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Performance Optimization of Voice Traffic against Background Traffic over Asymmetric Digital Subscriber Lines

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Abstract. One of major technical problems that hinder the promotion of voice communication over Asymmetric Digital Subscriber Lines is the unacceptable voice traffic performance in the presence of coexisting data traffic from other applications. In this paper, we study the delay, caused by transmission of long non-real-time data packets over the bottleneck of low upstream bit-rate of Asymmetric Digital Subscriber Line. In order to retain the delay bellow the acceptable limit, we propose using lower values for Maximum Transmission Unit for the traffic deploying the link in addition to the standard IP Quality of Service mechanisms. We investigated this approach through simulations and shown that performance of voice traffic is improved with respect to other data traffic even without increasing upstream bit-rate of the Asymmetric Digital Subscriber Line.

Keywords: voice over Internet protocol, asymmetric digital subscriber line, low upstream bit-rate, delay, maximum transmission unit size

Optimizacija zmogljivosti govornega prometa napram ostalemu prometu pri nesimetričnih digitalnih naročniških vodih

Povzetek. Ena perečih tehničnih pomanjkljivosti, ki povzročajo težave pri govornih komunikacijah prek nesimetričnih digitalnih naročniških vodov, je slaba učinkovitost (časovne zakasnitve in izgube) pri prenosu govornega prometa ob prisotnosti prometa preostalih aplikacij. Zato v tej študiji analiziramo zakasnitve, ki jih povzročajo prenosi dolgih paketov aplikacij, ki sicer nimajo potreb po prenosu podatkov v realnem času, prek obremenjenih povezav v smeri z nižjo bitno hitrostjo. Da bi zakasnitve navzgor omejili na sprejemljivo vrednost, poleg mehanizmov za zagotavljanje kakovosti storitev predlagamo omejitev največje dolžine prenosne enote za promet na tej povezavi. Analizirali smo uporabo tega pristopa in z uporabo simulacij pokazali večjo učinkovitost prenosa govornega prometa v primerjavi s preostalim prometom pri nesimetričnih digitalnih naročniških vodih v smeri z nižjo bitno hitrostjo.

Ključne besede: govor prek internetnega protokola, nesimetrični digitalni naročniški vod, prenos v smeri z nižjo bitno hitrostjo, zakasnitev, velikost največje prenosne enote

1 Introduction

The Asymmetric Digital Subscriber Lines (ADSL's) have been commoditized for broadband access to the Internet networks. Unlike most other broadband access technologies, the ADSL offers an asymmetric bandwidth, designed to support the one-way nature of most multimedia communication, in which large amounts of information flow toward the user and only a small amount of interactive control information is returned. In practice, the upstream bit-rates of ADSL vary from 128 kbps to 768 kbps, while downstream bit-rates are from 1 Mbps to 8 Mbps.

As converging networking is becoming more and more popular, it is likely that ADSL will become increasingly used for voice traffic [1], [2]. The voice over IP (VoIP) applications are symmetric and use the same bandwidth for communication in both directions. Because of this, they are not suited for the ADSL asymmetric concept. In theory, the upstream bit-rate of ADSL should be adequate to carry the VoIP data with some additional background data. In practice, however, an unacceptable delay and jitter are perceived when transmitting VoIP data over low upstream bit-rate of ADSL. The problem is even more obvious when constant bit-rate (CBR) VoIP packets are interleaved with long non-real-time data packets generated by traditional Internet non-real-time applications, such as FTP. During the transmission of these long packets over the low upstream bit-rate link short VoIP packets have to be queued, which causes a variable queuing delay of VoIP packets. This issue will be addressed in the following paper.

There is a number of reports in which the delay time of queuing the CBR VoIP streams over different networks was analyzed [3], [4], [5], [6]. The importance of modem delay versus other causes was assessed in [4]. Internal operation of analog telephone modems was

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examined with respect to delay when transmitting VoIP data streams. Privalov and Sohraby [5] provide an exact analysis of the jitter process in slotted networks like Asynchronous Transfer Mode (ATM). However, this research does not deal with a case where the packet length is heterogeneous. The end-to-end delay performance and delay variance were also addressed in [6], where the switching nodes of IP network were modeled as M/G/1 type queuing models. The study in [7] proposes packet size-based delay differentiation in order to improve jitter characteristics of DiffServ network. However, none of these reports investigates a queuing delay caused by the transmission of long non-real-time data packets over low bandwidth access networks. They do not provide and analyze a solution on how to reduce this delay bellow an acceptable limit.

The remainder of the paper is organized as follows. Sect. 2 provides a brief background and presents an analysis of the delay, relying on our simple mathematical model. Since this problem can not be eliminated by using the IP Quality of Service (QoS) mechanisms (e.g. Differentiated Services), we propose to use lower values for Maximum Transmission Unit (MTU) for the communication over ADSL. The proposed approach is introduced in Sect. 3 and investigated by means of computer simulations in Sect. 4. The results are compared to the existing approaches and delay reduction is demonstrated when using lower MTUs. We also evaluate the effect of the proposed approach to the reduction of the bit-rate of the background non-real time traffic caused by additional overhead. The paper is concluded in Sect. 5.

2 Analysis of delay constrains

In this section we examine the VoIP traffic queuing delay characteristics of a low bandwidth link. The small CBR VoIP packets are delayed when they are affected by the coexisting non-real-time data packets, which share the same bottleneck upstream bandwidth of ADSL (see Figure 1). VoIP packets being mission critical packets, they have to be transmitted as fast as possible. In order to meet this requirement, we assume that the priority scheduling mechanism prioritizes the VoIP traffic against other traffic sharing the same link.

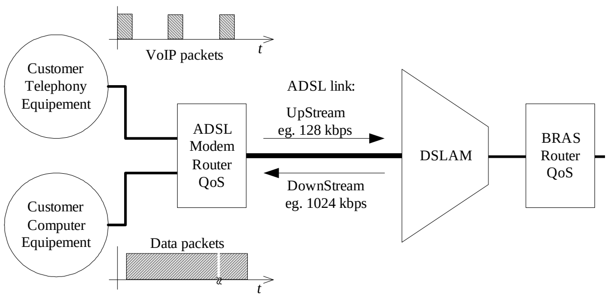

Figure 1. Reference model of VoIP using ADSL
Slika 1. Referenčni model VoIP preko ADSL

2.1 Acceptable delay

According to the ITU-T Recommendation G.114 [6], the VoIP communication quality is good if the total one-way delay is upper limited to 150 ms. This total one-way delay comprises the following delays of components: voice activity detection, packetization delay, IP network delay, codec delay, and packet size-based queuing delay, which is investigated in this paper. In our computations values of component delays as given in Table 1 were assumed.

	Delay		
Codec	G.711	G.729	G.723
Voice activity detection	5 ms		
Packetization delay	10 ms	10 ms	30 ms
Codec delay	0.125 ms	10 ms	30 ms
Lookahead delay	0	5 ms	7.5 ms
ADSL transmission delay	20 – 30 ms (25 ms)		
IP network delay	20 – 80 ms (50 ms)		
Partial sum-one-way delay	90.1 ms	105 ms	127.5 ms
Packet size-based delay	59.9 ms	45 ms	22.5 ms

Table 1. Component delays for different codes
Tabela 1. Prispevki različnih kodekov k zakasnitvi

According to our expectations, the largest acceptable packet size-based queuing delay is given in the last row of Table 1 and can be up to 59.9 ms for a G.711 codec, 45 ms for a G.729 codec and a 22.5 ms for G.723 codec in order to achieve the sum of one-way-delay bounded to 150 ms, as recommended by ITU-T.

2.2 Analysis

With the basic understanding of the problem, we move on to derive the number of packets that are being delayed because of the long data packet transmission and the distribution of delay among the VoIP packets.

This analysis is based on the assumption that the low upstream bit-rate of ADSL is shared by VoIP CBR packets and long non-real-time data packets, generated by a computer-based application like FTP. The non-real-time data is usually carried on the TCP protocol with a congestion control mechanism at the end nodes, so that TCP sources try to get as much link capacity as possible. For the purpose of this analysis the data traffic is modeled as packets of the same length. The IP QoS mechanisms make this traffic occupy all the remaining bandwidth in addition to the VoIP CBR traffic.

We assume the following terminology. Let L_1 be the length of the frame with a VoIP packet and let L_2 be the length of the frame of the data packets at the Ethernet layer. Let R be the upstream bit-rate of ADSL presenting a potential bottleneck for VoIP connections. Time needed for transmitting the data packets can be computed as $T_i = L_i/R$. Let $T_{1\text{interarrival}}$ be the time between transmissions of two consecutive VoIP packets. The length of this interval depends on the codec that is used for VoIP communication (as shown later in Table 2).

If VoIP packets appear after the start of the long packet transmission, they have to be queued until the end of the transmission. This causes delay that cannot be avoided by using a priority scheduling, since the priority packets can not disturb the lower priority packet that is already being transmitted. During the transmission of long packet, N CBR VoIP packets are delayed for a time D_n of a different length, where n denotes the consecutive number of nth VoIP packet which appears after the start of transmission of the long data packet. A pictorial presentation of the problem is shown in Figure 2.

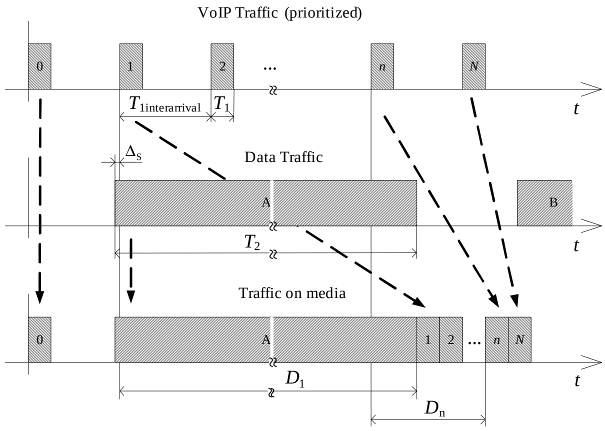

Figure 2. Delay of VoIP packets interleaved by long packets
Slika 2. Zakasnitev VoIP paketov zaradi vrinjenih dolgih paketov

The worst situation occurs when the long packet appears immediately before the next CBR VoIP packet ($\Delta_S \rightarrow 0$). The number of delayed VoIP packets during large data packet transmission is

$$N = \left\lceil \frac{T_2}{T_{1\text{interarrival}} - T_1} \right\rceil \qquad (1)$$

The delay of nth consecutive packet which appears after the start of transmission of the long packet can be expressed as

$$D_n = T_2 + (n-1)\left(T_1 - T_{1\text{interarival}}\right) = T_2\left(1 - \frac{n-1}{N}\right) \qquad (2)$$

It is obvious that the VoIP packet which arrives earliest ($n = 1$), suffers the maximal delay

$$D_{\max} = D_1 = T_2 = L_2/R \qquad (3)$$

With the assumption that the long data packets are following closely each other without vacations, the average packet size-based delay can be calculated as

$$\bar{D} = \frac{1}{N}\sum_{i=1}^{N} D_i = \frac{1}{N}\sum_{n=1}^{N}\left(T_2 + (n-1)\left(T_1 - T_{1\text{interarival}}\right)\right) =$$

$$= T_2 + \frac{(N-1)}{2}\left(T_1 - T_{1\text{interarival}}\right) \qquad (4)$$

For large N the packet size-based average delay can be approximated as $\bar{D} = T_2/2$.

2.3 Numerical Results

First, we determine the parameters used in numerical calculations. The calculations were made for three standard codecs: ITU G.711 [9], G.723.1 [10], and G.729 [11]. Table 2 tabulates the lengths of payloads and lengths of frames on the Ethernet layer, transmission time and interarrival time for traffic of the above-mentioned codecs, as well as for long data packets. Since the purpose of this subsection is to analyze the worst-case scenario of a packet size-based delay, the frames carrying data packets have the maximum length (1518 bytes). The headers of RTP (12 bytes), UDP (8 bytes), IP (20 bytes), PPPoE (8 bytes) and MAC (18 bytes) layers were already taken into consideration.

	L_i	T_i	T_i	$T_{i\text{interarrival}}$
R	payload/frame	128 kbps	256 kbps	
G.711	80/146 bytes	8.9 ms	4.5 ms	10 ms
G.729	10/76 bytes	4.6 ms	2.3 ms	10 ms
G.723.1	24/90 bytes	5.5 ms	2.8 ms	30 ms
long packet	1492/1518 bytes	92.7 ms	46.4 ms	/

Table 2. Packet length, transmission time and interarrival time for different codes and upstream bitrates
Tabela 2. Dolžina paketa, čas prenosa, ter čas med prihodi paketov za različne kodeke in prenosne hitrosti

Using Eq. (1) we calculate the number of packets (N) that are delayed because of the transmission of a long data packet. Figure 3 shows N plotted against upstream bit-rate R, where R varies between 32 kbps to 512 kbps, because most commercial implementations of ADSL support upstream data transmission in this range. The first obvious observation is a significant reduction in N for codecs with higher values of $T_{1\text{interarrival}}$. For instance, at a typical ADSL upstream with bandwidth of 128 kbps, a 1518 bytes long data packet delays 109 VoIP packets if G.711 is used, but only 4 packets in case of G.723. On one side, this might imply that codecs, which produce packets less frequently, prove to be less sensitive to delay caused by long packets. On the other side, we must take into account that G.723 introduces a large algorithmic delay. Therefore, the packet size-based queuing delay should be minimized in order to compensate the algorithmic delay and satisfy one-way delay requirements, in particular when this codec is used.

From Eq. (2), the distribution of delay D_n as a function of sequence (n) of VoIP packets, arrived after the start of transmission of the data packet, is computed. The distributions of packet size-based delay for the link bit-rate of 128 kbps, G.711 codec and different sizes of data packets are shown in Fig. 4.

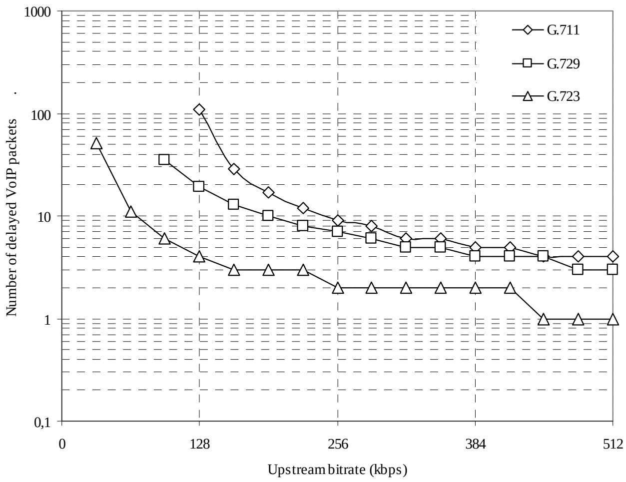

Figure 3. Number of delayed packets (N) as a function of upstream bitrate of ADSL (R)
Slika 3. Število zakasnjenih paketov (N) v odvisnosti od nižje izmed prenosnih hitrosti ADSL (R)

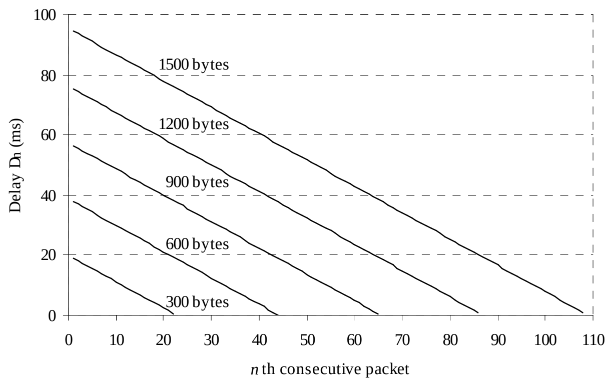

Figure 4. Distribution of the delay (D_n) amongst the consecutive VoIP packets (n)
Slika 4. Porazdelitev zakasnitev (D_n) med zaporedne VoIP pakete (n)

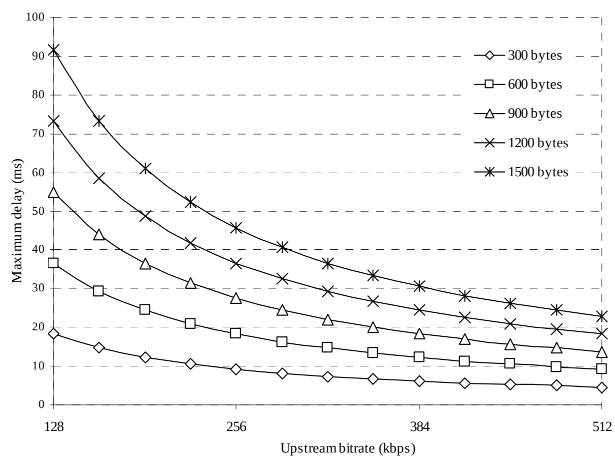

Figure 5. Maximum delay $D_{\max}$ at different bitrates (R) as a function of packet length of background data packets
Slika 5. Največja zakasnitev $D_{\max}$ pri različnih bitnih hitrostih v odvisnosti od dolžine vrinjenih paketov

Using Eq. (3) we computed the maximal packet size-based delay of VoIP packets. For the link bandwidth of 128 kbps and data packets of 1518 bytes the maximum delay D_1 is 92.7 ms. As shown in Eq. (3), the maximum delay of VoIP packets is inversely proportional to the bandwidth of the upstream link R and proportional to the length of the data packets L_2 that delays VoIP packets because of their transmission over the link. Figure 5 shows the impact of the packet size on the maximum queuing delay for G.711 VoIP packets ($D_{\max}$) at different upstream bit-rates of ADSL ranging from 128 kbps to 512 kbps.

3 The Proposed Method with Lower MTU

From Figure 5, a heavy impact of the packet size of the background data traffic on the maximum delay of the VoIP packets is apparent. For this reason, we propose using lower values for MTUs for the communication over the ADSL line – between the ADSL router or consumer equipment and BRAS.

The main idea of this approach is to set the MTU to a value which does not affect small VoIP packets (sizes bellow 200 bytes) but fragments the long data packet to shorter packets which do not cause exceeding the delay over the limit value.

From Eq. (3) the optimal value of MTU can be obtained

$$\text{MTU}=L_2 = RT_2 = RD_{\text{ACCEPT}} \tag{5}$$

By using the values of the packet size-based delay, identified in Table 1, as maximum acceptable values D_{ACCEPT} with Eq. (5), the optimal values for MTU can be determined as a function of upstream bit-rate for different codecs, as shown in Figure 6. Note that the MTU is upper limited to 1518 bytes for the Ethernet network.

The most important observation is that G.723 requires lower MTU values in comparison to G.729 and G.711 codec, in order to satisfy the ITU-T G.114 Recommendation. This is interesting conclusion is drawn from the fact, that although G.723 has much lower bit-rate requirements (6.4 kbps in comparison to 64 kbps of G.711), it introduces a large algorithmic delay (7.5 ms of lookahead delay, 30 ms of packetization delay and 30 ms of codec processing delay), which has to be compensated by minimizing the queuing delay or by using higher upstream bit-rate in order to satisfy one-way delay requirements.

Based on our assumptions for the delay, as shown in Table 1, and with 128 kbps upstream bit-rate of ADSL, the MTU size is 370 bytes for G.723 codec. The maximum packet size-based delay $D_{\max}$ caused by this MTU is 22.5 ms, which with the superposition with other contributions (Table 1) leads to a still acceptable one-way delay below 150 ms. In a similar way, the optimal MTU sizes for the G.729 and G.711 codecs are 730 bytes and 980 bytes, respectively.

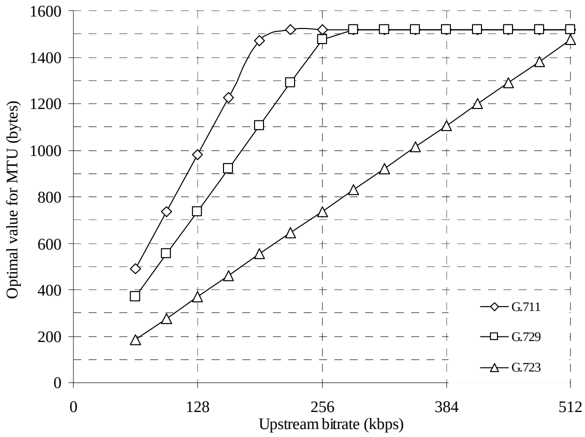

Figure 6. Optimal MTU size for different upstream bitrates and codecs
Slika 6. Optimalna velikost MTU za različne bitne hitrosti

From Figure 6 it can be observed that the links with high bit-rates do not suffer considerable queuing delay caused by the long data packets, because their transmission time is relatively short. This fact is also evident in Figure 6, where, at bit-rates over 512 kbps, the size of MTU no longer needs to be adjusted to a lower value in order to receive an acceptable delay.

4 Performance Evaluation by Simulations

In this section, we use the network simulator *ns-2* [12] to study the performance of VoIP traffic in conjunction with long data packets when using different values for MTU. We consider a single bottleneck link of the capacity ranging from 128 kbps to 512 kbps. The forwarding is done by DiffServ prioritization rules, which ensure the priority of VoIP traffic. The simulated network is shown in Figure 7.

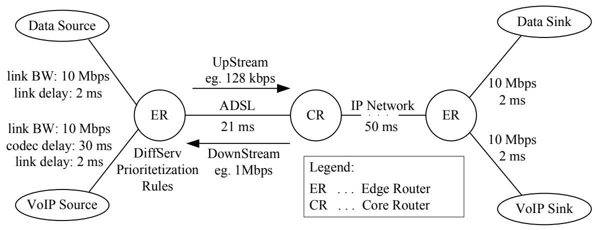

Figure 7. Simulated Network
Slika 7. Konfiguracija simuliranega omrežja

In the first simulation, we investigate total one-way delay of VoIP packets as a function of time. The results, obtained by using G.729 codec for VoIP traffic and data packet of size 1518 bytes (718 bytes) as background traffic at the bottleneck link with bit-rate of 128 kbps, are shown in Figure 8. We can see that the delay of the VoIP traffic is compounded of contributions defined in Table 2 as well as the packet size-based delay caused by waiting for the transmission for long data packets. The values of the delay, obtained by simulation, correspond to the delay computed by the mathematical model presented in Sect. 2.

It can be seen that the one-way delay exceeds the upper limit recommended by G.114 when the non-real time data is sent by packets with the length of 1518 bytes, but is upper limited to the acceptable delay (150 ms) when MTU limits the length of packets to 718 bytes.

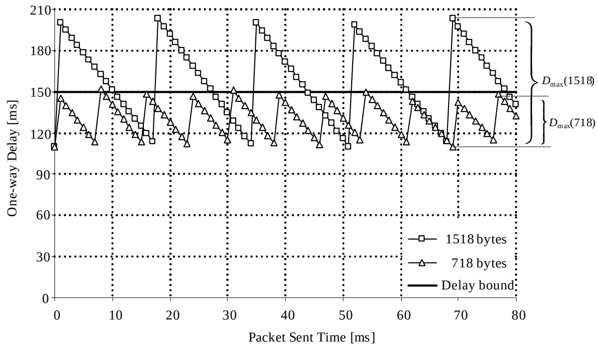

Figure 8. One-way delay of the G.729 VoIP traffic in the time scale with the packet size-based delay component indicated. The simulations analyze two different sizes of the background traffic (1518 bytes and 718 bytes) at 128 kbps upstream bit-rate of ADSL
Slika 8. Časovna odvisnost enosmerne zakasnitve pri G.729 VoIP prometu s komponento, odvisno od dolžine vrinjenenih paketov

In the second series of simulations, the one-way delay is observed as a function of different MTU sizes for different codecs and different upstream bit-rates of ADSL. The simulation results are summarized in Figures 9 and 10, respectively. It is apparent that the one-way delay is smaller for lower values of MTU. The optimal MTU for different codecs and at different bit-rates can be noticed by finding the crossing of the delay function with the constant value of the acceptable delay (150 ms).

The simulations confirmed the computations from theoretical analysis (Fig. 6). G.711 codec allows to use larger MTU values for the background traffic on account of a smaller delay introduced with the process of packetization and coding. Thus, it should be the preferred codec in comparison to G.729 and G.723 although it requires a higher bit-rate.

The use of lower values for MTU introduces additional overhead to the communication over the ADSL link. Since the MTU size is expected to be set to a value that does not affect the VoIP packets, only the long data packets are fragmented. In case of the ADSL link, the bottleneck is observed only at the upstream. Thus, only the packets that are generated by equipment connected in the local area network have to be limited by the MTU size. This causes a slight decrease in the performance of the background non-real-time traffic. To evaluate the amount of the performance loss, we calculated the effectiveness of background traffic communication as a function of different values of MTU. The results in Figure 11 demonstrate that a sharp change in MTU, which performs in a good reduction of the network delay, results in a slight decrease of the network performance compared to the maximum value of MTU. For example, taking into consideration the first simulation (Figure 8), the lost of effectiveness caused by decreasing the MTU size from 1518 bytes to 718 bytes at the ADSL with the upstream bit-rate of 128 kbps is only 2%.

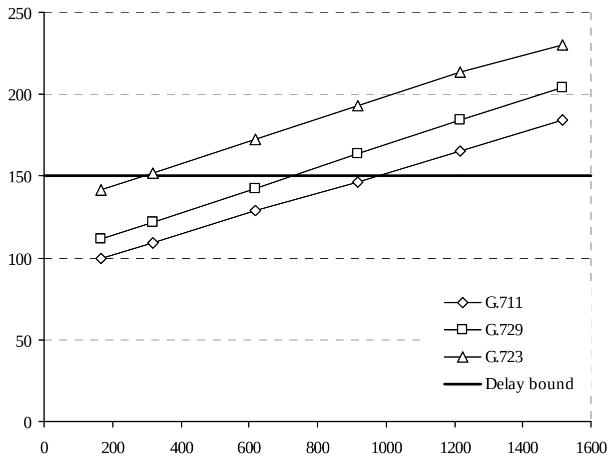

Figure 9. One-way delay as a function of the MTU size for different codecs at 128 kbps upstream bit-rate of ADSL
Slika 9. Enosmerna zakasnitev v odvisnosti od velikosti MTU pri različnih kodekih in bitni hitrosti 128 kbps

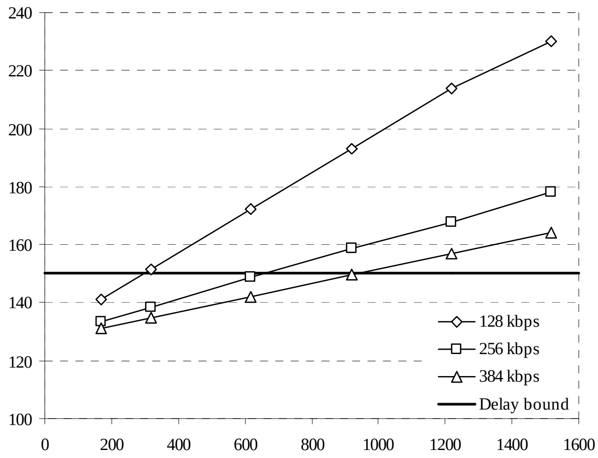

Figure 10. One-way delay as a function of the upstream bit-rate of ADSL for G.723 codec
Slika 10. Enosmerna zakasnitev v odvisnosti od bitne hitrosti pri G.723 kodeku

In conclusion, the simulation results confirm the validity of the proposed approach of using a lower MTU size for the communication over the last mile, in order to achieve a lower packet size-based delay of CBR VoIP streams. It is shown that this approach only slightly affects the network performance and responsiveness to the background traffic.

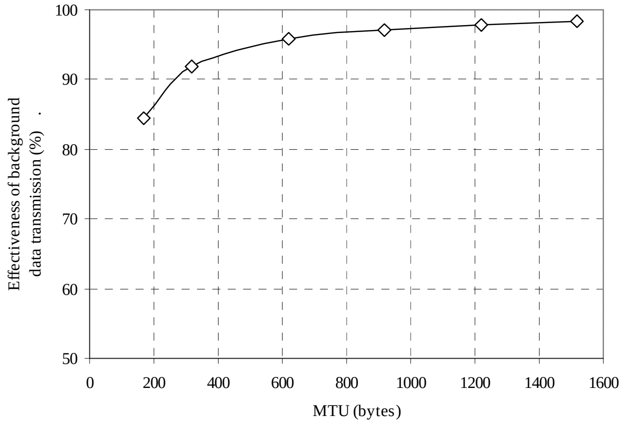

Figure 11. Effectiveness of background data transmission as a function of selected the MTU
Slika 11. Učinkovitost prenosa prometa v odvisnosti od izbrane velikosti MTU

5 Conclusion

In this paper, we studied, through theory and by means of computer simulations, the queuing delay, caused by transmission of long non-real time data packets over the bottleneck of a low upstream bit-rate of ADSL.

We found the following: (1) the delay of VoIP packets introduced by transmission of long non-real-time data packets superposed to the other component delays results in unacceptable delay for an real-time VoIP communications; (2) since the standard IP QoS mechanisms do not adequately address this problem, we suggest using lower MTU sizes for communication over ADSL; (3) the simulations demonstrated that performance of the VoIP traffic is improved comparised to the classical approach; (4) the loss of performance of the background non-real-time data traffic when using the optimal size of MTU is negligible.

It is demonstrated that the packet size-based queuing delay of large data packets (1518 bytes) over a slow upstream bit-rate of ADSL (128 kbps) is 92.7 ms. Together with the delay introduced by VoIP codec processing, framing, nodes processing and network transmission, the queuing delay results in an unacceptable one-way delay for real-time VoIP communications regarding ITU-T Recommendation G.114.

The standard IP QoS can prioritize the CBR VoIP packets, but cannot interrupt the transmission of a low-priority packet that is already being transmitted over the link. The usage of lower MTU sizes for communication over ADSL reduces the queuing time and retains the delay bellow the acceptable limit. We indicate the optimal MTU sizes for different codecs and different bit-rates of ADSL. For example, when we use a G.711 codec over a link with an upstream bit-rate of 128 kbps and network with the conditions as assumed in Table 1, the optimal MTU size is 970 bytes.

Our simulations demonstrate that the performance of the VoIP traffic is improved by using the optimal size of MTU corresponding to the upstream bit-rate in comparison with the standard approach. Although the use of the lower MTU size introduces additional overhead, it is shown that the use of 970 bytes as MTU decreases the performance of the background traffic by only 0.99 %. The change in the MTU size has no impact on the VoIP traffic since the size of CBR VoIP packets is lower than MTU.

Using lower MTU sizes for the transmissions over ADSL offers the possibility to implement sufficient VoIP communication over a low bandwidth upstream bit-rate of ADSL without the necessity of increasing its upstream bit-rate.

Acknowledgement

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