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Improving QoE over IPTV using FEC and Retransmission

Master's Thesis in Electrical Engineering with emphasis on Telecommunication

By

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Tariq & Abualhana

Abstract

IPTV (Internet Protocol Television), a new and modern concept of emerging technologies with focus on providing cutting edge high-resolution television, broadcast, and other fascinating services, is now easily available with only requirement of high-speed internet. Everytime a new technology is made local, it faces tremendous problems whether from technological point of view to enhance the performance or when it comes down to satisfy the customers.

This cutting edge technology has provided researchers to embark and play with different tools to provide better quality while focusing on existing tools. Our target in dissertation is to provide a few interesting facets of IPTV and come up with a concept of introducing an imaginary cache that can re-collect the packets travelling from streaming server to the end user. In the access node this cache would be fixed and then on the basis of certain pre-assumed research work we can conclude how quick retransmission can take place when the end user responds back using RTCP protocol and asks for the retransmission of corrupted/lost packets.

In the last section, we plot our scenario of streaming server on one side and client, end user on the other end and make assumption on the basis of throughput, response time and traffic.

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Chapter 1

Introduction

1.1 Motivation

Broadband access network has given a new interface to usage of IP networks, in fact it made the out of fashion television services i.e. instead of using substitute for distributing channels, a dynamic bandwidth access network makes it possible to go against the old method. IP telephony and television are some paradigms. This gave a huge range of options to us to use some of the very well known internet multimedia applications along with the immense increase in the usage. A window of opportunities was clearly viewed by the ISP's to install Internet Protocol (IPTV) services. A new world of applications was anticipated.

- Full/Vibrant usage of complete bandwidth would give us a wide range of selecting TV channels.
- Different subscribers could come up with different choices.
- Better quality of TV channels.
- Interactional application.

Most of the applications relating IPTV are prone to packet loss. This fact can make a real impact on Quality of experience of the consumers and it should be, somehow, offended. IPTV application and services are spread over IP supported network, which certainly makes use of RTP and RTCP [1]. Thus a packet loss due to the scenario which encodes the video can lead to image destruction quite easily.

How to become flexible against packet loss?

By selecting the following two best techniques, we can surely recover/avoid packet loss, keeping in view their working process.

- Addition of redundant data can make recovery from packet loss quite easier (FEC). Redundant data can be used by the receiver to recover from so called the lost data or packets. This extra data can be either inserted during the encoding process or during the transport.

- Resending/retransmitting the data or the lost packets. Client can ask for the retransmission of the lost data.

Though the above two techniques provide recovery to most of the packet loss, yet they posses some portion of drawbacks stated as under.

- In large multicast networks, FEC cannot be as fruitful as seen because such a network, the packet loss might occur for a small tree and it might be able to occur for a larger network. Thus it means that for some parts of the network, FEC can be very strong which means that the consumption of bandwidth exceeds, which is wastage. Conversely, FEC can be too weak which doesn't offer sufficient recovery.
- To get enough recovery, the protection process should be improved, which will lead to enhancement is bandwidth and affects all the users.
- For retransmission to be fruitful enough, IPTV users need packets to be buffered. As the buffered packets arrive too late, for this reason we need buffer packet so that packet wastage is stopped.
- Users of IPTV expect a witty response from the provided service, so the delay at the start must be as little as possible.

In a large multicast network, to apply the technique of retransmission from the source of TV to the outdoing number of subscribers might not be clever enough just because multiple subscribers can ask for retransmission at the same time instant, which probably can burst and overload the streaming server.

So, for large multicast distributed networks retransmission may not be suitable enough. An alternative can be, to apply the technique to the specific sub trees of the large multicast network. This can be easily adopted by the local networks and it will not also influence the entire network. Also retransmission can be applied to parts of the sub network where packet loss happened.

1.2 Goal/Aim

The main focus of this dissertation is to come up with a better understanding of IPTV and an overview of how mechanism of IPTV multicast distributed network takes place. Limitations of IPTV and making comparison of two famous protocols TCP and UDP and providing an overview of both, how they perform in audio/video streaming. A slight introduction to how packet loss takes place in IPTV and methods of recovery.

1.3 Parameters under focus

- To achieve our goals, we have to focus on following set of parameters
- Impact of packet loss on IPTV's distributed applications.
- Techniques good enough to provide flexibility, error recovery for IPTV streaming application.
- How can we use real-time protocol to provide fast retransmission in IPTV multicast stream delivery.
- Measuring packet retransmission on basis of error resiliency.
- Factors that manipulate the act of RTP retransmission mechanism.
- Kind of network for which the RTP based packet retransmission can be implied to successfully encounter error resilience mechanism.
- Application layer multicast, VoD (Video on Demand)
- Data throughput, application response time, average response time.

1.4 Methodology

We will investigate about the above defined parameters as follows:

- IPTV and transport protocol
- IPTV technologies
- Encoding techniques
- Resiliency and error recovery techniques
- Results of packet loss in a IPTV application
- Measuring feature metric and techniques.
- Using TCP and UDP as model protocols
- Extracting results on the basis of simulations.

Chapter 2

Background

2.1 Overview

IPTV stands for Internet protocol television. With IPTV we usually mean distribution of television/video content over a controlled IP network. The process works by delivering the consumers information through a set-up box, which in a way is connected to its broadband connection. The name here doesn't fully gives the meaning of delivering all the information over internet, it means that only IP protocol is used. Video streaming over internet shouldn't be looked upon as IPTV [3].

Certain key attributes managed by an IPTV service are as [3]:

End to end/partially closed network Services are offered by sole provider which provides means of IPTV services available: Access to content, network composition/ structure, a setup box or a decoder at the user's end, decode and receive data of IPTV content. The services delivered may depend upon multiple distributors offering.

Services are bounded The available services of IPTV are bounded to perform according to the structural composition. The strength of the service depends upon the area where the service provider has control, moreover, sufficiency in bandwidth is observed.

Services are provided by the service giver The IPTV services are used by the subscribers, which are solely provided by the service provider, later on, in future the services may be offered by the end user's as well.

Authorized and admission controlled service before any user can access for IPTV services, it is checked for appropriate authorization, weather the user has the access to make use of the contents or not. The services are only offered when a minimum required bandwidth is claimed and available, otherwise not.

On the other hand the daily internet video streaming [4] has aspects which are for instance:

- The service is open to everybody and anyone with internet access of any bandwidth can make use of video streaming.
- Services can be provided by anybody e.g. a television channel providing a live stream of any movie or an individual making a video for small number of users.
- No authorization/admission required to view video streaming of any kind. No resource reservation or guaranteed delivery of required video etc.

	Broadband TV	Internet video streaming
Footprint	Local (limited operator coverage)	Potentially supranational or worldwide
Users	Known customers with known IP addresses and known locations	Any users (generally unknown)
Video Quality	Controlled QoS, "broadcast" TV quality	Best effort quality, QoS not guaranteed
Connection bandwidth	Between 1 and 4 Mbit/s	Generally below 1 Mbit/s
Video format	MPEG-2 MPEG-4 Part 2 MPEG-4 Part 10 (AVC) Microsoft VC1	Windows Media RealNetworks QuickTime Flash, and others
Receiver device	Set-top box with a television display	PC
Resolution	Full TV display	QCIF/CIF
Reliability	Stable	Subject to contention
Security	Users are authenticated and protected	Unsafe
Copyright	Media is protected	Often unprotected
Other services	EPG, PVR (local or network)	
Customer relationship	Yes; onsite support	Generally no
Complementarity with cable, terrestrial and satellite broadcasting	Potentially common STB, complementary coverage, common metadata	Pre-view and low-quality on-demand services

Fig. 2.1 A common comparison of internet and IPTV

2.1.1 Features

The main difference between IPTV and today's television is that one gets a rare choice of selections between the limited numbers of qualities offered at using IPTV. There is a versatile range of services/options being offered in typical Internet protocol television. A two way high aptitude communication takes place between the user and service provider, which makes a more elaborated service going on between the user and service provider. For instance the user requests for a movie from a TV-guide and it is delivered to the user. Other well known features

that can be provided are interactive and transactional applications which can include a video blog or online shopping. Other prominent traits are as:

- VoD (video on demand) which is a kind of own personal video store with the facility of watching movies of your own choice.
- EPG (electronic Program guide), a full interaction with our own personal needs.
- PVR (Personal Video recorder) with all possibilities of playing, pausing, forwarding and reversing your video
- Personalized advertizing.

2.1.2 Tools

The two main protocols used for IPTV are IGMP version 2 and RTSP (explained in section 2.2). There are of course other protocols as well, for sending live TV multicasting is used. The providers of IPTV in Sweden use MPEG-2 for media distribution. In the table below, it is quite obvious that to view full quality, SDTV would require 4-6 Mbps bit rate [5].

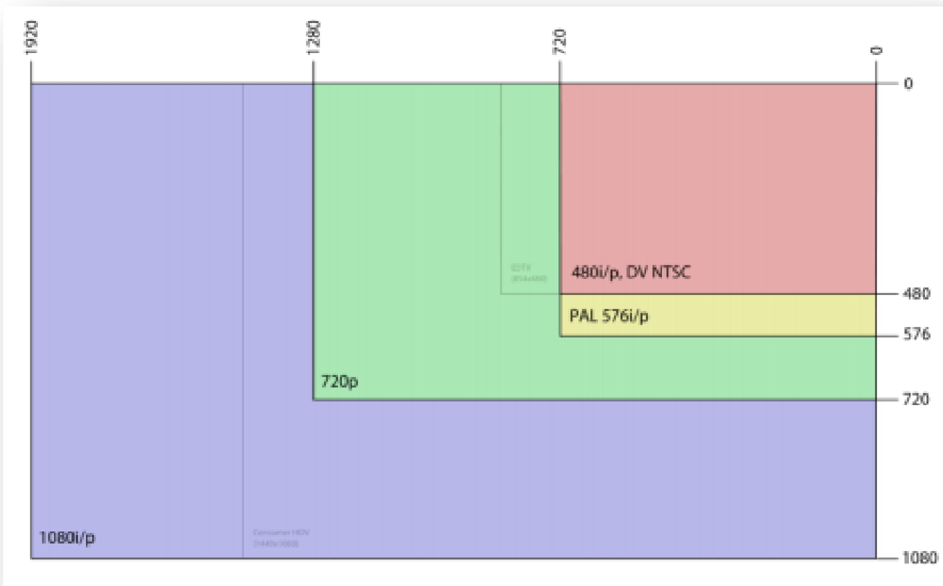
SDTV (Standard Digital Television)	HDTV (High Definition Television)
MPEG-2: 4-6 Mbps	MPEG-2: 20-24 Mbps
H.264, WM9: 2-3 Mbps	H.264, WM9: 10-12 Mbps

Fig 2.2 Common IPTV standards

2.1.3 Pros & Cons

The major drawback is the lack of high bandwidth, though now-a-days it is indeed very common to have a high speed internet connection, yet it is quite a problem in under developed countries, unlike Sweden where 80% of the houses today have high broadband facility. Entire networks have to be built so that point-to-point QoS can be satisfied, which is of course not the case today. Set-top box is an expensive digital portion of the IPTV technology, yet obligatory if one has to view another channel on a different TV. On the contrary, IPTV can provide a higher quality for subscribers. It can be offered in HD format, providing the viewers a different experience of television. In fig 2.1.4 a graphical resolution of PAL, NTSC, SD and HD television is presented. From the graph it is evident that a high definition signal can elaborate information then the current television solutions.

A higher value of services for the users is also delivered. Properties which can be used with the traditional TV are also on list of IPTV's aspects. For instance, pausing of a live TV and then resuming it later on is a feature. A variety of list of channels is another feature along with interactive applications etc. An effective decrease in cost can be observed when services provided of IPTV over the IP network are easily brought down to combine in a structure of ISP. The IPTV services are available when a broadband connection is at service which can be easily provided by the ISP.



[5] Video resolution in various standards

2.2 Protocols and distribution scheme

Our focus in this section is to provide details on the protocols used in IPTV technology; furthermore an introduction to video encoding techniques is given to provide a general concept of how packet loss might impact the IPTV services. Over all the following topics shall be discussed.

- Video encoding techniques.
- Technology used in IPTV
- Transport protocols practiced.
- Sources and outcomes of packet loss
- Error resiliency techniques

The compression and digitizing of all audio/video must be done in order to distribute the material of IPTV. Different multimedia streams which include audio and video and also some other optional streams e.g. subtitles can be sent as single entity or combined. There are advantages and disadvantages of separate and combined delivery. Combined delivery is not as complex because it doesn’t require out of band synchronization. Moreover, the concept of multiplexing at this stage guides to a lessened consumption of network address and ports, which is in a manner an advantage whenever there is a shortage of addresses. On the hand, separate delivery provides lot of elasticity regarding the distribution of a single or multiple streams.

In combined delivery, the streams are to be multiplexed and positioned in a transport container, providing an efficient delivery via streams. The most common type of multiplexing method is MPEG-TS (MPEG- transport stream). This multiplexing procedure provides many purposes regarding audio/video synchronization features, such as synchronization of streams that are transported in a way such that the receiver can synchronize the stream and decides when or when not to display streams. MPEG-TS will also provide us with the property of error correction, which shall be discussed briefly in the upcoming chapters.

The encoded streams are transmitted directly whenever the audio and video streams are not multiplexed. By not adding the transport container or transmission using a suitable protocol, a separate stream delivery is ensured. At last, packets are received by the IPTV user, when they are sent using a transport protocol over an IP network.

2.3 Transport Protocols

A number of protocols can be used for the deliverance of IPTV contents but it depends upon on a number of factors, which are to be viewed before they are implemented. These factors can be reduced down to a number of three by summing up all the facts. Firstly, dependency on the type of video service offered: for instance live broadcasting of a television is a totally different THING in comparison with on demand videos. Secondly, when multiple users receive broadcast contents simultaneously, only a few protocols provide guaranteed delivery of broadcast or multicasting. Lastly, the suitable selection protocol also depends upon two factors of IPTV applications.

1. Delay
2. Latency

So, keeping in view the above mentioned facts, the following protocols are adopted as a mean of practicing IPTV applications. We will discuss the above protocols and there structure, how they make up together to perform the audio/video streaming.

- *Real transport protocols* Transport Control Protocol(TCP),User Datagram Protocol(UDP) and Datagram Congestion Control Protocol(DCCP*)*
- *Application layer protocols* Microsoft media Server Protocol (MMSP) and Real-time transport Protocol (RTP)

2.3.1 Transport Control Protocol (TCP)

TCP is mostly used for reliable data transmission, in other words a connection oriented protocol which uses full-duplex association for reliable transfer of data [7]. A unique process in TCP is that the receiver can process and receive sufficient data or as much as it is required in time by the receiver by two ways. One is by means of sequence numbers in which TCP provides arranged delivery secondly a flow control mechanism which ensures the amount of data that can be absorbed by the receiver. TCP can adopt congestion and provide reliable data transfer by the two mechanisms opted i.e. a packet retransmission mechanism and congestion avoidance mechanism. This utility however imposes couple of restrictions on distribution of streaming data: TCP provides us with the surety of data delivery in time. If data loss or packet takes place, then for sure the receiver would wait for retransmission of data, in the meanwhile the buffers may under run. If TCP adapts to congestion, a constant throughput cannot be assured. This means here that the application receiving the contents needs a buffer to settle in the dynamic

transfer throughput. Moreover, TCP establishes the connection by three way hand shake, which is time consuming. These aspects make TCP become suitable for applications that necessitate low latency content delivery and fairly not suitable at all for applications that require lost of data over high latencies.

2.3.2 User datagram Protocol (UDP)

UDP is a connectionless protocol with limited functionality, which means that the sender and receiver have no dynamic connection [8]. It means that this connection type doesn't provide us reliable data delivery, flow control, congestion control, suiting of data rate to capacity of network and receivers processing speed. In UDP transmission, the sender only determines the transfer rate and no idea or process to make sure that the data is received by the receiver or not. Also there is not feedback on transmission control. By using broadcast or multicast, UDP oriented connection send data to multiple users. Unlike TCP, UDP doesn't take time to establish connection, which doesn't lead to delay in data delivery and transfer.

2.3.3 Datagram congestion control protocol (DCCP)

DCCP is a recently developed protocol, which fuses together the concepts of TCP and UDP. It provides controlled yet unreliable delivery of datagram over bidirectional unicast connection. DCCP is eccentric in a way that it provides a tradeoff between timing factor from UDP and congestion control from TCP, thus making it suitable for applications that have strict timing constraints yet benefitting from congestion control e.g. Voice over IP and video streaming.

2.3.4 Microsoft media server protocol (MMS)

Microsoft's MMS protocol is a collection of protocols used to stream multimedia from streaming server to media player. It can use UDP, TCP or RTP for the delivery of the contents. The protocol is however closed, but it is used by several other multimedia applications such as VLC and Winamp as the protocol supported only Microsoft's media applications.

2.3.5 Real time Protocol (RTP)

The protocol shall be discussed in the later section. [1]

2.4 Content Distribution

Four methods are used as for distribution of content in IPTV service. These four methods are stated below, depending on the type of application to be used.

1. *Unicast distribution*
2. *Multicast distribution*
3. *Peer to peer distribution*
4. *Hybrid distribution*

2.4.1 Unicast Distribution

Unicast distribution protocols: UDP, TCP, RTP, DCCP and MMS are the common choices used for Video on Demand services because protocols like TCP are used where high latency is required along with the factors of reliability and connection orientation, So low latency such protocol is not used for services that require low latency. What happens is that, an IPTV user connects itself to a streaming server for IPTV contents. Once the connection has been established, the users then continue using the IPTV content. Prerecorded data can also be sending as disintegrated data. In this case, the rate of data transferred is higher than the required rate to receive the application.

For broadcast TV, unicast distribution is ineffective because of the services provided by the service provider for IPTV. For instance, N users need N numbers of IPTV streams are to be transmitted over the same network.

2.4.2 Multicast distribution

For efficient delivery of IPTV to multiple clients of IPTV, multicasting distribution is preferred for such simultaneous process e.g. delivery of a live broadcast.

A combination of UDP and RTP is together used as transport protocols, as UDP's limited functionality for low latency multimedia content distribution; RTP is fused with it providing an availability of feedback mechanism.

Classically, TV channels are multicast in the core network and are remain in the access network, not forwarded unless clients request for the respective channels. Access network has limited bandwidth capacity as compared to the core network just because of the fact that the IPTV stream is only forwarded to the user when the user has made request for it, otherwise not. An IPTV service provider can offer more channels, which technically possible with the traditional broadcast TV. The only disadvantage of this mechanism is that in IPTV with limited bandwidth, the channel is present, only the stream has to be requested, while in the traditional broadcast channel is present in user's range

To enable multicasting, two protocols of immense importance are to be activated. Protocol independent Multicast-Sparse mode [11] (PIM-SM) and internet group membership protocol (IGMP) [12]. PIM-SM is a routing protocol for multicast group; it executes some of the most important multicast tasks. Also allows routers to alert each other for number of available multicast channels and multicasting routing functionalities, including of setting up of a new

multicast path from a source to multi content seeking receivers. IGMP is a subscription protocol which allows clients to subscribe to multicast groups by means of sending membership reports. Access node routers make use of IGMP protocol to determine which users are interested in definite multicast group and also determine which packets are to be send from a specific multicast group, to the which routers port.

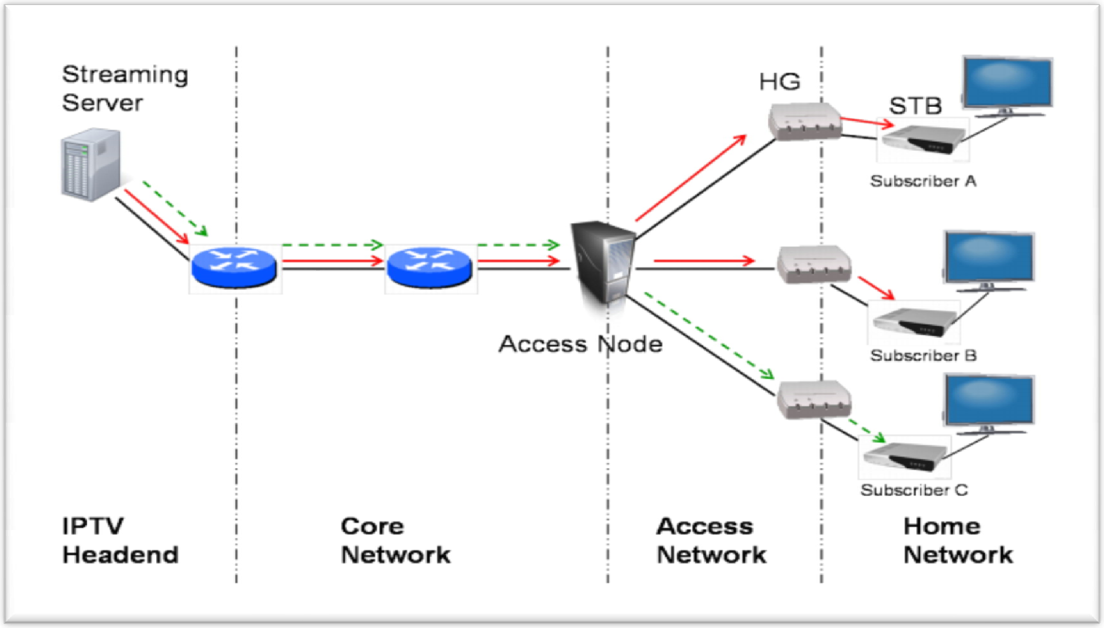


Fig 2.4.2 Multicast environment

In the above figure, a typical example of IPTV distribution network is provided. The whole stream is multicast from the streaming server to the Set top box (STB). STB is equipment with the user that decodes the video stream and plays it to the TV. During the whole process, the video stream navigates across three different networks; Core network, access network and home network. Core network is maintained by the service provider, access network which is the middle connection between the service provider and home network and lastly, home network is the network found in users region. Access network and home network are interconnected by a home gateway (HG). HG is a component which allows home devices such as PC or STB to get connected with the outside world. The access network and core network are connected by Multi Service access node (MSAN). MSAN is a device which assimilates different multimedia services like telephony, television and internet on a single platform, from there offering connections to different access nodes. In the figure, a stream is received by subscriber A and B, while the other is forwarded to C. When a user requests for IPTV service, the STB will issue a request to its belonged multicast group by means of IGMP report. Clearly home gateway HG will receive this

information, when HG will receive the packets; the packets are forwarded to the user. Otherwise it will forward the request to MSAN. The MSAN upon the reception of the request, will forward the packets to the requested user. The stream shall be received by STB and displayed.

2.4.3 Peer-to-peer distribution (P2P)

A new and upcoming technology is the use of Peer to peer overlay networks in IPTV technology to distribute IPTV contents from a service provider to IPTV clients. In a p2p overlay network, the contents of IPTV are distributed partially or entirely amongst peers. One interesting thing in this scenario is that the client receiving the IPTV content will not only be consuming data, but also allowing it to be shared by other peers, which are interested in data. If we look at this distribution from operator's point of view, it is highly preferred by the operator, as it is a cheap distribution technique because the bandwidth required to view the IPTV contents is shared by the IPTV nodes and the network becomes highly balanced.

For streaming in IPTV applications, users would like to watch the streaming TV right from the moment when they start using the application, unlike the packet distribution in typical p2p applications e.g. Bittorrent [13], Gnutella etc in which the order of the data received is not important as the user would directly use the file when the transferring of data is done. In IPTV, this implies that the order, in which the data is received, is important; a user here is only interested in data that is immediately followed by data currently being decoded and displayed. Thus for this kind of application, a tree is needed with nodes receiving data from higher nodes in the distribution tree. This means that a large playback interval can infer between the nodes transmitting the contents from top and the nodes that are at the edge of the tree. This also means that the dynamic availability of resources is also very dynamic as the users of IPTV are constantly joining and leaving the services. A large pre buffer is needed here to avoid buffer under run due to the dynamic behavior.

2.4.4 Hybrid distribution

Hybrid distribution is also very common now a day. Hybrid distribution combines content distribution methods to optimize content delivery. The two factors to be optimized are startup delay and network distribution costs. In multicast IPTV start delay is caused by the IGMP subscription. It can be reduced by starting up an IPTV connection via a unicast method with a streaming server. This will setup a connection very quickly and the data would be received as fast as possible. While the content is being displayed by the IPTV, client joins the multicast group and once the data from multicast group is received, the client can then switch from unicast stream to multicast stream.

The same rule can be used to lessen startup delays for p2p based IPTV distribution. In this case, the streaming server can be used for operation of two functions: one is to allow small startup delays and secondly function as a backup data resource, for which IPTV can directly connect to streaming server for the missing part of the stream.

2.5 Real-time Transport Protocol

The real time transport protocol RTP [1] is type of protocol designed for the transfer of real time applications over internet. RTP was designed to support real time applications which have low latency e.g. telephony, video conferencing and IPTV. Along with UDP, RTP also supports TCP. UDP runs on top on RTP [8]. RTP neither provides guaranteed delivery of data nor it provides any reliability, but its specific feature is the streaming of multimedia data.

2.5.1 Components of RTP

- Real-time data transport

  It can be a combination of audio/video streams as a single or both. For the transport of the stream, a transport protocol such as UDP or TCP is needed. Support of UDP is mandatory, TCP is not a must.

- RTCP (Real time control protocol) for monitoring and signaling of a session.

The edge that RTP has over other protocols is that it provides means of transport for multimedia streaming and it is used for low latency applications such as IP telephony and video conferencing. RTP provides support for multiple stream transmission at the same time, which provides flexibility in delivery of single or combined audio or video streams and the synchronization factor in RTP provides room for streaming situations. Above is the header format of an RTP protocol. It has certain fields that make RTP useful enough to support low latency multimedia applications.

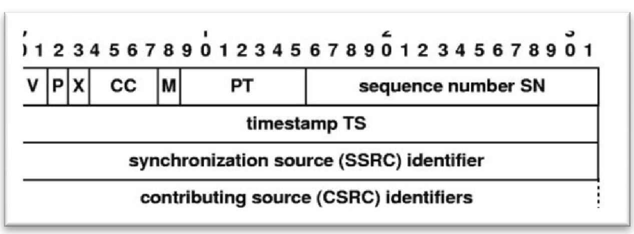


Fig 2.5.1 RTP header format

- *Payload type PT (7 bits)* Used to define several types of existing payload type's along with a variety of dynamic payload types. See section [14] and [15].

- *Sequence number SN (16 bits)* is used to send and receive packet loss and provides guarantee of recovery of lost packets by reordering them as they went out of order.

- *Timestamp TS (32 bits)* is used for providing synchronization of multiple steams, enables receiver to playback the samples at appropriate intervals and calculate jitter between the sender and the receiver.

- *SSRC Synchronization source (32 bits)* the source of RTP packets. SSRC contains a 32-bit randomly created identifier such that all the members of RTP session can uniquely identify the source of RTP packet stream without depending upon the network address. This is a convenient because RTP packets may get combined or mix up during the transport.

- *Contribution source CSRC* RTP uses CRSC to know the source of all the packets that were mixed up from different streams by a source.

2.5.2 Real time control protocol (RTCP)

Real time control protocol provides a check process on sender and receiver involvement in the RTP session. Monitors RTP's function for the ongoing session, mechanism for providing identification of the participants involved in the transport and least control over RTP session. For this reason RTCP provides reports on sender and receiver:

- A sender report is used by the senders to provide statistics on transmission and reception.

- Whereas, a receiver report provides stats on a member that is not involved in updating the data.

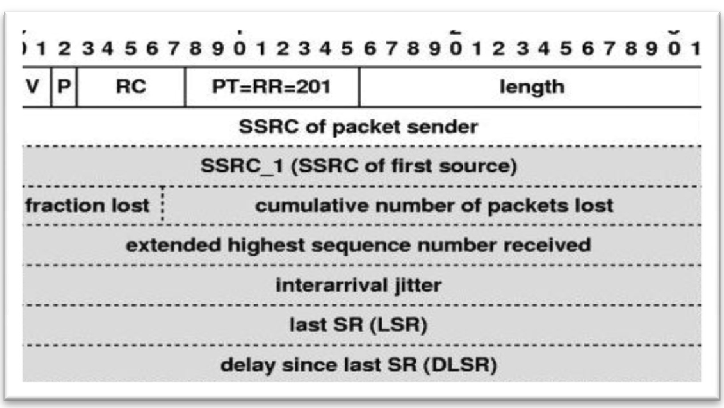
RTCP reports are sends out at periodic intervals in the session to all the participants. By sending out RTCP reports every participant keeps a record of number of members in a session and therefore it can calculate its share of RTCP bandwidth and interval of RTCP report. There is however constraints on applications provided by the RTCP, such as; higher the number of participants joining the session, higher the interval for transmission of RTCP reports. Thus for

group with larger number of broadcast group, the feedback process will subtle because feedback process duration will be too high to provide and detect solutions to problems. RTCP reports are useful because the problems during the RTP session can be easily identified, reported and solved very quickly. For example, the sender could easily reduce the transmission rate of packets when receiver is facing a huge packet loss. A figure of receiver report is shown in the upcoming topic, which informs the receiver about the packets the receiver received, packets that were lost during the transmission and jitter occurred between the arrivals [16].

2.6 Session classification

Before the start of an IPTV session, it is important to let the receiver know about the number of available streams, how the stream shall be delivered and the important constraints associated with the streaming session. All the required info and the parameters shall be exchanged in advance and before the establishment of the session. A common protocol for describing the session durations of the multimedia is the Session Description Protocol (SDP) [17]. The description of SDP contains media and transmission properties along with, maybe, a description of the content. To send and receive SDP information and contents, different protocols are used

- Session initiation protocol SIP
- Real-time streaming protocol RTSP
- Hyper text transfer protocol HTTP
- Session announcement protocol SAP



[20] Fig 2.6 RTSP structure

HTTP [18] is usually used for transfer of data over World Wide Web (www) and can also be used to sporadically reclaim information on streaming sessions. The session announcement protocol (SAP) [19] gives notice of multicast session via multicasting. Those entities which are keen to extract information about the available sessions listen to a familiar multicast address to receive updates provided by the SAP. The SAP description is announced after regular intervals. The Real time Streaming protocol RTSP [20] is used to control streams of continuous media such as audio/video. RTSP is described as a remote control for the network of streams, allowing the applications users to pause and resume the stream and search for particular contents. RTSP message syntax is similar to HTTP syntax. The session initiation protocol (SIP) [21] is used as a signaling protocol used to create, modify and terminate streaming sessions with one or more service users. SIP protocol is also used for instant messaging.

Chapter 3

Texture

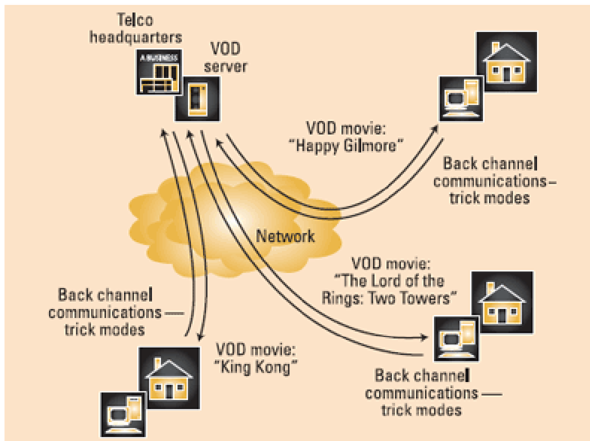
Before we explain the concept of multicasting in detail for IPTV, let us provide a simple discussion where we would be able to make a clear difference between choosing either Unicast or Multicast for IPTV. Some of the very fine aspects of IPTV are to be focused in this chapter. Before we proceed forward into providing details on multicast, let us say on what are we going to provide in this chapter. A unique discussion on defining multitask process in IPTV would be a major task under focus in this chapter, including a list of techniques and providing a quick review of how multitasking takes place in IPTV environment. Architecture, design of a multicast environment, protocols used a service model and how all these concepts merge at application as well as network layer, shall be discussed.

3.1 Why Unicast and multicast?

Our main focus in this chapter would be to provide architecture for reliable delivery on IPTV stream, focusing on different parameters. IPTV was designed to support primarily unicast protocol, to move data from a single source to single receiver. However, it can also support multicast, with more than one destination to deliver the service. IGMP more commonly *Internet Group Management Protocol* manages [12] the entire multicast in IPTV stream. Every piece of document provides a long discussion on IGMP but in our refined work, we shall focus on Application layer management (ALM) and topologies associated.

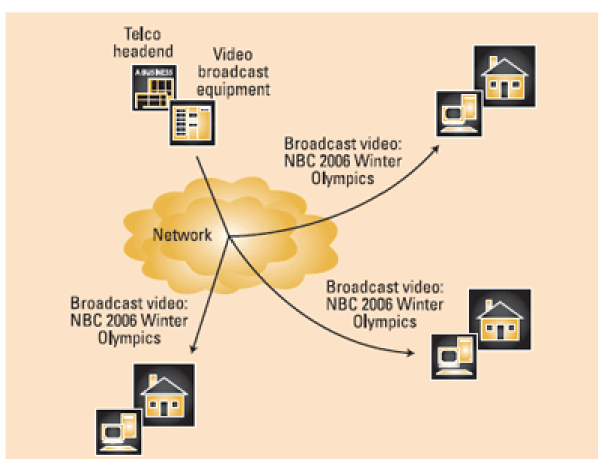
Let us consider multicast, when we compare it with unicast, it is obvious that the stream can be delivered to many users at the same time in multicast [23]. Every channel that has to be broadcasted has a unique multicast group. Using IGMP the end users receive broadcast packets and then the entire stream via the network. One drawback in multicast is that it there is no reliability to packet loss, once a packet is lost there is no such thing as error recovery.

The figure below provides multi users are using an IPTV stream via broadband setup. Each end user (home) in the above figure is a part of multicast stream delivered by the IPTV server. This scenario shows how multi users access different demands at the same time consuming bandwidth.



[22] Fig 3.1 Broadcasting in multicast

Figure below provides a contrast to a multicast environment, unicast Video on Demand (VOD) [6] as described in chapter 2. IPTV stream is sent from a VOD server to an end user. There is a separate content delivery stream for each user to deliver VOD. If the above stream is to be delivered as an MPEG [26] or H.264 [27] a higher stream of data may be required. So an obvious facet of unicast protocol is that bandwidth consumption for a particular network can increase instantly.



[22] Fig 3.2 VoD in unicast

3.2 VOD in Unicast

At this stage we will discuss a specific unicast feature VoD (Video on Demand), which would provide a clear idea of why we used unicast technique for above designed cache. IPTV uses multicast and VoD uses unicast technology. Unicast scenario is unique, because it has a single IP address and associated with a single IP is a single user. However, if the same data is to be sending more than once to the same unicast address, then the data has to be sending again and again for the same user. To support either unicast or multicast, the end user needs to support two major protocols: RTSP [20] (Real-time streaming protocol) for VoD and IGMP for multicast.

RTSP is used for two purposes, to establish and to control one or single streams of continuous data e.g. audio and video. However, RTSP is not able to deliver the continuous data itself rather it uses the services of RTP to transfer streaming media but its transport mechanism is not influenced by the way how continuous data is carried out.

IGMP on the other hand is used in multicast to carry out the management of the entire IPTV network. The entire core and access network is provided diagrammatically in chapter 2. IGMP is used in an efficient way in an IPTV environment by providing ease for which leave and join the network simultaneously.

For both techniques, weather VoD or IPTV, some very well known common algorithms are used. Some of them are discussed in chapter 2, Mpeg-2 [26], Mpeg-4[26] [28], H.264 etc. In all these cases the compressed as well as the encrypted data is send via using MPEG-TS (also discussed earlier in chapter-2).

3.3 Unicast metrics

As we have already discussed some of the important metrics that are required in making a retransmission mechanism more efficient, some common metrics at this stage of an optimal unicast/VoD scenario can also be discussed [29].

3.3.1 Packet loss

Loss in packet takes place when a single or multiple packets fail to reach the desired destination. If the packet loss is related to I, B or P frames, well judged opinions can be made here. If packet loss is related to I frames, the chances are that the video signal can be distorted. If the packet loss is reclaimed in B or P frames, the chances is that less distortion is observed but still image can be damaged.

3.3.2 Packet jittering

Packet jitter is estimating on variance of the data arrival time and interleaving of RTP data packets which are calculated on the basis of RTP timestamp field [15]. It can be defined as the mean difference of time when packets are sent by the server and then received at the destination.

3.3.3 Delay

Networks that are usually packet based, the aspect of delay always exist as it is common for packets to change their routes and sometimes may arrive at different time intervals to be out of order. RTP protocol tries to reduce this problem. As every RTP comprises of a sequence number, it is not possible for the packet to be lost and they are well placed for decoding.

3.3.4 PID

Every TS part of MPEG has 188 bytes fixed length, 4 bytes for the header and the rest of 184 bytes for payload. PID here comes from those 4 bytes and it carries a unique identifier for the type of packet or payload carried by MPEG-TS. Every video/audio needs to have a unique PID. This allows the STB to carry out sequenced decoding.

One thing has to be made sure at this stage for later procession that PID is tasking is carried out systematically and there has to be a precise difference between Payload structure identifier (PSI) and the video/audio stream that has to be reconstructed at the STB for its audio/video contents.

3.3.5 PCR (Program clock reference)

TS as said above, being used as a part of MPEG, contains seven packets of 188 bytes each. In these TS streams are clock synchronizers, which are referred to as PCR, are simultaneous values of 27 MHz system time clock which is located at the mpeg video encoder. A problem here with the jittering of the PCR is that if the PCRs are not received at regular intervals, then the associated clock may jitter.

3.3.6 MDI (Media delivery index)

RFC 4445---is a counting mechanism that transports good or better quality of video in a network by combining jitter and packet loss ratio together, regardless of the encoding techniques. MDI can be used as a tool or as a quality indicator for such networks which intend to deliver quality media such as video, audio, VoD, VoIP etc.

3.3.7 Delay Factor

DF or the delay factor is the maximum delay that is noticed at the end of each IPTV stream packets, when they start to move and at the time when they fully get consumed. This is rather a constant traffic rate at which traffic is consumed by the stream or the certain chunk of packets in the stream. Greater DF values indicate network latency required by a certain stream to be delivered (RFC 4445)

3.3.8 Media Loss Rate (MLR)

Media loss rate is the calculation of packets that are loss or out of order in a certain stream over a specific period of time. These packets are called flow packets and they carry information on streaming application.

Though it is quite clear that unicast is more effective than multicast, but the future of this technology lies within multicast environment because in today multicast is more favored by the server/user environment for video distribution. In the coming sections, we will describe a satisfactory multicast environment for IPTV and discuss how IGMP and PIM protocol work together in IPTV based scenario.

At first we will talk on why multicast is reliable enough for an IPTV environment and how it relates to provide end users the quality they need. Multicast sprouts as an important element when it comes to deploy networks such as IPTV. It provides guaranteed delivery in case of distinctive source applications as IPTV. But reliability in multicast environment for real time applications has always been an existing fact. Betterment in providing QoS for IPTV is necessary. When it comes down to end users to view IPTV, whether it is VoD or IPTV in multicast environment, they always demand best of quality. For a new user to IPTV, the service is really new as well as experimenting. For them, leaving the traditional TV and moving to the new internet-based service might be a shift in expenditure as well as taking chances on making substitutes. Some extra qualities must be provided along QoS, with reducing delay, jitter and congestion.

- Service without interruption
- Corroborated documentation on services associated with IPTV like VoD, TV, broadcasting etc.
- Fast transmission of service and fast re-transmission of service in case of failure of network or if goes down.
- Ease of using the hardware and Low cost.

3.4 Why choosing IP multicast over Application layer multicast?

The above discussion shows that users in IPTV expect more in order to make them quit or at least consider IPTV as a modern life alternative. In the recent years, there has been a quite drastic improvement in ALM (Application layer multicast) for using in multimedia applications and services like IPTV aswell. ALM is differentiated from IP multicast [24] on couple of facts which really provide it an edge. For instance ALM doesn't require facilitation of routers and structure of network, though highly cost-effective yet easy to install. Following are some major differences which draw a clear line between ALM and IP multicast [25].

- In ALM, there is no need of routers to maintain information of every group associated with each router in case of IP multicast. Rather this state is maintained at the end user. In case of IP multicast maintain this information at the routers increase transparency and complexity.
- ALM consumes more bandwidth than IP multicast procedures. The reason for this fact is that IP multicast is implemented by routers and instead of sending the same packet over again, it constructs a tree with best possible packet forwarding approach. On the contrary, in ALM no routers are used and packets that are to be send are resend on the same link multiple times on the end user. So this consumes fairly more bandwidth.
- As compare to IP multicast, ALM superimposes a network on top of existing network that is to carry out unicast.
- For delivering multicast data and routing information, a routing multicast tree is constructed. Any single host that enters or joins an IP multicast session informs the concerned router. This status is then shared amongst all the routers via the tree. All this information is carried out at the tree about the user status is carried out through IGMP protocol.
- ALM is more reliable than IP multicast, as reliability is an important factor for a packet sensitive service like IPTV.
- The absence of routers in ALM is drawback when we compare some important facts like latency, low efficiency which are only known in traces to the end users. So the end users in case of ALM have a very rare or no knowledge about what is going on in the network structure.

- Multimedia applications require reliability and in-time delivery of streaming data. In case of ALM more possibilities are offered for error and congestion control for the reliable delivery in multicast group.
- ALM can be easily implemented within the current internet infrastructure as compared to the IP multicast. Why? Because IP multicast requires routers installation of multicast behavior.

By far points, Application layer multicasting is more advantageous than IP multicast system. The reasons are immense and a glimpse is provided. Even in case of IPTV, one can easily deploy this system of guaranteed delivery of streaming data by focusing on certain factors. We will proceed with providing implementing different topological scenarios implemented in IPTV environment.

3.5 Application layer multicast protocols

We will provide a synopsis of all the protocols that by now have been implemented since the very first day when p2p was first implemented on application layer. In an Application layer multicast environment, end users handle all the information regarding which peers want to become a group member and making copies of packets etc. The end system maneuvers itself in such a way that it behaves as a rational overlay network end nodes as edge nodes for multicast to receive the message once, so the network would send all the users a single message once data in unicast environment. Figure shows the architecture of ALM.

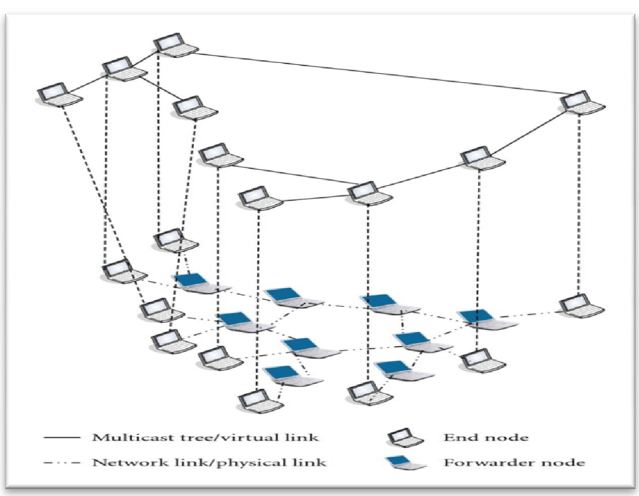


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Fig 3.5 End node and forward node

3.5.1 Application layer multicast types

Over the years many drawbacks have appeared after implementation of ALM in a network. Solutions to these drawbacks have shown up as awareness for topology multicast in an overlay network. This is apprehended to favor multicast environment somehow, depending on the selection of nodes. This can be classified into the following three ways [31].

I. Application layer multicast for overlay network

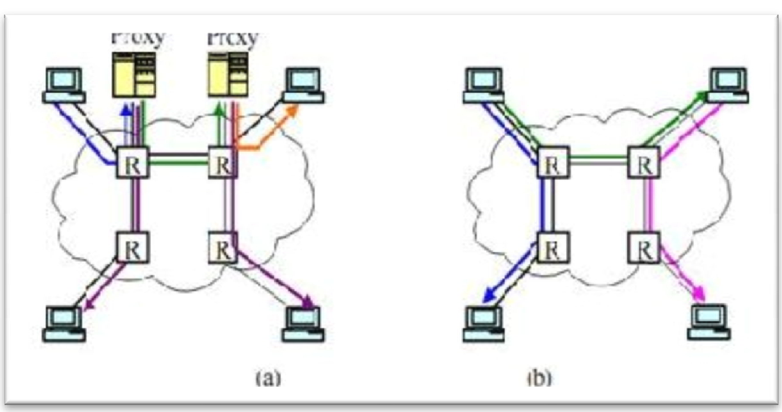
II. Application layer multicast for p2p network

III. Application layer multicast for point multicast.

3.5.2 Deploying Application layer multicast protocol

As said earlier, application layer protocols are deployed either at end users or at proxy level [32]. In case of proxy-based ALM, the advantage is taken of the existing IP multicast "mass" and by including an envoy as an overlay node, whereas the proxies themselves are organized into an overlay network and providing maximum security and efficiency. While on the other hand, ALM end users protocols provide forwarding of data as well as multicast utility. End user protocols are rather more flexible, adaptable, provide ease in deployment and handled easily when compared to proxy-based protocol but the end-user protocol can well maintain increasing number of users when it comes down to provide fruitful services.

Fig (a) Proxy based ALM (b) End user ALM



3.6 Topologies in Application layer multicast protocol

In application layer multicast, two topological orders are deployed that are used to organize group components. First is data topology and the second is control topology. Data topology provides convenience for the packets to follow a certain data path. While the control topology time to time swaps refreshment messages to identify and follow the non-ethical leave of members from a certain group. Now, it depends upon the construction sequence and data path topology, and follows the regular topology under following approaches [34].

- Tree first topology
- Mesh topology
- Lopsided tree topology
- Implicit approach
- P2P overlay

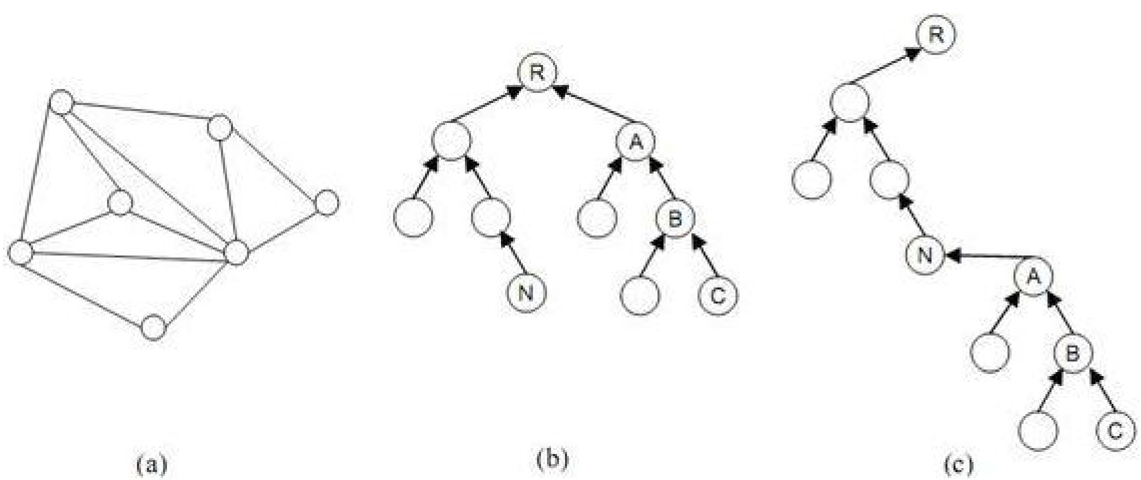


Fig 3.6 Various topologies (a) Mesh topology (b) Initial Tree (c) Lopsided tree [34]

3.6.1Mesh first approach

In this approach, the existing group members transform themselves into overlay mesh topology, so that each member is linked with multiple paths that can be accessed by other nodes. Then it follows an explicit tree which is formed at any member node via a well known multicast process e.g. DVMRP, NARADA and SCATTERCAST.

DVMRP

Distance vector multicast routing protocol [33] [34] [35], is a more of a multicast addition to unicast concepts that are used in Routing information Protocol (RIP). Routing information is started by the receiver, a source based routing protocol we can call this. Therefore, a spanning tree is created which performs best with respect to delay, at every node. Using Reverse path multicasting (RPM) each multicast data unit is routed. Techniques like graft-data units and poison-reverse are used to avail control mechanism in such conditions. The figures below represent group of routers that named from A to E. Every router is linked with a cost to get connected to another router. The number with the cost here shows some of the constraints like

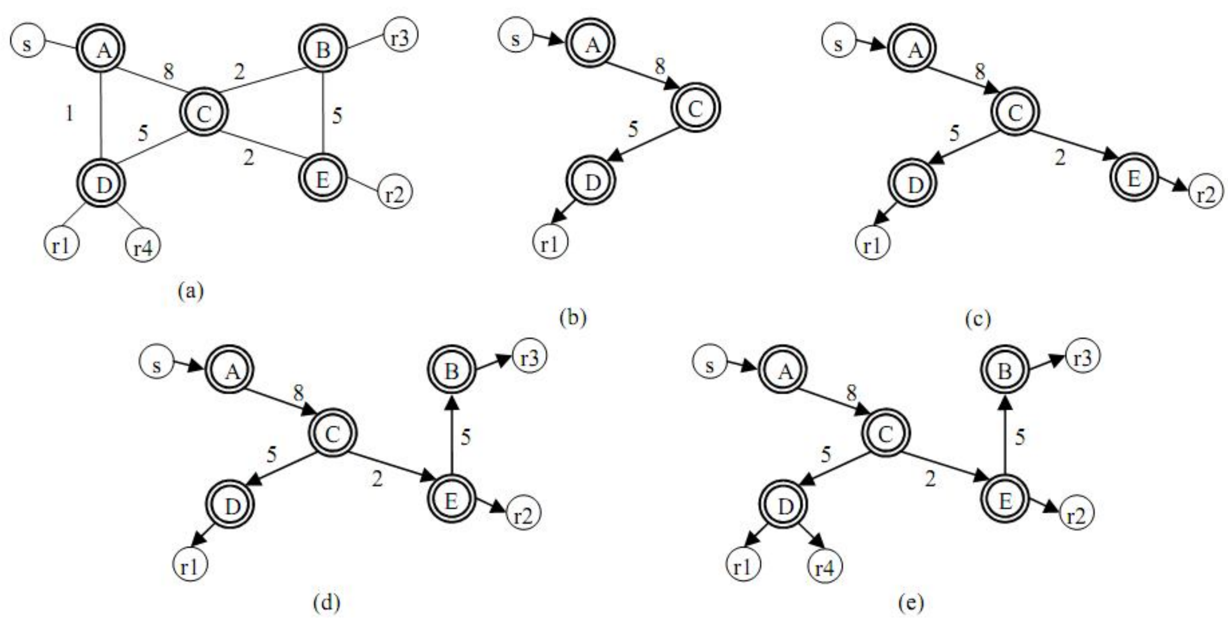


Fig 3.6.1 showing DVMRP (a) Multicast (b) r1 joining (b) r2 joining (c) r3 joining (d) r4 joining

end-to-end delay, maximum bandwidth etc .For instance, in the above figure (a) the link from D to C is more costly then the link from C to B. Some of the routers are attached with constants like r1, r2, r3, r4. In reality, these constants can be home users as well, sharing a common video or an online game. For instance, there are more connectivity points from A to D: A→D or A→C→D. The cost of the first path is greater (18) than that of the second path (13). (b) Shows the shortest path to r1 is from source S to the connected router R. When the rest of the receivers' r2, r3 and r4 join the network, the tree then further extends to (c), (d) and (e) respectively. In (e) it is quite clearly shown the composition of a distribution tree in a multicast group. It is to be noted here that this tree formed is for one source S, when dealing with the other sources, it becomes more complex.

3.6.2 Tree first topology

Unlike mesh approach, in tree first topology a shared resource data tree is constructed directly. To choose a parent node is an important both from efficiency and performance point of view. To choose an organized parent node is very important for the members of the group because then it becomes easy for the group members to know about the other group members under the same parent node and know about other neighboring members under the same overlay network and establishing links with them. YOID, banana tree, host multicast tree and ALMI are a few examples under this topology.

YOID

Like the above briefly tacit defined protocol DVMRP, YOID [35] [33] is also the first protocol to be deployed in tree topology. YOID directly creates a data delivery tree that has control over the aspects of different tree attachments.

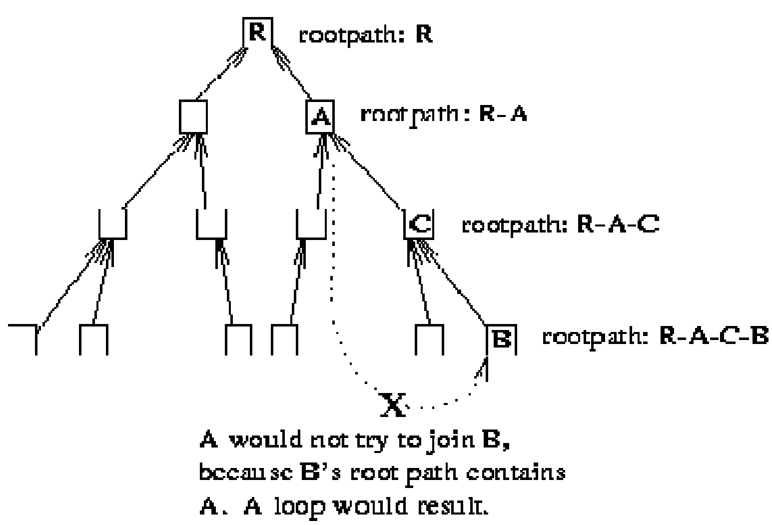


Fig 3.6.2 YOID in tree first topology

As it is clear from the above structure it is quite clear that unlike mesh, the quality of data is independent of mesh. It also obvious from the figure that YOID has a shared tree and all the members in the group are associated with their own parent node, also each parent has to choose the number of children it can support. Whenever a member wants to join a certain group, it looks for a parent and attaches itself to a certain meeting point. The main design of this protocol is to provide vigorousness and providing a simple tree structure. The structure however may cause loops and bad structuring can lead to poor performance. Certain mechanism had been proposed to avoid this flaw. To use SWITCHSTAMP, and certain algorithms have been proposed such as latency refined algorithm and loss rate refined algorithm.

HMPT

Host multicast protocol tree is another kind of protocol applied in tree topology and uses tree first approach [33] but in performance aspect, it is found to be much better than YOID. A tree in this case is built in a way that highest level at each node is constant and cost to connect every link is minimum. Round trip time for every member in of this group is added as an optimization matrix. The tree of HMPT is more or less same as YOID. All the members in the group are responsible for finding the related parent node, and then attach it via finding the common attach point to the tree.

From the extracted value of Round trip time (RTT), the new comer node finds itself to the closet node among the possible parents and their corresponding children. If the closed node itself is parent node then the new comer would be attached at node R. new then would be accepted as the child to the parent node. The whole process is repeated again for the same new comer node which wants to join the tree by knowing the child of the tree and the possible parents.

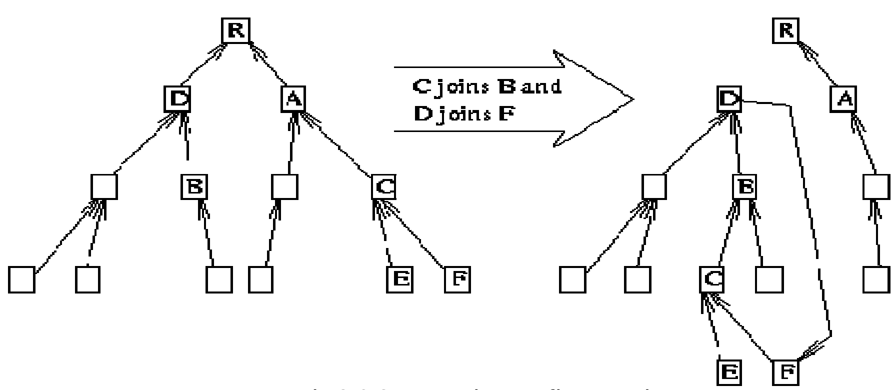


Fig 3.6.2 HMPT in tree first topology

3.6.3 Implicit approach

In this approach, all the protocols make up a central topology with specific properties. There are a couple of protocols of this centralized topology named as KUDOS and NICE. KUDOS is within two layered hierarchy and NICE is within three.

NICE

When a new member wants to join or leave a group, this protocol maintains information hierarchy amongst the group members [33]. It is useful for real time applications in large group. It is more of a hierarchy controlled protocol. All the group members are assigned to the lowest level of layer and there they group to form clusters. Cluster size usually varies from K to 3K-1 where K is a constant. There is a leader of each cluster, and the new member joins the cluster on the choice made by the leader.

However the for a new member to join this cluster, the process is very unique and complex too. First the RP is consulted, from which the information of the highest layer member is obtained $\mathbf{L_n}$ and then the closest to this one is obtained. The leader then notifies that the new comer is on the layer $\mathbf{L_{n-1}}$. This process is carried on until the layer $\mathbf{L_0}$ is reached with a proper position.

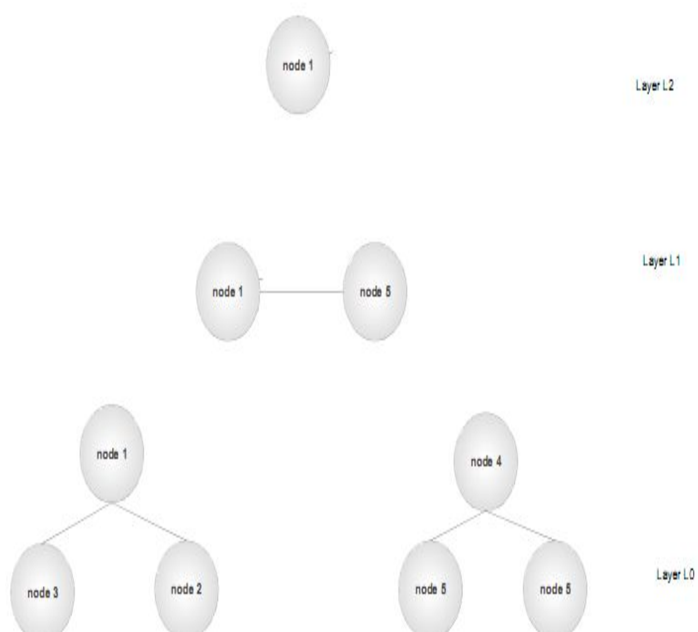


Fig 3.6.3 NICE in Implicit approach

3.6.4 P2P Overlay

Construction of an overlay network upon an already existing IP network has allowed p2p overlay to gain popularity in the recent times as this scheme has the latency to carry large scale functions successfully. There are two sorts of p2p [33] overlay architecture. One is the structured and the other is the unstructured. In structured overlay connected nodes are managed in a certain order while in unstructured overlay the connected nodes haphazardly. One of the main problems with structured overlay is that agitation is high. This is the only reason why structured overlay is used for application layer multicast. Common examples are KOORDE, TAPESTRY, CAN.

3.7 Constraints based application layer multicast

A certain set of constraints come into action when practically implementing application layer multicast (ALM). However, we can use these constraints to further explore our research and enhance the quality of IPTV stream.

- Extra transparency is caused at the end user while using application layer multicast, as it is implemented as the end user, so this can cause a compromise in IPTV quality.
- ALM is not quite effective when it uses network resources such as when end user is using
- ALM, the data packets are get imitated by the end user, so the same set of packets are send over the same link again and again. This causes congestion on the network.
- As in an overlay network, overlay nodes are pre-ordered haphazardly without knowing which layer is beneath; this is a flaw in application layer as compared to physical layer which is lying in top of application multicast.
- Large trees can built up when same packets are send over the same link again and again. This would cause using more bandwidth.

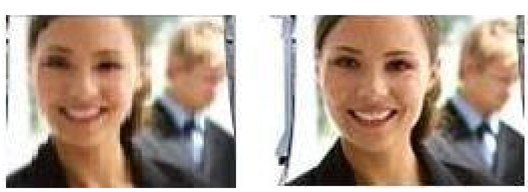
Chapter 4

Quality metrics & encoding techniques

In this section, we shall discuss the newest technologies in use for video compression along with basic principles. This portion of discussion here will provide us with facts on how packet loss can influence the features of IPTV stream. To discuss particular video format in detail here will be of no use but a summary of common IPTV video formats is discussed. A video compression format called as Codec is derived from *co*mpression/*dec*ompression. Video footage is either stored or scattered just when the video encoders compress the size of a video signal. This is done by using resources with inadequate storage output capacity, e.g. DVD or a broadband internet connection. To compress a video, 3 basic principles are used to make compression.

4.1 Fidelity

Is defined as a process in which how accurately a compressed image is able to reproduce or generate the original image [36] [37]. There are however couple of options opted by the video encoder; for instance by reducing the color space or decreasing image resolution etc. By reducing the image resolution some details might be lost, but the resulting image after image resolution can be much smaller in size and distorted. In case of reduced color spacing, similar colors might be replaced by color/colors that approximately match the original color. Fidelity in this case can be low, if maximum colors are replaced by a single color. An example here could be replacement of colors with black and white. The figure below shows the example of fidelity in case for image resolution and color spacing.



(a) (b)

(c) Color comparison

Fig. 4.1 Fidelity Compression (a) Original, (b) resolution reduced (c) reduced color space

4.2 Spatiality

Shows a relationship between different parts of an image [36] [37], an image, when split up into different small blocks, it is more likely to observe that the neighboring blocks may share the same color, because they are the part of the same object in the image. For instance, if the image is of a big blue box and it is divided into 500 blocks, it is likely to observe that multiple blocks contain same information. A video compressor tires to remove this unnecessary information and save storage space.

4.3 Temporality

It is described as a relation between video frames and subsequent video image. Subsequent video frames tend to contain the same information. Subsequent frames show the same object or parts belonging to the same object because of the changing location of the object. Most of the times, neighboring frames have high similarities so compression can be reached only by storing the differences between two subsequent frames. The resulting compressed image is much smaller in size as compared to the original image without encoding similarities in it. The below two figures show temporal relationship between two subsequent frames. The left hand and the mouth are the two differences in the frames; rest is the same, so it is encoded.

Fig 4.3 Temporal relationship between two frames

There are different types of encoders for different frames; predictive frames and reference frames. Reference frames are those frames that have no temporal relation with existing frames. On the other hand, predictive frames do have temporal relation with the other existing frames. Frames that belong to a reference frame are referred to as Group of pictures (GOP) and the interval amongst the both is knows to be GOP length.

4.4 Standards for video compression

Setting standards for video formats is an important aspect from development point of view. One, a video standard can be used on a large scale, if it had been developed by any manufacturer for interoperable solutions. Two, the cost can be reduced on extending the already developed version by certain manufacturer, as experts would emphasis more on the development of the standard. For instance, a quite doing well standard is the MPEG-2 standard for DVD’s.

There are two very well known organizations working on video and audio coding standards;

- *MPEG* moving picture expert group. MPEG is working under ISO/IEC and in charge of audio video coding standards. Standards for products such as CD, MP3, set top box and DVD’s are developed by MPEG [26].
- *VCEG* video coding expert group works under International Telecommunication union Standardization sector (ITU-T) focusing on development of new video coding standards [38].

The three key video formats shall be discussed in the next topic, which are most common in use and known: MPEG-2, MPEG-4 and H.264 [28] [26].

4.4.1 MPEG-2

MPEG-2 was the starter and was developed in the early nineties by moving picture expert group [28] [26]. This video standard is widely used as it is used in DVD and most of the IPTV format channels, cable TV and satellite TV. MPEG-2 uses two types of predictive frame scenarios: P-frames which only store information with prior reference frame while B-frames awaits on both preceding and following frame. A high compression rate can be attained in this technique.

4.4.2 MPEG-4

Is a group of standards for audio/video coding, used for storing digital multimedia [28] [26]. The goal of designing MPEG-4 coding technique was to compose a standard that could be used to provide support to low-bit rate applications as the purpose was MPEG-2 was to provide support for high-bit rate applications. MPEG-4 standard was more of a combined standard that had properties of high and low bit rate applications. More over the standard was comprised of different sub-level standards for audio/video coding. It is more of a structure that consists of affluent multimedia applications. Furthermore, MPEG-4 is extended into two more encoding standards that are currently being used in many applications, these are:

- MPEG-4 part2
- MPEG-4 part 10

When we use the term MPEG-4, it is considered to be MPEG-4 part 2, by default.

Now-a-days, a very advance codec is in frequent use, which provides the edge to comprise features like error resiliency and scalability called MPEG-2 MPEG-4 Advanced simple profile. MPEG-4 ASP provides with more refined results as compared to MPEG-2 in aspect of video compression. Some of the very common MPEG-4 parts 2 are DivX, Xvid and Apple Quick time codec.

4.4.3 H.264

H.264 is the most of the recent advance video coding standard available. The term H.264 has this name to describe the format. This format is used for high definition videos, IPTV contents and contents displayed over mobile multimedia applications. It is expected that H.264 will replace MPEG-2 standard in IPTV content delivery. The reason is that H.264 provides high compression proportion for less deposit cost, which will definitely lead to a decrease of bandwidth in a network that provides IPTV streaming.

4.5 Packet loss: Origin and outcomes

In this section, we shall discuss how packet loss takes place in IPTV. What actually, in general causes packet loss in IPTV, the origin and effects of packet loss on an IPTV stream? How quality of IPTV is disturbed when packet loss occurs. A set of techniques is followed to emphasis more on packet loss and how are these techniques used to reduce packet loss.

Packet loss can take place due to many reasons such as:

- Congestion in network
- defective tools
- weakening of signals over the network
- wrong routing paths

Weakening of signals and jamming in a network are two reasons that can effect IPTV applications. Weakening of electrical signals is caused due to a poor quality line, which can be in either DSL or coaxial cable for an access network. A weak signal will either lead to misinterpretation of a signal or no signal reception at all. So a packet loss will occur, either no packet would be received or wrong values stored in the packet header.

Congestion in a network happens when the output through a network link exceeds the maximum limit of the link or when the output exceeds the routing rate of a router, in a link. When the device, for instance the router is not able to process any more incoming packets, the packets will not be buffed anymore and the result will be packet loss. All this can be avoided by enhancing the network and the device procession speed along with providing priority to the packets that more important in the network and designing a QoS mechanism to match the network speed in accordance with the incoming packet.

4.5.1 Packet loss impact on IPTV content

Set top box decodes the incoming coded streams but it will not be possible enough after packet losing is commenced in an incoming IPTV stream. This will cause an error in the contents viewed. As an example, two images are compared and the result of packet loss is quite obvious in it. The image on the right is showing clear signs of missing packets, resulting in strong destruction of the decoded image: clearly head and body of the person in the image is disrupted. The image on the left however, a loss in data is causing the dislocation of the image, as in the image the body part with the tie is relocated on the right side of the person's body.

This error transmission may continue to flow for multiple seconds. This issue of error can have a big impact on the video stream and Quality of Experience as seen experienced by the IPTV users.

4.5.2 QoE/QoS metrics for IPTV

In this section we shall discuss how certain metrics act as beginners for providing much better quality of IPTV service, when it comes down to experience IPTV streams. If a user is using services of IPTV, but the services are not according to user's standards, the result will be that user will stop using that provided service. The minimum requirements that are required by a specific service are referred to as QoS. QoS can be defined as network level services agreed by the network metrics like packet loss, jitter delay and latency.

Users of IPTV are in habit of using services that are provided by the "Traditional" analogue cable broadcast TV. So if they suffer small delay in services received by the new IPTV technology, it will not be a very pleasant experience. The metrics mentioned above however doesn't provide a glimpse of what is going on at users end. The user experience now-a-days is becoming more and more vital, especially when it comes down to multimedia applications and with restriction of timing and quality for the subjective users.

Following group can be looked upon, when looking for metrics of IPTV regarding QoE and QoS:

- Metrics that associate focus on quality of network by relating the transport of IPTV contents and their delivery.

- Metrics associated to audio/video features of IPTV contents.

The coming section onwards will provide discussion on quality of network metric and audio/video metric along with types of measures that can be adopted for quality.

4.6 Quality of Network Metrics

The metrics of network are described as follow: packet loss, bandwidth, latency, jitter, delays. These metrics along with their descriptions are explained in the table below. The metrics here generally are used to make a synthesis of suitable applications for transport of data. For instance, real time application like VoIP, delay and jitter is very important, as when the two factors are on an increase, the application cannot continue. For services like IPTV, metric like bandwidth is very important, as if the bandwidth required to carry out the application is low, the process will not continue.

Latency	Time taken by the packet to go from sender to receiver. Two types of latency measures. Single trip delay and round trip time delay. Single trip specifies delay from sender to receiver while round trip time delay specifies from sender to receiver and receiver to sender.
Bandwidth	the amount of data that can be send over a network in a specific period of time, measured in bits per second
Loss	the amount of packet loss in a network from sender to receiver. It is measured in percentage as the total number of packets loss in the entire duration.
Jitter	the inconsistency over time of latency through a network.

For a high throughput in IPTV, a strict need of jitter, loss and delay is required. Exact requirements for services of video are not available yet, but there are several guidelines available for from ITU-T and DSL forum IPTV bodies. An example given below by ITU-T can be used to achieve the service quality required, in case for packet loss. These metrics are derived from IETF work group for IP.

Packet Loss Rate	QoS
$PLR \leq 10^{-5}$	excellent service quality (ESQ)
$10^{-5} < PLR \leq 2 * 10^{-4}$	intermediate service quality (ISQ)
$2 * 10^{-4} < PLR < PLR\_out = 0.01$	poor service quality (PSQ)
$PRL\_out = 0.01 < PLR$	IP end-to-end service not available

Fig. 4.6 Classification for digital services.

4.7 Quality metrics Video

Packet loss in IPTV can lead to many errors in the video but there is no such way to foresee the type of problem that will be encountered. In most of the cases we come across visual impairment, with change of color etc. The result of packet loss in IPTV can be due to multiple factors, they can be:

- Codecs;
- Settings of Codec with group of picture (GOP) size. The purpose of GOP is to determine those frames which have a temporal relation with the frames in GOP. The bigger the GOP, higher will be the flow of visual impairment in the stream.
- To which category frame lost belongs to. For MPEG-2, the type of frame lost can be either from B-frame or P-frame or I-frame.
- Ratio of compression
- Concealment of error role for video codec may cause users not to notice visual impairment due to packet loss.

Just like network metrics, video standardization metrics also exist, for reasons such as explicit metrics by video standardized and those based on measurement metrics are replaced by interoperable products. Examples of such standardized metrics are IPTV QoS and QoE created by IETF and creation of an RTP extension for using extended RTCP reports.

4.8 QoE and QoS metrics IPTV

The following metric are introduced by IETF [40] [25]:

- *Video stream metrics* provides an insight of different various types of video frames that are affected by the packet loss
- *Video stream description metrics* provide information on the type of codec in use, with GOP structure and length, size of the image.
- *Perceptual quality metrics* provides QoE figures for high level audio and video

- *MPEG metrics* provides information on MPEG-TS used for transport of contents of video along with associated metrics.
- *Transport metric* provides detailed information on jitter, loss, delay. Also information on how effective error mechanisms are like FEC and retransmission.

[1] RTCP reports extended version

The Extended Report (XR) packet type was defined for RTCP in the beginning of 2003 [41[1]]. This extended report can be used to provide more information than expected, in the reception block of RTCP receiver and sender.

4.9 Measurement techniques

There are 3 different methods to measure quality of video in IPTV service, they are:

1. **Objective measurements**: Comparison of signal being transmitted over the network with the original source signal
2. **Subjective measurements:** includes controlled video experiments in which the individuals involved rate the video quality by already defined scale.
3. **Indirect measurement:** is done by make use of network measurements. The quality of video is measured by using the effect of video impairments.

4.9.1 Objective measurements

Is a comparison if source video with output video. The results are then used to make assumptions in original stream is different from transported stream. The more then difference, lower the quality of received signal.

There are however three types of reference based measurements:

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[1] *RTP control extension protocol reports (RTCP XR) is beyond the scope of our research, however the details are available under "Internet Engineering Task Force" RFC 3611. but, it was studied to get the idea of feedback in RTCP.*

- *Full reference-* in full reference, the source signal is fully compared with received and the decoded signal. Both the streams are needed to be present at the same moment, which is not possible in case of IPTV, as in IPTV only one stream is received by the client.

- *Reduced reference*- involves measurements where a few measurements are needed from the transmitted signal and received signal for comparison.

- *Zero reference*- measurements from the transmitted video signal is not available for comparison, only received video signal can be used for measurement. This reference can be used for IPTV stream, as it is the received stream.

Peak signal to Noise ratio (PSNR)

An object reference technique which is used very often by computing peak signal to noise ratio is PSNR. It is done by taking the root mean squared value of (RMS) of differences of original to received video frames in dB [43]. This measurement technique is used to compute packet loss. However the measurement technique doesn't tells us about frame drop rate, delays in the playback or freezes in the playback due to buffer under runs. The disadvantage of PSNR is that it is a process highly depending upon calculations and so the calculations must be accurate, means sensitive to calculations.

4.9.2 Subjective measurements

It is uses humans viewers of IPTV to measure the quality of the content. It can be used to perfect evaluation of video quality, as it involves experiences from end users. Mostly used subjective measurement is mean opinion score.

Mean score

Is a subjective measurement signal, which ranks video quality services, based on user feedbacks? In MOS, the quality of the video displayed, ranges from 1 (very bad) to 5 (very good) and then the average is taken. As a comparison with PSNR, in MOS time is taken by the process, as all the errors in the video will be notice by the subjects testing [43].

MOS	Perceived Quality	Impairment	PSNR (dB)
5	Excellent	Imperceptible	> 37
4	Good	Perceptible, but not annoying	31-37
3	Fair	Slightly annoying	25-30
2	Poor	Annoying	20-24
1	Bad	Very annoying	<20

Table [44] MOS converting to PSNR.

4.9.3 Indirect measurements

The final category of IPTV service measurements is the indirect measurements. Network parameters are impaired in this scenario to forecast the video quality and quality of services for IPTV. A technique called EPSNR is used, which estimates PSNR [45] to determine video quality techniques. This technique uses relation between packet loss and portion of pixels that get impaired due to packet loss. This technique seems to deliver results, as it can be applied in practice, even in case of broadcasting.

Summary

In this chapter, a possible introduction is provided to IPTV and the services associated to it. A list of protocols can be used to deliver IPTV stream, but the protocol selection depends upon a number of factors:

- Delay in IPTV
- Applications distributed in the entire structure
- And the configuration of the entire network

A couple of very simple protocols are used in the transmission of the IPTV stream, which are the most simplest and yet result oriented. User datagram protocol UDP and Real time transport protocol RTP. UDP doesn't provide any mechanism on feedback, while RTP has the edge of providing feedback. It runs on the top of other transport protocols. However, in multicast scenario, the mechanism of RTP reports is not suitable.

In the later sections, we discussed video compression techniques and examples of techniques used by IPTV services. The answers to" *what effects can packet loss have on IPTV streams and* *applications?"* can be that packet loss in IPTV streaming applications can lead to visual impairments of the video stream decoded. Errors that flow in the reference video frames can make their way into the subsequent video frames, in the temporal relationship between sequenced frames. A single packet loss can result into a disturbed image of multiple frames which will also disturb Quality of Experience for several seconds. The techniques used above will be used in chapter 3 to provide an overview of error resiliency techniques. The impact of packet loss can be measured on the video quality by several ways:

- Subjective measuring by human experience
- Objective measuring by comparing video signal over original signal
- Indirect measuring by impairment of network

Chapter 5

Error resiliency techniques with focus on retransmission

After looking at the structure of IPTV transmission and providing an overview of how quality of video is affected by packet loss, we shall now create a scenario in which error correcting techniques are possibly applicable, i.e. how FEC and retransmission can be applied to and IPTV stream. Before going into details and providing a scenario where IPTV can make use of error correcting techniques, let us provide a brief introduction to some error correcting techniques. As it is quite obvious in the last chapter that either prevent packet loss or provide means to restore the lost packets to make the errors least notable. In other words, we must come up with an error correcting mechanism that reduce packet loss and provide quick recovery.

There are however, multiple ways to provide resiliency against packet loss. Two factors can be kept under focus here: One is to provide means of recovery for packet loss and the second is to conceal/reduce packet loss. The two most common error recovery techniques are forward error correction and retransmission. While other well practiced techniques for packet loss effect reduction are packet interleaving, error concealment, and application payload and bandwidth adaptation.

5.1 Forward error correction (FEC)

Forwards error correction (FEC) is the process of adding surplus/redundant data along with the data being transmitted. This EXTRA data can be used by users to reconstruct the data that was lost. However, the amount of reconstructed data entirely depends upon the amount of data loss and the surplus data as well. FEC only suits environment which has issues concerned with latency. Usually networks with uni-directional transmission make use of FEC, as in case of satellite transmission and cellular networks. In these cases, no mechanism of feedback exists from sender to receiver. Latency is high throughout the system of networks with single direction transmission.

FEC can be applied on the different layers of OSI model[2], from physical layer to application layer. For IPTV applications, FEC can be implemented either on application, transport or network layer. The most implemented FEC scheme is Raptor FEC encoding [46]. An advantage of FEC from network operator's point of view is that the bandwidth overhead for redundant FEC can be calculated prior to bandwidth reservation in IPTV stream delivery. Additional data is inserted before and it can be used for data recovery. However delay at this [2]stage can be introduced by the addition of surplus but this duration of delay is less then that caused by the retransmission mechanism.

Fig 5.1 Propagation of error in sub-sequent frames

5.2 Adaptive forward error correction

Adaptive forward error correction technique is unlike the conventional FEC. In this technique, redundant data is transmitted with the data that may contain loss attribute reported by the receiver but it is efficient in a way that the redundant data is only transmitted when needed [47]. This leads to the fact that adaptive forward correction is only applicable to distributed networks that posse's property of providing a feedback from sender to receiver.

[2] OSI, Open System Interconnection reference Model is a structure to design protocols. Defined by ISO for standardization and consists of 7 abstract layers in a top down hierarchy as Application, presentation, session, transport, network, data link and physical layer.

5.3 Packet retransmission

Packet retransmission is the method of transmitting packets that are considered to be lost. In retransmission process, communication between sender and receiver is obligatory, for a reason that the receiver has to ask/notify the sender about packet retransmission e.g. TCP [7]. Unlike FEC, in packet retransmission, bi-directional communication between sender and receiver happens, to indicate packet loss and to retransmit particular packets that are lost. Two types of message are used to indicate packet loss:

- *NACK* (negative acknowledgement) is used to verify that either one or more packets were not received. NACK is used in networks where feedback from receiver to sender is kept low, to avoid network congestion.
- *ACK,* acknowledgement message is used to indicate packet loss by acknowledging the reception of one packet or the queue of packets.

Packet retransmission is different in erratic networks; no packet retransmission takes place when there is no packet loss. It means that the bandwidth is needed only when packet loss occurs. As the packet is retransmitted, it will cause delay, which leads to a restriction of introducing packet retransmission methods only to those networks that are delay free.

The drawback of using packet retransmission in IPTV can be that IPTV services are not made to perform under jamming. When congestion takes place during the transport of IPTV stream and so is packet retransmission enabled, the introduction of packet retransmission leads to network congestion and the result will be bandwidth decrease for IPTV streams.

5.4 Error concealment

Error concealment functionality can be used in audio/video formats. When missing data is detected by the decoder, it camouflages the data by using certain algorithms. An example here can be that to conceal missing data, video data from neighboring region of an image are filled with it, or parts from the previous of upcoming frames can be used. Error concealment is practiced by both MPEG-2 and H.264.

5.5 Payload interleaving

One of the techniques used to minimize packet loss effect. There are certain loss tolerant applications which are linear to packet loss and even if packet loss occurs during the execution of such applications, it can be ignored. Such applications are telephony and video streaming, as in these applications, data is valuable for very short amount of time and packet loss is not an issue. For application, such as file transfer and online banking and services, this can be a serious matter. One method can be to interleave data, to avoid packet loss. By interleaving the transmitted data, packet loss will result only a small portion of the successive data. This will minimize the effect of packet loss on the whole throughput as well as enhancing probability of error concealment.

In temporal relation of neighboring video frames, the error would then disappear over multiple frames. This will allow small errors often not noticed easily by the user, as compared to key errors that can be easily viewed and percept by the user. Fig can provide a clear view of how errors are washed away in multiple frames by using payload interleaving. One another advantage of payload interleaving is that small errors can be concealed easily as compared to concealing one major error.

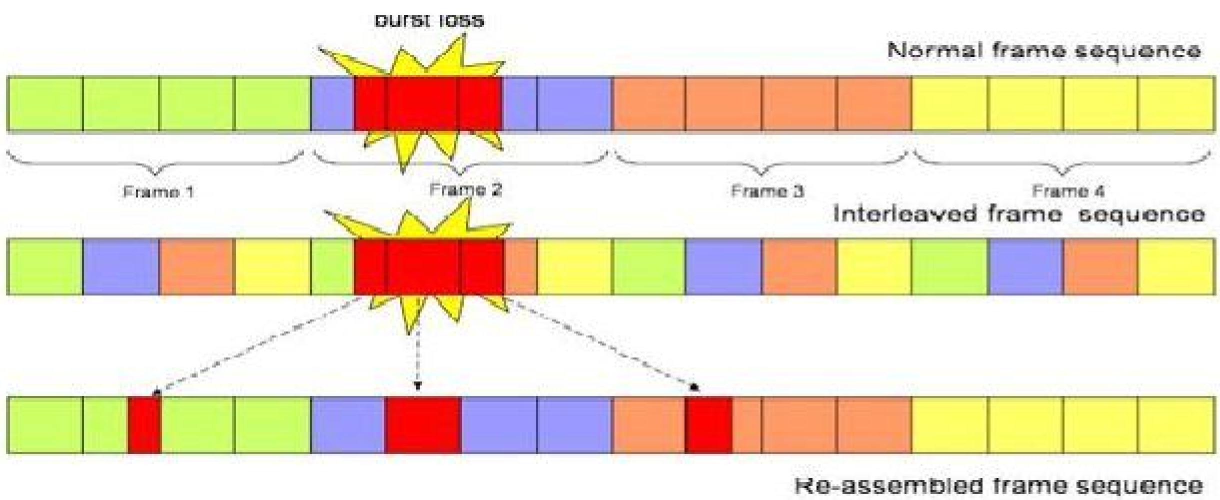


Fig 5.5 Payload interleaving: due to interleaving burst lost is scattered to multi frames. Loss is spread in frames.

5.6 Bandwidth adaptation

The form of error resiliency technique in which adaptation to network conditions is met by decreasing bandwidth used by IPTV streams or changing encoding report, making robust transmission or reducing congestion by controlling the network. By reducing all these parameters, a tradeoff can take place between audio and video quality but end-to-end delivery is confirmed. However, for a multicast network, this approach is not favored as the number of receivers influenced by distortion is more.

Before going in detail of the process to describe how FEC and retransmission can be implemented, let us look into the matter of how, by emphasizing on certain factors, we can enhance the quality of IPTV and provide an overview of certain self created scenarios, which we think are good enough to get the maximum throughput and error resiliency.

The upcoming sections would provide us to investigate how FEC works in multicast IPTV services and make certain assumptions on how can a prototype be developed for providing better IPTV services in a multicast environment.

5.7 Scenario

As it is clear from the above discussion that linear IPTV services are provided to its customers via a multicast distributed network. All the services attached are provided by IPTV service provider, not over any ordinary internet connection. All the IPTV traffic is maintained in the core network, as clear from the figure and is forwarded to the specific viewers when the viewers ask for the application to be delivered. As the channel is selected by the user, the set top box at the very moment becomes a part of the multicast network. Set top box decodes all the information and starts displaying.

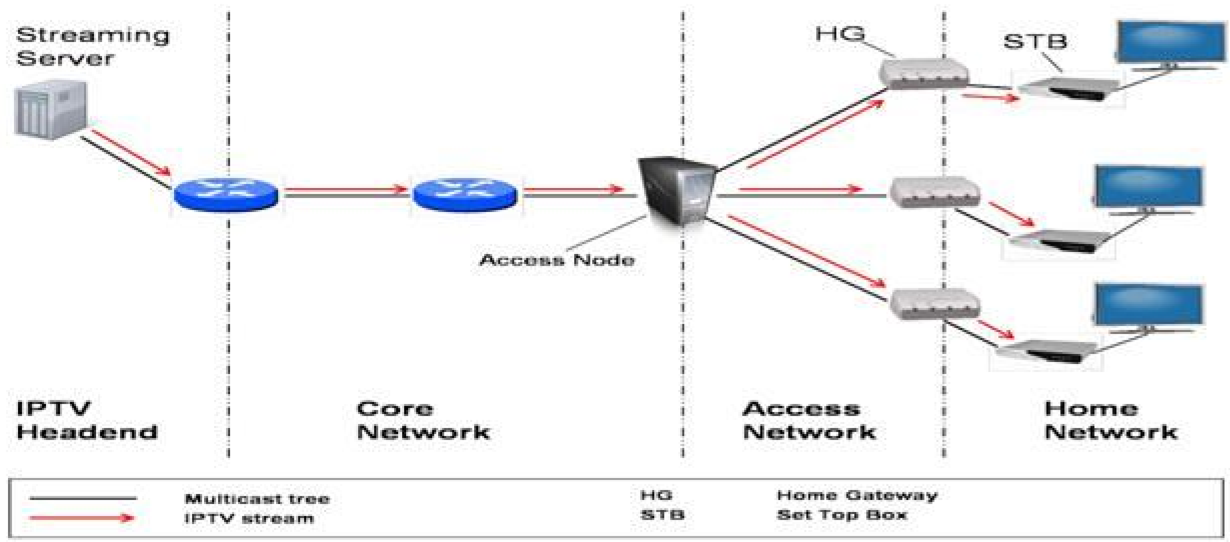


Fig 5.7 Network distribution

IPTV can be provided via different access networks e.g. Coaxial cable, DSL, and other networks offered by certain IPTV providing companies. The quality of IPTV stream delivered depends upon its delivery from subscriber to the core network and can suffer packet loss on its way. Packet loss can have harsh effect on picture quality and it can cause the user to stop using the service or look for another alternative in the market, may be a better one.To overcome packet loss, packet retransmission is introduced in the access network, to maintain standard of IPTV. One approach to provide packet recovery is provisionally caching packets that are to be forwarded to the users in an access node e.g. DSLAM or MSAG. This will provide resiliency in a certain position against packet loss between users and the network. When the set top box during a certain stream comes across packet loss, it will suddenly ask for packet retransmission, which will be offered quickly by the access node in a deliverable time. This will facilitate the set top box by placing the retransmitted packets in its reception buffer, before the data has to be sent. Our example can be shown here, with the access node in the middle of the network providing caching of packets.

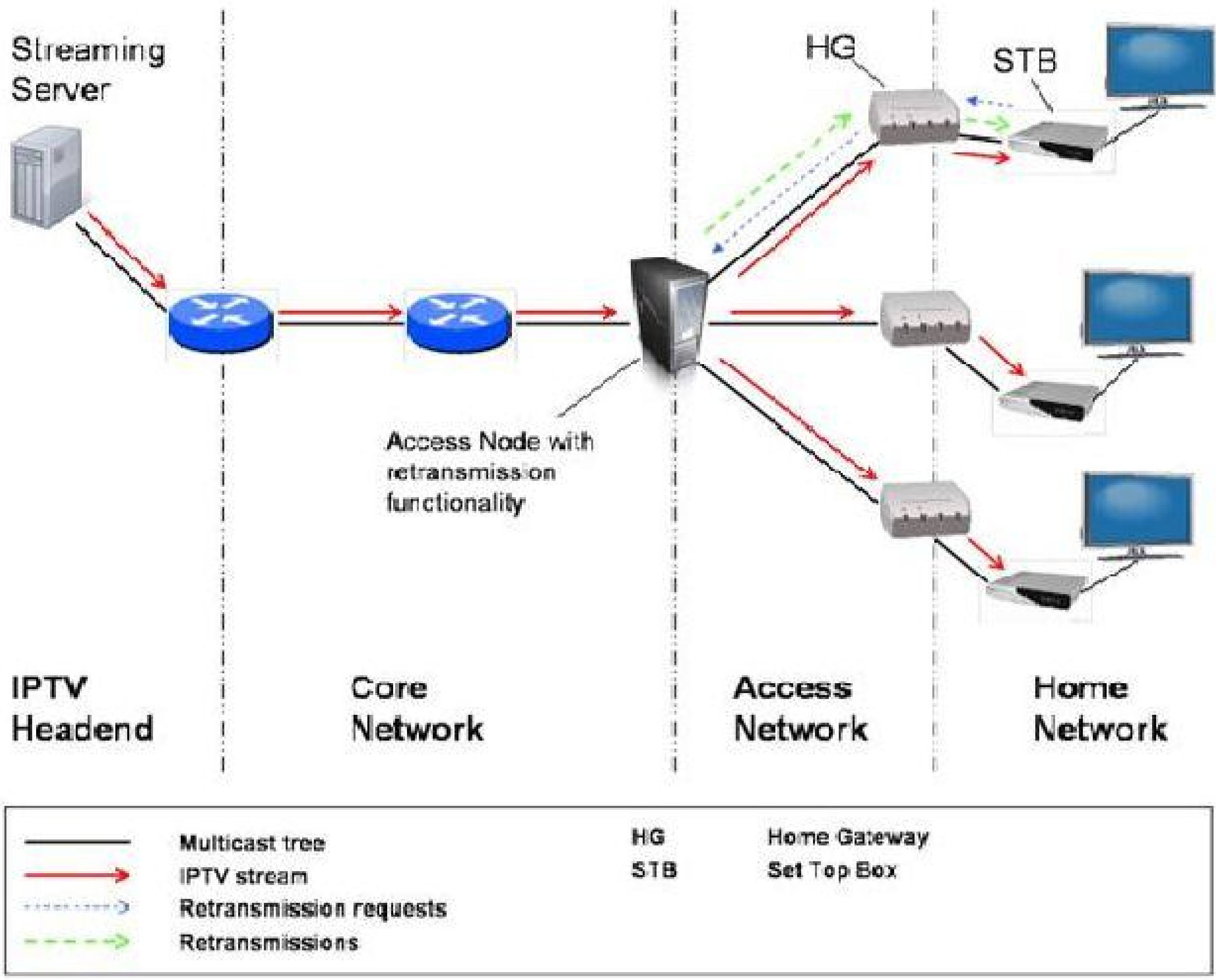


Fig 5.7 Providing cache in the middle of the access node to provide retransmission mechanism

The above scenario is for live IPTV transmission which means that there has to be strict time duration to recover from packet loss and amount of delay to be reduced.

5.7.1 Procedural description

The retransmission function cannot be implemented on the whole multicast network. In fact we can measure its results if packet retransmission is implemented on a single subtree of the entire stream. The figure shows end to end delivery of IPTV stream from a server along with the entire network till the set top box [48]. The intermediate aspects provide functionality for the guaranteed delivery of the IPTV stream. Transport protocol hasn't been defined in the above figure, as there is more than single possibility of protocols used on this layer, which is already discussed in chapter 2. IP in this stack provides forwarding and routing of packets.

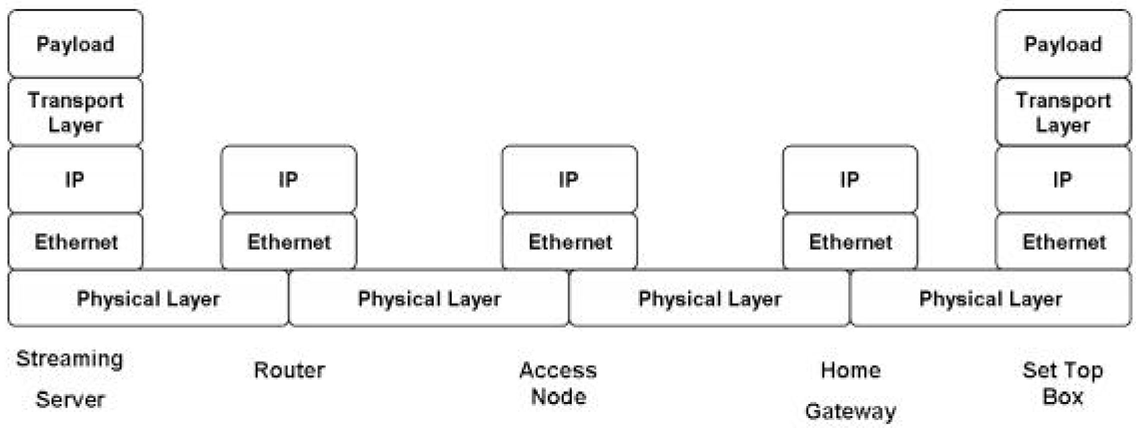


Fig 5.7.1 IPTV embedded protocol management

5.7.2 Hypothesis

A set of suppositions can be made here, for providing a model that can be used as a prototype.

- From access network, all the links must be providing good enough streams to users; otherwise if the services fail from this point, a guaranteed delivery on the IPTV stream is Not possible.
- The packet loss mostly happens between the temporary cache and the IPTV user. The hypothesis here is not providing any idea for packet loss between the server (streamer) and the temporary cache (DSLAM or MSAG).

- The above IPTV multicast network is assumed. The figure above has multicast routers beside the temporary cache server, set top boxes and streaming server providing main streaming. These routers are capable forwarding the packets to set top box via a multicast path.

5.7.3 Necessities

After the stating the above assumptions, we can now come up with a few steps of How can we provide our solution to our own created dilemma, surely that would decrease the packet loss in the whole IPTV scenario but to verify everything, we would come up with a couple of assumptions that can really satisfy our work and answer what we have just questioned above.

- The system we make up shall produce multicasting in IPTV streaming and it will provide resiliency to error as well on the basis of packet retransmission. The packet retransmission would be offered by a retransmission cache, which will be located in the access node, as shown in the figure.

- A stream server will send IPTV stream and the information would be decoded by the set top box located at the end user's side.

- Error resiliency shall be provided by the cache to all the packet loss, which is placed in the access node.

- Retransmission shall be done sequentially

So, the above mentioned system requires 3 particular components that are:

1. a client receiving services
2. a retransmission cache in access node and
3. a server providing constant streaming

In this section we will compose and somehow break the above described model into portions, while keeping a track of the protocols used both above and in the middle layers. This will let us provide a small proof of assumptions we made above. The goal of the coming sections is to put our concept on stakes and see if the formulated ideas work or not.

5.7.4 Validation

We will focus on a couple of points at this stage before we proceed and these points are actually the key points that we need to, somehow, develop and bring into practical consideration by making experiments on set of data collected regarding IPTV and making use of it as prototype. The three factors are:

1. size of the retransmission cache
2. buffer size of packets received at the end user
3. time taken from the retransmission cache to end user (back and forth)

5.8 Design

Here will come up with a small design of the system we want to develop and work on as a base for our experimental results. This will provide provision to for protocols as well, to be executed practically and hence building up a system on the basis of what we discussed.

5.8.1 System compilation

The most suitable protocols for IPTV stream delivery, as we discussed in chapter 2 are RTP and UDP. In IPTV stream, while providing a retransmission from access node to the end user via the cache, UDP can be used but it is alone not sufficient as we need to have information of the missing packets, so RTP is used as well which provides with this edge. However, RTCP is used as well which provides us the mechanism of feedback. So RTP would be used with UDP and RTCP for reporting back, as it is the process of protocol to do so.

Another edge to this scenario comes by using MPEG-TS (from chapter 2). It will be used for multiplexing of the audio/video streams from the IPTV stream server. So, this will provide to stream a single transmission of RTP audio/video. This will decrease the malfunctioning for the parts of the system involved.

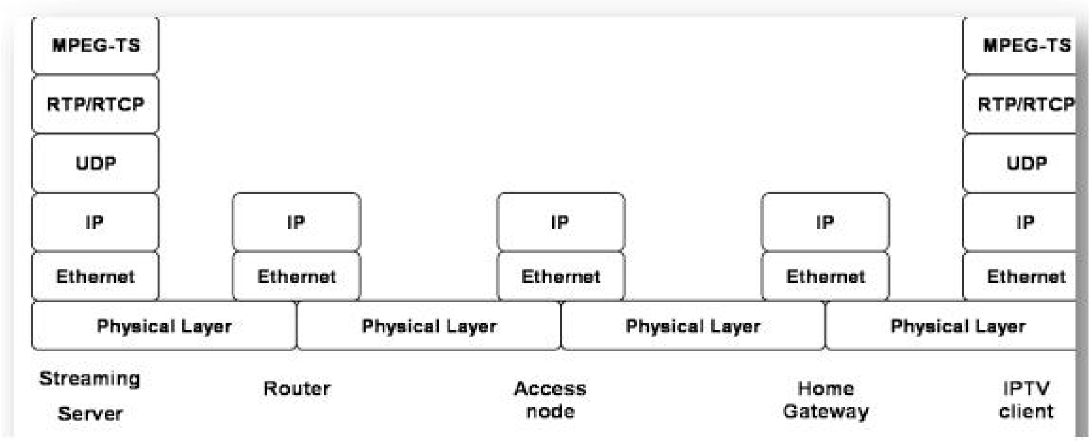


Fig 5.8.1 Multicast in IPTV embedded protocols

To make sure that the packets are transferred from the access node to the retransmission phase, support of UDP would be required as well, at the access node. So we introduce UDP as well on the access node. An interesting thing to notice here is that how will the retransmission cache proceed with its work. As it is located in the access network, how would it receive the packets from the streaming server?.

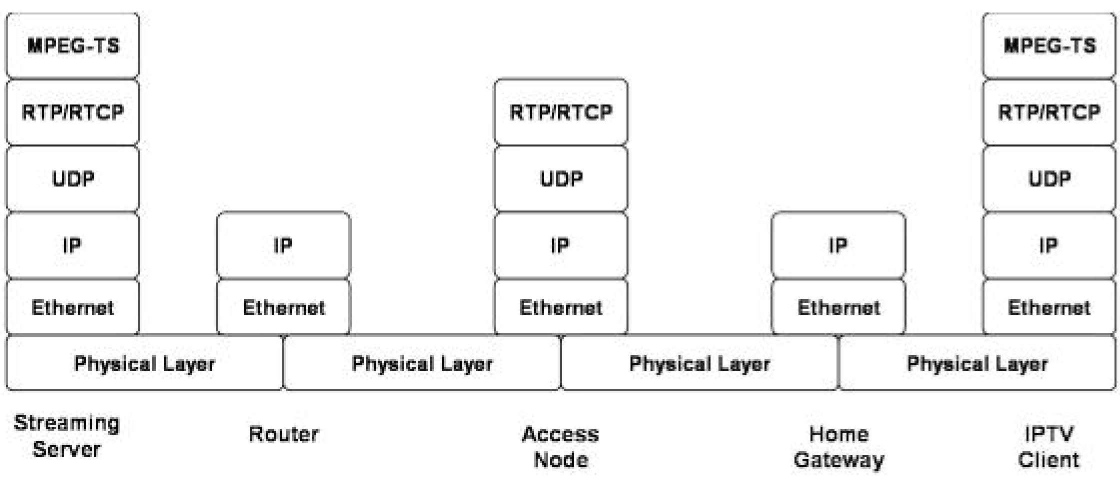


Fig 5.8.1 for retransmission in protocol stack

As the packets are sent from the server to the IPTV client, in the middle of the streaming process, the cache will somehow sneak the packets and keep them in the cache. The lost packets are then retransmitted by the cache when it comes to receive the feedback report from the client, which is of course the duty of RTCP. There are two approaches that can be adopted for the retransmission request:

5.8.2 Request approaches for RTCP

One approach is the direct approach, to directly ask the cache for the retransmission and the next is the indirect approach. The indirect approach is a hidden approach aimed by the cache to transfer the lost packets. In this approach the packets of RTCP are sent to the same address which was used to send RTCP feedback reports. This can be either the streaming server i.e. RTP server or the RTCP node [49]. When the RTCP report is received, the cache starts sneaking the packets and then retransmits them when asked.

The second approach is the direct approach, in which the client asks the cache for a direct transmission of the lost packets, only by knowing the network address of the cache.

5.8.3 Request approaches for RTP

The packets originating from the IPTV server (sender) also follow two approaches. Two sessions on whole will be used, one would be from the server to the client, and the other would be from the cache to the client. On the other hand, the IPTV client would be tackling with two sessions, either the session from the streaming server oriented or the session made from the cache to the client. The retransmission session, however, is optional as it the choice of the client is to either ask for it or not. Only those clients that have suffered packet loss and are interested to get them back could be joining the retransmission streaming.

We will now split the system into sub-components, this will make us understand and split the process into refined entities and then to implement the already constructed system for retransmission.

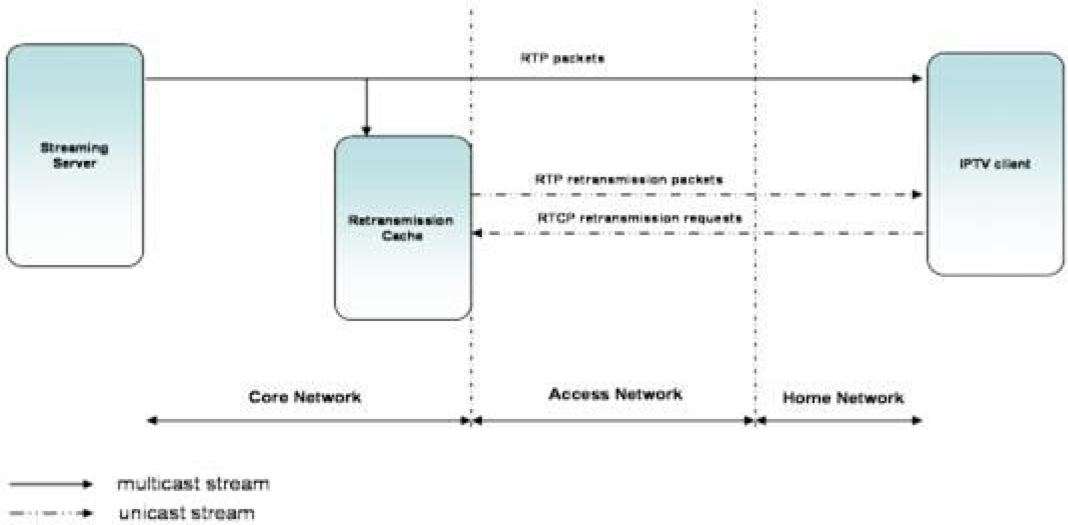


Fig 5.8.2/3 Retransmission components

5.9 Retransmission protocol

In the client, we will have a packet buffer. This buffer is unique in a way that the client when asks for the packets from the server, this buffer stores a limited number of packets to avoid jitter. This is because of the fact that the application only processes the entire application frames when the audio/video frames as a whole. This packet buffer can also be used for retransmission purpose. It will work the same, when it receives the request for the lost packets.

Cache will continuously receive packets from the streaming server and sneaking the packets and storing them for a short duration and later on discarding them. The cache also checks continuously that what retransmission requests are, the cache also checks whether the required packets are in the cache or not. If the packets exist, the cache will transmit the packets to the client.

5.9.1 Messages for retransmission protocol

There are two types for messages that are required for packet retransmission [50]. They are:

- retransmission request message (for retransmission of a single or more packets)
- retransmission response message (resending the lost packets to the client)

5.9.2 Retransmission request message

As said, the request or the feedback is send by RTCP. The feedback message contains the sequence of the lost packets. For this feedback/RTCP request message, we will be using RFC 4585. This RFC will help us to make conclusion on the resulting packet format for RTCP.

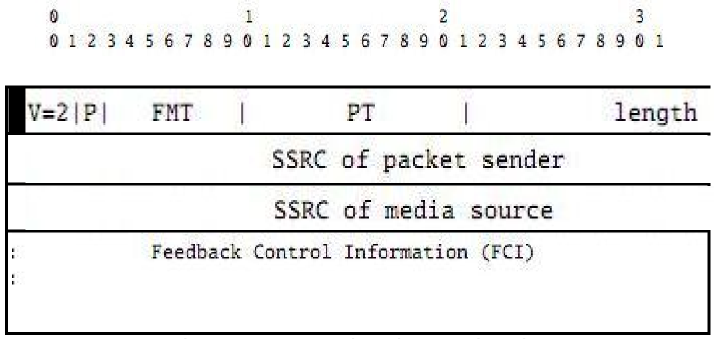


Fig 5.9.2 conventional RTCP header

With the above packet type, in the referred RFC comes a NACK (negative acknowledgement) factor, which is send when identifying for the sequence number of lost packets. NAK here consists of two fields. With one NACK message, a receiver can mention a loss of 17 packets, if multiple NACK messages are used then the number can get enhanced immensely.

a) Packet ID: this field contains sequence number of lost RTP packets

b) Bitmap of the lost packets: it is a 16 bit field and each bit is associated with any 16 packets that are followed by packet ID field.

5.9.3 Retransmission response message

A RTP payload format is needed here because IPTV packet retransmission mechanism doesn't require any altered version [50] to be introduced at application layer. However, the RTP payload format has been discussed in chapter 2. A set of adaptations is shown here, for the RTP header field.

- SSRC field in the packet provides same synchronization process which is used for providing multicast in IPTV.

- Simple RTP rules are followed by the sequence number; just one higher number is given to the prior packet in the cache while doing the retransmission mechanism.

- The original packet timestamp will be copied by the timestamp of retransmission.
- Dynamic payload

The RTP retransmission payload header consists of header of 16bit and the original sequence number and RTP's original payload.

5.10 Types of transmission

It is quite clear that IPTV content is transferred through multicasting to the IPTV clients, but the problem arises in when retransmission takes place and to deicide for the type of transmission. There are two possibilities that exist: via *Unicast* or via *Multicast.*

5.10.1 Unicast retransmission

In unicast there is single time retransmission from the cache to the client. Retransmission request is send from the client and the cache then responds back by sending the packets to client. Unicast retransmission is effective in a way that it operates independently, without depending on the other clients asking for retransmission. The hold-back mechanism in multicast (next topic) is not needed in a unicast session, which will definitely reduce the complexity of the whole retransmission. Performance of the cache in the retransmission becomes an issue because the processing performance of the cache can be effected in the worst case scenario where all the clients request for same packet at the same time.

5.10.2 Multicast retransmission

In multicast distribution, the retransmitted packets are simultaneously received by all the clients that request for packets. This is very useful when there is similarity between the packets lost by various clients. So all the clients will be treated as a multicast group, thus receiving the same packets they all asked for. So a single retransmission request shall be taken by the cache to serve all. As an advantage to unicast, in multicast retransmission all the clients can see which client is requesting for which packet. It is not possible in unicast. If all the clients send the request for same packet, then it will lead to retransmission flare.

To avoid retransmission burst, the IPTV server can hold back retransmission by keeping record of recently transmitted packets and doesn't allow the same packets to be transmitted back in a short tenure of time. If a client is asking for the same packet to be retransmitted again, then can be a clash here. As an alternative, all the feedback reports are transmitted by the RTCP protocol via multicast distribution. A random back-off timer can be used here for sending a retransmission request. When the time ends up, if the requested packet is not received by the IPTV client, then it will send the request, otherwise it will be stifled.

There are however, certain problems related to multicasts scenario: the worst possibility is that all the clients asking for different packets and then all the packets need to be retransmitted, which will enhance the bandwidth as twice for the delivery of IPTV stream, So every packet for every stream. The amount of bandwidth needed for retransmission depend upon following factors:

- Clients in a IPTV multicast network
- Lost packets type

Some conclusion can be drawn by making comparison of the above discussed factors studied in the retransmission types.

- In multicast, the number of users in the whole network becomes a problem, when IPTV clients ask for packet retransmission. The more the clients, the more the blockage. Though a hold-back mechanism is introduced but that only delays the retransmission by adding the time factor.
- In unicast, the performance of the cache can be a problem when more clients ask for retransmission. However, there is less complexity in this scenario.

Therefore, our personal preference would be a unicast retransmission process to be used in IPTV client environment, with low number of users. This will definitely avoid congestion on the link and provide better results.

Summary

In the above chapter, we have been able to discuss many aspects of IPTV when packet loss occurs. Specifically we have been successful enough to come up with the idea of applying retransmission to IPTV stream. Later in the chapter we came up with an idea to design our own cache that can be used to retransmit packets to the end user. It is quite interesting that our imaginary model is a unique and it would certainly provide us with great change in ratio of packet recovery.

The whole scenario is quite useful for a unicast environment as we were quite convinced by the facts which came forward in our study. After reading many papers and certain specific thesis previously done on IPTV, we were able to conclude that retransmission is very effective in IPTV environment. Moreover, other common error correcting techniques were also discussed, as a matter of fact, all the techniques discussed above were able to recruit lost packets but our focus was to retransmit packets by providing an imaginary design of a cache that could “Steal” packets and resend them when asked by the end user.

Chapter 6

Simulation

For our simulation, we came up with a simple model of TCP and UDP and providing a focus on these three factors.

- focusing on average time
- throughput
- Response time.

These factors are important in simulating any streaming scenario, usually for real-time multimedia applications. IPTV as we discussed thoroughly is a new and upcoming technology with lots of phases to be explored yet. In our 2nd and 3rd chapters, we have been able to make up a prototype of a cache and a path which would provide us less packet loss when it comes down to retransmission. Before going into results and snaps of TCP UDP model, we would first draw a list of areas which we focused and assessed for our simulation and we concluded that some areas need to be focused more by researchers for more effective video streaming.

Our main reason to provide discussion on IPTV retransmission and metrics associated with it was to bring QoS in a network and implement such prototype in an application which enhances QoS of an IPTV network. The structure below is a network in which the models of QoS are shown, implemented in a TCP UDP scenario [51].

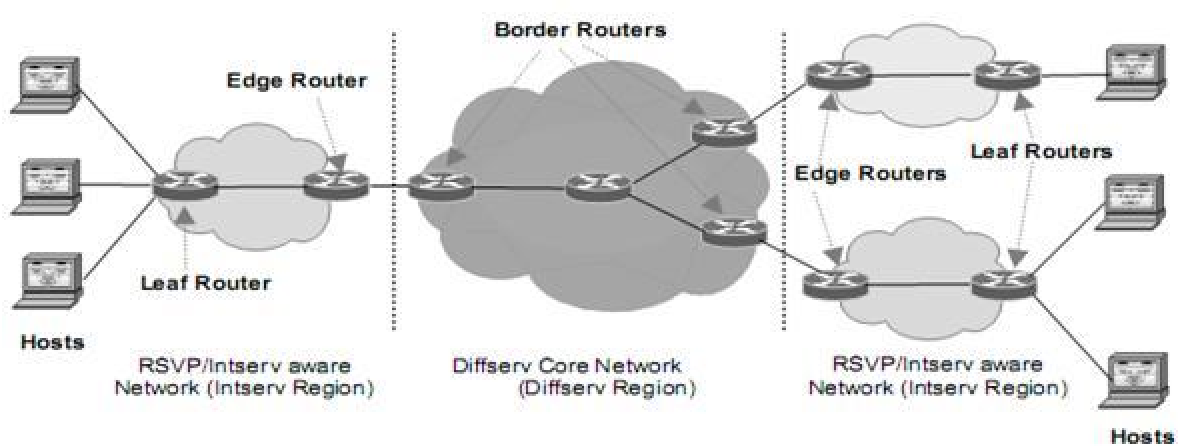


Fig 6.1 A general QoS architecture for TCP/UDP

To provide support for UDP in real time application (IPTV) over internet, it is of sheer importance that enough bandwidth is provided for certain applications so that the performance of this protocol is not diversely affected. Flow of UDP is not lagged back when congestion is followed, infact UDP flows more passionately then TCP. Therefore keeping an eye on bandwidth associated with applications having these two protocols, it is good to watch over response time and throughput time. It also important to protect reactive TCP flows from non-reactive UDP flows.

6.1 Overview

The difference between TCP and UDP is that the former provides connection-less services and the later provides connection-based services. The result can be that data trade using TCP can take longer while UDP would be the opposite. It is quite clear that TCP requires a three way hand-shake process to establish a connection before sending the data and a four way hand-shake before it is to terminate the connection. On the whole, destination has to acknowledge the data before it is received.

UDP on the other hand doesn't take time to build up connections, rather a big margin in saving delay and bandwidth. UDP is always used when doing small data transfer, for instance customer card information etc. However, if large data is to be send then the fact of extra overhead in TCP can be ignored.

6.2 Goal

The main objective of our simulation is to compare response time, get average time and throughput of the both protocols and build up our suggestions on the basis of results.

6.3 Scenario

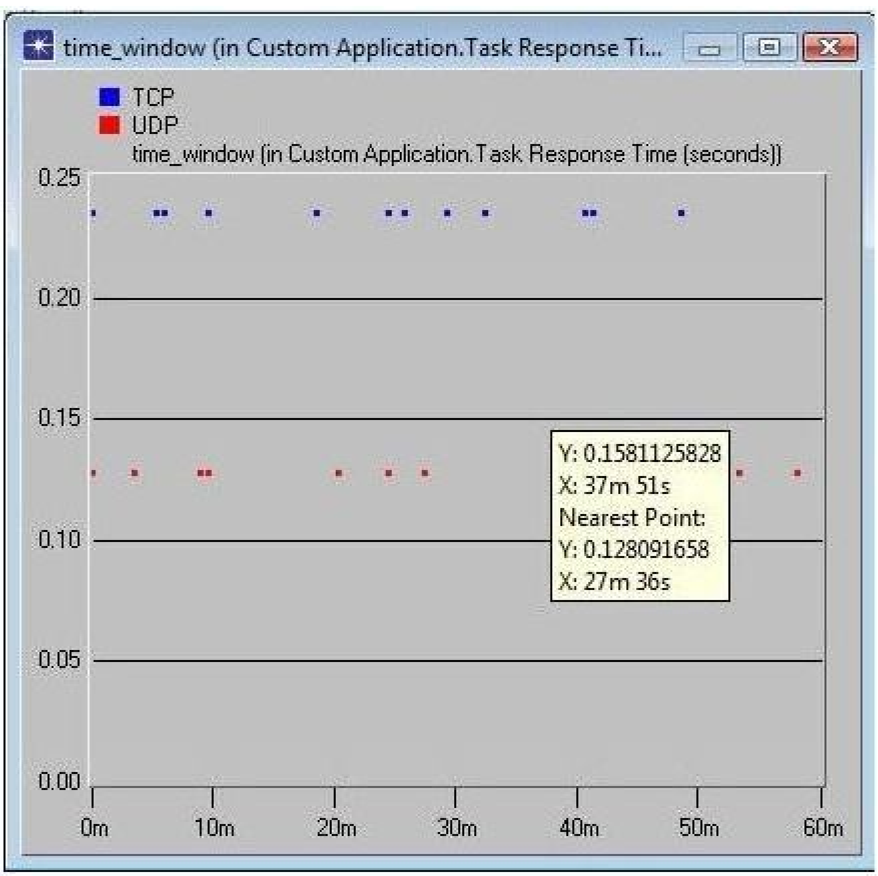
We created a scenario with an application client (we consider it is our end user in IPTV environment) and an application server (in this case it is the streaming server). The response time, average response time and task response time means that how the packets are send by the streaming server and why we felt the need to introduce our prototype of cache for retransmission. This set of application server and application client also helped us to make certain decisions on allowing certain data speeds. We connect these two servers via an IP cloud and using a DS1 link which allows a minimum speed for an IPTV service i.e. 1.5 mbps.

A custom application is made to run on either top of UDP or TCP. This application required a single request from a client and a single response is generated for every request and so the whole response time is simulated for a particular duration of time. This association is then maneuvered into a PROFILE that can be linked with our clients and servers. At first we create our scenario for TCP and run it, then later on run our same scenario for UDP. The first piece of simulation we get is for “Task Response Time”.

6.4 Results

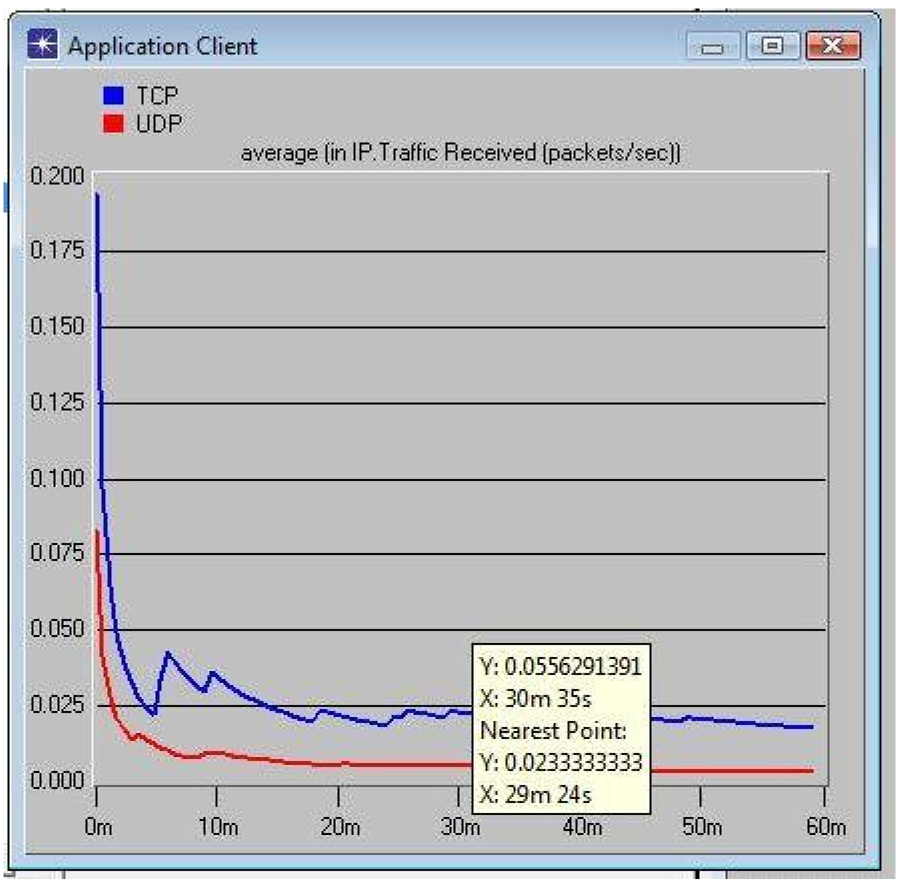
6.4.1 Task Response Time

This piece of statistic is for “Task Response Time (seconds)”. This statistic shows how long each task took to complete while transferring packet from streaming server to the end user. There is one point for each time the application was run and one set of points for TCP and one for UDP. We can see it clearly that the application took twice as long to complete each task when it is using TCP as it did while using UDP. This is only due to the fact that there is a lack of connection setup and terminating a connection in UDP.



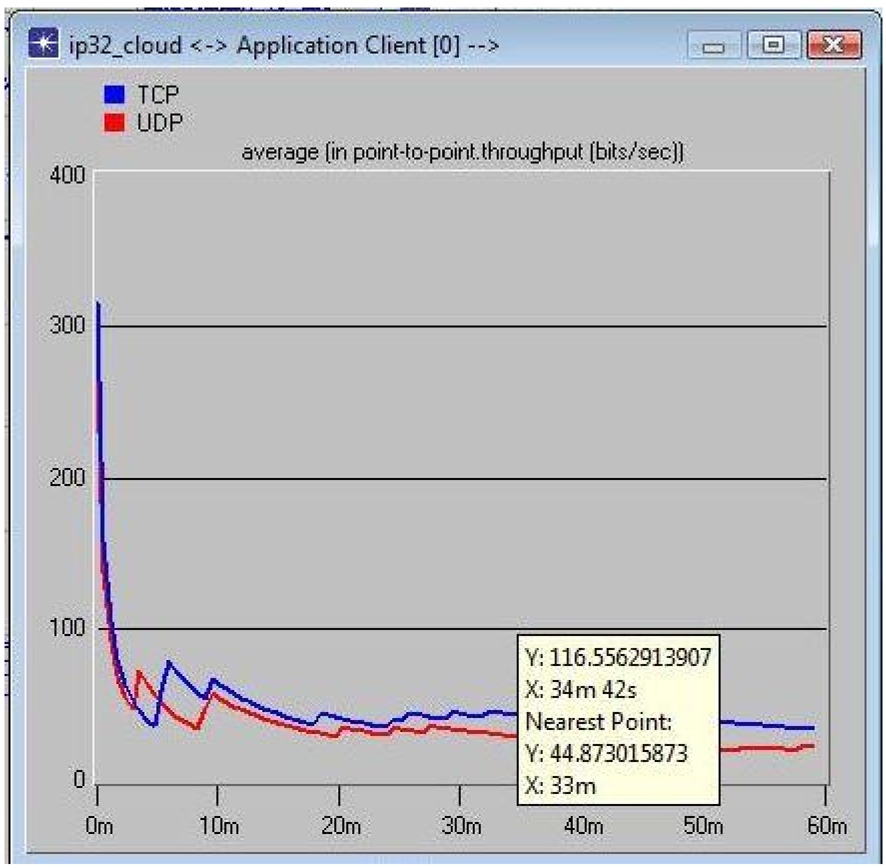
6.4.2 Traffic Received

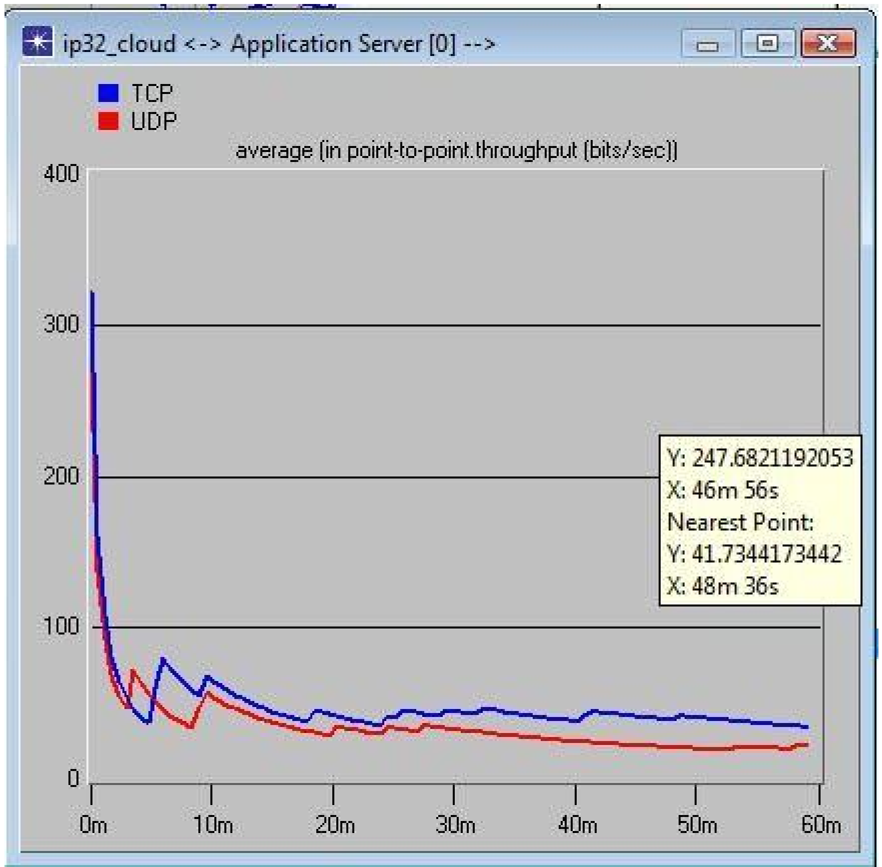
The next task provides us with Traffic Received (seconds) statistics and we used average mode to view the statistics. It is quite clear that the client received a large number of packets in every second in case of TCP as compared to UDP, nearly 5 times as many. This happened because of the building up and terminating a connection is TCP.



6.4.3 Throughput

To view throughput for point-to-point items, we view the results from both the sender and the receiver. The statistics shows us the results of the amount of the traffic that is seen on the link between the client, WAN and server. We provided a single phase traffic result from client to IP cloud and from server to the IP could. Though we came to see that almost the same number of response and requests were created and executed, the bandwidth usage is less in UDP case. This is only due to the fact that UDP doesn't need to provide acknowledgement from receiver when making or terminating a connection, like TCP.





6.4.4 Analysis

From the above graphs, we can easily conclude the relationship between TCP and UDP. However, if we make provisions in the parameters which are grouped together to make the above displayed results, a list of tasks can be further achieved and further work can be proposed if changes are made keeping in mind the following parameters.

- On the basis of configuration parameters such as task configuration and profile configuration in task specification, **Source → Destination field**, **Destination → Source traffic**, in profile configuration (duration, repeatability and inter- repetition time), what would be the results if there are no control packets such as connection packets and teardown packets? How can this effect the packet retransmission in IPTV? Answers can be found regarding size of request, size of response and the intermediate delay.

- Over head can be calculated after making assumptions in the above step for UDP or the percentage of bytes on the link.

- **DEST → Source Traffic** in the **Task Specification** can be modified for both TCP and UDP for a constant packet response. What effects can this cause? What is the size of response? Task Response comparison for both TCP and UDP?

- If we edit the attributes of **ip32_cloud** and modify the **Packet discard ratio** to 5%, how would this effect the response time for both TCP and UDP?
- Setting up tearing down UDP connection setup and layering RTP protocol on the top of UDP, how to achieve this?
- After all the above assumptions proposed, the next step can be to use UDP against TCP in retransmission phase at access node, which would allow a guaranteed delivery of packets.

All this above can be used beyond this point for research purpose and to expand, analyze and modify the model which we presented and thus emphasis more on different packet loss concepts in IPTV environment. However, this must be kept in mind that these are just assumptions and working on them would lead to results not applicable on the IPTV environment but as a general TCP UDP concept sharpening process.

Chapter 7

Conclusion and Future work

In this small chapter of few pages, we will discuss our analysis on the basis of simulation results and the retransmission cache we provided and of course some suggestions for the future work are also provided.

What consequences can packet loss impart on streaming applications, one like IPTV?

The Set-top box cannot convert the whole stream of data and decode it when it comes across packet loss from the stream because of the missing data. Visual impairments would be caused, which are discussed in previous chapters along with image distortion. A number of factors are considered here for impairment. An error in a visual stream can flow in the entire stream under the compression process it has been compressed. At time when an indicated frame is impaired, the following frames would also be impaired unless the new indicated standard frame has arrived. The time span of the error lasts until the entire supposed frame lasts and so does the error.

What error techniques can be used to avoid packet loss?

At this stage, we have a diverse range of techniques that can provide error resiliency and explicit results can be obtained if techniques like error concealment, bandwidth adaptation, payload interleaving is brought to use along with righteous use of the reference/indication frame. Congestion reduction can be avoided when bandwidth adaptation is practiced but the technique is not to good enough to be implemented in an application layer multicast. However the best two techniques for recovery are Forward error correction (FEC) and retransmission. The most favorable would be to provide a tree mechanism for multicast using retransmission (topologies discussed earlier). This would recover the lost packets quickly and also avoid feedback rush in case of data from streaming server and the cache.

How to achieve retransmission in multicast using Real-time protocol (RTP)?

RTP doesn’t possess the ability to provide a feedback message in case when individual packets are lost. This issue has been kept under notice for long an extension has been provided for RTP that provides a mechanism for feedback messages. A generic RTCP feedback can be used to alert for packet loss and RTP retransmission payload format can be used also to provide retransmission (RFC 4588).

The retransmission process is done by our proposed small cache, which makes use of this protocol in the access node. The cache sneaks onto the packets and then those packets are only transferred when they are needed.

How to measure error techniques effects on packet retransmission?

In 2nd chapter, we talked about several metrics and techniques to measure the evaluation of IPTV video stream. We cannot use different techniques at the same time during a live TV because these metrics/techniques need a reference video frame or point to interact with the user.

What factors impact the retransmission mechanism in RTP?

After our thorough study of implementing RTP as a protocol for streaming applications, we made up couple of conclusions that can effect retransmission mechanism performance. Some of them are as under.

- Size of the buffer of the assumed cache placed in the access node.
- Number of attempts cache makes to send the packet with a particular duration.
- Packet loss and detecting process.

From the simulation results we can easily conclude on how TCP and UPD can be used for improving retransmission, avoid overheads, transmission speed, suitability of data size and what types of applications are feasible to use these protocols and reliability.

In UDP, retransmission cannot take place, applications like IPTV are to detect lost and perform retransmission itself. We can clearly see from our graph that in UDP overhead is very low; in TCP it is low as well but higher than UDP. Similarly transmission speed is high in UPD but in TCP it not as high as UDP.

Why TCP is rarely used in IPTV environment?

The main focus of delivering this service and our thesis was to provide in-time delivery of the IPTV service and in good shape without tainted packets. So we need to make sure that we use a protocol which is able to provide the above said tasks. TCP does provide an extensive range of application with guarantee as well in comparison to UDP, but it is not a famous choice of transport protocol for subscribers. IPTV cannot afford to have delay in delivering stream, the following conclusion we make up on the basis of our results.

- Delay and sensitivity trade-off
- TCP is connection oriented
- Error correction maintenance

When an error is encountered, the end user would either wait for the loss packet to be retransmitted and allow a gap in the flow of the video or else to discard the packets when they arrive late.

A lack of supporting multicast in case of TCP is observed, while UDP is able to provide such capability and also providing broadcasting services in multicast environment, UDP is quite effective. A set of following points are listed after simulating UDP against TCP and choosing it over for IPTV environment.

- UDP can manage the stream of IPTV and no pause would be observed in delivering the IPTV stream.
- Low overhead
- A speedy connection would setup
- One way transmission, so the cache would easily support any UDP stream.

Regarding our cache, we can further implement it in a famous subscriber company that are providing IPTV services and then make evaluations on it. It can lead to make more changes in the cache we had been able to implement or proposing design for. We can use our retransmission mechanism for IPTV payload. Example can be providing retransmission mechanism for sample frames that are to carry the error further and carry low bandwidth for retransmission.

8

Addendum

ACK	Acknowledgement
AL	Application Layer
ALM	Application Layer Multicast
CODEC	Compression/Decompression
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexer
EPSNR	Estimated PSNR
FB	Feedback
FEC	Forward Error Correction
HD	High Definition
HG	Home Gateway
HMTP	Host
HTTP	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	International Telecommunication Union
MOS	Mean Opinion Score
MPEG	Moving Pictures Experts Group
MPEG-TS	Mpeg Transport Stream
NACK	Negative Acknowledgement
PIM	Protocol Independent Multicast
PIM-DM	Protocol Independent Multicast Dense Mode
PIM-SM	Protocol Independent Multicast Sparse Mode
PLR	Packet Loss Rate
QoE	Quality of Experience
QoS	Quality of Service
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SAP	Session Announcement Protocol
SD	Standard Definition
SDP	Session Description Protocol
SSRC	Synchronization Source
STB	Set-Top Box
TCP	Transmission Control Protocol
UDP	User datagram Protocol
VCEG	Video Coding Expert Group
VOD	Video on Demand

VOIP	Voice over Internet Protocol

9
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