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QUALITY OF THE SERVICE FOR THE TRANSMISSION OF VIDEO SIGNALS OVER IP/MPLS NETWORK

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Abstract: *This article recommends the solution for video streaming into two parts :distribution via IP network and distribution via MPLS VPN network. Based on the theory and basic premises of network routing it is possible to establish good quality streaming and videoconferencing. It is necessary for the quality to be tested in the network congestion conditions, so as to determine how increased traffic influences the quality of service. Advancement of network technologies and continuous increase in access speed to the end user enables good quality video streaming and provision of new services which include various multimedia contents.*
Key words: *video streaming, video signal, IP network, MPLS network*

1. INTRODUCTION

While the quality of service (QoS) in ATM (Asynchronous Transfer Mode) was supported from the very beginning of the network development, in the IP environment the QoS was developing gradually. IP (Internet Protocol) network was not constructed initially for video and audio streaming, therefore did not have the supporting mechanisms. QoS developed in accordance with market needs and in order of acceptance of new protocols.Because of that, practical implementation of QoS is complex, untypical and diverse. Application of QoS went beyond the boundaries of provision of quality services for end-to-end streaming. It is used for telephony, video streaming, data transmission which is classified (Keagy, 2000). There is data which is assigned with greater priority, such as routing and network control protocols. Based on theory and basic settings for the transmission of video stream via routers through the network, it is necesssary to set the router's QoS setting that will meet the needs of video stream transmission and videoconferencing. To transfer the video stream it is necessary to provide the required network capacity, which allows higher priority for transmission of video stream packets ahead of other traffic. QoS is necessary to be tested in the conditions of network congestion, in order to investigate how increased traffic affects the QoS. It is necessary to configure the associated routers to support QoS for video stream.

2. MAJOR CHARACTERISTICS OF QUALITY SERVICE

Within a converged network, QoS is by far the most important implementation consideration (Kelly, 2002.) Networks in which QoS has not been implemented are described as *best-effort* networks. In such networks all packets are treated equally. If there is sufficient network capacity and the routers can carry the traffic, all packets will arrive at their destination. But, this usually is not the case. Major characteristics which refer to QoS are:

- Loss of packets – refers to loss due to network congestion, rather than due to connection failure and similar errors
- Time delay – time required for the packet to arrive to its destination. It is very important for the real-time services that the amount of delay is reduced. The delay occurs due to a number of factors: packaging (signal sampling and coding, serialization (packet transmission to interface depending on the speed of the interface), network data flow (packet propagation through the network)
- Packet delay variation (jitter) – is variability over time of the packet latency across a network. On the side of the coder and decoder there are always buffers which prevent the variation of delays, but the size of buffer is definite and therefore the amount of delay is also definite. Consequently, overflow between buffers is possible. For that reason it is necessary to prioritize certain types of traffic, so that the impact of network on the delay would be minimized.

3. MODELS OF QUALITY SERVICE

There are two basic models of quality service provision: *InterServ (Integrated Services)* and *DiffServ (Differentiated Services)* (Evans & Filsfils, 2007). *InterServ* model assumes that the sender sets up sending specifications. The speed is specified, MTU (Maximum Transmission Unit), etc. The recipients side is specified on the other side. Besides this, signalling is conducted between the recipient and the sender through the protocol RSVP (Resource Reservation Protocol). RSVP reserves the network capacity for the transmission of specific data flow between two locations in the network. There are two basic types of load controlled by the RSVP, controlled load and guaranteed load. The controlled load is service which gives priorities to certain types of load, has no quantitative characteristics, but simply ensures priority through access control. Guaranteed load is a type of load which ensures exact mathematical description of load: delay and speed of flow.There is also a third group, that is *best effort* where there are no guarantees of transmission and all packets are the same. *InterServ* model is simple, it enables QoS per flow and supports the control of connection which serves end points which test the network capacity sufficiency for the transmission. On the other side, through every element of the network there must be an exchange of signalling messages which take up a significant part of the bandwidth in large networks and all nods must support the RSVP which is an additional loading on the network. *Diff Serv* model does not work on every data flow. Instead, "behaviour" rules regarding prioritization are defined on every nod. Classes are created and if a packet which is approaching a certain nod satisfies the defined class, it is being sent according to its own priority, similar to postal works where there are several classes of postage: surface, certified, urgent, etc. Therefore, there is no reservation of network capacity reservation of network capacity from end to end and no signalization from end to end. So every packet can be marked in the IP header with priority bytes and the packet is being transmitted through the network. If the packet falls within a certain class defined on the router through which it passes, than that packet is routed in accordance with the defined priority. Advantage of the *DiffServ* model is scalability, because it is unnecessary to transfer the information (signalization) about specific data flow.

4. QoS REQUESTS FOR THE TRANSMISSION OF VIDEO STREAM

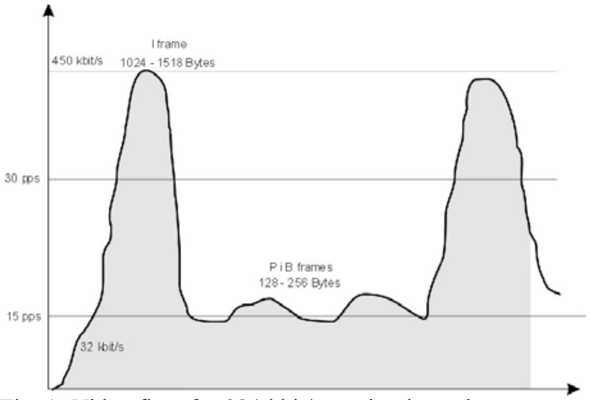

Fig. 1. Video flow for 384 kbit/s session in packets per second (pps)

There are two types of video traffic, so the implementation of QoS differs for those two types (Grgic et al., 1999). It concerns the conferences and distribution of video signals (unicast and multicast). As distinct from some audio-codec such as G.711, video stream has substantially grater variations in speed. I, P or B frames are not transmitted with equal speed. I frame (Intra Coded Picture) is coded independently of the neighbouring images. P frame (Predicted Picture) is coded depending on the previous I or P frames, while B frame (B-predictive picture) use previous and next I or P frames as reference for supplementing and preempting the movement. Figure 1. shows the appearance of video traffic. Also, it should be noted that the video signal consists of audio and video stream also contains audio stream, but needs to contain the IP/UDP/RTP (User Datagram Protocol / Real Time Protocol) header. Therefore, it should be checked that the video stream (and accompanying traffic) have variable speed and variable packet sizes.Transmission of distribution video signal is somewhat different. This is a multicast (or unicast) video stream transmission to the end user without direct interaction.

An important difference in relation to video conference is the delay and absence of requirement to set up delay variation. Realization of QoS for the transmission of video stream is standard. Packets are chosen and marked. Packets are chosen through filtering and chosen according to IP address or UDP port (for example, access lists) then class-maps are defined where DSCP (Differentiated Services Code Points). Packets are placed in WFQ (Weithed Fair Queing) buffers and are being further transmitted. Of course, the whole network should be configured so that it supports QoS for video. IP network is ready to ensure QoS for real-time audio and video services. Also, implementation through MPLS (Multi Protocol Label Switching) private networks for videoconferencing and multicast video streaming assumes usage of QoS. In IP/MPLS QoS is translated into MPLS QoS. In MPLS class for data transmission in real time is defined which supports transmission of video stream.

5. VIDEO STREAM DISTRIBUTION SOLUTION THROUGH IP/MPLS NETWORK

The solution for the distribution through IP/MPLS network is divided into two parts: distribution through IP network and distribution through MPLS VPN (Virtual Private network).
The solution for distribution of video stream through IP network is simple. It is sufficient to bring one PE (Provider Edger) router to the CPE (Customer Premisses Equipment). Routing scheme is shown in Figure 2. UDP multicast coded MPEG-2 (Motion Pictures Expert Group) traffic comes out of coder, it is redirected via CPE router into IP network if there is an IGMP (Internet Group Message Protocol) message for group access on the side of CPE equipment then the signal

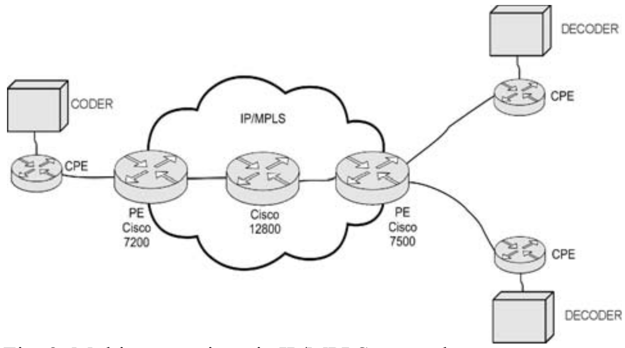

Fig. 2. Multicast routing via IP/MPLS network

leads to the decoder via CPE (Szigeti & Hattingh, 2004). Traffic marking is done on PE routers. First, the access list defines traffic that will be marked. This traffic is sent to a network marked and can be used under QoS. Classification of streams for the user and the network (service provider) is not the same.It is therefore necessary to re-mark the packets at the CPE equipment. Packages, classified according to certain classes can easily be re-marked into other classes. However, it is done only in the case there is a trust between user and provider. In most cases, all packages will be marked by the service provider according to its rules. For packets that are sent over the MPLS VPN network, it is necessary to convert DSCP on the PE router into MPLS QoS. According to recommendations the routers in the IP network should support routing PIM (Protocol Independent Multicast) *sparse.* Besides this, SSM (Source-specific Multicast) where multicasting routing is based on original (Inocast) address (S) and group (multicasting address) G, (Williamson, 1999). Future researches should test the influence of router configuration on QoS for video transmission via IP/MPLS network.

6. CONCLUSION

For implementation of multimedia conferences and multimedia exchange, including video, audio and data transmission through the packet network, it is necessary to obtain synthesis of network, audio and video technologies. Thanks to continuous expansion of telecommunications infrastructure, as well as installation of broadband networks to the small users, significant increase in exchange of multimedia contents became possible. This essay describes the technology which enables the transmission of video stream through the IP/MPLS network, through the ability of multicast routing, QoS protocols and standards which serve as relays for video streaming via IP/MPLS network which must have QoS implemented for the transmission of video stream.

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