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Internet Protocol Television (IPTV) Services

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This thesis mainly deals with IPTV (Internet Protocol TV) technology and how it changes the business of television; its development and architectural design; its applications and progress into the future. The goal of the project is to enhance professional networking on both live TV and radio platform, know how the IPTV functions and how it differs from internet TV, how it is formatted, transported and delivered to the end users. Equally important, how providers charge for it and make a living.

The study itself was carried out by retrieving information from different sources such as the library, the Internet, through self-observation, and discussions with the chief supervisor and instructor. Different aspects of IPTV are discussed in different phases of the thesis. First, the study introduces IPTV technology, its background and means of transmission. Then, the study entails the architectural design of IPTV, multimedia methods and applications, compression techniques and finally its purpose and role to the growing technology services.

The purpose of the project was to gain adequate practical experience, skills, techniques, and theory by applying previous classroom knowledge to actual principal-like situations in a strategic, organized and supervised environment. The end result came from the fact that in the near future it is likely that IPTV can replace traditional TV technology since it delivers a good supplement business model for service providers, offers better quality of service to consumers, and play a significant role on the fast growing and evolving interactive TV applications such as VOD.

Keywords	IPTV, VOD, DSL, QoS, IP, RTP, STB

List of Abbreviations and Acronyms

HDTV	High Definition Television
IPTV	Internet Protocol Television
VOD	Video-On-Demand
TV	Television
QoS	Quality of Service
IP	Internet Protocol
EPG	Electronic Program Guide
RF	Radio Frequency
DSL	Digital Subscriber Line
ADSL	Asymmetric digital subscriber line
HDSL	High data rate Digital Subscriber Line
SDSL	Symmetric Digital Subscriber Line
VDSL	Very high bit rate Digital Subscriber Line
IETF	Internet Engineering Task Force
IntServ	Integrated Services
DiffServ	Differentiated Services
SLS	Service Level Specification
PHB	Per-Hop Behavior
STB	Set-Top Box
NVOD	Near Video-On-Demand
QVOD	Quasi Video-On-Demand
TVOD	True Video-On-Demand
USB	Universal Serial Bus
RIP	The Routing Information Protocol
RTCP	Real-time Transport Control Protocol
RTSP	The Real Time Streaming Protocol
PIM	Protocol Independent Multicast
SM	Sparse mode
DR	Designated Router
RP	Rendezvous Point
DM	Dense mode
RPF	Reverse Path Forwarding
ICMP	Internet Control Message Protocol
IGMP	The Internet Group Management Protocol
DSAD	Digitally Sampled Analog Data
AAC	Advanced Audio Coding

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1 Introduction

Over the past decade, the only way to watch television was through over-the-air broadcast and cable signals. The emerging of satellite, digital cable, and High Definition Television (HDTV) services have made it possible for telecommunication providers to discover a new technology in the television broadcast system. The innovation of digitization of television technology around the globe has facilitated access to multiple services, with better quality of service on all devices at all point of time.

Internet Protocol Television (IPTV) has provided the means to securely deliver high quality triple play services to the end usersover a private or managed network. IPTV functions just like a standard pay TV (Television) service and one of its key benefits is to offer IP (Internet Protocol)based services in one integrated package, for example receiving and displaying live or pre-recorded audio and video, as well as covering live TV or Video on Demand (VOD).

IPTV systems are a significant aspect in the telecommunication field of technology as they enhance professional networking on both live TV and radio platform. The author's interest to have a better understanding of how to build a video on demand, know how the television signals are transmitted, how they are formatted, transported and delivered to customers was a key drive to choosing this topic. Equally the importance to know how providers charge for it and make a living.

This thesis is based on a project the author worked on during an internship period at Streamafrik. The study itself was carried out successfully by retrieving information from different sources such as the library, the Internet, through self-observation, and discussions with the chief supervisor.

2 Theoretical Background

The aim of the project was to explore the development of internet protocol television and its different phases, as well as the transmission distribution mechanism that allows for immediate interactivity and multimedia experience.

2.1 IPTV Vs Internet TV

IPTV is often mistaken for internet TV since they both use IP technology for video delivery. This section discusses the key differences between this two IP technology services as shown in Table 1. IPTV services are delivered via private and managed network using the internet protocol suite whilst Internet TV services are distributed over open, public or global internet [1, 21-25]. This enables IPTV delivery to allow for higher quality of delivery with secure delivery of content to the end users. Internet TV video delivery, by contrast, can be subjected to longer waiting times due to lower bandwidth, high traffic or poor connection quality.

Table 1 illustrates some of the key difference between IPTV and Internet TV. Modified from [1, 26].

	IPTV	Internet video
Nature of content	Continuous streams of content.	Delivers discrete content segment.
Content selection	Hundreds of programming channels.	Millions of content offerings.
Content format	One or two formats are selected by the provider.	Dozens of format with multiple players.
Delivery platform	Uses a secure dedicated private IP network.	Uses a public internet.
Access Mechanism	Viewed on Consumer TV via set top box.	Consumer PC display or portable device.
Geographical reach	Networks controlled by telecommunication providers and operators.	No geographical limitation
Costs	Mostly based on monthly subscription	Multiple content can be viewed free of charge

In Table 1 above it is clearly arguable to note that both IPTV and internet TV play a significant role in delivering video across a network platform. They both rely on IP technology for delivery, their approaches in delivery differs in the way in which the signal travels and how the content is delivered over the internet. Internet TV model is open to any rights holder and anyone can create an endpoint and publish on a global basis offering a direct communication between the provider and the consumers.

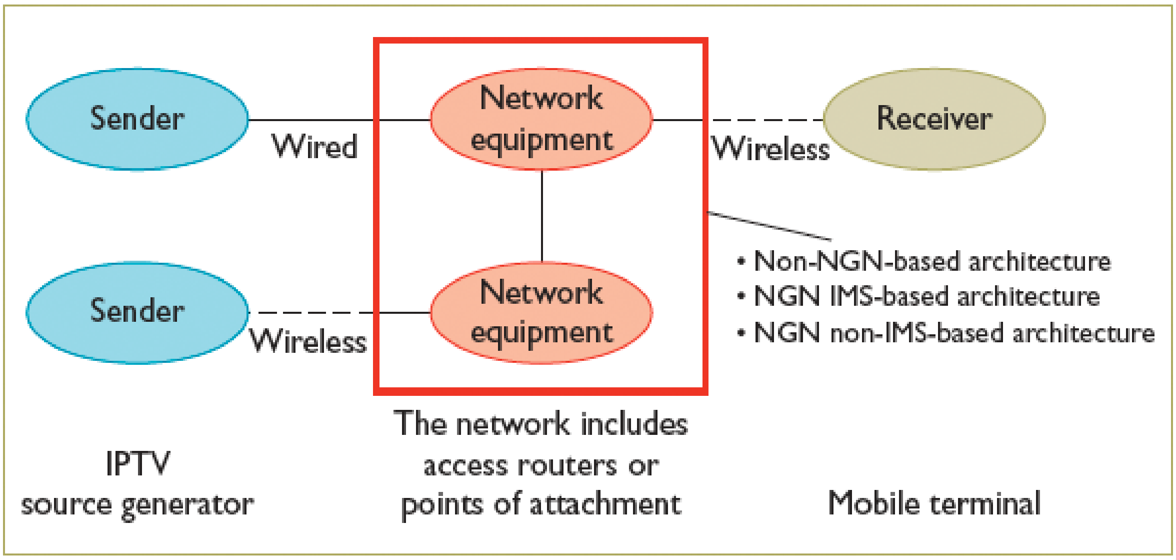
2.2 Mobile IPTV

Mobile IPTV is a wireless mobile transmission platform that enables users to receive multimedia content such as audio, graphics, video, and text over a wireless IP network to a mobile medium with support for mobility, security, quality of service (QoS), quality of experience, and reliability functions [2].

Mobile IPTV supports the following capabilities and features:

- It supports browsing of IPTV content information by using the Electronic Program Guide (EPG).
- It has mobility capabilities to end users.
- Mobile IPTV allows for streaming of high image quality TV on mobile device platform.
- It supports multiple languages and enhances faster interactivity and optimization.
- End user device of mobile IPTV services can be a Smartphone or a tablet using iOS (Originally iPhone OS) or Android.

Figure 1 shows a representation of mobile IPTV Architecture [33, 24]



The Next Generation Networks (NGN) enable unrestricted access for users to networks with a wide range of services offered by different service providers as shown in Figure 1. In this scenario both the sender and receiver are assumed to using a mobile device [33]. This mobility capability enables communication between the sender (service provider) and the receiver (at the mobile terminal) over a wireless interface.

There are two main approaches used to deliver mobile TV, that is, across a cellular network and across a dedicated broadcast network.

2.2.1 Cellular Network Approach

In cellular network approach, any broadcasting distribution network can be adopted as it may incur the loss of quality of the mobile operator's services since it competes with data and voice services for bandwidth. Cellular network use a secure infrastructure that would essentially reduce stationing to users of mobile TV services.

2.2.2 Dedicated Broadcast Network

Dedicated broadcast network on the contrary is a traditional digital broadcast network that combines broadcasting service and internet to deliver IP based broadband, data services, and optimize the provision of mobile TV via IP networking. The system delivery mechanism can be terrestrially based, satellite based, or a combination of both[3].

2.2.3 Mobile IPTV Benefits

There are many advantages that mobile IPTV offers, such as:

- Mobile IPTV provides variety of real-time streaming data such as VOD and video services much better on mobile phones.
- It allows for mobility of services based on Multimedia Subsystems (IMSP) and wireless characteristics to IPTV.
- It provides digital television experience via interactivity.
- Mobile IPTV allows channel switching and casting.
- It provides Information access and entertainment for the users.
- It allows for content synchronization and offers an opportunity for watching TV everywhere and anytime.

It is easy to see that mobile IPTV provides variety of new interaction level between Internet, voice and video. Its wireless capability help to speed deployments and reduce costs in the best possible way to reach out to users. The users only need a normal colored screen phone with fair display resolution and mobile data connectivity to use this technology service.

2.3 Signal Transmission

A signal is an electric or electromagnetic pattern of one or two independent variable and representation of data to convey information. Signaling is a mechanism in which data is transmitted across different medium ways such as twisted-pair, coaxial cable, fiber-optic cable, radio frequency (RF) waves, satellite, and cellular telephony [4].

Signaling is carried out with the help of signals that indicate to the connected end device what data is requested across a media platforms. It plays a significant role in receiving information such as video, audio or encoded data.

Types of signal transmissions

There are two types of signal transmission used to transmit information in form of audio or video usually through electric signals, that is, analog and digital signals. A well-known example of analog vs. digital is that of clocks as illustrated below (Figure 2).

Figure 2 Analog vs. Digital. Modified from [5]

Dubbing analog signals can often be skewed by just a few frames or by service seconds. Every dub might be different due to generational loss. For example, in Figure 2, digital clock specifically indicates time as 7.00, but analog signal time appears to be either 7.00 or closer to 7.01 [5].

2.3.1 Analog Signalling

An analog signal is a continuous signal that contains time varying amplitude, voltage, current, and frequency. Information is converted into an analog signal of varying amplitude of high and low physical property (such as voltage or current) over time. For examples sound, voice, temperature vary continuously in frequency and amplitude.

In Figure 3, the sine wave's amplitude value can be seen to be either positive or negative between higher and lower points of the wave respectively, while the frequency (time) value is amplified in the sine wave's physical length from left to right. Each time the signal is amplified, the noise is also amplified.

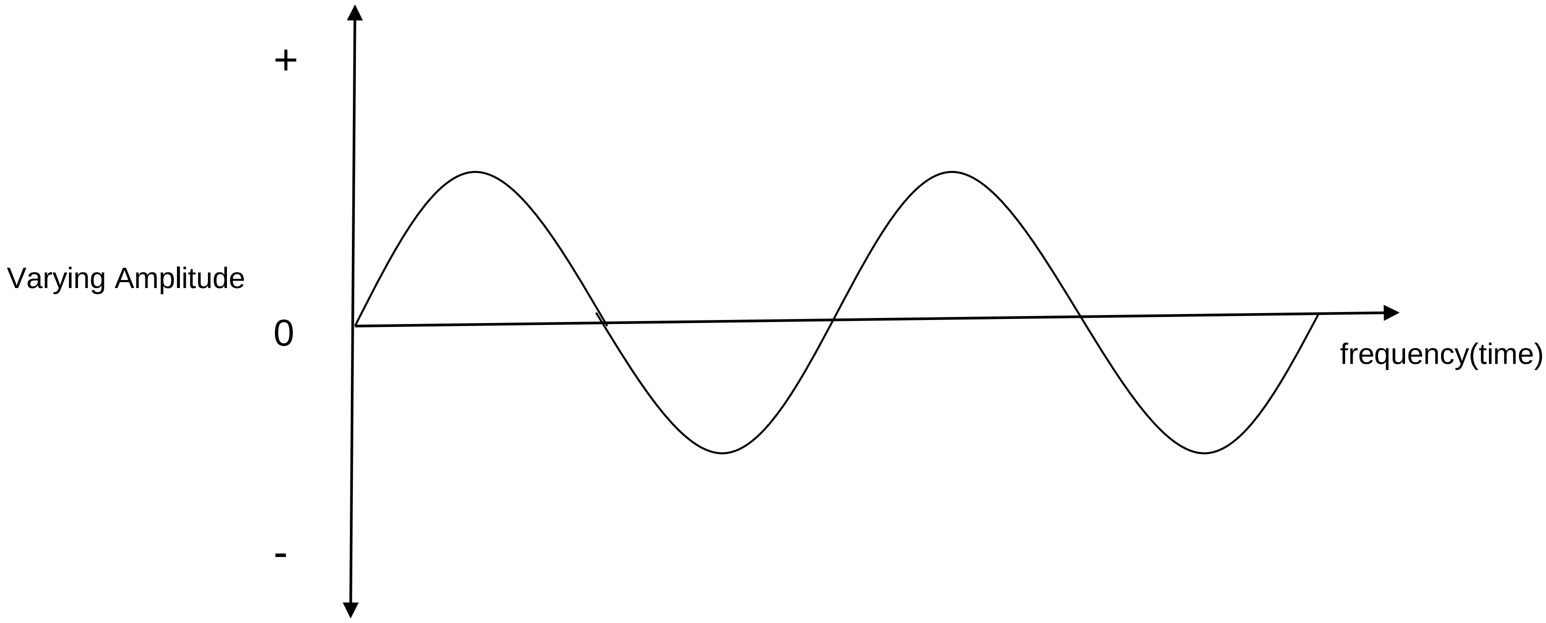


Figure 3 Sine wave of analog signal

Signals can be periodic or non-periodic analyzed in the frequency (time). Sine wave and square wave are the common representations of analog signals. Square wave is distinctive from digital signal by a negative minimum value.

Advantages of analog signaling

- Analog signaling suffers less attenuation than digital signal over long distances [6].
- Analog signaling defines infinite amount of signal resolution. Analog devices are equipped to handle the infinite values between 1 and 0 [6].
- It is a simpler implementation for easy processing and reproducibility.
- Analog signaling has a much higher density which can be multiplexed to increase bandwidth at the same time make good use of the bandwidth [34, 8].
- Analog is better for higher frequency applications, where low cost and computation portable are required in real time.

Most sounds such as music and speech are analog signals. The main advantage of analog signal is the potential for an infinite amount of signal resolution. Compared to digital signals, analog signals are of higher density.

Disadvantages of analog signaling

- Analog systems are less immune to noise, that is random unwanted variation, over long distances. The noise becomes dominant creating disturbance and distortion [34, 8].
- Analog systems are more likely to get affected by generation loss.

Even though analog signals is used in many systems today, its uses are declining with the introduction of the more reliable digital signal.

2.3.2 Digital Signalling

A digital signal can be defined as a series of pulses consisting of discrete (discontinuous) values of binary format(zero's and one's), where each bit is representative of two distinct amplitudes. Digital signals can be uniform, square, or discrete in shape. Figure 4 shows an example of digital signal format.

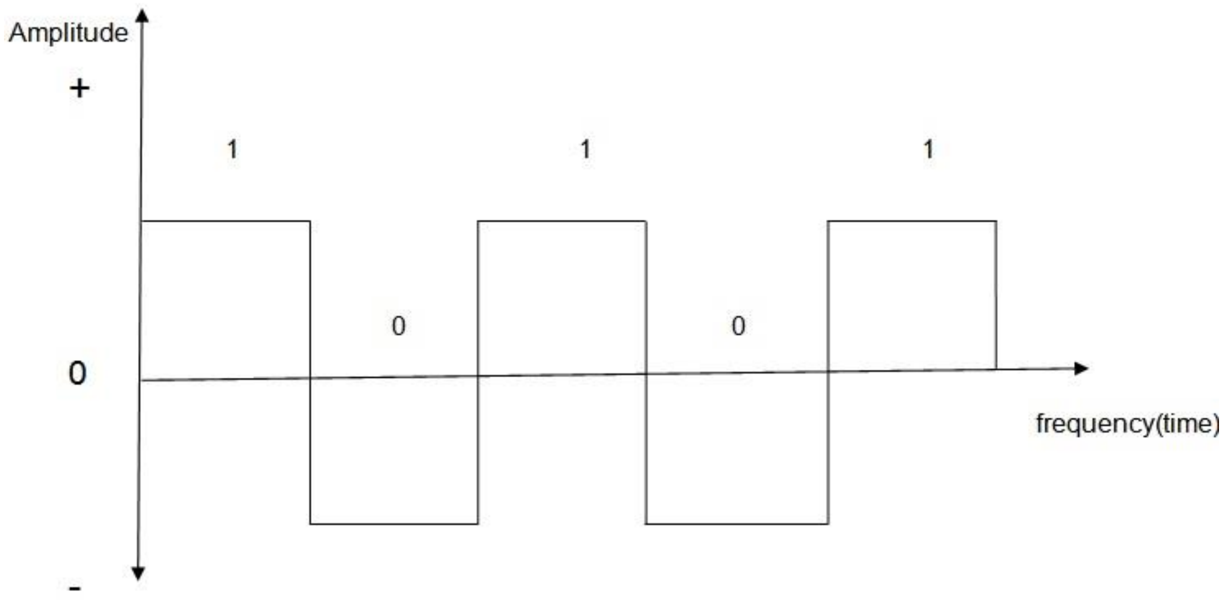


Figure 4 an example of digital signal format

Each pulse represents a signal element and monitored periodically by network. Binary data are transmitted by the presence or absence of signal elements. '1' represents the presence of transition and '0'represents the absence of transition [6].

Advantages of digital signal processing

- In digital signaling, the quality of the signal is maintained due its higher interference immunity to external background noise. Noise does not accumulate on a digital signal as it does on an analog signal during transmission.
- Digital signaling is compatible with integrated digital data and telephone signaling which can be implemented with a relatively low equipment cost.
- Digital signaling offers various transmission options over long distances due to its linear and nonlinear capabilities.

Disadvantages of digital signaling

- Digital signals like radios can be costly and not easily portable.
- Digital signals are intolerant to RF noises, where sampling can be complex resulting in signal error [34, 11].

It is often recommended to convert analog signals to digital signals for more effective signal processing. Video and audio transmissions are often transferred or recorded using analog signals.

3 Architecture of IPTV Systems

3.1 Digital Subscriber Line (DSL)

Digital Subscriber Line is a broadband connection technology that enables telecommunication network providers to offer high speed internet connection over existing 2-wire or 4-wire standard copper telephone lines, thus eliminating the need for costly infrastructure upgrades. DSL modem's ability to transmit data simultaneously in uplink and downlink communication allows telecommunication providers to use their existing networks to provide high speed connectivity for on-demand streaming audio and video entertainment to their subscriber bases. DSL connections are point-to-point dedicated circuits that are cost effective, fast and reliable.

In DSL the computer connects to the phone line which then connects to the DSL modem that has filters in place for different frequencies of voice and data. In other words, the data connection is on the same line as the phone while travelling at different frequencies. The data and the voice goes back and forth through the internet at the same time. This way, DSL makes it is possible for the user to experience high speed internet connection even when talking on the phone.

DSL modems establish a connection from one end of a copper wire by utilizing more of the bandwidth on the analog line, thus allowing for greater bandwidth to the other end of that copper wire and connecting digitally on both the uplink and downlink connection. DSL modems can enable downlink connection speeds greater than 6Mbps and uplink speeds up to 1MHz of bandwidth in both directions, thereby preventing the signals from interfering with each other [7,2]. Figure 5 illustrates a connection setup of the DSL network.

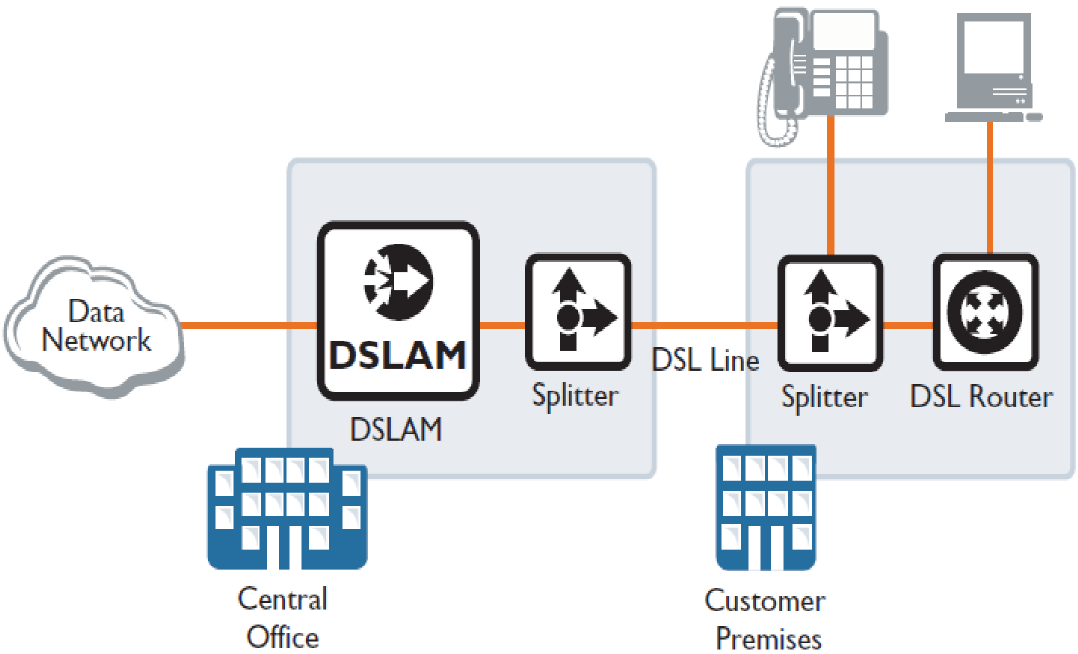


Figure 5 connection setup of the DSL network. Copied from [16, 5]

DSL modem's digital signal is not limited to 4 kHz of voice frequencies, making it much faster than 56K analog modems in bandwidth capacity. The bandwidth rate available are more consistent to the end user that are within 18000 feet [7,2]. Longer distances must operate at lower bit rates to allow more subscribers to be served from a single central office at a lower price.

Variants/subtype of DSL

There are many different DSL service types' options for broadband and IPTV, such as ADSL, HDSL, SDSL, and VDSL referred collectively as XDSL. These variants of DSL technology provide different data communication capabilities to different users. The following section briefly states how each variant functions.

3.1.1 ADSL (Asymmetric Digital Subscriber Line)

ADSL is a subtype of DSL technology that uses a single pair and transmits higher data rate downstream than upstream, implying that the download speed is greater than the upload speed [17, 273].Upstream in this scenario refers to the transmission of data from the subscriber back to the network or central office, and downstream refers to the transmission of data in the direction towards the subscriber. An example of this type of application is VOD. Figure 6 shows a connection setup of the ADSL network.

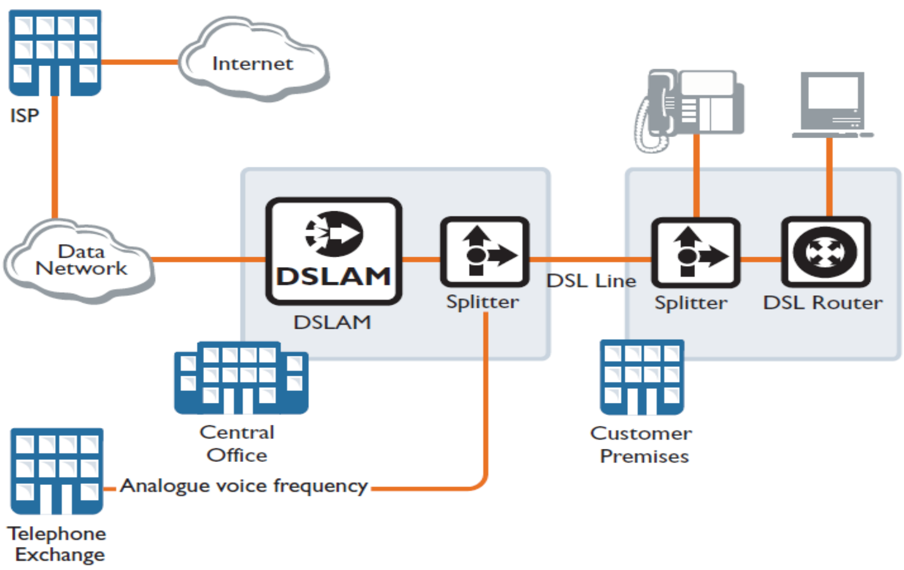


Figure 6 connection setup of the ADSL network. Copied from [16, 11]

Installation maintenance is a major problem for ADSL network operators. In order to make installation easy, a version of ADSL known as ADSL Lite was developed by the

International Telecommunication Union (ITU). ADSL Lite does not require a filter in the customer premises, it can reach speeds of up to 1.5Mbps downstream and an upstream rate of 640Kbps [17, 275], enough to provide internet surfing access, remote LAN access, multimedia access, software downloads, video-on-demand and home shopping. ADSL is ideally suited to home and small office users who are downloading more rather than uploading contents.

ADSL is the commonest form of DSL and by farther the most stable and affordable way to access broadband internet. It uses existing telephone lines but splits it into two channels, one for voice and one for data. This way the user can use the phone while accessing the web at the same time.

3.1.2 HDSL (High Data Rate Digital Subscriber Line)

HDSL is a particular type of SDSL, symmetrically delivering 1.544 Mb/s in both downstream and upstream directions over two sets of copper twisted pair lines of up to 12000 feet, which is the same rate as T1 digital line type connection. It is possible to extend the distance by using repeaters along the line to the customer.

HDSL is a better way of provisioning and transmitting T1/E1 over copper wires, using less bandwidth and requires no repeaters up to the standard range [9]. It is heavily used in cellular telephone build outs.

3.1.3 SDSL (Symmetric Digital Subscriber Line)

SDSL is similar to HDSL since it transmits same data rates (1.544 Mb/s) for the upstream and downstream channels simultaneously in both directions across a single telephone line. SDSL connections typically allow transmission of up to 6 Mbps in both directions, but usually require a 4-wire connection. This limits SDSL's reach of approximately 3km. SDSL service is more expensive than ADSL, therefore it is ideally suited to individual subscriber premises for connecting LAN's over short distances and video conferencing.

3.1.4 VDSL (Very High Bit Rate Digital Subscriber Line)

VDSL provides the highest data rates of the DSL technologies, being able to deliver data at a transmission speed of up to 52 Mbps across a single copper cable but only over short distances. VDSL is limited to distances of up to 2 km [17, 276]. This type of DSL technology is particularly useful for supplying high data rate services for hotels, university campuses and business parks that are closer to the telephone company's central office.

VDSL can also be used to connect premises distribution network to the optical network unit to handle a whole range of high bandwidth applications, such as multichannel of high definition TV broadcasting, VPNs, file downloading or uploading, video on demand and surveillance systems.

3.2 Quality of Services

This section covers the requirements of QoS, conceptual model, implementations and management of various QoS mechanisms to enable network administrators and architectures to deliver good quality traffic, full duplex communication, low levels of delay, and allocate bandwidth in a way that improves applications performances across a network in both directions.

3.2.1 Quality of Service Conceptual Model

QoS is a crucial element of any administrative policy since it measures the ability of network to deliver data (end-to-end) with predicted results and computing systems to provide different levels of services to networked applications such as, video conferencing, internet telephony, or voice over IP applications and associated network flows. This services include error rates, network traffic loads, up-time, latency, and bandwidth[12, 89].These applications require explicit quality of service guarantees in terms of good quality traffic, full duplex communication, and low levels of delay.

The Internet Engineering Task Force (IETF) started working on a new group to develop a framework for defining the services and service model and at the same time, an architecture for an internet which can give quality of service guarantee [12, 89]. There are three different types of service models for providing QoS on a network namely Best-effort, IntServ, and DiffServ.

3.2.2 Best Effort

The internet routing architecture is based on a best effort approach in which network delivery of IP packets are treated in the same way to provide a scalable and reliable network foundation. In best effort model, QoS is not applied to packets, the packets arrives anytime in any order and no preferential treatment is guaranteed. For example, critical data is treated the same as email. Best effort is the best suitable model for non-real time applications such as telnet, file transfer or web browsing email in which QoS is not necessarily needed.

3.2.3 Integrated Services

The Integrated Services (IntServ) model was designed to supplement the best-effort delivery by reserving bandwidth, buffer and central processing unit time for applications that require bandwidth and guaranteed packet delivery to end-to-end QoS over the network [12, 89]. IntServ expects applications to signal their requirements to the network to provide very high QoS to IP packets.

The signaling protocol which is used to set up the resource reservation in the IntServ model is known as Resource Reservation Protocol (RSVP). RSVP is a general signaling protocol where the receiver can specify its traffic characteristics and reserve network resources through a network for an IntServ service. This quality of service attributes can be either at the individual application flow level or at aggregate level.

One drawback of this type of service model is that the individual applications and flows must be maintained in the intermediate nodes and routers, making it impossible to maintain large number of states. A new architectural framework known as differentiated services model was introduced to solve this problem.

3.2.4 Differentiated Services (DiffServ) Model

DiffServ framework was designed to overcome the limitations of both the best-effort and IntServ models. It provides an almost guaranteed implementation QoS to a variety of end-to-end services across the network IP packets while still being flexible, cost-effective and highly scalable [31, 585]. The Differentiated Service model provides the ability to assign different levels of services and QoS treatment to different network traffic.

Essential features requirement for scalable QoS model

- Resources must be provisioned at aggregate class level but not at per-flow level.
- End to end quality of service should evolve gradually.
- DiffServ should not be independent on any dynamic signaling.

Requirement of differentiated services model/architecture

- It needs to accommodate a very wide variety of services and provisioning policies. Diffserv micro flows are subjected to policing and marketing according to service level specification (SLS). SLSs specifies the service to be given to each aggregate in the transmitter.
- It need a Per-Hop Behavior (PHB) which provides an appropriate way of traffic conditioning and aggregate flow forwarding of traffic on a DiffServ interior node to perform relatively coarse level of traffic classification within the network core routers and switches in the path.
- No end to end services will be defined and no per flow or per customer or per aggregate state will be maintained in the core router. Core routers will only look at the index or label in the packet and router handles each packet differently.
- The resource will not be reserved by any hop by hop signaling.
- The admission control modules must ensure that new reservations do not exceed the aggregate traffic capacity to provide end to end services.
- It must be able to achieve scalability and gradual upgrade of the existing system.

With Differentiated Services, the scaling properties are achieved by marking each packet's header with one of the standardized code point to deliver a particular kind of service based on the QoS specified by each packet.

3.3 Video on Demand

Video on demand is an interactive TV technology that allows simultaneous real-time access of audio and video materials to users and viewers over a cabled network. The concept of VOD is based on the transmission of videos and audios in a coded and compressed format and stored on hard disk or individual Set-Top Box (STB), and later decoded and decompressed by set-top converters and then sent to the local server.

The main VOD system level consist of; a local database and server to store and provide access to programs, and a standard TV receiver along with a set-top box that allows users to browse and play back a selected video as if they are watching from videotape or a video player. Some of the main types of VOD systems are Quasi Video-On-Demand (QVOD), True Video-On-Demand (TVOD), and Near Video-On-Demand (NVOD). These systems and based on the amount of interactivity allowed[11, 4].

Quasi Video-on-Demand

QVOD is a service in which users are grouped based on their interest. Programming will only be presented if a minimum number of subscribers sign up for it. Users can choose between different programs by switching to a different group.

True Video on Demand

True video on demand is a service where the user receives an individual video stream and has full control over requested playback media item. The user has full control of continuous interactions such as start, stop, pause, forward, reverse at different speeds, and full-function virtual video cassette recording capabilities [1, 36]. True video-on-demand is achieved by paying a fee for each service request.

Near video-on-demand

NVOD is a service in which a particular program is simulated to start in discrete time intervals(of about 5 or 10minutes window) over a particular channel or multiple channels. NVOD are used for pay-per-view services, where a subscriber pays electronically and selects time and day to start watching a particular program. Even though NVOD only limits the subscriber to a 5-10 minutes waiting before the start of a program, it significantly reduces the cost of services to subscribers [1, 36].

3.4 Triple Play Services

The concept of triple play service is considered as delivering telephony Voice over IP(VOIP), IPTV, blended IP multimedia streaming services, and high speed internet services over a single network using either fiber optic cable, copper cable, or satellite transmitter.

Triple play services can be useful in multiple ways that include:

- Delivery of multiple services such as voice, video, and data over one single network [12, 203-204].
- Flexibly to adapt to the next generation of multimedia-enabled networks and scalable for future upgrade and maintenance.
- Triple play is cost effective enough to reduce operational and management costs. This has made investors to maximize profit and increase return on investment.
- Triple play ensures a flawless user experience by offering mobility to enable subscribers to do what they want anytime, anywhere.

To deploy a viable triple play service, the network must be more distributed to cost effectively deliver video and broadband; reliable, to allocate bandwidth to provide optimal quality of experience for the subscriber; and flexible, to adapt to the next generation of multimedia-enabled networks.

3.5 IPTV Set-top Box

STB is a device that decrypt incoming signals into a synchronized format that directly connects to an end-to-end IPTV services to enable subscribers to access a variety of different types of digital entertainment content and video-on-demand programming content [12, 53-56]. A STB has a variety of TV interfaces at the back and front for connectivity to a variety of different networking infrastructure [14].The set-top box back channel allow two-way communication to support interactive features like, adding premium channels, option to play or stop live transmission, and the ability to record or save programs for future watchable purpose. Figure 7 shows a typical example of IPTV set-top box.

Figure 7 typical example of IPTV set-top box. Modified from [15]

A typical digital set-top box has a physical height of 2.5 inches and a width of 18 inches. As can be seen in Figure 7,the installation of STB is simple and involves plugging one HDMI cable into the TV set and another into STB interface. Subscribers are provided with a handheld remote or wireless keyboards to choose what they want and gain access to different channels and contents supported by the STB.

There are many different types of STBs based on different standards and geographical locations. The most commonly known types are IP Set-Top Boxes(STBs), Hybrid IP STBs, Hybrid IP Satellite STBs, Hybrid IP cable STBs, Multicast and Unicast IP STBs, Digital STBs [14].

4 Multimedia over IP

Multimedia is an important aspect in IPTV. It utilizes the combination of different media types including text, audio, video, animation, and graphics interactivity content forms. This section gives a clear picture of different multimedia transmission methods, applications and protocols used to transmit packets over a network.

4.1 Video Conferencing

IP Videoconferencing is a live communication technology that allows users to be seen, be heard, share information, effectively interact and communicate face to face from separate locations around the world without having to travel to a single location. It usually involves simultaneous transmission of audio, video and often text. The two most popular videoconferencing software being used over the Internet are CUSeeMe and Microsoft NetMeeting.

Videoconferencing is useful in general applications such as on demand meetings or scheduled meetings, telemedicine, classroom practices and cross collaboration between partner institutions. It offers the ability to exchange ideas and knowledge, collaborate and communicate, share documents, share information, and even presentations thus allowing decision making to be faster and easier. Videoconferencing purposes have been found to be extensive and vary in nature.

4.1.1 Purpose and Benefits of Video Conferencing

The benefits and the most common purposes of videoconferencing are listed below.

- Video conferencing enhances face-to-face communication and collaboration in real-time between two or more people regardless of location.
- It reduces cost of travel, maximizes productivity, improves work life and learning experience, and accelerates decision making.
- Video conferencing provides live sharing of full-motion video images, text, and high quality audio between two or more geographical locations providing an experience that is effective.

- Videoconferencing is often used in educational institution such as colleges and universities for live broadcasts of lectures and seminars to distance learning students. It also promote cross cultural exchanges between students and tutors, workforces and teams as well as stimulating collaborative learning.

- Keeps people in business organizations connected and unified and improves communication between them. In this manner, employee training and group work is facilitated.

It is easy to see that there is a growing need for videoconferencing both in business and the educational fields.

4.1.2 Components of Video Conferencing System

In order to understand how video conferencing works it is important to recognize the component parts of the system.

1. Camera

A webcam is used to record and send video signal that is required for adequate inter-activity between other distant people during a live video conferencing session. These can range from a simple desktop camera, small Universal Serial Bus (USB) camera to other more high definition camera quality systems equipped with remote control pan, autofocus, status indicator, automatic pan and zoom features.

2. Video Display /Monitors

The monitor displays the far end images to connected distant people received from the videoconferencing codec. The monitor devices come in multiple options that is plasma screens, liquid crystal display, projectors, and cathode ray tube.

3. Video Conferencing Codec Unit

The codec unit is used to digitize and compress video information into a digital signal using encoding program to decompress the received transmission for play-back[18].Common video codec's used in video conferencing applications are H.261, H.263, H.264, MPEG2, and MPEG4.

4. Microphone / Audio Sub-System

Many of stand-alone video conference systems automatically come with either a small USB or analog microphones attached to a computer to enhance the audio capabilities of the system and help with larger group interaction. Microphones can be of two types; a unidirectional microphone picks up sound from one direction and an omnidirectional microphone picks up sound from all directions [19, 68].

5. Computer with fast processor speed and large RAM

A computer with fast processor speed is needed to run the videoconferencing software. It compresses and decompresses video streams and maintains the data link to the network [18].

6. Speakers

A good set of speakers is essential to enable one to hear the audio from the far end of the videoconference. Videoconferencing systems require a guaranteed symmetric bandwidth for a point-to-point connection and multiple video performance (explained in section 4.1.3).

4.1.3 Types of Video conferencing

There are two main types of videoconferencing systems; point to point videoconferencing and multipoint videoconferencing. This section briefly covers how these two types of videoconferencing systems function.

1. Point-to-point video conferencing

Point to point videoconferencing is simplest scenario of videoconferencing system where participant in different location communicate with each other using desktop videoconferencing software. Point to point video conferencing does not require a bridge to function and each participant or group must use the same type of connection protocol. Desktop systems are typically designed for a single user. They may be software client based, web based, video-enabled IP telephone or dedicated standalone video appliance [20].

2. Multipoint video conferencing

Multipoint video conferencing are mostly commonly designed for more than two levels of interactivity. It includes integrated audio, life-sized images, large flat panel display devices and visual enhancements, which enhances the reality of the interaction. Multipoint conferences give the participants the feeling of being present in an actual meeting, as well as the ability to see any content being shared during the meeting even though the participants are geographically dispersed.

Multipoint conferences are created using a multipoint conference unit (MCU). The multipoint control unit either sends or receives calls from participants who dial the network ID of the MCU to initiate a multipoint videoconference [20].

4.2 IPTV Networks

There are three methods used to transmit packets over a network: unicast, multicast, and broadcast. They are introduced in detail below.

4.2.1 Unicast

Unicast by definition is one-to-one communication private session usually between a server and a client. During a unicast session, each distinct stream is sent to only those recipients that request the stream. VOD is an example of unicast application for IPTVs in which the source servers establish an individual communication line to each requesting end user.

The end users during the VOD sessions will have the ability to pause, rewind and have overall control to the video being steamed because of the established direct session with the source server. A big concern for the video of demand services is higher bandwidth demand requirements due to multiple individual direct connection between the source server and the requesting end user. Figure 8 illustrates how data flows under unicasting.

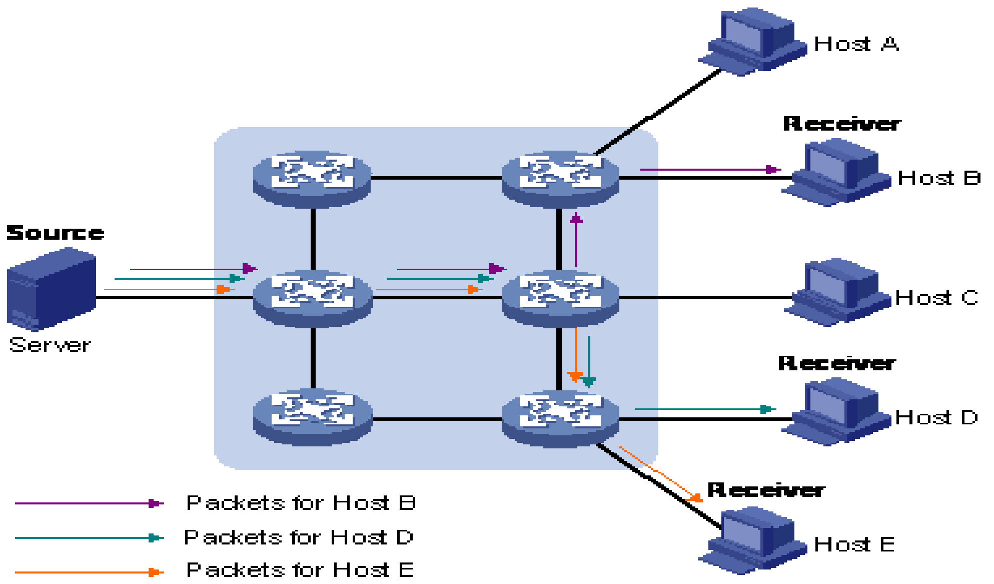


Figure 8 shows how data flows under unicasting [23].

The information source sends a separate packet to a single host over the IP network as shown in Figure 8. In this scenario, the server creates a separate transmission channel for each host (E, D, and B) and the packets are received only if the destination address matches one of its own IP addresses.

Unicast routing

Unicast routing is the process of forwarding unicasted traffic from a source to a unique address on an internetwork [21]. Its goal is to determine a good path (sequence of routers) through the network from source to destination. There are two main types of unicast routing used. In distance vector routing, each node can only know the distance between itself and shares its routing table with its immediate neighbors periodically.

In link state routing everyone gets a copy of a topology and computes their own routes, including the type, cost (metric), and the condition of the links (up or down). All nodes run the same algorithm (Dijkstra algorithm) concurrently to compute their forwarding table in the same distributed setting as for distance vector.

The nodes know only who they are connected to, their neighbors and the cost to their neighbors. They do not know the whole topology. The nodes can talk only to their neighbors using messages to find what is going on for the network at large since they have no other ways to gain information about the network. The Open Shortest Path First (OSPF) protocol is based on link state routing.

4.2.2 Multicast

Multicast is typically the opposite of unicast. With IP multicasting communication mechanism, the host uses a specific IP address that send packets to a set of multicast recipients to get traffic. The source server will broadcast the same session at the same time to multiple users. A real time example of multicast session will be standard TV programming whereby, the same prime time show will be playing at the same time on all TVs that are tuned to the same channel.

IPTV provides the ability to set up multicast and stream videos via multicast as opposed to unicast. This capability maintains a session with same stream of data which will also result to lower network congestion, lower bandwidth requirement and a reduction in the load of the sender and the overall load demands on the source server.

Multicast is a controlled and managed network thus making it much easier to predict the bandwidth capability by having a certain backbone going through the network. Multicast sessions offers a couple of advantages over unicast sessions. Multicasting support is optional in IPv4, but mandatory in IPv6. It supports User Datagram Protocol (UDP) only and its address can only be used as a destination address. Figure 9 illustrates how data flows under multicasting.

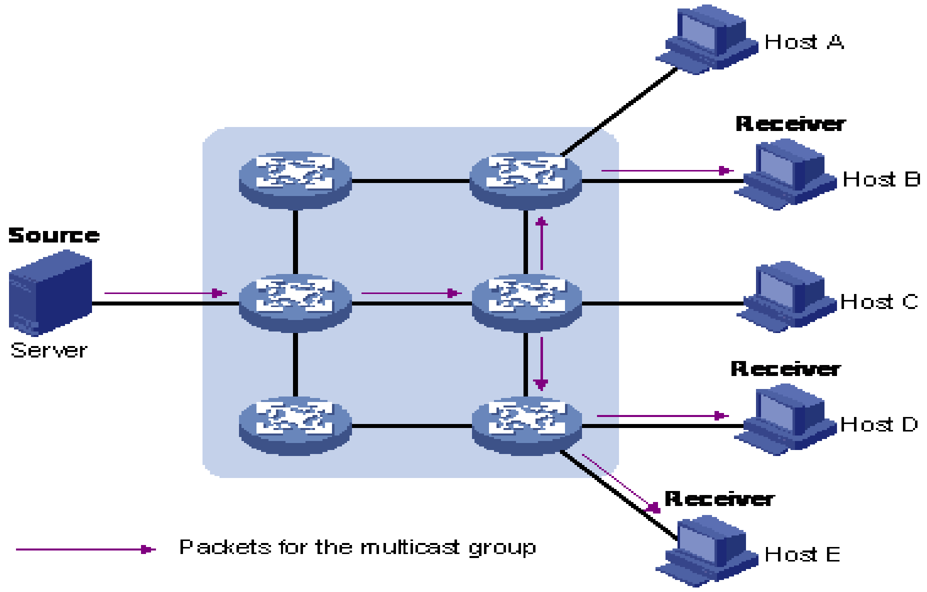


Figure 9 shows how data flows under multicasting [23]

Suppose that Hosts E, D and B need receive information from the source server, they need to join a receiver set or a multicast group as shown in Figure 9. The routers on the network duplicate and forward the information based on the distribution of the receivers in this set [23].

The information source sends packets over the IP network to hosts (E, D, and B) that joins the multicast group as shown in Figure 9. In this scenario, the server creates a single transmission channel for all the hosts.

IP multicast addresses

IP addresses are defined in RFC 1112 and class D address are used as multicast in the destination IP addresses field to identify a multicast packet. The Class D address range is 224.0.0.0 to 239.255.255.255 and the numeric overall range of multicast addresses is 224.0.0.0 through 224.0.0.

The source IP address of a multicast packet is always a unicast address (Class A, B, or C) [23]. Smaller address ranges are reserved for special use within the Class D address range.

Table 2 shows common multicast addresses reserved for various communications protocols. Modified from [23].

Address	Description
224.0.0.1	All multicast receivers on a subnet
224.0.0.2	All multicast routers on a subnet
224.0.0.5	All OSPF routers on a subnet, used by OSPF routers store exchange link-state information.
224.0.0.6	All OSPF designated routers on a subnet, used by OSPF routers to exchange link-state information.
224.0.0.9	All RIPv2 routers on a subnet.
224.0.0.10	All EIGRP routers on a subnet
224.0.0.13	All Protocol Independent Multicast (PIM) routers on a subnet
224.0.0.18	All Virtual Redundancy Router Protocol

Reserved multicast addresses are used by network protocols on network routers for different purposes as listed in Table 2. For example, an Open Shortest Path First (OSPF) router must sends a "hello" packet to an assigned multicast address, which is 224.0.0.5, and the other routers will respond. In case of multicast communication , by pinging 224.0.0.1 address, all multicast capable hosts on the network must join that group at start-up on all its multicast capable interfaces. A group of clients listening to same multicast address is known as host group.

4.2.3 Broadcast Transmission

In broadcasting a single packet is delivered from one sender to all connected receivers on the local network simultaneously. Each device that receives a broadcast packet must process the packet in case there is a message for the device [24, 250].
Broadcast packets are normally restricted to the cable network and are undesirable for streaming media, since even a small stream could flood every device on the local network with packets that are not of interest to the device.

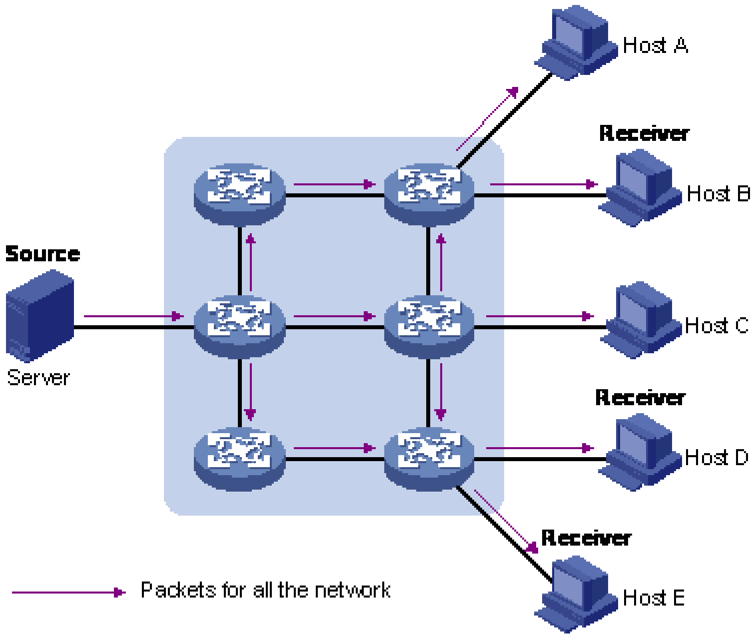


Figure 10 shows how data flows under broadcasting [23].

In the diagram shown above, the source server sends a packet to all hosts (A, B, C, D, and E) on the network segment. Suppose that hosts A,C and D do not need the information and only B and E needs the information, the source will still broadcast the information to all the host, but only the one with that IP address will respond. To use a broadcast transmission, map upper layer addresses to lower addresses, send a query to request an address and then exchange routing information by routing protocols [23].

IP Broadcast Addresses

IP broadcast addresses can be used only as the destination IP address for single-packet one-to-everyone delivery under the same LAN [24]. There are two different types of IP broadcast addresses;

1. Limited Broadcast

The limited address is the broadcast limited to single LAN and represented by setting all 32 bits of the IP address to the 255.255.255.255 [24]. It is used as the destination address of an IP datagram during the automatic configuration process such as Boot Protocol (BOOTP) or DHCP, and when the host does not know its subnet mask or network ID. For example, with DHCP packets, the client must use the limited broadcast address for all traffic sent until the DHCP server acknowledges the IP address lease [25].This datagram is never forwarded by routers, it will only appear on the local network segment. The destination MAC address for such frames will be FF:FF:FF:FF:FF:FF [25].

2. Directed Broadcast

Directed Broadcast address is the local subnet broadcast address sent to all hosts FF:FF:FF:FF:FF:FF on an Ethernet interface [24]. The broadcast address uses the highest address in the Ethernet interface range. For instance, if the subnet network ID is 192.168.0.0, the directed broadcast address will be 192.168.255.255, which will be heard by all in the same subnet hosts. NetBIOS Name Service (NBNS) uses directed broadcast packets.

IP directed broadcast is used in implementing remote management or administration task such as backups and Wake-on-LAN (WOL) on hosts in a subnet that is not directly attached to the Internet. If no IP directed broadcast is configured, directed broadcasts are dropped [26].

4.3 IPTV Protocols Network

There are many types of IPTV protocols used. The most commonly used are introduced below.

4.3.1 Real-Time Protocol (RTP)

RTP usually runs over UDP and does not reserve bandwidth or guarantee QoS. RTP is designed to support end-to-end delivery of real-time data such as voice and video from the source to the receiver, and also supports a wide variety of media-on-demand applications such as internet telephony, IPTV services and online games.RTP randomly picks even ports from UDP port or transport layer ports and encapsulate voice or video data packets.

RTP packet format

Table 3 illustrates the RTP packet format.

Bits 0 2 34 5 678 9 0 1 2 3 4 5 16 7 8 9 0 1 2 3 4 5 6 7 8 9 0 31

V	P	X	CC	M	PT	Sequence Number
Timestamp						
Synchronization Source (SSRC)						
Content Source (CSRC) (0-15)						

- V represents the version number 2.
- The padding bit (P) indicates if the padding bit is inserted at the end of the packet containing padding octets, which are not part of the payload. Padding may be required by some applications to fill up a block of certain size.
- The extension (X) bit indicates if the extension bit is set, there is an experimental one extension header after the fixed header.
- The count field (CC) tells the count of contributing source identifiers that precedes the fixed header.
- The marker bit (M) is intended to indicate the frame boundaries of a speech burst to be marked in the packet stream.
- The payload type (PT), 7 bits; this field identifies the payload format and determines its interpretation based on the information on the network feedback on application quality. RTP defines a profile specification and one or more actual payload formats, which may also describe some application specific extensions or modifications to RTP. The profile provides a range of information that ensures a common understanding of the fields in the RTP header for a particular application class [28]. These payload types include for example G.721, GSM Full Rate, G.722 and G.728 speech codecs, Joint Photographic Experts Group (JPEG) and H.261 video codecs.

- The sequence number (16bits) is an incrementing counter which is started by a source from a random number and used by the receiver to detect out-of-order packet delivery.
- The timestamp (32 bits) corresponds to the generation instant of the exact time at which first octet in the RTP data packet or payload are sampled. It provides a mechanism that enable the source to playback samples at the appropriate time intervals of the multimedia bit streams and to enable different media streams, such as audio and video to be synchronized [12, 90].
- The SSRC 32 bits is a randomly generated value that uniquely identifies synchronization sources within the same RTP session. It avoids two sources in the same RTP session to have the same SSRC. Examples of synchronization sources include the sender of a stream of packets derived from a signal source such as a microphone or a camera, or a RTP mixer [27, 85].
- The Contributing Source (CSRC)is a list of the SSRC identifies of the sources that contributes to the payload contained in the RTP header of a packet.

RTP is therefore responsibility for payload type identification, source identification, sequence numbering and time stamping. Translators and the mixers usually resides in between senders and receivers to translate and forward RTP packets from one payload to another. Mixer assigns itself as the sender of the packet, combines RTP streams from different sources into a single stream and then forwards a new RTP packet.

4.3.2 Real-Time Transport Control Protocol (RTCP)

RTCP is used during multicast audio or video transmission to receive streams of RTP data packets from one or more sources and combines them into a single stream. RTCP packets are distributed to all the participants using IP multicast. It is distinguished from RTP through the use of distinct port numbers.

RTCP packets contain sender/receiver to exchange periodic reports such as;

- Number of packets send.
- Number of packets lost.
- Inter-arrival time jitters.

It is up to the application to make use of RTCP packets. Different application may come up with different algorithms and mechanisms to best use of these applications.

Receiver reception packets

Will contain information such as;

- SSRC of the RTP stream,
- fraction of packet loss,
- last sequence number received in the stream of RTP packets,
- Inter-arrival jitters.

Sender report packets

Will contain information such as;

- SSRC of the RTP stream,
- timestamp and wall clock time of the most recently generated RTP packet in the stream,
- number of packet sent in the stream,
- Number of bytes sent in the stream.

Source description packets

Will contain information and about session participants;

- email address of the sender,
- senders name,
- SSRC of the associated stream.

RTCP is commonly used to create links on web sites that point to streaming media files [1, 225].RTCP monitors periodic transmission statistics of control packets, reports quality of service (QoS) feedback, and helps to synchronize multiple streams. For example; if RTCP packets are getting lost from the receiver, then it is obvious that the internet is congested and it is recommended to appropriately adapt the sending rate or change quantization levels.

4.3.3 Real Time Streaming Protocol (RTSP)

RTSP establishes and controls the delivery of multimedia streams with real-time properties, such as audio and video across IP networks between client and server. RTSP allows the client using a network remote control to communicate to the server information to deliver channels such as UDP, multicast UDP and Transmission Control Protocol(TCP) [12, 92].

It is designed to work with lower-lever multimedia streaming protocols such as RTP. A great protocol tool for VOD applications that have a unicast session between client and the VOD servers. Some applications that use RTSP include YouTube, Spotify, Windows Media Player, VLC player, and Skype.

Methods of control request

RTSP supports the following methods of control request made from the client to the server.

1. Record

Initiates recording of a range of media being streams.

2. Option

An option request tells the client what request types the server accepts.

3. Describe

A describe request provides the client with a description of the media to start the appropriate media applications. Figure 11 shows an example of a describe request.

```
C->S: DESCRIBE rtsp://server.example.com/fizzle/foo RTSP/1.0
      CSeq: 312
      Accept: application/sdp, application/rtsl, application/mheg

S->C: RTSP/1.0 200 OK
      CSeq: 312
      Date: 23 Jan 1997 15:35:06 GMT
      Content-Type: application/sdp
      Content-Length: 376

      v=0
      o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
      s=SDP Seminar
      i=A Seminar on the session description protocol
      u=http://www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps
      e=mjh@isi.edu (Mark Handley)
      c=IN IP4 224.2.17.12/127
      t=2873397496 2873404696
      a=recvonly
      m=audio 3456 RTP/AVP 0
      m=video 2232 RTP/AVP 31
      m=whiteboard 32416 UDP WB
      a=orient:portrait
```

Figure 11 example of a describe request [29].

4. Play

Tells the server to begin sending the bit stream for playing the media file. Figure 12 illustrates an example of a play request

```
C->S: PLAY rtsp://audio.example.com/audio RTSP/1.0
      CSeq: 835
      Session: 12345678
      Range: npt=10-15

S->C: RTSP/1.0 200 OK
      CSeq: 835
      Date: 23 Jan 1997 15:35:06 GMT
```

Figure 12 example of a play request [29].

5. Pause

Tells the server to pause media streams temporarily.

- Announce

Announce control request is used to send information to the client to register a new entry or description of a media stream. From the client's side, it displays the description of the media while from server's side, it updates the description in real time.

6. Setup

Tells the server how to transport the media for an identified media stream and which port to use.

7. Teardown

Terminates the media streaming session delivery and frees all associated network resources associated with the session.

8. Get_parameter

Retrieves the value of a parameter a specified stream.

9. Set_parameter

Changes a parameter of a stream from a URL by enabling the client to issue request to set the value of a parameter of a specified stream.

10. Redirect

Informs the client that it must connect to a different server and then moves it to that server.

RTSP enhances interactions between the client and the server by using a network remote control and adds a number of new requests to the existing HTTP requests as shown in the list above. The client first requests the description of the media using the DESCRIBE method, then requests that the session is SETUP and receives a session identifier in return. The client requests that the media streams of the session are PLAYED and at any point the client may PAUSE the media stream temporarily. When the client has completed, the client issues a TEARDOWN request to terminate the media streaming session.

4.3.4 Protocol Independent Multicast (PIM)

The protocol independent multicast is a type of multicast routing protocol that does not depend on any particular protocol for unicast traffic for its operation but it can leverage whichever unicast routing protocols used to populate the unicast table. All PIM routers multicast group, or unicast to a specific destination [30].There are two main modes of PIM that allow one-to-many and many-to-many transmission of information:

Sparse mode (SM)

PIM Sparse Mode (PIM-SM) is a multicast routing protocol designed on the principle that recipients for any particular multicast stream will be sparsely distributed throughout the internet with periodic multicast traffic [30]. PIM-SM by default uses shared trees distribution, which are multicast distribution trees rooted in multiple Internet domains designed to provide efficient communication, limit multicast traffic and used by all recipients to join and leave multicast distribution trees. Figure 13 illustrates the PIM-SM mode design with sparsely distributed streams being sent.

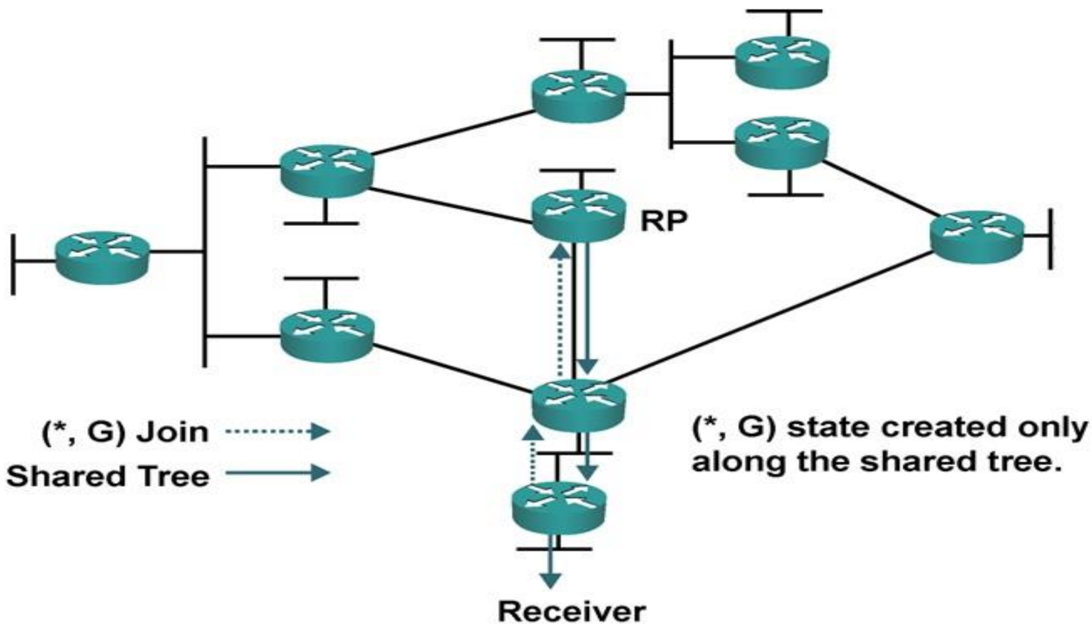


Figure 13 PIM-SM mode design with sparsely distributed streams being sent. Copied from [35].

One router is elected the 'querier' on each local/physical network, querier periodically sends membership query messages to 'all system group' (224.0.0.1) with TTL = 1.

When active receivers actively request to join a specific multicast group, routers along the path of these receivers register to join that group. Host sends a leave group message to group address G if it was the most recent host to report membership in that group. Router sends join messages to RP and sources register with RP, intermediate routers update state and forward join messages. RP can send stop messages to source if no receivers joined the group.

Characteristics of sparse mode (SM)

Characteristics of SM are listed below in details.

1. Designated Router

There must be one PIM Designated Router (DR) in each subnet in the network. Any PIM-SM interfaces on the subnet elect the DR with the highest DR priority. If there is more than one router with the same priority, or no priority, they choose the interface with the highest IP address. If the current DR becomes unavailable, the remaining switches elect a new DR for the subnet by DR priority or IP address [32].

The DR on the subnet containing a multicast source sends multicast packets towards the Rendezvous Point (RP). DRs with group members connected RP sends join messages towards the group's RP.

2. Rendezvous Point

Each multicast group must have a RP. The RP for a group or range of groups is found by an election process. The lowest preference value is elected from all the RP group range of multicast addresses.

To create a routing tree for a group with rendezvous point as a root for the tree a receiver send join messages towards the RP and the sender sends register towards the RP.

3. Bootstrap Router

Each PIM-SM network must have at least one Bootstrap Router (BSR) candidate. The Bootstrap Router for a network is chosen by election. The highest priority is elected to be the BSR. The elected BSR listens to PIM Candidate RP bootstrap messages to determine the RP for each multicast groups.

Dense mode (DM)

PIM dense mode also known has push mode is assumed that all downstream systems wants to receive multicast feed or view the multicast feed.PIM dense mode flooded across the network uses Reverse Path Forwarding (RPF) interface to receive multicast traffic. It forwards the multicast traffic through every single segment whereby some segments don't have group members interested in a multicast feed. Packets arriving via the non-RPF interface are discarded. PIM-DM will prune off the data packet destined for the group and forwarded by instantiating prune state.

PIM dense mode is recommended for small networks, to avoid more configurations and easy management. Figure 14 shows the PIM-DM mode flooding example, pruning unwanted traffic.

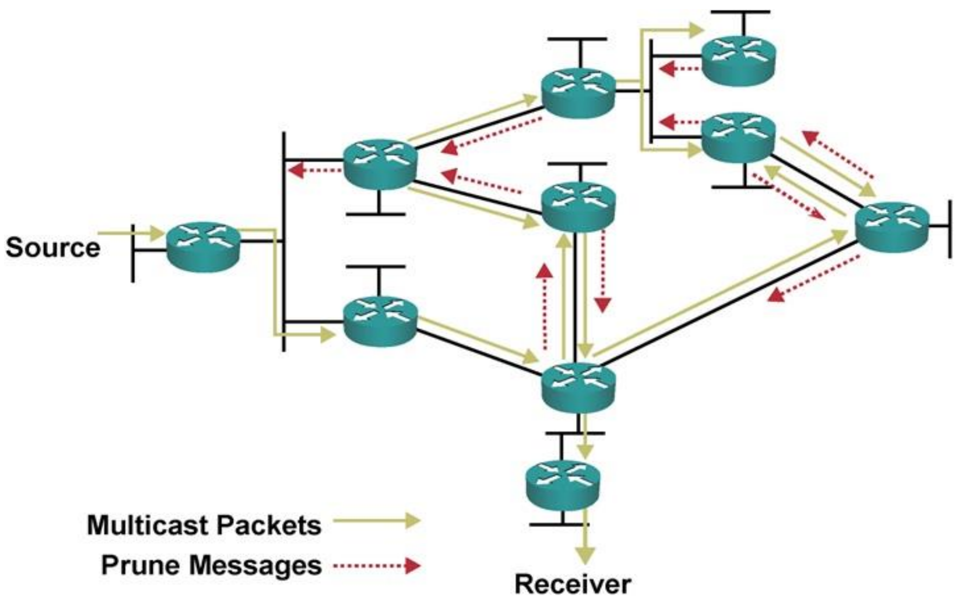


Figure 14 PIM-DM mode flooding example, pruning unwanted traffic. Copied from [35].

Figure 14, the (S, G) state is created in every router in the network, multicast traffic is flooded throughout the entire network. Figure 15, on the other hand, illustrates the PIM-DM results after pruning.

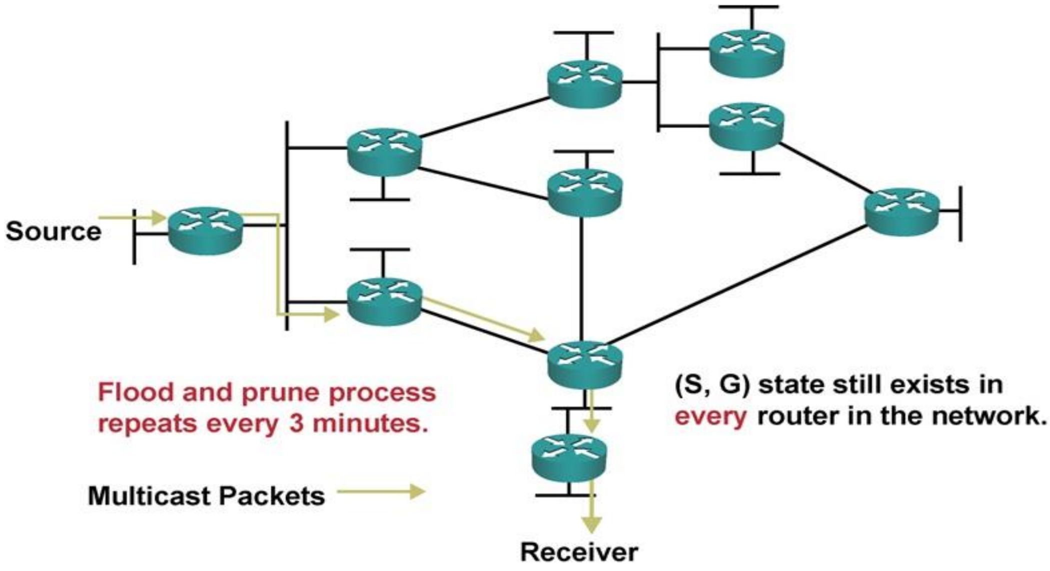


Figure 15 PIM-DM results after pruning. Copied from [35].

Other modes of PIM are source-specific multicast (SSM) and bidirectional, which are not widely used throughout a multicast domain.

4.3.5 Internet Group Management Protocol (IGMP)

The Internet Group Management Protocol (IGMP) is used on secure stack to snoop the multicast traffic only to those ports that need it. IGMP operates on a physical network that is a single Ethernet segment. It is used by multicast router to manage membership in IP multicast groups. It supports joining a multicast group, query membership and sending membership reports.

Multicast router will send queries to the host from time to time when it joins a multicast group. Report is sent only for the first process about multicast group membership related to neighboring router interface. This means that the host has the right to respond or not respond in receiving transmissions addressed to a specific multicast group. If there is no response or response time expires, then it is treated as if the host left the group and does not respond to the next query. Therefore, it will remove that host’s router interface from the group.

IGMP protocol is widely used in online streaming video and gaming. In IPTV it is used to connect to a TV channel and to change from one TV channel to another.

IGMP host reports

Host sends a report when it joins a multicast group and no report is sent when a process leaves a group.

For example:

IGMP report, TTL = 1
IGMP group addr = group address
dest IP addr = group address
src IP addr = IP address

IGMP router query

Router sends query at regular intervals to determine if known group members are still active. Hosts belonging to a multicast group must reply to query if wishing to join or stay in the group.

For example:

IGMP query, TTL = 1
IGMP group addr = 0.0.0.0
dest IP addr = 224.0.0.1
src IP addr = router IP address

IGMP message Types

Figure 16 shows an example of the components of IGMP messages.

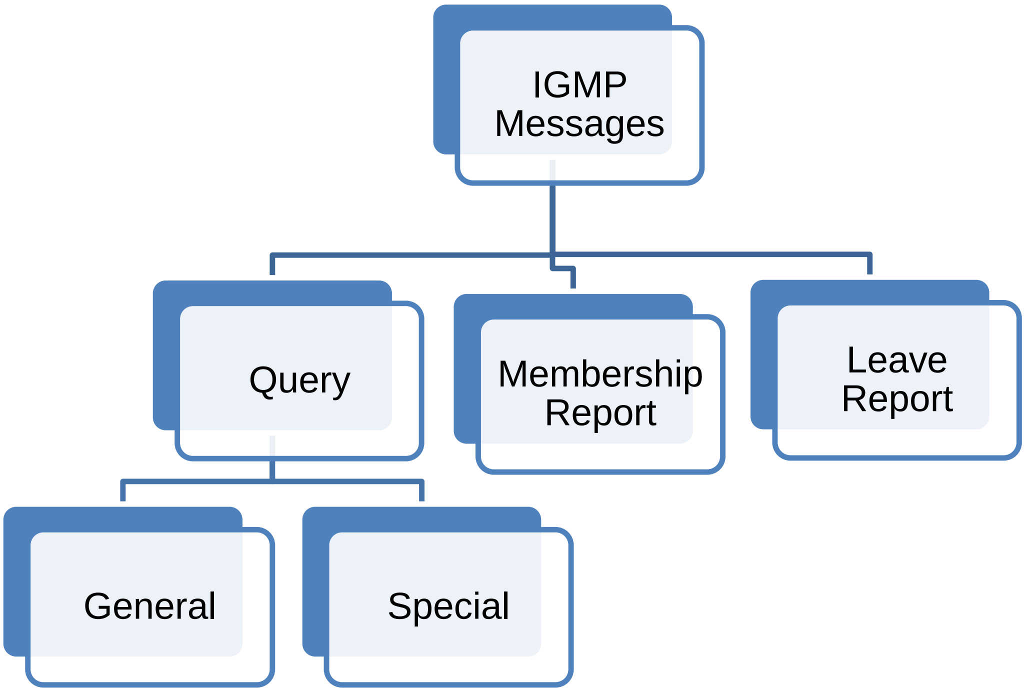


Figure 16 Components of IGMP messages.

The host may join a multicast group at their own will by sending a report message. There is no restriction as the host can choose to leave a group at any time by sending a leave report. Hosts can join as many group as they want to at a time. Membership query are used to discover which hosts are members of a particular multicast group.

4.3.6 Internet Control Message Protocol (ICMP)

ICM Presides in the IP layer and it is part of the TCP protocol used for error handling in the network layer. Since different types of errors can occur in the network, ICMP provide information messages concerning the routing of IP datagrams. It monitors, controls network traffic and reports the errors after network debugging to diagnose problems within the network layer.

For example; if the router cannot send the packet to a particular destination, ICMP communicate layer information between end hosts and routers sends an error message indicating that it cannot deliver the message to the end destination. Error reporting messages are used when an end host or router wants to report an error using

ICMP, it puts the information that it wants to send back to the source into an ICMP payload and delivers it to IP to be sent as a datagram.

5 Compression in IPTV System

The goal of this section is to provide an understanding of how compression works and types of compression techniques used. A good compression involves removing information from a file without someone hardly telling the difference between the compressed file and the original file. The compression techniques have made it possible to transmit multimedia signals via the internet.

Compression is the discarding of information by using different algorithms from a video or audio file in order to reduce overall file size, save storage capacity, speed file transfer, and lower cost for network bandwidth. The higher the quality of the source file, the higher the potential quality of the compressed files derived from it. Compressed file can be made look better than the source file by making the frame size smaller. The size and format of the source file have to have almost no relevance, all compression software automatically determines the specs of the source file and creates the transcoded file based on compression techniques applied. There are two compression algorithm techniques used.

5.1 Compression Algorithm

The compression algorithm consists of two different kinds of compression, lossless and lossy compression. They are both detailed below.

5.1.1 Lossless Compression

In lossless compression, data is compressed and the algorithm does not lose any single bit of data when the file is uncompressed. Algorithms stores and transmits data into smaller encoded files to restore the original information when uncompressed. Lossless data compression is ideal for situations where any loss of textual information cannot be tolerated. Example of this type of compression are ZIP compression and LZW compression. Formats such as GIF and PNG use lossless compression.

5.1.2 Lossy Compression

Lossy compression is a technique where a file is compressed or encoded to make it smaller by minimizing the bit rate of original data and irretrievably discarding away any unimportant data or unnecessary data. The data discarded from the original can never be restored. Therefore it is recommended to compress files as few times as possible to preserve as much data as possible.

This compression technique is used Digitally Sampled Analog Data (DSAD), where a loss of quality of data can be tolerated. DSAD consists of picture files, audio or video and graphics. Lossy compression is delivered in the form of Advanced Audio Coding (AAC), MP3,MP2, and many more. Digital audio is most often served in formats that use lossy compression to save bandwidth transmission cost and storage space.

5.2 Compression Techniques

The following compression types, spatial and temporal compression, are commonly used to compress audio and video.

5.2.1 Spatial Compression (Intra-frame)

Spatial compression is applied only to individual frames and mainly used to compress still images such as JPEG by removing spatial redundancy that exist in each individual frame. When a JPEG is created, color information of the image is reduced in a process called chroma sub-sampling and then the image is split in a session of 8 by 8 pixels called macro-blocks. Discrete cosine, transform and quantization is done to further reduce the file size.

Freeze frame is the pausing of a moving video which results in the view resting on one frame. The computer reviews video in slow motion and looks at each frame individually, and goes through a process of compare and contrast. It then reviews and take notes of the elements that are similar or not and on the new changes that are taking place.

Each frame does not have to be repeated making it much easier to reduce the overall file size for upload.

5.2.2 Temporal Compression (Inter-frame)

In temporal compression a series of frames or pixels are looked at and see what is different through the next ten frames. This compression technique looks at the data of the current frame and then goes to the succeeding frame. It does not keep track of every single pixel if the succeeding frames are identical or correlated, but only keeps track of the changes over time by reducing redundancy. Temporal compression is to motion compress and mainly uses lossy compression techniques such as temporal prediction to reduce the file size significantly without too much quality loss.

The key goal of compression is to get the highest possible image and video quality from the smallest possible bit rate. Image and video quality is a balance of five factors:

- Codec: Different codecs require different levels of efficiency settings for significantly similar quality.
- Larger frame sizes require faster bit rates.
- Faster frame rates require faster bit rates.
- More movement between frames in the master file from camera, transitions, and effects require higher bit rate needed to properly compress the frames for higher quality.
- Video compression yields the smallest file sizes when there is very little movement between the frames.
- Higher bit rates yield higher quality, but much larger file sizes.

The majority of television broadcast consists of 30 frames per seconds. YouTube for example broadcasts video at 25 frames per second. Discarding those 5 frames per second leads to a reduction of the overall size of video file. Many of live video streams such as Skype are 15 frames per second which is a reduction of half of the original frames.

In general, Intra-frame will always be better than inter-frame as long as the bandwidth is affordable. Inter-frame normally comes of importance in situations where there is not adequate bandwidth.

5.3 Audio Compression and Video Compression

Audio and video compression is a very important aspect of IPTV that indicates how audio and video streaming aspects e.g. in YouTube works. Video compression works by minimizing redundancy in the video data and offers a number of standards encodings as shown in Table 4 below.

Table 4 Image and Video Compression Standards. Copied from [36, 33].

Standard	Application	Bit Rate
JPEG	Still image compression	Variable
H.261	Video conferencing over ISDN	P x 64 kb/s
MPEG-1	Video on digital storage media (CD-ROM)	1.5Mb/s
MPEG-2	Digital Television	2-20 Mb/s
H.263	Video telephony over PSTN	33.6-? kb/s
MPEG-4	Object-based coding, synthetic content, interactivity	Variable
JPEG-2000	Improved still image compression	Variable
H.264/ MPEG-4 AVC	Improved video compression	10’s to 100’s kb/s

Audio compression is typically the compression of audio signals into formats such as MP3 or AAC, which reduces the size of the audio signal or file. Audio compression is used to remove useless audio data from its dynamic range that goes from absolute high to absolute low.

6 Results and Conclusions

While cable companies worked on moving their content on the web devices, telecommunication companies streamed television content over private network to a set-top box that is connected to a TV. IPTV is an important aspect in Television today dramatically changing the way people communicate and operate especially when its full potential is implemented and converged with mobile TV. It is a highway through which end users will see whatever video, whenever they want and on what device they prefer, be it on a television, a computer or mobile device.

With the emergence of internet, the TV experience has undergone a fundamental shift enabling cable operators and phone services to offer triple play of cable TV and high speed internet in digital voice services. With triple play services, consumers are now able to watch high quality programming over the internet using free or low priced over the top services from the like of Hullu, Netflix, YouTube and Apple iTunes.

To match up cable quality, IPTV providers rely on content delivery networks that store content across a geographically distributed network server rather than just one location. Distributing content close to the end customer, helps reduce bottleneck that result in startup delays or streaming video at a lower quality resolution. For instance when a request is made from Finland to view a video that originates from the United States, it can take a long time to cover that distance as compared to Sweden. This contributes to the importance of QoS to the end user satisfaction and experience.

To assure the highest quality viewing experience, IPTV providers must monitor their content delivery networks to isolate and quickly fix problems that might lead to subscriber defections. Phone companies such as Sonera can optimize streaming video performance, reduce costs and display an encoded video stream of IP packets via a STB. Some Telco network operators such as AT&T utilize DSL technologies to deliver IPTV and broadband services to users over their access network.

IPTV systems have the advantage over traditional cable TV services in comparison in that they are relatively independent on the operating point and offer quality of network solutions. The compression techniques of digital television data allows for more storage of content with less space. IPTV can be applied in different institutions and organizations such as education, finance, and medical fields to stream live operations with higher quality of content delivery, protection of content and control of video quality over a private network. It frees up bandwidth for distributers allowing them to deliver greater content to their customer.

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