Cell sectoring can be implemented using directional antennas. The main goal of the sectoring techniques is to decrease the signal-to-interference ratio (SIR). Based on the decreased SIR we can then reduce the cells in a cluster and increase the reuse of frequencies.

The switching-specific and mobile-specific processes must also guarantee high availability of traffic or control channels and prevent these channels being seized unsuccessfully.

2.1.3 Cost effectiveness

The amount of radio equipment to be installed and maintained within a PLMN (Public Land Mobile Network) service area must be kept to a minimum.

Cost effective planning must take into account any possible expansion and future developments.

Sub-multiplexing a number of channels into one physical transmission link contributes to cost-effective use of transmission paths.

2.2 GSM network structure

The structure of the cellular mobile radio network [2] for GSM900/GSM1800 mobile subscribers is shown on Fig. 2.2

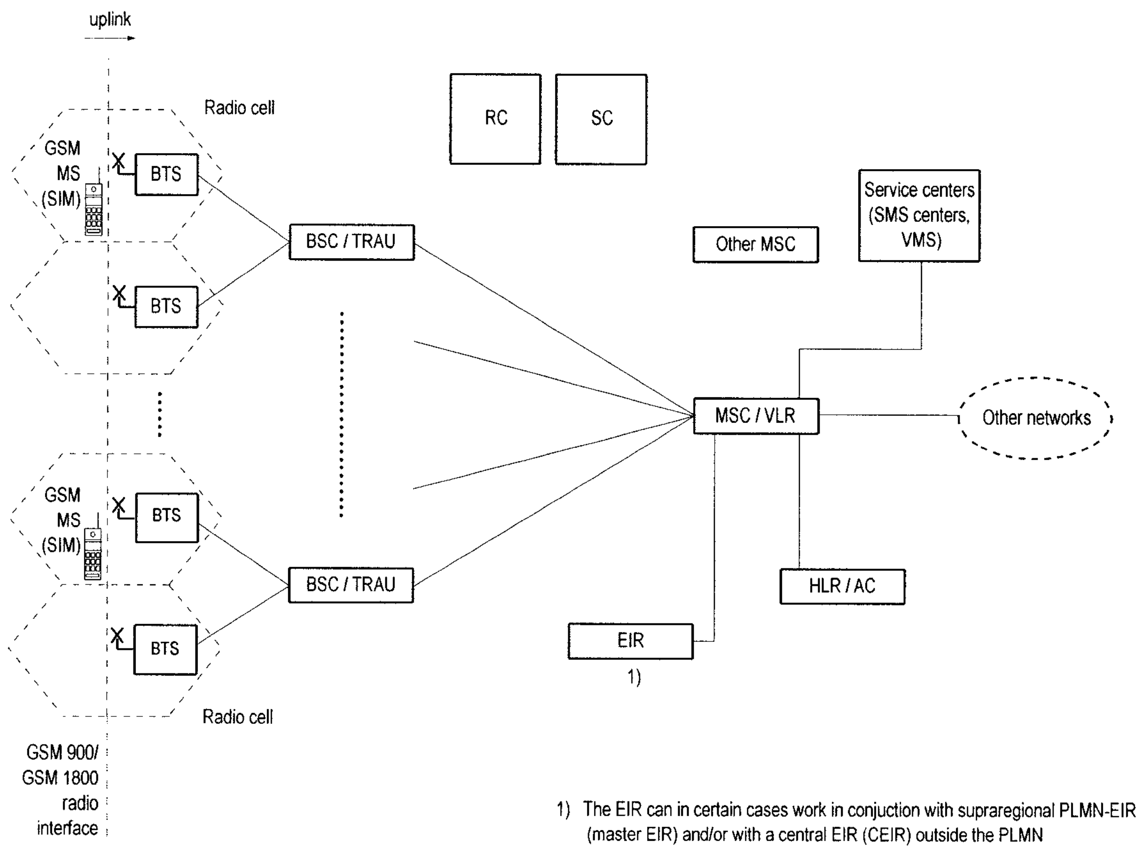

Figure 2. 2: Cellular mobile radio network GSM900/GSM1800 standard.

MSC Mobile-Service Switching Center

VLR Visitor Location Register

HLR Home Location Register

AC Authentication Center

TRAU Transcoding and Rate Adaptation Unit

EIR Equipment Identity Register

BSC Base Station Controller

BTS Base Transceiver Station

The overall PLMN area is divided up into a large number of radio cells. Broadcast antennas or directional antennas are used to cover the specified region or cell. The connection to the subscribers mobile stations is established via the radio interface. The radio interface consists of channels for transmission from the MS (Mobile Station) to BTS (Base Transceiver Station) (uplink) and channels for transmission from the BTS to MS (downlink).

2.3 GSM radio network

PLMNs use the FDMA (Frequency Division Multiple Access) access technology for sharing the available radio spectrum. In this approach each traffic channel or control channel is assigned a Radio Frequency Channel (RFCH) a subdivision of which the Radio Frequency (RF) bandwidth is divided. Each pair of radio channels requires a transmitter and a receiver. The radio frequency channels are separated by analog filters.

3GPP/GSM standards specify also the radio interface between the BTS antenna and the MS which is known as U_m interface.

The basic available standards are:

- The GSM900 primary band

890 - 915 MHz for uplink, 935 - 960 MHz for downlink; duplex spacing 45 MHz

- The GSM900 extended band

880 - 915 MHz for uplink, 925 - 960 MHz for downlink; duplex spacing 45 MHz

- The GSM1800 frequency band

1710-1785MHz for uplink, 1805-1880MHz for downlink; duplex spacing 95MHz

- The GSM1900 frequency band

1850-1910MHz for uplink, 1930-1990MHz for downlink; duplex spacing 80MHz

GSM operates with a combination of FDMA and TDMA (Time Division Multiple Access). TDMA provides the ability for users to share the same frequency channel by dividing the signal into different timeslots. For GSM networks eight time division traffic channels or control channels are transmitted over one radio frequency channel. The definition of a physical traffic channel is done by the frequency range of an RFCH pair and a time slot in the TDMA frame.

2.3.1 Radio frequency channels and bands for D1800

Most of the systems are constructed based on the GSM1800 standard, therefore we will discuss this particular standard in more detail. GSM1800 supplies 374 FDMA discrete duplex radio frequency channels: 374 uplink channels for transmission from the MS to the BSS (Base Station System) and 374 downlink channels for transmission from the BSS to the MS. The number of channels is increased by a factor of 8 to 2992 physical duplex traffic channels in the case of full-rate channels by implementing of TDMA. The specifications of the radio frequency bands are represented on Fig. 2.3

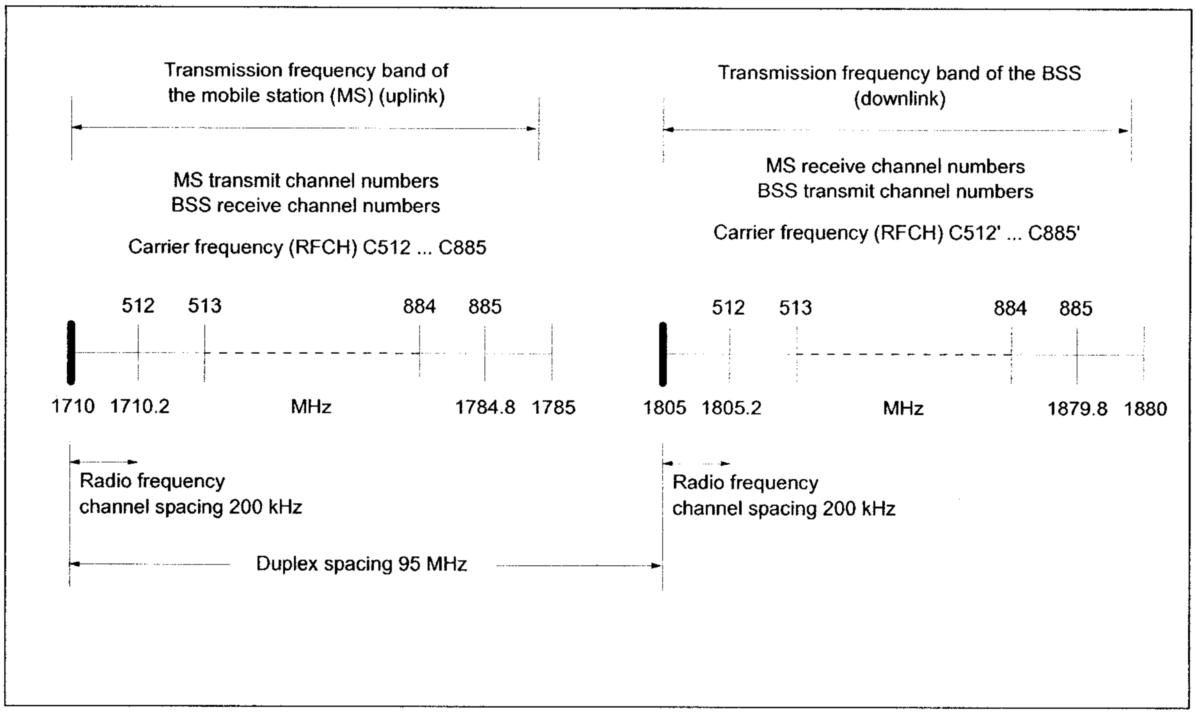

Figure 2. 3: GSM1800 distribution of the radio frequency bands

The channel spacing is as follow:

- Radio frequency channel spacing: 200 kHz
- Duplex spacing: 95 MHz

The exact channel frequency for each of the Absolute Radio Frequency Channel Number (ARFCN) can be found using the formulas below:

Carrier frequencies of the BSS receivers (uplink):

$$f_{up\,link}(n) = (1710.2 + 0.2(n - 512)) \quad [MHz]$$

(Where ARFCN n is: $512 \leq n \leq 885$)

Carrier frequencies of the BSS transmitters (downlink):

$$f_{down\,link}(n) = f_{up\,link} + 95 \quad [MHz]$$

2.3.2 Time Division Multiplex Access (TDMA) frame

The GSM900/GSM1800 standard based system utilize a combination of frequency division multiple access (FDMA) and time division multiple access (TDMA) techniques with eight traffic or control channels displaced in time and transmitted via one radio channel. Only one transmitter and one receiver are required for eight traffic or control channel pairs. This leads to reducing of space and energy in the base transceiver stations.

A TDMA frame is shown on Fig. 2.4

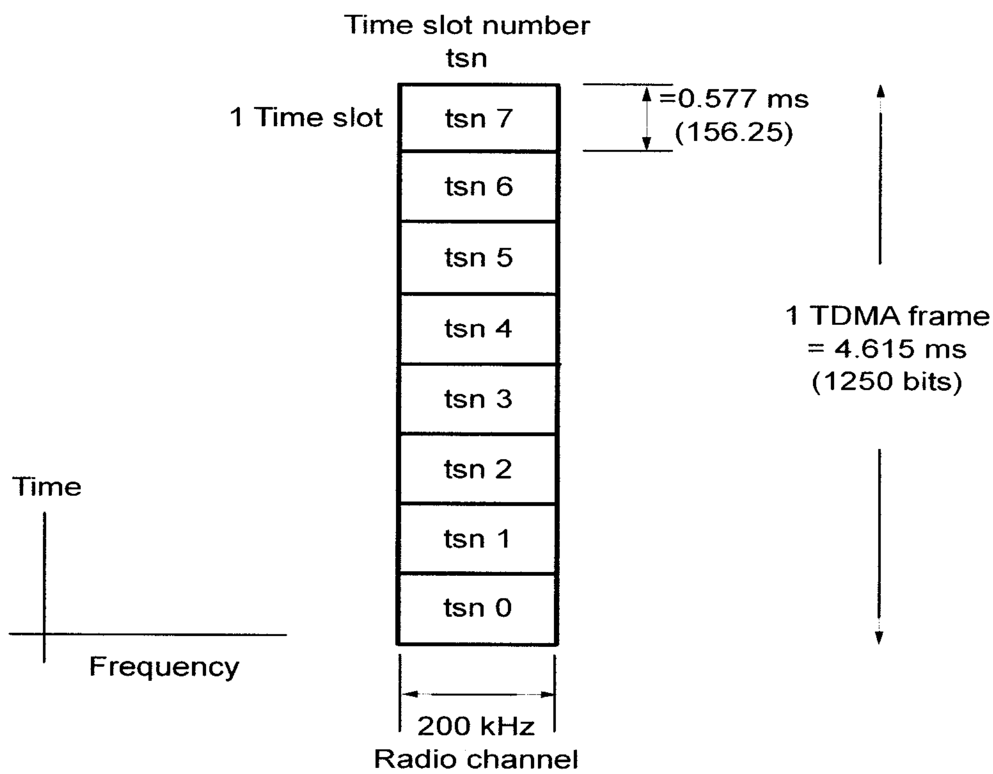

Figure 2. 4: GSM1800 Time Division Multiple Access frame structure

Each frequency radio channel is time multiplexed. The TDMA technique makes the 200-kHz dedicated transmission bandwidth available to eight full-rate channels or sixteen half-rate channels, at time intervals (time slots) repeated in a fixed pattern.

Each traffic or control channel is carried by a particular time slot number. These signals are split into portions, compressed to one eighth of their duration and then entered into a predefined time slot. After the information transmission is completed the compressed time segments are regenerated by expanding them to their original duration, and are put together to form the original signal.

In the case of voice transmission over a GSM network a regular pulse excited long term prediction (REP_LTP) codec is used. The RPE_LTP codec combines the advantages of earlier proposed (RELP) Residual Excited Linear Predictive and (MPE_LTP) multi-pulse excited long term prediction codec. By modifying the RELP codec to integrate features of MPE_LTP which traditionally provides excellent speech quality the bit rate was reduced from 14.77kbit/s to 13.0kbit/s. One of the most important modifications which led to these results was the addition of the long term prediction loop.

Utilizing the REP_LTP the electric analog voice signals produced by the microphone are initially converted for full-rate channels into a 13 kbit/s bit stream (for half rate channels into a 6.5 kbit/s bit stream). In order to enhance information stability against different noise factors, the process also provides an error control (forward error correction), allowing the transmitted information to be recreated to a certain degree at the receiver, even if the transmission path is destructed. This increases the bit rate for full-rate channels to 22.8 kbit/s (for half-rate channels to 11.4 kbit/s).

The information bits are also interleaved and separated at the transmitter and the receiver, to be able to manage the error bursts occurring on the radio path. Synchronization, control information, and transmission-free intervals between the time slots further increase the bit rate to a total of 33.9 kbit/s. The transmission rate for the overall TDMA signal is eight times as high, i.e., about 270 kbit/s. The modulation method implemented is called Gaussian Minimum Shift Keying (GMSK). A TDMA frame corresponds to 1250 bits transmitted in $120/26 \approx 4.615$ $[ms]$, a time slot corresponds to 156.25 bits transmitted in $15/26 \approx 0.577$ $[ms]$.

2.3.3 Time slot structure

Different kind of bursts may be carried by the time slots: frequency correction, synchronization, access, dummy, and normal bursts. A burst is a period of the radio frequency carrier which is modulated by a data stream. [3] The modulation is applied for the useful duration of the burst. The useful duration of the burst is the duration equivalent of 147 bits (from 0.5 to 147.5 bit time equivalent), except the access burst, which has a useful duration equivalent of 87 bits (from 0.5 to 87.5 bit time equivalent). A burst represents the physical content of a time slot. The bit with the lowest bit number (bn) is transmitted first.

To enable the possibility for the mobile station to adjust its receiver and transmitter frequencies frequency correction bursts are used. To establish an initial bit and frame synchronization synchronization bursts are used. The access burst has an extended guard period which helps to control the signal time delay due to the distance differences between the mobile stations and BTSs. After the main time delay correction the lag from the alteration of distance of a moving mobile station is controlled with the aid of the normal guard periods of 8.25 bit durations. If there is no data to be transmitted a dummy burst is applied. The structure of a normal burst is shown on Fig. 2.5

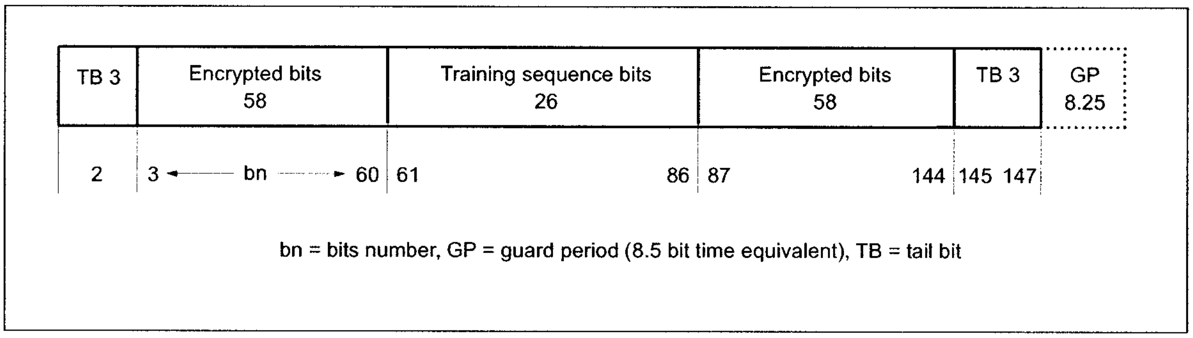

Figure 2. 5: Burst structure

The training sequence in the normal burst has 26 bits which is unchangeable synchronization pattern. This was selected in order to be enough to maintain the bit and frame synchronization once it has been established with the synchronization and the access bursts.

Different training sequences are assigned to neighboring radio cells so they to be uniquely identified by the mobile stations. GSM standard has defined eight different training sequences.

2.3.4 Frame structure

Multiframes are created by a different number of traffic channels and control channels: 51 of traffic channels or 26 of control channels.

The frame structure is shown in Fig. 2.6

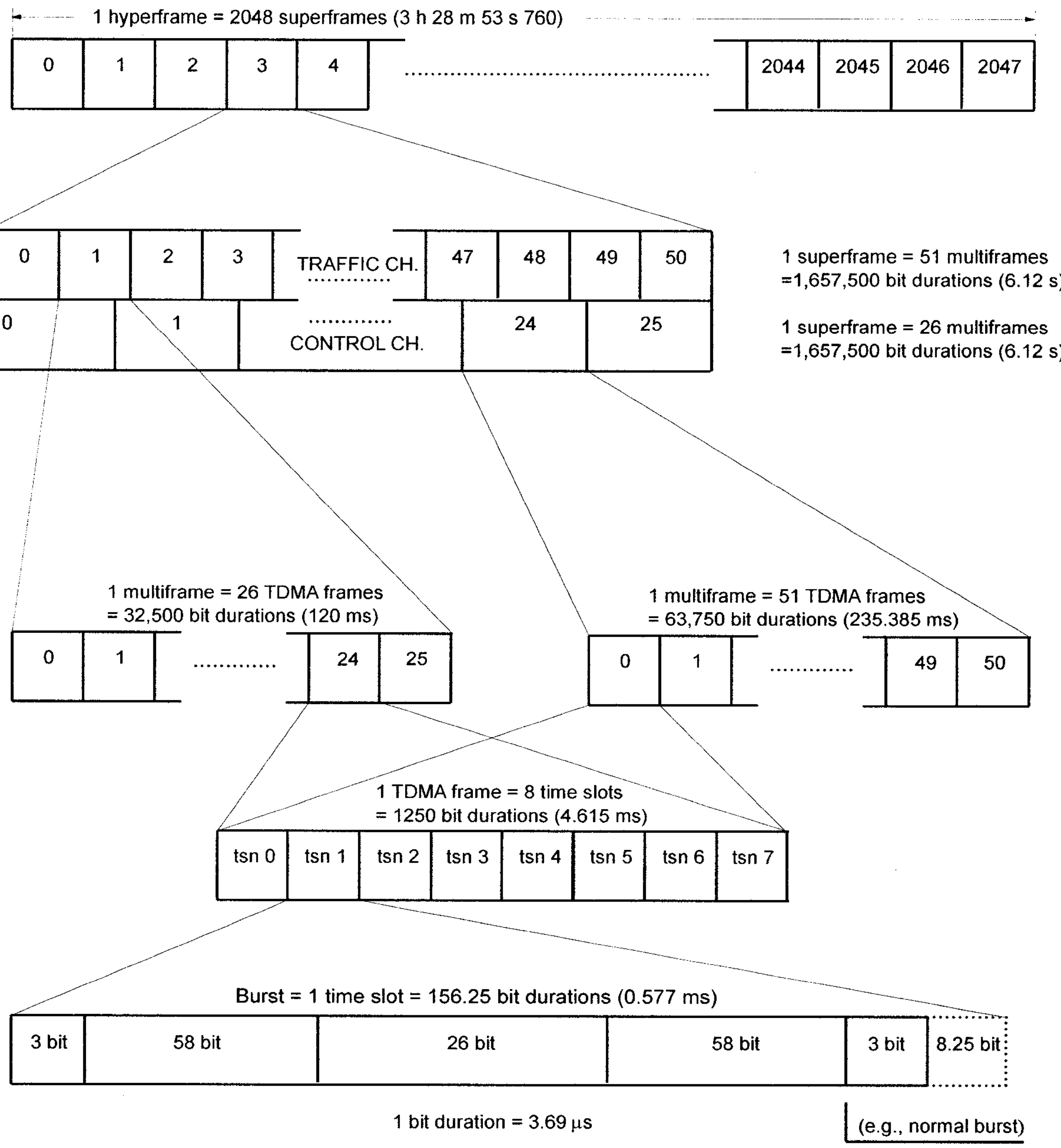

Figure 2. 6: Mapping of logical channels into Physical channels

One hyperframe includes 2048 superframes.

Traffic channels and control channels are build from different numbers of multiftames: 51 multiframes in the case of traffic channels, 26 multiframes in the case of control channels. [4]

Each multiframe contains eight time slots (one TDMA frame) and eight bursts create one TDMA frame.

2.3.5 Logical channels between BSS and MS (circuit-switched)

Many logical channels are defined for each of the physical traffic channels. They are identified by the specific parameter group in each case. Traffic Channels (TCH) carry user information (speech/voice, data) and Control Channels (CCH) deliver control signaling.

- **Traffic channels (TCH)**

Full-rate traffic channels (TCH/F) transmit information at a total rate of 22.8 kbit/s, half-rate traffic channels (TCH/H) at a total rate of 11.4 kbit/s.

The table below shows an overview of the GSM traffic channel types.

<table>
<tr><th colspan="2">Full Rate</th><th colspan="2">Half Rate</th></tr>
<tr><th>Speech</th><th>Data</th><th>Speech</th><th>Data</th></tr>
<tr><td rowspan="4">TCH/FS (Full rate speech/ Enhanced full rate speech)</td><td>TCH/F 14.4 (HSCSD)</td><td rowspan="4">TCH/HS</td><td>-</td></tr>
<tr><td>TCH/F 9.6</td><td>-</td></tr>
<tr><td>TCH/F 4.8</td><td>TCH/F 4.8</td></tr>
<tr><td>TCH/F 2.4</td><td>TCH/F 2.4</td></tr>
</table>

Table 2. 1: GSM traffic channel transmission rates

- **Control channels (CCH)**

Control channels transmit signaling information for call control, mobility management or radio channel management. Control channels can also be used for transmission of user data as Sort Message Services.

The following nine logical control channels are normally used for circuit-switched:

- Dedicated control channel (DCCH):
  - standalone dedicated control channel (SDCCH) - is used in the GSM system to provide a reliable connection for signaling and SMS (Short Message Service) messages
- Associated control channel (ACCH):
  - fast associated control channel (FACCH) – take place of the traffic channel when lengthy signaling is required between a GSM mobile and the network, typically during cell handover

- slow associated control channel (SACCH) - signalling channel that provides a relatively slow signalling connection, can be used to transfer SMS (Short Message Service)

- Broadcast channel (BCH):
  - broadcast control channel (BCCH) - is a downlink channel that contains specific parameters needed by a mobile in order that it can identify the network and gain access to it
  - frequency correction channel (FCCH) - is transmitted on the channel which generates a beacon 67.7KHz from the cell carrier frequency
  - synchronization channel (SCH) - is a downlink signal channel used for cell search and conveying of synchronization information.
- Common control channel (CCCH):
  - paging channel (PCH) - is used by the network to page the destination MS in a call termination
  - random access channel (RACH) - is used by the network to indicate radio link allocation upon prime access of an mobile subscriber
  - access grant channel (AGCH) - is used by the MSs for initial access to the network

2.4 GPRS PLMN for Mobile Subscribers

Looking ahead to the new generation of mobile services the General Packet Radio Service (GPRS) become a essential step in utilizing the existing GSM network and integrating a new high level data transmission network over it. As GSM phase 2+, GPRS offers an end-to-end packet service by the GSM network similar to fixed packet data networks such as the Internet.

The main goal of every commercial network is to maximize the number of subscribers while the quality of the provided service and the blocking rate remain unaffected. Therefore unused traffic channels should be available to cover the peak traffic at rush hours. A new approach of channel utilization was brought with GPRS. It enables a cost effective and efficient usage of radio resources (by statically allocating them to circuit-switched or packet-switched parts). GPRS enables services that use the same services and/or applications as in the fixed data network carried over packet-switched data network.

GPRS enables high-speed, user-friendly mobile data usage by using existing bandwidth more efficiently.

Typical applications for the GPRS services are:

- Wireless Internet access
- Traffic guidance
- Fleet management
- Train control system
- Point-of-sale application
- Remote control
- Database queries

2.4.1 GPRS network structure and interfaces

GPRS uses the same radio infrastructure as the circuitswitched networks (GSM). However to integrate the packet-switched functionality over the existing GSM network requires some new network elements represented on Fig. 2.7. The two new main elements of it are:

- The Gateway GPRS Support Node (GGSN) connects the GPRS network with the Internet.
- The Serving GPRS Support Node (SGSN) serves the mobile station.

The radio part of the GSM network (the BSS) has to be extended with the hardware unit Packet Control Unit (PCU) in the BSC and the software unit Channel Codec Unit (CCU) in the BTS.

Together with the new elements on the network, new interfaces were determined and standardized by the 3GPP.

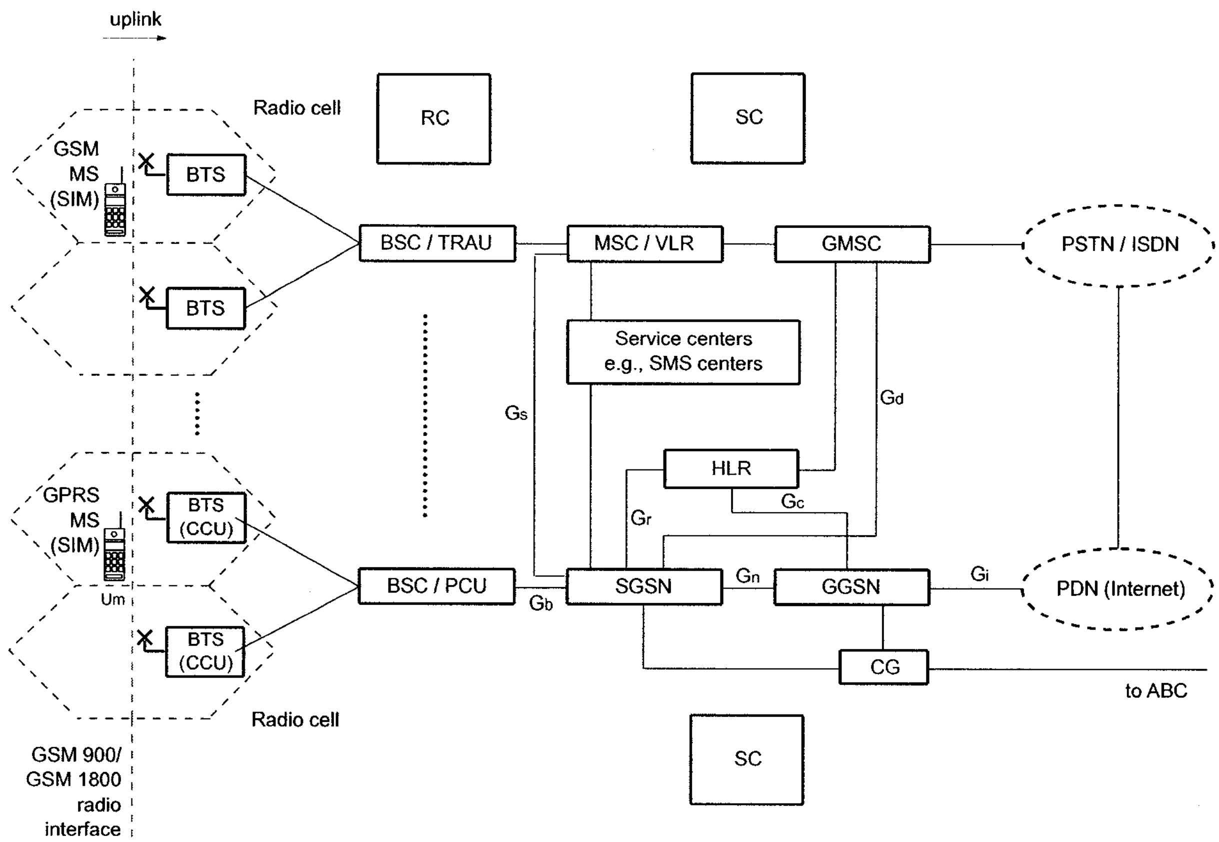

Figure 2. 7: GPRS network structure and interfaces

- Um - interface (air interface) between BSS and MS.
- Gb - interface between BSS and SGSN
- Gc - interface between GGSN and HLR
- Gd - interface between SGSN and SMS GMSC
- Gi - interface between GGSN and Packet Data Network
- Gn - interface between SGSN and GGSN or SGSN and another SGSN (represents the GPRS Backbone network)
- Gr - interface between SGSN and HLR
- Gs - interface between SGSN and MSC/VLR

2.4.2 GPRS Transmission Plane

Fig. 2.8 illustrates the layer protocol structure and the transmission of user and control information. [5] It includes the control procedures allied with the information transfer, flow control, fault detection and correction.

New introduced signaling protocols are Base Station Subsystem GPRS Protocol (BSSGP), Subnetwork Dependent Convergence Protocol (SNDCP) and GPRS Tunneling Protocol (GTP)

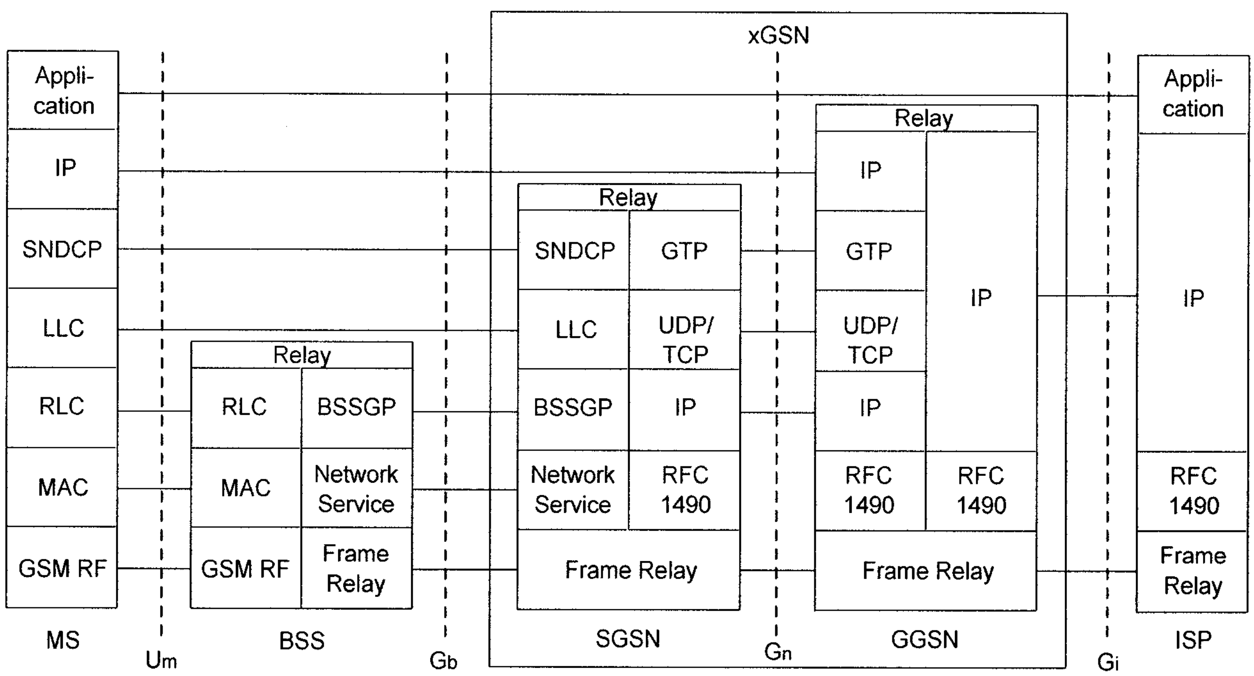

Figure 2. 8: GPRS transmission plane

BSSGP provided information related to the Quality of Service and routings between the MAC/RLC (Medium Access Control/Radio Link Control) of the PCU and the

SGSN. It is responsible of delivering of paging requests and the flow control from the SGSN to the BSS

Some of the main functions provided by SNDCP are:

- Transparent Transfer of Network Protocol Data Units (N-PDUs). It means that the introduction of new network layer protocols to be transferred over GPRS must be possible without any changes to GPRS. Therefore, all functions related to transfer of Network layer Protocol Data Units (N-PDUs) are carried out in a transparent way by the GPRS network entities.
- Data compression and decompression. With this feature the SNDCP improves the channel efficiency. Compression parameters are negotiated between the MS and the SGSN
- Multiplexing of several PDPs (Packet Data Protocol) - includes the multiplexing of N-PDUs from e.g. several network layer entities onto the appropriate LLC connection
- Buffering - N-PDUs are buffered at the SNDCP for acknowledged service.

GTP tunnels multiple protocol packets in-between GSNs. In a real environment where multiple subscribers can have multiple open sessions at the same time the IMSI and the Network Service Access Point Identifier (NSAPI) are used to uniquely identify the connections

2.4.3 Sharing of Resources in a Cell

GPRS packet switched (PS) users will share the same media and radio resources in a BTS with the GSM circuit switched (CS) users. Physical channels could be used either for GSM CS or GPRS PS traffic but not both at the same time interval. Priorities have been established to control the channel dedications depending on the traffic load in the cells. As GPRS has dynamic channel association which depends on the quality of service required for the provided service there will be more or less channels available for GPRS, CS connections.

2.4.4 Sharing of Physical Channels

One of the main characteristics of circuit switched connections is that the physical resource (the time slot) is reserved for only one subscriber. Therefore active subscribers (with established sessions) using circuit switched networks as GSM cannot share their channels with others. At GPRS networks packet switched subscribers can share physical channels. The management of the channels association to different sessions (the multiplexing of subscribers onto the same time slots) is done by software (protocols, MAC) and hardware (PCU). Sharing of network resources is a major feature of GPRS with regards to better utilization of resources on the radio interface.

2.4.5 Quality of Service and Multislot Classes

The GPRS subscribers will have different needs of radio resources depending of the mobile station multislot class and the quality of the service associated with the specific data transmission and used applications. GPRS can vary the Quality of Service (QoS) over a wide range of attributes. The QoS attributes are specified on recommendation (Rec. 02.06, 03.06) and permit the selection of the following attributes:

- Precedence Class - is related to message handling priorities, higher values of Precedence lead to higher levels of network service. The network uses this setting to prioritize service under abnormal conditions as temporally limitation of resources or overload. Three different classes have been defined: high, normal and low precedence.
- Delay class - identifies the maximum delay intervals for a packet during transmission through the entire GPRS network. GSM Rec. 02.06 defines 4 delay classes:

	128 Bytes		1024 Bytes	
Delay Class	Mean transfer delay (sec)	95% delay (sec)	mean transfer delay (sec)	95% delay (sec)
1	< 0.5	< 1.5	< 2	< 7
2	< 5	< 25	< 15	< 75
3	< 50	< 250	< 75	< 375
4(Best Effort)	unspecified	unspecified	unspecified	unspecified

Table 2. 2: Delay class specification

- Reliability class - guarantees an agreed quality of data delivery. The requirements of network protocol layers are identified in the reliability class. Five reliability classes (1 to 5) have been defined, 1 guaranteeing the highest and 5 the lowest degree of reliability. High reliability (class 1) is required for error sensitive not real time applications. Class 5 is used for real time applications which can get over data loss.
- Peak throughput class - specifies the maximum data transfer rate across the network in bits per second. However there is no guarantee that this throughput can be achieved over a certain period of time. It is the maximum rate at which the system operates regardless of frames dropped. The maximum rate can actually occur when the loss is zero. Nine throughput classes have been defined the maximum data rate doubles from one class to the next. Class 1 with 1000 byte/s (8 kbit/s) to 256 000 byte/s (2048 kbit/s)
- Mean throughput class - specifies the average data transfer rate across the GPRS network in octets per hour. Nineteen classes have been defined. Clas 1 is best effort - data rate is made available on the basis of demand and availability of resources

A large number of QoS profiles can be achieved by using the various possibilities of selected attributes.

A different QoS models is based on the concept of graceful degradation. The goal is to provide a guaranteed seamless service. The exiting bandwidth is allocated to the calls/PDP contexts in progress. In order to provide service to all of the active subscribers the number of the assigned multislots to different sessions can be reduced.

As a result we will achieve an improvement of the system's grade of service, by the cost of bandwidth lose for sessions that can tolerate bigger delays.

The network (PCU) implemented in the BSS is responsible for the channels allocation (how many time slots could be bundled by the MS uplink and downlink). This process is based on the identification of these different MSs by their multislot class. One of the simplest GPRS mobiles will be a GSM mobile that is able to handle the protocols and coding schemes of GPRS and use one time slot UL and one time slot downlink. Mobile station with multislot class 29 means that this station is able to receive and to transmit in eight time slots UL and DL simultaneously. In consequence such a MS has to have two synthesizers. The MS will send its multislot class and the PCU will only assign time slot combinations which can be handled by this equipment. [6]

An example of time slot dynamic allocation is shown on Fig.2.9

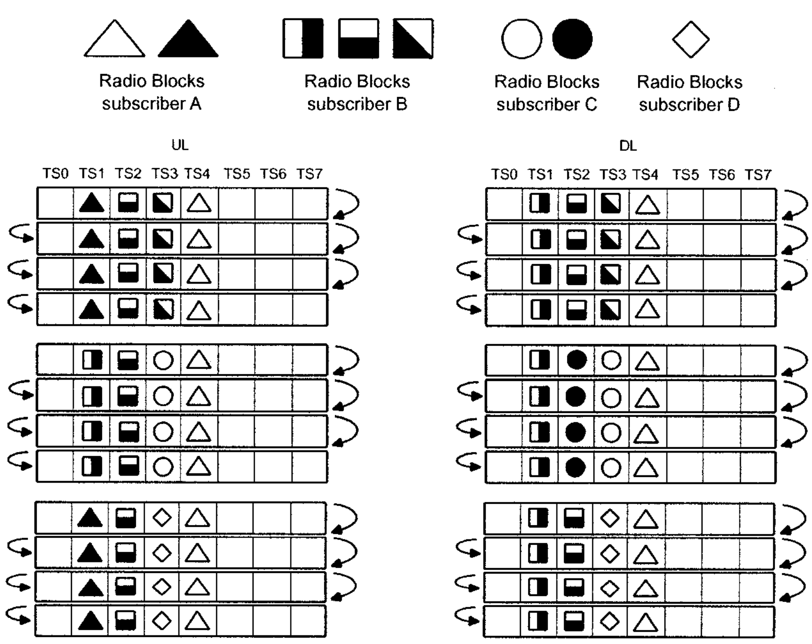

Figure 2. 9: Dynamic Time Slots distribution/sharing

2.4.6 Radio Block

Recommendation (GSM Rec. 03.64) introduces the channel coding modification for GPRS systems.

Channel coding starts with the division of digital information into transferable blocks. These radio blocks prior the encoding include:

- a header for the Medium Access Control MAC (MAC Header)
- signaling information (RLC/MAC Signaling Block) or user information (RLC Data Block) and
- a Block Check Sequence BCS.

The functional blocks (radio blocks) [7] are protected in the framework of convolutional coding against errors. Channel coding includes also a process of interleaving. In the case of GPRS, interleaving is carried out across four normal bursts NB in consecutive TDMA frames and, respectively, to 8 burst blocks with 57 bit each. Fig. 2.10 and Fig. 2.11 represent the radio block structure and the channel coding.

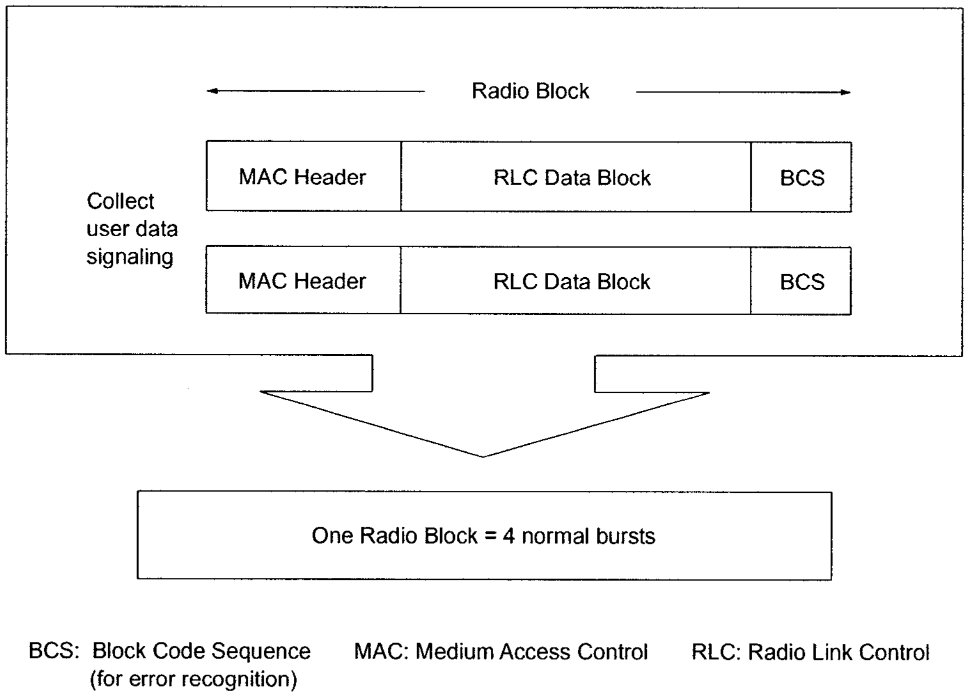

Figure 2. 10: Radio Block structure

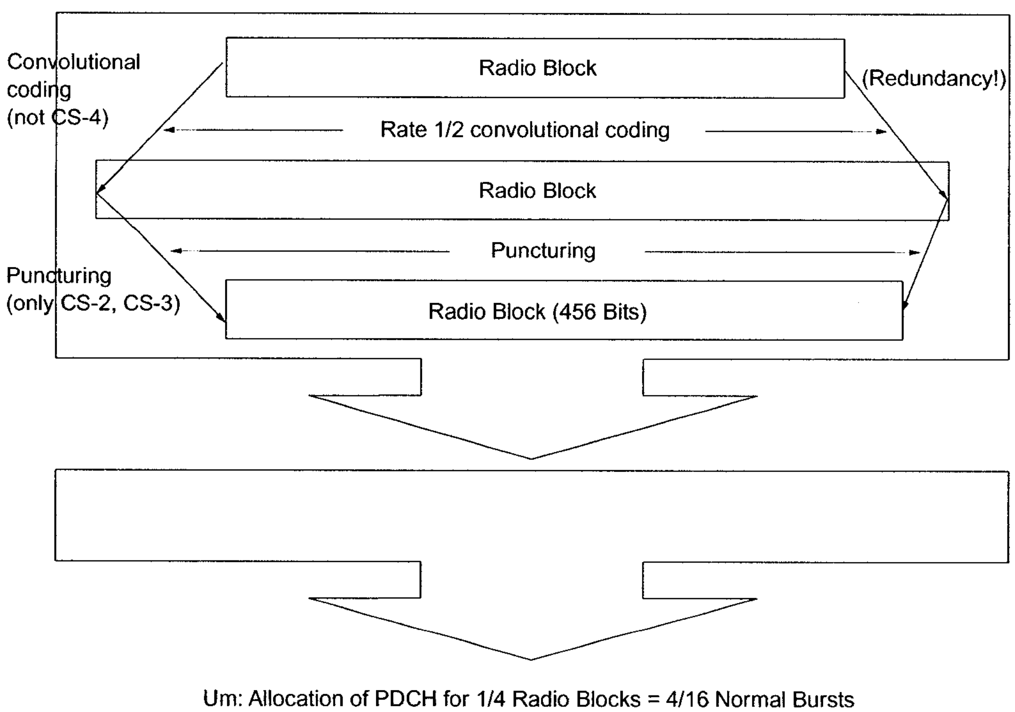

Figure 2. 11: Channel coding

2.4.7 Coding Schemes

Four GPRS coding schemes (CS) CS1 – CS4 have been defined for the transmission of packet data traffic channel PDTCH (Rec.03.64). These can be used alternatively depending on the information to be transferred and on the radio interface's quality. Usually, groups of 4 burst blocks each are coded together.

CS-1 makes use of the same coding scheme as has been specified for SDCCH in GSM Rec. 05.03. It consists of a half rate convolutional code for forward error correction FEC. CS-1 corresponds to a data rate of 9.05 kbit/s.

CS-4 has no redundancy in transmission (no FEC) and corresponds to a data rate of 21.4 kbit/s.

CS-2 and CS-3 represent punctured version of the same half rate convolutional code as CS-1.

CS-2 corresponds to a rate of 13.4 kbit/s, while CS-3 corresponds to data rate of 15.6 kbit/s.

In principle, 1 to 8 time slots TS of a TDMA frame can be combined dynamically for a user for the transmission of GPRS packet data. [8] Theoretically it is possible to achieve peak performances of up to 171.2 kbit/s (8x21.4 kbit/s) with GPRS. Current limitations are based on the mobile unit's ability of combining several times slots at the same time, which is referred to the unit multislot class. Common utilized mobile stations have multislot class 10 which ensure combining of four time slots for downlink and four time slots for uplink, but no more than five time slots combined uplink and downlink. Fig. 2.12 shows the resource management scheme.

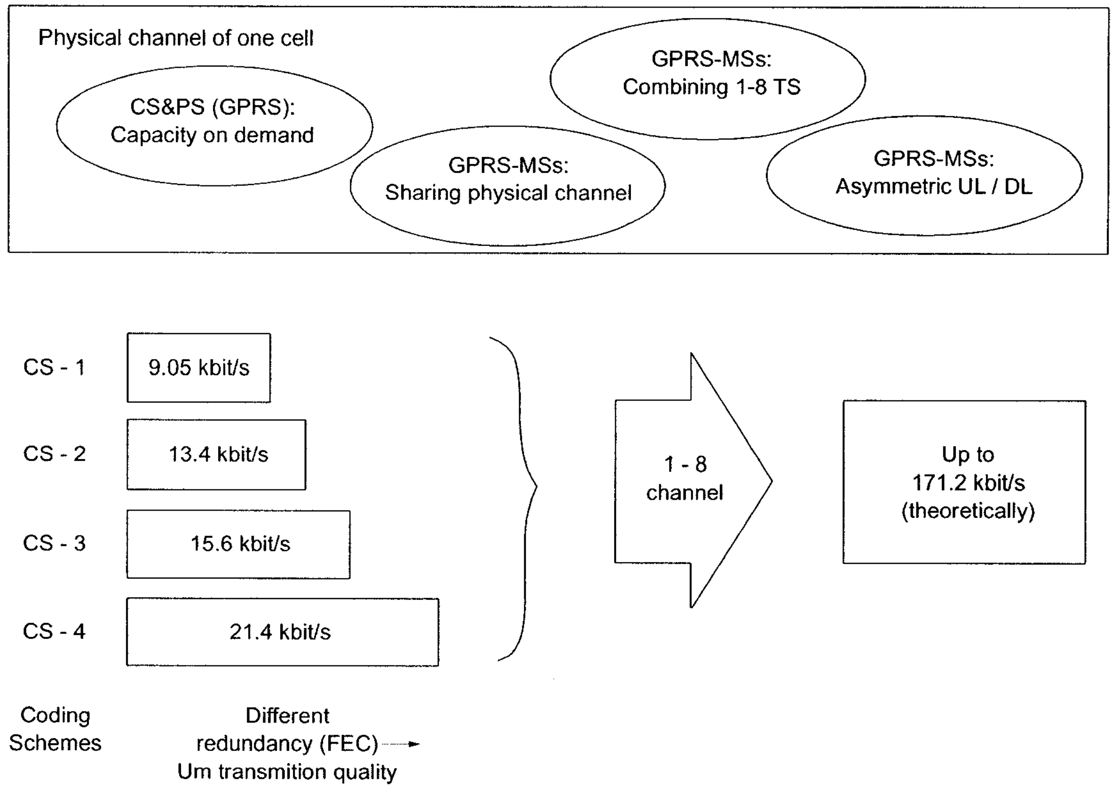

Figure 2. 12: Radio resource management coding schemes

GPRS has become a step toward the next generation of mobile technology. Today it is a widly utilized method which makes more new services available for mobile users. Together with high speed data transmission it implements the ideas of radio resource sharing, time slot combining, and asymmetric up-link and down-link channels. All of this leads to a more optimal use of the network and a more efficient usage of the existing insufficient radio resources.

Furthermore bringing the multicasting approach together with the existing GPRS makes significant enhancements in the long term goal of maximizing the network resource utilization.

Chapter 3

Multicast on cellular networks

3.1 Introduction to Multicasting

According to the present GPRS specification [9], multicast service is defined as follows:

Multicasting is a unidirectional Point-to-Multipoint service in which data packets are transmitted from a single source entity to group members currently located within a geographical area. In order for different multicast groups to be identified the message contains a group identifier.

Multicasting is a widely used solution in the world of new technology and a high demand of network resources. It provides an efficient way of transmitting data from one sender to multiple users simultaneously. Since in multicasting the data frames are sent to all the receivers instead of broadcasting the data or creating a link and transmitting the data to every receiver, there are highly esteemed advantages associated with as bandwidth efficiency and lower network and node overload leading to an improvement of the network's throughput.

Due to the advantages which can be gained from multicasting and the fast development of applications requiring more network resources such as video and audio conferencing, distance learning, e-commerce, distributed and multiplayer online games this area has become the most focused in the field of networking and many multicast related protocols have been developed in wired networks.

From the other side the dramatic change of a subscriber's mobility and the increase of the number of subscribers and popularity of mobile networks, the advances in the hardware design, the rapid growth of the communication infrastructure, increased user requirements and geographic dispersion have led to new research and implementation of multicasting in the world of wireless communications.

Fig. 3.1 and Fig. 3.2 shows the considerable resource savings in CN (Core Network) and RAN (Radio Access Network), independent of the number of users **n**

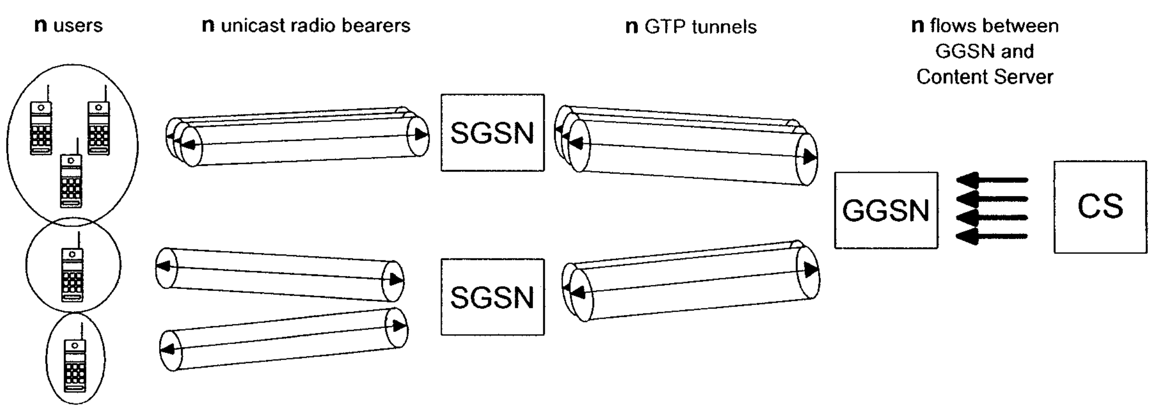

Figure 3. 1: Content delivery via existing unicast bearers

- The number of users determines required resources
- Does not scale for large numbers of simultaneous users

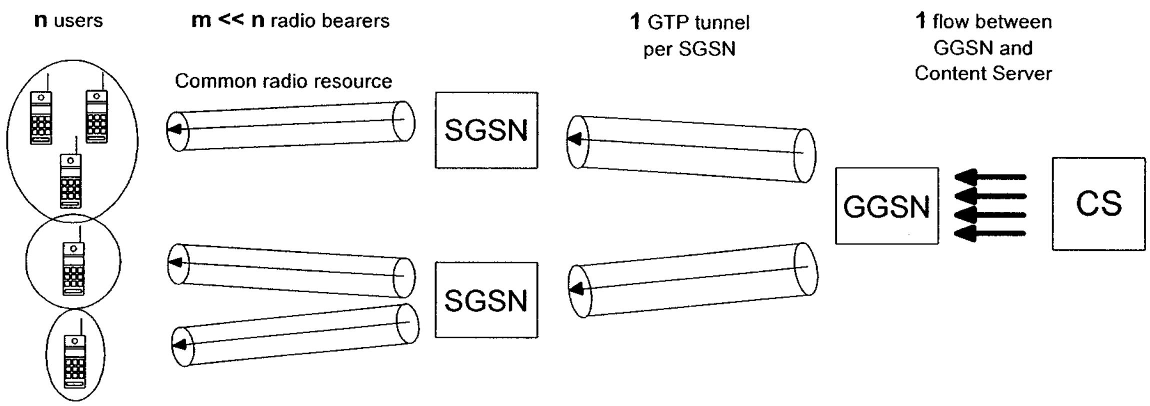

Figure 3. 2: Content delivery via multicast bearers

- Significant resource saving
- Number of content channels determines required radio resources not the number of users

3.2 Multicast development

Regardless of the network environment, multicast communication is a very efficient means of supporting group-oriented applications. Multicasting is designed for group-oriented computing. There are more and more applications where one-to-many dissemination is necessary. The multicast service is critical in applications characterized by the close collaboration of teams with requirements for audio and video conferencing and sharing of text and images.

With the network resources limitation as bandwidth in wireless data transmission, the support for multicast is even more important. It reflects itself as resource management in multipoint communications, therefore designing, development and implementation of multicast protocols and study of the impact of the realizations is highly required. Multicasts offers also much better cost effectiveness in comparison to unicast or broadcast approaches.

Furthermore, the range of applications that are well suited to multicasting has been widened. Speech transmission growth has been smaller than data transmission due to the fact that the need for mobile data transfer is becoming acute in the mobile working world of tomorrow. Following the prediction in Fig. 3.3, year 2007 data transmission will reach 50% of the total traffic volume and will thus be equal to the speech transmission in mobile networks. Note also that there is an underlying rapid increase in the total amount of traffic.

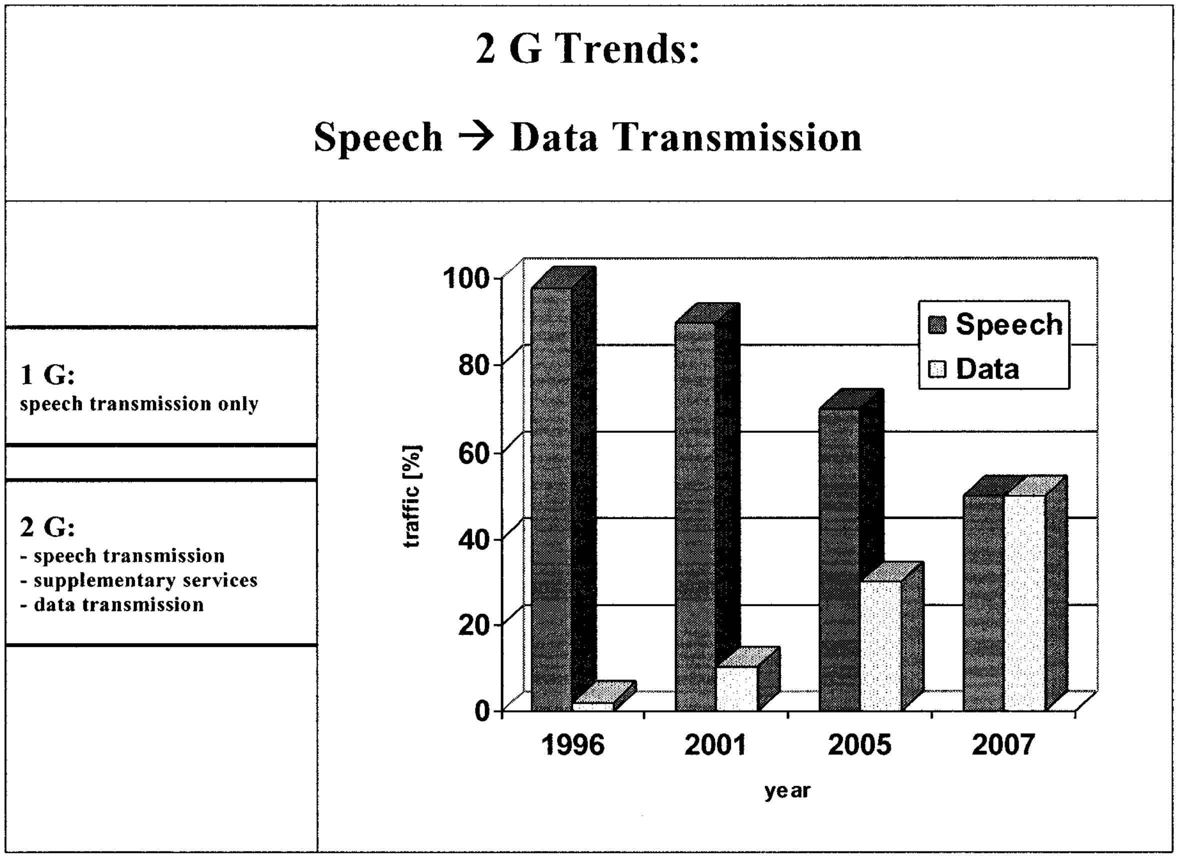

Figure 3. 3: Trend in traffic transported by mobile communication systems

Consequently, as wireless data transmission (for example Internet access) becomes more attractive, the demand for high data-rate connections will soon exceed the available capacity. It is therefore important to implement new technologies such as multicasting in order to improve the throughput of these networks.

3.3 Multicast Service Examples

There are a large number of applications that could be examined as multicast applications. Three major groups are defined based on application reliability and latency requirements.

In interactive real time applications, end-to-end latency is significantly small and varies by up to 100ms. However such applications have significant tolerance in regards to the packets loss.

On the other hand we have applications with very strong ratability requirements. For these error free information transfer (reliability) is the leading QoS factor. Such applications do not have a fixed latency period, but should maintain reasonable transmission time from the customer point of view.

The third category of applications is streaming applications. This category is defined between the real time and the reliable applications. It has less severe requirements from the real time group in regards to the latency and not as strict reliability requirements.

Examples may include the following:

- Distribution services such as news, weather and traffic reports, as well as product or service advertisements
- Multimedia services such as audio, video, and data
- On demand based High quality audio and video streaming
- Remote downloading - documents distribution or software distribution

3.4 Introduction to multicast protocols

In order to build a network supporting multicast packet transfer we have to determine how to differentiate the members of the multicast group and how to distinguish between the different multicast groups in the network. Another issue is how to build an optimum spanning tree in order to achieve good performance in an efficient manner.

Internet Group Management Protocol (IGMP) [10] was introduced in 1985 by Steve Deering. ICMP supports the membership management operations. It specifies the timer value during which all hosts have to confirm their memberships with the multicast agents. The host's membership is dynamic. It provides an opportunity that multicast groups can be joined and left at any time. There is no restriction on the location or number of members in a host group, however private access key authentication can be used for user identification.[11]

More definitions of IGMP are widely described on Request for Comments RFC966, RFC988, RFC1054 and RFC1112. Currently there are 4 versions of the IGMP, the latest one version 3 is specified on RFC 3376.

Many multicast protocols were introduced providing different approaches of realizing a loop-free optimum spanning tree.

Different goals could lead to different solutions. Some of the main criteria are measured in minimum network resources occupation and minimum packets delay.

The protocols can be classified into three different groups:

- Shortest Path Tree algorithm
- Minimum Cost Tree algorithm
- Constrained Tree algorithm

Shortest Path Tree (Spanning Tree Protocol) [12] is based on finding the minimum distance between the sender and each receiver along the tree. Well known algorithms for calculating the shortest path are Dijkstra and Bellman-Ford [13]. Varieties of the static algorithm are the dynamic Distance Vector and Link State algorithms [14].

The goal is to build a tree with the shortest distance between the sender and all of the receivers. The shortest path first does not always provide the optimum and cost efficient tree.

In order to build a spanning tree to provide the required Quality of Service parameters a new approach was introduced - Minimum Cost Tree algorithm. Metrics calculation can be based on several characteristics as bandwidth, delay, load, reliability, hop count and cost. Such algorithms focus on determinating a path with smallest metric values. Additional requirements were implemented as it should not include any nodes that are not members of the multicast group.

To improve the performance and further optimize the structure, the Steiner tree algorithm [15] neglects the restriction of every node to be a member of the group and may include also non-group members.

The Constrained Tree algorithm [16] combines the shortest path and the minimum cost goals and seeks the creation of a spanning tree based on minimizing the overall cost and the end-to-end delay simultaneously.

Some example implementations of the described protocol classes are Distance Vector Multicast Routing Protocol (DVMRP) [17], Multicast Extension to Open Shortest Path First (MOSPF) [18], Protocol Independent Multicast (PIM) [19], Core-Based

Tree (CBT) [20], Ordered Core Based Tree (OCBT) [21], Hierarchical DVMRP (H-DVMRP) [22], Hierarchical PIM (H-PIM) [23], Border Gateway Multicast Protocol (BGMP) [24].

DVMRP is a distance vector routing protocol developed for multicast routing based on the Routing Information Protocol (RIP) [25]. It was further improved with the graph mechanism to get low latency tree modification when new group members are attached.

MOSPF is a multicast routing protocol enhancement from the OSPF link-state algorithm. It uses cost as metric to build the spanning tree based on the Dijkstra's algorithm. Each router develops a link-state database which provides a complete picture of the topology of the Autonomous System.

PIM is a "protocol independent" multicast routing protocol. It has two variations PIM Dens Mode (PIM-DM) [26] and PIM Sparse Mode (PIM-SM) [27]. These modes were designed for networks with different density populations. PIM-DM is for networks with a dense group member population. It floods the data the same as DVMPR. For networks with sparse multicast members, distribution data flooding is not reasonable as it causes network overload and a waste of bandwidth. Therefore PIM-SM introduces the Rendezvous Point (RP), which acknowledges the join request sent by new multicast users and determines the correct group based on the multicast group address.

CBT is a network layer multicast protocol. It maintains a single delivery tree regardless of the number of senders. This approach reduces the state information that have to be managed at every transfer point as there is only one spanning tree per multicast group. CBT version 2 was developed in order to improve the loop free states.

OCBT enhances CBT and provides solutions to the issues with loops in the event of unicast routing instability and multicast tree creation under certain conditions.

Hierarchical protocols were created to overcome the drawbacks of existing protocols in the implementation on Wide Area Networks (WANs).

H-DVMRP divides the network on non-overlapping areas and levels which makes possible the utilization of any routing protocols on the intra-region multicast. It reduces the information exchange on second level only to the regions. Furthermore this also reduces the number of entries on the routing tables.

HPIM improves the existing PIM protocol by arranging the Rendezvous Pints (RPs) in an hierarchical structure, which overcomes the sub-optimal placement of RPs.

3.5 Multicast protocols utilization in GPRS and other mobile networks

The selection and implementation of the discussed protocols in a new multicast GPRS network should be based on the following criteria:

- Minimized network resource consumption
- Multicast group management
- Data privacy and integrity
- Charging mechanisms
- Sender and receiver mobility handling

Many studies have shown that IGMP protocol in collaboration with PIM-SM protocol can provide an acceptable performance and management functionality to build a multicast GPRS/UMTS network. [28]

Furthermore, new protocols were proposed to overcome the issues with efficient spanning tree updates for systems with rapid topology changes as in Mobile Ad hoc Network (MANET) [29]. The Multicast MANet Routing Protocol (MMARP) [30] provides effective routing mechanism, it is scalable and can manage the complexity of supporting traditional IP nodes. It could be utilized to build GPRS multicast networks together with the IGMP. [31] [32]

3.6 Comparison between Multicast, Broadcast and Unicast transmission sessions

Table 3.1 provides some general comparisons between broadcast, multicast and unicast service parameters. Multicast uses authentication and ciphering as the unicast sessions added, there is no acknowledgement and signaling for user authentication as in the broadcast case.

Some of the multicast specifics are defined as the following:

- User subscription to the multicast subscription groups is required. This can be made by the operator, the user or a third party.
- One user can join several multicast groups at the same time

	Broadcast	**Multicast**	**Unicast**
Recipient	All service subscribers	Identified subset of all service subscribers	Unique subscriber
Authentication	No	Yes	Yes
Ciphering	No	Yes	Yes
Coverage area	All	Limited	All
Acknowledgement	No	No	Yes
Signaling for User Identification (recipient)	No	No	Yes
Signaling for Group Identification (recipient)	Not available	Yes	Not available
Group Identity	No	Yes	No
User Identity	Yes	Yes	Yes

Table 3. 1: General Comparisons, Broadcast, Multicast and Unicast

3.7 Multicast subscribers identification and addressing

Every multicast group is uniquely identified by the International Mobile Group Identity (IMGI). It supports two levels of identifications: a service provider level and an application level. Other important addressing information is related to the location of the multicast MSs and it is determined as geographical addressing. The geographical addressing specifies the area over which the multicast data will be transmitted.

3.8 Multicasting operation model

The basic operational mode for a multicast system is shown on Fig. 3.4

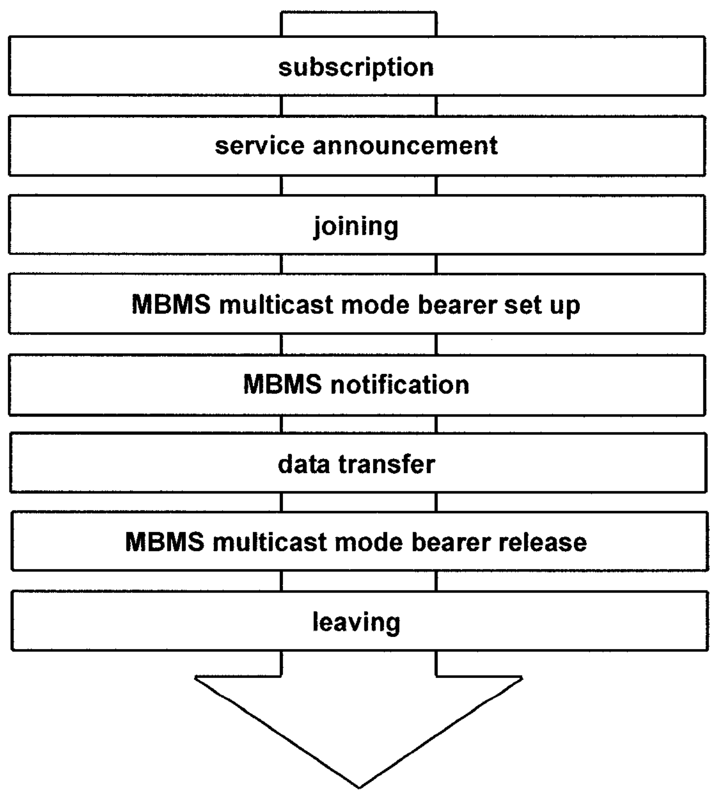

Figure 3. 4: MBMS Multicast mode

- Subscription - Determine the agreement between user and operator
- Service announcement - User is informed about the range of Mobile Broadcast/Multicast Service (MBMS) services available.

  Client receives notification about the available multicast service and how to access the service. The MBMS service announcements distribution to mobile subscribers could be done in different ways.

  HTTP (hypertext transfer protocol) or WAP (wireless access protocol) can be used to get the stored announcements from a web server.

  Another method is to use existing services as SMS or MMS or a special MBMS service announcement channel to send the announcements to end users.
- Joining – after the service announcement is received and the customer has selected to use the service and become a member of the corresponding multicast group a session join request must be sent to the network. The join request should contain parameters extracted from the service announcement in order for further initialization to be performed on the switch side.
- MBMS multicast mode bearer set up - Start of a multicast session (network resources established). Before starting the transmission GGSN according to the request from the BM-SC (Broadcast/Multicast Service Center) allocates the required internal resources and forwards the request to the involved SGSNs. SGSNs transfer the information to the PCUs responsible for the allocation of radio resources necessary for providing the required quality of service (QoS).

- MBMS notification - All MSs of the corresponding MBMS service group are notified that the transmission of user data can start. "Multicast data transfer is ready".
- Data transfer – receive/transmit data
- MBMS multicast mode bearer release - This is a service cancellation from the server side. The server sends a stop notification to indicate that the data transmission has ended. Network resources are released.
- Leaving – An end user who wants to leave the multicast service sends a service leave request to the network. The responsible node removes the user from the related MBMS service group. It is no longer possible for this user to receive information sent to the multicast group.

3.9 Mobile Broadcast/Multicast service - MBMS Architecture

On the basic MBMS architecture a new node Broadcast Multicast - Service Center (BM-SC) has been introduced in addition to the existing GPRS platform Fig. 3.5.

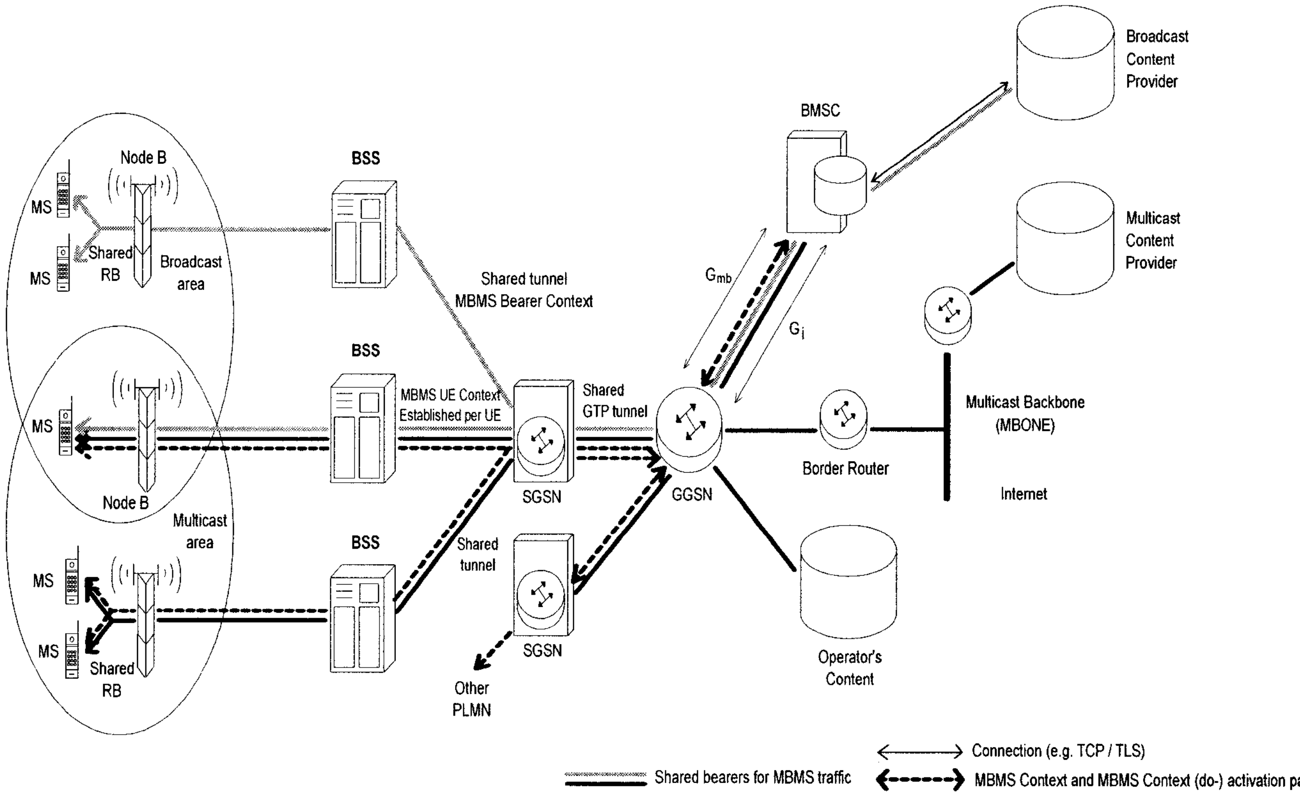

Figure 3. 5: Multicast mode Mobile Broadcast/Multicast service architecture

BM-SC - Broadcast Multicast - Service Center

SGSN – Serving GPRS Support Node	GGSN – Gateway GPRS Support Node
GTP – GPRS Tunneling Protocol	BSS – Base Station System (PCU&CCU)
RB – Radio Bearer	MS – Mobile Subscriber

3.9.1 Broadcast Multicast - Service Center (BM-SC)

The BM-SC is responsible for the authorization of the user specific MBMS service activation at the GGSN. It has two independent interfaces to the GGSN, the Gmb interface and the Gi interface. Gmb is used for transmission of multicast control information. Gi is used for transmission of the data. In scheduled data transmission, when the data transfer is requested by the content provider the BM-SC has to permit the session and could apply charges to the content provider. The data can originate from within the mobile network or from other sources such as the Internet.

3.9.2 Serving GPRS Support node (SGSN)

Some of the main functions of the Serving GPRS Support node are access control, mobility management and packet data protocol activation. The SGSN hold information for the location of the GPRS-MSs, subscriber states, the supported packet data protocol(s) and the corresponding GGSNs.

In MBMS service environment SGSN concentrates all individual users of the same MBMS service into a single MBMS service and also performs an individual service control functions. The SGSN is aware of all multicast sessions at any time and maintains a single connection with the source of the MBMS data.

3.9.3 Gateway GPRS Support Node (GGSN)

The GGSNs maintain specific routing tables with addressees of corresponding SGSNs involved in the multicast transmissions (depending on the end users Local Area/ Routing Area information). It performs gateway functions and connects the GPRS system to external service providers (Internet Service Providers). GGSN utilizes the GPRS Tunneling Protocol (GTP) to forward packets to the related SGSN where the destination user resides. The GGSN can assign dynamic packet data protocol addresses.

3.10 Session and mobility management signaling processing

This section covers the basic signaling processing for GPRS multicasting.

GPRS Mobility Management and Session Management (GMM/SM) are part of the GPRS signaling plane. The mobility management functions are used to keep updated information related to the current location of an MS within the mobile network. The MM functionality is integrated in MSs and SGSNs. The session management is used for activation, modification and deactivation PDP (packet data protocol) contexts, MBMS multicast service activation, set-up and release. The SM functionality is integrated in MSs, SGSNs and GGSNs.

3.10.1 MBMS Mobility Management

Fig. 3.6 shows the most complex Inter SGSN Routing Area Update Procedure. The Routing Area Update is detailed discussed as it plays critical role in managing the multicast sessions for mobile subscribers and defining the spanning tree topology.

- The MS sends a "Routing Area Update Request" containing (old Routing Area Identity (RAI), old Packet - Temporary Mobile Subscriber Identity (P-TMSI), Signature, Update Type) to the new SGSN. The Base Station System (BSS) adds a call identifier before passing the message to the new SGSN. The new

SGSN can extract the new RAI and Local Area Identity (LAI) from the Cell Global Identity.

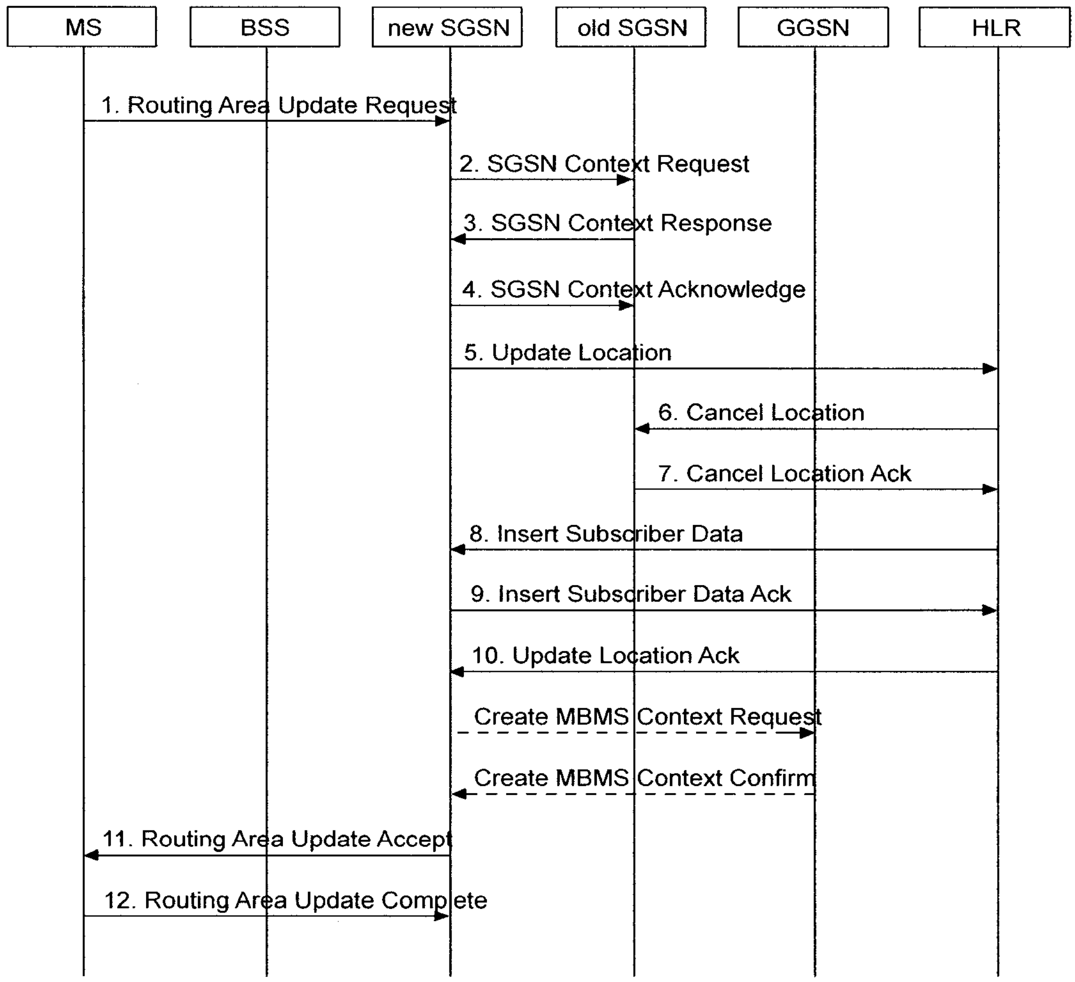

Figure 3. 6: MBMS Mobility Management

- The new SGSN sends a "SGSN Context Request" including the following information (old RAI, temporary logical link identity (TLLI), old P-TMSI Signature, new SGSN Address) to the old SGSN in order to get the MM and

Packet Data Protocol (PDP) contexts for the MS. The old SGSN validates the old P-TMSI Signature. If it does not match the value stored in the old SGSN a security functions is initiated to confirm the identity.

- If the MS authentication is successful, the old SGSN responds with "SGSN Context Response" (MM Context, PDP Contexts, and Logical Link Control (LLC) Acknowledgment). The old SGSN also stores the new SGSN address to allow data packet forwarding to the new SGSN. LLC ACK contains the acknowledgements for each LLC connection used by the MS. Each PDP Context includes the GTP sequence number for the next downlink PDU (Packet Dada Unit) to be sent to the MS and the GTP sequence number for the next uplink PDU to be transmitted to the GGSN.
- In the case when the MS has at least one activated PDP context, the new SGSN sends an "SGSN Context Acknowledge" message to the old SGSN. This informs the old SGSN that the new SGSN is ready to receive data packets belonging to the activated PDP contexts.
- If the MS authentication is not successful, then the routing area update is rejected. The old SGSN continues buffering and transmitting packets according to the old MS registry.
- The new SGSN sends "Update Location" signaling packet to the HLR informing about the new MS location by providing the following information - SGSN Number, SGSN Address and International Mobile Subscriber Identity (IMSI).

- In response the HLR sends Cancel Location (IMSI, Cancellation Type) to the old SGSN. After finishing the forwarding of PDUs, the old SGSN removes the MM and PDP contexts. It guarantees that the MM and PDP contexts are kept in the old SGSN for a certain period of time in case the MS initiates another routing area update before completing the ongoing routing area update to the new SGSN.
- The old SGSN confirms the location cancellation to the HLR with "Cancel Location Ack." including the MS IMSI.
- The HLR sends "Insert Subscriber Data" containing (IMSI, GPRS subscription data) to the new SGSN. The new SGSN validates the MS's presence in the routing area.
- If due to regional subscription the MS location update is rejected, the SGSN rejects the Attach Request with an appropriate cause and returns an "Insert Subscriber Data Ack." (IMSI, SGSN Area Restricted due to regional subscription) message to the HLR. If the routing confirmation is successful then the SGSN creates an MM context for the MS and returns an "Insert Subscriber Data Ack" (IMSI) message to the HLR.
- The HLR confirms the updated Location sending "Update Location Ack" (IMSI) to the new SGSN.
- At this point MM and PDP contexts were created at the new SGSN and logical link is established to the MS.
- New multicast sessions can be established following the activation flow procedure.

- The new SGSN responds to the MS with "Routing Area Update Accept" including (P-TMSI, LLC Ack, P-TMSI Signature).
- The MS confirms the new P-TMSI with a "Routing Area Update Complete" providing (PTMSI, LLC Ack). LLC Ack contains the acknowledgements for each LLC connection used by the MS. Based on the provided information the successfully transferred PDUs are confirmed and they are discarded by the new SGSN.

3.10.2 MBMS Multicast Service Activation and RAB Set-up

Fig. 3.7 depicts the signaling flow between the MS and the GPRS modules in regards of the Multicast Service Activation. [33]

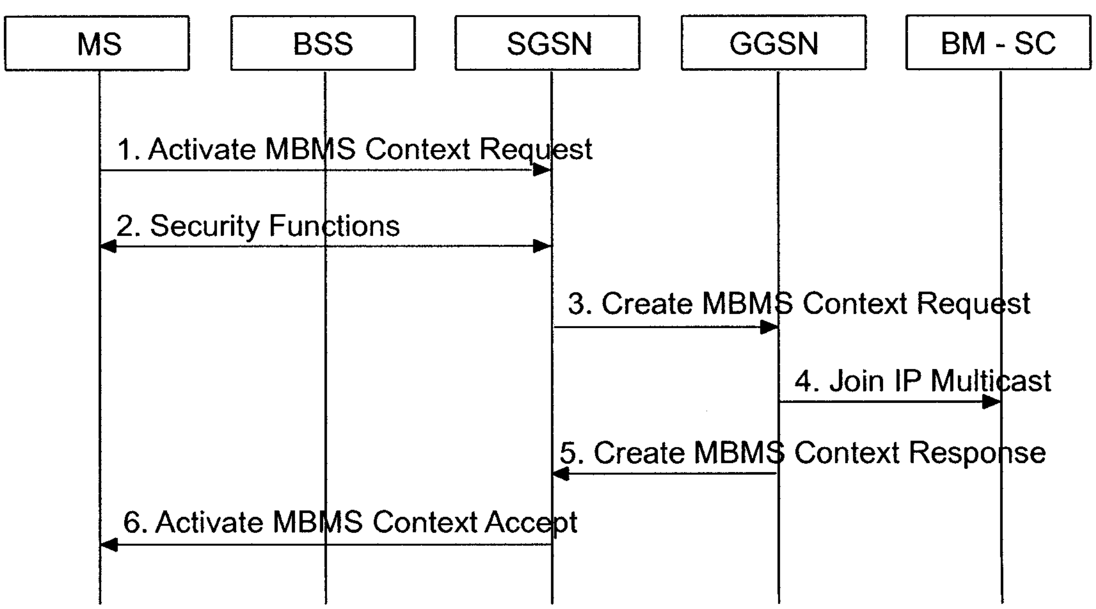

Figure 3. 7: MBMS Multicast Service Activation

An important part of the creation of the connection is the allocation of required network resources at the core of the network and on the radio side. This allocation has to be based on the required Quality of Service for the specific multicast applications.

3.11 Sticky Point-to-Multipoint channel for multicasting

A new area of research [34] has arisen in the field of multicasting, focusing on better utilization of the radio resources.

The multicast service could be extremely beneficial in terms of radio resource saving in areas with a high density of multicast subscribers using the service. The high density could be because of the popularity of the offered service or if the supported base station can cover and support multicast users from neighbors areas using the same service.

It is known that for more than 40% of the time mobile users are covered by more than one base station, especially in a city environment where base stations support small cells (geographic areas) with a large capacity. On Fig. 3.8 are represented two cells with 120 degree sectoring, base stations overlap areas and three mobile users.

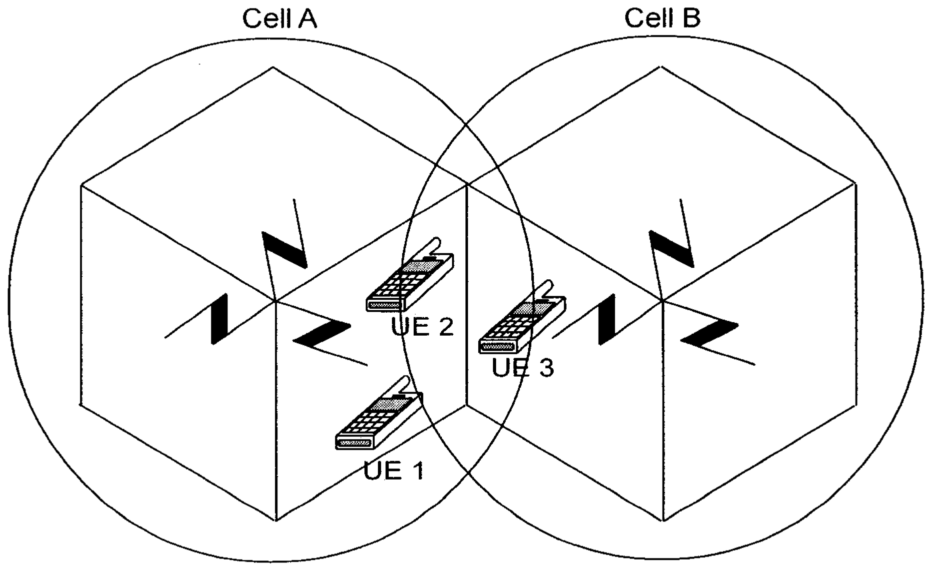

Figure 3. 8: Hexagon cells 120o sectoring with overlap area representation

If all of the users UE1, UE2 and UE3 are using the same multicast service they could have the same dedicated network resource. In standard configured system separate resources will be dedicated at cell A to support users UE1 and UE2, and resources from cell B for the support of the service provided to user US3. In this case the system will waste radio resources and face backbone network overflow.

If we implement a Sticky multicast channel in cell A, it could support all of the mobile users on the cell's overlap area and use the same multicast service.

This new technology will contribute to achieving better resource utilization and further improve the main benefit of the multicasting as bandwidth efficiency and lower network and node overload leading to an improvement of the network's throughput.

Chapter 4

Simulation of GPRS multicasting

4.1 Simulation objectives

In previous chapters we have discussed some of the new leading technologies in the modern world of telecommunications - GPRS and multicasting. Bringing them together opens new opportunities for technological improvement and new areas of study, research and development.

With our studies we would like to make an objective conclusion on the impact of multicasting over mobile networks and in specific over GPRS networks. We seek answers to the question - How the multicast implementation will facilitate structured and functional GPRS networks to improve the current resource possibilities and support new services requiring high utilization of core network and radio resources?

4.2 Simulation input parameters - initial user definition and distribution over the examine area

In this section we describe the basic parameters of the mobile subscriber distribution over the simulated system and some of the main characteristics

4.2.1 Number of mobile subscriber and dedicated transmission time slots:

The number of active subscribers (performing data transmission) has significant impact on the system performance as this is a radio resources limited system. The active subscriber number is defined as a simulation input parameter. We examine four cases for **N=200, 400, 600** and **1000** subscribers. On the other hand considering the limited bandwidth, this parameter affects the number of the possible transmitted packets and has direct impact on the results. This parameter is related to the overall load of the system.

We define three categories for packet transmission sessions in one iteration.

- from 0 to 2 packets transmitted per iteration
- from 3 to 5 packets transmitted per iteration
- from 6 to 8 packets transmitted per iteration

For more precise studies we include an option the number of possible packet transmission also to be defined as fixed values as 1, 2, 3, 4, 5, 6, 7, and 8.

The maximum number of possible packets transmitted in one iteration for any GPRS system is limited to eight. Overloading of the system leads to reducing the total number of available channels. In case of equal priority of system resource distribution of TDMA/FDMA based systems such as GPRS. Limited channels numbers per supported area (cell) will result in decreasing the probability of a specific data session to transmit the maximum of eight packets. As a consequence this will decrease the data transfer speed. For applications that have high demands of data transmission speed QoS setup will play a significant role as it can change the priority of specific sessions.

The attributes of each user are defined by the following parameters:

- **Location**
- **Velocity**
- **Type of session**
- **Buffer size**

4.2.2 Location

Even though a real cell is amorphous in nature, a regular cell shape has to be defined for system design, simulations and further development. The selection of the form for the cell presentation is relegated to the following three choices a square, an equilateral triangle, and a hexagon. The selection is based on the rule the entire area to be covered without any adjacent cells overlapping. For the multicast simulation, we have defined an area of 16 square cells 4x4 (each 2 km x 2 km), with a total area covered of 64 km^2.

The real radio coverage (the size and the shape) of a cell is determined from field measurements and the propagation prediction models in the field depend on the urban area and the surrounding objects. In most cases each cell is modeled by the hexagonal geometry structure. For simulation simplicity as we should monitor the distribution of the subscribers on different cells, we have defined the cells as squares. This will not affect the simulation results as this simulation is made to show the affect of multicasting over the system performance. This is related to the number of the subscribers per cell and the traffic they generated rather than to the exact position of the subscribers. The simulation is built in a way that will facilitate the expansion of the research area and involve further systems tests proving the system resource utilization improvement in case of covering multicast users located on the cell borders from the calls with existing multicast sessions.

Mobile stations are randomly distributed at the beginning of the simulation over the considered area, running the generator or random numbers in the range 0 to 2000 for the x-axis and the y-axis.

We assume that all users are on GSM/GPRS covered areas and have GPRS subscription on the HLR which is essential for performing packed data transmission.

4.2.3 Velocity

We have defined three types of users with different velocities. The percentage of users belonging to different categories is set as input parameter.

V_1=0km/h for **50%** of the users – stationary users

V_2=5km/h for **30%** of the users – slow moving (walking) users

V_3=50km/h for **20%** of the users – fast moving users

This parameter has influence over the distribution of the user from different sessions at certain cell and modifies the numbers of the users belonging to a specific multicast session. All this reflects over the number of saved channels and the determination of the multicast performance. The simulator is created in a way to provide output data for user distribution for time intervals of 5 minutes.

To study QoS parameters the simulation is run for 3000 iterations. The approximate time interval for the iterations to be completed is a few seconds. The probability of users to change the base station supported routing area during this time frame is negligible, this is the same also for users with the highest velocity. However the velocity will not be considered for the simulation and studies of delay and overflow parameters.

The simulation and studies of delay and overflow parameters is done without taking into consideration the mobile station mobility from one cell to another based on the negligible probability that this event occurres in the time frame. However, we have defined this parameter as this simulation could be run also for larger number of iterations and hence we could identify the effect of the QoS over much larger time frame. It is expected that in an ideal case the number of subscribers leaving the cell should be equal to the new ones entering, which will not change the results as the load for the particular cell will remain the same.

4.2.4 Types of Sessions

Each simulation is run for a different distribution of multicast users versus unicast users. All of the users have established data transmission sessions: multicast sessions or with another user (unicast sessions)

The total number of the active multicast subscribers varies for the different simulations and could be 10%; 30% and 60% out of the total number of subscribers, i.e.

$$\mathbf{a=0.1N;\ 0.3N\ or\ 0.6N}$$

Where a represents the number of multicast subscribers in terms of the totals number of subscribers N

Each multicast user can belong to one of the three existing multicast sessions. The percentage distribution is specified below:

- Multicast session one: $\mathbf{a_1=10\%}$ of total number of multicast users **a**
- Multicast session two: $\mathbf{a_2=30\%}$ of total number of multicast users **a**
- Multicast session three: $\mathbf{a_3=60\%}$ of total number of multicast users **a**

4.2.5 Simulation realization of user definition and placement

Flow chart for user creation, belonging to different sessions, defining the velocity of each user and randomly user distribution over the examine cells is represented on Fig. 4.1

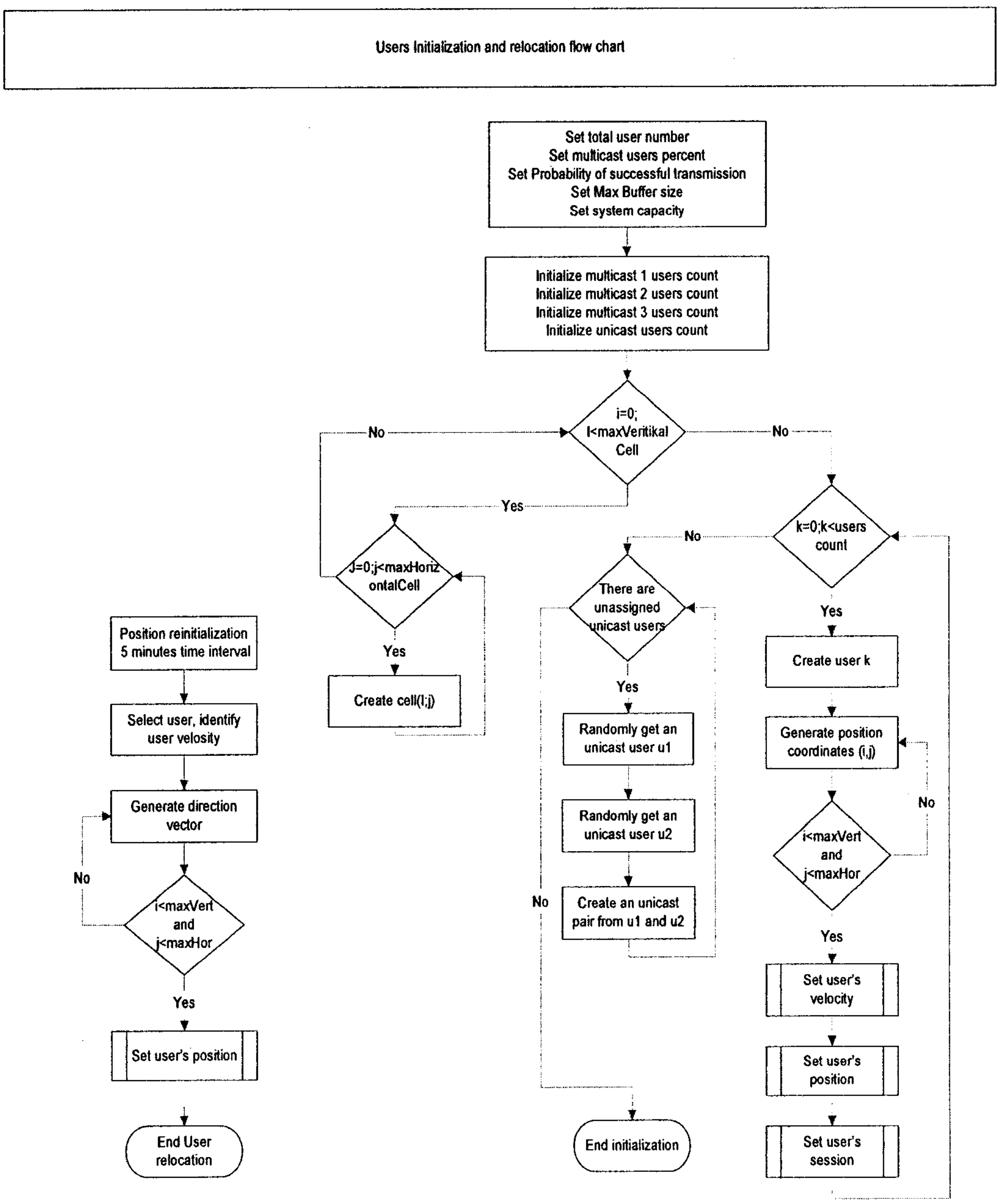

Figure 4. 1: User definition, distribution and relocation flow chart

We have studied the system user distribution for time intervals of 5 minutes. Users have been moved based on their velocity and selecting random vectors for directions. If the new location is outside the examined system (16 cells) a new direction is selected to keep the users inside the specified area.

Fig. 4.2 and 4.3 show the initial distribution of the users across the network for two cases: 200 total users with 10% multicast users and 1000 total users with 60% multicast users

- ◆ - unicast
- O – multicast session one
- × - multicast session two
- ⚐ - multicast session three

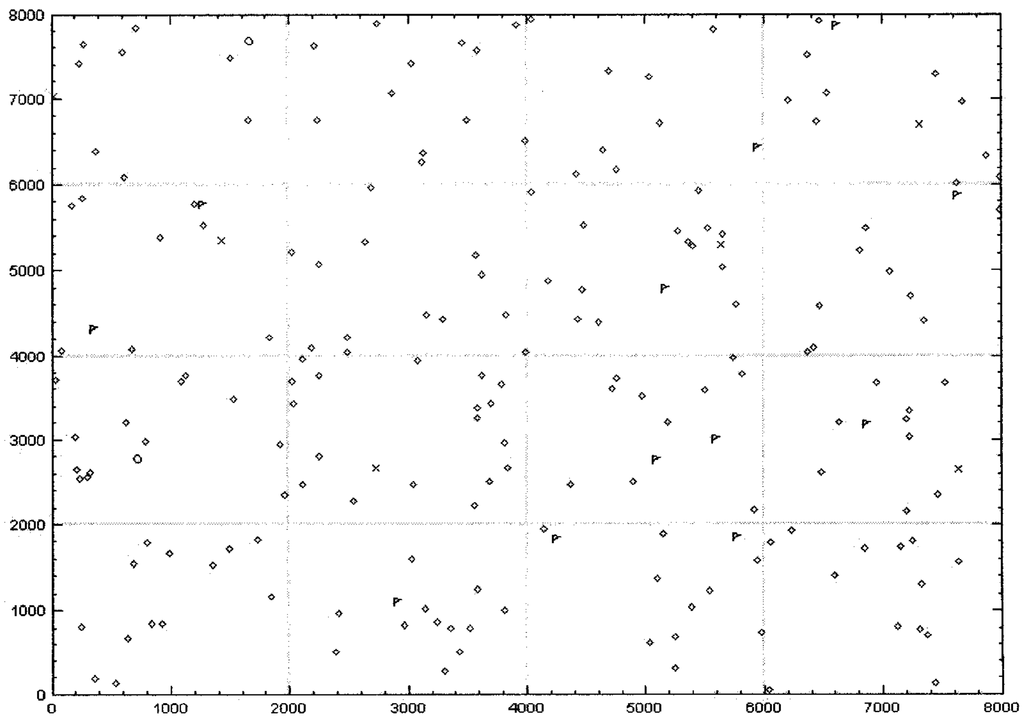

Figure 4. 2: Initial user distribution - 200 total users 10% multicast users

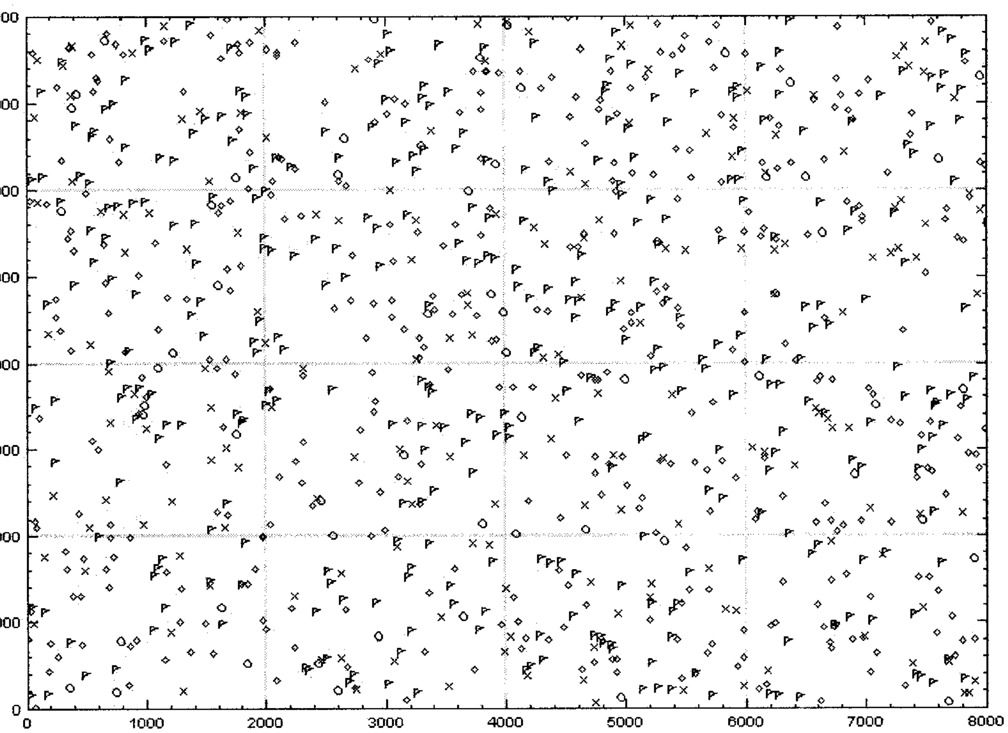

Figure 4. 3: Initial user distribution - 1000 total users 60% multicast users

4.3 Description of session's packet generation, probability of successful transmission, buffering and overflow parameters

Definition of simulator parameters related to subscriber's activities and network performance values are discussed below

4.3.1 Probability of successful packet transmission

The probability of successful packet transmission is one of the important parameters, together with the system load it defines the systems throughput. It has a direct influence on packet queuing for all sessions multicast, and unicast and in consequence over the packets transmission delay. For the simulation we considered different probabilities of successful packet transmission and measured the affect over the system performance. The probability of successful packet transmission $\mathbf{P_c}$ is defined as input simulation parameter and can take one of the following values:

$$\mathbf{P_c = 0.6, 0.7, 0.8, 0.9} \text{ and } \mathbf{0.99}$$

In practice this probability parameter changes from user to user and from location to location. It is a function of multiple factors such as the fading situation, the exact location and hence the probability BER (Bit Error Rate). This research is focused on the effect of the multicasting over the system performance. The fading situation is strongly related to the physical layer details which we do not cover in detail so, only one value of $\mathbf{P_c}$ for all of the users in the whole system is used.

In some cases of interactive real time applications multicasting there is no packet retransmission, because the failing of one packet to reach one of many other subscribers does not result in retransmission of the same packet over the network to all of the participants of the multicast session. Such applications also have significant tolerance in regards to the packets loss. However there are applications with very strong low loss requirements as for them error free information transfer (reliability) is the leading factor. Therefore we have incorporated the probability of successful packet transmission.

4.3.2 Buffer size. Iteration process and dependencies

The performance of the packet transmission is related to the free network resources, generated traffic, probability of successful transmission and the buffer size which define as result the number of overflowing packets dropped by the system. In real time applications as videoconferencing or teleconferencing which generate discontinuous data traffic (packets) it is essential that the buffer size be biger enough to accumulate all of the generated traffic and the percentage of the overflow dropped packets to be minimized. As simulation criteria we could monitor the influence of the buffer size and the probability of successful packet transmission over the dropped packets.

For the simulation a buffer is associated with each unicast subscriber. We have defined the buffer size for iteration **I** as **B(N,I)**; where **N** indicates the subscriber number in the system, **I** indicates the iteration cycle.

For example: B(3,2)=5 means Mobile Subscriber 3; Iteration 2; Current buffer size 5 (five packets were generated before and are waiting in the queue for transmission)

Different buffers are also associated for each session (unicast and multicast) on each Base Transmit Station (BTS).

The buffer size is indicated as **B^*(S,M,I)** where **S** specifies the BTS number; **M** indicate the session number for the cell; **I** indicate the iteration cycle. We consider geographic area divided to 16 cells (a square of 4x4 cells) therefore **S** varies from 1 to 16.

The maximum buffer size is an input parameter and will take the values shown below

$$\mathbf{B_{max} = B^{*}_{max} = 2;\ 4;\ 8;\ 12;\ 16;\ 32}$$

Each program iteration (for each user with non empty buffer – have one or more packets accumulated) will call a random number generator which returns a value **x**.

- If $\mathbf{x \leq P_c}$ then the packet is considered successfully transmitted. Therefore the packets on the queue (the current buffer size value) will decrease by one

$$\mathbf{B(N,I+1) = B(N,I) - 1}$$

- If $\mathbf{x > P_c}$ the packet is deemed to have failed transmission and have to be retransmitted. The number of the packets (the current buffer size value) remains the same.

$$\mathbf{B(N,I+1) = B(N,I)}$$

We ignore ground network transmission delay and errors in-between the Base Transmit Stations being connected with high speed links with BER less than 10^{-12}. If the packet is successfully transmitted to the mobile originated BTS we assume that the same packet has arrived at the buffer on the mobile destination BTS, immediately, without intermediate transfer points (as BSCs and MSCs) to cause any transmission delays. We acknowledge the fact that this delay could have a significant impact but for the GPRS system that we consider the GPRS coding schemes (CS) CS4 which allows data transfer up to 21.4 kbit/s and maximum 8 channels the speed is limited to 171.2 kbit/s the impact is negligible and could be ignored.

When the packet has arrived at the destination BTS the packet is placed on the queue. The BTS will transmit the data packet with probability of success $\mathbf{P_c}$.

As one BTS supports many unicast transmissions together with the three predefined multicast session each BTS has to keep separate buffers for each unicast user and different buffers for each multicast session.

Similar to the algorithm above, a random generated number is called by the simulator for each iteration. For each BTS session the probability of successful transmission will have value of **x** and

- If $\mathbf{x} \leq \mathbf{P_c}$ the packet is successfully transmitted and the number of the packets on the queue (the current buffer size value) will decrease by one

$$\mathbf{B^*(S,M,I+1) = B^*(S,M,I) - 1}$$

- If $\mathbf{x} > \mathbf{P_c}$ the packet has failed successful transmission and have to be retransmitted. The numbers of the packets (the current buffer size value), remain the same.

$$\mathbf{B^*(S,M,I+1) = B^*(S,M,I)}$$

4.3.3 Packet generation

The packet generation parameter has strong relation with the type of the service provided to the user. Different service i.e. video and audio conferencing, distance learning, e-commerce, distributed and multiparty online games and others require different QoS which reflects on the packet generation processes.

To have a precise monitoring of the system and flexibility of the simulator input parameters corresponding to different classes of multicast services we define packet generation per iteration from 0 to 8. The simulator is build in a way to have fix packet generation parameters for the different sessions or random generation from specified range.

If a range is specified, we will call random generated number identifying the new packet generated during the cycle.

Therefore the buffer size can be identified by:

$$\mathbf{B^{*}\ (S,M,I+1) = B^{*}(S,M,I) + G}$$

Where **G** represents the number of packets generated as random number from the selected range or as a fix value.

Every generated packet will increment the buffer on the unicast user transmitting data and base stations multicast buffers.

It is expected that the BTS buffer size which will transmit the packet to the destination unicast user depends not only on the packets generated by the unicast transmitter but also on the success of the transmission to the base station. Only if the generated packet is successfully transmitted to the base station (ignoring the intermediate points) this packet will increase by one the value of the unicast buffer assigned to this session on the retransmitting BTS (which covers the area where the receiving mobile station is).

4.3.4 Queuing and transmission delay parameter

The delay of the packet is calculated as the time difference between the creation and the successful transmission period. Thus the time spent on the queue is taken in consideration for the results.

4.3.5 Overflow and packet drop

If the buffer is full and new packets are generated for the multicast sessions or on the transmitting MSs the overflow counter is incremented by one for each packet dropped by the user/base station buffer. For the unicast transmission the successfully transmitted packets to the transmitter's BTS will generate new packets on the retransmitting BTS (the BTS which covers the area where the receiving MS is). If the buffer is overflowing and the sent packet will exceed the capacity, the packet will be dropped and the overflow counter is incremented by one.

4.3.6 Simulator implementation of buffer monitoring and data transmission

The basic flow chart is represented in Fig. 4.4. The flow chart of packet generation and transmission based on the probability of success is shown in Fig. 4.5. Consideration is also taken to monitor the the delay and packet overflow.

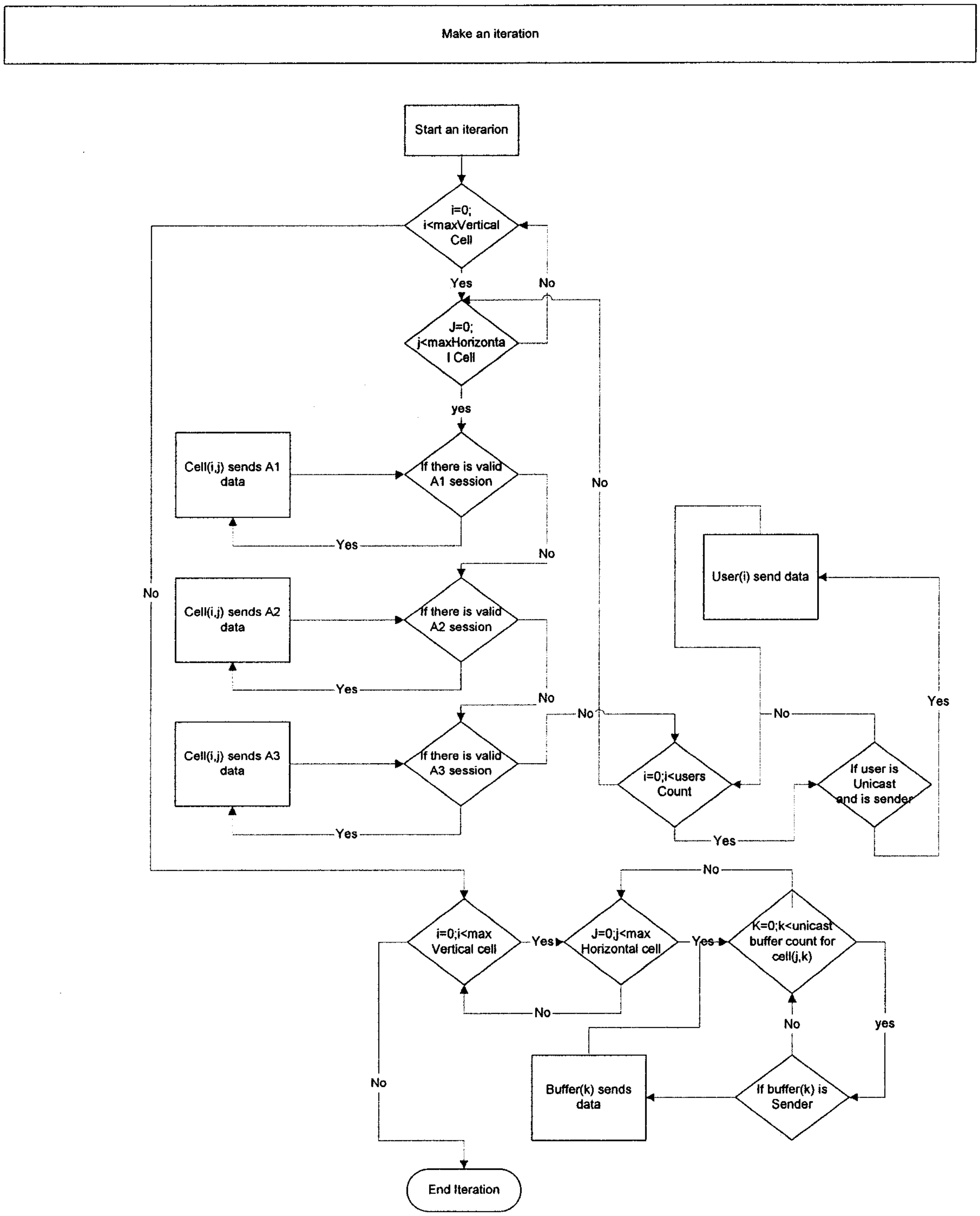

Figure 4. 4: Basic iteration cycle

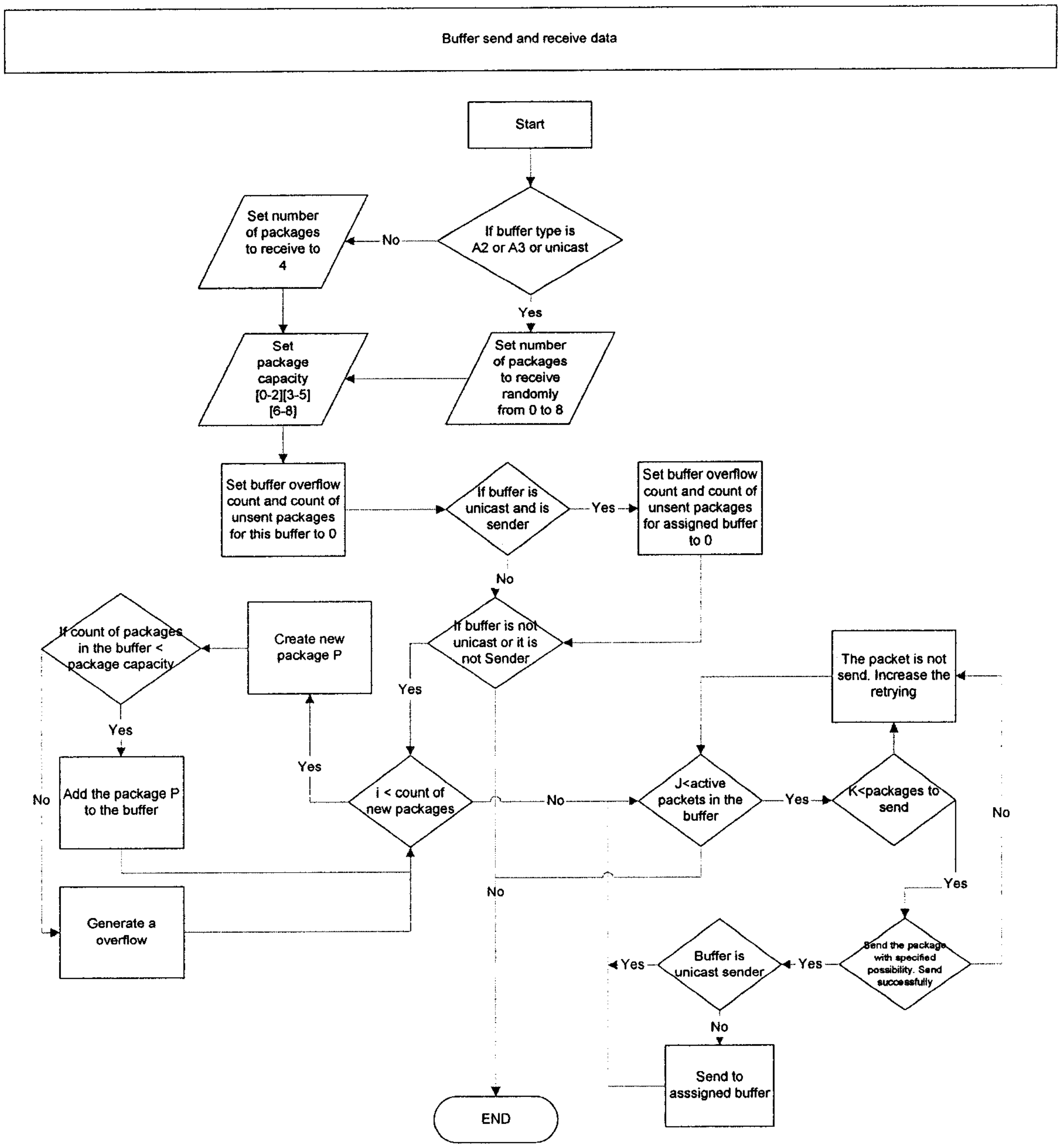

Figure 4. 5: Packet generation and transmission

4.4 Simulation outputs and results

In the following sections we present the simulations output results. Analysis are made and conclusions are drawn confirming the expected theoretical justification

4.4.1 Multicast channel utilization effects

Based on the simulator user distribution we can define the exact number of users belonging to different sessions -unicast and multicast per cell. Fig. 4.6 and 4.7 represent those metrics for different density modes of multicast users and different total number of subscribers.

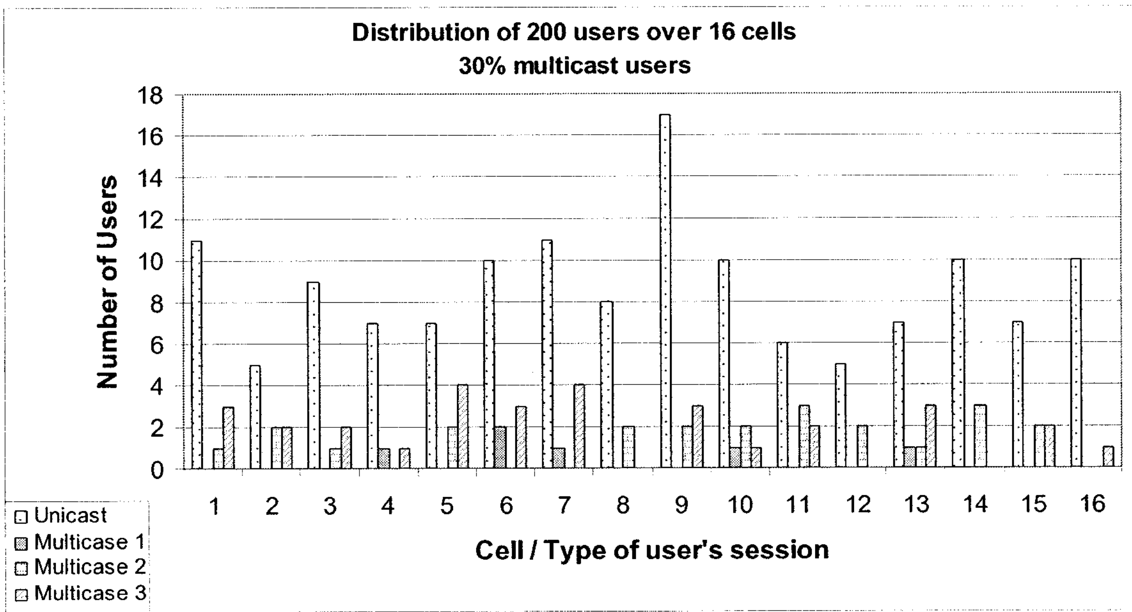

Figure 4. 6: 200 user distribution based on type of user sessions
30% multicast users from the total number

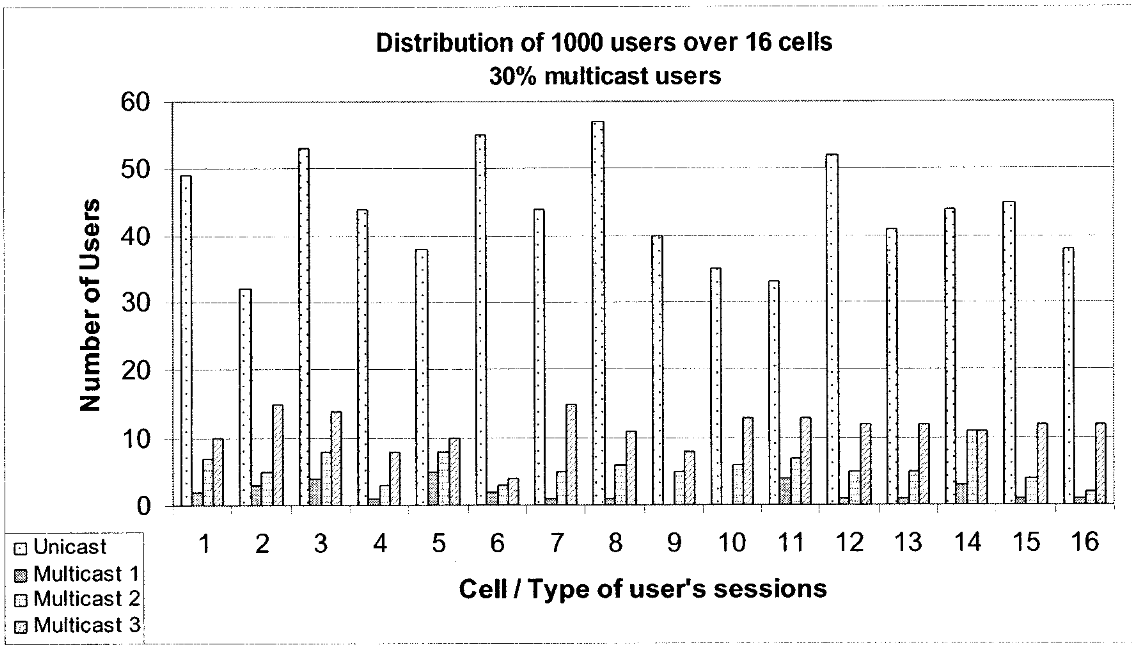

Figure 4. 7: 1000 user distribution based on type of user sessions
30% multicast users from the total number

From the represented data we can extract how many users in a cell participate in a specific multicast session and determine how many channels can be saved if multicasting is implemented. The minimum saved channels as result from multicasting utilization are the number of the multicast users minus one. One channel will always be used to carry the data to all of the multicast session participants. If a multicast session uses more than one channel the number of saved channels will be multiplied by the number of channels for the current session.

We have collected data for different cases and multicast user density which varies from 10% to 60% of the total number of users. We took into consideration the mobility of the users, as this will affect the number of specific multicast users per cell and therefore the minimum number of save channels. Simulations results were taken for four different time frames in intervals of 5 minutes. The average values of the minimum saved channels are represented on Fig. 4.8

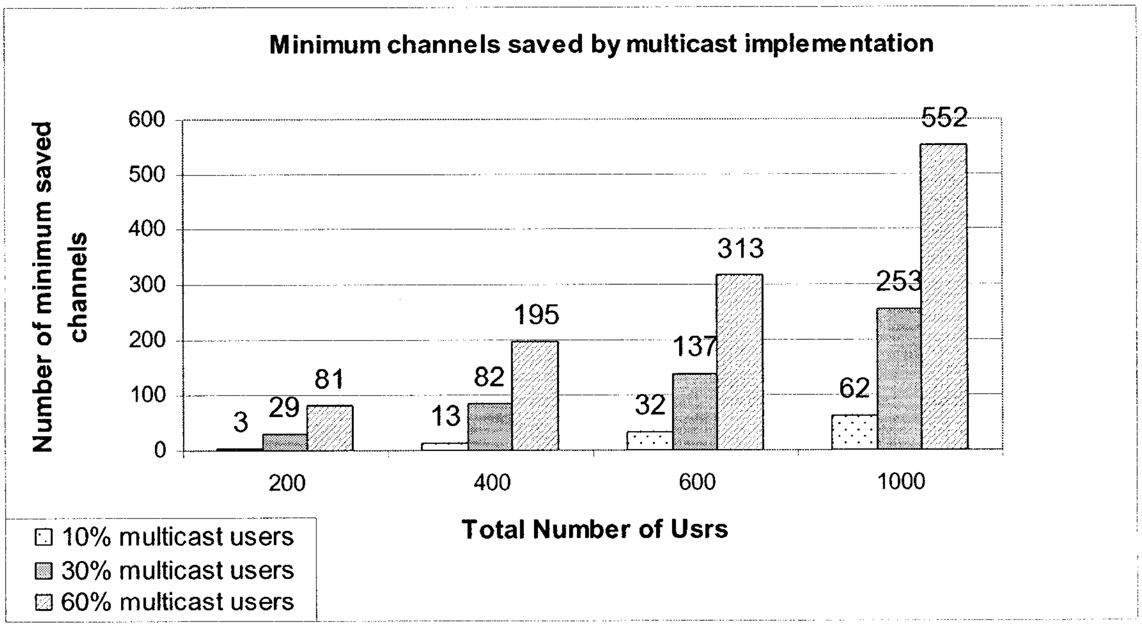

Figure 4. 8: Number of minimum saved channels average values

As mentioned before the values show only the minimum number of saved channels. The total number of rebased channels depends on occupied resources by the multicasting sessions. GPRS technology makes possible up to eight channels to be engaged in one session. Therefore as an example in the case of 30% multicast users from total of 1000 gives minimum of 253 saved channels and total of 253x8, 2024 saved channels over 16 cells. Those channels can be used for an additional of 2024 voice calls or 253 high speed data transfers at speeds of 171.2 kbit/s (8x21.4 kbit/s). Fig. 4.9 and 4.10 represents the increase of the number of minimum saved channels by increasing the total number of users or an increase of the density of multicast users. The results are showing a linear increase which is a result of the increase of the probability more users in one cell to join the same multicast session.

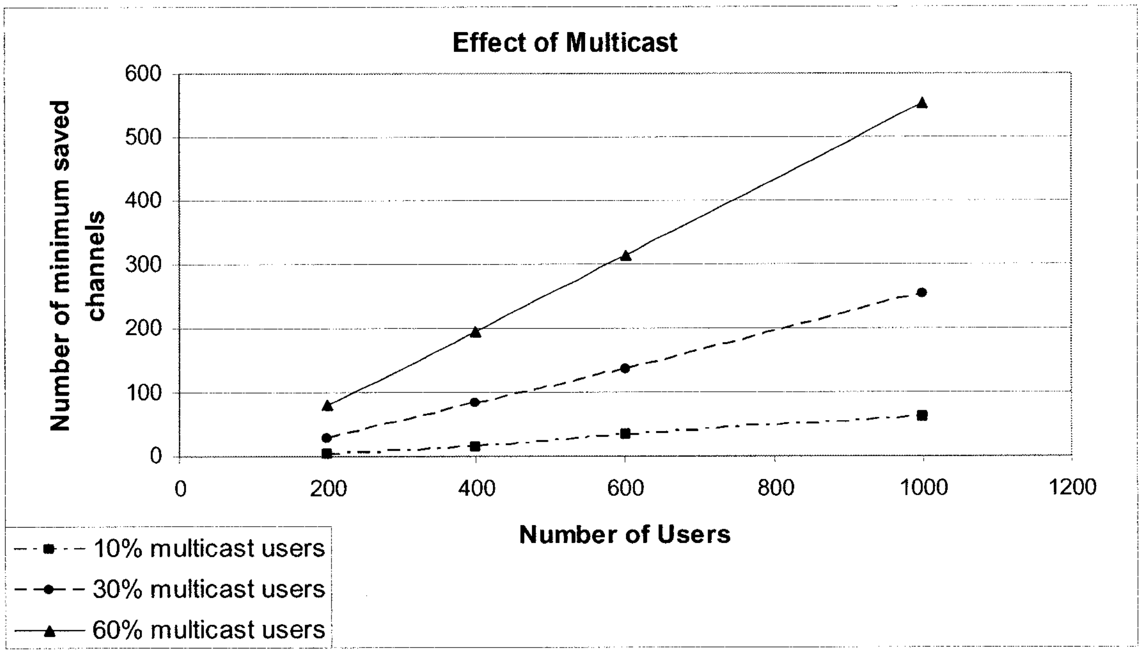

Figure 4. 9: Number of minimum saved channels average values by increase of the total number of users

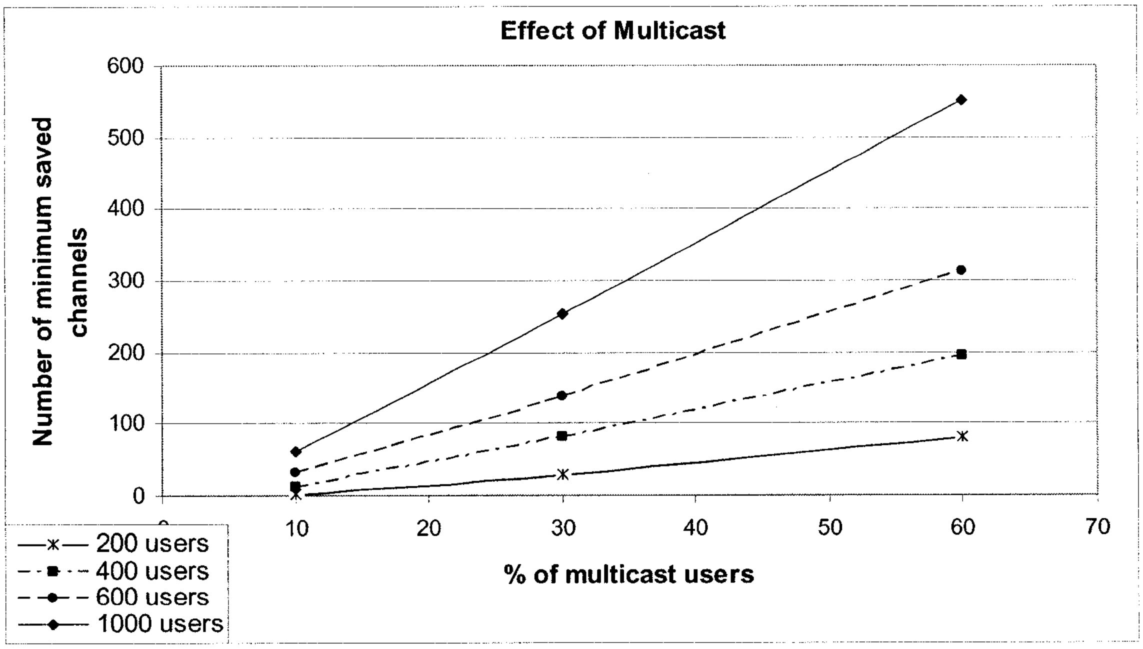

Figure 4. 10: Number of minimum saved channels average values by increase of the multicast density

4.4.2 Quality of Service dependencies

We have examined different criteria in relation to the system performance and provided QoS.

- **Delay**

The delay is monitored for every user/BTS and packet generated and transmitted over the system.

First we define,

$$D_{i,n} = IT_N - IT_1 \qquad \text{(Eq. 4.1)}$$

D - packet's delay on the queue caused by insufficiency of network resources or unsuccessful transmission $\mathbf{P_c} < 100$.

i - User ID number (corresponding buffer ID) on the MS site if MS is transmitting or on the BTS if the MS is receiving. The first three numbers are reserved for the multicast session ID numbers

i=1 Multicast session one; i=2 Multicast session two; i=3 Multicast session three

$\mathbf{i_{max}}$ - is equal to number of unicast users plus the buffers for the multicast sessions which is three times (three multicast sessions) the number of considered cells (4 cells x 4 cells) total of 48 (for the multicast sessions)

n - Packet number

$\mathbf{IT_N}$ - the iteration on which the packet is successfully transmitted

$\mathbf{IT_1}$ - the iteration of first attempt of transmission

The average delay of the packets for user/buffer **i** could be calculated from the following formula:

$$D_i = \frac{\sum_{n=0}^{N_i} D_{i,n}}{N_i} \tag{4.2}$$

Where $\mathbf{N_i}$ represents the total number of packets registered on $\mathbf{i}^{th}$ queue (generated packets from the transmitting MS and packets received from unicast and multicast users)

The total delay arithmetic mean

$$\overline{D} = \frac{\sum_{i=1}^{U} D_i}{U} \tag{4.3}$$

U is equal to the total number of buffers which is the number of unicast users plus 48 (buffers in 16 BTS for 3 simultaneous multicast sessions).

Standard deviation is the most common measure of statistical dispersion, measuring how spread out the values in a data set are. The standard deviation is the root mean square (RMS) deviation of the values from their arithmetic mean. The delay sample (or estimated) standard deviation is given by:

$$\sigma_D^2 = \frac{\sum_{i=1}^{U} (D_i - \overline{D})^2}{U-1} \tag{4.4}$$

- **Overflow**

If there is an incoming packet to the buffer when the buffer has reached the occupation of the entire dedicated capacity, the packet is discarded. A counter is assigned to each buffer.

The average number of discarded packets for each buffer (**i**) could be found using:

$$O_i = \frac{\sum_{G=0}^{K} O_{i_G}}{K} \tag{4.5}$$

Where **K** is the total number of iterations for which the simulation was run. We have run 3000 iterations during the simulations(K=3000).

The average number of dropped packets - the arithmetic mean is given by:

$$\overline{O} = \frac{\sum_{i=0}^{U} O_i}{U} \tag{4.6}$$

The variation of discarded packets (or estimated) standard deviation is given by:

$$\sigma_O^2 = \frac{\sum_{i=1}^{U} (O_i - \overline{O})^2}{U-1} \tag{4.7}$$

4.4.3 Simulation outputs and results

We have examined the effects of different parameters over the average packets delay, the delay sample, average values of discarded packets per simulation run and their standard deviations. All of these criteria are functions of the system capacity, probability of successful packet transmission and buffer size.

To study the impact of the system capacity over the system performance we have selected a system with 600 users with 30% multicast density. As mentioned, all of the parameters are related to each other. Therefore to eliminate the effect of the other parameters we selected the following input data:

Probability of the successful packet transmission to be on it's maximum and is set to 0.999%. It means that the delay will not be influenced by failed transmission and every packet will be transferred almost in a single attempt.

Buffer size was set to it's maximum for the selected simulation, i.e. 32 packets.

In order to present a wide picture, we have assigned a range of values to packet generation of the different multicast sessions. In practice this depends on the provided service (video streaming, news reports, remote downloading). GPRS limits the transfer to 8 time slots occupation per session, therefore all of the available applications are optimized for the corresponding data speed rate or up to 8 packets generation pre frame. We have selected packet generation per iteration as the following:

Multicast session 1 – 8 packets per frame

Multicast session 2 – 4 packets per frame

Multicast session 3 – 2 packets per frame

Unicast sessions – average of 3 packets

Simulation was run for values of the Packet transmission from 1 to 8 with step of 1.

The summarized simulation output data for the average packets delay and delay sample is shown on Fig. 4.11, 4.12 and 4.13.

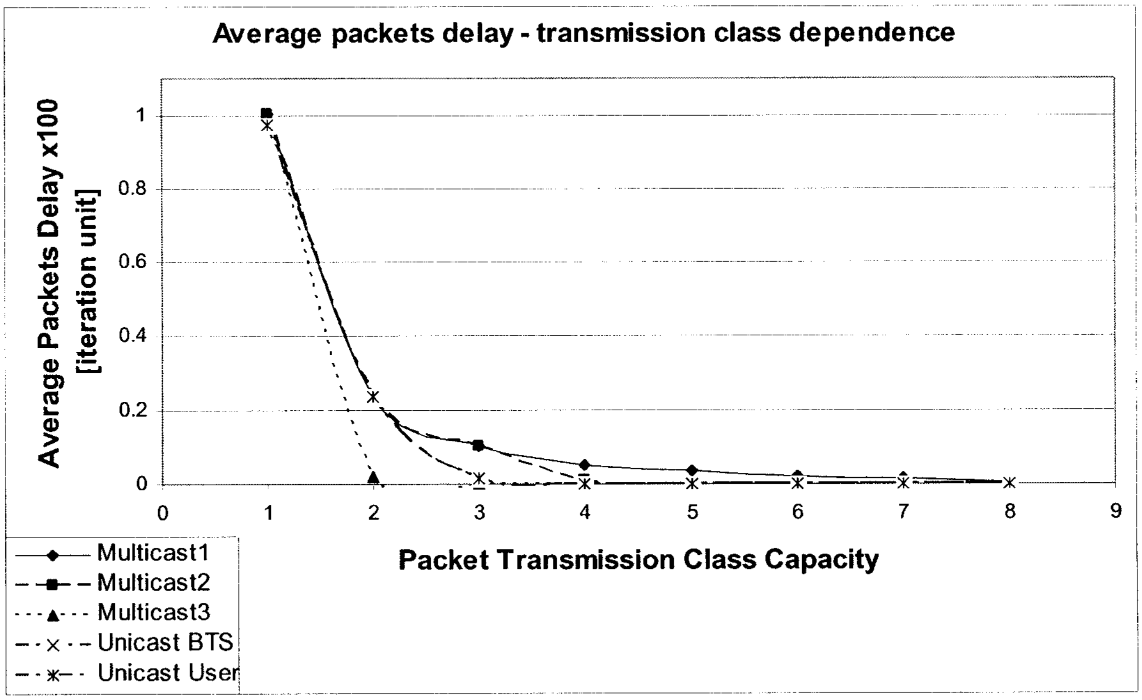

Figure 4. 11: Average packets delay - transmission class dependence

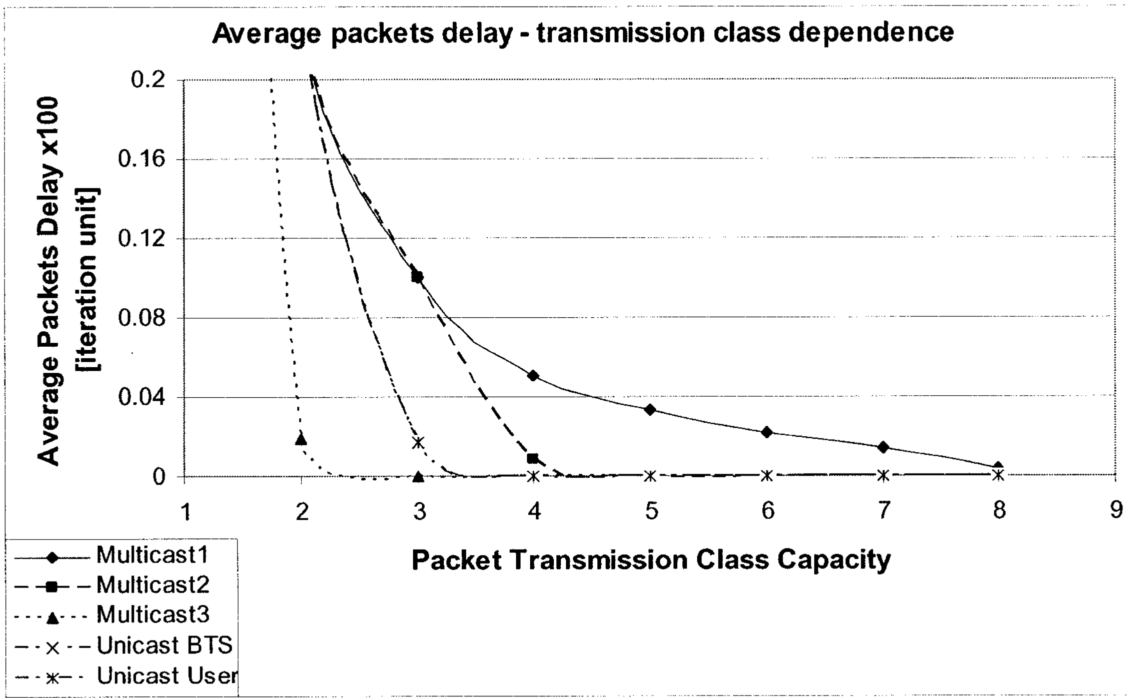

Figure 4. 12: Average packets delay - transmission class dependence Expanded View

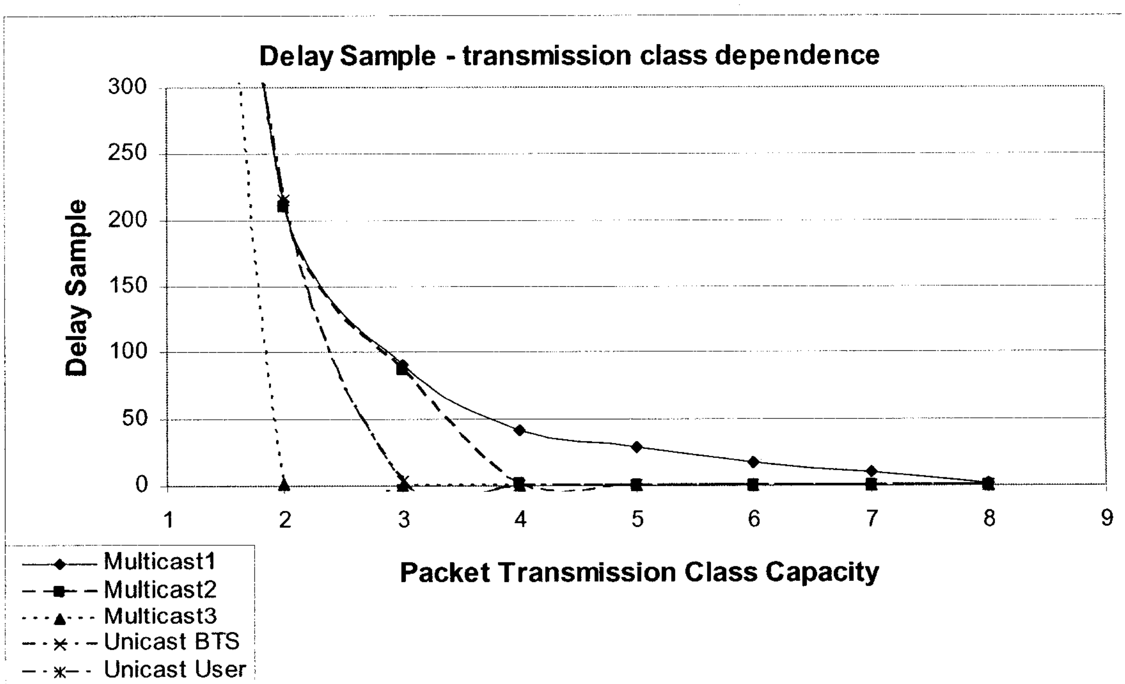

Figure 4. 13: Delay Sample - transmission class dependence

The Computer Simulation Iteration Unit represent a cycle with duration of 0.577 ms (one burst – 156.25 bits). As seen from the charts the average delay has its threshold values when the packet generation becomes equal to the system ability to transfer the packets or the Packet transmission class. The packet transfer rate could also be limited by the user subscription for certain QoS.

Fig. 4.14 and 4.15 represent the information for the discarded packets average values and the overflow deviation.

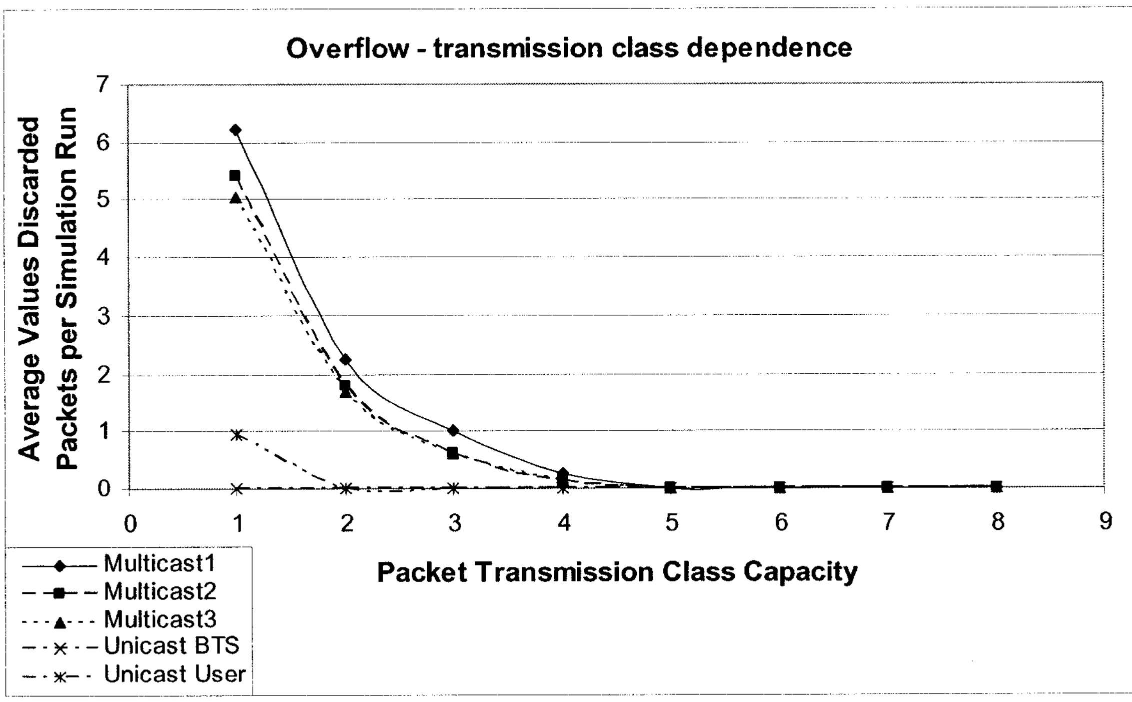

Figure 4. 14: Average Values Discarded Packets per Simulation Run
Transmission class dependence

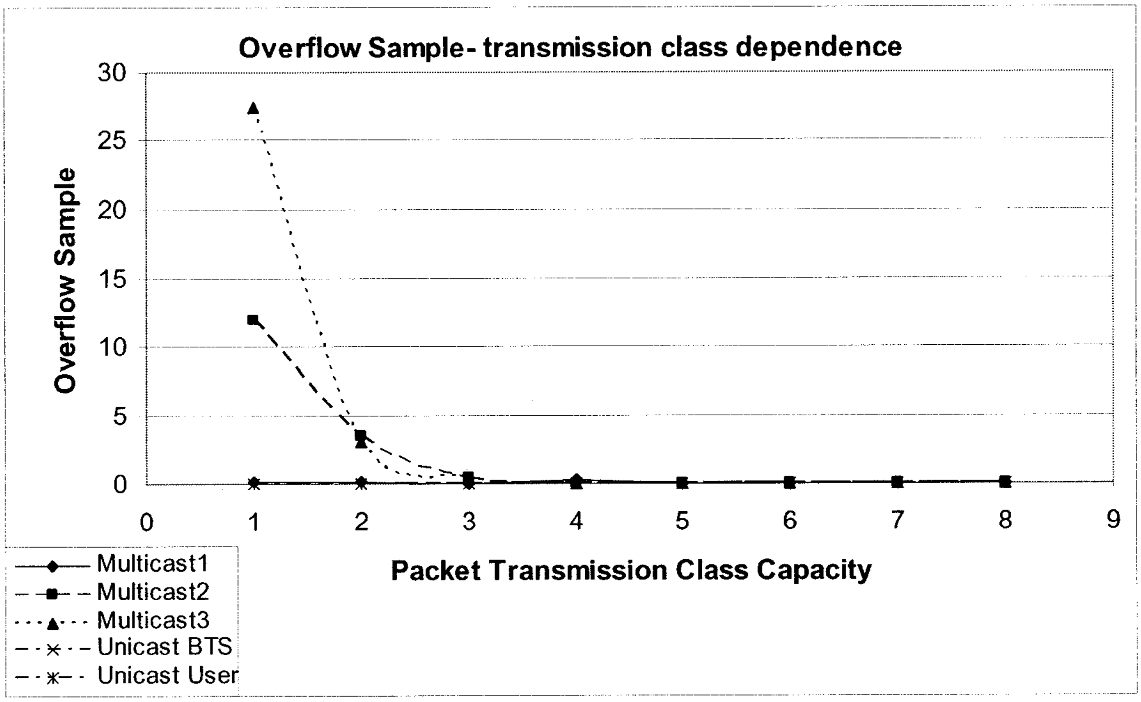

Figure 4. 15: Overflow Sample- transmission class dependence

The average values of discarded packets also decrease significantly when the system provides bigger capacity and can transfer more packets per iteration.

As generalization of the presented outputs we can define the strong dependence of the packets delay and the number of the discarded packets from the system capacity. This has leaded us to expand the research in the field of multicasting. By implementation of multicasting we improve the system performance by saving of radio channels, and gain smaller delay and packets overflow.

However the number of free channels that can be utilize play significant role for the system's QoS, in the previous study we neglect the probability of successful transmission in order to have more precise monitoring of the system capacity influence.

To define the effect of the probability of successful transmission we examine a system defined as the following:

- 600 users with 30% multicast density
- Buffer size of 32 packets per buffer
- Rate of packet generation for the different sessions per iteration:

    Multicast session 1 – 8 packets per iteration

    Multicast session 2 – 6 packets per iteration

    Multicast session 3 – 4 packets per iteration

    Unicast sessions – average of 3 packets per iteration

- Packet Transmission Class (System Capacity) of 2 packets per iteration

The packet transmission of two per iteration was selected to be less that the packets generated by any of the sessions, if it is bigger it will result in reaching the threshold conditions. Threshold conditions will result in no discarded packets and average delay close to zero. Computer simulations were run for different values of the probability of successful transmission 0.6, 0.7, 0.8, 0.9 and 0.99.

Fig. 4.16, 4.17, 4.18, and 4.19 show the study results

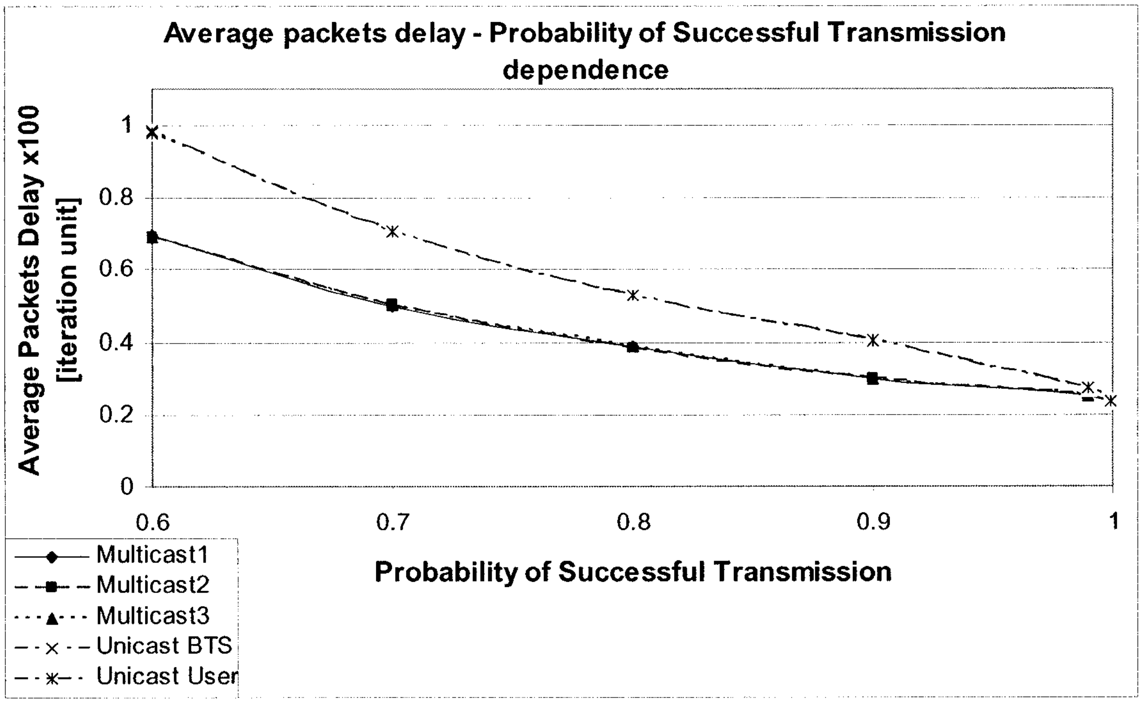

Figure 4. 16: Average packets delay - Probability of Successful Transmission dependence

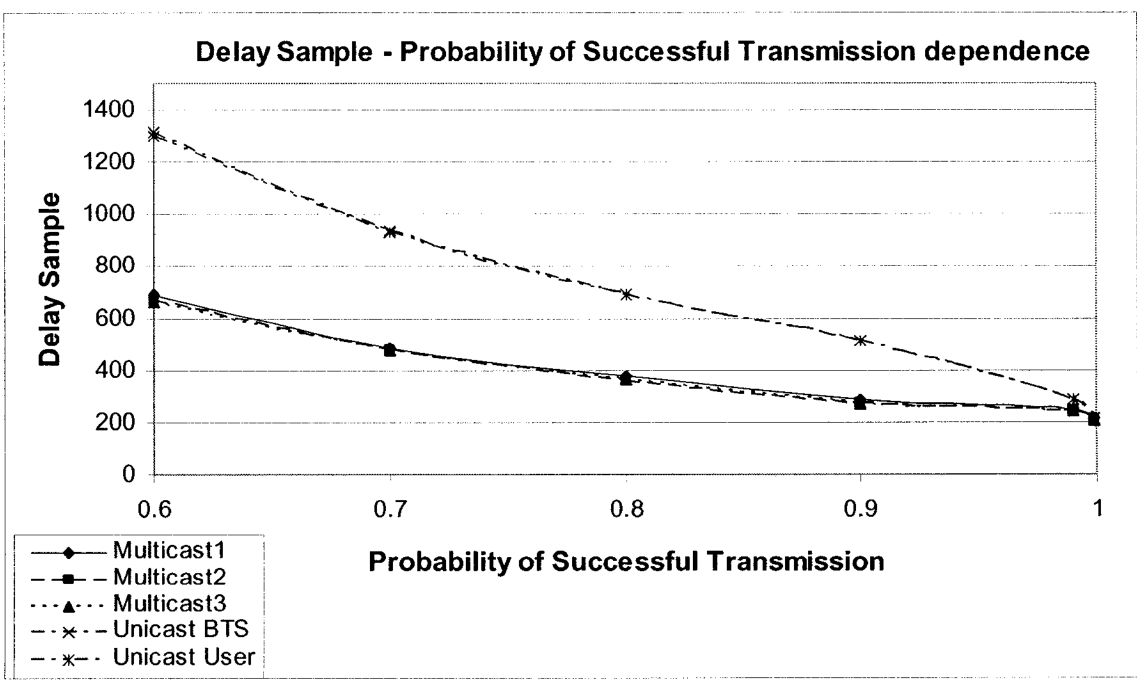

Figure 4. 17: Delay Sample - Probability of Successful Transmission dependence

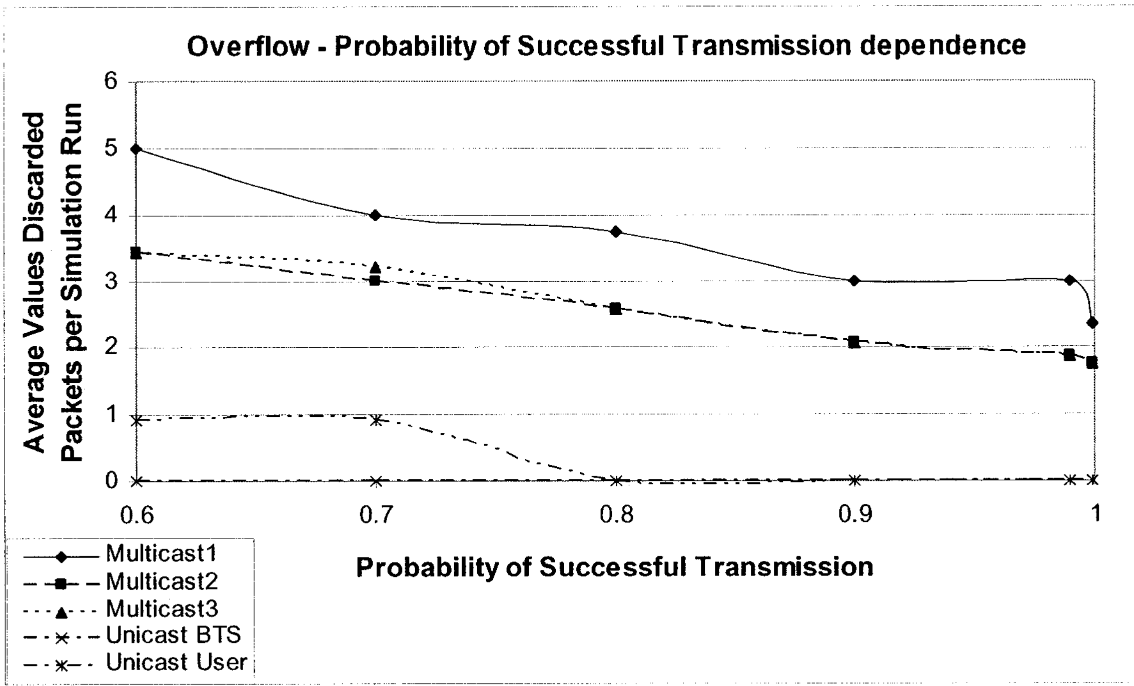

Figure 4. 18: Average Values of Discarded Packets per Simulation Run - Probability of Successful Transmission dependence

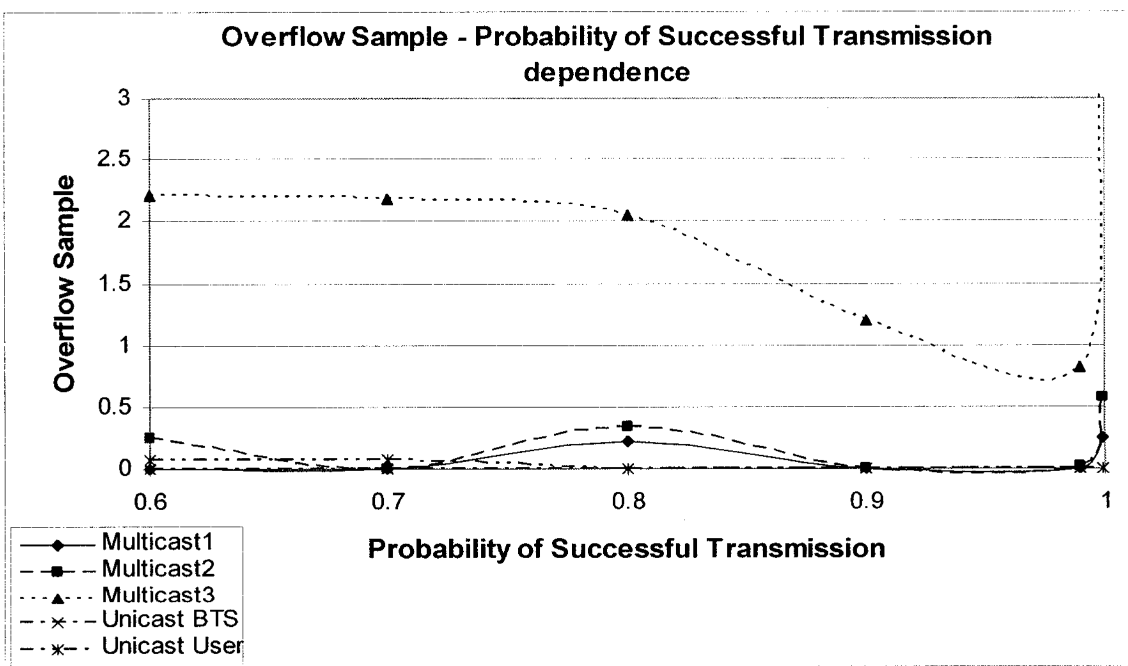

Figure 4. 19: Overflow Sample - Probability of Successful Transmission dependence

Simulations were run for 0.6, 0.7, 0.8, 0.9, 0.99, and 0.999% for Probability of Successful Transmission parameter.

As expected with the improvement of the quality of the connection (better Probability of Successful Transmission) the values of average Delay and dropped packets have decreased.

This parameter is more linked to the user location and the fading situation, however it plays major role with the collaboration of the system capacity to characterize the overall system QoS.

An other parameter influences the system that we study i.e. the effect of the system buffer capacity.

We have examined the same system as the previous case but with fixed Probability of Successful Transmission at the value of 0.999% and Packet Transmission Class (System Capacity) of 3. Simulation was run for the following values of the buffer size: 2, 4, 8, 12, 16, and 32 packets per buffer. Summarized output data is shown in Fig. 4.20, and 4.21.

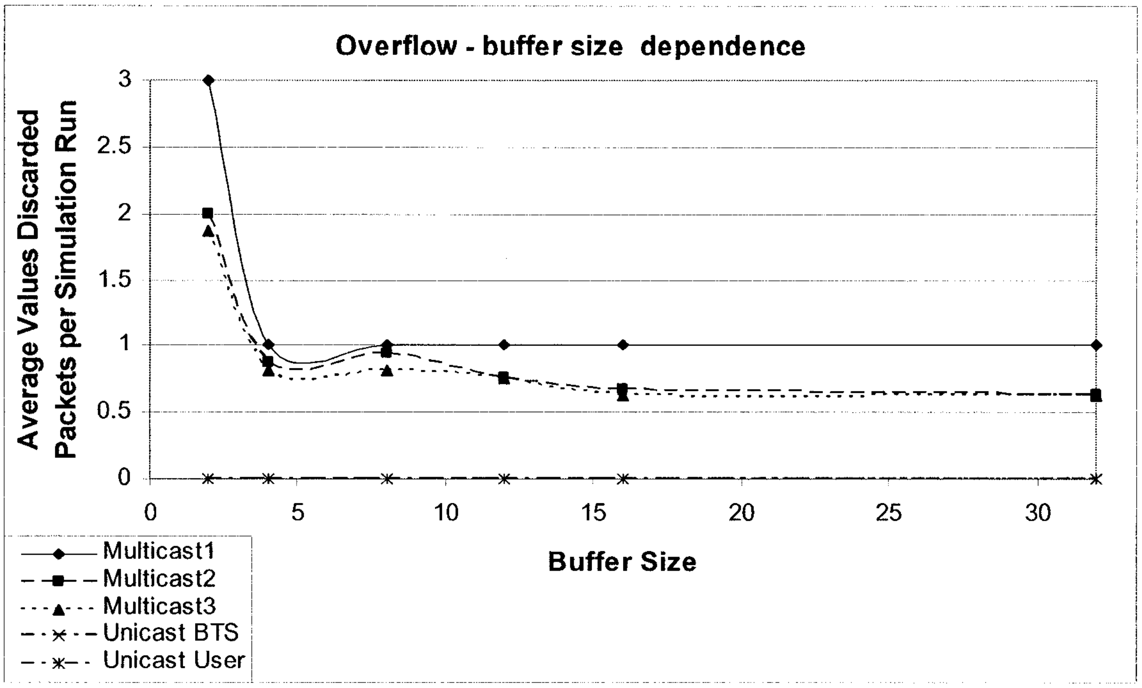

Figure 4. 20: Average Values Discarded Packets per Simulation Run - Buffer Size dependence

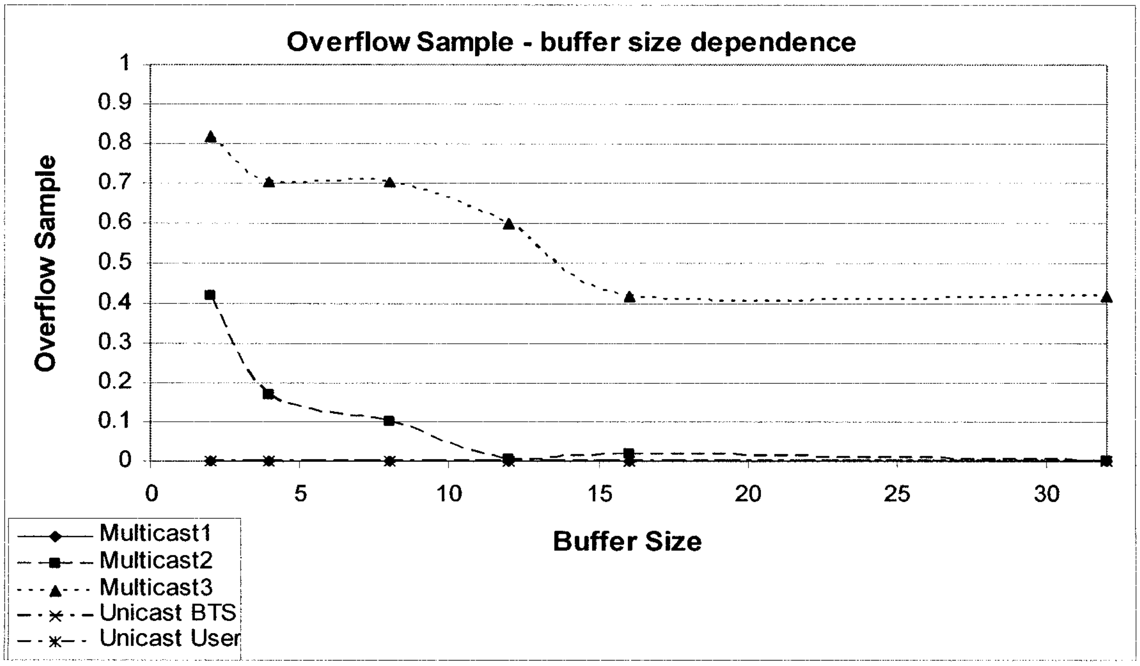

Figure 4. 21: Overflow Sample – Buffer Size dependence

The provided results show that the Average Discarded Packets per Simulation Run depends from the buffer size and can significantly be reduced by extending the buffer size.

The improvement is much better for small values of the buffer capacities; therefore with small extension to approximately 5 packets per buffer we can achieve much better outcome for discarded packets.

Based on the simulation results we have proved that multicast implementation on GPRS platforms lead to improvement of the system capacity and furthermore results in improvement of the provided QoS to the customers.

Chapter 5

Conclusions and further works

5.1 Contribution and conclusions

The goal of this thesis was to show the possibility of a significant improvement of network resource utilization and improvement of quality of service by the implementation of multicast technology on GPRS platforms.

We have introduced the concept of GSM networks as a base for the GPRS model. Complete studies were done on radio resources management, frame structure and multiplexing mechanisms. GPRS structure elements, interfaces and protocols were examined.

Detailed presentation of the multicast concept and expected effects over fixed and mobile networks were made. We justified the demand of multicast technology development. Examples of multicast services and available multicast protocols were presented. Theoretical base validation of the implementation of the multicast service on a GPRAS platform and expansion of the provided services by the GPRS structure elements was presented.

Simulations have been made to validate the expected theoretical effects of multicasting and came close to real environment.

In conclusion from the theoretical basis and the simulation results we proved that by multicast implementation there is a considerable improvement of network resource utilization which leads to maximizing the system capacity. As a consequence, the newly saved channels expand the opportunities for new services to be made available to the customers and better quality of service to be provided.

5.2 Suggestions for further work

In a world of a wider resource utilization and the popularization of applications which are suitable for multicasting, defining multicast protocols for mobile networks will play a significant role. The newly introduced protocols can be further developed and observed in real networks. This will play a significant role in regards to the traffic reduction on the core network and optimization of the network cost effectiveness.

The distribution and the location change especially for users located on the area borders and covered by more than one base station could give us a basis for further research in the area of Sticky Point-to-Multipoint channel for multicasting.

Further development of multicast protocols could play a significant role in regards to the traffic improvement on the core network and reducing the signaling

In future if we can expand the research for UMTS technology where the mobile data transmission is up to 2Mbit/s we could take in consideration also the intermediate delays as they will be comparable to the delay on the radio site.

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