"Congestion Control in UMTS Under Multicasting"



By DATA ENTERED

Muhammad Irfan (29 - FET/MSEE/F05)

Shahzad Ali Khan (21 - FET/MSEE/F05)

This dissertation is submitted to IIIUI in partial fulfillment of the requirement for the degree of MS Electronic Engineering

(Majors in Telecommunication Engineering)

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External Examiner

T -

Dr. Abdul Jaleel, Associate Professor, Department of information and Computer Sciences, PIEAS, Islamabad

Internal Examiner

Dr. Tanveer Ahmad Cheema Assistant Professor, Dept. of Electronic Engineering, Faculty of Engineering & Technology, International Islamic University, Islamabad

Supervisor

Engr. Raza Hasan Abedi Assistant Professor, Department of Electronic Engineering, Faculty of Engineering & Technology, International Islamic University, Islamabad

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Muhammad Irfan 29–FET/MSEE/F05

Shahzad Ali Khan 21–FET/MSEE/F05

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International Islamic University, Islamabad

Pakistan

Undertaken By:

Muhammad Irfan

29-FET/MSEE/F05

&

Shahzad Ali Khan

21-FET/MSEE/F05

Supervised By:

Engr. Raza Hasan Abedi

Assistant Professor,

Department of Electronic Engineering,

Faculty of Engineering & Technology,

International Islamic University, Islamabad

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Chapter # 1
Introduction to UMTS

1.1 Introduction

One of the most advanced communication systems is the Universal Mobile Telecommunication System (UMTS) that offers a series of broadband services to wireless and mobile communication world. UMTS is considered in third generation communication systems. There is provision of very low-cost mobile communication in UMTS at 2 Mbps data rate. It conserves capacity the global roaming of second generation communication systems offering new improved capabilities. A variety of data delivery services are being offered by UMTS now days like pictures, graphics, video communications, and other multimedia information, as well as voice and data, to mobile wireless subscribers.

The extension in existing second generation communication system like GSM/GPRS and adopting the Wide-band Code Division Multiple Access (WCDMA) technology results a new communication system as UMTS. In this advanced communication systems, there is complete support design for Handover between both the technologies as UMTS and GSM. The role of GPRS in the whole design of UMTS is just as a junction point between second generation technologies and the packet-switched domain of the third generation communication system of UMTS.

Few of the mobile broadband services are offered by all UMTS operators. Also, a number of PC vendor present laptops with built-in HSDPA potential that will definitely boost the usage of data. Supplementary data services having less dispersion today but with high expectations for future development, include mobile email access, mobile TV, mobile gaming and full track music downloads.

1.2 UMTS Services

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Currently, UMTS maintain services of both types as voice and data. There are some date rate targets being used in UMTS as given below:

- 44 kbps This date rate is used for Satellite and rural outdoor
- 384 kbps This date rate is used for Urban outdoor
- 2048 kbps This date rate is used for indoor and low range outdoor

There are different parameters of quality-of-service (QoS) regarding data services for proper data transfer. Four types of traffic are there which are operated by different QoS classes available in UMTS:

Conversational Class covers Voice, video telephony, video gaming

Streaming Class covers Multimedia, video on demand, web cast

Interactive Class covers web browsing, network gaming, database

access

Background Class
 E-mail, short message service (SMS), file

downloading

1.3 UMTS Architecture

The UMTS architecture is constituted with three major categories of network elements as discussed below:

GSM Core Network Elements

This elements of UMTS architecture consists of Mobile services switching center (MSC), visitor location register (VLR), home location register (HLR), authentication center (AuC), and equipment identity register (EIR)

GPRS Network Elements

This element of UMTS architecture consists of Serving GPRS support node (SGSN) and gateway GPRS support node (GGSN)

• UMTS-specific Network Elements

This element of UMTS architecture consists of User equipment (UE) and UMTS Terrestrial Radio Access Network (UTRAN) elements

1.4 UMTS Interfaces

There are four new open interfaces defined in the UMTS. These interfaces are also shown in figure 1.1.

- U_u Interface User equipment to Node B (Air Interface)
- I_u Interface RNC to GSM/GPRS (MSC/VLR or SGSN)
 - I_u-CS Interface for circuit-switched data
 - I_u-PS Interface for packet-switched data

- I_{ub} Interface RNC to Node B interface
- Iur Interface RNC to RNC interface
- The I_u, I_{ub}, and I_{ur} interfaces are based on the transmission principles of asynchronous transfer mode (ATM)

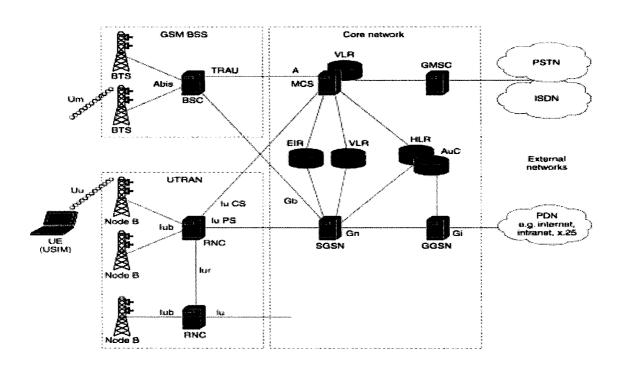


Figure 1.1: UMTS Architecture

1.5 General Packet Radio Systems

The General Packet Radio System provides the progressive change from GSM networks to 3G UMTS networks. The GPRS supports GSM networks by enabling packet switching and allowing direct access to external packet data networks (PDNs). Data transmission rates above the 64 kbps limit of integrated services digital network (ISDN) are a requirement for the enhanced services supported by UMTS networks. The GPRS conducts the optimization of core network for the substitution to elevated data rates. Therefore, the GPRS has the importance for the preface of the UMTS as a prerequisite. UE and host are named as TCP/IP protocol suite and the applications are positioned at the end-nodes. The transmission and reception of network Packet Data Units (PDUs) is

conducted by the Packet Data Coverage Protocol (PDCP). PDCP is also responsible for holding the one to one association from one network protocol to one RLC entity. It also functions for compression and decompression at transmitting node and receiving node respectively of redundant network PDU control information

1.6 UMTS Terrestrial Radio Access Network

The air interface transmission is the apparent difference between GSM/GPRS networks and UMTS networks. There are two access techniques being in use in GSM/GPRS network as Time division multiple access (TDMA) and frequency division multiple access (FDMA) while wide-band code division multiple access (WCDMA) air interface access technique is selected for UMTS networks. This air interface access technique of WCDMA has two basic modes of operation: frequency division duplex (FDD) and time division duplex (TDD).

This new air interface access method requires a new radio access network (RAN) called the UTMS terrestrial RAN (UTRAN). The network elements of the UTRAN are shown in Figure 1.2

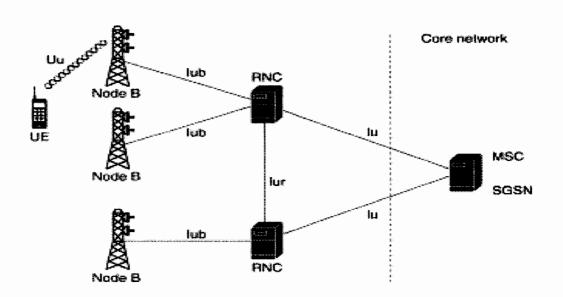


Figure – 1.2: UTRAN Architecture

The UTRAN is accommodated by making few changes in the core network.

There will be addition of two network elements as in the UTRAN. One is called radio

network controller (RNC) and second one is Node B. A combination of more than one radio network systems (RNSs) constitutes UTRAN where every RNS is proscribed by one RNC. RNC is responsible for the connections between one or more Node B elements. Every Node B can function for multiple cells. The functionality of RNC in UMTS is the same as the base station controller (BSC) functions in GSM/GPRS networks. In UMTS networks, Node B provides the functionality like base transceiver station (BTS) in GSM/GPRS networks.

So this is one sort of protecting the investment of mobile wireless operators in UMTS by making enhancements in existing GSM and GPRS networks. It also makes possible to launch new services over vacant interfaces such as A, Gb, and Abis, and new interfaces. These new interfaces are introduced between Node B, the RNC (Iub) and between two RNCs (Iur).

1.6.1 Radio Network Controller

The functionality of the radio network controller (RNC) is the same as to the base station controller (BSC) works in GSM/GPRS networks. Node B elements in their covering areas are centrally supervised by the RNC's. All sort of protocol exchanges between UTRAN interfaces (Iu, Iur, and Iub) are handled by RNC's. Being the interfaces as ATM-based, switching of ATM cells are performed by the RNC over the interfaces. The transmission is made over the Iur, Iub, and Uu UTRAN interfaces to and from the user equipment (UE). The RNC's supports joint multiplexing for circuit-switched and packet-switched data from the Iu-CS and Iu-PS interfaces.

The operations and upholding of the radio network system (RNS) is performed by the RNC including access to an operations support system (OSS) using the Iur interface. Contrarily to the GSM/ GPRS networks, radio resources in UTRAN are managed differently. Core network is responsible for radio resource management in GSM/GPRS network. While in UMTS, a space is provided to the core network to handle other functionalities by freeing it from radio resource management and this task of managing the radio resources in UMTS to the RNC's. Now in UMTS network, a single RNC is responsible for different types of control functions such as connection establishment to

the UE, congestion control, and handover procedures. There are few functions of the RNC given as below:

- Radio resource control
- Admission control
- Channel allocation
- Power control settings
- Handover control
- Macro diversity
- Ciphering
- Segmentation and reassembly
- Broadcast signaling
- Open loop power control

There are three different modes of functionality of Radio Link Control (RLC) layer as: acknowledged mode, unacknowledged mode and transparent mode.

- The delivery of reliable data is the responsibility of the acknowledged mode over the error-prone radio interface. This is done by retransmitting erroneous RLC PDUs.
- In the *unacknowledged mode*, the data transfer over the radio interface is error some but retransmission reduces additional delay.
- The *transparent mode* works same to the unacknowledged mode but PDU is not edited by adding extra protocol information in it.

Medium Access Control (MAC) manages data streams directed to it from RLC and RRC and provides an unacknowledged transfer mode to the upper layers. Logical channels also termed perform the communication between RLC and MAC.

Similarly, the transport channels also called SAPs handle the communication between MAC and PHY. Furthermore, the utilization of physical channels at the radio interface is controlled by Physical (PHY) layer. It is the responsibility of PHY to map transport

channels on to physical channels. PHY also responsible for different operations related to the communication reliability features like forward-error correction and error detection, interleaving, spreading, modulation, rate matching, and radio frequency processing.

1.6.2 Node B

It is the radio transmission and reception unit to communicate between different radio cells. One or more cells are served by each Node B unit. Existing GSM base transceiver station (BTS) and UMTS Node B unit are physically positioned for making the UMTS accomplishment possible with a reduced cost. Node B is responsible for connecting the user equipment (UE) over the Uu radio interface using wide-band code division multiple access (WCDMA). A single Node B unit can support both frequency division duplex (FDD) and time division duplex (TDD) modes.

There are different types of functions performed by Node B which of them are given as below:

- Air Interface Transmission and Reception
- Modulation and Demodulation
- CDMA Physical Channel Coding
- Error Handling

Downlink transmission power control is used by UE to adjust its power that is supported by Node B. These are the RNC power control parameters which initial values for power control.

1.6.3 UMTS User Equipment

As SIM in GSM/GPRS networks, the UMTS user equipment (UE) is constituted with the combination of the subscriber's mobile equipment and the UMTS subscriber identity module (USIM). The USIM is just like a card which is inserted into the mobile equipment and it makes identification possible for the subscriber to the core network. The physical characteristics of USIM card to the SIM card in GSM/GPRS.

Following are the functions of USIM:

- It provides supports to multiple user profiles
- It performs updates USIM information over the air interface
- It provides functions related to security
- It also works for user authentication
- Payment methods are supported in USIM
- It supports secured downloading of new applications

Chapter # 2

Multicasting in 3G Wireless Networks

2.1 Introduction of 3G Multicast

Third generation networks are the next phase of wireless communication systems to support mobile users with improved services. 3G networks are aimed to design a global wireless infrastructure that will change the static communication networks environment. These networks are able to facilitate users with internet access all the time. As an adoption of these enhanced communication infrastructure, users will be aspired have applications and access services similar to in case of conventional wire line networks where all these services are being run currently.

The multicasting is such a method in which data-grams are transmitted from a single source to several destinations. This method can also be considered to make the transmission possible from one point to multi-points in a group. This is an important objective to implement multicast over wireless networks. There are lots of issues to be resolved in such an infrastructure where many group applications are to be installed on a large scale. As compared to uni-casting or broadcasting, multicasting is a proficient way to support group communication. Also some network resources are utilized to allow transmission and routing of packets to multiple destinations.

Premeditated findings are multicast to users, tanks, and planes in military environments. Nominal delays, secured and reliable wireless multicast are required to such types of applications required. These services are also very proficient in case of distance education and entertainment services to be offered to mobile or remote users. For the observance the quality, such type of applications demand high bandwidth and near-real-time wireless multicast. Dynamic routing of vehicles is managed by intelligent transportation systems. Few main issues in wireless multicast are the reliability and correctness of location information.

2.2 Multimedia Broadcast/ Multicast Service In UMTS

It is an IP data cast type of service which is accessible via existing GSM and UMTS cellular networks. Data is transmitted from a single source entity to a group of user of specific area in MBMS. MBMS is a unidirectional point to multipoint service. There are two modes of operations. One is the Broadcast mode and second is the Multicast mode.

Broadcast service is such a service in which data is transmitted from a single source to multiple destinations associated with broadcast service area. Multicast services can be defined as a unidirectional point to multipoint service in which data is transmitted from a single source to a multicast group in the associated multicast service area. In this scheme of communication user must be registered to specific multicast group which can receive the multicast services.

Multimedia Broadcast / Multimedia Service consists of two services: one is MBMS bearer service and other is MBMS user service. In MBMS bearer service, the user is served with multimedia contents while in MBMS user service different type of operations like user authorization, charging and Quality of Service, improvement to preventing unauthorized reception are performed.

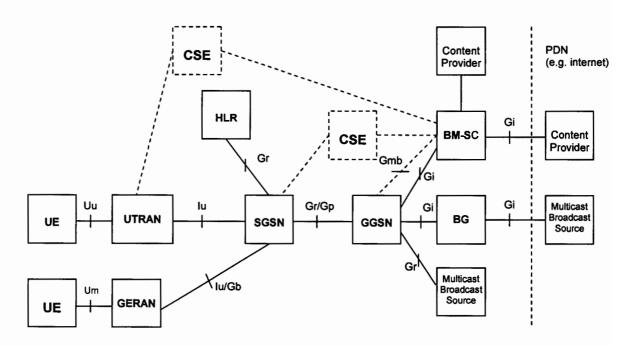


Figure – 2.1: MBMS Architecture

The architecture of MBMS is shown in Figure 2.1 above. The existing GPRS architecture is modified by the addition of new entity named as Broadcast Multicast-Service Center (BM-SC). This is responsibility of newly added entity as BM-SC to make the communication possible between UMTS, GSM and the external Public Data Networks (PDN's). There are certain procedures to enable reception of an MBMS multicast service.

These are the different phases involved in MBMS multicast service. These phases are Subscription, Service Announcement, Joining, Session Start, MBMS Notification, Data Transfer, Session Stop and Leaving. The phases which are performed for each user subscribed for this service separately are Subscription, Joining and Leaving.

If a deep fading occurs for a considerable amount of time, then the channel may cause burst errors. Also, the performance of TCP gets affected and TCP misinterprets the packet losses due to congestion when random errors occur in channel. To keep away from this situation, TCP shortens the size of the packet to be sent even in case of no congestion.

2.3 Packet Forwarding Mechanism

Apart from the User Equipments (UEs) in UMTS, while implementing multicast mechanism there is an introduction of Routing Lists (RL's) in every node of the network. The routing list of any node contains the information about all nodes lying under that level connecting to it with such user equipments (UE's) which belong to a specific multicast group. As a result, it is observed that there is only one routing list for each multicast group in each node. To illustrate this mechanism a network topology is discussed in Figure 2.2 shown below. It is assume that UE1, UE2, UE11 and UE12 belong to multicast group MG1. GGSN preserves a RL for this multicast group which contains all the Serving GPRS Support Nodes (SGSNs) which connect GGSN with the UEs belonging in MG1.

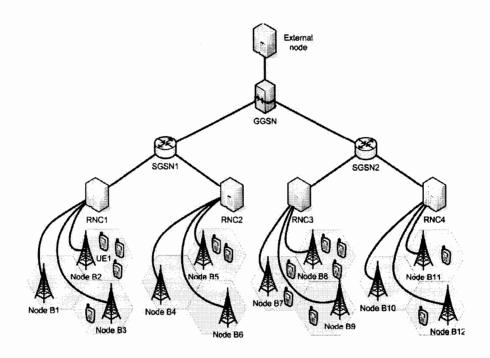


Figure 2.2 Multicast Packet Transmission

The connecting SGSNs are SGSN1 and SGSN2. On next level, a RL with the Radio Network Controllers (RNCs) is placed with each SGSN which connect it with the UEs belonging in the examined multicast group. The RL of SGSN1 contains RNC1, the RL of SGSN2 contains RNC3 and RNC4, while RL of SGSN3 for MG1 is empty. Similarly, the RL of RNC1 contains UE1 and UE2, the RL of RNC3 contains UE11 and that of RNC4 contains UE12. The rest RNCs have empty RLs.

Considering the case when incoming traffic for the previously defined multicast group MG1, reaches the examined PLMN. Suppose that there is an incoming packet to GGSN and the packet addresses to MG1. Routing lists are checked by the concerned GGSN. If this RL contains lower level nodes, the packet is forwarded once to each one of them. The mechanism of the packet forwarding is repeated recursively by the lower level nodes until the packets reaches the UEs belonging to the specific multicast group.

It must be very clear that there are two other kinds of lists to be maintained. These two lists are Drift Routing Lists termed as DRLs shortly and Multicast Group Lists termed as MGLs shortly. The RNCs uses Drift Routing Lists (DRLs) while the Multicast Group Lists (MGLs) are used in the GGSN.

The DRLs used for the purpose of inter-RNS soft handover. Each DRL communicate to a multicast group and enclose pairs of RNC-UE. Multicast transmission over the Iur interface is the major objective of these lists. The functionality of MGLs is to represent a specific multicast group and to maintain the UEs which are subscribed to the specific multicast group. Regular updates are compulsory for the purpose to achieve the correct and reliable multicast data transmission.

2.4 Multicast Group Management

In case of UMTS network which is offering MBMS service, subscription phase of a UE is accomplished and its joining to the multicast group is required. Now, the execution of service announcement phase would be required for the purpose of joining of UE to a multicast group. The list of currently existing multicast groups is requested by sending a message to the GGSN. The list of multicast groups is sent back to UE in reply to its request when GGSN received message by UE. Decision is made by UE to join a specific multicast group from the list provided by GGSN. There are number of steps are involved for the multicast group(s) Joining phase which are discussed in next paragraph as given below.

The UE sends a join-request message to the GGSN, specifying the multicast group MG which the UE wants to join in. The GGSN checks the subscription profile and, assuming that it accepts the request, the GGSN adds the UE in the corresponding MGL. Then, it checks if the SGSN which serves the UE, exists in the relevant RL. If it does not exist, it adds it in the RL. Finally, the GGSN sends an acknowledgement message to the serving SGSN. When the SGSN gets the acknowledgment, it examines if the serving RNC of the UE exists in the RL that it maintains for the MG. If the RNC does not exist, the SGSN adds it in the RL. Then it forwards the acknowledgment to the RNC. When the

RNC gets the acknowledgment, it adds the UE in the proper RL and forwards the acknowledgment to the UE. When the UE receives the acknowledgment, the establishment of the context is finally confirmed.

Leaving phase is initiated if any UE is in need of leaving a multicast group. Similar steps are involved as in joining phase. Also the message sequence to be sent for this purpose during this phase is similar to the methodology discussed in joining phase of UE to multicast group(s).

2.5 User Mobility

Regarding user mobility, a way of implementation is conferred which hold the user mobility to guarantee MBMS service stability. Impacts of soft handover over the proposed methodology are inspected. The adaptation of the methodology to the SRNS relocation practice is also discussed properly.

2.6 Soft Handover

Lets a case is assumed where a UE which is subscribed to multicast group and MBMS multicast service in Data Transfer phase. During the transmission of multicast packets to multicast group members, the cell is changed by UE. Now three different possible situations can occur for analysis for soft handovers.

- 1. Inter-Node B/intra-RNS soft handover.
- 2. Inter-Node B/inter-RNS/intra-SGSN soft handover.
- 3. Inter-Node B/inter-RNS/inter-SGSN soft handover.

The existing routing lists at the network nodes are not affected, therefore the proposed methodology also does not effected in above outlined scenarios. The last two scenarios are managed in same profession. The transparency of handover to SGSN(s) is the fundamental concept to be considered. For this purpose, a new interface, the Iur interface is introduced.

There are number of different steps are involved in inter-RNS soft handover procedure which are discussed in the figure 2.3 shown below. The existing handover methodology of UMTS is a base of the proposed mechanism with number of additions to assure MBMS service stability. During the analysis phase all these additions are discussed briefly.

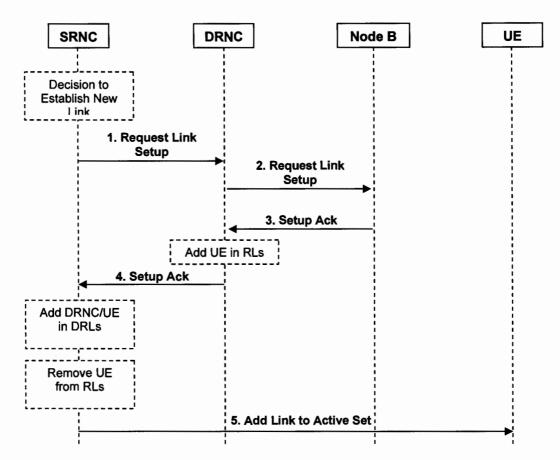


Figure 2.3: Inter-RNS Soft Handover

2.7 Multicast Routing for Ad Hoc Wireless Networks

Very high degree of user mobility and router mobility is offered by Ad hoc wireless networks. During a session, the path established between two users can be changed due to unimpeded mobility. In this sense the usage of the multicast routing protocols designed for infrastructure-based wireless networks is not possible in ad hoc networks.

Ad-hoc network does not have any fixed infrastructure like base stations and switches therefore all nodes including mobile hosts are to be computed, maintained, and routing information is to be stored properly.

2.8 Reliable and Secure Wireless Multicast

Reliable and secure multicast is an essential need for many applications. These applications must be able to provide end-to-end reliability that requires detection of packet loss and error recovery as well as.

There are two different approaches to detect packet losses:

- Sender-initiated. Receivers return acknowledgments for correctly received packets; timers can be used to detect packet losses at the sender. However, if every receiver sends an acknowledgment for a packet it receives, feedback implosion can occur.
- Receiver-initiated. Negative acknowledgments (as in the receiver-initiated approach) are used by receivers to inform the sender about packet loss via negative acknowledgments. Loss recovery can be performed through selective retransmission to the receivers that did not receive packets.

The security issues can be addressed in a number of ways; for example, packets in wireless multicast can be encrypted using symmetric (private-key) or asymmetric (public-key) schemes. But the key distribution and re-key processes, in light of user mobility and membership changes, can create significant processing and network overhead. In group key encryption, a group key management protocol securely delivers a common key to all users of a multicast group [4]. Group authentication is implicitly provided by possession

of the key; sender authentication can be provided via other means, including digital signature.

2.9 Wireless Multicast Adoption

There is great importance of wireless multicast to service providers like internet, cellular and personal communication, content, and multiple sites based businesses in need of instantaneous updates. Multicast is utilized by service providers to support services related to the distribution of contents for which their customers charge premium to them. Wireless multicast can also be used to distribute software and data updates to branch offices and stores spread all over the world.

Apart from this sort of valuable importance, there are also many important issues which must be tackled before the wide deployment of multicast. Some of these major issues are like: charging wireless customers in new business scenarios, distribution of revenue among service providers; maturity of wireless multicast software, applications, middleware support, vendor support, router interoperability and trust of user that is very much necessary for conducting mobile transactions.

A continuous communication among group users may become sophisticated due to user mobility combined with wireless link characteristics. Several problems may be lead if no response is observed from a user some time. In this situation, occurrence of time out or the user's re-connectivity is awaited to get the normal mode of application. It is not clear that how the length of a time-out is calculated. Therefore, frequent removal of mobile users from the group can occur by choosing a short value whenever they experience brief disconnectivity

2.10 Effect of Loss of Packets and Acknowledgement

Mobile commerce applications are affected in different way by the loss of packets and acknowledgments. The loss of packet can affect the overall application's result in a mobile sale while loss of one or more messages in a mobile advertising application is not a big deal. Reliability findings are based upon two levels; one is transaction and second is the application. At the transaction level, a sequence is followed to execute all the steps. Hierarchy of step is taken into account very carefully. The transaction may have to be aborted and scheduled on latter times if any one of the steps is unable to complete in a certain time. These are the essential reliability measures to be ensured properly before the deployment of such techniques in case of Mobile commerce applications.

Chapter # 3

Congestion Control in 3G Wireless Networks Υ.

3.1 Introduction

The wired networks use Adaptive Window based control as it is also used by TCP. It becomes essential aspect to check its proper functionality in wireless networks. This is the need to understand the differences between wired and wireless networks.

In TCP, the window size is reduced whenever packets are lost. This hypothesis is invalid in wireless networks because the channel may have been performed poorly thereby causing loss of packet at the physical layer. Due to the traffic congestion occurred; the figures of this sort of losses may vary as compared to packet drops at a router queue.

There are few points to understand properly as discussed below:

- 1. To distinguish the reason of data losses in wireless networks whether they are due to traffic or channel.
- 2. To set proficient ways to achieve objective if reasons for data losses have been found.
- To formulate methodology to tackle the issue of corrective measures against any found cause of data loss.

3.2 CC STRATEGIES IN A GERAN/UTRAN SCENARIO

The discussion has been accomplished on such area that is served by GERAN and UTRAN sites provide service over a same area. Due to the different medium access nature of GERAN and UTRAN systems (TDMA vs. CDMA), congestion will be detected and solved differently in each of the RATs.

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The following sections tackle the congestion detection mechanisms in each of the available RATs along with congestion resolution strategies both in a common and local perspective.

3.2.1 Congestion Detection

The procedures defined for Congestion Detection required keeping away from two situations. One of them is false Congestion detection and other one is the non-detected congestion. The first one is related to the case when a congestion situation is detected when the overload is not been happening on air interface. The second one is concerned with unobserved congestion when the air interface is overloaded. To avoid the aforementioned problems, it is necessary that fast reactivity and high measurement must be exhibited reliability the Congestion Detection.

3.2.2 Congestion Resolutions

In following subsections, to overcome congestion situations in the network three mechanisms can be discussed as given below.

3.2.2.1 Vertical Handover Congestion Resolution (VHO-CR)

This method of congestion resolution is related to improve congestion avoidance by performing a VHO over a group of prioritized users in the congested RAT/Cell. While we intend to increase radio frequency by directing users to UTRAN in case of congestion situation in GERAN. If the target cell is congested then a VHO will not be implemented. The procedure involved in VHO is the selection of a base station. It is suppose that the BS having best signal strength is selected in GERAN and base station having higher signal strength is selected.

3.2.2.2 Bit-rate Reduction Congestion Resolution (BRR-CR)

This method of congestion resolution required to minimize the transmission rate to lower congestion by data users subscribed for a specific data service. So, there will be definite reduction in QoS regarding throughput significantly by adopting this mechanism.

There are two types of BRR strategies which can be defined so that a proper reduction in bit-rate can be achieved.

a) Maximum BRR (MAX-BRR):

Maximum allowable transmission rate reduction is applied to achieve the reduction in bit-error.

b) Minimum BRR (MIN-BRR):

In this scheme, allowable reduction on each user is the minimum. Congestion metrics are observed after a BRR is performed to see whether the congestion avoidance is achieved or not. Congestion resolution process will be finished if the congestion avoidance is achieved. In other case BRR is performed to the prioritized list users.

3.2.2.3 User Dropping Congestion Resolution (DROP-CR)

This CR strategy reduces the overload of the system by selectively dropping users in the congested cell/RAT. By doing so, load factor is reduced in UTRAN and RF is increased in the same fashion than in the VHO-CR scheme. However, the DROP-CR presents the highest negative impact on users' perceived QoS and should, therefore, be only used if other strategies fail to solve congestion.

3.2.2.4. User Prioritization Considerations

To reduce the overload of the system, congestion resolution strategies are applied on a number of users. There can be number of criteria to select users. Few of these criteria have been discussed here

a) Service-class prioritization

To get the expected QoS needs, users are placed from high to low priority. Setting the lower priority to the users' congestion may be resolved. Real Time (RT) and Non Real Time (NRT) service demands can be considered examples of these methods applied. The congestion resolution algorithm may start dropping non real time users to solve the congestion because RT services are more inflexible in QoS requirements.

b) User-Type Prioritization

To suppose QoS the users must be premium-based and then preferential treatment can be expected. To diminish the overloaded situation consumer users will be downgraded.

c) Capacity-Consumption Prioritization

Different amounts of resources are consumed by service users like in GSM, a voice user use a whole slot for its transmission in both directions and in GPRS, lower resource consumption is contributed by data user because they can share the same timeslot with other users. Therefore the users having the highest resource consumption are de-allocated by the congestion control algorithm to avoid congestion on the network.

3.3 Approaches of End To End Multicast Congestion Control

All the receivers are affected by sender which uses a single transmission rate and user infers network congestion from feedback collected from receivers. This approach is very useful for such applications having bulk data transfer while the primary goal is to deliver data reliably to all receivers but some receivers may suffer delay in waiting for others. To deal problems such as implosion control and drop-to-zero problems, the sender-based approach also requires extreme careful design.

A receiver-based mechanism is also used to favor other types of sensitive to delay but tolerant to some amount of quality reduction applications such as audio/video. Data is structured into layers and transmitted onto different multicast groups in this mechanism. A proper amount of data is chosen by receivers which can be accepted under the current network condition. For reliable bulk data transfer, problems stem from the receiver-based approach as data does not come with a natural layering. It is hard to organize data into layers, yet maintaining data consistency and ordering. The number of

available layers is normally confined to be small due to the complexity of encoding and decoding, thus limiting the adaptive range of congestion control.

The receivers must coordinate to join and leave a multicast group. This causes overheads and dependency on proposed multicast routing protocols in receiver-based approach. To control congestion in sender-based and end-to-end for reliable multicast this strategy is referred.

3.4 Multicast Congestion Control Issues

The efficient use of network resources is the main goal of congestion control by being responsive and adaptive to the network congestion occurred. When this principle is applied to multicast congestion control, there are two fundamental issues to be taken place:

- Scalability
- Fairness.

3.4.1 Scalability

For all multicast based protocols, the scalability issue is the most essential aspect. Apart from the need to be scaled to a large number of receivers, multicast congestion control protocol also needs to scale in a more heterogeneous environment with different link capacities and delays. There are two resulting problems to be discussed. One of them is feedback implosion and other one is rate drop-to zero.

In all mechanisms extra delay in feedback is considered very carefully. There is direct relationship of feedback delay to congestion control schemes the receptiveness. The responsiveness would be less if delay is longer. This irresponsiveness of a multicast flow is especially dangerous to the network. Because, irresponsiveness creates fluctuated link conditions potentially. It is also instability is expected into network.

The source uses these signals to regulate its transmission rate without proper aggregation when receivers use packet losses as congestion signal. This is considered as a problem. Receivers downstream of paths will all sent congestion signals to the source resulting in multiple rate drops at the source when packets are lost on multiple paths

independently. The data source does not know the receiver topology in the current IP multicast model, hence it cannot combine the congestion signals.

3.4.2 Fairness and Responsiveness

TCP is the prevailing transport protocol and its success is almost featured to its congestion control and error control mechanisms in the internet scenario now a days. It is important to design a multicast congestion control scheme which coexists and shares the bandwidth fairly with TCP. The fairness is defined as to accomplish TCP-compatible throughput on the worst sender-to receiver path in reliable bulk data transfer. The responsive time of TCP's window based congestion control mechanism is typically one fast retransmission or one retransmission time-out.

However, the sender is not necessarily informed of all packet losses and packets can be retransmitted locally in multicast situation. Additional complexity will be added by the combination of error control and congestion control and un-scalability to the protocol will be observed. However, it is still an open issue as how local retransmissions should be limited. The decoupling of the error control and congestion control implies that in time of extreme congestion when the packet loss ratio is very high, the different degree of responsiveness taken by multicast. Also degree of unfairness in the TCP congestion control will be observed.

3.5 Problem Formulation in Networks

To achieve reliable transmission of data over the network infrastructure is the most important goal in networking environment. Traffic congestion on network nodes is the leading source of packet loss and delay in wired networks. TCP uses the methodology of adaptive window control to deal with congestion issue. Apart from the above problem of congestion in wireless networks, unreliable data transmission and packet loss occur due to time varying channel. In this sense TCP looks to have very conservative designs if the congestion and fading time scales vary. To increase throughput there must be a joint strategy to contest both congestion and fading.

3.6 TCP Congestion Control

To control packets being offered to the network Transmission Control Protocol (TCP) uses Adaptive Window Management in wired networks. To maintain a particular flow of data, this mechanism allows the sender to adjust its sending rate to the current traffic conditions in the selected path. For the packets which have been received, the mechanism of acknowledgement is used by the TCP protocol to deal with sender-receiver flow control, from the destination to the source node. The source TCP uses a Window parameter which is the maximum number of unacknowledged packets that it can have at any given time during the connection. This controls the maximum number of packets offered by sender to the network without having received the acknowledgement for any of them. Adaptive Window Control involves changing the size of the transmission window based on the acknowledgement packets that the source has received adaptively.

3.7 Modification for Wireless Environment and Congestion Control

Link capacities vary randomly with the passage of time in a wireless network. So, the channel capacities are tracked and control decisions are taken ideally to distinguish data losses due to fading and due to congestion. To exploit the dependence of channel capacity on power can be the one method to be considered. The perception is to vary the transmit power adaptively to minimize the capacity variations due to fading.

Chapter # 4
Literature Survey

4.1. Literature Review

Various mechanisms for Multicasting in UMTS have been discussed in the literature. In [Mariann Hauge and Øyvind Kure, "Multicast in 3G Networks: Employment of existing IPMulticast Protocols in UMTS" WoWMoM'02, September 28, 2002, Atlanta, Georgia, USA, Copyright 2002 ACM] the author proposed the multicasting technique for UMTS adopted from the direct IP networks. In his work the author discussed three ways to implement the multicasting in UMTS network. The first architecture is included as an optional feature in the current UMTS standards. In this architecture multicast routing is terminated in GGSN. This solution requires minimal changes to the existing packet data network. It is able to use most of the existing UMTS mechanisms for charging, mobility, flow control, etc. This architecture does not utilize multicast's ability to save network resources.

The second architecture terminates multicast routing in the RNC. This solution requires moderate modifications to the UMTS packet-data network. This scheme reduced the resources from GGSN and core network. The third architecture terminates multicast routing in the base station. This solution requires substantial modifications to the packet-data platform and does not make much of the use of existing UMTS mechanism.

However, the first architecture does not provide any bandwidth savings in the UMTS network. The two other designs are Internet multicast architectures where the multicast functionality is pushed successively further out towards the UMTS terminal. Higher complexity is introduced to achieve network resource savings. The presented multicast mechanism employs the Internet Group Management Protocol (IGMP) for group management and relies on the standard hierarchical tunneling of UMTS for distributing multicast packets to the group. The hierarchical tunneling mechanism of UMTS, however, does not lend itself to efficient multicast packet delivery, since each tunnel may only be established for a single subscriber. Depending on the distribution of

the multicast users within the coverage area, this may lead to an inefficient usage of resources within the network [11].

The IP approach for multicasting cannot be adopted because of various reasons. Firstly As the multicast receivers are non stationary, and consequently, may change their point of attachment to the network at any given time. Second, mobile networks are generally based on a well-defined tree topology with the non stationary multicast receivers being located at the leaves of the network tree and the mechanism for mobility is not provided in IP multicasting scheme adopted in UMTS [11].

4.2 IP multicasting in UMTS

In [Rummler R, Chung Y, Aghvami H., "Modeling and Analysis of an Efficient Multicast Mechanism for UMTS". IEEE Transactions on Vehicular Technology 2005 the author does not adopt the mechanism of IP multicasting in UMTS instead used the Multicast packet data Protocol context for multicast group in GGSN and SGSN. For this the author introduced the number of new tables in GGSN, SGSN and RNC while some trivial changes in GTP are required for forwarding multicast packets.

4.3 Multicast Mechanism For Circuit-Switched GSM Networks

In Lin Y., "A multicast mechanism for mobile networks". IEEE Communication Letters 2001 a multicast mechanism for circuit-switched GSM networks is outlined that only sends multicast messages to Location Areas (LAs) in which multicast users reside. This mechanism uses the existed UMTS/GSM short message architecture in order to perform multicast routing. In particular, two new tables are considered in the Home Location Register (HLR) and in the Visitor Location Register (VLR). The multicast table at the HLR records the Mobile Switching Centers (MSCs) that serve multicast users, while the VLR keeps track of the LAs that have multicast users. However, the multicast messages are delivered to all the cells of an LA, independently of whether or not multicast users are located in all cells. This is inefficient if an LA is large or only sparingly populated with multicast users.

4.4 Point-To-Multipoint Channel For Multicast In UMTS

The cellular network is made of Cell and sectors in order to enhance the capacity of the systems .Transmission from the one antenna almost cover one sector. *In Mariann Hauge Unik*, "STICKY POINT-TO-MULTIPOINT CHANNEL FOR MULTICAST IN UMTS"- University Graduate Center Øivind Kure4 the author uses the concept that the Subscriber is covered by more than two sectors for the 40% of its call processing. This scheme will increases the coverage area of one multicast-channel by letting the multicast channels from selected BSs stick to subscribing terminals that are positioned well into a neighboring sectors' range. Three different physical channel types are adopted to serve the subscribers that are the "Normal point-to-point" distribution to each multicast subscriber in the sector, a point-to-multipoint channel to distribute the multicast data within the sector boundaries and a Sticky point-to-multipoint channel that covers multicast subscribing terminals in own, and parts of neighboring sectors.

The draw back of this approach is the Intercell and Intracell interference increased because of increase in power however capacity of cell will increase because we use fewer channels to transmit data to the set of terminals.

4.5 Routing Mechanism in UMTS

In Antonios Alexiou, Dimitrios Antonellis, Christos Bouras and Andreas Papazois, "An Efficient Multicast Packet Delivery Scheme for UMTS MSWiM'06", October 2-6, 2006, Torremolinos, Malaga, Spain the author proposed a routing list for multicasting in UMTS at every node in the network except the UEs and record the node of next level to which the message for multicast group should be forwarded. Drift Routing Lists(DRL) were added For the purpose of soft handovers.

In Antonios Alexiou1,2, Dimitrios Antonellis1,2, Christos Bouras1,2, Andreas Papazois and F. Boavida, "Multicast Approach for UMTS: A Performance Study" NETWORKING 2006 The author measure the performance of above mentioned schemes in terms of cost of the network, number of multicast users within the multicast group, the amount of data sent to the multicast users and the type of transport channels used for the

transmission of the multicast data over the air and show that the cost of the network increases with the increase of number of multicast UEs.

In Antonios Alexiou, Dimitrios Antonellis, Christos Bouras and Andreas Papazois, "A Multicast Packet Forwarding Mechanism for WCDMA Networks using Routing Lists" WMuNeP'06, October 6, 2006, Torremolinos, Malaga, Spain the author introduced the permanent routing list (PMRL) and Temporary Multicasting Routing List (TMRL) at each node of the network except UEs in routing mechanism for multicasting in UMTS. In the PMRL we record the nodes of the next level that the messages for every multicast group should be forwarded. The TMRL is useful for the temporary record of information from the path from the mobile users to the GGSN. These lists decrement the number of transmitted packets and use the network resources more efficiently. This will result in minor modification in the UMTS architecture.

In Gluhak, Kar Ann Chew, Klaus Moessner and Rahim Tafazolli, "Multicast Bearer Selection in heterogeneous Wireless Networks Alexander" the author proposed a bear selection method, suitable for different optimization goals and implement heuristic solutions, concentration on user preference and resource effectiveness. This paper studied the problem of selecting the optimal bearer paths for multicast services with a group of heterogeneous receivers in wireless networks with overlapping coverage. The author clearly showed that coordinated selection of multicast bearers over multiple access networks can significantly increase overall resource efficiency.

In Khaled A. Ali Hossam S. Hassanein Abd-Elhamid M. and Taha Hussein T, "Directional Cell Breathing: A Module for Congestion Control and Load Balancing in WCDMA Networks". IWCMC'06, July 3-6, 2006, Vancouver, British Columbia, Canada. Copyright 2006 ACM The author introduced the concept of Directional cell Breathing which is based on controlled Directional antenna where a base station sectorized coverage area is varied reactively in instance of congestion and proactively in the instance of exercising load balancing. This scheme will overcomes the drawbacks of non-sectorized and non-controllable WCDMA by optimizing the coverage within sector

under constraint. The coverage of the directional smart antenna can be controlled based on the traffic load of its sector. In this scheme the highly loaded sector will handoff most of its traffic to less loaded sector.

4.6 Performance of UMTS for Different Traffic Types

In Alexiou, Christos Bouras and Vaggelis Igglesis, "Performance Evaluation of TCP over UMTS Transport Channels Antonis" the author discusses the performance of UMTS for different traffic types. The performance parameters are end-to-end packet delay, delay in Radio Access Network and throughput in wireless link. The author Consider two types of transport channels in this evaluation: Dedicated Channel and High Speed downlink Shared Channel. The performance of TCP over UMTS is evaluated for two main traffic models: DCHs and HS-DSCH.AS TCP does not efficiently use the guaranteed QOS of UMTS radio channels in term of delay and throughput that will lead to congestion in the network and retransmission decrease the window size that will result the less throughput Very high delays and small throughput lead to unsatisfied mobile users. The solution presented in paper works well in mobile environment but the performance TCP is detoreated due to above mentioned reasons.

In Roberto Cusani1, Filomena Del Sorbo, Francesco Delli Priscoli, Giuseppe Lombardi and Dario Pompili "UMTS access network architecture for multimedia services" the author proposed a mechanism that allows the access network to provide guarantee QOS agreed between the user and the UMTS access network at the time of connection setup by fair allocation of network sources with property to control the congestion in the network. The author Implemented the congestion control mechanism in the RRC and subsystems both on the UE and on the RNS (Radio network Subsystems) by means of Dual Leaky Buckets (DLBs) whose input parameter could be either fixed or dynamically changing. This mechanism provide the efficiency and QOS to the user.

4.7 Radio Resource Management in UMTS

In the Universal Mobile Telecommunications System (UMTS) the radio resource management functions are jointly handled in two different protocol layers, the Radio Resource Control (RRC) and the Medium Access Control (MAC). In Joachim Sachs, Thomas Balon and Michael Meyer Wireless'99 "Congestion Control in WCDMA with Respect to Different Service Classes" ITG-Fachtagung "Mobile Kommunikation" October 6 - 8, 1999, Munich, Germany the author divide the user requirements into different classes and share the network resources on the basis of class of the user. The mechanism provides the QOS by congestion resolving ability by selecting the optimum threshold for congestion indication measure. The author discusses the various methods of detecting the congestion in the network in order to resolve the congestion and radio resource reconfiguration by RRC in UTRAN.

4.8 High Speed Down Link Packet Access Extension in WCDMA

In Sara Landstr'om, Lars-Ake Larzon and Ulf Bodin, "Congestion Control in a High Speed Radio Environment" the author proposed the mechanism how congestion is control at the transport layer and its interaction with physical layer in High Speed Down Link Packet Access Extension in WCDMA. The author makes use of TCP SACK and TFRC for controlling the congestion in network. There are a number of fundamental differences between TCP and TFRC - TCP is sender-oriented and uses a sliding window to control the send rate whereas TFRC is receiver-oriented and uses an equation based scheme. The author studied the performance of TFRC and TCP over a HSDPA link layer with both a RR and a SIR scheduler.

In X. Gelabert, J. Pérez-Romero, O. Sallent and R. Agustí, "Congestion Control Strategies in Multi-Access Networks" ©2006 IEEE the author suggest the mechanism for congestion detection and give some strategies, for solving congestion with minimum effect on QOS of user in heterogeneous network consisting of GERAN and UTRAN RATs. The author detects the congestion in UTRAN and GERAN by using their RATs differently.

4.9 Congestions Mechanisms in UTRAN

In UTRAN, overload situations may be detected by means of the load factor which can be measured, for the uplink (UL) and downlink (DL). Then the criteria will decide weather the congestion occur and we have entered in congestion situation by checking (Performance of Uplink and Downlink greater then the Congestion detection performance) during a certain percentage of frames within a period of time. And in GERAN the congestion is detected by mean of resource allocation technique in EGPRS. Whenever the total sum of resource demands by the users in a particular RAT is higher than a given amount of available resources we may encounter a congestion situation. The author discusses different mechanism to overcome Congestion.

The schemes includes Vertical Handover Congestion Resolution (VHO-CR) in this case if congestion is detected in a UTRAN cell, a successful VHO attempt of user from UTRAN to GERAN will contribute to decrease the uplink load factor. In Bit-rate Reduction Congestion Resolution (BRR-CR) the congestion is overcome by mean of user requirements. In User Dropping Congestion Resolution (DROP-CR) the congestion is overcome by selectively dropping the number of uses in the congested cell/RAT. The selection of user depends on the Service-class prioritization, User-type prioritization and Capacity-consumption prioritization.

In Jordi Perez-Romero, Oriol Sallent, Ramon Agusti and Juan Sanchez, Department de Teori Del senyal I "On managing radio network Congestion in UTRA_FDD" Communicacio- University politecnica de Catalunya the author provides the survey of several congestion control algorithms and provides some guidelines to deal with overload situation in WCDMA in context and particular in the UTRA-TDD. The author also discusses the various methods of congestion detection and congestion resolution.

Chapter # 5
Introduction to OMNet ++

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5.1. Introduction

OMNet++ stands for Objective Modular Network Testbed in C++. It is a discrete event simulation tool designed to simulate computer networks, multi-processors and other distributed systems. Its applications can be extended for modeling other systems as well. It has become a popular network simulation tool in the scientific community as well as in industry over the years.

5.2. Components of OMNET++

- Simulation kernel library
- Compiler for the NED topology description language (nedc)
- Graphical network editor for NED files (GNED)
- GUI for simulation execution, links into simulation executable (Tkenv)
- Command-line user interface for simulation execution (Cmdenv)
- Graphical output vector plotting tool (Plove)
- Utilities (random number seed generation tool, makefile creation tool, etc.)
- Documentation, sample simulations, contributed material, etc.

5.3. Platforms of OMNET++

OMNet++ works well on multiple platforms. It was first developed on Linux. OMNet++ runs on most Unix systems and Windows platforms. The best platforms used are:

- Solaris, Linux (or other Unix-like systems) with GNU tools
- Win32 and Cygwin32 (Win32 port of gcc)
- Win32 and Microsoft Visual C++

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5.4. OMNet ++ USAGE

OMNet++ is free for any non-profit use. This is also in use for commercial project. The GNU General Public License can be chosen on OMNeT++.

5.5. Simulation Modeling In OMNet ++

The following are types of modeling that can be used:

- Communication protocols
- · Computer networks and traffic modeling
- Multi-processor and distributed systems
- Administrative systems
- Any other system where the discrete event approach is suitable.

5.5.1 Library Modules

Object libraries can be made using simple modules. The best simple modules to be used for library modules are the ones that implement:

- Physical/Data-link protocols: Ethernet, Token Ring, FDDI, LAPB etc.
- Higher layer protocols: IP, TCP, X.25 L2/L3, etc.
- Network application types: E-mail, NFS, X, audio etc.
- Basic elements: message generator, sink, concentrator/simple hub, queue etc.
- Modules that implement routing algorithms in a multiprocessor or network

5.5.2 Network Modules

A model network consists of "nodes" connected by "links. The nodes representing blocks, entities, modules, etc, while the link representing channels, connections, etc. The structure of how fixed elements (i.e. nodes) in a network are interconnected together is called topology.

OMNet++ uses NED language, thus allowing for a more user friendly and accessible environment for creation and editing. It can be created with any text-processing tool (perl, awk, etc). It has a human-readable textual topology. It also uses the

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same format as that of a graphical editor. It also supports submodule testing. OMNet++ allows for the creation of a driver entity to build a network at run-time by program.

5.5.3 Organization of Network Simulation:

OMNet++ follows a hierarchical module structure allowing for different levels of organization.

• Physical Layer:

- 1. Top-level network
- 2. Subnetwork (site)
- 3. LAN
- 4. Node

• Topology within a node:

- OSI layers. The Data-Link, Network, Transport, Application layers are of greater importance.
- 2. Applications/protocols within a layer.

5.5.3.1. Network Description (NED)

Modular description of networks is given in NED language. The network description consists of a number of component descriptions such as channels, simple and compound module types. These component descriptions can be used in various network descriptions. Thus, it is possible for the user to customize his or her personal library of network descriptions.

The files containing the network descriptions should end with a .ned suffix. The NEDC compiler translates the network descriptions into C++ code. Then, it is compiled by the C++ compiler and linked into executable simulation.

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5.5.3.2 Components of a NED Description

A NED description can contain the following components, in arbitrary number or order:

- import statements
- · channel definitions
- simple and compound module declarations
- system module declarations

5.6 USER INTERFACES

The OMNet++ user interface is used with the simulation execution. OMNet++'s design allows the inside of model to be seen by the user. It also allows the user to initiate and terminate simulations, as well as change variable inside simulation models. These features are handy during the development and debugging phase of modules in a project. Graphical interface is a user friendly option in OMNet++ allows access to the internal workings of the model.

Currently, two user interfaces are supported:

- Tkenv: Tk-based graphical, windowing user interface (X-Window, Win95, WinNT etc..)
- Cmdenv: command-line user interface for batch execution

Simulation is tested and debugged under Tkenv, while the Cmdenv is used for actual simulation experiments since it supports batch execution.

5.6.1 Tkeny

Tkenv is a portable graphical windowing user interface. Tracing, debugging, and simulation execution is supported by Tkenv. It has the ability to provide a detailed picture of the state of the simulation at any point during the execution. This feature makes Tkenv a good candidate in the development stage of a simulation or for presentations. A snapshot of a Tkenv interface is shown in figure 5.1.

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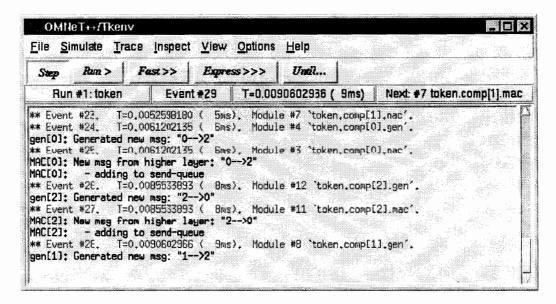


Figure 5.1: OMNeT++ Tkenv Environment

Important features in Tkenv:

- Separate window for each module's text output
- Scheduled messages can be watched in a window as simulation progresses
- Event-by-event execution
- Execution animation
- Labeled breakpoints
- Inspector windows to examine and alter objects and variables in the model
- Graphical display of simulation results during execution. Results can be displayed as histograms or time-series diagrams.
- Simulation can be restarted
- Snapshots (detailed report about the model: objects, variables etc.

5.6.2 Cmdenv

Cmdenv is designed primarily for batch execution. It is a portable and small command line interface that is fast. It compiles and runs on all platforms. Cmdenv use simply executes all simulation runs that are described in the configuration file.

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5.7 Expected Performance of OMNet ++

One of the most important factors in any simulation is the programming language. The common languages used are C/C++ based. OMNet++ performance is of a particular interest since it reduces the overhead costs associated with GUI simulation library debugging and tracing. The drawback found in OMNet++ was its simulations were 1.3 slower than it C counterpart.

5.8 NED overview

Files containing network descriptions generally have a .ned suffix. NED files can be loaded dynamically into simulation programs, or translated into C++ by the NED compiler and linked into the simulation executable.

5.8.1 Components of a NED description

A NED description can contain the following components, in arbitrary number or order:

- import directives
- · channel definitions
- simple and compound module definitions
- network definitions

5.8.2 Reserved words

The writer of the network description has to take care that no reserved words are used for names. The reserved words of the NED language are:

Import	channel	endchannel	simple	endsimple
Module	endmodule	error	delay	datarate
const	parameters	gates	submodules	connections
gatesizes	if	for	do	endfor
network	endnetwork	nocheck	ref	ancestor
true	false	like	input	numeric
string	bool	char	xml	xmldoc

5.8.3 Identifiers

Identifiers are the names of modules, channels, networks, submodules, parameters, gates, channel attributes and functions. Identifiers must be composed of letters of the English alphabet (a-z, A-Z), numbers (0-9) and the underscore "_". Identifiers may only begin with a letter or the underscore. If you want to begin an identifier with a digit, prefix the name you'd like to have with an underscore, e.g. _3Com. If you have identifiers that are composed of several words, the convention is to capitalize the beginning of every word.

5.8.4 The import directive

The import directive is used to import declarations from another network description file. After importing a network description, one can use the components (channels, simple/compound module types) defined in it. When a file is imported, only the declaration information is used. Also, importing a ned file does not cause that file to be compiled with the NED compiler when the parent file is NED compiled, i.e., one must compile and link all network description files – not only the top-level ones.

5.8.5 Channel definitions

A channel definition specifies a connection type of given characteristics. The channel name can be used later in the NED description to create connections with these parameters.

The syntax:

channel Channel Name

//...

endchannel

Three attributes can be assigned values in the body of the channel declaration, all of them are optional: delay, error and datarate.

- delay is the propagation delay in (simulated) seconds;
- error is the bit error rate that speficifies the probability that a bit is incorrectly transmitted;
- datarate is the channel bandwidth in bits/second, used for calculating transmission time of a packet.

The attributes can appear in any order. The values should be constants.

Example:

(:

```
channel LeasedLine
```

delay 0.0018 // sec

error 1e-8

datarate 128000 // bit/sec

endchannel

5.8.6 Simple module definitions

Simple modules are the basic building blocks for other (compound) modules. Simple module types are identified by names. By convention, module names begin with upper-case letters. A simple module is defined by declaring its parameters and gates. Simple modules are declared with the following syntax:

simple SimpleModuleName

parameters:

//...

gates:

//...

endsimple

ξ.

5.9 Compound Module Definitions in NED

Compound modules are modules composed of one or more submodules. Any module type (simple or compound module) can be used as a submodule. Like simple modules, compound modules can also have gates and parameters, and they can be used wherever simple modules can be used. It is useful to think about compound modules as "cardboard boxes" that help you organize your simulation model and bring structure into it. No active behavior is associated with compound modules — they are simply for grouping modules into larger components that can be used either as a model or as a building block for other compound modules.

By convention, module type names (and so compound module type names, too) begin with upper-case letters.

Submodules may use parameters of the compound module. They may be connected with each other and/or with the compound module itself. A compound module definition looks similar to a simple module definition: it has gates and parameters sections. There are two additional sections, submodules and connections.

The syntax for compound modules is the following:

module CompoundModule

parameters:

//...

gates:

//...

submodules:

//...

connections:

//...

endmodule

All sections (parameters, gates, submodules, connections) are optional.

5.10 Connections

The compound module definition specifies how the gates of the compound module and its immediate submodules are connected. You can connect two submodules or a submodule with its enclosing compound module. This means that NED does not permit connections that span multiple levels of hierarchy – this restriction enforces compound modules to be self-contained, and thus promotes reusability. Gate directions must also be observed, that is, you cannot connect two output gates or two input gates. Only one-to-one connections are supported, so a particular gate may only be used occur in one connection. One-to-many and many-to-one connections can be achieved using simple modules that duplicate messages or merge message flows.

```
Example:

module CompoundModule

parameters: //...

gates: //...

submodules: //...

connections:

node1.output --> node2.input;

node1.input <-- node2.output;

//...

endmodule
```

The source gate can be an output gate of a submodule or an input gate of the compound module, and the destination gate can be an input gate of a submodule or an output gate of the compound module. The arrow can point either left-to-right or right-to-left. The gate++ notation allows you to extend a gate vector with new gates, without having to declare the vector size in advance with gate sizes. This feature is very convenient for connecting nodes of a network:

```
simple Node
gates:
in: in[];
```

```
endsimple
module SmallNet
      submodules:
      node: Node[6];
       connections:
      node[0].out++ --> node[1].in++;
      node[0].in++ <-- node[1].out++;
      node[1].out++ --> node[2].in++;
      node[1].in++ <-- node[2].out++;
      node[1].out++ --> node[4].in++;
      node[1].in++ <-- node[4].out++;
       node[3].out++ --> node[4].in++;
       node[3].in++ <-- node[4].out++;
       node[4].out++ --> node[5].in++;
       node[4].in++ <-- node[5].out++;
endmodule
```

5.11 Network definitions

Module declarations (compound and simple module declarations) just define module types. To actually get a simulation model that can be run, you need to write a network definition. A network definition declares a simulation model as an instance of a previously defined module type. You'll typically want to use a compound module type here, although it is also possible to program a model as a self-contained simple module and instantiate it as a "network". There can be several network definitions in your NED file or NED files.

The syntax of a network definition is similar to that of a submodule declaration:

```
network wirelessLAN: WirelessLAN
parameters:
    numUsers=10,
```

```
httpTraffic=true,

ftpTraffic=true,

distanceFromHub=truncnormal(100,60);
```

endnetwork

5.12 Large networks

There are situations when using hand-written NED files to describe network topology is inconvenient, for example when the topology information comes from an external source like a network management program.

In such case, you have two possibilities:

- 1. Generating NED files from data files
- 2. Building the network from C++ code

The two solutions have different advantages and disadvantages. The first is more useful in the model development phase, while the second one is better for writing larger scale, more productized simulation programs. In the next sections we examine both methods.

5.13 Generating NED files

Text processing programs like awk or perl are excellent tools to read in textual data files and generate NED files from them. Perl also has extensions to access SQL databases, so it can also be used if the network topology is stored in a database.

5.14 Building the network from C++ code

Another alternative is to write C++ code which becomes part of the simulation executable. The code would read the topology data from data files or a database, and build the network directly, using dynamic module creation. The code which you need to write would be similar to the *_n.cc files output by Nedtool.

Chapter # 6
Implementation and Results

6.1. Problem Identification and Scope of Project

In all the literature reviewed, no body had discussed the congestion control over UMTS under multicasting scheme. The main objective is to design a method for detecting the congestion and resolve it in UMTS under multicasting scheme using the routing list scheme. There are two problems to be faced while detecting congestion in wireless networks that are whether the losses are due to bit error rate BER or due to congestion at the receiver side. Wireless links are not as robust as wired -line links, since the radio quality may vary considerably over time, the bandwidth is usually lower, and transmission errors occur more frequently [15].

The TCP performs lower in wireless networks than in wired because of the fact that problems occurring in wireless networks from congestion cannot be distinguished by TCP. Also, the TCP is not able to distinguish between the losses that whether they are due to BER or congestion. It is assumed that the losses are due to Congestion and then TCP decreases its window size affecting the decreased throughput.

Many techniques are used to improve the TCP performance over wireless i.e. link layer, split connection, explicit notification, and end-to-end link layer optimizations for improved TCP performance. In order to distinguish congestion related losses from BER losses in the wireless network explicit notifications can be used between an intermediate node and the end hosts.

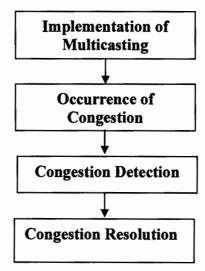


Figure 6.1 Different Steps for Congestion Detection and Congestions Resolution

6.2 Proposed Methodology for Congestion Control

In problem multicast congestion control in UMTS, firstly it must be distinguished between the losses due to the congestion and losses due to BER in the transmission link between the Sender and the receiver. Several technique have been developed to over come the multicast congestion in wired networks that are broadly categorized as Single Rate base congestion control and Multi rate congestion control. In figure 6.2, different multicast congestion control mechanisms can be viewed in hierarchical structure.

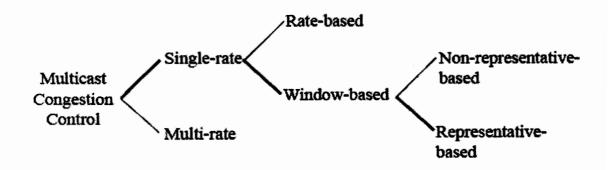


Figure 6.2: Hierarchy of Different Types of Multicast

Congestion Control Mechanisms

In Single rate congestion control where all the receivers receive data at the same time and at same rate that is the rate of slowest receiver and multi-rate congestion control, where different receivers receives data at different rates. Though the multi-rate congestion control is more scalable but it has other concerns such as the complex encoding of data, possible multiple paths in the layered approach, and the effects of receivers joining and leaving layers.

In the given solution to Multicast congestion control in UMTS, window base single rate congestion control is used with explicit notification for the losses due to BER. Routing list mechanism is used for the multicasting in the UMTS. For this mechanism, it is attempted to build the temporary and permanent routing list in all nodes of the UMTS network except the user node as shown in figure 6.3 given below.

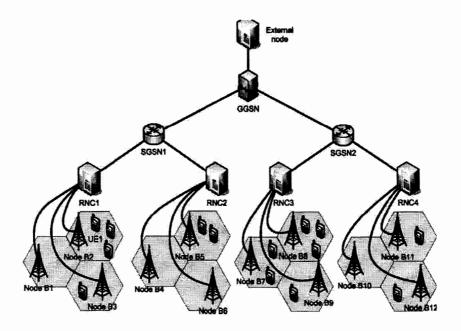


Figure 6.3: Temporary Routing structure in UMTS Network

The multicast receivers are lies in the node B and radio resources are controlled by the RNC. In solution, firstly a mechanism is designed for selecting the slowest rate user on which the sender transmits data to the receiver. The slowest receiver is selected on the basis of its throughput and directly communicates with the Sender this is the ACKER Receiver if at some other time the through put of the other receivers are less then the ACKER receiver then receiver will become the ACKER receiver. The selection of ACKER receiver is show in the figure 6.4 [29].

To overcome the losses due to BER in the transmission line the explicit notification from the receiver is used. The sender made the decision whether the losses are due to congestion or due to BER on the basis of explicit notification. The receiver sends the NACK to the Sender and the Acker sends ACK to the sender. If the receiver NACK window size is less then the previous window size then the sender assume the losses are due to congestion at receiver side and it will result in the selection of new ACKER by the sender and decrease its window size as to size of the ACKER receiver window and if the size of NACK is not less then the previous window size then the

sender assume the loss are due to the BER and does not decrease its window size without sending the lost packets.

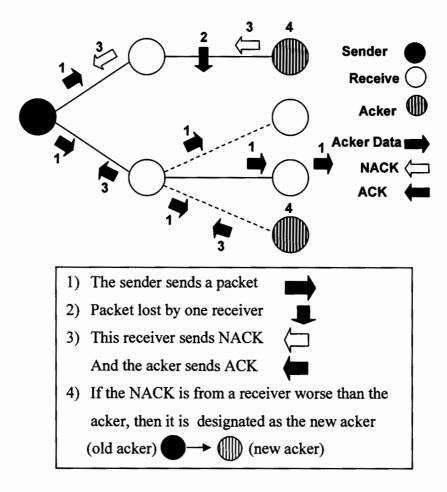


Figure 6.4: Acker Selection Mechanisms

In order to test above mentioned mechanism it is attempted to made many performance measures using OMNeT ++ Simulator. Some of the measurements which are being used are throughput for various random BER and congestion control Metrics.

6.3 Designing the Simulation

6.3.1 Basic Files to design Simulation

While working in OMNeT++, different files are required to designed with prefixes like .cpp and .ned etc. therefore, there are different files designed in Visual C++ and GNED for different components of UMTS network as shown in figure 6.5.

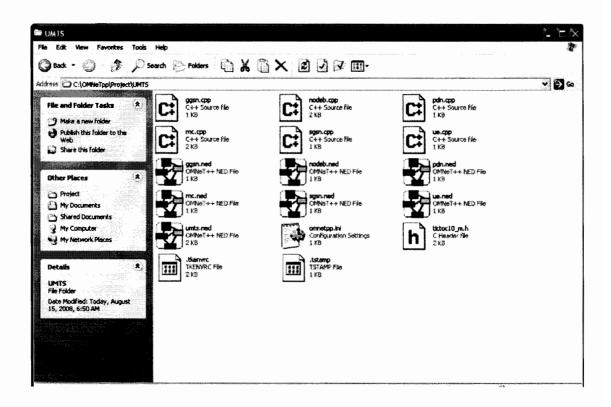


Figure 6.5: A view of different Types of Files designed for Simulation

6.3.2 Different Views of UMTS.ned, Representing UMTS Core Network

6.3.2.1 NED Source View

While designing a simulation in OMNeT++, different Simple Modules are to be designed for representing the functionality to be performed by different nodes which are part of UMTS network. Apart from the Simple Modules, there will be Compound Module that combines all simple Modules on the one platform to function as an independent node of UMTS network. Figure 6.6 shows GNED view of compound

Module *UMTS.ned* which combines all the simple modules and create connections in between them.

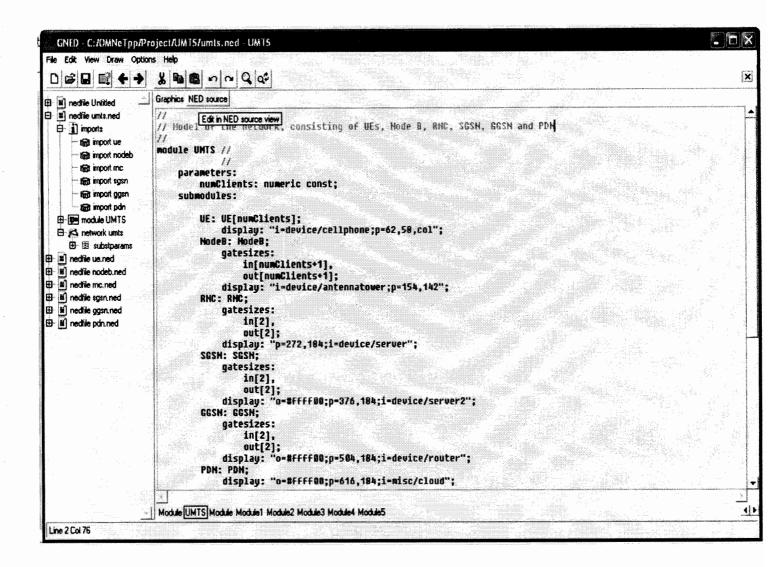


Figure 6.6: GNED Source view of UMTS.ned

6.3.2.2 NED Script for UMTS.ned

A script written in OMNeT++ GNED for *UMTS.ned* is presented as below. It is cleared that all the definitions made against existing nodes of UMTS network and configurations of their connections established with each other.

```
// Model of the network, consisting of UEs, Node B, RNC, SGSN, GGSN and PDN
module UMTS
parameters:
    numClients: numeric const;
submodules:
    UE: UE[numClients];
       display: "i=device/cellphone;p=62,58,col";
    NodeB: NodeB;
       gatesizes:
         in[numClients+1],
         out[numClients+1];
       display: "i=device/antennatower;p=154,142";
    RNC: RNC;
       gatesizes:
         in[2],
         out[2];
       display: "p=272,184;i=device/server";
     SGSN: SGSN;
       gatesizes:
         in[2],
         out[2];
       display: "o=\#ffff00; p=376,184; i=device/server2";
     GGSN: GGSN;
       gatesizes:
         in[2],
         out[2];
       display: "o=\#ffff00; p=504, 184; i=device/router";
     PDN: PDN;
       display: "o=#ffff00;p=616,184;i=misc/cloud";
  connections:
```

```
PDN.out --> GGSN.in[0];
PDN.in <-- GGSN.out[0];

GGSN.out[1] --> SGSN.in[0];
GGSN.in[1] <-- SGSN.out[0];

SGSN.out[1] --> RNC.in[0];
SGSN.in[1] <-- RNC.out[0];

RNC.out[1] --> NodeB.in[0];
RNC.in[1] <-- NodeB.out[0];

for i=0..numClients-1 do
    UE[i].out --> NodeB.in[i+1];
    UE[i].in <-- NodeB.out[i+1];
endfor;

display: "b=697,417";
endmodule</pre>
```

6.3.2.3 Visual C++ Code for UMTS Networks

OMNeT ++ GNED describes only the general structure of different nodes of UMTS network. The actual functional to be preformed by the different nodes of the network has been described inside the Visual C++ files. So, here is the code for different Nodes of the UMTS network representing their functionality.

6.3.2.3.1 Visual C++ Code for Node B

```
//#include <omnetpp.h>
//#include <netpkt m.h>
#include <stdio.h>
#include <string.h>
#include <omnetpp.h>
#include "tictoc10_m.h"
class NodeB: public cSimpleModule
 private:
  double propDelay;
protected:
virtual void initialize();
void handleMessage(cMessage *msg);
virtual void finish();
};
Define_Module(NodeB);
void NodeB::initialize()
{
ev<<"This is Initilization of"<<name()<<endl;
propDelay = (double)par("propDelay");
cMessage *msg = new cMessage("Data Packet");
if (index()==0)
scheduleAt(0.0, msg);
```

```
}
}
void NodeB::handleMessage(cMessage *msg)
{
ev<<"Packet Arrived at "<<name()<<"from"<<msg->getSource()<<"\n";
ev<<"Message Length"<<msg->length()<<"\n";
ev<<"Message Kind"<<msg->kind()<<"\n";
ev<<"Buffer remains"<<msg->buffer();
msg->setSource(2);
msg->setLength(8*24);
msg->setKind(1);
msg->setBuffer(200);
int k = intuniform(0,3);
ev<<"\n\n New Packet being sent to"<< k <<"from "<<name();
if(k==0)
bubble("Going to RNC");
else
bubble("Going to UE");
sendDelayed(msg, propDelay, "out", k);
}
void NodeB::finish()
{
  // This function is called by OMNeT++ at the end of the simulation.
  ev << "Sent:
                " << numSent << endl;
  ev << "Received: " << numReceived << endl;
  ev << "Hop count, min: " << hopCountStats.min() << endl;
  ev << "Hop count, max: " << hopCountStats.max() << endl;
  ev << "Hop count, mean: " << hopCountStats.mean() << endl;
  ev << "Hop count, stddev: " << hopCountStats.stddev() << endl;
```

```
recordScalar("#sent", numSent);
  recordScalar("#received", numReceived);
  hopCountStats.recordScalar("hop count");
}
6.3.2.3.2 Visual C++ Code for UE
#include <omnetpp.h>
class UE: public cSimpleModule
{
       protected:
       virtual void initialize();
       virtual void handleMessage(cMessage *msg);
};
Define Module(UE);
void UE::initialize()
       ev<<"This is Initilize of"<<name()<<"\n";
{
       cMessage *msg = new cMessage("Data Packet");
       msg->setSource(1);
       msg->setBuffer(200);
       bubble("Going to NodeB");
       send(msg,"out");
}
void UE::handleMessage(cMessage *msg)
{
       msg->setSource(1);
       msg->setBuffer(intuniform(0,200));
       bubble("Going to NodeB" );
       ev<<index<<"\t"<<msg->buffer();
       send(msg,"out");
}
```

6.3.2.4 GRAPHICS View of UMTS.ned

UMTS.ned can be viewed in its Graphics Mode in OMNeT++ GNET as in figure 6.7.

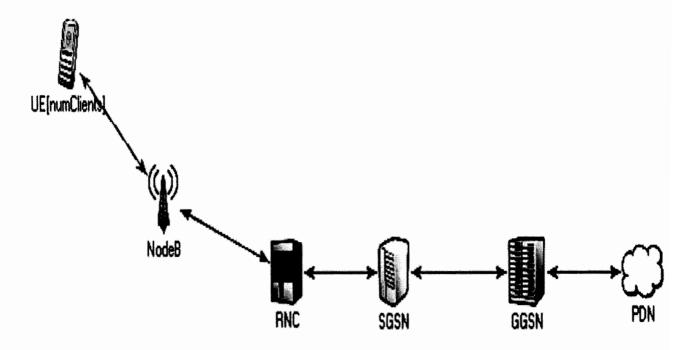


Figure 6.7: Graphics View of UMTS.ned

6.4 Compiling the Simulation

After the complete design of different nodes of UMTS network in OMNeT++ GNED and their code in Visual C ++, Here are the different steps to Run the simulation designed. For the this purpose, Command Line Interface is used where different kind of commands are applied which can be seen in different snapshots given in next sections below.

6.5 Different Steps for Running the Simulation

6.5.1 Creating the MakeFile.vc



Figure 6.8: Command for creating the Makefile.vc

After the successful creation of file as "Makefile.vc", the command prompt will be view like this described in figure given below.

```
C:\WINDOWS\system32\cmd.exe
C:\OMNeTpp\Project\UMTS>opp_nnakenake -f
Creating Makefile.vc in C:\OMNeTpp\Project\UMTS...
Makefile.vc created.
Please type 'nmake -f Makefile.vc depend' NOV to add depend
C:\OMNeTpp\Project\UMTS>_
```

Figure 6.8: A view of Command Prompt after successful creation of Makefile.vc

In next step, the intension is to compile the makefile.vc to generate the executable file. For this purpose, the command applied in Command Prompt can be viewed the figure 6.9 given below.

```
C:\OMNeTpp\Project\UMIS>opp_nmakemake -f
Creating Makefile.vc in C:/OMNeTpp/Project/UMIS...
Makefile.vc created.
Please type 'nmake -f Makefile.vc depend' NOV to add depende
C:\OMNeTpp\Project\UMIS>nmake -f makefile.vc
```

Figure 6.9: A Command to compile the Makefile.vc

6.5.2. Number of Files generated after compiling the Simulation

While compiling the makefile.vc, which is a collection of all files with prefixes as .ned and .cpp etc, different kinds of files will be generated in the simulation folder. During this process, all the files designed in OMNeT++ GNED are translated into .cpp files and .obj files. Some other supportive files to the executable file will also be generated. Following figures 6.10(a) and 6.10(b) are showing the details of generating different files during compilation process.



Figure 6.10(a): Process of Compiling the Makefile.vc

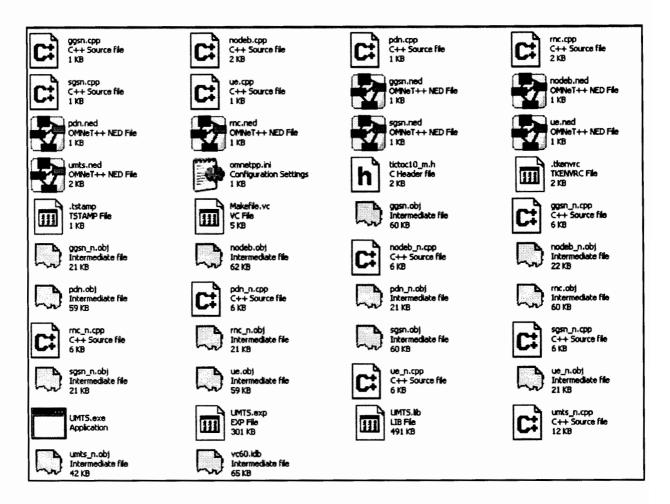


Figure 6.10(b): A view off different additional files generated after compilation

In the above figure, it is observed that different supportive files have been created during the compilation process. Here, the most important file is UMTS.exe which is main source to run the simulation. Apart from the UMTS.exe, omnetpp.ini is also very important file in this simulation folder. Omnetpp.ini contains the basic configurations off the simulation including parameter's values, different RUNS and other very important sections of different sort of settings for simulation.

6.6 Running the Simulation

After the successful creation of .exe file, .exe file is executed. When it is run the .exe file, OMNeT++ Tkenv GUI will be displayed along with other addition supportive windows as a platform to run the simulation successfully. The different views can be seen here in figure 6.11.

6.6.1 Simulation's View in Executable Form

The following figure 6.11(a) displaying the whole image of Simulation design in running form before playing it.

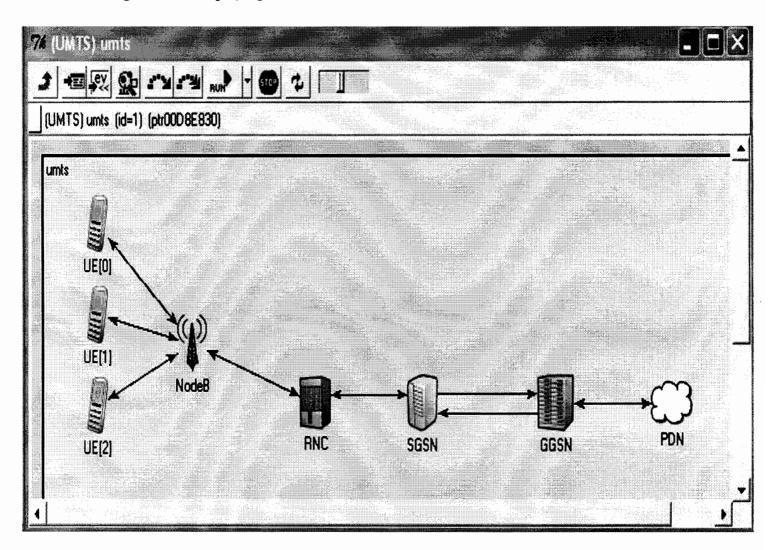


Figure 6.11(a): OMNeT++ Tkenv Graphical User Interface

```
Loading NED file: C:\OMNeTpp\Project\UMTS1\nodeb.ned
Warning: replacing existing declaration for simple nodule type
Warning: replacing existing declaration for compound nodule type
Loading NED file: C:\OMNeTpp\Project\UMTS1\pdn.ned
Warning: replacing existing declaration for simple nodule type
Warning: replacing existing declaration for compound nodule type
Loading NED file: C:\OMNeTpp\Project\UMTS1\rnc.ned
Warning: replacing existing declaration for simple nodule type
Warning: replacing existing declaration for compound nodule type
Loading NED file: C:\OMNeTpp\Project\UMTS1\sgsn.ned
Warning: replacing existing declaration for simple nodule type
Varning: replacing existing declaration for compound nodule type
Loading NED file: C:\OMNeTpp\Project\UMTS1\unened
Warning: replacing existing declaration for simple module type
Loading NED file: C:\OMNeTpp\Project\UMTS1\unened
Warning: replacing existing declaration for compound nodule type
Loading NED file: C:\OMNeTpp\Project\UMTS1\unts.ned
Warning: replacing existing declaration for compound nodule type
Loading NED file: C:\OMNeTpp\Project\UMTS1\unts.ned
Warning: replacing existing declaration for compound nodule type
Loading bitmaps from .\bitmaps: *: 0
Loading bitmaps from C:\OMNeTpp\bitmaps: *: 0
Loading bitmaps from C:\OMNeTpp\bitmaps: *: 0
Loading bitmaps from C:\OMNeTpp\bitmaps: *: 11
Plugin path: ./plugins
```

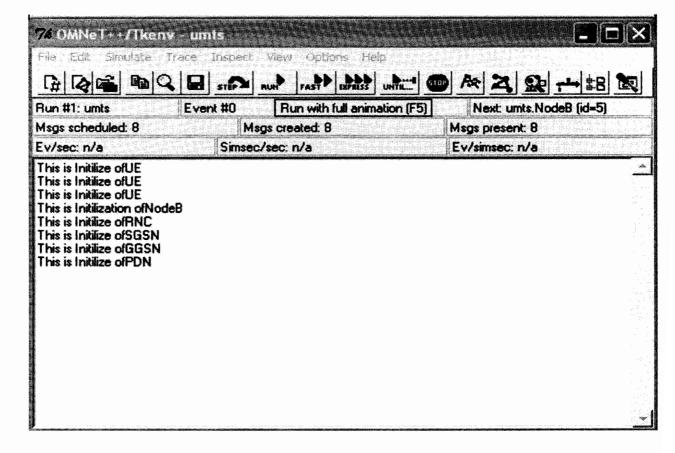


Figure 6.11(b): OMNeT++ Tkenv Graphical User Interfaces

In the above figure 6.11(b), the upper window representing the command prompt for loading all the necessary supportive files to the simulation's .exe file. The lower window is representing OMNeT++ Tkenv, which is major platform to show different statistics of simulation being run.

6.6.2 Simulation in Progress

Here is the snapshot of simulation while running, different data packets are being transferred in between UE nodes, Node B and other nodes of UMTS Core network. In our project scope, more concentration has been made on the communication activities on NodeB especially. Because, the main activity regarding Multimedia Broadcasting / Multimedia Services (MBMS) and all sort of other group management related activities are done on Node B in UMTS network.

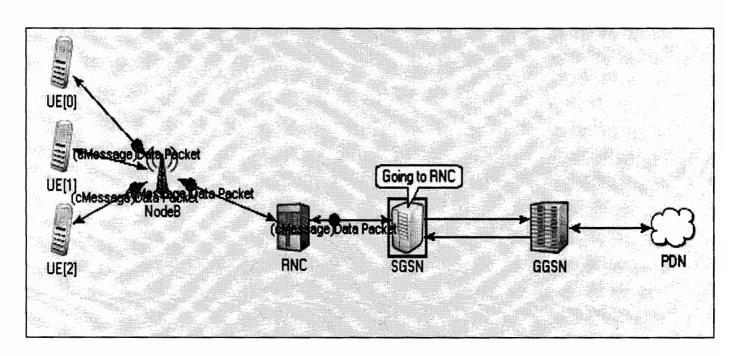


Figure 6.12(a): Running Simulation

Different statistics on the whole simulation will be displayed on Tkenv GUI parallel to the simulation running as shown in figure 6.12(b)

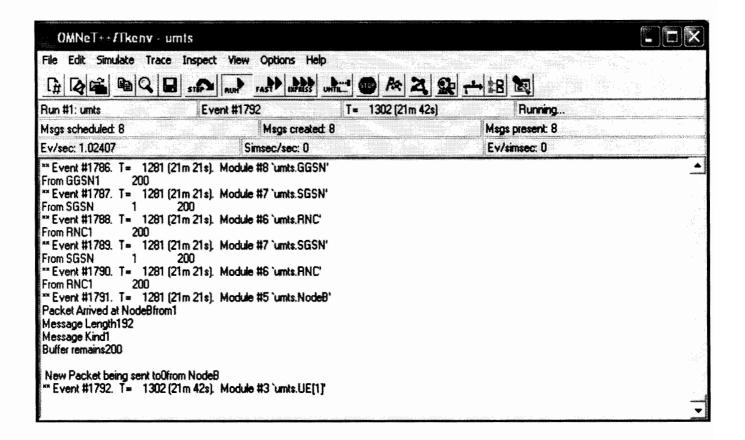


Figure 6.12(b): Running Simulation

6.7 Results and Analysis

After the successful definition to resolve the issues being faced to UMTS network regarding congestion, few of the performance related parameters of the UMTS network are analyzed. Basically, the main purpose of the results to be analyzed is to see the different effects of this congestion resolution on the network. There are further analysis of UMTS network are expected in this regard in near future depending upon our proposed methodology to resolve congestion in very suitable high speed data networks of future.

6.7.1 End – to – End Delays

Delay is the moment when the application passed the packet to the transport protocols and on the other side when it is received by the application. Delay is of great importance to some applications. For example, voice over IP application cannot tolerate one-way delay greater than 250 ms.

We have calculated Delay using the formula as

$$Delay = Tr - Ts$$

Where Ts is equal to the time stamp of packet sending and Tr is time stamp of packet receiving. We have taken mean delay for our results and calculated with this formula

$$MeanDelay = \frac{TotalDelay}{N}$$

Where N is the total number of packet send in a simulation time.

6.7.1.1 End -to-End Delays on Node B

As in UMTS network, Node B is very important node that transmits data to the User Equipment(UE). In simulation discussed above, we extracted the information about end –to-end delays occurred at Node B which are being shown in figure

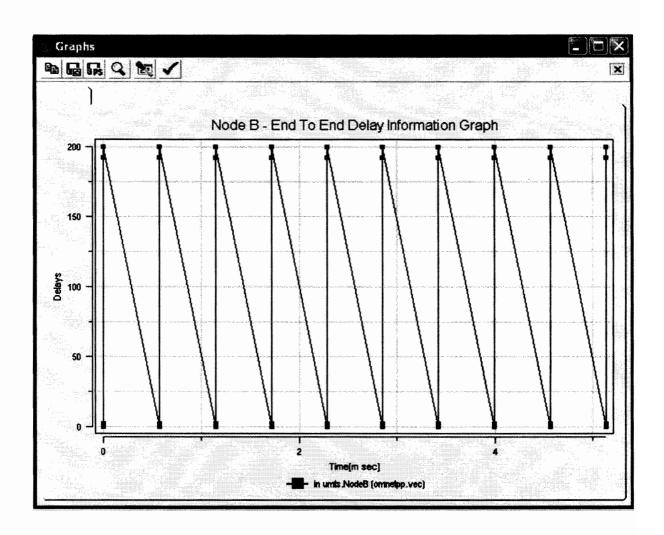


Figure 6.13: End – to –End Delays at Node B

As NodeB is constantly transmitting data packets to User Equipments and other nodes of UMTS network, so its delays related behavior is looking as periodic patterns. Delays are representing peaks and downs with respect to transmission session initial stage and its termination respectively. We can see that when transmission going to be start and delay ratio is on the peak while in case of packet sending session completed the expected

delay is its lowest point and these patterns are going to bee repeated for next transmission sessions with other nodes.

6.7.1.2. End to End Delay on User Equipments

Being the counterpart of NodeB, User Equipment (UE) observed some sort of End – to- End Delays as shown in figure 6.14.

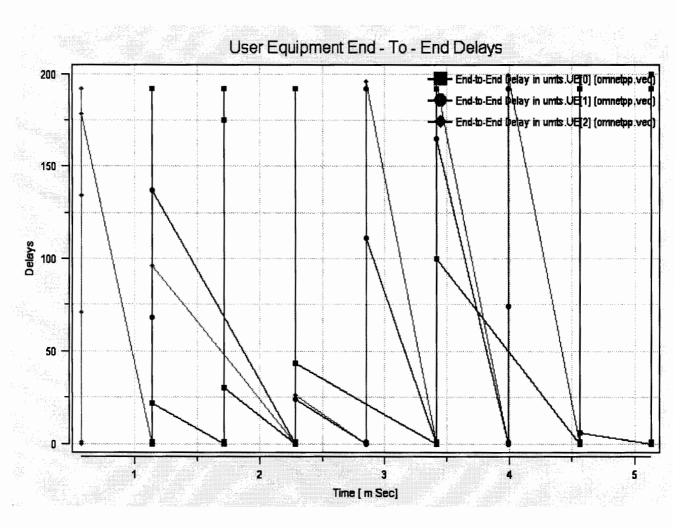


Figure 6.14: End – to – End Delays at User Equipments

There are very random delays as being displayed in the above shown figure. Continuity and discontinuity in the linearly displayed delays depends upon the session of transmission started between specific UE and NodeB. NodeB select UEs randomly and

based upon the messages created for UEs. Whenever sessions starts and goes to end, delays information is recorded in to vector files.

In this section, the figure 6.15 is comparing the End-to-End Delays at UE's and NodeB both. We can see the response of Node B in UMTS network under multicasting, a bit different from User Equipment's response.

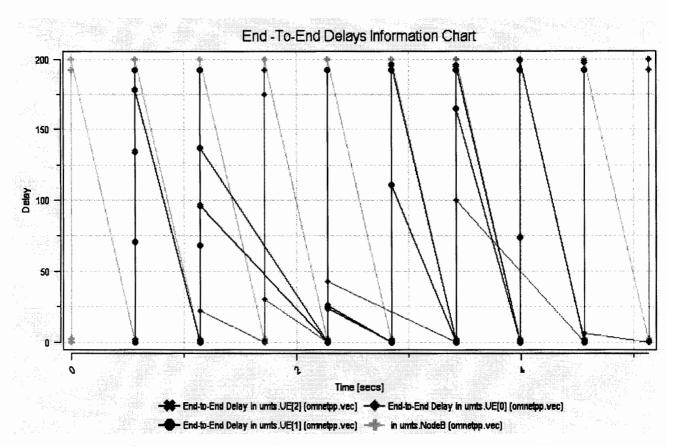


Figure 6.15: A comparison of End -to-End Delays at UE's and Node B

6.7.2 Throughput

This is a measure of the data transfer rate through a complex communication or networking scheme. Throughput is considered as an indication of the overall performance of the system. In communication throughput id usually measured as the number of bits or packets processed each second. For the purpose of this work, we used number of packets

$$Throughput = \frac{Packet(Mb)}{Time(s)}$$

The following Figure 6.16 is representing the Throughput analyzed at UE's.

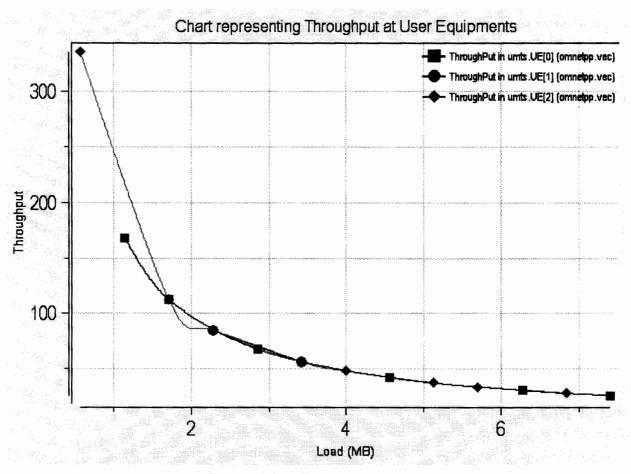


Figure 6.16: Throughput at UE's

In accordance to the above results, Throughput is totally based upon the offered load on the network. Throughput will be the maximum at minimum load offered while it gets sharp decrease with the rapid increase in load offered. The offered load is affecting on throughput due to limited buffer size of ending nodes as UEs and also overhead of processing of high speed data load offered.

6.7.3 Bit Error Rate at Node B

Loss is defined as the percentage of transmitted packets that never reach the intended destination due to deliberately discarded packets. In this section, Packet Loses due to Bit Error Rate has been analyzed in figure 6.17.

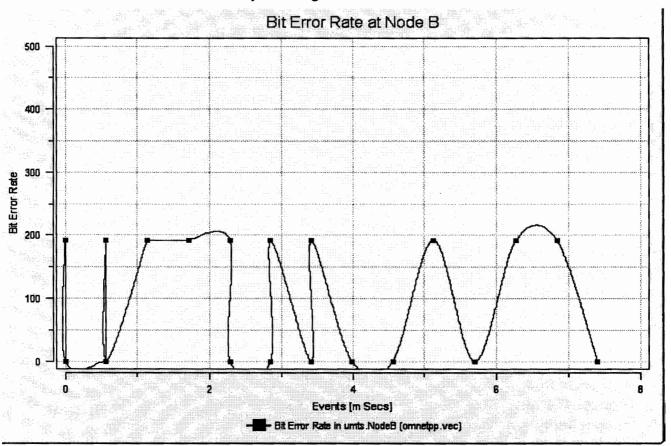


Figure 6.17: Bit Error Rate at Node B

Results shown in the above figure 6.17, Random behavior of NodeB with respect to Bit Error Rate in terms of only bits on ongoing simulation events. Normally, UMTS networks are very much robust in this sense but still there is effect of randomness of wireless channels which are naturally variable and we can't be strictly efficiently.

As it is discussed in scope of project earlier, packet losses can be due to two reasons i.e. due to congestion or due to BER. The congestion control mechanism is defined to avoid packet losses but on the other hand BER still can be reason of packet losses in transmission but that is on very minor level.

Chapter # 7

Conclusion and Future Works

7.1 Conclusions

In overall effort, this is tried to simulate the UMTS network in OMNeT++ from very scratch and put all possible efforts to analyze very high speed data rate network regarding congestion control under multicasting. It is practiced to cover different aspects of functionality of UMTS network regarding congestion. From this work, it is expected that it can be a very strong and reliable platform to work on UMTS in near future. The major problem was being faced as no extra information available to support this work related to UMTS networks in OMNeT++.

As it is explored that no extra efforts has been made to simulate perfect module for UMTS network in OMNeT++ until now. So, here are lot of encouragements in this work to any one who is intended to work in UMTS network analysis in different prospective in near future. Robustness and reliability features are the main targets to achieve to this scale of analytical based work.

UMTS is rapidly getting advancements in providing significant data capacity, performance and featured functionality benefits which have been described improvements in different Releases laid down by 3GPP recently. Apart from these advancements, UMTS also going ahead in deployment progress and the availability of terminals, applications and services which have been discussed and set goals according to suggested improvements by 3GPP.

In this prospective while UMTS is aimed to achieve lot of milestones still to come, it is very important to cover most important areas of its core functionality related to data services and bringing the most efficient mechanisms to handle large scale data loads as demanded by the users day by day. As it is hoped to discover new problem domains for achieving the goal of robust communication between two communicating nodes. After a comprehensive phase of analysis, it is quite expected to get proper solutions which would be the most feasible structures to the existing systems.

7.2 Future Works

As the work goes ahead, more and more prospective will come into consideration and better efforts will be put up to resolve all sort of complexities regarding proper analysis of work around UMTS role in high speed data network in near future that would the essential of part of our communicating world.

The 3GPP also struggling hard to set different goals day by day and defining the working standards to deliver the undeniable advantages of the GSM family of technologies like GSM, GPRS, EDGE and UMTS/HSDPA to near to a quarter part of the whole population of this digital world today.

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