Performance Evaluation of Quality of Service Assurance in MPLS Networks

Karol Molnar[†], Jiri Hosek[††], Lukas Rucka[†††], Dan Komosny[††††] and Martin Vlcek[††††]

Brno University of Technology, Faculty of Electrical Engineering and Communication,
Purkynova 118, 612 00 Brno, Czech Republic

Summary
The paper introduces the main features of the mechanism of Multiprotocol Label Switching (MPLS) and evaluates the behaviour of MPLS implementation in current IP networks with respect to Quality of Service support. For this purpose a simulation model was created to examine several aspects of service differentiation in MPLS networks and to evaluate mutual influences of transmission parameters for different traffic classes.
Key words:
Label, MPLS, Network, NS-2, QoS.

1. Introduction

The packet forwarding mechanism used in IP networks is based on a hop-by-hop forwarding paradigm. The destination address of each IP packet arriving to a router is decapsulated and evaluated in the route lookup process. Based on the results of this process in the majority of the situations the packet is forwarded to the corresponding output port of the router and finally sent to the corresponding neighbour (next hop).
The mechanism of MultiProtocol Label Switching (MPLS) implements a similar packet-forwarding paradigm, but in this case the route-lookup process is based on a short, fixed-length identifier called label. These MPLS labels are assigned to the packets at the border of the MPLS network represented by an ingress router of an MPLS domain. Within the MPLS domain the packet forwarding decision is controlled directly by the MPLS labels. Since MPLS labels virtually replace the long and variable IP network addresses this solution simplifies and speeds-up both the look-up and forwarding processes. When the packet leaves the MPLS domain the labels are removed at the egress border node. A router supporting the MPLS mechanism is called Label Switching Router (LSR). The routing path that starts with ingress LSR and ends at egress LSR that means it traverses through the MPLS cloud is called Label Switched Path (LSP).
From the point of view of packet routing there is an important difference between MPLS and classical IP forwarding which lies in the fact that IP forwarding requires packet classification and subsequent route-lookup in every router, whereas with MPLS forwarding the classification is done only by the ingress LSR.

2. Quality of Service Assurance in MPLS

One of the most important features of the MPLS technology is that it can significantly improve network performance and increase the efficiency of quality of service (QoS) support mechanisms. MPLS offers multiple service classes, each associated with different types of traffic. The QoS assurance in MPLS networks is closely related to the usage of an MPLS label. RFC 3031 [1] defines a label as “a short fixed length physically contiguous identifier which is used to identify a Forwarding Equivalence Class (FEC), usually of local significance.” The label makes possible to decouple routing from the forwarding paradigm. The label is an identifier assigned to a packet that tells the network where the packet should be sent. It is located at a header called the Shim header. The 32 bit long Shim header resides between the layer 2 and layer 3 headers. Besides the label itself it also contains other fields, like an experimental Exp field, the indicator of the bottom of the stack called S-bit and the Time to Live (TTL) field. The structure of the MPLS Shim header is shown in Fig. 1.

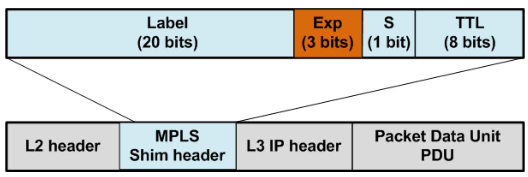

Fig. 1 MPLS Shim header

From the point of view of QoS assurance the 3-bit Exp field is especially important because in most MPLS implementations it is used to hold a QoS indicator. Often the copy of the IP precedence bits of the encapsulated IP packet is stored here. If the Exp bits are used to indicate the differentiated packet treatment than the LSP is called E-LSP indicating that the LSR will use these bits for packet scheduling and policing.

Manuscript received November 5, 2009
Manuscript revised November 20, 2009

Another option how to implement QoS in MPLS networks is to use one label per QoS class for each traffic-flow between two LSP endpoints. In this case a signalling protocol is required to indicate the usage of different labels for the same LSP or prefix. Such an LSP is called L-LSP, showing that the label implicitly holds a part of the QoS information. In the case of L-LSP, the experimental bits still hold a part of QoS-related information. More precisely, these bits are used to express the assigned drop precedence, whereas the label indicates the traffic class. As it was mentioned earlier for an E-LSP, the experimental bits express both the traffic class and the drop precedence [2].

3. Evaluation of QoS assurance in MPLS Network

Although the mechanism of Differentiated Services (DiffServ) is presently the most wide spread QoS support technology for IP-based networks the MPLS can be a preferable alternative in many data networks. In contrast to DiffServ MPLS also controls packet forwarding and due to this feature it is able to use different paths for distinguished traffic classes. There are various types of network services with different requirements on transmission parameters. For example, MPLS can assign faster network path with lower delay to the real-time video flow and a more reliable path to the traffic-flows of classical data services. In this way the application requirements can be better satisfied.
To evaluate the behaviour of the MPLS mechanism with QoS assurance a simulation model had been built in Network Simulator version 2 (NS-2) environment. For this purpose the classical NS-2 environment was extended with the MPLS Network Simulator (MNS) tool and with additional modules for label switching, constraint based routing label distribution protocol (CR-LDP) and class-based queuing (CBQ) scheduling. The MPLS simulation scenario is shown in figure 2.

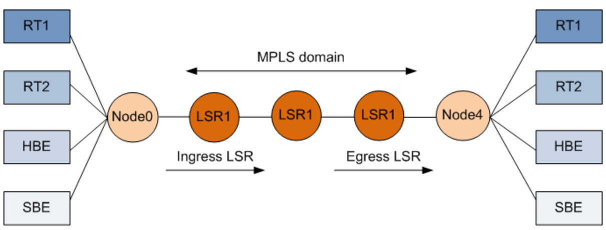

Fig. 2 Topology of the simulated MPLS network

3.1 Simulation Scenario Topology

The network topology in our simulation scenario consists of 5 nodes. Three of them are MPLS capable nodes and the remaining two nodes are standard IP nodes. LSR1 is an ingress edge router of the MPLS domain which ends with the LSR3 egress edge router. On the left side of the figure there are traffic sources, on the right traffic sinks. Thus, the traffic is injected into Node0 and terminated at Node4. The link between Node0 and LSR1 is of the capacity of 2Mb/s to avoid congestion before packets arrive to the MPLS domain. The remaining links are of the capacity of 1Mb/s. The slower links were chosen in order to be able evaluate the aspects of QoS support in the MPLS technology without a need for large number of traffic sources.
There are four types of sources attached to the agents. Each of them generates different type of traffic. The highest priority traffic in the scenario is represented by the model of an end-to-end delay- and jitter-sensitive real-time service. Classical data services are divided into two classes: High Best Effort (HBE) class with higher priority and Simple Best Effort (SBE) class with lower priority. There are four LSPs established from the LSR1 ingress router to the LSR3 egress router for each traffic class. Two ER-LSP paths are used for SBE and HBE traffic and two Constraint-Based Routing Label Switched Path (CR-LSP) for real-time traffic 1 (RT1) and real-time traffic 2 (RT2).
Using constraint-based routing the LSP computation is automatically started by the edge router. After the path computation process, the explicit route is passed to the signalling protocol. The CR-LDP signalling protocol, used in the simulation, establishes the LSPs through the MPLS routers, reserves corresponding network resources and distributes labels to support packet forwarding along the LSPs [3]. We assigned an LSP with a bandwidth of 450kb/s for real-time traffic 1 and an LSP with bandwidth of 350kb/s for real-time traffic 2. The delay of all communication links in the simulation scenario was configured to 10ms.

3.2 Traffic Sources

The traffic sources used in the simulation were configured to accurately model the behaviour of real services. For this purpose traffic generators with different traffic rates, packet lengths and traffic distributions were selected.
The source for real-time traffic 1 was configured to model Voice over Internet Protocol (VoIP) traffic. For this purpose it uses User Datagram Protocol (UDP) agents which generate constant packet flows. These flows are not affected by packet losses meaning, that there are no retransmissions neither mechanisms to adjust the transmission rate. The traffic generation is an On/Off process which means that packets are either sent at full

rate or not at all. Parameters describing the VoIP traffic are as follows:

- burst time (seconds),
- idle time (seconds),
- burst send rate (bps),
- packet size (bytes).

The ITU-T P.50 and P.59 specifications [4], [5] are related to the usage of artificial voice. In recommendation P.50, artificial voice is defined as a "signal that is mathematically defined and that reproduces the time and spectral characteristics of speech which significantly affect the performances of telecommunication systems". The temporal behaviour of human conversation, which includes pauses, mutual silence, etc, is described by recommendation P.59.

Thus the source of the RT1 traffic was configured to use an exponential On/Off distribution with burst time 1,004s and idle time 1,587s, with a bit rate of 6,3kb/s per one connection. The same configuration was used for each of the 100 connection of this type.

The class of real-time traffic 2 models audio streaming coded with G723.1 codec. This type of connections is based on UDP agents too. Traffic parameters of audio streaming were modelled as an exponential On/Off distribution with 0,00005s burst time and 1,8s idle time. The packet size of 240B was used for this type of connection.

The source of HBE traffic simulates random FTP transmissions. It consists of five different FTP agents, each of them randomly establishing TCP sessions. The length of the packets transmitted during the session is 1500B and the transmission is influenced by random delay.

The source of the SBE traffic has exponential character and models background traffic with packets of the length of 512B. The configured speed for this traffic is 200kb/s.

3.3 Packet Scheduling Algorithm

All MPLS nodes in the simulation scenario implement class based queuing. The preferential treatment of delay sensitive applications is achieved by the assignment of the highest priority value. It ensures that the priority-aware scheduler servers this traffic first. The Weighted Round Robin (WRR) scheduling mechanism was used to arbitrate between traffic classes within the same priority level. The WRR scheduler uses weights proportional to the bandwidth allocated to traffic classes. The weight determines the amount of data that a traffic class is allowed to send during a round of the scheduler. The WRR scheduling mechanism is shown in Figure 3.

When a traffic class is overloaded and it is unable to borrow additional bandwidth from parent classes, the scheduler activates the corresponding action handler for that class. For this situation the DropTail policy was chosen for all traffic classes of the simulation scenario. DropTail queues assigned to real-time traffic are smaller, because of undesirable additional delay introduced by queuing. In the scenario 80 percent of the total bandwidth was allocated to the real-time services, 5 percent for High-Best-Effort traffic, 10 percent for Simple-Best-Effort traffic and 5 percent for signalling traffic.

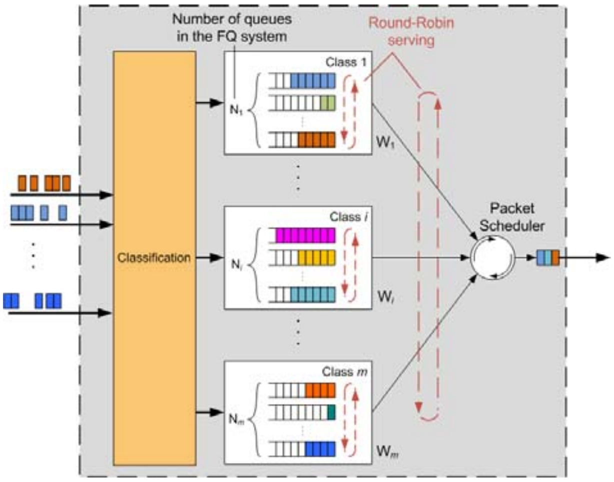

Fig. 3 Weighted Round Robin mechanism

Except the real-time traffic classes all the other classes were permitted to borrow bandwidth from their parent class.

4. Simulation Results

In the following section the most important simulation results are introduced.

4.1 Throughput

Figure 4 shows the throughput of Real-Time traffic 1. It can be seen that the throughput is not changing over the time. Figure 5 shows the throughput of Real-Time traffic 2. As it is obvious from the figure the traffic intensity (number of sessions) of RT2 increased by time. The maximum number of sources was set to 200 and they were all active at the time of 60s. The traffic increases within the period of 0 to 60 seconds linearly. After 60 seconds all sources became active and afterwards the throughput remained practically constant. Both of these real-time traffic classes were served by priority queuing, thus their throughput has not been influenced by other, lower priority flows and their fluctuations.

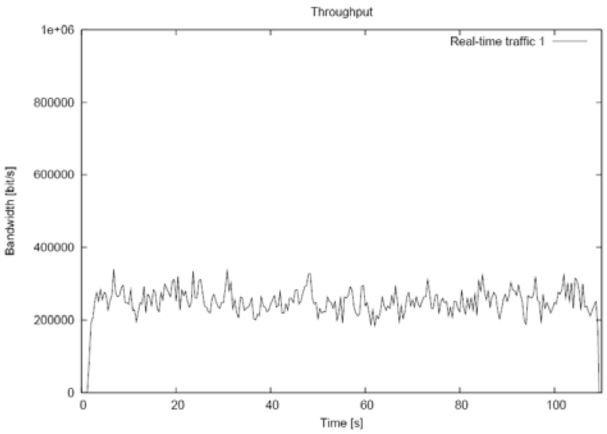

Fig. 4 Throughput of Real-Time traffic 1

As it was described in chapter 3 there is a bottleneck between LSR1 and LSR2 represented by a link with a bandwidth limited to 1Mb/s. It is thus obvious, that with the increasing amount of real-time traffic other lower-priority flows are influenced in their throughput. Figure 6 shows how the throughput of the High-Best-Effort traffic is gradually suppressed. The suppression of the Simple-Best-Effort background traffic is even more serious as it is evident form figure 7. It can be clearly seen that the HBE traffic has higher priority than the SBE traffic and thus the bandwidth limitation in the latter type of traffic is more significant than for the former.

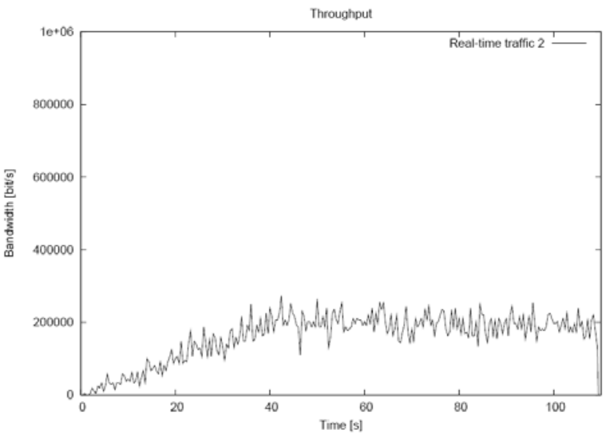

Fig. 5 Throughput of Real-Time traffic 2

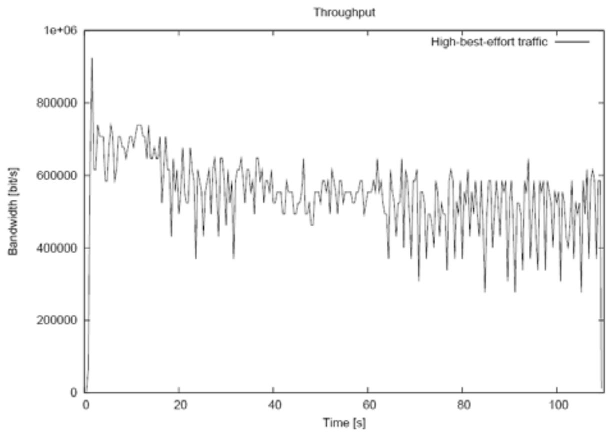

Fig. 6 Throughput of High-Best-Effort traffic

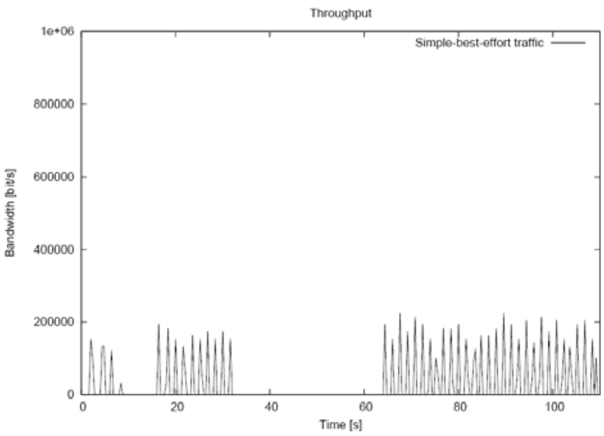

Fig. 7 Throughput of Simple-Best-Effort

4.2 Packet Loss ratio

Figure 8 shows the packet loss ratio measured at the egress node of the MPLS domain. As it is expected, the packet loss ratio for SBT increases with the increasing amount of traffic that traverses through the MPLS domain. This increasing amount of traffic is caused by the RT2 traffic-source. It is evident, that the SBT traffic class has the highest packet loss, because of its lowest priority value. After the stabilization of the TCP connections the packet loss ratio for HBT is around 10 percent. Packet loss ratio for real-time traffic 2 is under 1 percent and in the case of real-time traffic 1 it is almost zero. We can conclude that when the communication link can be over-provisioned it is very important to implement a QoS control mechanism. In our case the CBQ scheduling mechanism clearly ensured preferential treatment for real-time traffic flows prior to other traffic types.

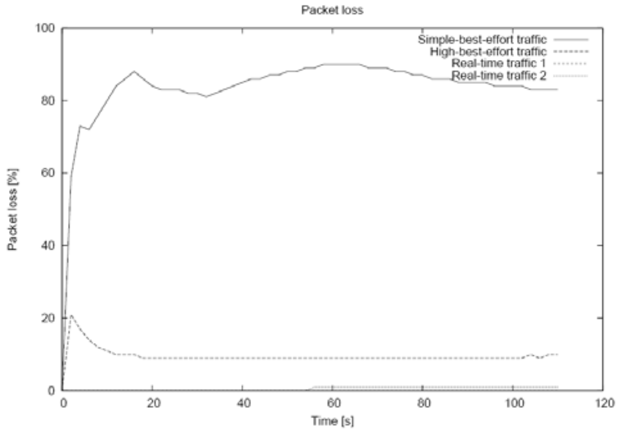

Fig. 8 Packet loss ratio

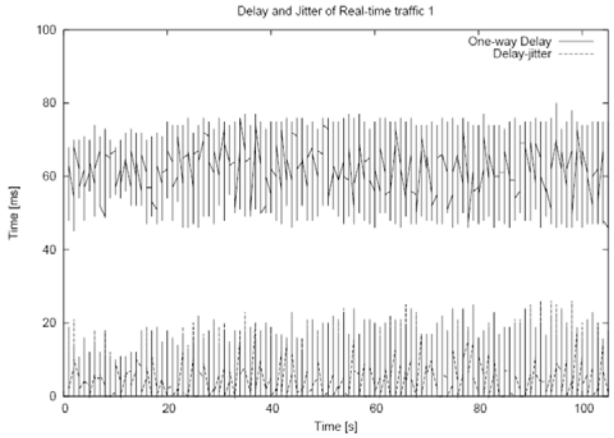

Fig. 9 One-way delay and jitter of real-time traffic 1

4.3 Time Characteristics

In the case of real-time services there are also other important transmission parameters that must be investigated. Since these services have strict requirement on timing we evaluated the one-way delay and delay-jitter for the corresponding traffic-classes.

One-way delay is defined as the time between the transmission of the first bit of a packet and the reception the last bit of that packet at the destination node [6]. This parameter is very important for real-time traffic, especially for VoIP services, where the ITU G.114 [7] recommendation limits the maximum value of one-way delay to 150ms. Figures 9 and 10 show the simulation results for one-way delay for RT1 and RT2 traffic classes. Typical values are also included in Table 1. For both real-time traffic classes the average value of one-way delay is below 65ms. In addition, the maximal delay values for both real-time classes are below 85ms. This value is very satisfying for this type of traffic. There is a small increase of the delay values after the 60th second of simulation time, because of the increasing network load.

Table 1: Measured statistics

	Packet loss [%]	Average delay [ms]	Maximal delay [ms]	Average jitter [ms]	Maximal jitter [ms]
RT1	0.01	62.45	80.41	6.31	26.68
RT2	1.39	63.21	84.90	7.07	39.33
HBE	10.01	252.52	491.95	76.67	295.81
SBE	83.40	789.63	1968.14	145.91	1846.60

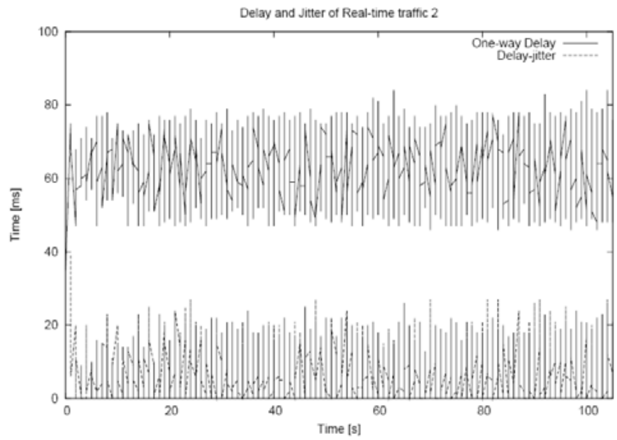

Fig. 10 One-way delay and jitter of Real-Time traffic 2

The delay-jitter of real-time traffic 1 is under 30ms that is again very satisfying. This value can be compensated at the jitter-buffer of the receiver but at the cost of the increased one-way delay. Thus it is important to hold this value low. There are some peaks in the jitter values. That is caused by differences in queuing times for two consecutive packets. The queue lengths of three packets were configured for both real-time traffic classes on all MPLS nodes. There is a considerable impact on one-way delay and delay-jitter when the queue length for real-time traffic is too large.

In the case of the remaining traffic types, the HBE traffic class performs better than the SBE class. Thus the one-way delay for HBE is significantly lower than the delay for SBE. From figure 11 it is evident that after the 60th second of the simulation time the delay value increases. The maximal value is about 500ms, which is still sufficient for classical data services. With the increase of one-way delay we can also see the increase of delay-jitter, but still in an acceptable range. On the other side, the one-way delay of the SBE traffic, show in figure 12, is very high. The packets with such a high delay are considered to be lost in real networks. Such a large delay value is the consequence of the starvation of the Simple-Best-Effort traffic class, because of its low priority.

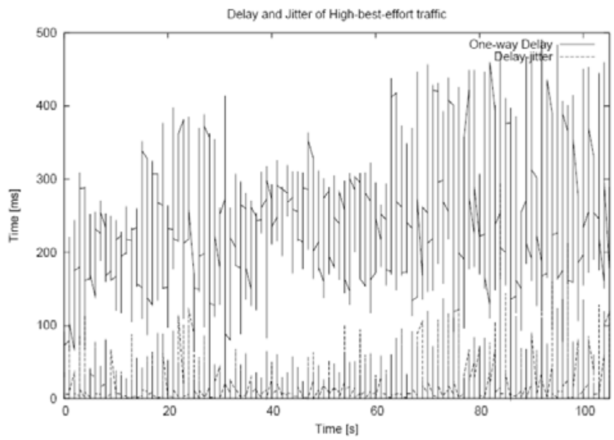

Fig. 11 One-way delay and jitter of High-Best-Effort traffic

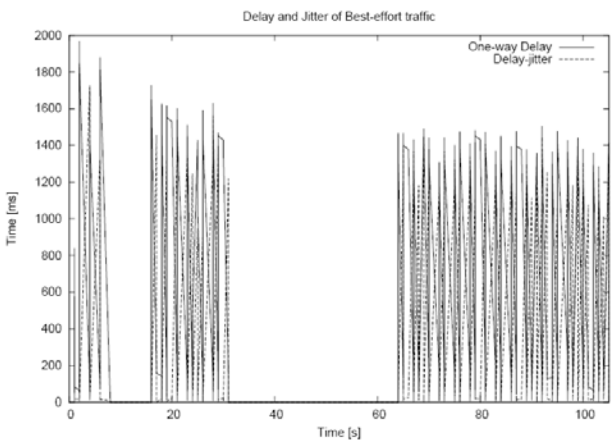

Fig. 12 One-way delay and jitter of Simple-Best-Effort traffic

5. Conclusion

As the amount of traffic processed by the IP networks is still increasing, there is a need to introduce new mechanisms to optimize and speed-up traffic handling. In addition, also other requirements like Quality of Service support are arising.

The technology of Multiprotocol Label Switching offers an efficient and scalable solution to these challenges. In our paper we evaluated the behaviour of the MPLS technology in combination with distinguished packet treatment. Within this evaluation we examined the effect on key transmission parameters like bandwidth consumptions, packet loss ratio, one-way delay and delay-jitter. The simulation results clearly demonstrated the impact of link overutilization on reliable transport protocols like TCP combined with the impact of the priority queuing on real-time services.

Acknowledgments

This paper has been supported by the Grant Agency of the Czech Republic (Grants No. GA102/09/1130 and No. 102/07/1012) and the Ministry of Education of the Czech Republic (Project No. MSM0021630513).

References

[1] E. Rosen, A. Viswanathan, R. Callon *Multiprotocol Label Switching Architecture*, IETF, RFC 3031, 2001.
[2] L. Ghen *MPLS Fundamentals*, Cisco Press, Indianapolis, 2006, ISBN 978-1-58705-197-4
[3] A. Halimi *Quality of Service Networking with MPLS*, TU Wien, Vienna, 2004
[4] ITU-T Recommendation P.50 *Artificial Voices - Series P: Telephone Transmission Quality, Telephone Installations, Local Line Networks - Objective Measuring Apparatus*, ITU, 1999
[5] ITU-T Recommendation P.59 *Artificial Conversational Speech*, ITU, 1993
[6] G. Almes, S. Kalidini, M. A. Zekauskas *A One-way Delay Metric for IPPM*, IETF, RFC 2679, 1999
[7] ITU-T Recommendation G.114 *One-way Transmission Time*, ITU, 2003

Karol Molnar received his MSc. degree in Electronics and Communications (1997) and Ph.D. degree in Teleinformatics (2002) at Brno University of Technology (BUT), Czech Republic. He is with the Dept. of Telecommunications of the Faculty of Electrical Engineering and Computer Science, BUT as Assistant Professors (2002-2007) and Associate Professor (2008 – up-to-now). In his scientific work he focuses on modern network technologies, especially on topics of QoS support in both fixed and mobile network technologies. During the last several years he actively participates in theoretical and research works closely related to the technology of Differentiated Services.

Jiri Hosek received the B.S. and M.S. degrees in Electrical Engineering from Faculty of Electrical Engineering and Communication at the Brno University of Technology in 2005 and 2007, respectively. Recently he is studying for the Ph.D. degree. He is currently an assistant at the Department of Telecommunications of the Faculty of Electrical Engineering and Communication at the same university. His research work has been concentrated on the design of new methods for Quality of Service (QoS) assurance in data networks.

Lukas Rucka received the B.S. and M.S. degrees in Electrical Engineering from Faculty of Electrical Engineering and Communication at the Brno University of Technology in 2005 and 2008, respectively. Recently he has started studying for the Ph.D. degree. He is currently a 2nd-year postgraduate student at the Department of Telecommunications of the Faculty of Electrical Engineering and Communication at the same university. His research work has been concentrated on the Java ME programming platform. Recently he has also been concerned with problems of mesh networks and Quality of Service (QoS) assurance in mobile and wireless networks.

Dan Komosny received his Ph.D. degree in Teleinformatics in 2003. In 2009 he became an associate professor at the Department of Telecommunications, Brno University of Technology. His research is aimed on multimedia systems; communication protocols; real-time communication; VoIP technology; IPTV technology, IP network quality of service and communication pragmatics for Wireless Sensor Network.

Marin Vlcek received his B.S. and M.S. degrees in Electrical Engineering from Faculty of Electrical Engineering and Communication at Brno University of Technology in 2006 and 2009, respectively. In his master thesis he dealt with the evaluation of advanced features of Multiprotocol Label Switching using simulation environments and field measurements.