Optimizing Voice over Multi-Protocol Label Switching (VoMPLS)

Amer Alkayyal [1], Stelios Sotiriadis [2,1], Eleana Asimakopoulou [1], Nik Bessis [1]
[1]School of Computing & Maths, University of Derby, Derby, United Kingdom
[2]Department of Electronic & Computer Engineering, Technical University of Crete (TUC), Chania, Greece
a.alkayyal3@unimail.derby.ac.uk, s.sotiriadis@intelligence.tuc.gr, eleana.asimakopoulou@googlemail.com, n.bessis@derby.ac.uk

***Abstract*— Today, the Voice over Internet Protocol (VoIP) is a very growing service because of the increasing utilization of IP networks in modern enterprises. Its major benefit is the reduction of the operational and infrastructural cost that is vital for disaster management planning. This paper presents a literature review on voice transmission techniques over IP networks and discusses the advantages of the Multi-Protocol Label Switching (MPLS) backbone for transmitting voice packets. This could be a case of an emergent communication case where there is a need for utilizing available infrastructure along with the VoIP protocol. We focus on the identification of the constraints that affects the quality of service provisioning to the end users and we suggest a set of modifications to improve it. Further, we propose a plan to develop a script format to achieve optimization of voice packet (real-time) transmission performance over a MPLS based network.**

Keywords— Voiceover Internet Protocol (VoIP), Multi-Protocol Label Switching (MPLS), real-time voice packet optimization

I. Introduction

Enterprises continuously demand better and less costly communication services regarding inter-enterprise and intra-enterprise telecommunications. There are commercial advantages in utilizing VoIP service in an enterprise not just in terms of infrastructure costs, but also in terms of network management. Enterprises are depending on VoIP service in a very growing rate. “The usage of VoIP service reduced a huge amount of cost for their infrastructure” [9]. Transferring voice packets over a public network such as Internet has come with some drawbacks such as delay. This type of packets must be processed in real time to be useful. A certain set of standards has been created to raise the quality of the VoIP service. At the same time a new infrastructure solution was developing ‘MPLS’ which can improve the network by reducing packet process time and empowers QoS. [2].

This paper encompasses four main sections. The first section presents the need of the VoIP technique and the reasons to choose the new solution to transmit voice packets VoMPLS. The second section performs an overview on the voice transmission fundamentals and proposes the identification of the voice transmission service parameters; Quality of Service (QoS) requirements and the methods to provide a sustainable and a high quality voice service over an IP based network, then over a MPLS based network. The third section contains a working plan of identifying the key factors and parameters that affect VoMPLS performance and suggestions to modify them. Then suggesting a MPLS-based model depending on those modifications to provide voice service and achieve improved QoS. Further in the plan a structure of a script (set of instructions) can be proposed to be applied on the network equipment to achieve the aim of improving voice service running on MPLS-based providers. The fourth and last section includes the study conclusion and proposed future work.

II. THE NEED FOR VOICE TRANSMISSION OVER IP NETWORKS

IP technology is dominating over every communication system in the globe because it supports a vast variety of services and it’s compatible with almost everything. Because of that, terms such as Everything over IP (EoIP), ALL IP and Next-Generation Network (NGN) started to appear in the network world. “The future of voice communications, much like most everything else, is going with some type of Internet Protocol (IP) implementation.” [1]. NGN is a telecommunication core and access network architectural change and its idea is that one network can be able to transport all types of data and services by encapsulating them into packets, similar to those used on the Internet.

There are a lot of important benefits that an enterprise can gain by implementing VoIP service [5] with regards to disaster management:

- VoIP service is charged differently than the traditional phone service, there are no charges on the connection itself because the use of IP infrastructure and a VoIP call is charged by the second. Additional to that, a VoIP call is free of charge if it was between two extensions in the same company even over Internet.
- Reducing the amount of required lines to operate the telephony service, because VoIP enables the feature of configuring several internal virtual numbers over a single Internet connection. This could be particular useful for cases of disaster management [12].
- Adding new VoIP equipment is simple because IP Phones runs over Ethernet, and every environment could have an Ethernet infrastructure. This could be of great assistance in disaster management situations [16].

978-0-7695-5094-7/13 $31.00 © 2013 IEEE
DOI 10.1109/3PGCIC.2013.79

- Managing the voice service in a computerized environment can offer a lot of features and control over it. For instance, this could extend the traditional message exchanging solutions [17] used in large-scale settings to include new protocols. This includes the management of voice services in terms of cloud services. In particular a cloud provider could offer such services on demand [13] and could be utilized based on various situations.

The main issue with VoIP was the difficulty of delivering high service quality. On the other hand, a new kind of technology 'MPLS' is emerging in the networking domain that can optimize the voice packets transmission and provide a better service. There are multiple reasons behind choosing MPLS to run on several service providers' backbone [5]:

- Compatibility: MPLS is compatible with multiple protocols and infrastructural technologies such as IP in on hand, and ATM.
- Scalability: Several issues related to IP over ATM/FR overlay can be overcome by the use of MPLS technology.
- IP QoS: MPLS uses several methods to ensure and guarantee QoS such as end-to-end traffic engineering and load balancing.

"The different QoS requirements of voice traffic can be met by using MPLS in conjunction with DiffServ, proper traffic engineering, and other techniques" [8]. The main challenge of transmitting voice packets is the need of processing them in real time. In other words, if there was a delay in delivering the information, it will be useless.

Transmitting voice is a real time application that requires critical factors to reach the customer predicted performance. Those factors are reliability and minimizing congestion which a new technology as MPLS can simply meet.

In conclusion, MPLS is being adopted as a key protocol within service providers and carrier networks because of:

- Traffic Engineering.
- Flexible signaling plane which support service differentiation.
- Flexible path definition (including redundant paths for security functions).
- Ability to transport almost anything (i.e. voice over IP or directly over MPLS).
- QoS techniques implemented with Diffserv [10].

Additional to all of that, MPLS can achieve several advantages that can affect the transmission of voice traffic [6]:

- During network instability, the number of dropped packets can be decreased.
- Service reliability is increased.
- The ability to assign priority flags to specific type of traffic without affecting other types.
- Quickly adjust to traffic flow changes and rerouting traffic to avoid a congested route while applying physical topologies upgrades.
- Meet the demands of customers' with high performance.
- Ultra-fast forwarding.
- Virtual Private Networks (VPNs).
- Implementation of tunnels and trunks.

All of above points out that transmitting voice packet over a MPLS backbone is more efficient than transmitting them over a traditional IP backbone. "It was clear from the simulation that MPLS model had a better overall performance for voice traffic transmission and would have better utilization." [9].

III. OVERVIEW ON VOIP & VOMPLS

When considering transmitting voice packets over an IP based network, there are two main protocols to transfer any kind of payload over this kind of networks. The first one would be the Transmission Control Protocol (TCP), but the issue with TCP that is quite slow because it doesn't offer much bandwidth for the payload since the header is large, and it's a connection oriented protocol. On the other hand, the User Datagram Protocol (UDP) is the more suitable protocol to accomplish the task, because UDP is fast compared with TCP, connectionless and has a small header which will offer more bandwidth to the payload. UDP is not designed to retransmit the packets that may be dropped or lost on the path to the destination [1].

By using UDP, packets will still be formed in the IP stack methodology, which might result an issue in the received packets order. Therefore a mechanism needs to be applied to ensure the packet stream reliability. Receiving the last half of a conversation before the first will be unusable. The Real-time Transport Protocol (RTP) can achieve the task of reordering the received packet to the order that they were sent by the source. Additional to that, there are extra voice compression methods that can improve the voice transmitting service by regenerating lost packets [1], [11].

When two users want to initiate a VoIP session, some information is required to be exchanged between these two users before the session starts. The protocol that handles that information exchange is the Session Initiation Protocol (SIP) which has a huge impact on the VoIP service. After all the previous, a voice packet will go through multiple protocols to eventually establish and maintain a high quality call, and they are in order: SIP, RTP, UDP, and IP. These protocols will get involve with several parameters that represent the voice packets transmission QoS. At the sender side, the speech samples are encoded and compressed into a frame then sent on the carrier media. On the receiver's side, the adverse process is done. The receiver will receive the encoded frames and regenerate the original speech samples. This process is done to meet the requirements of delivering voice service from one point to another.

The combination between header compression techniques and voice codecs can result of major enhancement in bandwidth consumption. For example, there will be a consumption of 20 gigabits per second of uncompressed headers when using the G729 codec to transmit 200 million calls per day, where the whole amount of bandwidth reservation is nearly to 36 gigabits per second [1]. Even there were substantial efforts to improve the VoIP services, there was a major issue of guaranteeing the QoS offered by the IP based networks, and that was the main drawback of VoIP services. Alongside with that, there was an emerging technology that can empower QoS in multiple applications such as voice transmission.

MPLS stands for Multi-Protocol Label Switching, it is clear from the name that this new approach is able to work with multi kinds of protocols and existing infrastructures such as ATM, Frame Relay, PPP and HDLC with higher compatibility. It depends on a new method to preform packet forwarding called label switching. It reduces the process time of transporting a packet by making the routing decision based on a simple label instead of the whole IP prefix look up. MPLS is very flexible in its signaling plane, which convinced a lot of voice service providers to apply this new technology on their backbone. MPLS strongly support VPNs, virtual trunks and tunnels and it is capable of achieving high levels of traffic engineering. That will enhance the QoS in traffic streams, and eventually will empower the voice service transmission and also optimizes network resource usage [3].

A router that supports MPLS is called a label switch router (LSR), and there are several types of LSRs depending on its position in the network [6]. The ingress LSRs: this type receives a non-labeled packet, insert a label on top of the packet, and send it on a data link. The egress LSRs: this type receives labeled packets, takes off the label, and sends them on a data link. The previous two types are routers that are positioned on the network edge. The intermediate LSRs: this type receives an inbound labeled packet, perform forwarding process on it, and send the packet on the correct data link. Each one of these routers is involved in voice packets transmitting process. After MPLS is applied on the backbone, multiple data stream transmitting methods (voice packets) can be configured between the previous backbone routers.

Those methods are [4] the transporting VoIP packets over an MPLS tunnel and the transporting voice frames over MPLS backbone directly. This method is not inefficient because the voice packet needs to go through many protocols as mentioned before, which creates a major drawback. This method is called VoIPoMPLS. The other preferable method is transporting voice frames over the MPLS backbone directly and it is called VoMPLS. Table 1 shows a comparison between VoMPLS and VoIP and why VoMPLS is preferable. "VoMPLS provides a very efficient transport mechanism for voice in the MPLS environment" [4]. Even though VoMPLS is the preferable method to transmit voice over a MPLS backbone, there are drawbacks in this method especially when the provided bandwidth is not constant such as a wireless connection.

IV. TOWARDS AN OPTIMIZED VOMPLS

Because of the traffic type (voice), there are two major areas that need attention to optimize this service. The first area is Traffic Engineering (TE) because it is the dynamic technique to optimize a network performance by evaluating and predicting the transmitted traffic behavior over that network. The second one is Quality of Service (QoS) which refers to multiple aspects of telephone services and computer networks to handle the transportation of traffic with special requirements. MPLS doesn't define a new QoS architecture; it supports the Differentiated Services (DiffServ) architecture. It also supports TE significantly because it provides most of the overlay model functionality. The flowing points describe the voice transportation procedure directly over MPLS [3].

Initially, the multiplexing structure of a voice packet transmitted over a MPLS backbone must contain three major fields: outer label, inner labels and the VoMPLS primary sub-frame that contain a header of a size of 4 octets or less. Secondly, each VoMPLS primary sub-frame can be linked with several voice connections to increase throughput. Thirdly, the packet primary payload can be a sequence of encoded voice sub-frames or a single Silence Information Descriptor (SID) sub-frame. Also, the length field size is 4 octets and it is located in the primary sub-frame header. Therefore, there is enough space to insert three padding octets in each sub-frame.

Table 1. A comparison between VoMPLS and VoIP [10]

	VoMPLS	VoIP
QoS	Comprehensive QoS on aggregated traffic.	Scalable with qualitative guarantees or non – scalable with quantitative guarantees.
Signaling	CR – LDP and RSVP - TE	Non – scalable.
Access network	Software upgrade to customer IAD.	Integration with CPE IP applications. Standard for cable networks.
Backbone network	Traffic engineering Interworking with IP/DiffServ. Implementable over ATM.	No quantitative scalable QoS.
Bandwidth efficiency	Very high with PPP.	Requires RTP multiplexing. Header compression improves efficiency in the access network.

The essential voice traffic is included in the primary payload, and a Channel Identifier (CID) identifies it. The primary payload is a flexible length sub-frame and it contains encoded voice and SIDs. To support the primary payload, control sub-frames can be sent along side with them but they are not included in the same multiplexing frame. The control sub-frames belong to the MPLS signaling plane. Each Label Switched Path (LSP) in the MPLS backbone can transport several numbers of voice calls. Finally, 248 VoMPLS calls are allowed to be included in the header of the CID and to be multiplexed in the same LSP.

There are three parameters that affect VoMPLS traffic and impact on the main parameters (mentioned in table 1). Firstly, the header size; this parameter affects the service because it reserves bandwidth on the link. Usually it size is 4 octets in VoMPLS, secondly, the packet-packaging scheme. Usually the packaging scheme states that one SID is carried by one sub-frame. Thirdly, the padding octets; these are managing octets that add extra load on the SID frames and voice generated packets. After identifying these parameters, the suggested methodology. Firstly, to minimize the header size as much as possible to reduce bandwidth reservation. Also, to apply some structural changes in the packaging scheme to enable the labeled IP packet to carry multiple voice calls and SID frames. Another way is to reduce the number of padding octets to reduce the load on the transmitted packets. Finally, to increase the number of VoMPLS calls that can be included in the same LSP by changing the specifications of the CID without affecting the QoS.

Depending on the modified parameters and the previous procedure, a virtual model can be built and tested. The test phase includes several observations and evaluations on the main parameters of voice service. Thy will be repeated several times to achieve the ultimate goal of delivering a high quality call over the MPLS backbone without glitches, delays and disconnects. The final phase is translating the results of the last testing stage and imbedding the optimized parameter values into a script to be applied on the MPLS-Backbone equipment.

V. CONCLUSION AND FUTURE STEPS

The current efficient solution to meet the demands of enterprises all over the globe regarding voice transmission between their branches is MPLS. That's why tuning and optimizing voice traffic over MPLS backbone is a major interest to researchers and developers. There are two main approaches to transmit voice traffic over a MPLS backbone, VoIPoMPLS and VoMPLS. This paper showed that VoMPLS is the better approach and it is the preferred one for many reasons. Depending on that, it proposed a work plan to optimize this approach and to deliver a better QoS by suggesting enhancements to some parameters that affect this approach. In the future these enhancements could be applied on the parameters and conclude the structure of a virtual model that can be built and tested and monitored to tune up the delivery of voice traffic on MPLS backbone. At the final stage all this information can assist in building a script that could be applied to the MPLS equipment directly to achieve high quality voice transmission on the network. A future aspect of this work is the exploration of communication algorithms and protocols with regards to optimization of communication of emerging infrastructures [14] e.g. inter-clouds. This could be adoptable by the VoMPLS solution. For example the work of [15] includes an analysis of requirements for such environments including inter-enterprises.

REFERENCES

[1] Alawieh, B. (2007). Efficient Delivery of Voice Services over MPLS Internet Infrastructure. IEEE, 483-486.

[2] Álvarez, S. (2003). QoS in MPLS Networks. Cisco Systems.

[3] Estepa, A., Estepa, R., Campos, I., & Delgado, A. (2009). Dimensioning Aggregated Voice Traffic in MPLS. University of Seville.

[4] Estepa, A., Estepa, R., Vozmediano, J., & Delgado, A. (2011). Traffic Engineering Parameters for VoMPLS. C/ Camino de los descubrimientos, 151-161.

[5] Fjellskål, E. B., & Solberg, S. (2002). Evaluation of Voice over MPLS (VoMPLS) compared to Voice over IP (VoIP). Masters Thesis, Agder University College, Agder.

[6] Ghein, L. (2007). MPLS Fundamentals (1st ed.). Indianapolis: Cisco Press.

[7] Hickorytech. (n.d.). Managed MPLS-VPN. Retrieved April 15, 2103, from Hickorytech: http://www.hickorytech.com/business-services/data/mpls-vpn/managed-mpls-vpn.aspx

[8] Jdsu. (2010). Application Note (Testing VoIP on MPLS Networks). Jdsu.

[9] Naoum , R. S., & Maswady , M. (2012). Performance Evaluation for VOIP over IP and MPLS. World of Computer Science and Information Technology Journal (WCSIT), 2(3), 110-114.

[10] Subirats, L. (2009). State of art of IP telephony and MPLS. Telematics Engineering Dept., Technical University of Catalonia, 1-9.

[11] Zubairi, J. A. (2008). Voice Transport Techniques over MPLS. IEEE, 25-29.

[12] Asimakopoulou, E. and Bessis, N. (eds 2010) Advanced ICTs for Disaster Management and Threat Detection: Collaborative and Distributed Frameworks, IGI ISBN: 978-1615209873

[13] Sotiriadis, S., Bessis, N., Xhafa, F. and Antonopoulos, N. (2012) From Meta-computing to Interoperable Infrastructures: A Review of Meta-schedulers for HPC, Grid and Cloud, 26th IEEE International Conference on Advanced Information Networking and Applications (AINA-2012), Mar 26-29, Fukuoka ISBN 978-1-7695-4651-3 pp: 874-883

[14] Sotiriadis, S., Bessis, N. and Kuonen, P. (2012). Advancing Inter-cloud Resource Discovery based on Past Service Experiences of Transient Resource Clustering, 3rd International Conference on Emerging Intelligent Data and Web Technologies (EIDWT-2012), September 19-21 2012, Bucharest, Romania, ISBN: 978-0-7695-4734-3, p.p.: 38-45

[15] Bessis, N., Sotiriadis, Cristea, V. and Pop, F. (2011). Modelling Requirements for Enabling Meta-Scheduling in Inter-Clouds and Inter-Enterprises, Third IEEE International Conference on Intelligent Networking and Collaborative Systems (INCoS-2011), Fukuoka, Japan, November 30 – December 2, 2011, ISBN: 978-0-7695-4579-0, p.p.: 149-156.

[16] Bessis, N., Asimakopoulou, E., and Xhafa, F. (2011) A Next Generation Emerging Technologies Roadmap for enabling Collective Computational Intelligence in Disaster Management, International Journal of Space-Based and Situated Computing, InderScience 1(1) ISSN: 2044-4893 pp: 76-85 DOI 10.1504/IJSSC.2011.039109

[17] Bessis, N., Sotiriadis, S., Pop, F. and Cristea, V. (2013). Using a Novel Message-Exchanging Optimization (MEO) Model to Reduce Energy Consumption in Distributed Systems, Simulation Modeling Practice and Theory, Elsevier, ISSN: 1569-190X