IP MULTICAST ADMISSION CONTROL FOR IPTV

A Thesis by

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The following faculty members have examined the final copy of this Thesis for form and content and recommend that it be accepted in partial fulfillment of the requirements for the degree of Master of Science with a major in Electrical Engineering.

Ravi Pendse, Committee Chair

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Abu Asaduzzaman, Committee Member

DEDICATION

God, the Almighty

My Parents

Mrs. Lalitha Jayaraman & Mr. Jayaraman

My Family

Mrs. Indira Subramanian and Mr. Subramanian

Mrs. Mythreyi Venkatesan and Mr. Venkatesan

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ABSTRACT

Video streaming over the Internet has become the most sought after application and is growing at a very fast rate. Internet Protocol Television (IPTV) is a technology that has been growing fast, replacing traditional cable TV. With the rapid development in the high speed networks, multimedia streaming over the internet has increased incredibly, of which video streaming is the major source of traffic in the core network. Hence IPTV video streaming over the core network has become one of the active topics for research. The major challenge associated with IPTV traffic is the fact that video traffic requires more bandwidth and is more sensitive to delay and packet loss due to congestion. Lots of research has been done to provide an Admission Control algorithm for IPTV traffic. Admission Control becomes an essential part as it is typically enforced to ensure QoS in the network. It helps prevent bottleneck in the core network.

This thesis proposes an efficient method to provide admission control for IPTV traffic in the core network by using multiple GRIP probe packets to check the resource availability in the core network for the new incoming channel request. Moreover, the algorithm proves that using multiple video qualities in the network helps increase the number of channels delivered to the end user, thus satisfying more users, as opposed to single video quality. Using multiple GRIP packets made the proposed method more reliable and it was seen that on an average, the number of channels delivered to the end user was increase over 90%.

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LIST OF ACRONYMS

FIFO	First In First Out
FTP	File Transfer Protocol
GoP	Group of Pictures
GRIP	Gauge and Gate Reservation with Independent Probing
HDTV	High Definition Television
HTTP	Hyper Text Transfer Protocol
HTTPS	Secure Hyper Text Transfer Protocol
IETF	Internet Engineering Task Force
IGMP	Internet Group Management Protocol
IP	Internet Protocol
IPTV	Internet Protocol Television
ISP	Internet Service Provider
ITU-T	International Telecommunication Union
MPEG	Motion Picture Expert Group

LIST OF ACRONYMS (cont.)

PDA	Personal Digital Assistant
PIM	Protocol Independent Multicast
PIM-SM	Protocol Independent Multicast - Sparse Mode
QoE	Quality of Experience
QoS	Quality of Service
RFC	Request for Comment
RP	Rendezvous Point
SDTV	Standard Definition Television
SLA	Service Level Agreement
SMTP	Simple Mail Transfer Protocol
VCEG	Video Coding Expert Group
VoD	Video on Demand
VoIP	Voice over Internet Protocol

CHAPTER 1

INTRODUCTION

1.1 Overview – Internet Protocol Television

Today's communication network offers its customers a wide range of services. These services include Voice over IP (VOIP), Video on Demand (VoD) and various data services including email (SMTP), web (HTTP and HTTPS), FTP and several more. The Internet Service Provider (ISP) generates their revenue by providing the above-mentioned services with contractual Service Level Agreement (SLA). There is a definite increase in the revenue with the addition of the new Internet Protocol Television (IPTV) technology. IPTV is different from an Internet TV. Examples of Internet TV are video available through YouTube, in which the end user receives the video content by streaming and downloading. IPTV, on the other hand, becomes a part of the "Triple Play" service which would be provided by the ISP. This triple play content would include VOIP, Internet access and IPTV [1-2]. The growth of IPTV is seen to be accelerating at a very fast pace over the past few years. Figures 1.1 and 1.2 show sectional regional wise growth in the number of subscribers and the revenue for IPTV technology. There are four regional sections including Asia, Europe, North America and Rest of the World (ROW) [3]. The growth forecast in the number of IPTV subscribers in the years 2009 through 2013 is depicted in Figure 1.1. The growth forecast in the revenue from IPTV technology in the years 2009 through 2013 is shown in Figure 1.2.

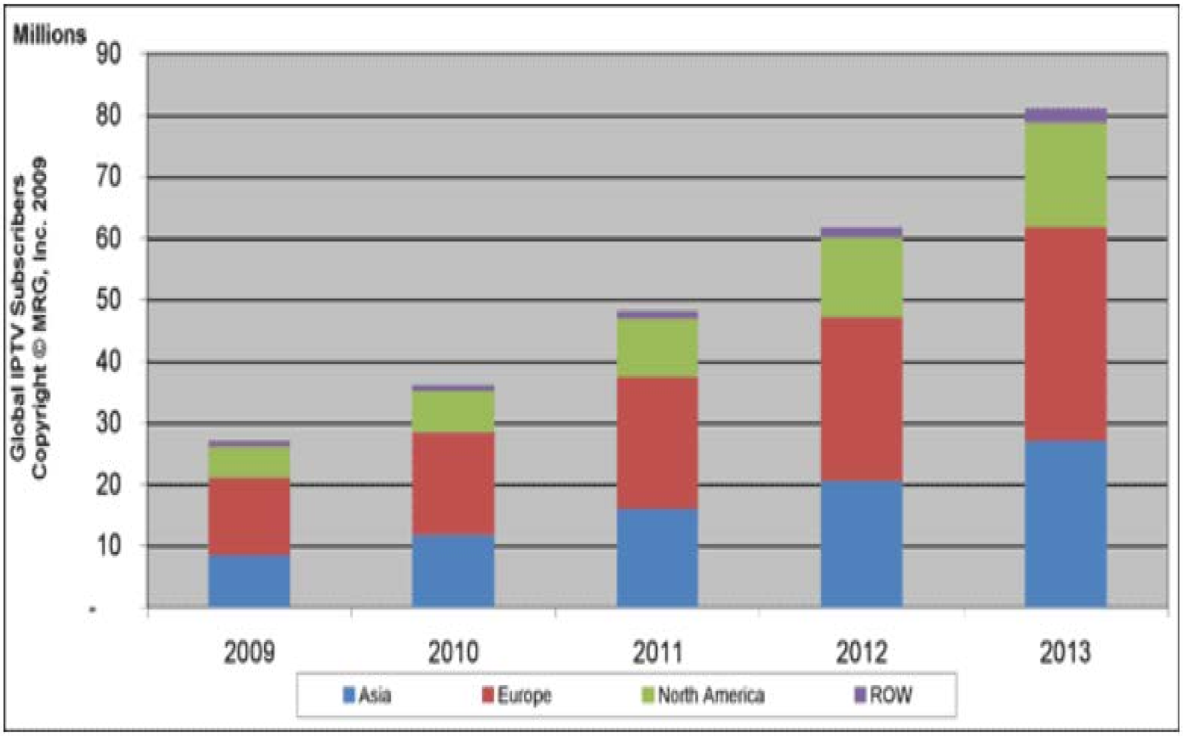


Figure 1.1 IPTV Subscriber Growth Forecasts [3]

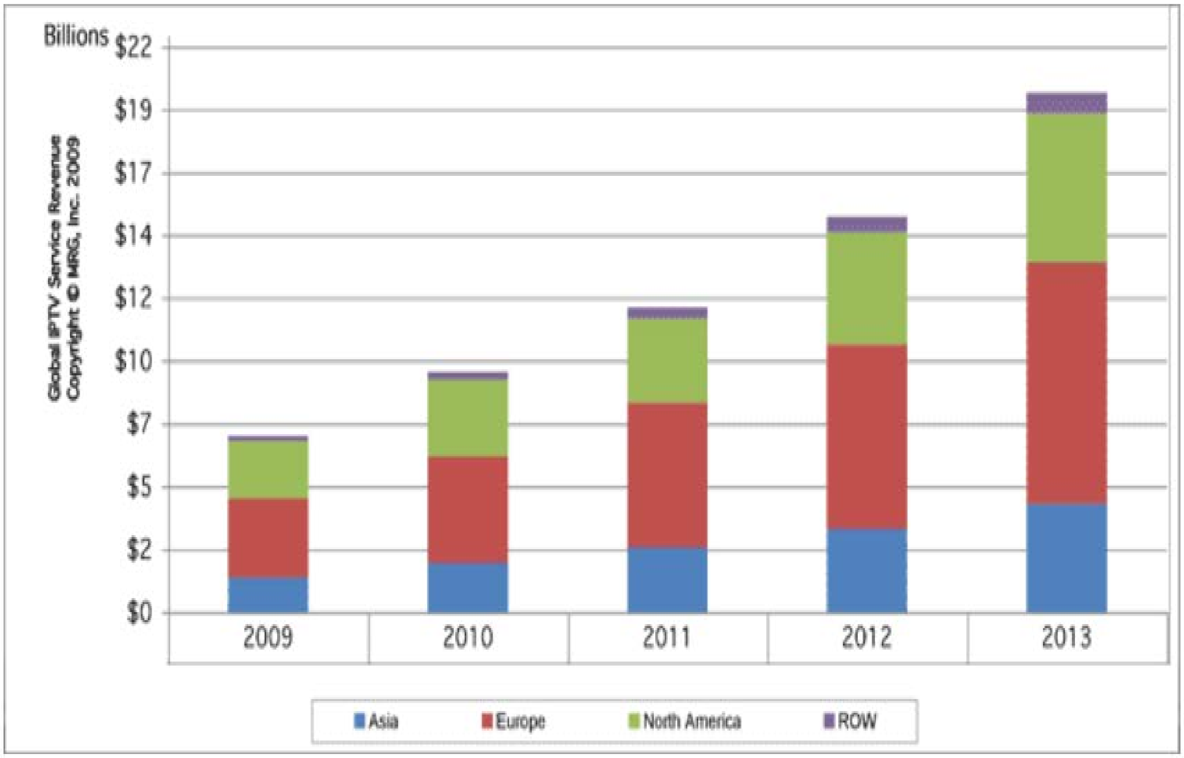


Figure 1.2 IPTV Revenue Growth Forecasts [3]

Internet Protocol Television [IPTV] is an emerging technology. Introducing a new technology in the existing infrastructure always poses its own non-technical and technical challenges. The most interesting and a challenging part of this new technology is that it makes use of the existing IP network. Since all of the various services like VOIP, Internet access and IPTV are handled by the same network service provider, it is very much essential that the Quality of Service (QoS), Quality of Experience (QoE), security, reliability and interactivity is maintained. One other interesting aspect of IPTV is that it does not restrict users to view only on traditional TV, but provides options to view on any hand held device like laptops, mobile phones, PDA's etc. It is important to remember that IPTV shares its bandwidth in the core network with the other existing applications like data and voice. Also IPTV, being a video application, would eventually use more bandwidth than any other applications. It should not affect the other existing applications. Initially, when the concept of IPTV technology was introduced, the quality of video delivered to the end user over the internet was very poor due to limitations in the bandwidth and compression techniques used. With the enhancement in the compression techniques and proper allocation and utilization of bandwidth, the quality of service provided and the quality of experience will improve.

IPTV uses Multicast technology to deliver packets to the end user. This method is preferred over the conventional broadcast method used by cable TV as it helps to prevent excessive usage of bandwidth. By using Multicasting, the service provider delivers only those channels requested by the end user. This makes it much more resource efficient than the broadcasting technique.

1.2 Multicasting

In 1988, Steven Dearing [4] proposed the first model for Multicast communication on the notion that a group of users are requesting for the same information. Multicast communication refers to a group of users interested in receiving one particular stream of information. The first multicast model was deployed in 1992. The advantage of multicasting is that it does not have any constraints on the physical or geographical location of the hosts. The only requirement is network connectivity. Members intending to become a part of the multicast group send a join request to the router based on an Internet Group

Management Protocol (IGMP). This makes sure that the hosts are explicit members of a group, so that the packets are delivered to the members in that particular group as long as they remain as members of the group. These hosts are identified based on Class D IP address, often referred to as the multicast IP address (224.0.0.0 to 239.55.255.255) assigned to them when they join the multicast group. This way they do not know the original IP address they are receiving the data from. The algorithm which allows a host to arbitrarily join and leave a group is explained in RFC 1054, RFC 1112 and RFC 2201.

1.3 Admission Control

It would be unreasonable to expect the core network to handle the growing IPTV needs of the end user. An Admission Control technique is typically enforced to ensure QOS in a network. It is ideally implemented between the edge device and the core network to restrict the flow of traffic that enters a network. This mechanism can be used to prevent bottleneck in the core network. The admission of IPTV traffic into the core network can be controlled using various parameters. Extensive research is being done in this area and several admission control mechanisms have been proposed [5-8]. The method of interest involves an end user, who wishes to view a particular feed, sending a message to the edge router requesting to join that multicast group. The steps involved in the process, where the edge router makes a decision to permit or deny the join request, is where the proposed admission control mechanism can be applied. This thesis document guides through a procedure where the edge router and the other routers in the core network communicate with each other with the help of probe packets which would help calculate the available resources in the network. This calculation is done on every router and is local to that router as proposed in [5-6]. Based on the response it receives, the decision making router (DM router) accepts or rejects the PIM multicast join request. Multiple QoS levels are introduced to increase the number of channels and users who would be utilizing the resources that are available. Additional servers are also introduced in the core network. These servers are capable of buffering the channel traffic and if necessary assist in changing the type of video coding used.

1.4 Framework

The remaining part of the thesis document is organized as follows. Chapter 2 provides a detailed analysis of the previously published literature associated with the proposed thesis. It includes the work related to Multicast networks, heterogeneous and differentiated service domains, an idea on network planning and the existing admission control algorithms for IPTV traffic. Chapter 3 explains the briefly the architecture involved in IPTV networks, the existing layout and implementations. A brief discussion addresses the problems that are present in the existing network and how the proposed algorithm covers these issues. Further on Chapter 4 explains the proposed model and covers the algorithm and its simulations. Chapter 5 is the analysis of the simulations and its results. Finally Chapter 6 provides a conclusion for this thesis work and opens ideas for future work.

CHAPTER 2

LITERATURE REVIEW

2.1 An Outline

This chapter provides a detailed study of the various literatures published, that deals with providing the best multicast admission control methods used for meeting the growing QoS targets of the end user. Yang Xiao et al [9] has provided a detailed study about IPTV and the various technologies that are involved in delivering IPTV to the end user. The paper also analyzes the various network scenarios involved and the challenges that are posed in meeting the QoS requirements when delivering IPTV to its end users. Section 2.2 introduces heterogeneous and differentiated services networks and the works done in that area to provide quality of service to the end user without disturbing the existing layout. Section 2.3 guides through network planning and its importance when it comes to making admission control decisions. In section 2.4 the empirical estimation of bandwidth with the help of network planning is reviewed. Finally section 2.5 deals with the proposal of GRIP packet and its effectiveness in admission control. It concludes with how this probe packet is effectively used in the proposed thesis.

2.2 Heterogeneous Networks and Diffserv Domain

Quality of service is one of the growing requirements in the IP network. Though the bandwidth in the backbone network is increasing continually, without provisioning the resources appropriately, it is very difficult to support QoS in the

Internet. The major existing problem is the heterogeneity in the IP multicasting network. Heterogeneous network is a term given to networks with different levels of QoS requirements. To provide QoS in the internet, the IETF proposed the Differentiated service architecture. Baijian Yang and Prasant Mohaptra [10] have conducted a detailed study on Multicasting and Differentiated Service domain and have listed the complexities and challenges involved in having them coexist. The authors have also mentioned the necessity of the coexistence of differentiated service domain and multicasting. In order to have a scalable and simple network core, the routers in the core of the core networks in differentiated service architecture, are simple and do not maintain any information regarding the per-flow states of the traffic flowing through them. A.Striegel and G. Manimaran [11] have proposed a method to provide multiple QoS levels in the heterogeneous diffserv network. The proposed DSMCast method uses the encapsulation approach. In this method, the information about the domain’s multicast tree is encoded. This encoded information is appended to the IP packet in the form of DSMCast header. This process helps in the simplification of tasks including the heterogeneity of the receiver. The DSMCast is extended with the help of dynamic DSCP, wherein the DSCP value in the packet changes as it crosses the diffserv domain. This process facilitates a single multicast tree to support QoS heterogeneity.

2.3 Network Planning

It is very important to know the existing network resources for having an efficient Network Planning. Network demands are to be known for QoS aware planning of network. At the same time, having to introduce a new hardware for calculating the network demands does not make it cost efficient for the Internet service provider. Figure 2.1 depicts a simple block diagram of the Network Elements of IPTV [1].

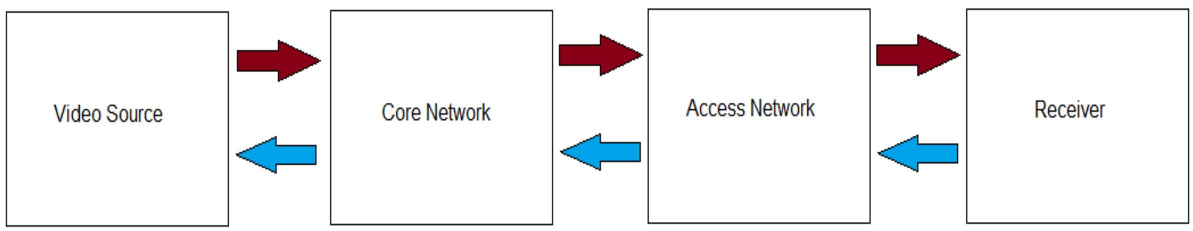


Fig 2.1 Network Elements of IPTV[1]

Alan Davy et al. [12,13] proposed a method for effective network planning by studying the network demands and harnessing the required information that is already gathered by the ISP for the purpose of accounting and charging. It efficiently uses the pre-existing information which makes this technique highly cost-effective and easier to deploy. Apart from this, the author also proposes a technique to calculate the effective bandwidth available for a new flow of traffic [12-15] based on the analysis of packet traces for that source. A FIFO queue of infinite size is assumed to be empty initially. With the trace packets of the same input traffic, the delays created for different traffic are analyzed. With the help of this, the percentage of traffic with delays and effective bandwidth for the traffic is estimated. Based on the estimated values, the admission control decision as to whether the traffic has to be accepted or rejected is made. This algorithm was extended to be employed with scenarios involving multiple levels of quality of service.

2.4 Bandwidth Estimation and Admission Control

In order to ensure that the end user receives a non-disrupted flow of traffic by maintaining packet-level QoS during periods of congestion and at the same time avoid bottleneck in the core network while still meeting the Service Level Agreement (SLA) of an end user, a set amount of bandwidth is allocated to a particular traffic flow by the service provider. This is termed as Admission Control. The key feature of Admission Control would be the ability to make a decision whether to accept or reject a new traffic flow request based on the prediction of effective bandwidth available for the incoming flow of traffic. Alan Davy et al. [15] proposed an Empirical Effective Bandwidth Estimation Algorithm. Based on this approach, of the various QoS parameters considered for the admission control technique, the delay parameter was emphasized. Delay, for a video traffic is unacceptable. When there is a lot of delay in the delivering of packets to the end user, the traffic becomes unusable. This is because out-of-sync packets (with respect to time) cannot be used for a video traffic. A video frame once missed cannot be used or delivered later. Having this in mind, there is a maximum acceptable delay for video traffic in IPTV. A typical value of this delay, maintaining the QoS targets is (50ms, 0.001) [15]. In this case only 0.1% of the total traffic that is being sent to the receiver can be delayed over maximum acceptable delay, i.e., 50ms. In this paper, he has defined $delay_{max}$

as the value of the maximum permissible delay, P_{delay} as the maximum percentage of traffic that can have a delay greater than $delay_{max}$ and R_{eff} as the effective bandwidth available for the traffic flow. An attempt has been made to prove that at any given time, with effective bandwidth R_{eff}, the percentage of traffic having a delay greater than $delay_{max}$ is always below $P_{delay.}$

2.5 Bandwidth Estimation Using Probe Packet (GRIP Test)

Olli Alanen et al. [8] researched on multicast in differentiate services environment and proposed a method for admission control in differentiated services network. Based on the proposed algorithm, when a new channel request comes in, the edge router, if not a member of the request multicast group, sends a PIM join request to the upstream router. The decision is made based on a GRIP probe packet that the edge router would generate to be sent to the branching node. This probe packet checks to see if all the intermediate links in the path to the destination has the available resources for the new incoming request. If every node in the path is found to have sufficient resources, the new channel join request is accepted. If any link in the path is found to have insufficient resources, then the request to join is denied and the packet is dropped.

This GRIP packet was first introduced by Giuseppe Bianchi et al. [5,6]. The GRIP which stands for Gauge and Gate Reservation with Independent Probing was initially introduced for unicast transmission of traffic. Later the GRIP was extended further [7] to work with Multicast transmission of traffic. When a channel join request comes in from a user, the edge router sends a PIM Multicast join request to the first upstream router that is a part of the requested Multicast group. The upstream router checks to see if it is a part of the requested multicast group. If it is a part of the requested group, this router becomes the decision making router responsible for the admission control in the network. This router sends a GRIP probe packet to the edge router which is connected to the user requesting for the new channel. Ever router on the path performs a calculation which is local to that router. This calculation triggers accept or reject switch in each router. The idea of this local computation is to check and see if it has enough resources available for the new request. If the switch in the router is in the accept state, the probe packet is allowed to pass through, else it drops the packet. When the probe packet reaches the edge router, the router replies back with a confirm message. When this packet reaches the decision making router, the router accepts the request and starts sending the requested traffic. This technique can be deployed in a heterogeneous environment where there are different QoS targets.

The proposed model uses the above mentioned GRIP packet in the admission control technique to gather information on the bandwidth availability in the link. With the help of the calculated values, a decision is made as to what quality of video can be delivered to the end user to meet the user requirements as well as maintain the network standards and resource availability for future incoming requests. Chapter 4 discusses the proposed model in detail.

CHAPTER 3

INTERNET PROTOCOL TELEVISION ARCHITECTURE

3.1 Introduction

Internet Protocol Television IPTV, being an emerging technology, would eventually be the most sort after video application. The most challenging part would be to accommodate the ever growing needs of the end user in the core and access network. For the major part of the work done on this research, emphasis is made on the admission control technique at the core network. IPTV traffic is basically video traffic and hence takes up more resources than any other application. There is a lot of work going on in this area to come up with an admission control technique that would best suit the core network requirements and resources. In the methods previously proposed by Giuseppe Bianchi et al. [5-7] admission control is done based on the GRIP (Gauge and Gate Reservation with Independent Probing) packet that travels through the nodes in the network to know the available resources for the new traffic.

3.2 Existing Architecture

The basic IPTV network is depicted in Figure 3.1. The IPTV live streaming makes use of Multicasting. The edge router takes care of the IGMP requests from the hosts and the PIM requests between routers. When a channel (multicast group join request) request comes from a user to the edge router, the router checks to see if it is already processing the requested traffic of the interested channel. If it is, it simply accepts the new channel request from the host allows it to join the multicast group and starts sending the channel traffic to the host. If it is not processing the requested traffic, it sends a PIM join request to the upstream router. The upstream router when it receives the PIM request, checks to see if it is a part of the requested multicast group. If it is, then this becomes the decision-making router, which takes care of admission control in the core network. If it is not a part of the multicast group, the PIM join request is forwarded to the upstream router until it reaches the first router that is a part of the requested multicast group or in the case of PIM-SM the Rendezvous point (RP) itself. The first router in the network which is a part of the requested multicast group becomes the decision making router. The decision is made based on GRIP test as shown in Figure 3.2.

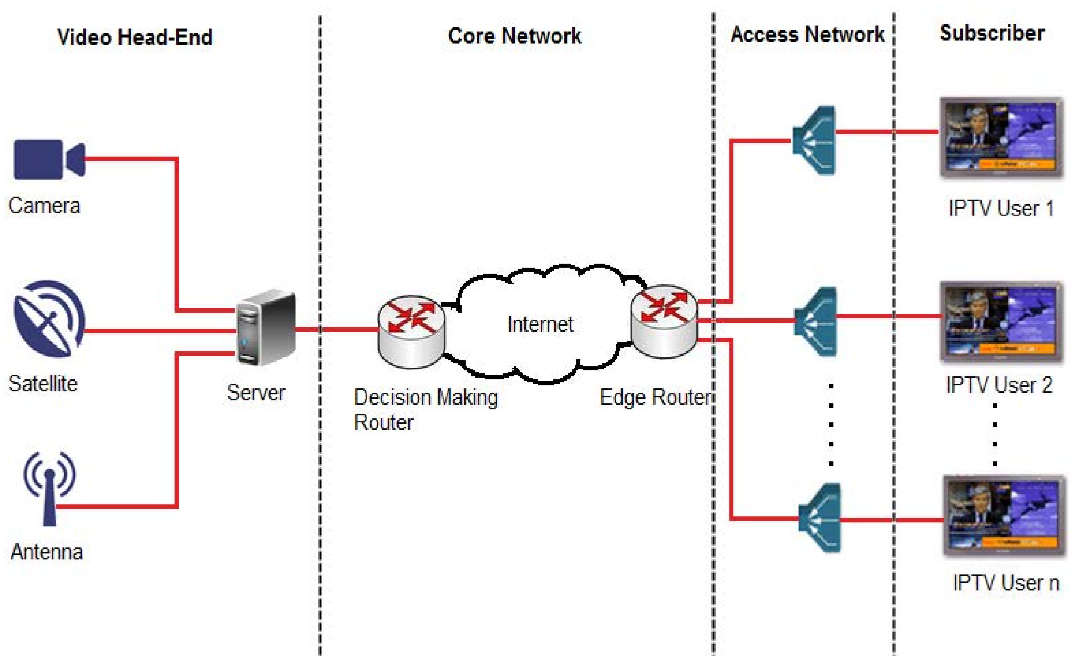


Figure 3.1 IPTV Architecture

When it receives the PIM join request, it sends a probe packet to the edge router and starts a timer set to an acceptable value. The probe packet passes through every router in the multicast path. Each router performs a local calculation [5,6] to check and see if it has enough resources available in the link to accept the new traffic. Based on the results of the calculation, an ACCEPT/REJECT switch is triggered. If when the router receives the probe packet, the switch is in the ACCEPT state, the probe packet is allowed to go through the router and if it is in REJECT state, the probe packet is dropped. If it happens that the switch in all the intermediate routers are in the ACCEPT state, it means that there is enough resources in the network for the new traffic. The edge router when it receives the probe packet, send a confirm message as an acknowledgement of the receipt of the probe packet back to the decision making router. When this confirm packet reaches the decision making router, the router knows that there is enough resource in the network for the new traffic and hence accepts the PIM join request from the edge router and starts sending the requested channel traffic. Once the edge router gets to know that it has been added in the PIM multicast group, it in turns accepts the IGMP join request from the host and starts streaming traffic to the end user. This is shown in the sequence diagram in Figure 3.2

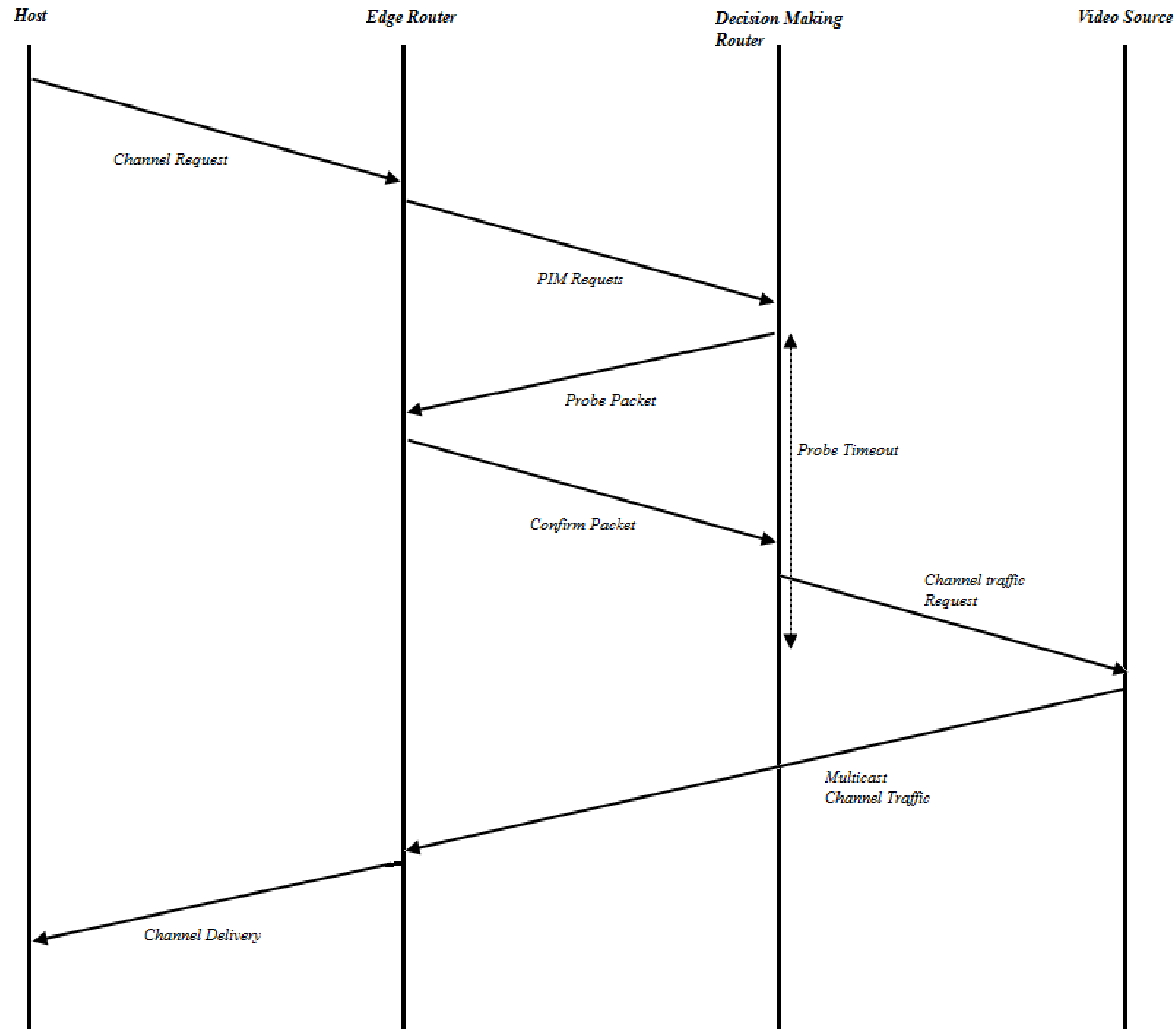


Figure 3.2 GRIP Operation for IPTV Admission Control

In the case where the probe packet is dropped due to the router in the REJECT state, the decision making router waits for a time equal to that set in the timer. If at the end of this period, it does not receive any response from the edge router, it assumes that there are not enough resources in the network for the new traffic and hence rejects the PIM join request. This is explained in Figure 3.3. When this PIM reject reaches the edge router, the edge router in turns sends an IGMP reject message to the host. Ideally, in such a condition, the host receives a service or channel unavailable message. This is not a very acceptable scenario as the end user will not be happy to receive this message.

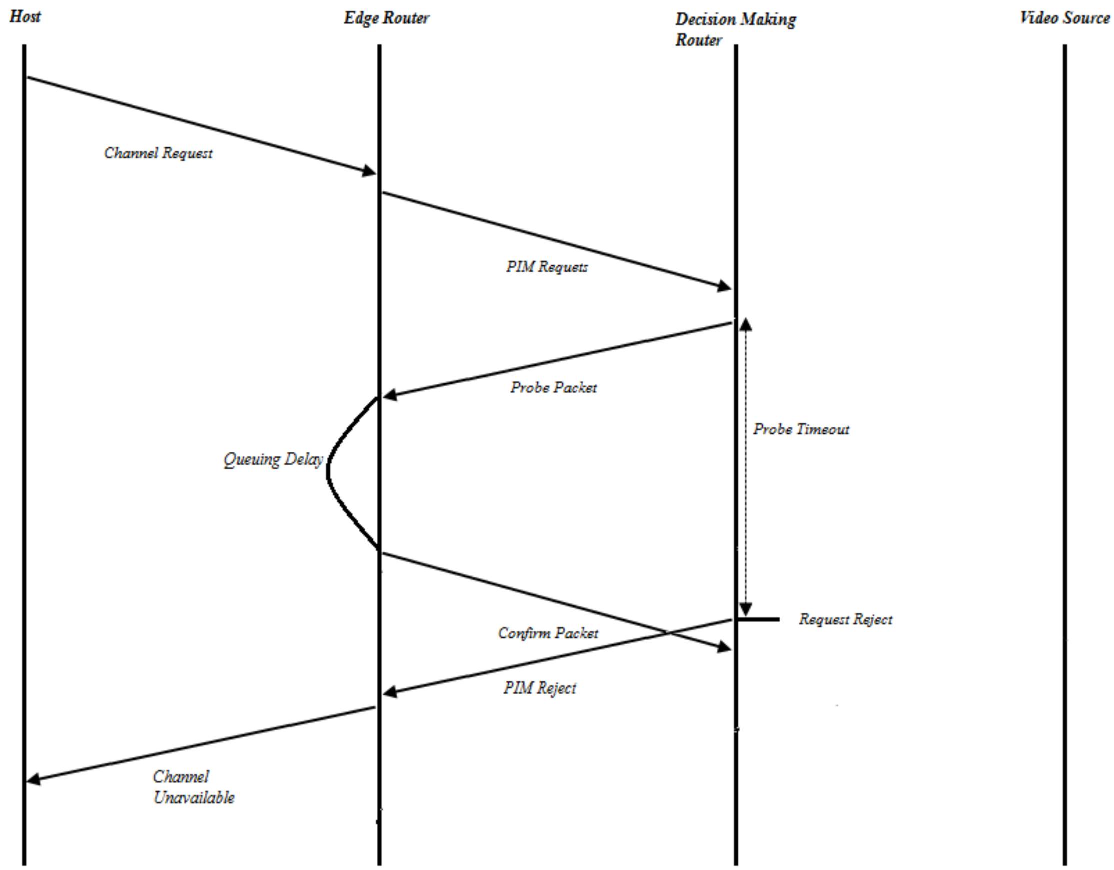


Figure 3.3 PIM reject due to GRIP queuing delay

3.3 Issues and Objectives

This technique did not consider the growing needs of quality of the video delivered. It either accepts or rejects the request but does not give any other options to the user. It should also be kept in mind that a lot of requests would be rejected if the users start requesting for different channels i.e., if the edge router has to be a part of multiple multicast groups. This means that the edge router has to process a lot of traffic associated with different channels. If every single time a user sends a request for a new channel to the edge router that is not being processed by the edge router previously, the links in the core network would be out of resources very quickly. Every single request after that threshold point has to be rejected.

To overcome this, the proposed workaround introduces different quality for videos in terms of coding compression techniques used. Additional servers are introduced in the core network. These servers are capable of coding and buffering video traffic. The proposed new technique of admission control would help accommodate more users in the network and also provide different quality videos to the users without utilizing excessive resources in the core network.

CHAPTER 4

PROPOSED APPROACH FOR ADMISSION CONTROL FOR IPTV TRAFFIC

This chapter covers an overview of the various existing video coding standards. It also discusses the proposed approach for IP Multicast Admission Control for IPTV traffic. It projects in detail the various steps involved in a new request entering a network and how the proposed algorithm effectively makes a decision as to accept or reject the new traffic based on the resources available in the network.

4.1 Internet Video Streaming

A video consists of a continuous sequence of frames. The number of frames displayed per second is termed as frame-rate. For a video to appear as a motion picture to human eyes, there should be a minimum of 15 frames displayed per second. The typical frame rate of a video is 24fps or 30fps. The more frames displayed per second, the better the quality of video. Resolution is the number of pixels per frame. It is the number of distinct horizontal and vertical pixels forming the display. A typical video frame contains 320x240 lines of resolution. As it is known, there are three primary colors: red, green and blue, often referred to as RGB. Hence digitally, for any color frame, each video frame consists of 320x240x3 pixels. This has a numerical value of 230400 pixels or, in binary terms, bits of data. For a minimum frame rate of 15fps for a video traffic, the total amount of data in bits per second would equal (230400x15) 3456000bps. For a standard television, video traffic uses 30fps. For this case the amount of data transferred would equal (230400x30) 6912000bps. This is a large amount of data and would definitely pose a major challenge to be transferred across a network channel. This is where video compression algorithms come in handy. The video compression technique helps reduce the amount of data transmitted through a channel by over 50%. Figure 4.1 shows some of the standard resolutions available today.

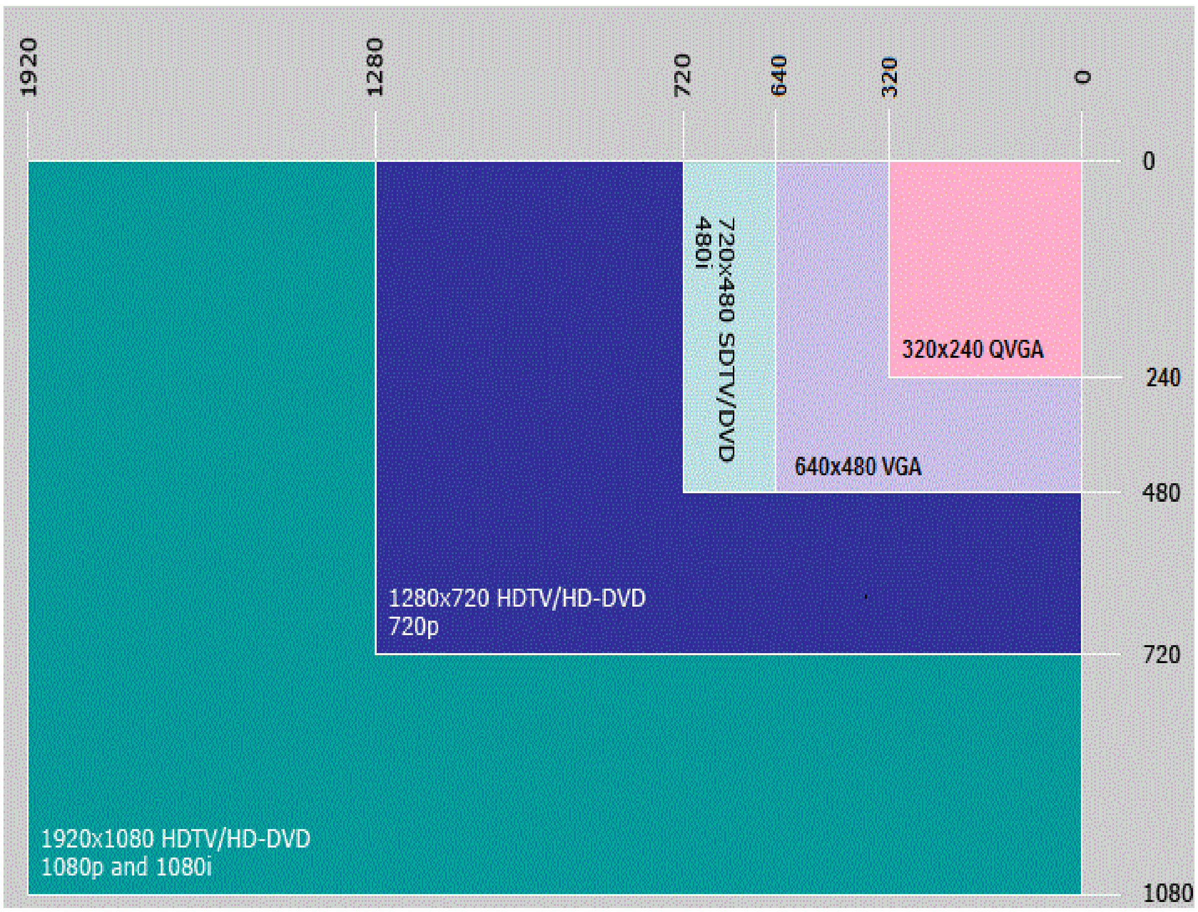
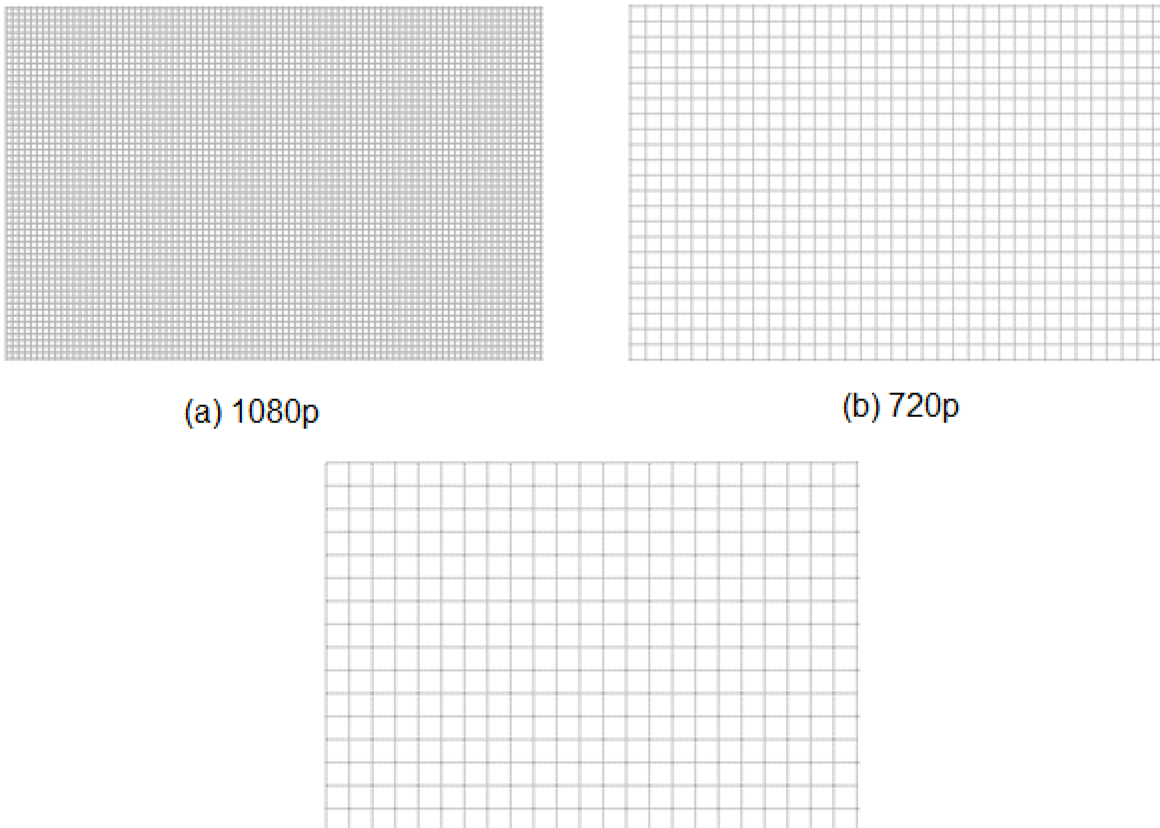


Figure 4.1 Standard Resolutions

The resolution of the display depends upon the number of pixels displayed on the screen. The pixel count and pixel size for a same sized display is shown in the Figure 4.2.



(a) 1080p

(b) 720p

(c) DVD Quality

Figure 4.2 Pixel Variation within same sized display

4.2 Concept of Video Streaming

It is a well-known fact that the raw video from its source is in the analog format. This analog video is converted into digital signals that can be transmitted over a network channel. The process of converting an analog signal to a digital format is termed as digitization. This digitized video is then fed to an encoder. The encoder is responsible for compressing the video data to make sure that the bandwidth of the video signal is reduced so that it can be transmitted over the network channel. This video data is further broken down into packets at the router. The router is responsible for routing the video packets to the destination. The router takes care of both unicast communication between server and client (generally VoD traffic) and multicast communication between server and multiple clients (live video streaming). At the destination, the data is depacketized and is sent to be decoded. This decoder would be responsible for the reconstruction of the video from its compressed format. These decoded video frames are sent to the output device, usually a screen. [16, 17].

4.3 Video Coding Techniques

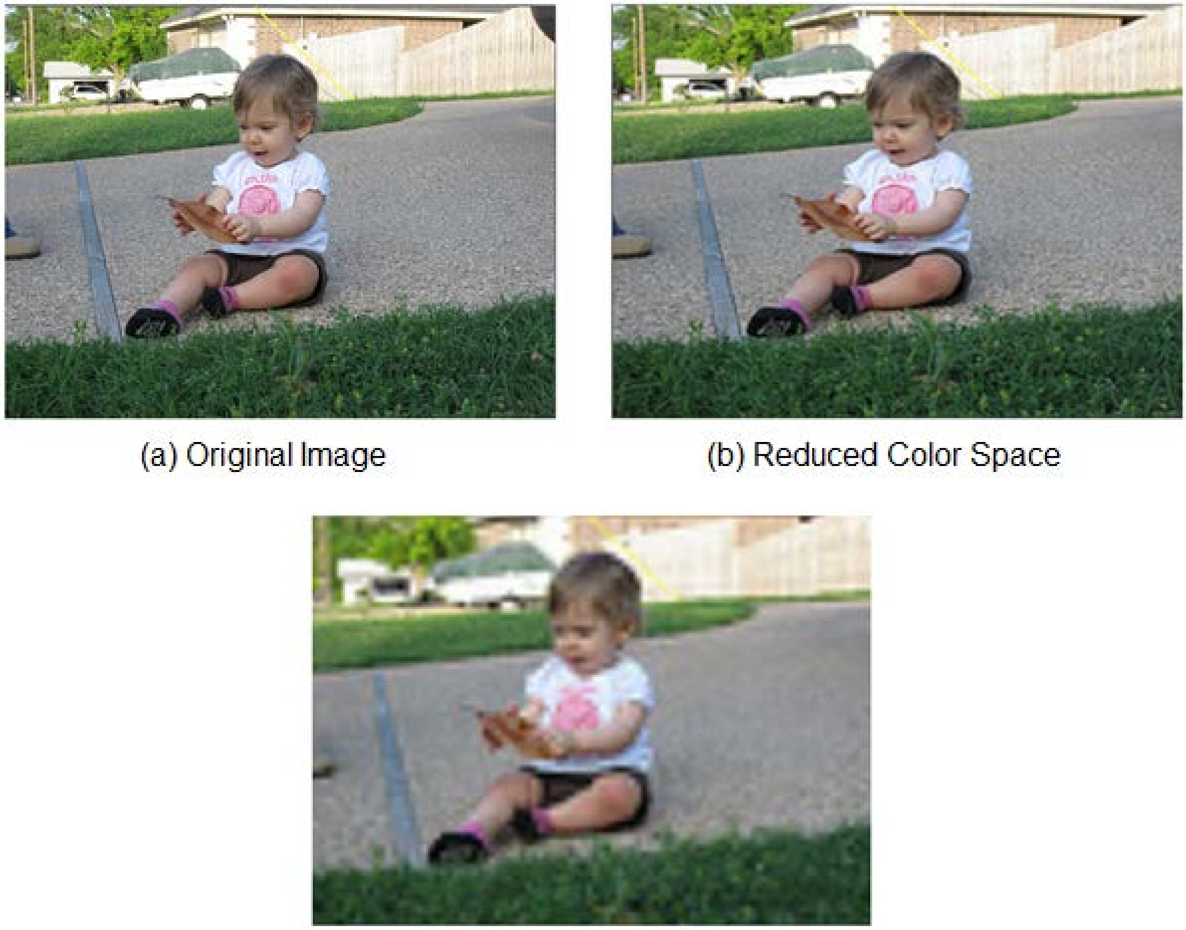
Video Coding helps in reducing the amount of data that has to be transferred. It involves a combination of spatial image compression, fidelity of the image and the temporal motion compensation [18].

4.3.1 Spatial Image Compression

This refers to the relation between different image parts. When an image is transferred, it is broken down into smaller blocks. These blocks belong to the same image. These blocks are transferred over the network channel. Since the blocks belong to the same image, it is most likely that two neighboring blocks have similar image content. In such a case, the video encoder identifies these parts and removes redundant information from the image. This improves the storage space availability.

4.3.2 Fidelity of the Image

Fidelity of image compression is defined as the accuracy with which an original image can be reproduced from the compressed image. Here, compression of image takes place by reducing the color space of an image. For instance, two subsequent blocks of image having similar content undergoes compression by replacing the details of a color with similar colors. In some cases, the resolution of the image can be reduced to reduce the storage size of the frame. By replacing the original color with approximate colors, the fidelity of the image is compromised. The more the replacement of color the lesser the fidelity of the image and greater the compression ratio. The example of fidelity of image compression is shown in Figure 4.3.



(a) Original Image

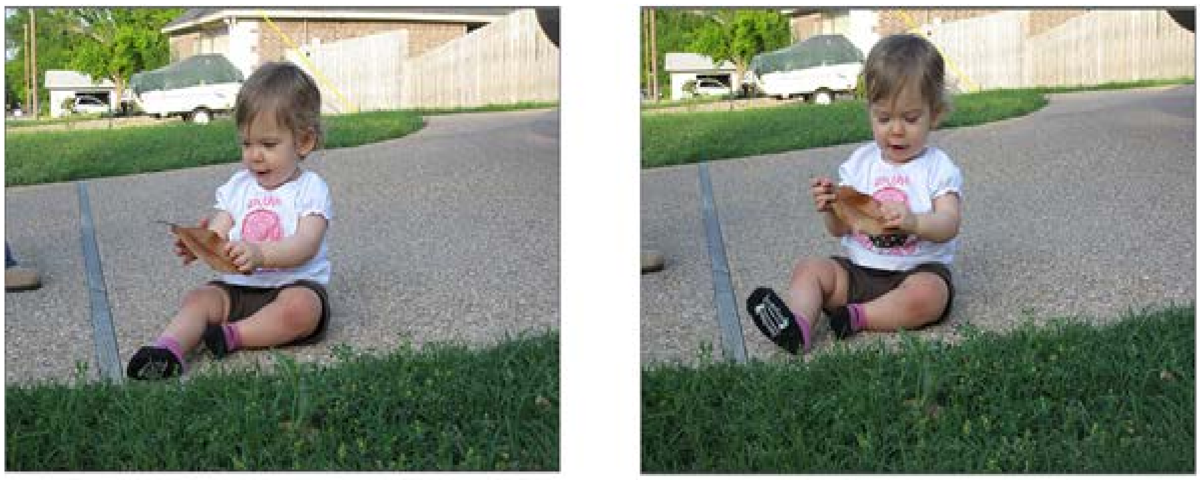
(b) Reduced Color Space

(c) Reduced Resolution

Figure 4.3 Fidelity of Image Compression

4.3.3 Temporal Motion Compensation

This is similar to spatiality except that it is a removal of redundant information between each image. This refers to the relation between images in the video present in a video sequence. Just like subsequent image blocks have similar image content, subsequent video frames tend to have similar or partially similar video content. That is, since the video is broken down to frames, the neighboring frames might have similar or partially similar image contents. It makes more sense for the video encoder to know this and compress the image in such a way that redundant video information on subsequent frames can be omitted. This redundancy tends to be found more in neighboring frames as there is not much change in activity from the previous frame. An example for temporality is shown in figure 4.4.



(a) Video Frame 1 (b) Video Frame 2

Figure 4.4 Temporal relations between video frames

Encoding can be done using one of the two major frames. These are the reference frames and the predictive frames. Reference frames have no relation with any other frames. These are stand-alone video frames. Whereas video frames having temporal relation with subsequent frames are termed as predictive frames. Group of Pictures (GoP) are frames belonging to reference frames. To improve the quality of video compression, on an average, there is one reference frame for every one to two seconds of video footage. This interval between reference frames is termed as GoP length [19, 20].

4.4 Standards of Video Coding

Two major organizations are responsible for developing and standardizing the various video coding techniques available: Motion Picture Expert Group (MPEG) and Video Coding Experts Group (VCEG) of International Telecommunication Union (ITU-T). Various standards have been developed by these organizations over the years. The most prominent among the rest would be MPEG-1, MPEG-2, MPEG-4 and H.264.

- MPEG-1 standard was introduced for video storage e.g. CD-ROM. The bit-rate of this standard is 1.5Mbps.
- MPEG-2 standard was designed basically for Standard Definition television (SDTV) and High Definition television (HDTV). The bit rates of this standard varied from 2 to 20 Mbps. This form of compression was initially used for internet television and IPTV.
- The MPEG-4 standard was more complicated than many other standards and was mainly used for 3G wireless applications. The extension of MPEG-4 standard is MPEG-4 part 10 and is commonly referred to as H.264 [21].

H.264 was a technique developed exclusively for Internet Video Streaming. It is known to produce a similar or even better quality of video for similar data rates as MPEG-2. An example for the bit rates of H.264 for various resolutions of video is shown in the table (4.1) below [22].

Table 4.1

H.264 Data Rates for Various User Requirements

User Requirement	Resolution	Frame Rate	Data Rates
Mobile Video Content	176x144	10-15 fps	50-60Kbps
Standard Definition/Internet	640x480	24fps	1-2Mbps
High Definition	1280x720	24fps	5-6Mbps
Full High Definition	1920x1080	24fps	7-8Mbps

4.5 Bandwidth Streaming

There is a minimum broadband speed requirement for viewing television through internet. This minimum requirement for television, like Apple TV, Google TV etc., is mentioned as 2.5Mbps and that for content of high definition is mentioned as 10Mbps. The thesis primarily focuses on live video streaming of IPTV traffic. The total data transferred during a live streaming at any given time is calculated using the following equation.

$$BW_s = \frac{DR \cdot T_{tr} \cdot V}{8 \cdot 1024 \cdot 1024} \tag{4.1}$$

Where

BW_s = Total data transferred or Bandwidth used for data transfer in (MiB)

DR = Data rates or Encoder Speed

T_{tr} = Time taken to transfer data

V = Total Number of viewers

1MiB = 8*1024*1024 bits (as in equation (4.1)) [23].

1MiB (mebibytes) roughly equals a megabyte. This is a general formula used to calculate the amount of data transferred at any given time for Unicast communication. Since IPTV live streaming mainly uses Multicasting, in which case the number of packets per channel is just one. Assuming all viewers to be watching just one channel, the number of users accounts to one. Also to calculate the amount of data transferred per second, the value of time of transfer is set a 1 second. Hence the equation becomes

$$BW_s = \frac{DR}{8 \cdot 1024 \cdot 1024} \tag{4.2}$$

The above equation holds good to calculate the amount of data transferred over a pipe at any given time. The same equation can be used to calculate the total amount of bandwidth occupied at any given second by knowing the number of channels being streamed through the network pipe.

$$BW_{total} = \frac{DR \cdot Ch}{8 \cdot 1024 \cdot 1024} \tag{4.3}$$

The above equations were used to calculate the amount of bandwidth used up by a particular stream of channel.

4.6 Estimation of Available Bandwidth

In the proposed method for admission control, the GRIP technology proposed in [5 – 7] has been used to gather information regarding the resource availability in the network. Based on the GRIP feedback, either a high, low or medium quality video traffic is delivered to the end user. When a user requests for a new channel, a channel join request is sent to the edge router that takes care of both IGMP for communication between host and router and PIM for communication between routers. When the edge router receives the new join request, it checks to see if it is a part of the requested Multicast (channel) group. If it is, the edge router accepts the request and starts forwarding the requested channel. If the edge router is not a part of the requested group, it sends a PIM join request to the upstream router. This request is forwarded upstream until it reaches a router that is a part of the requested multicast group or is a RP itself. The upstream routers on receiving this request becomes responsible for checking if the path between the decision making router and the edge router has enough resources to handle the new request. To do this, the decision making router generates probe packets called GRIP packets. Unlike the traditional method proposed in [5] where just one GRIP probe packet is generated, here three probe packets, one for each quality are generated as shown in figure 4.5.

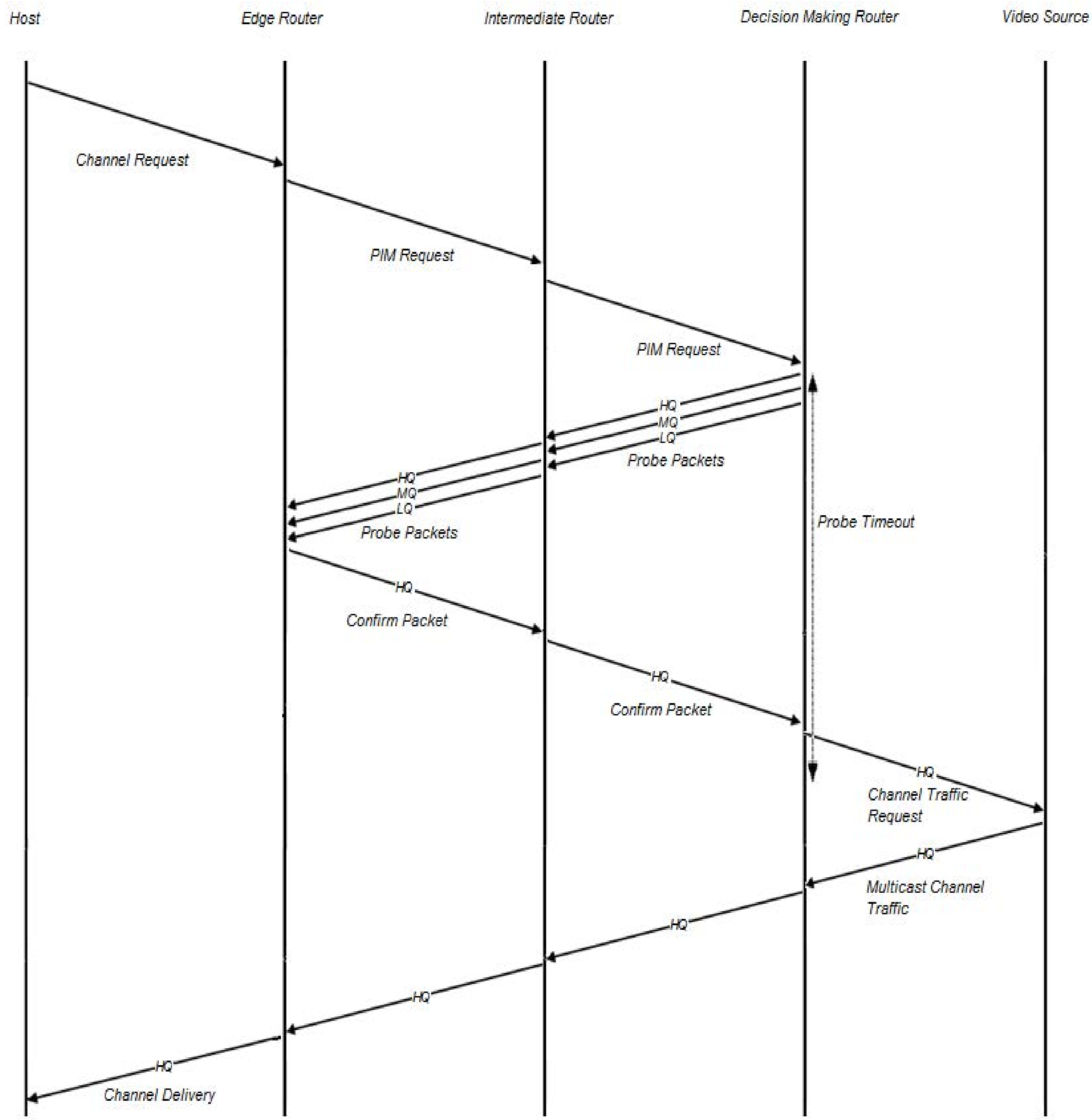


Figure 4.5 Proposed GRIP Operation for IPTV Admission Control

The calculation for the GRIP packet to go through a router is local to the router, the only difference being, based on the locally calculated values, all three probe packets could pass through if the switch is in the accept state for the high quality packet. If the switch in the router rejects the high quality packet, then the medium and the low quality packets are allowed to pass through as shown in Figure 4.6. If the local calculation only allows low quality packets then the other two probe packets are dropped allowing only the low quality packet to go to the next hop downstream router. The steps are repeated on every router in the path to the edge router. When the packet reaches the edge router, it responds back with a confirm packet for the best quality. This confirm packet, when it reaches the decision making router, will know that the path to the edge router has enough resources to handle a new request and hence accepts the join request. It starts sending video traffic based on the confirm response it received. If it receives a confirm packet as response to the high quality probe packet, a high quality video traffic is sent. Hence depending on the confirm response packet, the quality of video traffic is chosen and sent. This way more users/channels could be accommodated in the network. The results of the experiments conducted is shown and explained in detail in the next chapter.

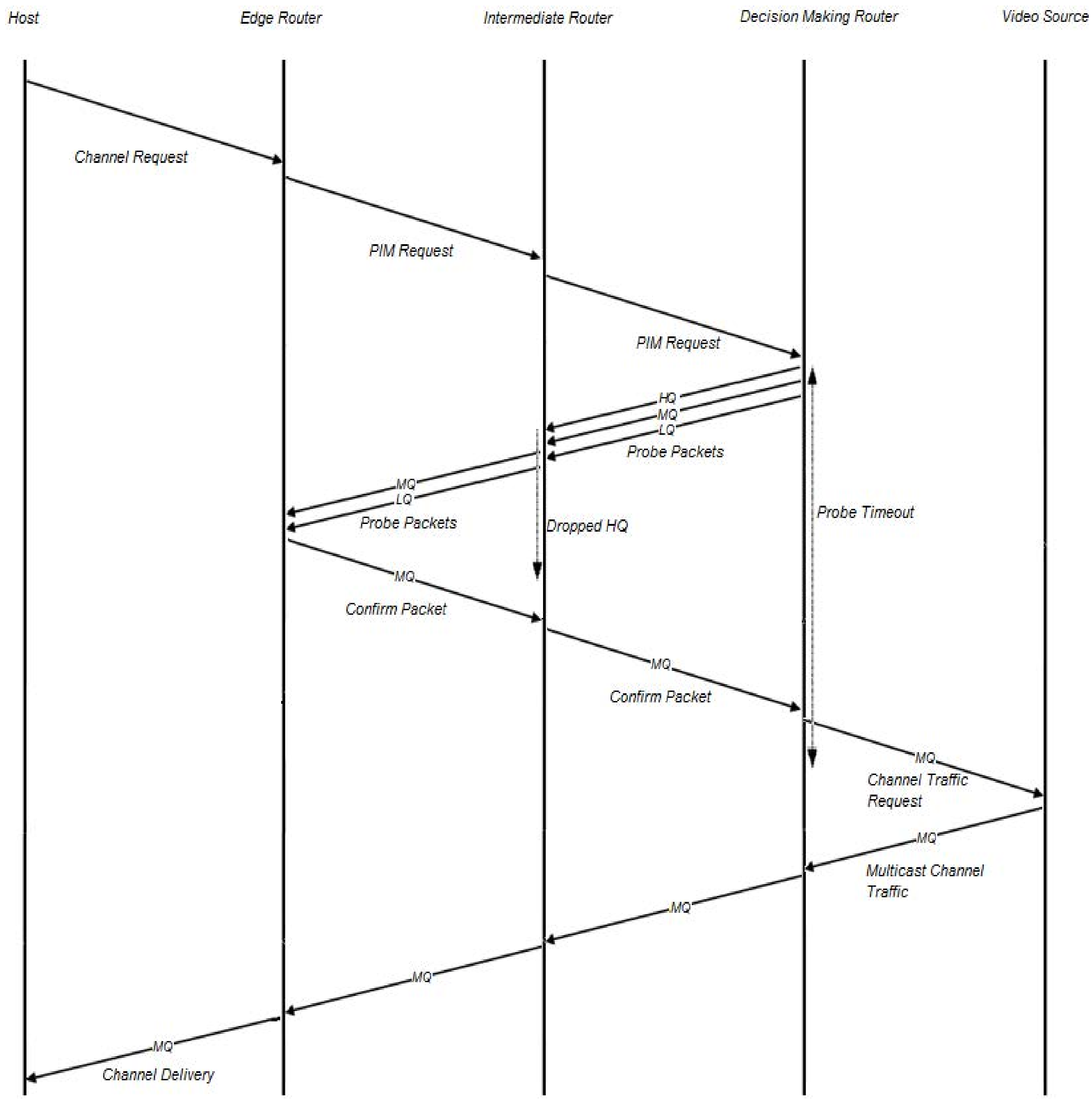


Figure 4.6 Proposed GRIP Operation for IPTV Admission Control during Network Congestion

4.7 Delay and Bandwidth Utilization

Another important factor that has to be taken care of while providing multiple quality of video to the users is to prevent the delay and the bandwidth consumed. Delay factor comes into play every time a request comes in and when the decision making router eventually decides that the end user should receive a quality that is different from what the router is already processing. Then the router has to request an appropriate video quality from the head end server. The time for the process to reach the server and return would be high because the GRIP calculation for this new traffic has to be processed again between the decision making router and the RP. Also, accepting this new quality request only increases the bandwidth utilization in the links. To avoid additional delay and bandwidth consumption, servers can be introduced in the core network connected to the routers as shown in figure 4.7. The servers are capable of buffering the video traffic processed by the router and change the quality of video frames if necessary. This way, when the decision-making router decides on processing the video with a different quality, all it has to do is communicate with the nearest server and have it modify the quality of video. By doing this, both the delay and bandwidth consumption can be minimized. The above mentioned scenario was tested and the results are shown in the next chapter.

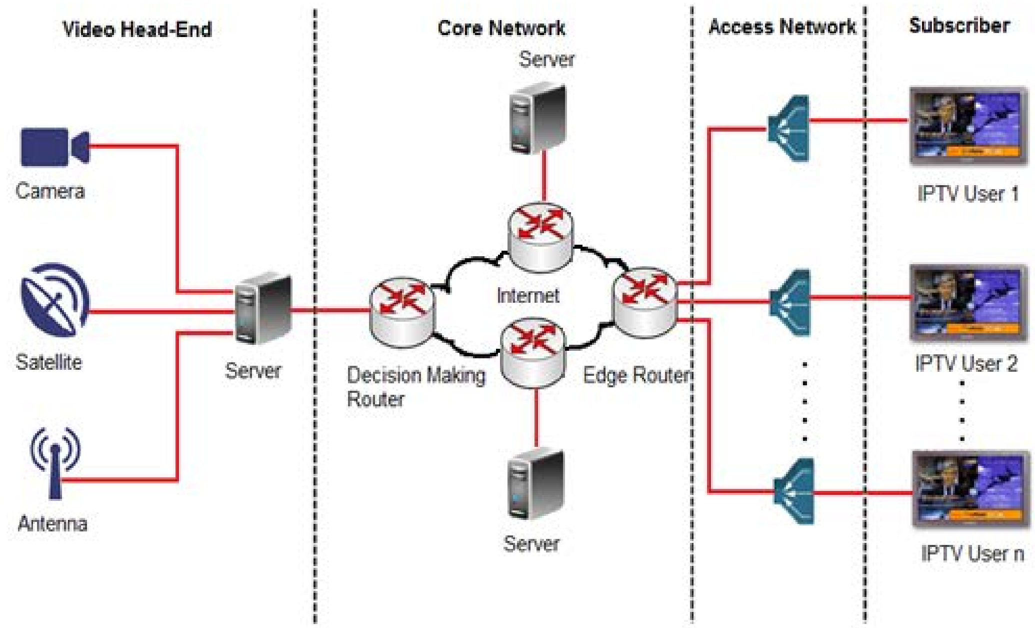


Figure 4.7 Proposed IPTV Architecture

CHAPTER 5

SIMULATION AND RESULTS

The following chapter explains the results involved in the proposed study. Two sets of tests were done in order to verify the performance of the proposed admission control method. The first test involved the existing admission control method proposed in [5 – 7]. The second set of tests involved the proposed admission control mechanism. The two results were compared and graphs were plotted.

5.1 Simulation Scenario

Simulations were done using MATLAB to calculate the bandwidth availability using GRIP algorithm. Two sets of tests were performed.

- The first set of tests was focused on checking the bandwidth consumption when every channel and every user request was responded with the same quality traffic, as per the existing scenario.
- The second set of tests was performed to check the same parameter, bandwidth consumption, when every request was responded with a different video quality based on the availability of bandwidth resources.

The following assumptions were made in simulating the above-mentioned scenarios.

Bandwidth: The total available bandwidth in the core network was assumed to be 2GB.

Data Rates: The data rates and the resolution of different quality videos was assumed to be values equivalent to table 4.1 in Chapter 4. The bandwidth used by every channel stream was calculated using the equation 4.2 in chapter 4.

5.2 Experiment

5.2.1 Case 1

In this case, a network of multiple nodes or routers was assumed for multiple video qualities. This was to show how the existing admissions control method and the proposed admission control method would function in a real network condition. In this scenario, a link’s traffic was studied. This link is considered to be branching out to multiple routers connected to users. It was possible to study how the link, though it had the capacity for the highest quality video, was processing low and medium quality video depending upon the links the traffic was passing through to reach the end user and their resource availability. This shows the efficiency of the proposed method. The simulation was repeated twenty-five times and the average number of channels delivered to the users was found.

5.2.2 Case 2

In this case, a similar network as in case 1 with multiple nodes or routers was assumed but for a single video quality. Again the traffic in the same link was studied. This link was branching to multiple routers connected to users. It was seen how the link bandwidth was consumed when all requests were responded with the same video quality. The results of this analysis are shown in the graphs below.

5.3 Results

5.3.1 Graph 1

For the cases where only a single quality (high quality) video was delivered to the end users, as in the existing scenario, it was seen that for a 2GB link about 2000 channels were accommodated. A test scenario with only low quality video was simulated to check the number of channels that can be accommodated through the same link and the number was found to be around 8000 channels. Then the proposed algorithm was used to calculate the total amount of bandwidth utilization to the number of channels. This test was run multiple times (25 times) to find the average number of channels that can be handled by a 2GB link. It was found that on an average, 4121 channels can be handled by the link. This was found to be very efficient as more number of users can be provided the service with a satisfying video quality. Figure 5.1 shows the necessary details.

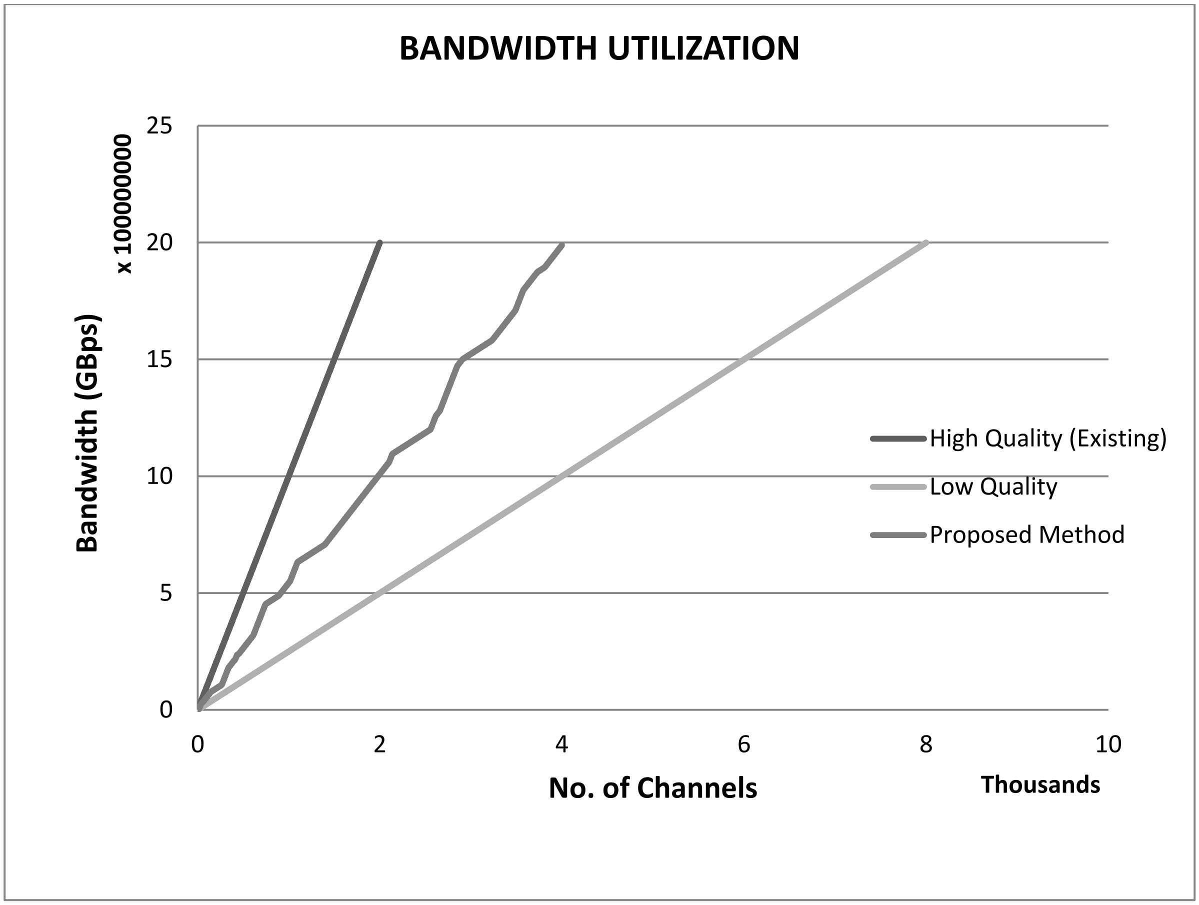


Figure 5.1 Variation in the Bandwidth Utilization and the No. of Channels with different Video Quality

5.3.2 Graph 2

Figure 5.2 shows a simple outcome of the number of channels that are delivered to the end user in the existing method and the proposed method.

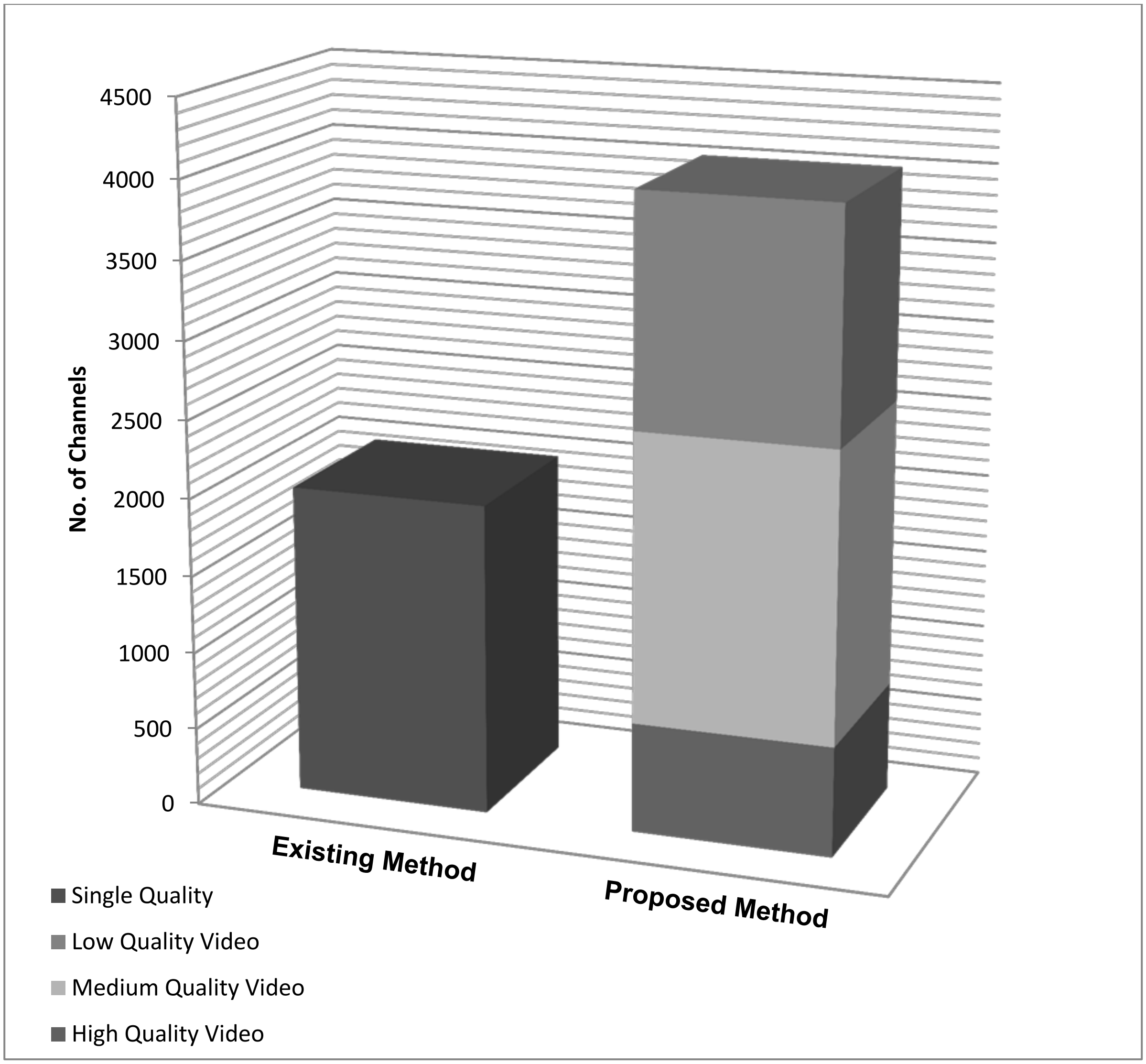


Figure 5.2 Total No. of Channels delivered by the Existing and the Proposed Method

This shows a comparison of the total number of channels delivered by the existing method versus the total number of channels delivered with the proposed algorithm. The different shades help differentiate the quality of the videos delivered to the end user. Table 5.1 and Figure 5.3 give us a better understanding of this scenario.

Table 5.1 Percentage of Channels for different Video Quality

Video Quality	No. of Channels	Percentage
High Quality	707	18%
Medium Quality	1852	46%
Low Quality	1468	36%

Varying the percentage of the channels receiving different quality videos varies the total number of channels delivered to the end-user.

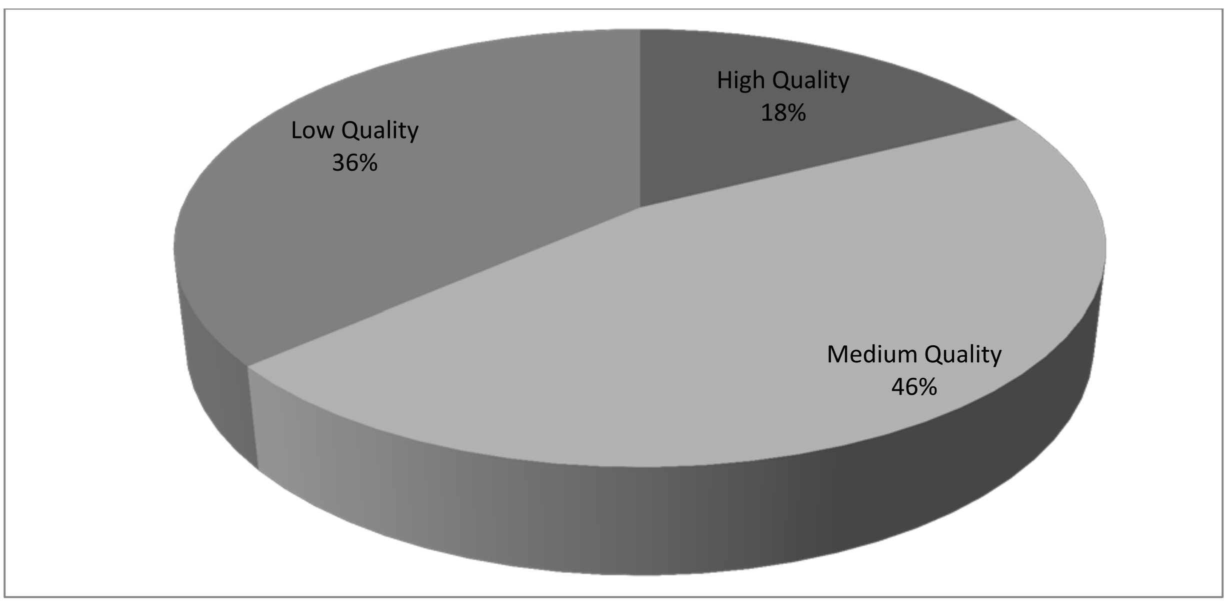


Figure 5.3 Percentage of Channel Allocation for Different Video Quality

CHAPTER 6

CONCLUSION AND FUTURE WORK

In this thesis, an algorithm was proposed for the admission control of the IPTV traffic in core network. This focused on the bandwidth utilization of video streaming. The proposed method made use of H.264 for video coding. In this approach, when a channel request was made by the user, the edge router and the decision making router (rendezvous point) communicate with each other with a help of a GRIP packet to check if there is bandwidth available for the new request. In addition to this, the GRIP packets are generated for different video qualities in order to accommodate the increasing demand of the end-user. A small network was assumed and simulations were done to check the efficiency of the proposed method. It was seen that on an average the number of channels delivered to the end-user increased by 94.32%. It was clearly seen that having multiple video qualities increased the number of channels that can be delivered than having the same video quality.

This method made use of GRIP packets to calculate the bandwidth availability. To get multiple video qualities, multiple probe packets were used. This can be further extended to have just one packet collect the resource availability details from the various routers in its path. Mobile technology has been developing rapidly. The proposed algorithm can further be extended to be used in mobile and Ad hoc networks which are the future of networking.

REFERENCES

REFERENCES

[1] Mohammad Taufiqul Islam and Azimul Hoque, “Study of Reliable Multicast for IPTV Service,” Master Thesis, Blekinge Institute of Technology, Sweden, August 2008

[2] Gerard O’Driscoll, “Next Generation IPTV Services and Technologies,” New Jersey: John Wiley and Sons, 2008

[3] http://www.international-television.org/tv_market_data/global-iptv-forecast-2009-2013.html, Data on Global IPTV Subscriber and Revenue Growth Forecast Retrieved on 04/05/2011

[4] Steven Dearing, “Host extensions for IP multicasting”, Networking Working Group, RFC 1112, August 1989

[5] Giuseppe Bianchi and Nicola Blefari-Melazzi, “Admission Control over Assure Forwarding PHBs: a Way to provide Service Accuracy in a DiffServ Framework,” *IEEE GLOBECOM 2001*, San Antonio, TX, 25-29 Nov. 2001

[6] Giuseppe Bianchi, Nicola Blefari-Melazzi and Mauro Femminella, “Per-Flow QOS Support over a Stateless DiffServ Domain,” Comp. Nets., Special Issue, Towards a New Internet Architecture, vol. 40, no. 1, Sept. 2002, pp. 73-87

[7] Giuseppe Bianchi, Nicola Blefari-Melazzi, Giuliano Bonafede and Emiliano Tintinelli, “QUASIMODO: Quality of Service-Aware Multicasting over DiffServ and Overlay Networks,” IEEE Network, Special Issue on Multicasting, 2003

[8] Olli Alanen, Mikko Paakkonen, Timo Hamalainen, Mikko Ketola and Jyrki Joutsensalo, “Multicast Admission Control in DiffServ Networks,” in *12th International IEEE Conference*, Vol. 2, pp. 804 – 808, November 2004

[9] Yang Xiao, Xiaojiang Du, Jingyuan Zhang, Fei Hu and Sghaier Guizani, “Internet Protocol Television (IPTV): The Killer Application for the Next-Generation Internet”, *IEEE Communications Magazine*, November 2007

[10] Baijian Yang and Prasant Mohapatra, “Multicasting in Differentiated Service Domains”, *Proc of IEEE GLOBECOM*, 2002

[11] A. Striegel and G. Manimaran, “Dynamic DSCPs for Heterogeneous QoS in DiffServ Multicasting,” *Global Telecommunications Conference*, Vol. 2, pp. 2123 – 2127, November 2002

[12] Alan Davy, Dmitri Botvich and Brendan Jennings, “An Efficient Process for Estimation of Network Demand for QoS-aware IP Network Planning,” in *Proc. Of 6th IEEE Internation Workshop on IP Operations and Management, IPOM*, Berlin/Heidelberg, 2006, vol. LNCS 4268, pp. 120-131, Springer

[13] Alan Davy, Dmitri Botvich and Brendan Jennings, “On the use of Accounting Data for QoS-aware IP Network Planning,” in *Proc. Of 20th International Teletraffic Congress (ITC-20)*, L. Mason and T. Drwiega, Eds., Verlag Berlin Heidelberg, 2007, vol. LNCS 4516, pp. 348-360, Drwiega, Springer

[14] Alan Davy, Dmitri Botvich and Brendan Jennings, “Revenue Optimized IPTV Admission Control Using Empirical Effective Bandwidth Estimation”, *IEEE Transactions on Broadcasting*, vol. 54, no. 3, September 2008

[15] Alan Davy, Dmitri Botvich and Brendan Jennings, “Empirical Effective Bandwidth Estimation for IPTV Admission Control”, in *Proc. Of 10th IEEE/IFIP International Conference on Management of Multimedia and Mobile Networks and Services (MMNS)*, Berlin/Heidelberg, 2007, vol. LNCS 4787/2007, pp. 125-137, Springer

[16] Ram Mandali, “Rate Control for Video over IP”, Thesis, Wichita State University, 2007

[17] Yao Wang, Jorn Ostermann and Ya-Qin Zhang, “Digital Video Processing and Communication”, New Jersy: Prentice Hall, 2001

[18] http://planete.inrialpes.fr/~roca/doc/mips03_streaming_part2.pdf, data on Video Coding Techniques and Frames, Retrieved on 06/10/2011

[19] Martin Prins, “Fast Retransmission for Multicast IPTV”, Thesis, University of Twente, 2007

[20] Alan Davy, “Measurement Based Quality of Service Control for Communications Network”, PhD. Dissertation, IFIP/IEEE International Symposium, pp. 827-832, 1-5 June 2009

[21] Thomas Wiegand, Gary J.Sullivan, Gisle Bjontegaard and Ajay Luthra, “Overview of the H.264/AVC video coding standard”, *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 12, no. 7, July 2003

[22] http://edutechwiki.unige.ch/en/Video_streaming, data on Video Streaming Techniques, Retrieved on 05/25/2011

[23] http://physics.nist.gov/cuu/Units/binary.html, data on International System of Units, Retrieved on 06/10/2011