# A Set-top Box for End-to-end QoS Management and Home Network Gateway in IMS

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**Abstract —** *As the next generation networks promise network convergence with services for both static and mobile networks, the research focus is converging towards standardizing a common infrastructure for all-IP networks. IP Multimedia Subsystem (IMS) is a recent initiative, which outlines a service delivery platform (SDP) framework. A flexible and robust QoS management system for IMS is essential to ascertain the achievability of service quality on these converged networks. As Triple and Quadruple Play services are the most celebrated services of IMS in the near future, the necessity of QoS assurance is becoming an imminent need of the hour. Multimedia services delivery in converged IP networks using IMS, concentrates on ensuring capabilities to ensure the best QoS. To meet these requirements, we have designed a set-top box which provides feedback in IMS management architecture to ensure end-to-end QoS as well as enables the functionalities of Regional Gateway for the Home Network. Results of the simulation confirm the improvement of performance parameters and the quality of service.*[1]

**Index Terms — IP Multimedia Subsystem (IMS), Quality of Service (QoS), Set-top box, Home Gateway, end-to-end QoS.**

## I. INTRODUCTION

IP multimedia subsystem (IMS) is the future for all IP next-generation converged networks with the potential of enabling service providers to create and provide value added services to users on heterogeneous networks [1]. IMS was defined by 3GPP (3rd Generation Partnership Project) [2] as a standard architecture which provides a horizontal, cross-functional layer of intelligence on top of IP, enabling the creation, control and execution of new and rich user-to-user services (video streaming), server-to-user offerings (IPTV) and multi-user media services (game-playing on the move and at home via PC).

The services provided by the IMS environment should ensure Quality of Service (QoS). As the traffic can be generated from anywhere at anytime, it is desirable that QoS management is dynamically adaptable to end users' requirements. These requirements are according to the Service Level Agreement (SLA), which specifies the connectivity and performance parameters for an end user service from a provider of service.

The next generation solutions are designed to provide converged services such as Triple Play [3] and Quadruple Play [4] services, which are going to become the most celebrated services in the near future. This demands a need of end-to-end QoS between the consumer and the service provider.

A Set-top Box (STB) [5] is used as the end device to provide multimedia services to the consumers. As this set-top box is a service providers' owned equipment, it can be used to ensure end-to-end QoS. Furthermore, this device can also provide the functionalities of a regional or home network gateway [6].

In this paper, we present a Set-Top Box; designed to provide both of the above mentioned functionalities. Next section provides a brief background and related knowledge in the field of STB, Home Gateway and end-to-end QoS delivery. Section III, discusses the management in the IMS environment. Section IV shows the design of our set-top box, and in Section V, we present the mechanism of the proposed STB. Section VI provides the mathematical calculations for the network performance parameters. Section VII discusses the simulation and Section VIII shows the results and evaluation. Finally, we conclude our work in Section IX.

## II. RELATED WORK

In this section the existing QoS solutions using Set-top Box or Regional Gateway are reviewed and their limitations are identified for end-to-end QoS provision in multimedia service delivery scenarios.

IMS has enabled provisioning of real time multimedia services; however these are resource demanding services, which breed new challenges in providing end-to-end QoS and resource management. Research focus had been concentrated on this particular topic to enable consumer satisfaction and to contend with the competitors (service providers). Many solutions are proposed to ensure end-to-end QoS in IMS architecture, such as [7] and [8]. The solutions that exist make use of network performance monitoring and ensure end-to-end QoS in the network.

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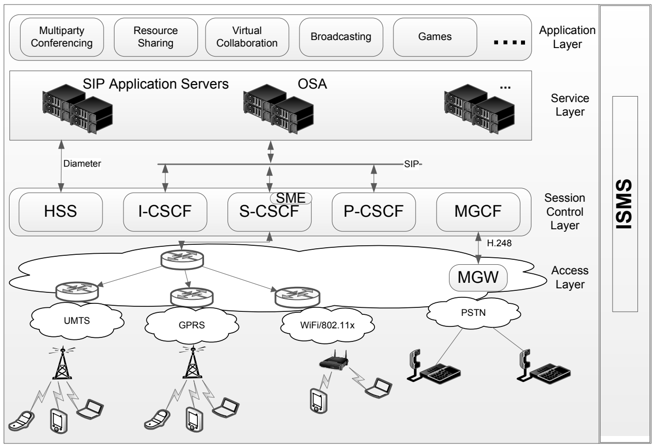

**Fig. 1. Simplified view of IMS/ SDP architecture with ISMS as a vertical layer.**

For Digital video broadcasting, STBs have been used for quite a while now. They provide audio and video decoding at the user's end. Some intelligent STBs have been designed to provide better QoS in the network, such as [5], [9].

With the concept of digital home, home network gateways are being used to provide centralized network control within a home network. With STB as the end device from the service provider, it can be used to perform the functionalities of the home gateway to extend the internet connection as discussed in [5], [9].

To the best of authors' knowledge, the proposed STB is one of the first systems which provide network performance feedback for ensuring QoS management in IMS based systems as well as works as the Home Gateway. Our STB also works as the SIP server and client for establishing session calls, manager for the home network, and provides decoding for the video and voice channels.

## III. IP MULTIMEDIA SUBSYSTEM & SDP

The most important aspect of next generation networks is the network convergence. Converged networks are thought to be the future of communication with support to every network from conventional POTs (Push-tO-Talk) and cellular mobile network to wired and wireless broadband networks. IMS, defined by 3GPP (3rd Generation Partnership Project) [2] promises to provide the access layer, for the converged networks, as an abstraction for the service layer. But IMS architecture must be made compatible with existing service delivery environment such as Service Delivery Platform (SDP) [10]. A service delivery platform helps to standardize all the service interfaces for a provider, creating a horizontal platform from which they can provide, control and bill for all the value-added services they provide; whether the services are created by third-party application developers or by the service providers themselves. By ensuring a consistent, highly automated and reusable service environment, a service delivery platform can dramatically accelerate a positive return on investment. A typical SDP+IMS solution is depicted in Fig. 1 and another famous one is discussed in [11].

As IMS is described to be the solution for the future services delivery, it must provide a cost effective solution to the companies. Therefore, the services provided by the IMS environment should be delivered according to the Service Level Agreements (SLA) and ensure Quality of Service (QoS). Furthermore, IMS nodes should provide best performance and error free environment to ensure the best delivery of the services. This gives rise to the issue of managing the IMS environment so that it provides cost effective solution and enable increase in revenue. Management of IMS system to provide service delivery should be designed to provide non-disruptive services with maximum quality [8]. Performance of the network nodes should also be monitored to ensure that there is no fault in the system. Error and faults in the system should be identified and reported. Also, efficient error recovery mechanism should be applied to ensure the services are delivered up-to the demand of the customers.

### *A. Management in IMS*

IMS is thought to be the common service platform for the next generation networks. However, IMS applications require adequate management solutions for the efficient delivery of services they promises to provide. The IMS is emerging overlay architecture and has not been standardized yet, hence, decisive measurements need to be taken so the near future technologies are in coherence with the standardized management framework designed for IMS/SDP [12].

Our research has been focused on identifying the key management roles in this system; so, we designed our management framework around a single Management server named IMS SDP Management Server (ISMS) [12], [13].

An ISMS is a fundamental part of IMS/SDP system as shown in Fig. 1. It supervises the vital states of IMS/SDP components. IMS/SDP process and traffic are constantly monitored, both actively and passively. The captured traffic is collected, correlated and analyzed and performance information of session control layer and quality of delivered services are diagnosed. Management server also gathers different performance, security, fault and configuration parameters from all over the network to maintain a healthy state of the system. For this it uses SNMP agents at each managed nodes to gather the required information.

ISMS uses its module named Session Management Enabler (SME), which resides in the Serving Call Session Control Function (S-CSCF), see Fig. 1. SME is responsible for the management of the multimedia sessions within the system. SME uses Session Initiation Protocol (SIP) [14] signaling to monitor and manage the call and media sessions between the service providers and users. For the management of services ISMS uses service managed objects (SMOs), which provide the performance, configuration and fault parameters of services provided by the servers in the application layer or third party servers [13].

### B. Policy Decision & Enforcement

Basic idea of policy enforcement is given in Fig 2. The centralized management server ISMS, gathers the network parameters using SNMP agents, managed objects and Service Managed Objects (SMO) [13] to provide network and service management. It uses SIP signaling for session management. For QoS management, Policy enforcement is done using the well known architecture of Common Open Policy Service (COPS) protocol [15].

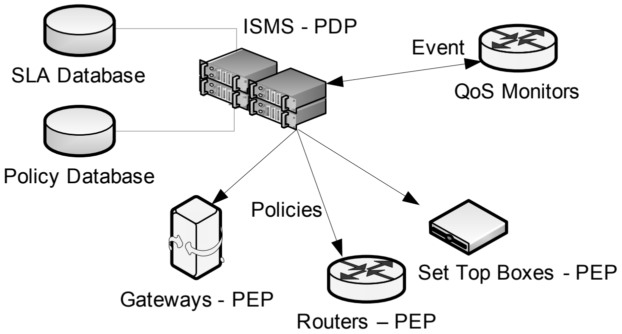

**Fig. 2. Policy enforcement using COPS.**

When a network element is started, its local Policy Enforcement Point (PEP) requests the Policy Decision Point (PDP) for all policies concerning DiffServ traffic marking [16] and resource reservation using COPS [15]. The policies sent by the PDP to the PEP, may concern entire node's QoS configuration or a portion of it, such as, an updating of a DiffServ's marking filter. The PDP may proactively provision the PEP reacting to external events generated by some monitors such as a bandwidth monitor.

The ISMS manages the SLA for each customer and ascertain the delivery of service according to the SLAs. ISMS works as the PDP, while the network and IMS nodes such as gateways, routers and STBs work as PEPs [15]. These PEPs ensures that the applied policies according to the SLAs are enforced. For example, a gateway would manage that the certain amount of resource, such as bandwidth, is provided to a certain flow (as stated by the SLA). So it has to guarantee that amount of bandwidth to this flow. Moreover, it also limits and checks if more than allowed resources are not being provided to a certain flow. In conclusion, these PEPs guarantee that each flow gets its rightful share.

### C. Signaling

This approach is completely SIP-based where all network entities use SIP for both, service and media signaling. Within this approach the SME is always aware of the user's current state because the SME component sometimes acts almost as a kind of B2BUA (Back-to-Back User Agent). The SIP INVITE message from the user is handled by the SME and forwarded to other components of the architecture. The response message from the MGCF (Media Gateway Control Functions; see Fig. 1) is forwarded to the SME. So, all user-related messages are handled by the SME. This implicitly results in the user's state update. In general, the stream from the media server is linked to the UA (User Agent). The SME receives a content query by the user and invokes the media server entity to push the stream to the querying user. Trick functions (such as play, stop, ff, pause etc.) are represented by SIP INFO messages. The SME relays these messages to the MGCF and performs a direct state update. Using SIP for session signaling and RTSP for media signaling is a relatively recent approach. So in case of media signaling, the UE (User Equipment) communicates in a direct way with the MDDCF (Media Deliver Distribution Control Function). This requires the MGCF to notify the SME (via SIP NOTIFY message) about state updates. This notification mechanism is based on a subscription-notification framework [4].

### D. Session Management Enabler

Session management is provided by the session management enabler (SME) [4], which is an integral part of the ISMS, shown in Fig. 1. The SME manages all user-to-content and content-to-user relationships. Furthermore, the SME manages all media server signaling that is necessary to deliver a demanded content to a demanding consumer. It provides session mobility and bearer mobility [13]. Both mobility issues are crucial for a full service delivery scenario such as IPTV and IP telephony. SME enables the session management and the control of multimedia streaming sessions. A small module of SME is integrated in S-CSCF, which monitors and controls the sessions and provides session management functions. SME updates ISMS using either SNMP MIBs or SIP INFO messages.

SME monitors all the unicast and multicast sessions between the users and service providers. Session monitoring is based on two protocols, SNMP (SMOs & MIBs) and Real-time Transport Protocol (RTP). RTP includes a companion protocol, called Real-time Transport Control Protocol (RTCP), to communicate information on data reception quality among the service providers and end users of an active unicast or multicast session. SME uses SIP INFO messages for control messages and RTP to implement the functionality of session monitoring.

SME has to control all the services such as the Internet, telephony, multimedia sessions etc. These services can be interactive and demand certain content. These interactions can switch between different contents, react to incoming or outgoing calls, process trick functions (ff, rw, and pause), perform notification of subscribed content etc. The session management is the main control unit for the entire service delivery scenario. A key task of the session management is the provisioning of session mobility and bearer mobility. Bearer selection can be performed based on various parameters (e.g. device capabilities, network resources, number of receptions, etc.). Deeper concerns are to be researched in future works.

## IV. SET-TOP BOX ARCHITECTURE

In digital video streaming, system and services are not yet robust and in poor QoS conditions, customer may experience static interference, delayed images and/or non-synchronized audio video phenomenon [9]. We have designed a STB with focus on ensuring QoS in the IMS architecture. As the STB is

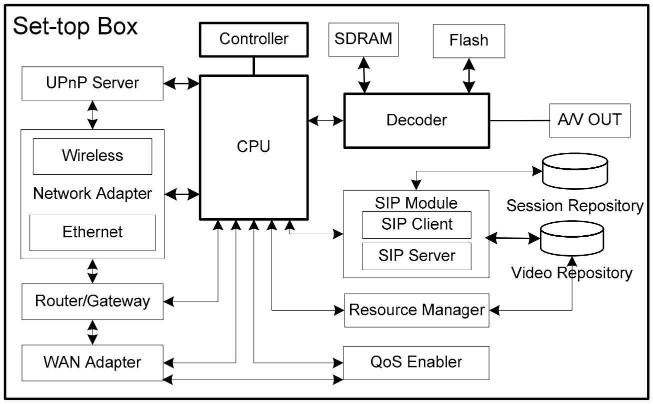

**Fig. 3. Modular design of the proposed Set-top box, which includes the functionalities of a Home Network Gateway and end-to-end QoS enabler.**

the end user device on which the service provider has some control; it provides network performance parameters as a feedback to the QoS session enabler in IMS network. The STB uses simple SIP signaling to communicate with the management server, which monitors the network and ensures the optimal QoS.

STB monitors network parameters (such as round-trip latency, jitter, throughput and packet loss etc) and signals them back to the management server. The server fine-tunes the network by prioritizing the flows, and service parameters accordingly, making it possible to create an end-to-end solution by integrating cooperative application (based on SIP) from different vendors.

### A. *Architecture*

Fig. 3 shows the modular architecture of our proposed STB. Along with the usual subsystems of a conventional STB, our STB includes a QoS enabler, a UPnP server and a SIP module. A QoS enabler is responsible for providing end-to-end QoS between the end user nodes and the service provider.

The UPnP server provides the centralized management for the home network. The UPnP architecture with plug and play functionality maintains the information of all the devices connected in home network. By using a simple interface at the STB, devices attached in the network can be managed providing simple functionalities such as turn on, turn off and set timer etc.

The SIP module, consisted of SIP Client and SIP server Module, is responsible for two functions:

1). Behave as a client to the SIP server in IMS (such as the CSCF) and
2). Behave as a server to the SIP clients in Home network.

Session repository is used by the SIP module to manage the multimedia sessions. On the other hand Video Repository is used to provide the functionality for audio/video trick functions such as rewind, forward, fast forward and pause etc.

### B. *Protocols*

Fig. 4 shows the protocol stack of the proposed STB. The simple control messages in between the network nodes are delivered using the ICMP protocol. The STB provides both IPv4 (with NAT) and IPv6 functionalities for the network access for the home network nodes. SNMP is used to gather the management information from the different nodes in the network to enable the functionality of the QoS enabler. UPnP protocol stack (includes HTTP, HTTPU, HTTMU and SOAP) provides the plug and play networking for the home network.

## V. SET-TOP BOX MECHANISM

Our proposed STB provides three basic functionalities:

1). Home networking gateway;
2). SIP session management, and
3). End-to-end QoS enablement.

Along with these significant functionalities, the STB also works as the SIP server and client for establishing session calls, manager for the home network, and provides decoding for the video and voice channels. The major functionalities of the STB are discussed as follows:

### A. *Home Gateway*

The STB extends the Internet (broadband) connection to the home network, providing the nodes in the home network access to the external network. The UPnP server makes use of its plug and play architecture and provides a centralized management point for the home network. As the home gateway, the STB can control all the traffic going out and coming in the home network. It classifies the traffic and also work as a firewall for the local network. Our STB is able to support Network Address Translation (NAT) for IPv4 networks and also full support for IPv6 networks.

### B. *SIP Session Management*

All the multimedia sessions are managed by the STB, with two-way signaling with the session management enabler (SME) at the management server ISMS. SIP Module within the STB is responsible for:

1). Session initiation and negotiation
2). Session renegotiation
3). Session termination
4). Session mobility

The state of each session is stored in the session repository, which helps in session reestablishment, session mobility and performance measurement. Furthermore, it provides all statistical information about active sessions.

SIP client creates, controls, and deletes sessions (e.g. TV sessions). All session related modifications are invoked and monitored here. It furthermore handles the communication

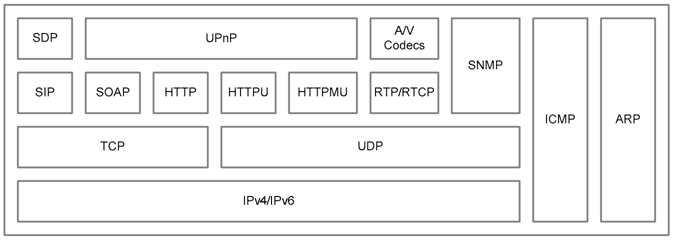

**Fig. 4. Working protocol stack for the proposed Set-top box.**

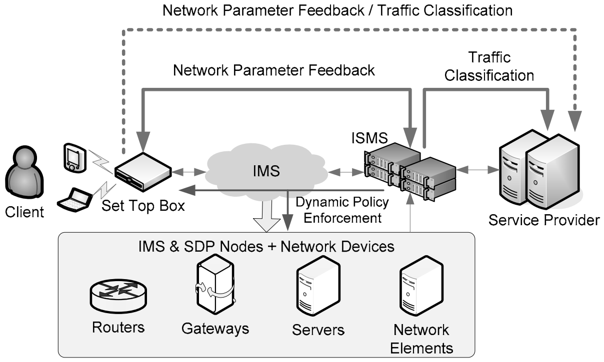

**Fig. 5. Network feedback and signaling between the STB & ISMS (SME).**

with the SIP server. It is also the connection point to the SME.

SIP server is responsible for session establishment and management within the home network.

### C. *End-to-end QoS Enablement*

Sometimes, due to a change in network parameters such as degradation of performance, a certain resource may not be provided to each flow according to its SLA. There are various ways (as discussed in Section II) to avoid such problems, such as, to compromise some flows for certain important flows. This noncompliance of SLA can cost a service provider a loss of customer to its competitors.

The STB includes a QoS enabler, which is responsible of assisting the ISMS in managing the overall system. The QoS enabler at the STB and the SME at the ISMS communicate with each other to make sure the multimedia sessions are properly managed and the sessions are served with optimal QoS. The working of the SME is discussed in section III-D, while the working of the QoS enabler at the STB side is as follows:

The QoS enabler and the SME identify each flow and monitor the flows such that performance measuring parameters of the flows can be calculated on a per flow basis. These parameters are compared to the threshold values defined in the SLA for the customer (available at the ISMS). When for a certain flow, the performance is below the required level, ISMS tries to increase it performance by increasing the resources for it. This is done in two ways.

First, ISMS create dynamic policies according to the required resources and flow performance and these policies are sent to the concerned policy enforcement point to make sure the flow gets more resources.

The second mechanism requires the MPLS mechanism. If the service provider supports the MPLS traffic classification, the ISMS contacts the service provider to increase the priority of the concerned flow. This is done by increasing the traffic class for the flow traffic (such as from AF12 to AF11 and so on). Both of the mechanisms are depicted in Fig. 5. The feedback from the STB to the ISMS is shown in solid arrow. After that the traffic classification message is sent to the service provider and the new policy is enforced to the network nodes along with the STB itself.

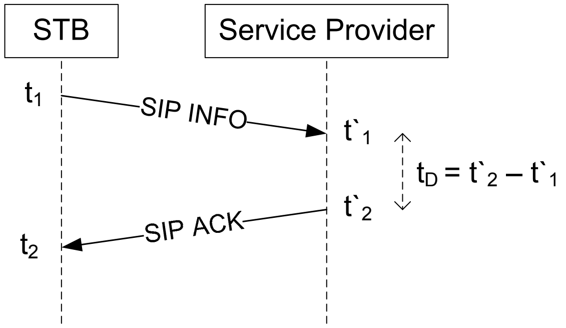

**Fig. 6. SIP INFO & ACK packets are used for measuring packet transfer delay from the service provider to the STB**

For the traffic feedback there are two solutions available. The first method (as already discussed) is to send the feedback to the ISMS which in turns communicate with the service provider to rectify or improve the link in between the service provider and the user. While the second approach is to send the feedback directly to the service provider, which can take action on its own for increasing the quality of the service it provides. However, for this, the service provider should be able to afford this service. Although, the second approach is not implemented but it is discussed as a future work. The detail of the first approach is envisaged in Fig. 5 and will be further discussed in simulation section.

## VI. CALCULATIONS FOR NETWORK PARAMETER

Currently the system is implemented to provide feedback of packet delay, jitter, packet loss, and throughput attained by the flow. The calculations of these parameters are done at the STB as follows.

### A. *Delay*

Delay is calculated as the round trip delay sustained by a packet and its response, as shown in Fig. 6. It can be calculated by the service provider also. The SIP client at the STB sends a SIP INFO message to the service provider. The service provider is bond to send a SIP ACK message in response [17]. The STB calculates the round trip delay D as in (1);

$$D = \left(t_2 - t_1\right) - t_D \tag{1}$$

$$t_D = \left(t_2' - t_1'\right) \tag{2}$$

Here, $t_2$ is the time the response is received at the STP. $t_1$ is the time when the SIP INFO packet is sent by the STB. $t'_2$ is the time the SIP INFO packet is received at the STP. $t'_1$ is the time when the SIP ACK packet is sent by the service provider. However, sometimes $t_D$ calculated as shown in (2), may not be available at the STB. In such a case, the system ignores $t_D$, as this value is not significant, as discussed in [18]. Since, in a normal case the connections are symmetric in nature; we divide the value *D* by 2 to get the packet delay from service provider to the STB.

### B. *Inter-arrival Jitter*

Jitter is calculated for each data packet, as it is received by the STB. The difference between the arrival times of each packet provides the inter-arrival time for each packet, as shown in Fig. 7. This inter-arrival time is compared with the difference in the timestamp of these RTP packets. Hence, the STB can calculate the variance of inter-arrival time by using (3).

$$\Delta = \left|\left(t_2' - t_1'\right) - \left(t_2 - t_1\right)\right| \tag{3}$$

Here, $t_1$ and $t_2$ are the timestamps in the RTP packets 1 & 2, respectively. $t_1'$ and $t_2'$ are the times at which the RTP packets 1 & 2 are received at the STP, respectively. $\Delta$ Similarly, $\Delta$ for each consecutive packet pair is calculated to measure the jitter in packet arrival.

### C. *Packet Loss*

Packet loss ratio ($\delta$) is defined as the number of packets lost divided by the number of packet expected. It is calculated as shown in (4).

$$\delta = \frac{n}{\sigma} \tag{4}$$

Here, $\sigma$ is *'highest sequence number received'*, which is the highest value of the sequence number received in the RTP packet header for that flow and $n$ is the total number of packet expected to be received at the STB.

## VII. SIMULATION & IMPLEMENTATION

For the testing & evaluation of our Set-top Box proposal, we have performed simulations on Pentium Dual Core processor using C# .Net. The simulation environment is depicted in Fig. 8. We have implemented typical nodes of the IMS architecture such as; Proxy, Serving, and Interrogating Call Session & Control Functions. Along with these nodes, we have also implemented the functionalities of the IMS-SDP Management System (ISMS) [13]. Moreover, many router nodes are included in the simulation for creating background traffic. These router nodes use static routes for maintaining routing tables and interconnectivity. These router nodes are for simulating the Internet environment such that it is crafted as an imitation of real scenario.

When a user connects to a service provider for example to get a video streaming, it communicates with the CSCF to start a session. CSCF in turn connects to the Service Provider and establish the session between the user and the service provider with already decided Service Level Agreement. As the session progress, the background traffic disturbs the network performance and the promised QoS is not accumulated at the user end.

STB in the simulation monitors the session that is used by the client and delivers the performance parameters to the SME. An ISM gathers the information from the SME. When the service is not up to certain threshold, ISMS communicates with the service provider and informs it about the problem i.e. to upgrade the QoS. If the service provider supports this functionality then it sets the priority of that session and puts it in a higher class till it achieves the required QoS. If the service provider cannot upgrade the service, the Session Management Enabler (shown in Fig. 1) performs this task. Typically, the network is supposed to support MPLS classes or some traffic prioritizing technique. SME just enhances (or sets the priority to high) the class of the concerned session's traffic, so that the session can be entertained by higher resources.

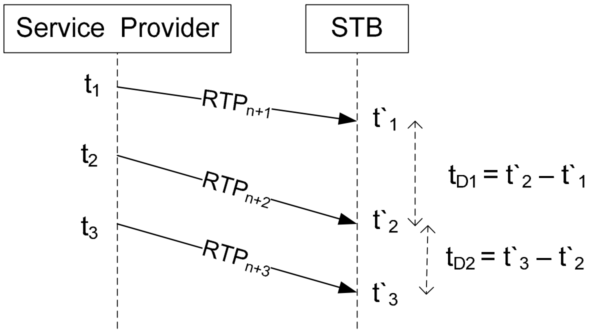

**Fig. 7. RTP packets are used for measuring packet Inter-arrival delay at the STB**

Along with the traffic classification, policy creation, modification and enforcement are also performed. The IMS nodes such as gateways, servers, routers, and the STB are the policy enforcement points, which make sure that the new policies (such as increase in bandwidth, bounded delay and higher priority queues) are provided enabling better resources to the affected flows.

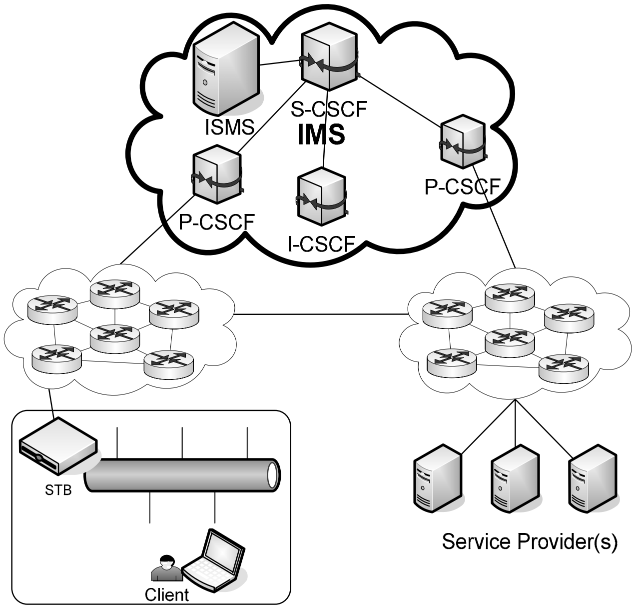

**Fig. 8. Simulation environment, with Client connected to STB and Video service is provided by the service providers' servers.**

## VIII. RESULTS & ANALYSIS

The simulation has been performed with different number of flows with video, voice and the Internet (HTTP & FTP) traffic. We have performed our analysis and evaluation on the basis of the simulations. A client uses a multimedia service, with intermediate nodes having high rate of entropy for network delay and available bandwidth. These delays are generated at each node randomly with simulating different flows. With the help of QoS feedback, our system is able to reserve better resources for each flow.

Fig. 11 shows the throughput achieved by a video flow from the service provider to the set-top box. Video flow generates two 1024 Kbits packets per second. We have compared the throughput in the normal case and in the case of using our proposed scheme for end-to-end QoS enablement.

At the start, the throughput is low as there are small sized packets for session establishment. More packets with small size show less throughput than few packets of larger size (although the rate is same). After session establishment, we see an improvement in the throughput for the video flow. As the time progressed, we analyzed the throughput for the flow by increasing the background traffic. We created 2 voice channels and 4-5 HTTP channels for back ground traffic. Results shown in Fig. 11, confirm the higher throughput achieved by our scheme. Due to the feedback gathered from the STB, the video flow is classified as higher prioritized. As the flow was lacking throughput, higher bandwidth is dedicated to the video flow and the throughput for the proposed scheme is enhanced.

The second decrease in the throughput is due to the start of a new video flow. For some time the new flow affects the old flow but after a while the throughput of the first video flow starts to improve again.

Fig. 10 shows the effect of the proposed scheme on the packet loss rate of the traffic between the service provider and an STB. Fig. 10 shows the packet delivery ratio in percentage which is calculated as; *100 – packet loss ratio*. As with the throughput, the loss rate is high at the beginning, and the packet delivery ratio at the STB is less. But after the receiving the feedback from the STB, the service provider increases the priority of the flow for higher bandwidth. This increases the average delivery ratio for the data traffic.

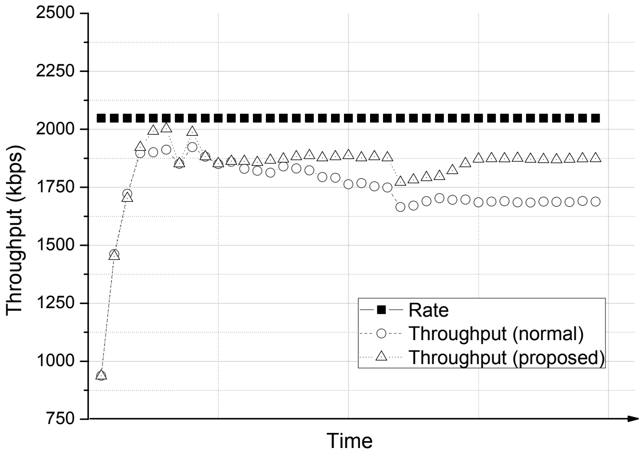

**Fig. 9. Simulation results for a multimedia stream with degrading internet channel. According to feedback from STB, management server makes sure that the multimedia stream gets more data delivery (throughout).**

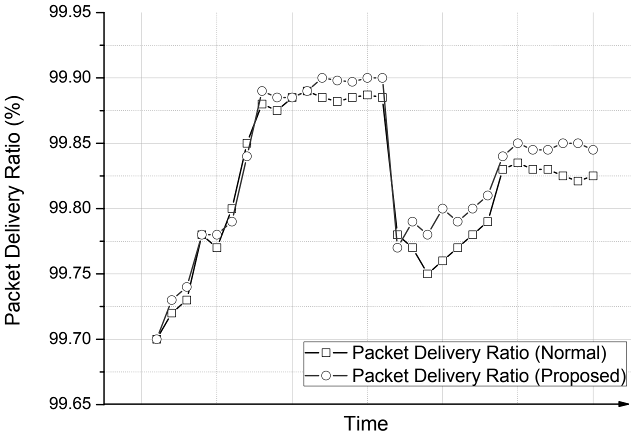

**Fig. 10. Simulation results for Packet Delivery ratio at the STB for the data traffic between the service provider and the STB.**

The sudden loss in delivery ratio is due to the addition of a new simultaneous flow. This is again due to the higher loss rate of smaller sized packets used for establishing the session. But as the time goes on, the packet delivery ratio increases and gets stable. In Fig. 10, it is clearly visible that the higher prioritization of the multimedia flows reduces the loss rate, although the difference is not much.

Fig. 11 shows the effect of the proposed scheme on the delay of packet delivery for the video flow. As showed by the Fig. 11 the proposed scheme is able to reduce delay of the delivered packets. This is by prioritizing the flow for high priority queues, thus the packets in the flow bare less delay at intermediate queues.

The sudden increase in delay is again due to the addition of a simultaneous flow. This interns the average delay for the packets received by the STB and it calculates higher values of delay for the incoming packet. But as the second flow is also prioritized, the average delay for incoming traffic starts to decrease.

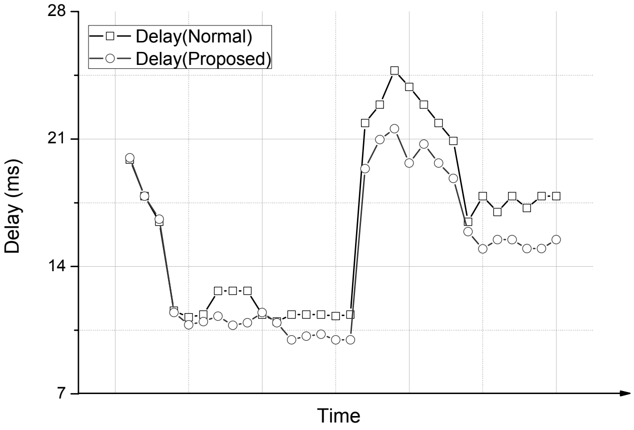

**Fig. 11. Simulation result showing the delay in milli-seconds for the multimedia flows.**

From the results we got from the simulation, we can see that feedback mechanism shows the potential for improving the QoS in multimedia sessions. Although the difference is not much but as more flows are added, the difference increases. Also, some window for the improvement at the SME side is also available.

## IX. CONCLUSION & FUTURE WORKS

In this paper, we have presented the design of our STB, which works as a home network gateway (regional gateway), as well as enables end-to-end QoS in the IMS architecture. According to the best of our knowledge, this is the first STB, which provides QoS to home network nodes utilizing services delivery on top of IMS. Our proposed STB is simple and effective for next generation converged multimedia networks.

Furthermore, we are working on the virtual platform within the STB. With this technology, an STB would be able to provide a virtual manager to each service provider, which in turn would be able to manage its own flows. This mechanism assures to reduce the message passing overhead and promises to be beneficial in the future.

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