# Optimization of multimedia flows in multipath networks

Ján Murányi*
Institute of Applied Informatics
Faculty of Informatics and Information Technologies
Slovak University of Technology in Bratislava
Ilkovičova 2, 842 16 Bratislava, Slovakia
jan.muranyi@stuba.sk

## Abstract

Multipath networks are used mostly to improve high availability or to increase throughput. These assets are also invaluable to a VoIP environment. Achieving them by simply adding a new path between two network endpoints is insufficient. Simple loadbalancing of multimedia sessions, as currently provided by routing protocols, can cause a jitter to increase or datagrams to be delivered out of order, which causes multimedia quality degradation. When a link experiences quality degradation or failure, the multimedia sessions utilizing this path are also affected. Routing protocols do not analyze link parameters such as round trip time, jitter or packet loss. The detection of link failure is limited to a physical interface status change or a loss of communication between routers, often a loss of hello packets. These detections are insufficient for multimedia sessions, as they cause longer periods of service outage and do not solve multimedia quality degradation.

This thesis defines a formulated problem in assuring the quality of multimedia sessions in multipath VoIP networks. As a solution we propose a new architecture capable of active management of ongoing multimedia sessions. This approach brings the control from "above" in contrast to the standard control model from "below", where the network is controlled by routing protocols.

## Categories and Subject Descriptors

C.2.1 [**Computer-communication networks**]: Network Architecture and Design; C.2.3 [**Computer-communication networks**]: Network Operations—*network management, network monitoring*

## Keywords

SIP Single Port, Multimedia session management, Multimedia delivery, Multimedia quality evaluation, Multimedia load balancing, Multimedia quality balancing

*Recommended by thesis supervisor: Dr. Ivan Kotuliak. To be defended at the Faculty of Informatics and Information Technologies, Slovak University of Technology in Bratislava, 2015.



## 1. Introduction

Multimedia over Internet protocol (usually covered also as Voice over IP) has developed as a mainstream platform. SIP protocol has been standardized by IETF in [11] and is commonly accepted as a standard. Even the 3GPP has adopted it in IMS (IP Multimedia Subsystem) as a signalling protocol. Despite this technology is mature and widely used it still faces problems, among which are load and quality balancing of multimedia sessions in multipath VoIP networks. We have identified these problems among SIP (Session Initiation Protocol) session management tasks. In [8] we have introduced the SIP Single Port architecture to enhance SIP session management related problems. We have showed this approach is valid and can be used as a solution to identified issues. SIP Single Port itself is however only a tool of a session management and has far more possibilities to be utilized in a proper conditions.

The SIP Single Port is another part of the converged architecture of Next Generation Networks (NGN). Its functionality might be brought various architectures including IMS (IP Multimedia Subsystem).

Formulated problem is in a field of assuring the quality of multimedia sessions in multipath networks. Standard approaches of VoIP architectures in similar scenarios are not optimal, when it comes to load and quality balancing [5]. As a solution we propose a new architecture capable of active management of ongoing multimedia sessions. This approach brings the control from "above" in contrast to the standard control model from "below". In the standard approach, the network is controlled by routing protocols, with limited options for multimedia needs. Approach from "above" allows us to focus the optimization directly on multimedia sessions' needs, without a need to reconfigure whole underlying network during each optimization execution.

This paper is organized as follows: In second chapter we give an overview of present State of the Art in the field of VoIP session management and we discuss available techniques of multimedia sessions' quality evaluation. Third chapter discusses a new architecture utilizing SIP Single Port idea as a solution to the stated problems. Fourth chapter shows achieved results and verifications of a proposed architecture. Conclusions, depicted future works, and final remarks are given in a fifth chapter.

## 2. State of The Art

### 2.1 Multimedia sessions management

SIP based multimedia communication is usually utilizing three separate streams on IP layer for each direction: SIP for the signaling, RTP for a transport of multimedia data, and RTCP for a management of messages related to RTP. Therefore, for a multimedia communication, each user agent needs to allocate three different ports. This complicates the management of SIP sessions as one session is composed of several streams. Even more, communication requires additional steps to be successfully established in NAPT scenarios as described in [9] and [7].

Multiplexing of RTP and RTCP as defined in [10] solved some of these mentioned drawbacks. These benefits can be improved even more by adding the signaling protocol to the multiplexed stream.

The SIP Single Port approach, presented by authors in [8], consists of merging SIP, RTP and RTCP streams together into a smaller transport-layer footprint so that they use only a single port for a communication on each side.

For the purpose of SIP Single Port in [8] we have defined 10 tasks as the session management: service identification, session billing, session recording, administration of firewalls, control of QoS, location control, mobility control, correct service assignment, security tasks and a customer service profiling.

SIP Single Port solution modifies the way SIP-based VoIP establishes a session on the transport layer by merging all streams into a single one. The definition of a network stream in this case refers to a packet flow that is treated using only one policy by all network elements between two consecutive SIP user agents. In order to achieve this behavior, we propose modifications of the content of SDP messages, and an alternate detection scheme, which replaces an address-based and port-based RTP and RTCP demultiplexing.

The footprint of the SIP Single Port stream is limited only to one 5-tupple (source IP address, destination IP address, source port, destination port and protocol). Therefore whole traffic can be managed only by one 4th OSI (Open System Interconnection) layer rule.

Advantages compared to the standard SIP-based VoIP include a smaller network footprint, a higher success rate of a session establishment, a better control of the session including RTP and RTCP and most importantly, an ability to configure border devices that allows problem-free VoIP. The need of relaying of the whole media session in all cases is considered as a negative aspect of SIP Single Port. [8]

To fulfill the idea of the SIP Single Port to multiplex all streams into one single, they have to be properly identified. Current SIP-based VoIP solutions uses transport-layer information to detect the purpose of each given network stream (to differentiate between SIP, RTP, and RTCP). In SIP Single Port idea, transport-layer information is lost and detection schema was modified. SIP protocol is identified using by "SIP/2.0" identifier in first row of the message. RTP and RTCP detection used in

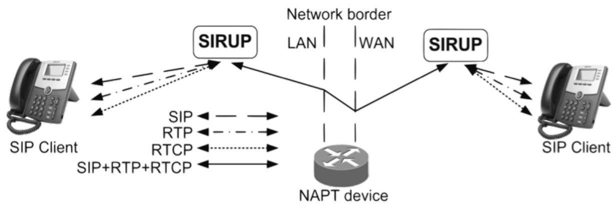


**Figure 1: Two instances of SIRUP in a dual-proxy mode. The communication's footprint between SIRUP nodes is made of one transport layer stream**

SIP Single Port is described in [10].

To demonstrate the proof of concept the program called SIRUP (SIp Rtp mUltiplexing Proxy) was introduced in [8]. As depicted on Figure 1, the solution enabled one-port footprint for the whole communication between SIRUP instances.

### 2.2 Multimedia quality evaluation

The quality of multimedia sessions in VoIP is crucial. For this reason the quality of multimedia sessions have to be evaluated [2]. The quality evaluation can be divided into two methodologies: subjective and objective [6], [4]. Subjective methodology uses users opinion on each particular session. Second methodology is called objective. Using this methodology networks can be evaluated faster and cheaper. The methodology is subdivided into two categories: intrusive and non-intrusive.

Intrusive methods use the original samples of the multimedia session. For the most accurate evaluation, this reference sample is recorded at the originating source or at the entrance to the network. Another sample is recorded at the end of the network or at the receiver. These two samples are compared and the quality estimation is given. Best known algorithms introduced by ITU-T are PSQM, PESQ and POLQA.

In non-intrusive method, the original sample is not a part of an evaluation [12]], [14]. This means, the result is always an estimation of the quality of the source, based on a used technology. Most well-known non-intrusive method is E-model. The E-model's output is a rating factor, $R$ . This is a number from 0 to 100, where 0 indicates the lowest quality, while 100 the best one. This number can be converted to MOS (Mean Opinion Score) using (1). Standard E-model is evaluated using signal noise ratio which is subtracted by various impairments.

$$\begin{aligned} &MOS = 1 \text{ for } R < 6.5 \\ &MOS = 1 + 0.035R + R(R-60)(100-R)7.10^{-6} \\ &\qquad\qquad \text{for } 6.5 \leq R \leq 100 \\ &MOS = 4.5 \text{ for } 100 \leq R \end{aligned} \tag{1}$$

As a standard E-model, introduced by G.107, is rather complex, several simplifications were introduced. All these simplifications aims to reduce the number of measured factors and to provide results closest to the original E-model. The thesis examines three simplified E-model. First simplification was brought by AT&T laboratories in [3]. Applying linear regression to these results, the second method was introduced by M. Voznak, F. Rezac and J. Rozhon in [13]. Another modification of simplified E-model was brought by H. Assem, D. Malone, J. Dunne, P. O'Sullivan in [1].

Based on an analysis of various approaches to calculate the quality of multimedia sessions we will use a model introduced by Voznak, Rezac and Rozhon in [13]. Their results are optimal in various scenarios and are adequately tested in a BESIP project. Further more our work will not require the exact estimation of a quality of multimedia stream. We will focus only on a comparison of results of various streams

## 3. Active Session Management architecture

The problem we address in this thesis is quality and load sharing in multipath VoIP networks. We intend to bring a tool for the optimal distribution of multimedia sessions among all available paths in order to guarantee the best overall multimedia quality, throughout their whole lifespan.

To achieve this goal, we propose a new architecture capable of load and quality sharing of multimedia sessions among multiple paths. This architecture is called Active Session Management (ASM) architecture.

ASM architecture has following fundamental principals:

- Underlying network is able to differentiate sessions based on predefined information (e.g.: transport layer port, IP address) and based on it deliver this traffic via different paths (links, time slots, frequencies). Quality degradation on one path does not affect other paths.
- Change of predefined information in ongoing traffic can occur anytime during an architecture runtime, even during a multimedia session lifetime.
- One and only one ASM server is assigned to one and only one path in the underlying network. Assignment of an ASM client to an ASM server causes all its traffic to be delivered using a path assigned to this ASM server.

The proposed architecture is defined over three planes and two interfaces between them to describe each of its elements as specifically as possible.

### 3.1 Planes

We define ASM architecture as a three-plane architecture. The reason for their differentiation is to guarantee transparency for each participant of multimedia sessions from the view of all OSI layers. We define following planes:

- delivery plane - is a lowest plane of the ASM architecture. It is responsible for data delivery and paths definition,
- multimedia plane - is a middle plane of the ASM architecture. Its purpose is to serve the multimedia service to users,
- control plane - is a highest plane of the ASM architecture. It is responsible for assignment of multimedia plane's sessions to delivery plane's paths.

#### 3.1.1 Delivery plane

Delivery plane is a lowest plane of the ASM architecture. It is responsible for data delivery and paths definition.

Delivery plane is technology independent. It can use any network technology that can assure paths creation based on any predefined information.

The main advantage of ASM architecture is that the network optimizations on ongoing multimedia sessions are not controlled by general methods used in lower layers of OSI model as routing protocols, but using higher layers' mechanisms. This means the delivery plane is unaware of data and any control mechanisms.

A main component of the delivery plane is a path. A path in terms of ASM architecture, ASM path, is a set of links, paths or tunnels through a network that is a subject of optimization of ASM architecture.

These ASM paths are predefined by a system administrator. The number of ASM paths is limited to the number of available unique paths in the network from one edge entry point to another one. This means, the quality change on one ASM path should not affect the quality on other ASM paths. As each ASM path can be utilized multiple times in parallel, there is no benefit creating multiple ASM paths over one path in the network. To guarantee the maximum availability in the network, each available path between edge elements should be utilized as a path of the delivery plane of ASM architecture, ASM path.

Example technologies used in the delivery plane can be for instance MPLS with IP address as predefined information, or SIP Single Port with transport layer port as predefined information as shown in [8].

#### 3.1.2 Multimedia plane

Multimedia plane is a middle plane of the ASM architecture. Its purpose is to serve the multimedia service to users. It is using the delivery plane of ASM architecture as transport layer of OSI model. Multimedia plane is unaware of paths defined in the delivery plane.

Multimedia plane is also technology independent. It can use any VoIP architecture. In this plane, data is generated on the users' user agents, encapsulated into lower layers of OSI model and delivered to proxies or gateways on the server side. From the user's point of view, all elements in the network are gathered in multimedia plane. Standalone it can offer the same service as in ASM architecture, with exception of an enhanced multimedia quality control.

#### 3.1.3 Control plane

Control plane is a highest plane of the ASM architecture. Its elements creates a complex architecture responsible for collecting information, controlling and adjusting the traffic of multimedia plane using interfaces. This plane utilizes the delivery plane to control data of the middle plane. Using interfaces described below, this plane adds information to multimedia plane data, so the delivery plane is able to identify and route the data to assigned paths. This plane is responsible for the optimization of multimedia sessions and therefore it is a core of the ASM architecture. It is also responsible for collecting quality data of streams and performing above tasks based on this information.

All planes are independent of each other and none of its elements is aware of the elements in other planes. This mandate will allow us to be independent of technologies used in any of them. None of the elements has to be modified in any way in order to adapt to changes in other planes.

## 3.2 Interfaces

In order to allow communication between planes and to avoid their changes, interfaces are used. Interfaces are adapted to each implementation of ASM architecture, so the changes to planes are avoided.

- There is no interface between delivery and multimedia plane.
- Delivery - control interface. This interface is a unique and defining property of lower layers. This property is used to define paths. These paths are used by control plane to route the traffic according to ASM network requirements. An example of such property can be an IP address of a multimedia server or a port of SIP Single server. The property is set to data by control plane via multimedia - control interface.
- Multimedia - control interface. Using this interface, the control plane modifies data of the multimedia plane based on a unique and defining property of the delivery plane (delivery - control interface). These modifications are transparent to multimedia plane. For example, if the delivery plane defining property is the IP address, this interface has to guarantee the correct IP address of server side of multimedia plane in the multimedia session will be chosen, so the delivery plane will be able to distinguish the session based on the IP address.

## 3.3 SIP Single Port Active Session Management Architecture

To study the unique properties of ASM architecture, we use SIP Single Port as a mechanism in control plane of ASM architecture. SIP Single Port idea was created as a multimedia session management tool. Together with ASM architecture we can provide solutions to the formulated problem.

Based on ASM architecture, SIP Single Port ASM (SSPASM) architecture is defined over three planes and two interfaces as depicted on Figure 2

### 3.3.1 Delivery plane

A technology used in the delivery plane is independent from SSPASM architecture. The underlying technology has to be able to identify streams based on transport layer port and use this information to route different streams via different paths. On Figure 2 the delivery plane, depicted in dark grey color, is an MPLS network with three tunnels.

### 3.3.2 Delivery - control interface

The delivery - control interface is transport layer port information and is depicted in green-dark grey gradient.

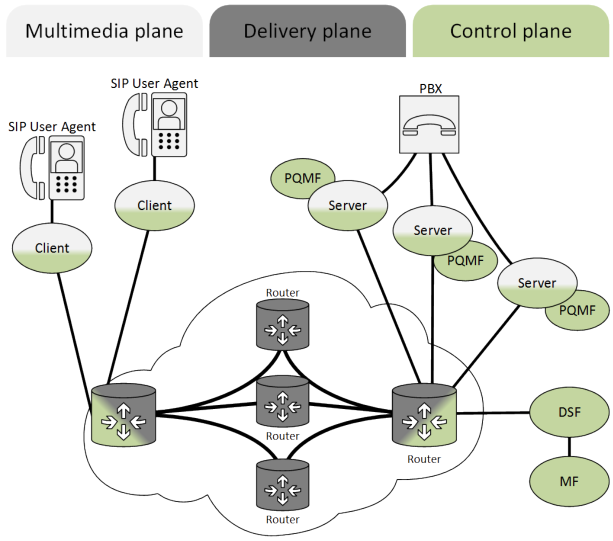


**Figure 2: Logical connections of elements in SSPASM architecture (interfaces between planes are depicted using gradient filling)**

### 3.3.3 Multimedia plane

Multimedia plane, similarly as in SIP Single Port architecture, can be any standard SIP-based VoIP architecture. On Figure 2 the multimedia plane, depicted using light grey, is made of SIP proxy and SIP user agents.

As ASM architecture is technology independent and SIP Single Port can be used with any SIP user agent or proxy, SSPASM architecture can be used in any SIP based VoIP architecture.

### 3.3.4 Multimedia - control interface

The multimedia - control interface is modified SIP Single Port architecture described in the State of the Art. On Figure 2 the multimedia - control interface, depicted in green-light grey gradient, is made of ASM clients and ASM servers.

**Active Session Management client interface.** As depicted in Figure 2 all multimedia plane's user agents are connected to the ASM architecture through ASM client interface. Similarly to the original SIRUP, the client is a stateless proxy, which multiplexes incoming traffic from a user agent and demultiplexes the traffic addressed to it. This task also remains in SSPASM architecture. The new added functionality is an ability to live adapt to required changes in the network. The clients are required to change the delivery plane property (TCP/IP transport layer port) based on the decision of the control plane. Based on this information, the client changes a port and data is sent to a new server. The technology in the delivery plane is not reconfigured in any way and the routing is changed to a more suitable path.

**Active Session Management server interface.** Based on the State of the Art, a server is a stateless proxy, which demultiplexes incoming traffic from client and multiplexes the traffic addressed to a client. Analogous to client, this task is also essential in a new architecture. The process of live migration of a multimedia session to a new path requires several modifications in the server side of the SIP Single Port architecture. A server side does not initialize the change of the OSI model transport layer port, but still it has to adapt to this change. A client always requests the change.

### 3.3.5 Control plane

Control plane is depicted in green color on Figure 2 and is made of three new elements:

- Managing Function (MF)
- Default Server Function (DSF)
- Passive Quality Monitoring Function (PQMF)

The reason to distinguish between these logical elements is to comply with ASM architecture principles of being technology independent.

**Managing Function.** Managing Function of ASM architecture is a central architecture element. MF is responsible for:

- collecting data from Default Server Function,
- collecting data from Passive Quality Measuring Function,
- perform an optimization based on received data,
- send control messages to Default Server Function.

MF is an element responsible for optimization of the traffic in the network. MF gathers data from PQMF and clients via DSF. Based on this information MF controls the network and ensures the quality of all calls is on the highest possible level.

MF does not communicate with SIP elements in multimedia plane or elements in multimedia - control interface. This makes MF technology independent.

The functionality of MF has to be added to SIP Single Port to perform objectives of the ASM architecture.

**Default Server Function.** DSF is an interface between SIP elements of control plane (ASM clients) and MF.

Clients in the ASM architecture have no transport layer port assigned for a communication, prior their start. In the original SIP Single Port architecture, clients had a predefined port that was used throughout the whole runtime of clients. During the start, one predefined port is assigned to all clients. This port is used by the Default Server Function. Every new client in the network at first connects to DSF. After the decision made by the management elements of the architecture, DSF assigns a client with a new server. Only this ASM server is multiplexing and demultiplexing the traffic. DSF itself does not relay any traffic.

Second role of DSF is to control SIRUP clients. Management elements send commands to DSF and DSF sends

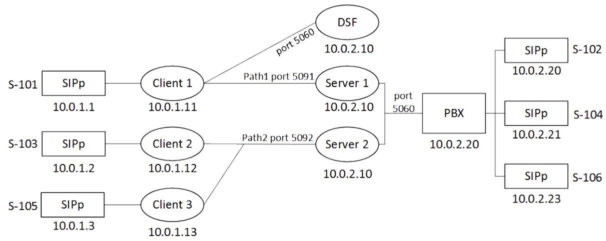


**Figure 3: Testbed topology of ASM network. Clients are connected to servers over NetEm simulated network. To simplify the scheme we have omitted lines connecting Client 2 and Client 3 to DSF**

these commands to appropriate clients. Default Server Function holds subscriptions of clients. The communication between clients and DSF is using SIP protocol.

The functionality of DSF has to be added to SIP Single Port to perform objectives of the ASM architecture.

**Passive Quality Monitoring Function.** Passive Quality Monitoring Function is an element responsible for collecting and sending data of a quality of ongoing multimedia sessions. It is a part of the server ASM interface (Server), where it passively monitors traffic. It collects following information:

- session identification,
- type of the data in the session,
- packet arrival time to calculate a jitter,
- sequence number to calculate a packet loss.

There are two types of policy of sending quality data to MF. Data is either send periodically or send on the quality change information only if any of the measured parameters exceeds the predefined threshold. When the second policy is used the traffic between PQMF and MF is significantly reduced and reaction time of the incident in the network is shortened. The disadvantage is higher resource consumption, as the PQMF has to store the previous values of the measured parameter, and the administrator has to configure the thresholds for all measured parameters.

The functionality of PQMF has to be added to SIP Single Port to perform objectives of the ASM architecture.

## 4. Results

Verification of proposed Active Session Management Architecture will be done upon modified SIP Single Port Architecture. The verification was done in a laboratory environment. We prepared several test-beds where we will verify the proof of concept implementation and where we will verify proposed optimizations over ongoing multimedia sessions.

The overall test-bed in our laboratory is composed of multiple virtual machines distributed in the testing network. This topology is depicted on the Figure 3.

The network topology of the delivery plane in all scenarios is simulated using developed system based on the Linux program called Network Emulator or NetEm. This system allows us to create as many virtual tunnels as needed. There are multiple setting the NetEm offers to simulate real network conditions like modifying a jitter, packet loss, duplication, reordering or setting a needed bandwidth. Using these configuration parameters, we can simulate any conditions in the network.

Elements of the multimedia plane of ASM architecture are deployed on the separate virtual machines. User agents will be simulated using SIP traffic generator SIPp. This will allow us to simulate multiple multimedia sessions from each client based on a scenario's requirements. To simulate a multimedia plane's server side we have deployed Kamailio based proxy and registrar server and as a multimedia handler, there are several other SIPp instances. These SIPp instances are configured to run as a SIPp server with RTP stream echo enabled. This configuration parameter forces the application to send every received RTP datagram back to its server and therefore making a two-way multimedia session. As both SIPp elements are separated and they are not using same resources, the outputs of test-beds will be more accurate.

### 4.1 Load balancing optimization

**Description.** This testing scenario verifies the ability of ASM architecture to optimize the overall bandwidth of all multimedia sessions in the network. In this scenario, we will monitor a bandwidth of each path in the network. Each path of the delivery plane of ASM architecture is depicted with different color.

#### Prerequisites

- DSF has to be registered to MF.
- A network connection has to be established. All reference points have to be available from the view of lower layers of OSI model (TCP/IP connection has to be established).
- The delivery plane of ASM architecture has to be properly configured and delivery - control interface has to be known to control plane of ASM architecture.
- Servers of ASM architecture have to be initialized and have to be configured to reflect the delivery - control interface of ASM architecture. Server 1 has Path 1 assigned using port 5091 and Server 2 has Path 2 assigned using port 5092.
- Clients of ASM architecture have to have a configuration with correct domain or IP address of the DSF.

**Expected results.** Clients are expected to create several multimedia sessions through servers assigned to them. A client assignment to paths is reorganized when the MF identifies the possibility to optimize the network. As each server is defined by one delivery plane's interface, these sessions will use paths assigned to these interfaces. Figure 4 depicts each path with bandwidth. This bandwidth should be equal to a number of calls multiplied by a bitrate of a used G.711 codec in a given time.

#### Testing sequence

1. Client 1 subscribes to DSF and registers to Server 1.
2. Client 2 subscribes to DSF and registers to Server 2.
3. Client 3 subscribes to DSF and registers to Server 2.
4. User 101 (SIPp client) behind Client 1 creates a multimedia session to User 102 (SIPp server) in time $t = 26$ s. At this moment, there is one multimedia session on Path 1 and no multimedia session on Path2.
5. User 103 (SIPp client) behind Client 2 creates a multimedia session to User 104 (SIPp server) in time $t = 103$ s. At this moment, there is one multimedia session on Path 1 and one multimedia session on Path2.
6. User 103 (SIPp client) behind Client 2 creates a multimedia session to User 104 (SIPp server) in time $t = 148$ s. At this moment, there is one multimedia session on Path 1 and 2 multimedia sessions on Path2.
7. User 105 (SIPp client) behind Client 3 creates a multimedia session to User 106 (SIPp server) in time $t = 201$ s. At this moment, there is one multimedia session on Path 1 and 3 multimedia sessions on Path2.
8. At time $t = 252$ s an optimization occurred. MF identifies a calls distribution is not optimal and executes change of an assignment of Client 3 to Path 2. After the optimization is finished, the bandwidth of paths is equal.

**Results.** Results of the test-bed comply with expected results defined above. The bandwidth of both paths during a testing scenario is depicted in Figure 4.

The optimiaztion was performed without a datagram to be lost. After a request of changing the assigned server, the server starts to send packets from a new port (and therefore using a new path). After the acknowledgement reply is received from the server, client modifies its forwarding tables and starts to send packets to the correct destination server (and therefore using a new path).

## 5. Conclusions

In the presented thesis, we have defined the formulated problem in a field of assuring the quality of multimedia sessions in multipath networks. Our contribution for a depicted problem is ASM architecture capable of load and quality balancing. ASM architecture comes from original SIP Single Port architecture. We have proposed several modifications to basic elements to enable live multimedia

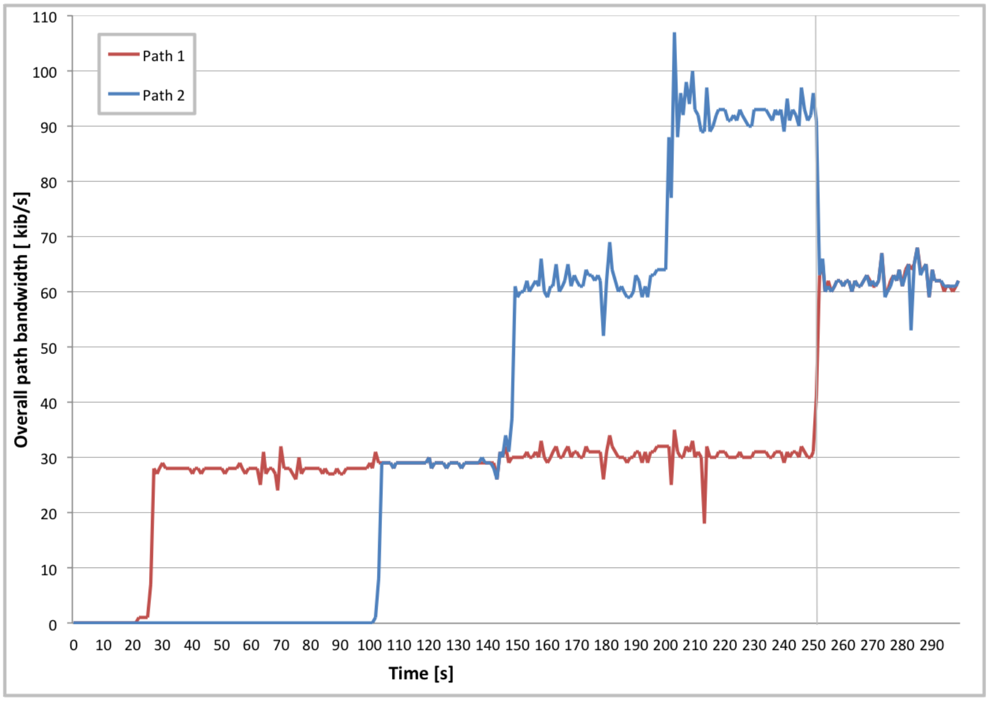


**Figure 4: Graph is showing a bandwidth of both paths during "Optimization of overall bandwidth" test scenario. Grey line indicates the time when an optimization was performed**

session migration. This brings several possibilities to perform various optimizations in the network from "above". We have proposed three plane model of ASM architecture. First two planes come from original SIP Single Port Architecture, while third control plane is introduced in this thesis. It consists of three new elements DSF, PQMF and MF. These are responsible for "above" control approach.

Provided verifications show the ASM architecture is a valid solution to the formulated problem of multimedia sessions' management in multipath VoIP networks.

Brining MF opens many possibilities for future work. One of the most important would be bringing algorithms that would control whole ASM network and would keep sessions balanced during a whole lifetime of the network.

**Acknowledgements**. This work has been supported by the Slovak National Grant agency under the number 1/0676/12, 7FP project HBB- Next and ITMS 26240120029.

## References


[1] H. Assem, D. Malone, J. Dunne, and P. O'Sullivan. Monitoring voip call quality using improved simplified e-model. In *Computing, networking and communications (ICNC), 2013 international conference on*, pages 927–931. IEEE, 2013.

[2] P. Chochol. Qualitative factors that impact real implementing voip in private networks. *Acta Electrotechnica et informatica*, 9(4):61–65, 2009.

[3] R. G. Cole and J. H. Rosenbluth. Voice over ip performance monitoring. *ACM SIGCOMM Computer Communication Review*, 31(2):9–24, 2001.

[4] R. ITU-T. 800: Methods for subjective determination of transmission quality. *International Telecommunication Union-Telecommunication Standardisation Sector (ITU-T)*, 1996.

[5] S. Kandula, D. Katabi, S. Sinha, and A. Berger. Dynamic load balancing without packet reordering. *ACM SIGCOMM Computer Communication Review*, 37(2):51–62, 2007.

[6] J. Karam. A comprehensive approach for speech related multimedia applications. *WSEAS Transactions on Signal Processing*, 6(1):12–21, 2010.

[7] P. Koski, J. Ylinen, and P. Loula. The sip-based system used in connection with a firewall. In *Telecommunications, 2006. AICT-ICIW'06. International Conference on Internet and Web Applications and Services/Advanced International Conference on*, pages 203–203. IEEE, 2006.

[8] J. Murányi, I. Kotuliak, and J. Nemecek. Simplifying the session management using sip single port. In *Next Generation Mobile Apps, Services and Technologies (NGMAST), 2013 Seventh International Conference on*, pages 148–152. IEEE, 2013.

[9] V. Paulsamy and S. Chatterjee. Network convergence and the nat/firewall problems. In *System Sciences, 2003. Proceedings of the 36th Annual Hawaii International Conference on*, pages 10–pp. IEEE, 2003.

[10] C. Perkins and M. Westerlund. Multiplexing rtp data and control packets on a single port. Technical report, 2010.

[11] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler. Sip: session initiation protocol. Technical report, 2002.

[12] M. Voznak. E-model modification for case of cascade codecs arrangement. *system*, 1:10, 2011.

[13] M. Voznak, J. Rozhon, and F. Rezac. Speech quality assessment using computational e-model and its implementation. *International Journal of Mathematics and Computers in Simulation*, 7(1):277–285, 2013.

[14] M. Voznak, M. Tomes, Z. Vaclavikova, and M. Halas. E-model improvement for speech quality evaluation including codecs tandeming. *Advances in Data Networks, Communications, Computers, Faro*, pages 119–124, 2010.


## Selected Papers by the Author


J. Murányi, I. Kotuliak, J. Nemeček. Simplifying the Session Management using SIP Single Port. In *NGMAST 2013: the Seventh Int. Conf. on Next Generation Mobile Applications, Services, and Technologies*, pages 148–152, Prague, Czech Republic, 2013, Los Alamitos: IEEE Computer Society.

J. Murányi, I. Kotuliak. Identity Management in WebRTC domains. In: *ICETA 2013: 11th IEEE Int. Conf. on Emerging eLearning*

*Technologies and Applications*, pages 289–293, Stary Smokovec, Slovakia, 2013, IEEE.

F. Burda, P. Havrila, M. Knězek, K. Konôpková, I. Kotuliak, J. Murányi, J. Nemeček. Optimization of Network Redundant Technologies Collaboration in IMS Carrier Topology. In: *Lecture Notes of the Institute for Computer Sciences, Social Informatics and Telecommunications Engineering: 6th Int. ICST Conf.*, pages 705–712, Berlin, Germany, 2011, Springer Berlin Heidelberg

T. Kováčik, I. Kotuliak, K. Konôpková, J. Murányi. Service Availability Enhancement in Multimedia Networks. In: *Wireless and Mobile Networking Conf. (WMNC)*, Third Joint IFIP, pages 1–6, Budapest, Hungary, 2010.