Traffic Measurements and Flow Analyses in 3G Network

Rossitza Goleva[1], Seferin Mirtchev[2], Dimitar Atamian[3], Ljubomir Khadjiivanov[4] and Kiril Kassev[5]

Abstract – **This paper presents traffic measurements and statistical analyses performed in 3G network. The aim of the work is to estimate packet flow characteristics. Data, voice and video sessions are captured and analyzed. As a result statistical distributions of the pure and mixed traffic are specified and simulated. The simulation is used to analyse difficult behaviour of the network.**

Keywords – **3G UMTS networks, Traffic sources, Traffic measurements, Quality of Service.**

I. INTRODUCTION

Wireless technologies and their rapid convergence are a matter of fast development recent decades. Because of the new services involved in the networks the Quality of Service (QoS) and Quality of Experience (QoE) estimation is actual topic of research all the time. Their measurements and management procedures are not developed as fast as the market requires and it is a reason for complains from the service quality worldwide.

This paper is focused on traffic measurements and its analyses in 3G UMTS network. The derived results will allow authors to specify proper traffic models for further network design. They are also used to analyse the network behaviour in difficult conditions. The measurements and simulations are based on data, voice over IP (VoIP) and video over IP traffic sources. Attention is paid to the real-time services and services with high packet dispersion.

Active and passive traffic measurements [1] are explained in many papers. All of them concerns performance of the wireless access links that are a key points when considering the performance of the entire Internet connection. Laboratory based measurements of similar type can be seen in [1]. The traffic source in a multi-point videoconferencing application and data for One-Way-Delay, IP-Delay-Variation and Packet Loss Ratio are shown. The asymmetric nature of the UMTS link is demonstrated. There are no rules for traffic analyses that will allow further network behaviour forecast.

The foundation of QoS and QoE analyses can be seen in [2]. Clear mapping between QoE and QoS parameters is still missing.

[1, 2, 3, 5]Rossitza Goleva, Seferin Mirtchev, Dimitar K. Atamian and Kiril Kassev are with the Faculty of Telecommunications at Technical University of Sofia, 8 Kl. Ohridski Blvd, Sofia 1000, Bulgaria, e-mails: rig@tu-sofia.bg, stm@tu-sofia.bg, dka@tu-sofia.bg, kmk@tu-sofia.bg

[4]Ljubomir Khadjiivanov is with the Mobiltel EAD, Kukush str. 1, Sofia, 1309, Bulgaria, e-mail: Khadjiivanov.L@mobiltel.bg

In [3] QoS importance is explained. Measurements of IPTV and VoIP are presented to prove the explained ideas. One-way end-to-end QoS performance statistics in live HSDPA (High Speed Downlink Packet Access) is shown. The phenomenom of CBR-type (Constant Bit Rate) VoIP source traffic affecting with 3G network timing mechanisms is analyzed. The nature of distributions at packet layer and call layer is not explicitly specified.

The nature of TCP session and IP packets levels that affects the packet inter-arrival time is shown on [4]. Clear cross layer approach to the call level is not visible. An overview of existing measurements technologies is done in [5] and [6]. Common guidelines for measurements do not exist.

QoS and QoE parameters and its mapping with WiMAX – DiffServ platform are explained in [7]. Delay and throughput end-to-end budget is shown. There is no way to guarantee end-to-end traffic parameters in DiffServ domains because they work with aggregated traffic.

A very complicated correlation between video, voice quality and loss is published in [8]. The authors also show that the compression algorithm is important. The effect of codec bit rate and losses for video traffic sources is analysed in [9]. Round trip time and how it is influenced by different segments of the network is shown in [10]. Direct mapping to the call layer is missing.

The aim of this paper is to show passive traffic measurements in 3G network domain that will allow mapping of call, session and packet layers with traffic parameters. The distributions of the flows seen in the measured domain are applied in simulation model for further network behaviour analyses and QoS to QoE mapping.

II. TRAFFIC MEASUREMENTS IN 3G NETWORK

Passive traffic measurements are performed in 3G network (Fig. 1) for ftp, Voice over IP (VoIP) and Video over IP services. On the figure the Node-B serves the antenna and transceivers in 3G network, RNC (Radio Network Controller) schedules the resources for base stations, MSC-S (Mobile Services Switching Center Server) manages the network, MGW (Media Gateway) connects GSM and 3G networks, GMSC (Gateway Mobile Services Switching Center) connects the mobile network with Public Switched Telephone Network or Integrated Services Digital Network, HLR (Home Location Register) contains user data including QoS specific parameters, VLR (Visitor Location Register) contains data for all servicing terminals in the network, SGSN (Serving GPRS Support Node) supports packet switching and routing in UMTS network, GGSN (Gateway GPRS Support Node) interconnects UMTS network with other packet switched networks, AUC (Authentication Center) contains all necessary data for user access.

The experiment is configured between Mobile Connect Card connected to a laptop for packet access via the 3G network. The data flow passes through Node B – RNC –

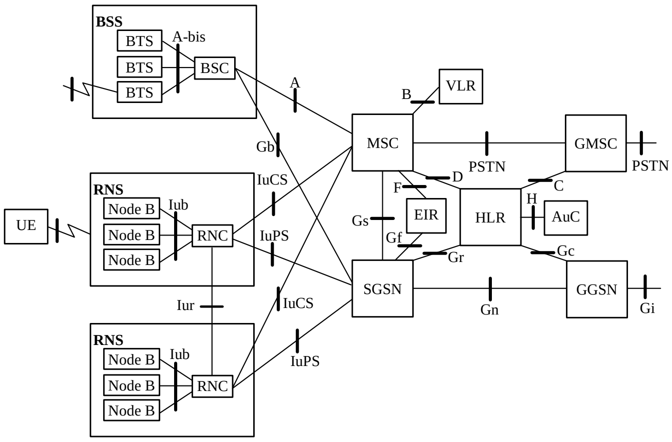

Fig. 1. Packet switched part of the 3G network

SGSN – GGSN –IP backbone transmission network – SBC (Session Border GW Controller) forward and backward following the same path.

Calling-party and Called-Party are UMTS users in the network. Different codecs are applied:

- For VoIP - GSM 6.10, G711a Law, G.729

- For video calls - H.263, H.263+ (1998), H.264 (low end), H.264 (high end)

The "low end" and "high end" are notations for mobile phones with low and high bandwidth accordingly. For tests with ftp traffic a server that is internal in the network is applied.

On Fig. 2 the VoIP packet flows with H.263 and H.264 codecs are shown. The observation is done in one minute interval. The codec characteristics influence the occupancy of the channel significantly. The dispersion of the signals is also significant. On Fig. 3 payload data rates are shown. The main result here concerns the limited capability of the mobile to transmit video with proper quality.

Fig. 4 shows the same video flows in packets per second. On the right side of the graph comparison with data payload can be seen. Low end mobile phones are not good enough to carry video traffic because of their low bandwidth and high difference between overhead and payload.

The video/voice packet sizes observed are mostly between 100 and 600 bytes. Because the traffic is mixed with signalling and data small packets of 20 bytes for acknowledgement and big packets of 1500 bytes from ftp are also visible in the channel.

Observation of packet inter-arrival times (Table I) shows three different groups of packets in the channels. Most of the packets are at packet layer. Some of the packets are at UDP/TCP session layer and few of the packets are at call layer. There is long range uncommon dependence between the session duration and number of packets in session, the distribution of number of packets and inter-arrival time. The traffic sources are modelled as ON/OFF sources with calls, sessions and packets layers. The long range dependence of the packets in buffers requires the use of special distributions like Pareto as well as a cross-layer approach.

Fig. 5 and Fig. 6 show the difference between low end and high end mobiles. The nature of the mobile phone bounds the

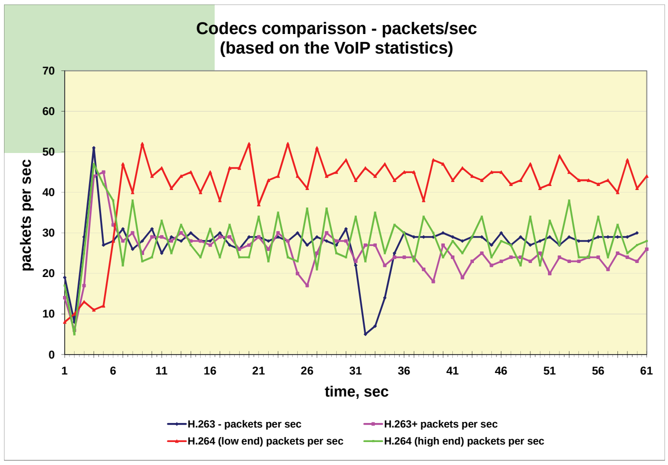

Fig. 2. VoIP flow for H.263 and H.264 codecs

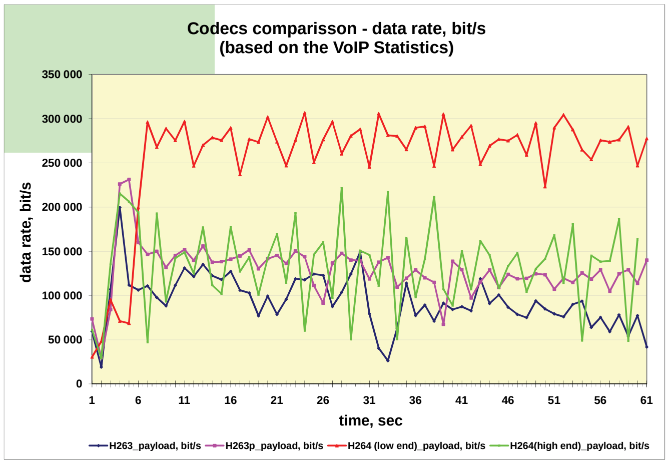

Fig. 3. VoIP flow rates for H.263 and H.264 codec

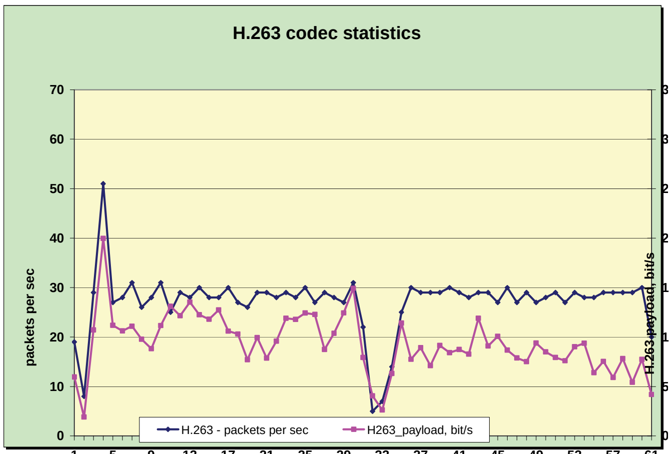

Fig. 4. Video service flows in packets per second and comparison to the payload

payload that might be transmitted in the networks in low end case. So, high end mobile or mobiles with better compression should be applied in order to satisfy the QoS and QoE requirements.

TABLE I
PACKET INTER-ARRIVAL TIMES

Inter-arrival time, ms	Number of packets observed	Layer of consideration
0,5 ms	250	Packet layer
1 ms	457	Packet layer
3 ms	50	Packet layer
5 ms	187	Session layer
8 ms	30	Session layer
11 ms	126	Session layer
13 ms	30	Call layer
15 ms	45	Call layer

III. TRAFFIC SIMULATION IN 3G DOMAIN

Based on the observed data from measurements a simulation model is built in order to analyse the behaviour of the 3G network in the case of mixed traffic. The applied QoS management serves in different ways the real-time multimedia services and the non real-time services. Usually, the real-time services are classified as conversational and streaming QoS classes in 3G network [7]. Because these traffic classes are not supported well by the equipment they are converted into interactive and best effort QoS classes in the simulation model (Table II).

TABLE II
SIMULATION PARAMETERS IN 3G NETWORK

Parameter	VoIP	Video	WWW	FTP
Packet size, bytes	168	208	300	540
Maximal rate, kbps	128	192	384	384
Transport protocol	UDP	UDP	TCP	TCP
Traffic source	ON/OFF with exponent-tial distribu-tion	Determi-nistic traffic	ON/OFF with Pareto distribution	Determi-nistic traffic
Ton, sec.	1.00	-	1.60	-
Toff, sec.	1.50	-	12.00	-

The QoS parameters are implemented in PDP (Packet Data Protocol) context and depend on HLR. The video streams are considered to be symmetric.

In this simulation the Pareto distribution is proposed for WWW traffic. WWW flows have long range dependence

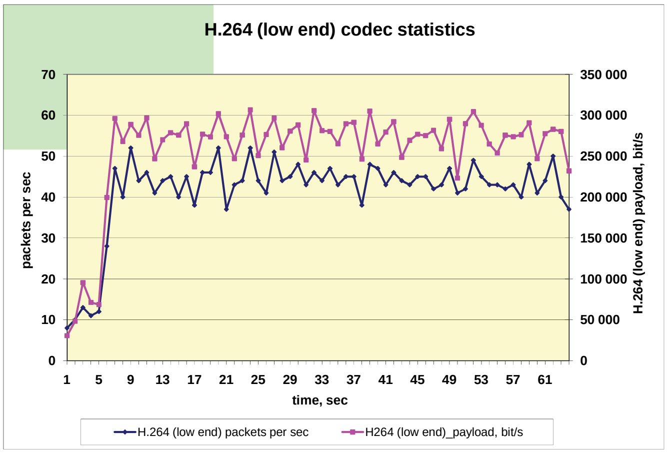

Fig. 5. Low end H.264 codec vs. payload

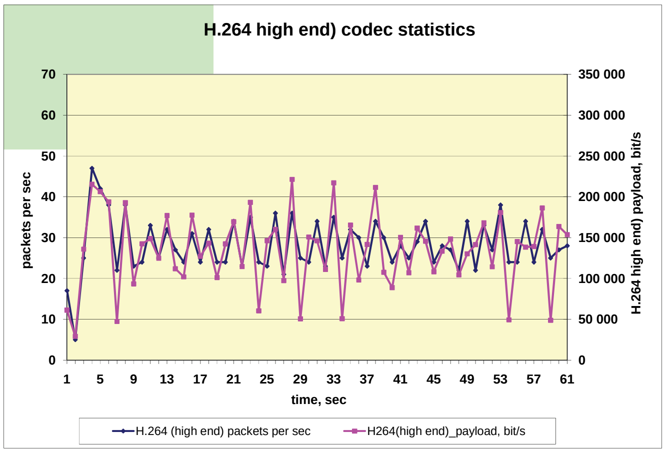

Fig. 6. High end rates vs. payload

between the intervals of web page load and web page reading. This feature increases the delay of the packets and influences the behaviour of the buffers. The family of Generalized Pareto Distributions (GPD) has three parameters: the location parameter μ, the scale parameter σ and the shape parameter ξ (Eq. 1).

The cumulative distribution function of the GPD is in Eq. 1:

$$F(x) = 1 - \left(1 + \frac{\xi(x-\mu)}{\sigma}\right)^{-1/\xi} . \quad (1)$$

We choose substitutions shown on Eq. 2:

$$\eta_0 = \frac{\xi}{\sigma}; \quad \frac{1}{\xi} = 1 + \frac{\lambda}{\eta_o}; \quad \mu = 0 . \quad (2)$$

Therefore, we receive another form of the generalized-Pareto distribution as in Eq. 3:

$$F(t) = 1 - (1 + \eta_o t)^{-\left(1 + \lambda/\eta_o\right)} . \quad (3)$$

The mean value of the generalized-Pareto distribution is on Eq. 4:

$$m_o = 1/\lambda . \quad (4)$$

The mean value is the average inter-arrival time for packets in our study. The parameter λ is the packet arrival intensity [11].

The core 3G packet switched network uses Diffserv for Quality of Service management. SGRN and GGSN are end nodes in the core DiffServ network. The DiffServ to 3G QoS mapping is different in radio interface and in core network. The operators may map 3G QoS classes and DiffServ for uplink in SGSN. Downlink mapping in performed in GGSN [12]. This functionality is part of the network configuration. On Table III possible mapping scenario is shown. Traffic Handling Priority (THP) is used to differentiate 3G UMTS interactive services.

The QoS observed during simulations shows that the error control conforms to the required 2% (Table IV). Low priority queues with THP=3 may introduce high delay. The jitter is less that 3 ms uplink and downlink and is considered acceptable. Bigger values are hardly expected not to exceed 10 ms.

TABLE III
DIFFSERV TO 3G QOS MAPPING IN CORE NETWORK

Service	3GPP UMTS QoS class	IETF Diffserv QoS class
VoIP	Interactive with THP=1	Explicit forward
Video	Interactive with THP=2	Assured Forward 21
WWW	Interactive with THP=3	Assured Forward 11
FTP	Best effort	Best effort

TABLE IV
3G NETWORK SIMULATION RESULTS

Service	Uplink	Downlink
VoIP	Rate 50 kbps	Rate 50 kbps
	Loss < 2%	Loss < 1%
	End-to-end delay = 150 ms	End-to-end delay = 80 ms
	Jitter = 1 ms	Jitter = 1 ms
Video	Rate 175 kbps	Rate 190 kbps
	Loss < 2%	Loss < 2%
	End-to-end delay = 200 ms	End-to-end delay = 100 ms
	Jitter = 3 ms	Jitter = 3 ms
WWW	Rate 20 kbps	Rate 30 kbps
FTP	Rate 40 kbps	Rate 60 kbps

IV. CONCLUSION

In this paper the voice and video service measurement is presented and analyzed. Proper traffic specification of traffic sources obtained is applied for simulation. QoS class mapping between 3G and DiffServ network allows better network design and configuration. During the work the mostly used codecs are considered. We conclude that the recent IP applications will work on a 3G or any wireless domain with small packet sizes. This will decrease congestion probability. The long-range dependence in buffers at packet, session and call layers will require overdimentioning of the network or other compression algorithms. The work continues with more precise measurements and analyses based on active and passive algorithms and additional statistical analyses.

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