IPTV-Oriented Network Architecture

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Abstract

Although it emerged as a means to improve switching speed at the core network, MPLS has spread rapidly as a very flexible and robust solution for network providers to enable triple play services and reduce costs at the same time. As network complexity grows and a higher scalability is required to meet customer demands and give them the necessary Quality of Experience, more advanced features and functions must be included at the 'brains' of the network. To this end, a more automated and video delivery oriented path reservation mechanism based on RSVP-TE has been developed and tested in a lab environment with real-world ISP network equipment. We have added additional video intelligence capabilities to the network using Juniper routers and taking advantage of the JUNOS SDK for third-party developers. Finally, some measurements have been taken to validate our IPTV-oriented architecture, showing that it is possible to achieve a reasonably better bandwidth utilization by using the proposed video LSP reservation scheme.

Keywords: IPTV, MPLS, QoS, RSVP-TE, video streaming.

Introduction

Telecom service providers tend to deploy more and more services over their managed IP networks. The high growth rate of new services, and especially high-bandwidth services such as IPTV, has caused that this process of convergence of multiple services over a single IP network seems unavoidable, and requires of both saving costs and looking for more flexible transport solutions that consolidate all the services and maximizes the efficiency of the investments.

However, the increasing video-related traffic will have as result a mismatch between the required bandwidth and the network capacity available. Consequently, the Quality of Service (QoS) and Quality of Experience (QoE) perceived by the users will be decreased. Nowadays, the only network technology able to offer new possibilities for this situation is MultiProtocol Label Switching (MPLS).

MPLS is widely used in core networks, combining the scalability of IP routing and the speed of layer 2 switching. It allows the forwarding of packets at a very high rate by using labels on label-switched paths (LSPs). The Resource ReServation Protocol with Traffic Engineering capability (RSVP-TE [1]) is one of the main signaling protocols for distributing labels in MPLS networks [2]. RSVP-TE extends RSVP allowing the establishment of explicitly routed label switched paths using RSVP as a signaling protocol. The result are tunnels that can be automatically routed away from network failures, congestion, and bottlenecks.

Tradicionally, IP networks usually manage every IP packet independently of its payload but, despite the fact that video traffic turns ultimately also in IP packets, it has some specific requirements that must be taken into account in the network. This fact can turn into an interesting limitation of IP networks when it comes to carrying video traffic. In IP networks the traffic is traditionally sent through the shortest path. This behavior has made sense for data traffic, but does not for video in all cases.

Furthermore, if we have a look at the IP network from a video perspective, and particularly from a latency-sensible video perspective, on traditional IP/MPLS networks video IP traffic will flow through the shortest path between two points. If congestion arises, packets will be dropped, and the end user experience will be severely degraded. However, an IP network has no mechanism to prevent it, some additional intelligence is required so that the IP network offers the necessary transport behavior for Video IP traffic, taking into account what is relevant for the service and what is not, but specially, what is relevant for the service provider.

So far, the mechanisms used by service providers to guarantee video traffic requirements have been based on very manual and static configurations of traffic classification and queuing, with very basic admission control systems. This scenario tends to lead to over dimensioning of the IP networks that transport video traffic, as the only "safe" mechanism to guarantee the required service [3]. Over dimensioning means over investing, which is a policy very difficult to justify.

As regards to this context, there are different approaches to provide QoS and resource admission control to MPLS networks. For instance, an application-driven QoS control discussed in [4], which assures QoS with the help of DiffServ-Aware TE [5]. This QoS control implementation uses the ITU-T NGN Architecture [6] with QoS guarantees [7], relying on RACF [8]. The solution introduced in the present paper differs from others in that the allocation of MPLS tunnels is based on a dynamic scheme.

The remainder of this paper is organized as follows. Some technology insights are provided in Section 2, while the solution envisioned is described in Section 3, which involves the detailed network architecture, an in-depth view of the hardware and software components that made it up. The test bed and its experimental results are presented and discussed in Section 4. Finally, a brief conclusion and related future work are given in Section 5.

2. Background

2.1 Multiprotocol Label Switching

Nowadays, MPLS has become the industry standard to enable IP packet transport to meet the growing demands for real-time services. It has emerged as the ideal solution to provide triple play services in an efficient and cost-effective manner.

The essence of MPLS is the generation of a short fixed-length label field that acts as a shorthand representation of an IP packet header. In MPLS, the IP packets are encapsulated with these labels by the first MPLS device they find as they enter the network, also referred to as the MPLS cloud. The MPLS edge router analyses certain contents of the IP header and selects an appropriate label to encapsulate the packet with. Part of the great power of MPLS comes from the fact that, in contrast to conventional IP routing, this analysis can be based on more than just the destination address carried in the IP header. The MPLS label, and not the IP header, is used to make the forwarding decision for the packet at all the subsequent nodes within the network. Finally, as MPLS labeled packets leave the network, another edge router removes the labels.

The success of MPLS-based QoS classifications applied to the core network is now driving Internet Service Providers to begin deploying MPLS into other areas of the end-to-end service path. It is now being widely deployed in areas nearer the end user to support fixed and wireless broadband services by adding MPLS capabilities to edge-devices, such as Multi-Service Access Nodes (MSAN), Digital Suscriber Line Access Multiplexers (DSLAM) and cellular base stations gateways.

In this context, as IPTV services get deployed in increasing volumes, the challenges for network operators are becoming increasingly complex. Service providers want to ensure that their networks are able to meet the QoS levels sold as part of the service level agreement (SLA), and they want the ability to anticipate problems in the supporting infrastructure that could lead to non-compliance, and therefore to a significant degradation of the Quality of Experience perceived by end users.

2.2 Resource Reservation Protocol-Traffic Engineering

A set of MPLS control protocols based on IP technology have been developed by the IETF in order to make MPLS networks more flexible and usable. The reason is that controlling a highly dynamic network as MPLS only through the management plane —as it could be done— entails a significant overhead for the network operator, so a more automated approach is required to provide advanced features and functions to the network by means of signaling protocols.

In this regard RSVP-TE is an RSVP extension especially suited for providing traffic engineering capabilities to RSVP, so it supports the reservation of network resources throuhgout the network. RSVP is suitable for extension to the MPLS world because it deals with end-to-end reservation of resources for traffic flows, a concept similar to traffic-engineered MPLS. On the other hand, it does not address all of the requirements needed for MPLS (e.g. label distribution and control of paths through explicit routes). For this purpose, RSVP-TE permits the establishment of MPLS Label-Switched Paths (LSPs) taking into account network constraint parameters such as available bandwidth and explicit hops.

The statistical approach for antennas is part of a general trend and progress nowadays as regards the incorporation of uncertainty in electromagnetism

2.3 Quality of Service

The provision of high-quality media content through a reliable network is a key factor in order to offer IPTV services. Multimedia applications feature several media types, namely: text, graphics, animations, audio and video. Also, network-based multimedia applications are very common today. Due to the importance of networked multimedia applications, it is of fundamental importance for the IPTV network designer or content provider to understand the issues derived from this kind of delivery system.

Across the network, multimedia traffic can suffer from dropped packets, jitter in packet inter-arrival times, delayed and corrupted packets. Also, when it comes to QoS provisioning, video delivery over an IP network raises issues in ensuring QoS that are analogous to, but often more difficult than, those of data services and VoIP, as highly compressed video flows are often strongly sensitive to packet loss. Even when the Transmission Control Protocol (TCP) is used, the effectiveness of the IPTV service can be affected by the reliability and speed of the network.

The aim of Quality of Service is to make sure the network is able to deliver end-to-end data with expected and deterministic results. This includes error rates, latency, service time, utilized bandwidth resources and network traffic loads.

As already stated, QoS is core to IPTV. Only Internet Service Providers who also own and manage the IP network to customer homes can fulfill end user expectations with regards to QoS experience. IPTV services provided through the public Internet cannot guarantee the QoS necessary for a good user experience. Hence, IPTV is not Internet video.

2.4Real-time Protocols

As aforementioned, IP networks were not designed for real-time delivery of data and can have unpredictable packet loss, jitter and delay. To maintain QoS, the media data carried on the IP networks must arrive on time and in the same order it was sent. Real-time Transport Protocol (RTP) can be used to address the time-critical requirement of media flows. Specifically, RTP provides a sequence number and a timestamp to IP packets containing media data. These can be used by the receiver to synchronize playback and handle buffers minimizing network jitter.

The Video-on-Demand approach of IPTV uses UDP or RTP protocols for channel streams, whereas control is done using the control protocol RTSP, which has been developed by the Internet Engineering Task Force (IETF) for use in entertainment and communications systems to control streaming media servers. Specifically, RTSP is used for establishing and controlling media sessions between end points. To this end, media clients act as a remote control of media streaming servers by issuing VCR-like commands to them so as to facilitate real-time control of playback of media files from the server.

As noted earlier, the RTSP protocol is not responsible for the transmission of streaming data itself. Instead, most RTSP servers use RTP for media stream delivery. Nonetheless, the operation of RTSP does not depend on the transport mechanism used to carry continuous media.

3. IPTV-oriented Network Solution

3.1 Network Architecture

The network architecture whose description follows is made up of various network elements operating at different points of the network, interacting with each other. The main feature consists in giving the ingress nodes some knowledge from the MPLS cloud with regards to incoming video flows entering the core network. A diagram of the network architecture is shown in Fig. 1.

A brief description of the main elements which form the IPTV platform follows:

- Video Client (VC). This is the end user's equipment running a media player application, i.e. the final endpoint that will ask for media content to the IPTV platform.

- Streaming Servers (SS). These are the media servers under the Video Service Provider (VSP) control. They satisfy client requests for media streaming.

- Video Proxy (VP). It is an RTP/RTSP relay extension to the SS. Its main role is to handle the media requests from the VCs and manage to get the bandwidth resources allocated before allowing the SS to send the media streaming. Otherwise, it informs the client about the error using the RTSP standard protocol [9].

- Video Label-Switched Paths (VLSP). These are the MPLS tunnels with the necessary allocated bandwidth so the VSP server can

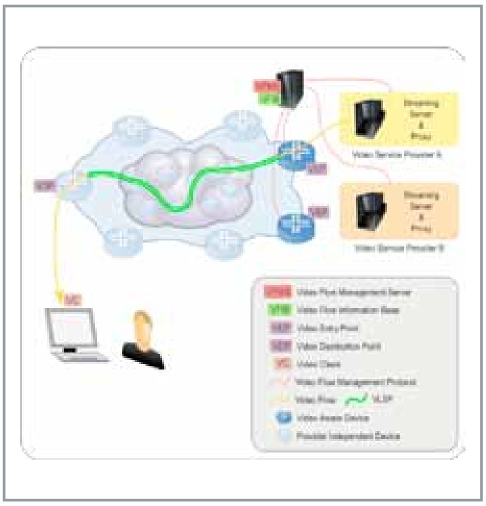

Figure 1. *IPTV-oriented network architecture.*

deliver the media streaming to the client with the required QoS. The term Video Flow (VF) is used to refer to the media encapsulated within the VLSPs.

• Video Aware Device (VAD). This is the name given to the MPLS routers with the necessary intelligence to establish VLSPs and forward video flows.

• Video Entry Point (VEP). It is an ingress router of the MPLS core network.

Video Delivery Point (VDP). It is an egress router of the MPLS core network.

• Video Flow Management Server (VFMS). It is the core network element where the major part of the video 'intelligence' resides. It is responsible for requesting bandwidth reservations to the appropriate VEP routers when needed, i.e. upon receiving a VP request to do so.

• The VFMS gathers, in turn, information from the VADs about existing VLSPs established through the MPLS network and the VFs carried by them. Upon this information is processed, a database is built, the Video Flow Information Base (VFIB), and stored by the VFMS.

3.2 Operation Analysis

Next, the network elements and the software modules are reviewed in greater detail. The video-aware devices can be split in two groups. On the one hand, there are devices and software operated by the VSP, i.e. those ones performing routing tasks and thus implementing the network intelligence. On the other, there are systems and software handled by end users which do not participate in video routing decissions.

The software operated by an ISP runs on VADs, specifically on VEPs. These ingress routers execute the JunOS operating system and therefore they can run customized JunOS SDK [10] applications as daemons. This customized development provides the necessary intelligence to the IPTV platform. The operation of the daemons that have been developed for this architecture is discussed below:

3.2.1 MPLS Reporter

The MPLS Reporter is a daemon that gathers key MPLS data. It sends the VFMS this MPLS information from all the VADs. The data consists of a number of MPLS parameters, such us the VLSP state, bandwidth utilization, RSVP-TE tunnel statistics, and so forth. With this information, the VFMS can get an up-to-date picture of the network state. The parameters available for configuration are: the inter-transmission interval used for sending the reports, and the VFMS IP address and UDP port. The data sent from VADs to the VFMS is encapsulated in VFMP Reports. These messages are character-oriented and have an SDP-like syntax. Each report section about a VLSP contains an identifier, the total bandwidth of the MPLS tunnel and its state.

3.2.2 VFMP Manager

In addition, the VFMP Manager daemon runs only on VEP routers and is responsible for managing the bandwidth allocation requests received from the VFMS. It is also in charge of the creation of firewall rules to send the allocated video flows through the tunnel established with strict QoS guarantees. When a new request arrives, this daemon performs several actions so as to allocate the needed resources. First of all, it checks if the received message is a valid reservation request for a VF. Afterwards, the daemon inspects the existing VLSPs looking for one with enough bandwidth available to allocate this VF reservation.

In case there is no VLSP with enough bandwidth, a new one is created. Both if it was not possible to create a new VLSP and everything went smooth, the VFMP Manager informs the VFMS about the situation with a VFMP Response message.

In addition, the VFMP Manager daemon keeps an internal forwarding table with information about VFs and the VLSPs currently established that have this VEP as its ingress node. Also, VLSP creation and the addition of filter rules to forward the video traffic through the tunnels are performed by executing some scripts on the routers.

3.2.3 Video Flow Management Server

As previously said, the VFMS is the brain of the platform, the device that centralizes the forwarding information coming from all the VADs. It has a global view of the IP/MPLS network, with its VFs and VLSPs. Moreover, it manages the resource reservation mechanism.
Which follows is a description of the network elements that are not directly under ISP control.

3.2.4 Video Proxy

The VP is a software application running on the same machine as the SS, and its task involves complementing the latter in order to guarantee a strict QoS for the VF before it is sent to the VC. This is accomplished by sending a request to the VFMS, as it has been explained previously. For this purpose, the VP implements an RTSP proxy and an RTP/UDP relay. This RTSP proxy follows the flow of messages, as depicted in Fig. 2.

As it can be seen in the diagram, there is a modification in the RTSP dialog. Therefore, this proxy adds the bandwidth reservation functionality to the SS. Once the reservation is done, the VP acts as an RTP relay to send the media requested by the client.

3.2.5 Video Client

The VC is a software application running on the customer system. Basically, it is an RTSP client when it comes to request media content located at the SS, and an RTP/RTCP client for media

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playback tasks. It is not completely transparent to the IPTV platform. Nevertheless, it neither participates in the forwarding decission making nor aids the VFMS sending any kind of IPTV-related information.

3.3 Video Forwarding Management Protocol

For the purpose of communicating the VFMS and the VEPs, a new protocol has been developed, referred to as the Video Forwarding Management Protocol (VFMP). Basically, it consists of a request-response scheme in order to allow the VFMS request a video flow allocation to the appropriate ingress LSR. Whenever the video flow is allocated, the VEP informs the VFMS about the RSVP-TE tunnel that will be used to encapsulate the video flow, otherwise an error is reported to the VFMS.

3.4 RTSP Extension

The dialog between the VPs and the VCs is based on the RTSP protocol. Moreover, a new RTSP method named Allocate has been added so the SS could request a bandwidth reservation to the VFMS before a video flow is setup and streamed. This way, the video flow is sent to the client only if the necessary bandwidth conditions are met.

It is important to notice that this architecture can continue serving new incoming VFs due to its intelligent operation, because new tunnels can be automatically created with a total bandwidth according to the network state. This is truly an improvement because the actual trend of ISPs is to have oversized tunnels which are set manually.

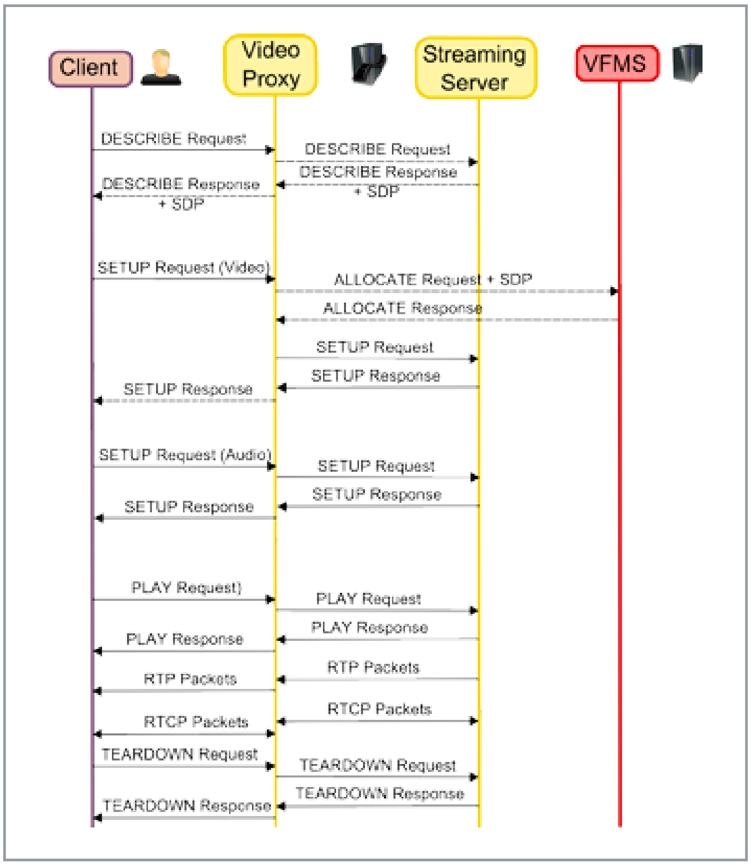


Figure 2. *RTSP dialog involving VADs.*

4. Test Bed

This section presents the results obtained when applying our automated approach to MPLS traffic engineering. A series of tests were carried out to evaluate the delivery of RTP video flows through an MPLS network. The aim of this experiment is to evaluate how well MPLS traffic engineering and QoS can improve the performance of video delivery in an IPTV network without overdimensioning network links, and development of new mechanisms to ensure the provision of traffic engineering and QoS features in an automated manner.

We verify how to optimize the use of the available bandwidth and minimize the effects of network congestion with MPLS traffic engineered paths automatically created and erased as needed.

4.1 Test Bed Design

The measurements analyzed in this paper have been got from a laboratory MPLS network specially designed for this purpose. The topology used to perform the different tests is displayed in Fig. 3. The major elements found in this picture are:

4.1.1 Core Routers

A Juniper M7i Internet router has been used to build the core network by taking advantage of its router virtualization feature. Three logical systems, i.e. virtual routers, have been configured and interconnected, providing flexibility to organize the network in the most convenient way.

4.1.2 Provider Edge Routers

The VEP and the VDP routers in our network scheme are represented by three Juniper SRX 210 MPLS-enabled routers running the same OS version than the core router.

4.1.3 Streaming Server

Darwin Streaming Server is the Streaming Server used, as it is a well-known open source RTSP/RTP server. It stores several video presentations, encoded with a wide range of standard codecs, bitrates, resolution and video quality.

4.1.4 Video Clients

A number of video clients have been used in our tests, all supporting both RTSP and RTP/RTCP protocols. These media players are steered by a custom client application, referred to as the Video Client, which is able to modify the RTSP dialog if appropriate.

4.1.5 Video Flow Management Server

Developed as a Java application for rapid prototyping, the VFMS runs on a common PC. At a later stage it will be implemented as a JUNOS daemon, but for this series of tests there exist no problem due to the small number of video flows allocated to get the measurements.

4.1.6 Video Proxy

It is a Java application running in the same machine as the Streaming Server and set up to communicate with the VFMS. Furthermore, the Video Proxy keeps a list of video presentations that are simulcast capable, so they can be interchangeably sent to the client based on the available bandwidth resources at a given time.

4.1.7 Additional Tools

With the intention of registering Peak Signal-to-Noise-Ratio (PSNR) and video frame measurements EvalVid [11] video evaluation tool is used. It processes the measurements obtained by a network analyzer, Wireshark, so transmission losses and jitter can be tracked for a video flow of interest, referred to as the target video. In addition, D-ITG [12] traffic generator is employed to inject the necessary background congestion traffic in the network.

4.2 Bandwidth Results

Firstly, a number of video flows are sent across the MPLS network without QoS when a certain amount of network traffic is filling the network links and therefore behaving as potential congestion traffic from the video delivery perspective. In Fig. 4 it can be observed that video packets entering the network are competing with existing traffic for bandwidth resources and eventually congestion arises, so a significant number of video packets are dropped as a result. Specifically, only 30 Mbps are encapsulated whereas a total of 38 Mbps were sent from the sources. From the client point of view, the player experiences video frame losses, which leads to an important quality decrease perceived by the user.

On the other hand, when our IPTV-oriented solution is deployed on the same network scenario and the video flows enter the MPLS cloud, a Video LSP is dynamically created and the flows are encapsulated through this path. Clearly, Fig. 5 shows this video traffic occupying 38 Mbps, i.e. the total bandwith sent, as it is guaranteed by the tunnel reservation. Thus, no video packets are dropped and the clients' media players can play the video feeds without any sort of degradation.

4.3 Jitter Results

The Probability Distribution Function (PDF) of the target video jitter measured in each test is shown in Fig. 6 and the results obtained with and within QoS mechanisms applied are compared. As it has been depicted, a higher jitter corresponds to the case without utilizing any QoS mechanism. This was the expected result.

4.4 Frame Loss Results

Moreover, the losses experienced by the target video in each network scenario have been recorded. As we can observe in Table 1, the percentage of lost frames when sending the video applying our QoS mechanism is nil, as it guarantees the required QoS for the proper transmission of every packet that makes every frame up. On the other hand, frame loss occurs when the IPTV QoS mechanism is not used, and in

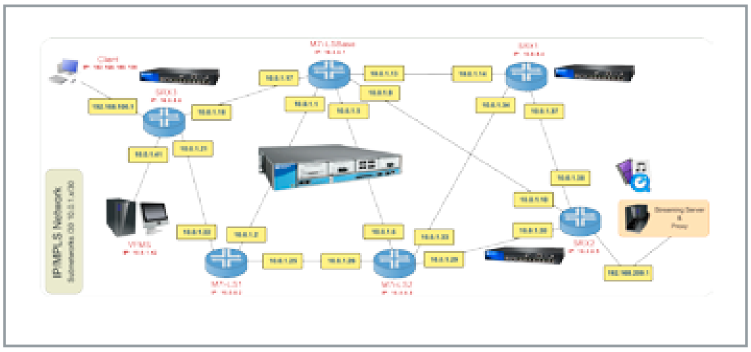


Figure 3. *Test bed network*

the table this loss is classified according to frame type (I, P, B). It worths noticing that each frame is made up by a set of packets at network level.

Without the IPTV-oriented QoS mechanism, some video packets are lost, as discussed in Bandwidth Results, but a frame loss is considered only when its first IP packet (the one that carries the headers) is lost. This way, when a packet (or more) of a frame is lost, and is not its first IP packet, the quality of the frame is degraded but it is not classified as a lost frame.

4.5 PSNR and MOS Results

Also, PSNR has been measured before and after the target video transmission took place, with and without IPTV-oriented QoS mechanisms applied. By utilizing the tools previously described, it has been calculated for each frame. Firstly, it has been calculated after encoding the raw video file, in order to take this PSNR as a video quality reference.

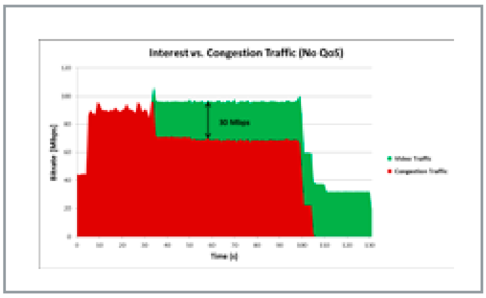
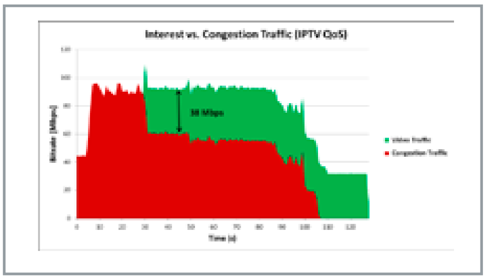


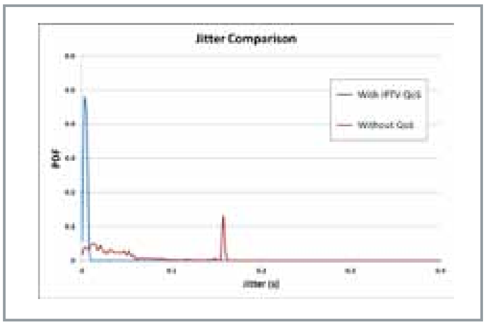
Figure 4. *Traffic bitrate when no QoS mechanism is used.*

As depicted in Fig. 7, we can see the effectiveness of our QoS scheme as the PSNR measured between the raw video and after the transmission coincides with our reference PSNR. In contrast, the PSNR obtained between raw video and after the video transmission, without using IPTV-oriented QoS, is worse than our reference PSNR since it is affected by lost and degraded frames.

In addition, the MOS (Mean Opinion Score) subjective quality metric has been illustrated in Fig. 8. The first column group corresponds to the video before being sent through the network, right after encoding it. This MOS acts as a reference to compare the results. As expected, MOS column group using IPTV-oriented QoS is

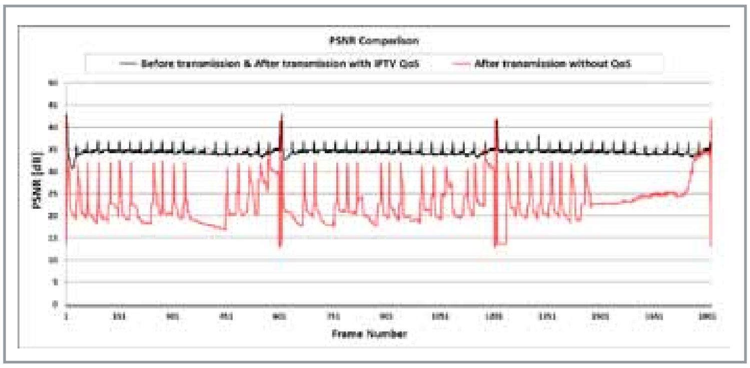


■ **Figure 5.** *Traffic bitrate when using VAIPA QoS mechanism.*

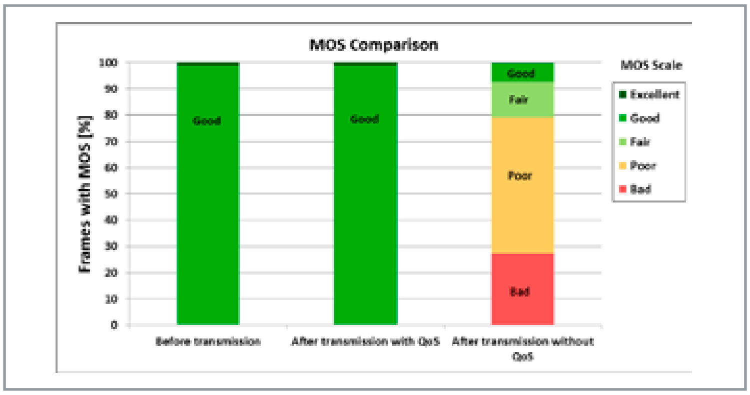


■ **Figure 6.** *Jitter improvement when using IPTV QoS mechanism.*

the same as the reference. On the contrary, the column group without using it has an increment of percentage of frames with lower scales, as the percentage of frames with higher scales decreases, so the quality of the video perceived by a human is worse than the case of using our IPTV-oriented network architecture.



■ **Figure 7.** *Comparison between PSNR measurements.*



■ **Figure 8.** *MOS comparison.*

Percentage of lost frames [%]		
With IPTV Architecture	I (including H)	0.00
	P	0.00
	B	0.00
	Overall	0.00
Without IPTV Architecture	I (including H)	15.62
	P	21.97
	B	0.00
	Overall	21.74

■ **Table 1.** *Lost video frames in each case. NOTE: A frame whose first packet (header packet) has been lost is considered a lost frame.*

Conclusion

Firstly, some insights to video QoS mechanisms and particularly to IPTV oriented ones have been given. Based on MPLS TE, a video flow admission control scheme and an automated LSP management is applied to provide end to end guaranteed QoS to video traffic in IP/MPLS networks. Furthermore, a number of measurements have been carried out so as to assess the benefits achieved by means of these mechanisms with successful results. The required bandwidth is reserved for each video stream delivered through the MPLS network, yielding a significant improvement in video delivery performance indicators, such as packet loss, jitter, PSNR and, eventually, QoE perceived by the end user.

In the future, some improvements to the given IPTV architecture will be added, beginning with the addition of more functionality to the VFMS to take advantage of the information that a Juniper router has inherently (forwarding tables, network state, etc.). Also, the VDPs could be VAD routers that implement JunOS SDK daemons to improve the functionality of the IPTV platform. Moreover, the VP will implement a SIP Proxy due to its flexibility, and it could be too a point of study to find the clients in this IPTV architecture in due time. Another enhancement will be the use of priority queues to assign properly the bandwidth utilized for data traffic and video traffic (LSP tunnels). Finally, redundancy will be added to the network to provide robustness and error resilience.

Acknowledgment

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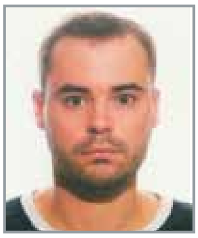
Biographies



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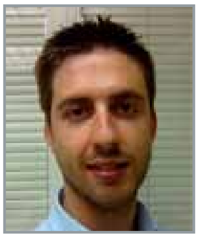

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