Performance Evaluation of IP Based Multimedia Services in UMTS

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This paper presents our work in the performance evaluation of UMTS network based on simulation. Enhanced UMTS Radio Access Network Extensions for NS-2 (EURANE) developed by SEACORN has brought us to the higher phase of UMTS simulation in third generation wireless telecommunication system. Wireless 3G is designed to be able to deliver various kind of multimedia package through an IP network for the purpose of easier interconnection with fixed network with various existing multimedia services. Multimedia services with their bandwidth consumption characteristics are able to be sent through a UMTS network with the existence of High Speed Data Packet Access (HSPDA) in Release 5. Quality of Service (QoS) is a major concern in multimedia services. This paper shows the performance analysis of a number of multimedia services and their QoS using HSDPA in UMTS. The experiments were based on EURANE extension for NS-2. From the simulation conducted, we found that Unacknowledged Mode (UM) in Radio Link Control (RLC) will perform better for QoS class number 1 (VoIP) and 2 (Video Streaming), while Acknowledged Mode (AM) mode are more suitable for QoS class number 3 (web server) and 4 (FTP).

***Keywords**: UMTS, 3G, HSDPA, Multimedia Services, QoS, EURANE, NS-2*

INTRODUCTION

Wireless technologies have gone through a speedy growth, supporting higher data rate as well as more service diversification. One of the emerging wireless technologies is the development of Universal Mobile Telecommunication System (UMTS) which is a part of the third generation (3G) mobile network technology. The development of UMTS, so called Enhanced UMTS (E-UMTS) is based on the basic idea of 3G which is to prepare a universal infrastructure which can deliver current and future services.

Wireless 3G network is intended to transmit multimedia traffic such as voice, combination of voice and data, image transmission, web request, email, and other applications. Wireless 3G network is also designed to increase the transfer data rate according to each application standard. Third Generation standardization in Europe, called 3G Partnership Project (3GPP) which is based on UMTS, has launched High Speed Data Packet Access (HSDPA) technology. HSDPA is a starting point for E-UMTS network that can reach downlink data rate up to 14.4 Mbps and uplink data up to 384 kbps.

In order to support wide traffic scope and satisfy high speed requirement, a new network must met Quality of Service (QoS). To achieve a satisfying QoS results, investigating the process of UMTS end-to-end performance in a form of network simulation is a required in implementing the network layer and data link layer functions. The simulation can evaluate parameters associated with E-UMTS network performance; delay, packet loss, and throughput in accessing new service, protocol, and architecture. Our experiment will be based on different combination model of traffic request available in urban user environment and many kinds of service accessed by user. There are three main node functions that build radio access network in Enhanced UMTS, i.e. User Equipment (UE), Base Station (BS), and Radio Network Controller (RNC).

This simulation is built using Network Simulator application (NS-2) version 2.30 and additional patch for Enhanced UMTS Radio Access Network Extensions for NS-2 (EURANE) version 1.60 developed by IST-

SEACORN (Simulation of Enhanced UMTS Access and Core Network). NS-2.30 runs on Fedora Core 6 Kernel 2.6. User environment are represented in matrices generated by MatLab 7.1 application. Final output result from NS-2 are visualized in graph and table. The experiment result analysis is presented.

UMTS THEORY

Universal Mobile Telecommunication System (UMTS) is one of the third generation system developed in Europe. UMTS standardized by European Telecommunication Standard Institution (ETSI) while International Telecommunications Union Telecommunication Standardization Sector (ITU-T) standardized similar system named International Mobile Telecommunication System 2000 (IMT 2000). Both standardization association cooperate to develop one integrated system in the future.

UMTS is designed to provide 2 Mbits/s bandwidth. UMTS service should fulfill the user demand in any places, which means UMTS is expected to serve such a wide area. In the case where no UMTS cell exists in an area, it can be routed using satellite.

UMTS can be used for office, home, and vehicles. The same services can be provided for indoor and outdoor areas users, public and private, as well as urban and rural areas.

Radio frequencies allocated for UMTS are 1885-2025 MHz and 2110-2200 MHz. These bands will be used for small cell (pico cell) that provides a large capacity for UMTS.

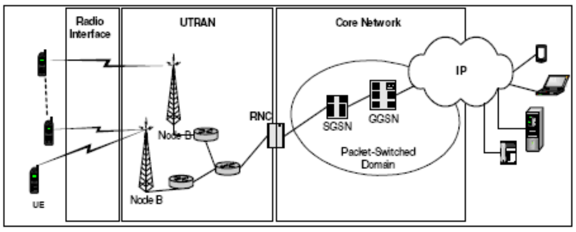


Fig.1. UMTS Network Topology [1]

Figure 1, shows several main components in UMTS network. UMTS mobile terminal (UE) has the ability to communicate with network and allowed to use UMTS service. UMTS Radio Access Network (UTRAN) handles all radio related functionality e.g. handover and relationship management. Core Network (CN) is responsible for switching and data transportation while the function related to terminal movement is implemented in Intelligent Network (IN).

There are 4 QoS classes [1] defined by UMTS. Table 1 shows the defined parameters for each class ordered from highest class to the lowest:

Table 1. UMTS Definition QoS Classes

	Conversational (VoIP)	Streaming (Video)	Interactive (Video Server)	Background (FTP)
Delay	minimum fixed	minimum variable	Moderate variable	large variable
Buffering	no	allowed	Allowed	allowed
Traffic	symmetric	asymmetric	Asymmetric	asymmetric
Bit rate	guaranteed	Guaranteed	No guaranteed	no guaranteed
Example	voice	music & video streaming	Web browsing	email

HSDPA

High Speed Downlink Packet Access (HSDPA) in UMTS enables users in UMTS network to receive multimedia data with a high bit rate. The basic characteristics of HSDPA are [2]:

• Use shared-channel transmission (in time domain)

- o Total downlink radio resources
- o Channelization code
- o Transmission power
- Enable possibility to rapidly allocate a large amount of downlink resources when required
- High-speed downlink shared channel (HS-DSCH)
- Adaptive Modulation and Coding (AMC)
- Hybrid ARQ
- Fast cell selection

Enhanced UMTS

E-UMTS network is an all-IP based network which support additional and modification for UMTS network. This modification purpose is to satisfy the requirements for capacity improvement in network access, flexibility on core network, and support integrated supplementary services, which has not unavailable yet in UMTS standard. In the other hand, E-UMTS is an evolutionary step which enables an effective end-to-end packet based transmission.

EURANE

Enhanced UMTS Radio Access Netwok Extensions for NS-2 (EURANE) was developed by the European Commission 5th framework through its project named Simulation of Enhanced UMTS access and Core Network (IST-SEACORN) [1, 3]. EURANE consists of 3 additional nodes which are Radio Network Controller (RNC), Base station (BS) dan User Equipment (UE). Their function can support transport channels such as FACH, RACH, DCH, dan HS-DSCH.

SIMULATION DESIGN

In this paper we simulated 12 users in 3 different environments accessing 4 multimedia data services in an Enhanced-UMTS. This simulation is conducted using NS-2 version 2.30 and additional patch for EURANE version 1.60 which was developed by IST-SEACORN [1, 3]. The simulation architecture of E-UMTS can be seen on Figure 2.

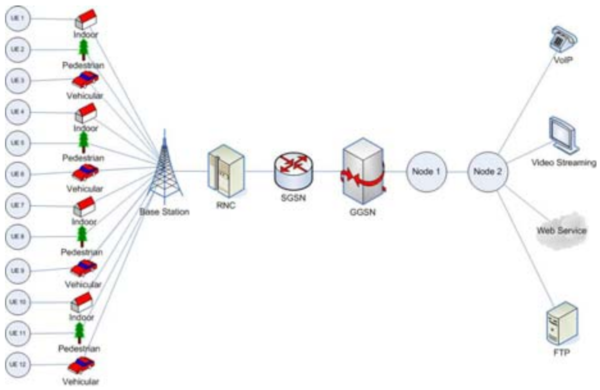


Fig.2. Simulation architecture

Simulation Environment

There are 3 environments used in this simulation:

1. Indoor environment [4]

In this environment both users and base station are located indoor. The main characteristics of this environment are small cells and low transmission power. The path loss rule varied due to scattering and attenuation by walls, floors, and metallic structures such as partitions and filing cabinets. These objects also produce shadowing effects.

2. Pedestrian environment [4]

In this environment, pedestrian users walk slowly while base station with low antenna heights is located outdoors. Similar to indoor environment, this environment is characterized by small cells and low transmission power.

3. Vehicular environment [4]

This environment is characterized by larger cells and higher transmit power. Rayleigh fading rates are set by vehicle speeds.

Table 2. The tapped-delay-line parameters for each environment [4]

Tap	Indoor A		Pedestrian A		Vehicular A	
	Relative delay	Average power	Relative delay	Average power	Relative delay	Average power
	(ns)	(dB)	(ns)	(dB)	(ns)	(dB)
1	0	0	0	0	0	0
2	50	–3.0	110	–9.7	310	–1.0
3	110	–10.0	190	–19.2	710	–9.0
4	170	–18.0	410	–22.8	1090	–10.0
5	290	–26.0	–	–	1730	–15.0
6	310	–32.0	–	–	2510	–20.0

Application

In this paper, we simulate some multimedia services for E-UMTS, such as File Transfer, Web Service, Video Streaming, and VoIP. Each multimedia services represent 4 UMTS defined QoS classes.

1. VoIP (Conversational class)

Voice over IP (VoIP) allows voice traffic, video, and data packet to be transmitted through IP network. This network uses packet-switched based data communication network in order to use IP network or Internet in telephone. Unlike conventional phone which has its own central/PABX port, VoIP can be installed in any type of Ethernet and IP address.

2. Video Streaming (Streaming class)

Video Streaming allows real-time audio and video digital files in computer network. The protocols video streaming used are Resource Reservation Protocol (RSVP), Simple Multicast Routing Protocol (SMRP), Real-Time Streaming Protocol (RTSP), Real-Time Transport Protocol (RTP), dan Real-Time Control Protocol (RCTP). The latter is a data packet control protocol in RTP used for guaranteeing the quality of streaming.

3. Web Service (Interactive class)

Web service is a web service application through Internet access connection which use Hypertext Transfer Protocol (HTTP).

4. File Transfer (Background class)

File Transfer allows the transfer of any type of data file between different types of computers or networks, using File Transfer Protocol (FTP).

Scheduling

There are 3 scheduling mechanisms which can be used in EURANE module [1], i.e. Round Robin, Maximum C/I and Fair Channel Dependent Scheduling (FCDS). We use FCDS for this simulation, due to its user fairness and power efficiency Characteristic [1]. FCDS is a trade-off of two scheduling mechanism, optimal power based C/I scheduling and fair scheduling based Round Robin [1].

Quality Of Service (QoS)

The following are some QoS parameters are used in this work [5]:

- Data Rate: transmission data rate parameter [in Kbps or Mbps]
- Latency (maximum packet delay): maximum time required from transmitter to receiver [in milliseconds (ms)].
- Packet Loss/Error: error rate parameter from packet data transmission [in percentage (%)]. Packet loss usually happened in limited buffer or incorrect packet sequence.
- Jitter: packet delay parameter which represent smoothness of audio/video playback.

Operation modes on RLC

Radio Link Control (RLC) in UMTS can be configured by Radio Network Controller (RNC) in 3 operation modes [1]:

1. Transparent Mode (TM)

No overhead protocol added to higher layer. Erroneous Protocol Data Unit (PDU) can be thrown and marked as error. Service Data Unit (SDU) can be transmitted without segmentation according to the type.

2. Unacknowledged Mode (UM)

In UM there are no retransmission protocols and data delivery is not guaranteed. Erroneous data can be thrown or marked as error according to the configuration. PDU structure includes sequence number so that PDU entity can be observed to the higher layer. Segmentation, concatenation, and padding included in the header added to data. The entity of this mode is unidirectional because there are no connections between uplink and downlink.

3. Acknowledge Mode (AM)

In this mode, Automatic Repeat Request (ARQ) is used for its retransmission mechanism. At the time RLC can not transmit data correctly, e.g. the maximum number retransmission number has exceed maximum retransmission or time up, SDU will be thrown and the corresponding user will be acknowledged. Segmentation, concatenation, and padding included in the header added to data. The entity of this mode is bidirectional and has the ability to piggyback the link status to the user in the other side.

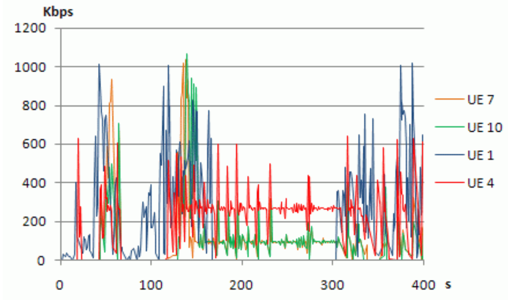
In this simulation Transparent Mode (TM) is not included, because TM supports circuit-switched services instead of packet-switched network[1]. In fact we are working on packet switched network in this work.

Simulation is initiated by generating user input matrices based on each desired environment using Eurane Matlab files on Matlab 7.1. The matrices will be useful to represent the error model of each user environment. Each multimedia data services is put on the last IP node (not a UMTS node). The architecture on Figure 2 above will be simulated in 2 scenarios of RLC mode, AM and UM. Finally, we compared the different RLC modes to find the best performer. In addition to showing the graph result, in this we calculated 3 QoS parameters; delay, throughput and jitter using AWK scripts of Marco Fiore [6].

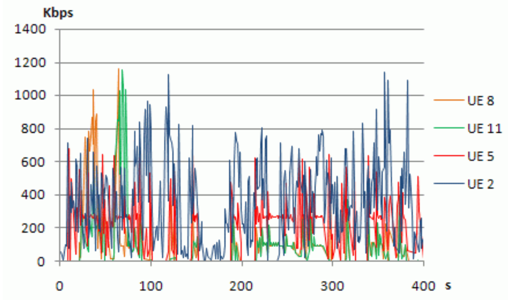
RESULT

Throughput

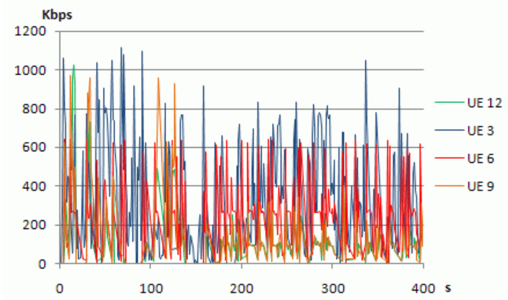
Throughput results acknowledged and unacknowledged mode (AM and UM) has significant differences. The throughput graph result in AM experiment for every service class is smooth and similar. In contrast in UM, only services with high priority have good enough quality, i.e. VoIP and streaming video. It may be due to no available mechanism to guarantee every sent data. Visualization of throughput in AM can be seen in Figure 3 and 4, while throughput visualization in UM can be seen in Figure 5 and 6.



(a)



(b)



(c)

Fig.3. Throughput in Acknowledge Mode (AM)
(a) indoor, (b) pedestrian, (c) vehicular
(note: UE is User Equipment)

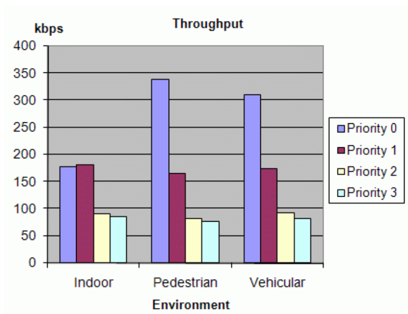
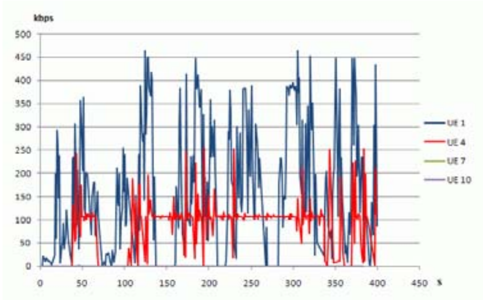
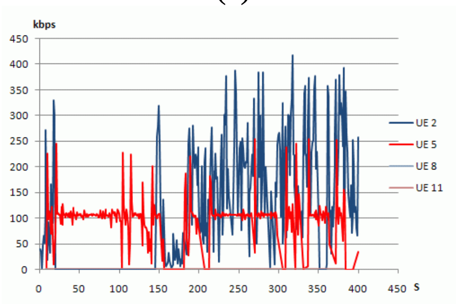


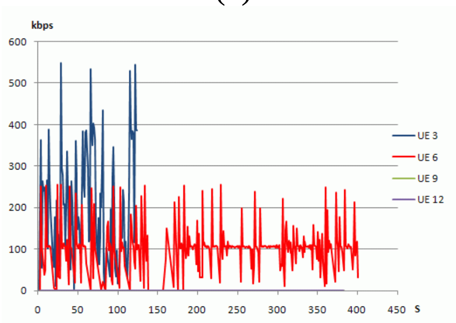
Fig.4. Overall Throughput in AM Mode



(a)



(b)



(c)

Fig.5. Throughput in Unacknowledged Mode (a) indoor, (b) pedestrian, (c) vehicular

(b)

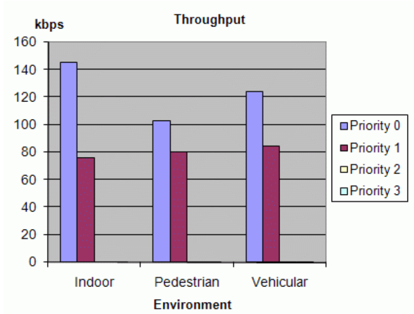


Fig.6. Overall Throughput in UM Mode

Packet Loss

It is interesting to note the experiment result from the number of packet graph in both AM and UM modes. In both of that modes, video streaming service have a large number of packet compared with other priority services. This is may be due to the difference in the number packet sent, which is 3760 packets for VoIP application, and 96 packets for video streaming application. The visualization graph of packet loss in AM is depicted in Figure 7, while the graph of packet loss in UM can be seen in Figure 8.

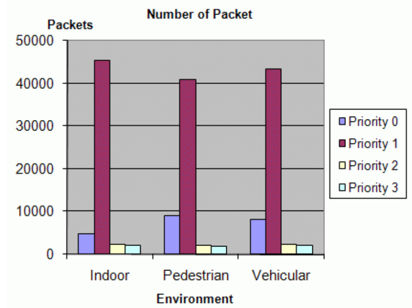


Fig.7. Number of Packet in AM Mode

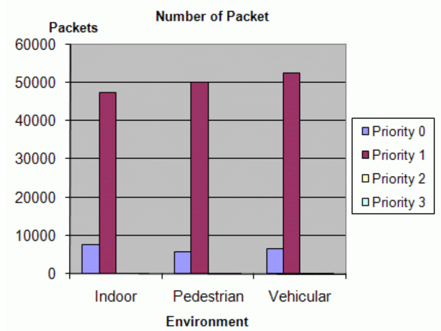


Fig.8. Number of Packet in UM Mode

Delay

The significant difference between delay in AM and UM mode is shown in web service and FTP services. In Indoor user environment for AM Mode, there is a raise in delay. In addition to that, a smooth delay happened for Acknowledge mode. The delay value increases periodically based on its priority grade. Delay visualization in Acknowledge Mode M can be seen at Figure 9, while Figure 10 shows the condition for UM.

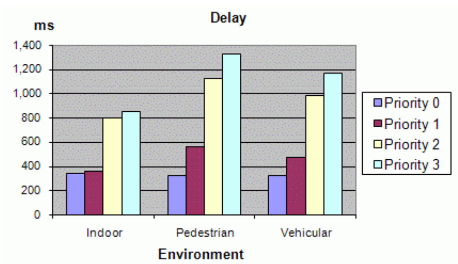


Fig.9. Delay in AM Mode

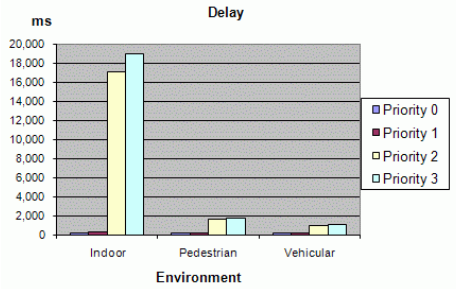


Fig.10. Delay in UM Mode

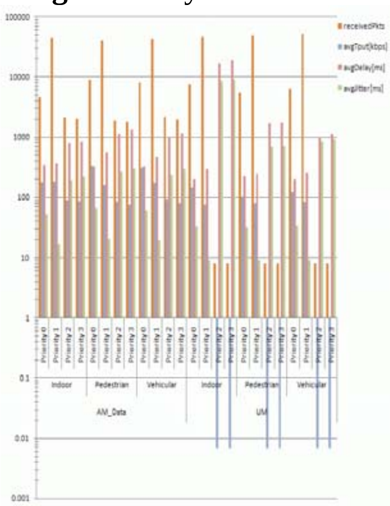


Fig.11. Overall Comparison of AM and UM

Based on the overall comparison of RLC operation modes AM and UM at Figure 11, it can be seen that the number of packet data in UM are larger than in AM, for the 3 users environment. The delay in UM for the web service and FTP are much bigger than delay in AM especially for web service and FTP. The largest raise in delay available for web service and FTP in indoor environment is shown by the value 17.089 and 18.967 seconds. Throughput for web service and FTP in UM has the value of 0 which means that the data rate speed is very small so that it is impossible to support transmission data process.

CONCLUSION

Based on simulation result and analysis conducted, UM operation mode is not suitable enough to be used for web service and FTP services, because the throughput value is near to 0 and the delay is significant. On the other side UM operation mode is suitable for VoIP and video streaming service, even if compared with AM operation mode. AM operation mode is suitable to be used for web service and FTP services. We can conclude that UM mode is suitable for high priority QoS while AM mode is suitable for low priority QoS in UMTS HSDPA network. In the future the use of dedicated channel (DCH) instead of only HS-DSCH in HSDPA UMTS it can be investigated. It also necessary to compare the performance of those two channel in UMTS in delivering multimedia content.

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