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VoIP Performance Comparison across MPLS-Based Networks with SIP, H.323 Signaling Protocols Enhancing the QoS through RSVP Protocol

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ABSTRACT

This study demonstrates the application of the OPNET simulation tool to assess VoIP over MPLS networks utilizing the RSVP protocol as a QoS mechanism, w0ith SIP or H.323 serving as signaling protocols for establishing VoIP conversations over the Internet. The findings indicated that H.323 with RSVP over MPLS networks yielded lower end-to-end delay compared to the SIP protocol, rendering it an optimal choice for signaling in VoIP over MPLS networks where minimal end-to-end delay is essential. Conversely, SIP demonstrated the least delay jitter and minimal call setup time.

مقارنة أداء المكالمات الهاتفية عبر الانترنت VoIP في الشبكات المعتمدة على تقنية MPLS باستخدام بروتوكولات الإشارة SIP و H.323مع تعزيز جودة الخدمة من خلال بروتوكول RSVP

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الملخص

تظهر هذه الدراسة تطبيق أداة محاكاة OPNET لتقييم خدمات الصوت عبر بروتوكول الإنترنت(VoIP) على شبكات MPLS باستخدام بروتوكول RSVP كآلية لجودة الخدمة، مع استخدام SIP أوH.323 كبروتوكولات إشارة لإقامة محادثات VoIP عبر الإنترنت. أظهرت النتائج أن بروتوكول H.323 مع RSVP على شبكات MPLS أسفر عن تأخير أقل من النهاية إلى النهاية مقارنة ببروتوكولSIP ، مما يجعله خيارًا مثاليًا للإشارة في خدمات VoIP عبر شبكات MPLS حيث يكون الحد الأدنى من التأخير بين النهايات أمرًا ضروريًا. على العكس، أظهر بروتوكول SIP أقل تأخير في التذبذب وأدنى وقت لإعداد المكالمات.

1. Introduction

In many Internet service provider (ISP) networks, new networking techniques like Multiprotocol Label Switching (MPLS) have emerged recently and are becoming an important and effective part of the networking infrastructure. Nonetheless, the MPLS technique is distinguished by the continuous and stable supply of Internet services using extraordinary transfer rates and minimal transfer delays, in addition to its traffic engineering (TE) with virtual private network (VPN) characteristics [1][2]. However, millions of multimedia streaming, including IPTV, video transmission, and VoIP calls, can now be transmitted over the Internet thanks to social networking websites and broadcasting. These applications demand communication with very little latency and delay jitter, enhanced routing, and acceptable quality of service (QoS) levels, all over the best-effort Internet infrastructure [3]. Consequently, all of that drove ISP providers to employ effective networking methods like MPLS. Moreover, QoS refers to the collection of methods required to control network capacity, packet loss, jitter, and transfer latency in order to satisfy the demands of the application. In IP networks, there are three fundamental approaches to managing QoS: (i) network resource reservation; (ii) packet prioritization; and (iii) traffic engineering [6].

Therefore, resource reservation protocol (RSVP) has been suggested as a signalling protocol which, allow applications to reserve resources in a way that allocates link bandwidth and buffers of intermediate nodes to the running flows over the network. Furthermore, throughout the path from sender to destination, the RSVP protocol makes it easier to write relevant flow status information into the routers. RSVP protocol has its place into *Integrated Services (IntServ)* QoS model [4] [6] [7].

VoIP, which combines voice call packetization and digitization, is a technique for delivering voice calls via IP networks. To cut costs

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and address the underuse of traditional dedicated phone circuits, VoIP is built to operate over the Internet. VoIP was also enhanced by using available signalling protocols like session initiation protocol (SIP) or H.323 in order to enable a number of sophisticated services, including contact centres, multimedia conferencing, advanced call routing, and voice mail. [4] [8] [3] [4]. On the other hand, the SIP protocol is constructed in a bottom-up manner and leverages current Internet protocols to meet its requirements, whereas H.323 is developed using a top-down method and defines a comprehensive framework with detailed protocols. Furthermore, the scalability of addressing in H.323 is problematic. Because SIP is more straightforward and provides text message capabilities similar to those of the conventional protocols such as Hypertext Transfer Protocol (HTTP) or Simple Mail Transfer Protocol (SMTP) it is preferred in most of implementations. H.323 is also a binary protocol, which makes it less user-friendly for technical purposes when performing maintenance. In addition, SIP was designed as a scalable protocol, whereas H.323 was not designed with easy expansion in mind, making it difficult to add new capabilities [3] [6].

An experimental study employing modelling and simulation techniques is presented in this paper [9], by employing open shortest path first (OSPF) [5] as an essential routing protocol with a RSVP protocol for a QoS means to manage a particular receiver-to-sender QoS delivery to specific receivers' VoIP flows, the investigation examines VoIP performance across MPLS networks using the OPNET tool. Furthermore, the case study presents a comparative analysis of SIP and H.323, two distinct signalling protocols used in VoIP communications.

Related Work

First Using the OPNET tool, Abdul-Bary et al. presented a simulation study in [10] regarding the performance analysis of IP and MPLS TE networks for multimedia applications like VoIP and video conferencing. The outcomes of the simulation demonstrated that MPLS-TE outperformed IP networks in terms of performance. The RSVP protocol and the VoIP signaling protocol that is being used were not mentioned in the research on [10]. Using the OPNET program, KeerthiPramukh Jannu and Radhakrishna conducted a VoIP performance analysis study in two settings that were similar to IP and MPLS networks in [11]. Using the OPNET tool and routing protocols such as enhanced inner gateway routing protocol (EIGRP), OSPF, and router information protocol (RIP), Sllame and Aljafari [12] demonstrated a performance comparison study of VoIP over IP/MPLS networks. Nonetheless, the simulation findings showed that MPLS improved multimedia streaming by lowering packet latency, jitter, and end-to-end latency significantly lower than IP networks. Consequently, it is determined OSPF routing protocol performs better than EIGRP and RIP with MPLS. However, the RSVP protocol has not been taken into account, and the authors of [12] have not stated which VoIP signalling protocol they are using in their experimental investigation. A comparison study between the Differentiated Services (DiffServ) model and DiffServ with MPLS TE was presented by Vesna et al. in [13]. The findings showed that the MPLS TE case study with DiffServ enhanced network performance beyond the scenario with DiffServ alone.

Multimedia Session's Requirements and QoS performance parameters

More bandwidth is needed for multimedia communication than for any other type of Internet-based data transfer. However, in order for multimedia over IP networking to be sent, received, and played back successfully with a satisfied quality of service degree, certain conditions must be met [3] [6]. *First* and foremost, packet sequencing is necessary to ensure that multimedia messages are appropriately constructed at the recipient even in cases of rerouting. Out-of-order decoding must also be finished, and any missing packets must be identified. Time-stamping and buffering are the *second* technique; the former is used to specify the precise moment of playback, while the latter reduces delay jitter by buffering and reordering at the receiver prior to playing the pack. *Third*, specifying the packet's payload type so that encryption and media interpretation can be applied correctly by the receiving end.

The detection and correction of packet faults during multimedia streaming is accomplished through error concealment, which is the fourth prerequisite. The minimal QoS level and/or service class needed to maintain a continuous multimedia session between senders and receivers are specified in the fifth criteria, known as the QoS level requirements. Finally, in order to prevent network congestion, rate control rules and changes based on bandwidth specifications and network capacity are needed. While QoS gives the network the chance to offer various applications specific service levels based on the bandwidth that is available to meet their needs for jitter, end-to-end latency, and throughput while maximizing the use of the network's resources [3] [6].

Delay jitter: the difference in time between two successive packets within the same flow is known as delay jitter. By employing appropriate streaming algorithms and enough buffering, delay jitter can be significantly decreased.

End-to-end delay (latency) the gap between the packet's sending time at the source and its receiving time at the recipient is known as the end-to-end delay, also known as latency. Processing delay, queuing and rerouting delays, propagation delay, and transmission delay are all included in the end-to-end delay.

Bandwidth: a network's capacity to carry data, often expressed in bits per second (bps). Nevertheless, the underlying physical layer technologies in use determine the bandwidth.

Throughput: the quantity of successfully transferred error-free data between a sender and a recipient per unit of time is known as throughput (bps). The error rate, network load, and network access mode (carrier sensing, for example) all affect throughput.

Packet loss: the quantity of packets lost during the network transmission process is known as packet loss, and it is expressed in bits per second (bps). On the other hand, network congestion, problems at lower layers, malfunctioning network devices, or application errors can all cause packet loss

Protocols Employed in This Paper

In this section the engaged protocols are briefly illustrated.

A. RSVP protocol

RSVP is a receiver-oriented service protocol that lets users choose their preferred QoS level, makes reservations, and keeps them active for the duration of the reservation. RSVP is able to manage rerouting and membership changes, and it can operate with heterogeneous receivers by taking into account both multicast and unicast transmission types. Since receivers in heterogeneous multicast groups have varying capacity and QoS levels, heterogeneous receivers can use RSVP to request varied QoS that meets their unique needs while staying within their capabilities. Additionally, senders have the option to divide traffic into many flows, each of which is confined within a distinct RSVP flow with a set QoS level. Each RSVP flow in the RSVP working environment is homogeneous, and recipients have the option to join one or more flows. "Traffic control" techniques work together to implement QoS to a specific stream flow [4-7]. These means include a packet scheduler and any link-layer system to decide what time specific packets are dispatched, admission control, and classifying unit of packets. For certain streams, a definite path with associated QoS class are identified by the classifying unit. The admission control module assesses if the node's resources are adequate to meet the required quality of service standards [4–7].

This is a brief summary of how the RSVP protocol works: Path (PATH) and reservation (RESV) messages are the two primary messages that form the basis of the RSVP protocol mechanism. Senders and recipients exchange these messages over the chosen path. PATH messages are only sent in RSVP environments, where recipients have to join a designated multicast group. All destinations in the multicast group get RSVP protocol path messages, which are generated by sender hosts and relayed on a regular basis via distribution tree. Sender-to-network signaling, on the other hand, uses path messages to transmit routing information from the reverse path to the sender and to relay sender information [7].

Alternatively, RSVP protocol reservation messages are receiver-to-network signalling. They reserve resources by flowing upstream from receivers to senders, adding more receiver-related states at routers along the chosen path, and generating soft states. To preserve the reservation states over the routers along the path, the RSVP control must, nevertheless, send refresh messages on a regular basis. Consequently, the reserve state will end if the refresh message not acknowledged in the interior of assured time interval. A unicast destination address, or multicast group, the refresh time, the previous hop of the upstream router/host ID to guarantee path building, and the flow specification—which details bandwidth requirements for the flows targeted by the reservation—are all included in the PATH message. RESV messages originate from the receivers. At each node, reservations are formed for one or more senders via the RESV messages, which travel the exact opposite path of PATH messages. To maintain the reservation states, the RSVP control was implemented, requiring periodic refresh signals to be sent. Therefore, if a refresh message not arrived by a definite amount of interval, reservation state will be cleared. Filter and flow specifications are among the reservation parameters included in the RESV messages. What packets in the flow the packet classifier should employ is described in the filter specification. The packet scheduler uses the flow specification, whose content is determined by the service [4–7].

B. SIP protocol

For online multimedia sessions, signal and control protocol (SIP) is utilized. SIP protocol offers phone and video conferencing over the Internet or online immediate sms messaging. Hence, SIP protocol specifies the necessary steps and operations for initiating the communication session and describes the particular message formats that are sent in order to create, maintain, and end multimedia sessions. The SIP may work with the UDP, TCP, and Stream Control Transmission Protocol (SCTP) because it works without any concerns with any underlying transport protocol. In addition, SIP protocol uses text description of messaging that used in SMTP and HTTP protocols. SIP works with several protocols to identify and transfer the data from the session media. Session Description Protocol (SDP) prepares the media description and session negotiation settings, while Real-time Transport Protocol (RTP) or Secure Real-time Transport Protocol (SRTP) transports multimedia streams (voice, video). Furthermore, multimedia sessions using the RTP protocol benefit from the improved QoS provided by RTCP. SIP is a centralized form protocol in which servers are crucial to call registration and session command forwarding until the necessary call recipient is located. As a result, SIP servers receive and process SIP queries.

A SIP server is an application that can provide guidance or information to a client on behalf of a SIP client. Proxy, redirect, and registration are just a few of the many varieties of SIP servers. The proxy server is not the source of SIP requests; rather, it is in charge of sending requests to establish sessions to one or more servers, creating a search tree to locate the necessary SIP users, and receiving responses to the sending user entity to create the multimedia session's path. The purpose of a redirect server is to send users to other servers and transmit communications to other servers in an attempt to locate the caller; alternatively, it may use the Domain Name System (DNS) to locate the necessary user information and return it. Registrar server receives registrations concerning the up-to-date user locations [3] [6].

A. H.323 Protocol

A signalling protocol that resembles the SIP protocol is H.323. As a standard protocol for initiating, regulating, and ending multimedia communications, it was created by the ITU-T organization. Usage reporting, caller identification, alias mapping, improved bandwidth management, enhanced fax functionality, and the ability to tunnel other protocols are some of the characteristics of H.323 protocol. However, H.323 protocol involves collection of protocols working together to carry out the multimedia session; for the multimedia sessions to be managed, the H.323 terminal must support these protocols. In order to make an associated link among H.323 calls, needs a H.225 protocol to provide call initiation and required call control procedures. The Registration, Admission, and Status (RAS) protocol is managing endpoint and gatekeeper registration, admission control, bandwidth modifications, state information, and termination processes. The H.323 endpoint's functioning is governed by end-to-end messages of control that are exchanged using H.245 control signalling. Within the H.323 protocol suite, multimedia packets are sequenced using RTP/RTCP. [3] [6].

B. MPLS Protocol

By attaching MPLS headers to IP packets, MPLS is an improved method of packet forwarding across networks. The label value and the Exp value, which indicate the QoS within the MPLS packet, are fields found in the MPLS header. Nonetheless, the forwarding procedure is altered to be rapid and effective by basing the decision about the packet's forwarding on the labels present inside MPLS header instead of reading of entire IP address of the packet in addition to consulting the routing table [14, 15, 16]. Furthermore, by assigning each packet in a particular flow the same label, the MPLS header guarantees that every packet in that flow follows the same path across the network's backbone path [17] [18]. Also such packets are forwarded through the same path as well as these packets provided the suitable QoS class which especially enhances the handling of the real-time multimedia streaming (voice and video) [1-2].

Experimental Study

The accomplished simulation's run [9] comprise four scenarios, each of which is represented by the initial layout example given in Figure (1) and which takes into consideration an identical network topology. As a result, the experimental study conducted in this paper uses VoIP over the evaluated system to send signals from an arbitrary beginning node to arbitrarily locations in various MPLS network instances employing two distinct VoIP signalling protocols—SIP and H.323 utilizing RSVP protocol. VoIP traffic defined through the "create traffic flow" Option of OPNET tool, as seen in Figure (2) with the following input parameters [19]:

- Call rate transferred into the network: 500 calls per an hour for every single scenario which produced about 272 voice traffic streams.
- Call duration in average: 300s (5 min).
- Voice flow duration: 3600s (60 min).
- VoIP encoder: G.711.
- Type of Service: described as interactive voice (6) with measuring of delay, throughput, and reliability, together with RTP/UDP/IP protocols overhead (bytes). Thus, the implemented network case study involves 4 scenarios as follows:
    - a- Scenario 1 is RSVP only.
    - b- Scenario 2 is RSVP with MPLS.
    - c- Scenario 3 is RSVP with MPLS and H.323.
    - d- Scenario 4 is RSVP with MPLS and SIP.

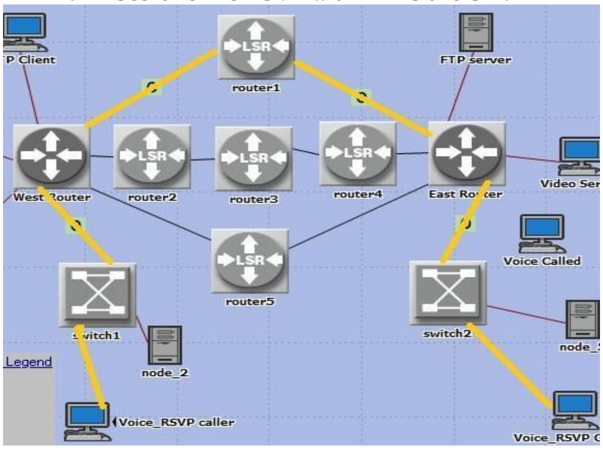

Fig. 1. The case study topology with RSVP path selected

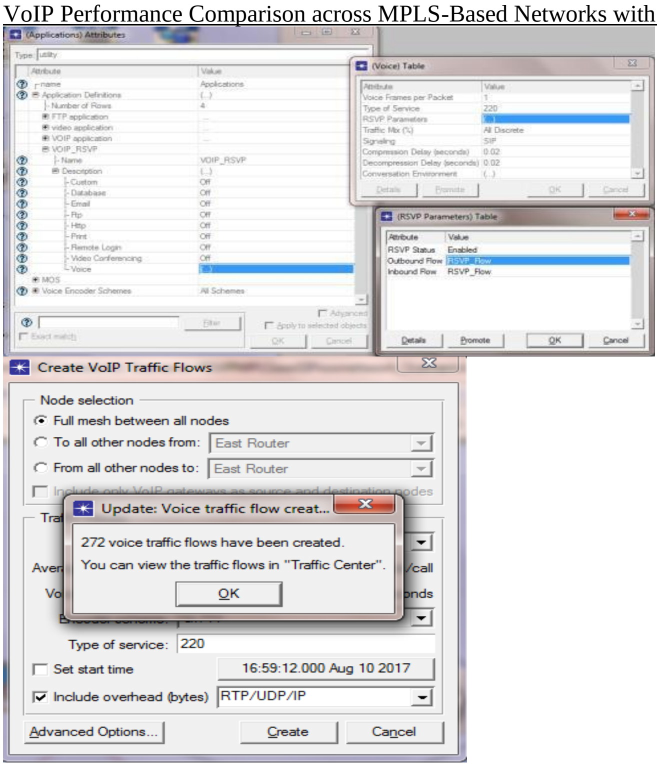

Fig.2. VoIP application configuration (SIP protocol and type of service)

The configuration of the Hosts and Routers with RSVP protocol using OPNET tool is seen in Figure (3).

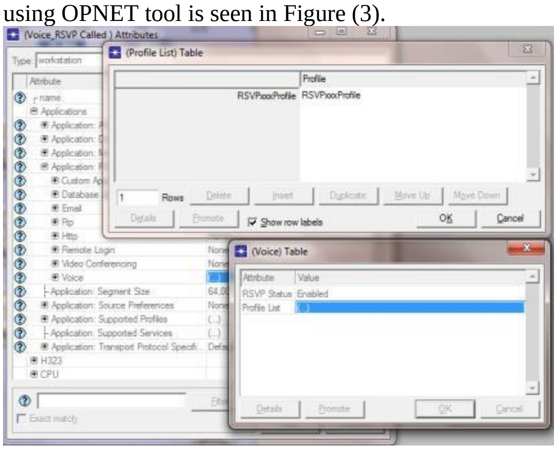

Fig. 3. The configuration of RSVP protocol

Simulation runs

Figure (4) shows the data used during the simulation run which have been generated by the traffic generator in OPNET and supplied to the case study for one hour simulation duration. The Figure (4) demonstrates a traffic measured in Giga bytes values (maximum value around 490,000 packets per second) which is calculated as 174.501 GB in total as VoIP traffic which indicates that the scenarios' performance are tested with a very massive data to exemplify a competency of applying MPLS networks with multimedia streaming applications.

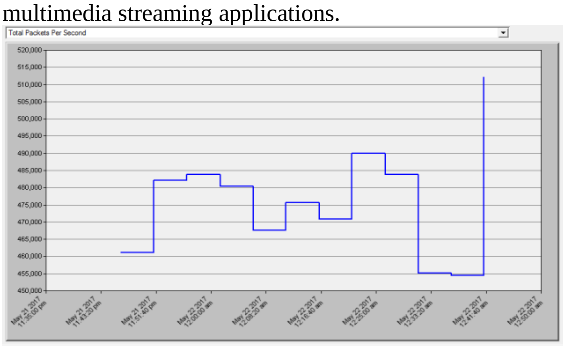

Fig. 4. The quantity of the VoIP data injected into the case study

subsequent performance metrics: VoIP Jitter (seconds), packet End-to-End latency (seconds), call setup time, and Throughput (bits/second) are employed to assess the efficacy of various protocols across distinct circumstances.

VoIP jitter

When utilizing SIP and H.323 voice signaling protocols, Figure (5) shows the voice jitter of the RSVP protocol over MPLS communication. The SIP protocol performs better in this performance metric than H.323 with MPLS networking as its jitter is lower (greater) with SIP than it is with H.323.

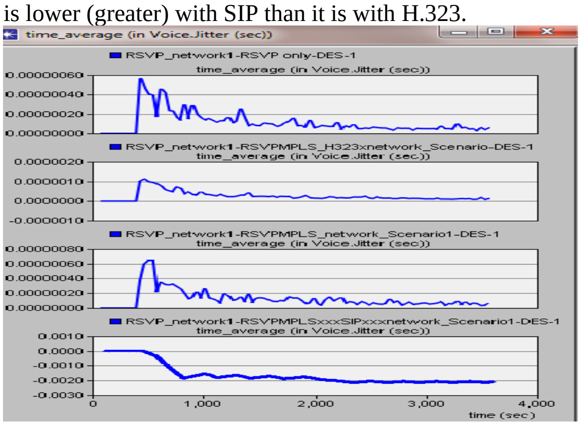

Fig. 5. Voice delay Jitter (Sec)

VoIP end-to-end delay

- Figure (6) demonstrates the end-to-end VoIP latency while employing SIP and H.323 voice signaling protocols, alongside the RSVP protocol and MPLS networks. The H.323 protocol is superior in this performance metric as it has significantly less end-to-end latency (less than 0.10 sec) than SIP, compared to around 3 seconds with MPLS networking. The end-to-end delay of SIP is much greater (worse) compared to the delay of the H.323 protocol.

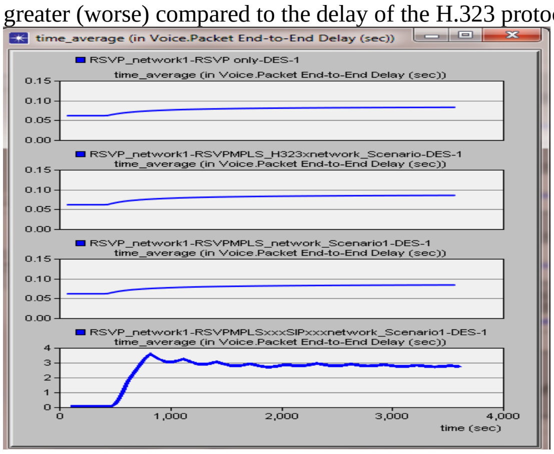

Fig. 6. VoIP packet end-to-end delay

Active connection count

Figure (7) illustrates the TCP connections that are open which correspond to the MPLS networking session configured for a VoIP service using the RSVP protocol, SIP, and H.323 signaling protocols. Consequently, H.323 recorded a higher TCP active connection count number than SIP, resulting in the smallest number of connections that are active.

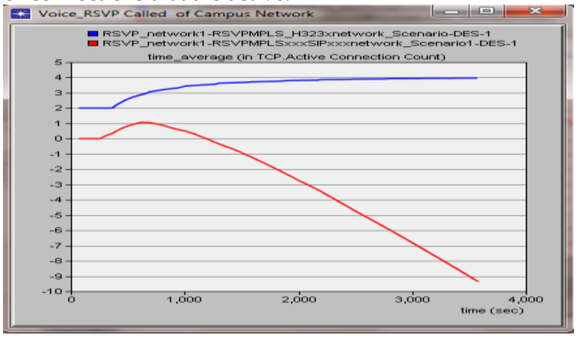

Fig. 7. TCP. Active Connection Count: RSVP with MPLS using SIP (Red) and RSVP with MPLS using H.323 (blue).

Number of path states of RSVP protocol

Figure (8) illustrates the number of path states of RSVP protocol with the four scenarios of MPLS network using (SIP, H.323). The figure shows that in the case of RSVP with MPLS using (H.323) has higher number of path states of RSVP while in the case of

RSVP with MPLS using (SIP) has reported the lower value of the number of RSVP path states.

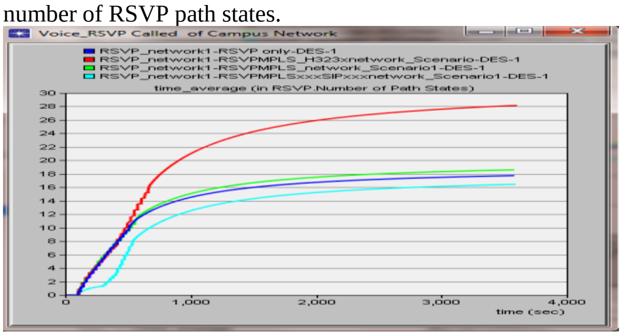

Fig. 8. RSVP. Number of Path States (packet/ sec) : RSVP only (blue) , RSVP with MPLS using H.323 (red) , RSVP and MPLS (green) , RSVP with MPLS using SIP (aqua).

Number of reservation states of RSVP protocol

- Figure (9) shows the number of reservation states of RSVP protocol in MPLS networks with SIP and H.323 signaling protocols of VoIP. However, the figure clearly describes that the number of the RSVP reservation states is almost the same in all the scenarios.

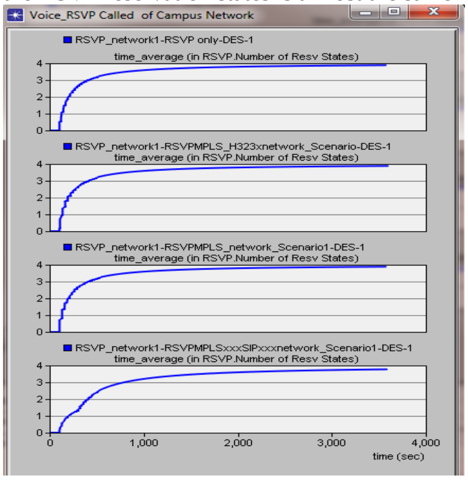

Fig. 9. The number of RSVP of Reservation States with the four scenarios

Call setup time (sec)

Figure (10) describes the call setup time of VoIP with SIP protocol using RSVP protocol with MPLS networks. However, the highest recorded call setup time of SIP is 0.000040 sec, and the lowest SIP call set up time value 0.000006 sec which is an acceptable value for initiating VoIP session using SIP protocol.

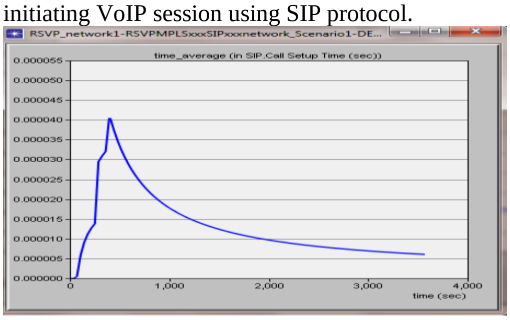

Fig. 10. SIP Call setup time (Sec) with RSVP protocol in MPLS networks

Active calls

Figure (11) shows the active calls of VoIP made by clients of SIP protocol using RSVP protocol with MPLS networks. The highest recorded value of active calls with SIP is up to 0.023 sec, and the lowest value is 0.004 sec of course as a percent from the total number of applied VoIP calls to scenarios.

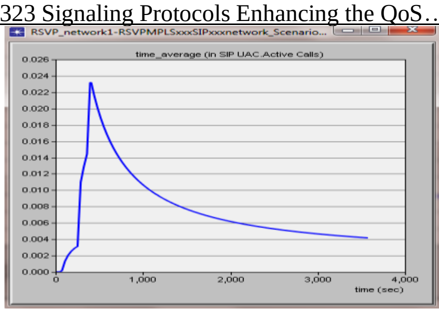

Fig. 11. Active Calls of SIP UAC (Clients)

Incoming calls

Figure (12) illustrates the average time of the incoming calls made by the clients of VoIP with SIP protocol using RSVP protocol with MPLS networks. However, the figure shows that the incoming calls of SIP clients recorded the highest call value up to 0.0023 sec while the lowest call value is 0.0005 sec.

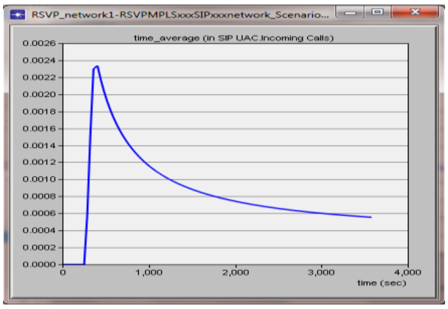

Fig. 12. Incoming Calls of SIP UAC (Clients)

Calls initiated by VoIP clients

Figure (13) demonstrates the average time of the calls initiated by the SIP clients using RSVP protocol with MPLS networks. However, the figure describes that the highest value of the call setup time is about 0.0079, where a smallest rate about 0.0031 as of a total value from the injected calls to the SIP scenario.

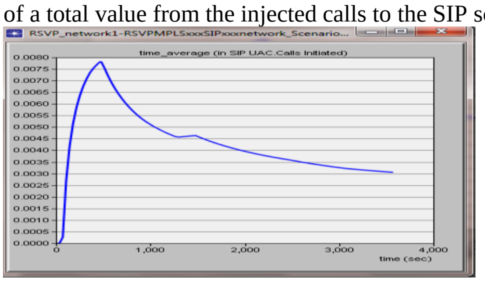

Fig. 13. Calls initiated by SIP UAC (Clients)

Result of H.323 Protocol

<u>H.323 Setup Time</u>

Figure (14) shows the average setup time made by the H.323 protocol to initiate calls with RSVP protocol on MPLS networks. However, the figure demonstrates that the highest value is of 0.0043 sec which is a very reasonable time to make VoIP calls with MPLS networking.

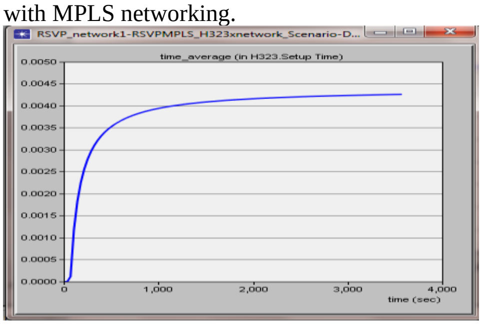

Fig. 14.H.323 average call Setup Time

Point-to-point Throughput

Figure (15) describes the VoIP Throughput with RSVP protocol and MPLS communication network, while employing the SIP and H.323 as signaling protocols for VoIP application. However, results shown in figure for the four scenarios -for some specific nodes- are very close saying that the all scenarios are working properly.

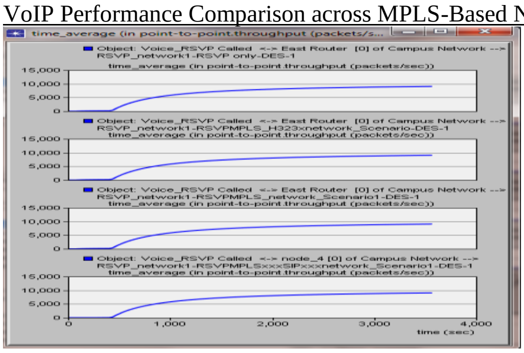

Fig. 15. Point-to-Point Throughput

Results

In this study, the effectiveness of SIP and H.323 protocols for signaling with VoIP performance evaluation was evaluated in terms of packet delay jitter, end-to-end latency, and throughput over MPLS-based infrastructures using RSVP as a call for quality of service.

The paper is produced the following results: -

(a) Delay jitter of VoIP for both signaling protocols

With MPLS and RSVP protocol, the SIP outperformed H.323 in terms of voice jitter delay.

a) End-to-End Delay of for both signaling protocols

When it came to end-to-end latency, H.323 outperformed SIP with MPLS and RSVP protocols. When using MPLS networking, the H.323 protocol has an end-to-end latency value of less than 0.10 seconds, compared to around 3 seconds for SIP.

b) Call setup time of both signaling protocols.

With call setup time of VoIP with SIP protocol using RSVP protocol over the MPLS networks is as follows:

The highest call setup time value of SIP is 0.000040 sec.

The highest call setup time value of H.323 is 0.0043 sec. Making SIP is better in call setup required time.

Conclusion

When transporting VoIP applications, H.323 is a viable option for MPLS with RSVP for businesses needing minimal end-to-end latency. Conversely, SIP is a strong contender for MPLS with RSVP in VoIP applications for businesses that need very low jitter values and a fast call setup time. By extending the study to compare the OSPF routing protocol with SIP and H.323 over MPLS networks using RSVP as a QoS tool, more research may be done.

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