Experimental Analysis of QoS Provisioning for Video Traffic in Heterogeneous Networks

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Abstract—This work in progress presents video traffic sources specification and analyses at IP layer. The aim is to analyze network segments with video services that tend to congest the channels and find proper configuration parameters in order to avoid overflow. We analyze the Quality of Service (QoS) and Network Performance requirements in wired and wireless LAN as well as 3G network. Measurements and simulations in different network domains help to obtain the presented data. Special attention is paid to networks with video streaming and interactive video services. QoS of heterogeneous sessions is analyzed. The presented ideas are applicable in network design to ensure QoS guarantee and congestion free transmission from user point of view. The study is also relevant to network access and quality management.

Keywords- Multimedia applications*;* Video services; Quality of Service; Bandwidth allocation

1 Video QoS in IP Networks

This paper investigates the Quality of Service (QoS) and Network Performance (NP) requirements of video traffic in IP networks. The necessity of such analyses is based on the network operators' experience of dissatisfactory quality provisioning and irregular congestions in the network segments. Video QoS to DiffServ mapping has been researched extensively last years. Its practical implementation may differ thanks to the traffic mixtures, offered network services, technology constrains. Detailed configuration parameter investigation is still not completed. We investigate how well mapping QoS provisioning to network dimensioning at the IP layer can support the required network performance. The mapping was tested by performing diversity of simulations with various traffic sources for fixed IP and 3G networks. In order to correctly represent traffic sources measurements in wired and wireless LANs and 3G networks were performed and their flow characteristics were analyzed. The analysis of the QoS and NP parameters can help us to determine appropriate ways to map QoS provisioning algorithms, traffic sources, and technology at IP layer in heterogeneous networks. This will allow application of end-to-end QoS management and more effective and adaptive scheduling.

Based on operator experience it shows that the specific requirements of video applications, e.g., low delays and large number of packets, may cause the (temporal) suppression of other traffic in the network. Therefore, in order to preserve a degree of service fairness, the management of video traffic should be done carefully. Many different approaches in the literature have been proposed to address this problem [1-3]. An idea how to adapt applications to the traffic currently present in the network is shown in [4].

QoS guarantee algorithms like DiffServ, IntServ-RSVP have been a matter of extensive research in the last decade. IntServ-RSVP proved to be better than DiffServ but more expensive. Currently, DiffServ algorithm is the most widely used mechanism on aggregated traffic that has proven to deliver satisfactory performance in practice [5-6]. Furthermore, Pre-Congestion Notification (PCN) and Next Steps in Signaling (NSIS) protocols are designed to guarantee end-to-end QoS [7-8].

A 3G network has defined four types of traffic classes for QoS management. Conversational and streaming classes [9-10] are proper for real-time multimedia applications. However, due to the lack of transparent support by the equipment they are converted into interactive and background classes. In [11] the authors demonstrate a complicated approach towards delay analyses and bandwidth calculation in 802.11e. H.264 video traffic laboratory measurements and simulation in 802.11 with TCP Reno algorithm is shown in [3], [12]. Reported results of delay jitter show that it depends of the assigned priority and is not directly related to the distribution of the inter-arrival time. In practice, network operators compromise by bigger resource provisioning and aggregate traffic prioritization.

The aim of this work is to map configuration management procedures with Quality of Service parameters in wire, wireless and mobile IP environments when the percent of the video traffic is not negligible. The addressed problem has been shown theoretically to exist but it has been rarely monitored in practice. We aim to fill the gap by providing an analysis of real network measurements and using them to argue about system performance. We show that the distribution of the inter-arrival time influences significantly the nature of the traffic.

The paper is organized as follows. First, we study the distributions of the packet sizes and packet inter-arrival times in section 2. Than, we apply these results in simulation models of LAN segment and 3G network. During the simulation, the QoS parameters are conformed and this is the final verification test for the idea presented.

2 Measurements on Video Traffic Sources in Fixed IP Domain and 3G Network

In a fixed IP network, we performed measurements on four traffic types, namely, HTTP transfer, point-to-point (P2P), TV, and VoIP flows. The derived traffic type characteristics are summarized in Table 1. Data traffic collection and analysis is done by Wireshark tool. HTTP measurements we collected in a period of 24 hours for a worldwide set of queried locations. A typical personal computer is used in all experiments. Queries are performed during peak network hours. The data presented below is the most typical set. The applied browser is Google Chrome. Point-to-Point (P2P) traffic is observed for over 500000 packets in Skype and µtorrent sessions of variable duration. For the TV traffic type, we observed a random channel at iptv.bg for 10 minutes, while for VoIP the traffic generated by Cisco IP Communicator between mobile and IP network was monitored.

From the results in Table 1 we notice that the packet intensity, i.e., the mean number of packets per second, and the session duration vary considerably depending on the type of application. Applications with typically large content transfer such as P2P and TV generate much more packets and have considerably longer sessions than lightweight applications such as HTTP or VoIP. Although this is somewhat expected result its identification is of great importance for QoS provisioning, i.e., the network should be able to provide for the fair treatment of traffic classes with distinct characteristics. Moreover, applications with longer sessions are more vulnerable to fluctuations in the QoS over the different domains of a heterogeneous network.

Table 1. Traffic source observed data in fixed IP network

Parameter	HTTP	P2P	TV	VoIP
Packets intensity, num/sec	50	600	400	60
Mean packet size, bytes	500	800	1000	100
Session duration and mean inter-arrival time (equal)	10 sec	1 hour	1 hour	360 sec
Distribution of session duration and inter-arrival time (equal)	Exponential	Exponential	Exponential	Exponential
Distribution of packets within single burst	Deterministic	Deterministic	Deterministic	Depends on application
Priority of packets	Low	Low	Medium	High

We also measured video traffic in mobile operator's 3G network for two different codecs, namely, H.263 and H.264. Data traffic was measured at the Gi interface in the forward direction on the path between a Node B and a Session Border Gateway Controller (SBC), as shown in Fig. 1. The path passes Radio Network Controller (RNC), Serving GPRS Support Node (SGSN), GGSN Gateway GPRS Support Node (GGSN) and backbone IP transmission network.

The measurements are performed in different busy hour's intervals from different network cells. The traffic at Gi interface is filtered by Wireshark and statistically post-processed. Quality of Service parameters are fully satisfied during the experiments. The observed and analyzed values are typical within the 3G network.

The distribution of packets within a single burst depends mainly on the used codec and the application type. Variable rate codecs generate packets with uniform distribution, while fixed rate codecs generate traffic with uniform or close to exponential distributions. For example, the distribution of packets after Xlite VoIP (codecs such as GSM, G.711) is deterministic whereas the distribution after proprietary Skype codec is closed to uniform [4]. The priority of the service can also change considerably the distribution of the packets after application.

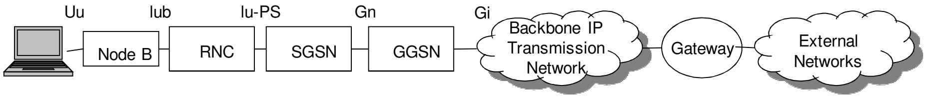

Fig. 1. The observed network architecture

The packet-level traffic characteristics of a single burst in an interactive video session have been studied before by [12-13] and we present them in Table 2. We extend these observations with more detailed analysis of single and multiple bursts.

Fig. 2 shows the packet inter-arrival times and distributions in the forward direction at Gi interface. Single and multiple bursts were considered. Most of the packets sizes are in the range between 180 and 600 bytes and generated with inter-arrival times of up to 20 msec. Differences in inter-arrival time distribution are due to the nature of service and application as well as scheduling.

Table 2. Traffic source specification in 3G network at Gi interface in forward direction

Experiment	Packet size (bytes)	Transport protocol	Inter-arrival time distribution	Packet size distribution
VoIP (GSM 06.01)	100	UDP	Exponential (mean 12 ms)	Deterministic
VoIP (A-law)	230	UDP	Gamma (mean 20 ms)	Deterministic
VoIP (G.729)	80	UDP	Log-normal (mean 12 ms)	Deterministic
VoIP and video (H263)	230	UDP	Exponential (mean 8 ms)	Deterministic
VoIP and video (H264)	230	UDP	Gamma closed to exponential (mean 8 ms)	Deterministic
FTP	40 or 1500	TCP	Gamma (mean 17 ms)	Multi-modal
TV	208	UDP	Almost deterministic	Deterministic

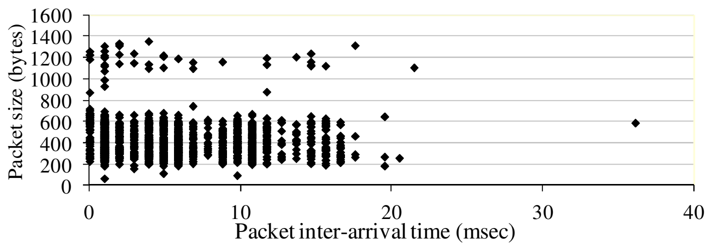

Fig. 2. Packet size versus packet inter-arrival times at Gi interface

Furthermore, the long-range dependence between session duration and number of packets in the session leads to non-trivial distribution of the number of packets versus the inter-arrival time (Fig. 3). Most packets are observed in the time interval of approximately one millisecond. These are packets in the burst when the packets have the same size and deterministic distribution. The second peak on the graph corresponds to inter-arrival times between short bursts in variable rate voice and video streams. The third and fourth peaks correspond to multiple bursts as well but for different session lengths. Our observations are similar to the conclusions drawn from Table 2, i.e., the distributions of the inter-arrival time within a single on-off traffic source in 3G network tends to be different from typical exponential.

Note that the uplink and downlink traffic characteristics are different and the video traffic in both directions cannot be described with one common model from teletraffic point of view. The packets at the receiving end are shaped based on the scheduling and Quality of Service algorithms applied along by the transmission devices. The shaping effect depends very much on the priority of the packets. Packets with high priority will have less end-to-end queuing delay in comparison to the packets with low priority.

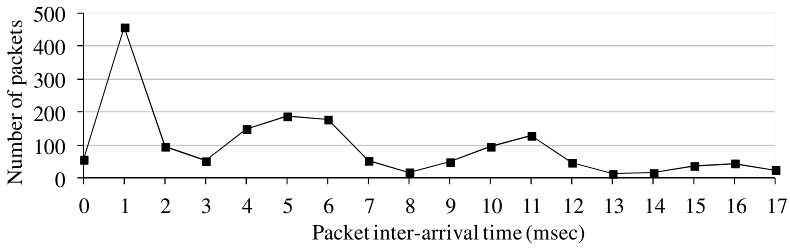

Fig. 3. Number of packets versus packet inter-arrival times at Gi interface for all services

3 Simulations of Video Traffic Sources in Fixed IP Domain and 3G Network

The measurements performed in fixed IP domain were used as input to a simulation model of wired and wireless LANs within the OMNET++ simulator. The observed exponential nature of the burst length allows its straightforward representation in a simulator. The aim is to analyze the fixed IP domain in circumstances not directly verifiable by measurements.

Table 3 shows results for 24 hours simulation. The total number of simulated traffic sources varies between 40 and 200. The number of sources of each service type is equal. Results show that the mean waiting time in a single queue is between 1 and 3 milliseconds, which are acceptable for most end-to-end applications [13]. The high variance of TV and P2P traffic, however, could lead to buffer over dimensioning and resource over provisioning [11]. The long-range dependence of some of the traffic sources, e.g., TV and P2P, is obvious.

In our opinion, traffic separation and partial reservation per traffic type may lead to better performance and QoS guarantee. This can be partially achieved by algorithms such as DiffServ [8] or by enhanced schedulers that dynamically prioritize based on current network load [14]. Given the expected increase in video traffic in wireless networks, however, the latter approach may be quickly incapacitated by the sheer amount of traffic. Therefore, in general, novel approaches towards QoS provisioning in heterogeneous networks may be need. Protocols and QoS mapping in technologies like 4G and DTN where the quality of the signal changes dynamically will require flexible approaches of QoS negotiation.

In similar fashion, we use collected 3G measurements to represent realistic data, voice, and video sources within the ns2 simulator. The simulation model is more complex than the LAN model presented above and covers the entire 3G network as shown in Fig. 1. The TV and the Closed Circuit Television (CCTV) streams are considered asymmetric, whereas the interactive video is considered symmetric. A total of 5, 50, 100, 150, 200 traffic sources are simulated with three priority levels, VoIP having the highest priority and ftp transfer the lowest.

Table 3. Results from simulation in fixed IP domain

Source	Mean rate	Variance	Source	Mean rate	Variance
10 LAN	0,134 Mbps	0,00178 Mbps	50 LAN	6,641 Mbps	0,159 Mbps
10 P2P	0,33 Mbps	1,09 Mbps	50 P2P	15,314 Mbps	50,810 Mbps
10 TV	0,713 Mbps	1,74 Mbps	50 TV	43,479 Mbps	132,882 Mbps
10 VoIP	1.269 Mbps	0.067 Mbps	50 VoIP	1,269 Mbps	0,067 Mbps

The QoS parameters are implemented in Packet Data Protocol (PDP) context and depend on the Home Location Register (HLR). The QoS management at the radio interface is performed in accordance to the 3GPP requirements [15]. The core 3G packet switched network uses DiffServ protocol for QoS management. User data is encapsulated and tunneled in GPRS Tunneling Protocol (GTP) over UDP.

The DiffServ to 3G QoS mapping is different at the radio interface and at the core network [11]. The mapping between IP and 3G network QoS packet marking is done at SGSN for uplink and at GGSN for downlink as well as at HLR. Theoretically, the mapping follows the algorithm shown in the first two columns of Table 4.

Table 4. Theoretical and proposed practical QoS mapping between 3GPP and IETF DiffServ networks

Service	3GPP UMTS QoS class	IETF Diffserv QoS class	3GPP UMTS QoS class	IETF Diffserv QoS class
VoIP	Conversational	EF	Interactive THP=1	VoIP
TV	Streaming	AF21	Interactive THP=3	TV
WWW	Interactive THP=1	AF11	Interactive THP=2	WWW
FTP	Background	BE	Background	FTP

However, the streaming and interactive classes are applied only when they are explicitly specified. Therefore, we propose a mapping as indicated by the last two columns of Table 4, in which there are no conversational and streaming QoS classes. Instead, VoIP and TV traffic types are represented as interactive classes with different priorities. The lower priority of the TV packets adds an additional delay to their transfer times. However, since the TV service is not interactive, we do not expect degradation of the quality perception.

In Table 5 QoS-related data from the performed simulation shows expectedly different parameters for the uplink and downlink, which is strongly influenced by the priority level and scheduling. In the case of packet delay the lower the priority the higher the delay. Hence, we can conclude that QoS can be guaranteed only in case of priorities, traffic shaping, partial bandwidth reservation and over provisioning.

Network dimensioning should take into account traffic mixtures in the network. QoS parameters could change drastically in case of TV flooding or VoIP overloading. Interconnection between mobile and fixed network segments will change the distribution of the traffic and its priority marking. This also can lead to the QoS change.

Simulation results are not directly comparable to measured ones. During measurements, we observed a real network in order to collect information on the behavior of traffic sources, i.e., packet size and inter-arrival time distribution. This information is used in simulation to test the potential network performance for a range of traffic scenarios, which are difficult to realize in a real network. Moreover, in real measurements we are more interested in whether QoS parameters are satisfied. Note that traffic sources are not the same in ns2 and OMNET++ simulation models due to different codecs and specific technology procedures.

Table 5. Video traffic simulation data in 3G network

Parameter	VoIP	TV	WWW	FTP
Packet size (bytes)	208	208	600	1500
Maximal rate (kbps), uplink	80	175	60	50
Maximal rate (kbps), downlink	80	192	125	50
Packet loss, uplink	Up to 2%	Up to 2%	0	0
Packet loss, downlink	Up to 1%	Up to 1%	0	0
Packet delay (ms), uplink	150	100-250	-	-
Packet delay (ms), downlink	80	70-120	-	-
Packet delay jitter (ms), uplink and downlink	2	Less than 3	0	0

Conclusion

This paper discusses first impressions on the QoS support for video traffic in wired and wireless networks. We try to map configuration management algorithms in IP and 3G networks with QoS mechanisms for a range of service classes. The work first presents real measurements of several types of interactive services, e.g., VoIP, streaming, in order to derive traffic characteristics. Different codecs as well as different packet sizes and IP flows are considered. Subsequently, the collected measurements data is used to set simulations such that to investigate the effects of the tested QoS mapping on the end-to-end performance for different traffic mixtures.

We conclude that recent IP cameras require traffic models with rather small packets without fragmentation in order to avoid congestion in the wired and wireless domains, long-range dependence in buffers and heavy-tailed queues. We believe that video traffic should be separated from other types of traffic with bandwidth reservation already at network design. Moreover, the priority level of video streams that tend to flood connections should be carefully chosen. Clear QoS mapping between network domains will improve end-to-end performance. In our future investigations, we will consider mapping QoS requirements at MAC, IP, TCP and application layer as well as QoE evaluation.

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