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Media Processing in Video Conferences for Cooperating Over the Top and Operator Based Networks

School of Electrical Engineering

Thesis submitted in partial fulfillment of the requirement for the degree of Master of Science in Technology

Espoo 05.04.2012

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ABSTRACT OF THE MASTER'S THESIS

AUTHOR: Syed Safi Ali Shah

TITLE: Media Processing in Video Conferences for Cooperating Over the Top and Operator Based Networks

DATE: April 05, 2012 LANGUAGE: English NUMBER OF PAGES: 104

FACULTY: Electronics, Communications and Automation

DEPARTMENT: Communications and Networking

PROFESSORSHIP: Networking Laboratory CODE: S-38

SUPERVISOR: Professor Jörg Ott

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Telecom operators have dominated the communication industry for a long time by providing services with guaranteed quality of service. Such services are provided by the operator at the cost of maintaining a high grade network. With the introduction of broadband and internet, many over the top (OTT) services have emerged. These services use the underlying operator networks as a mere bit pipe while all service intelligence resides in the application running on the client device. Introduction of OTT services has seen a good response from general users who are no longer bound to services provided by the network operator. This in turn has caused operators and telecom companies to loose the ownership of their customers.

This thesis takes media processing in video conferencing as a case study to compare the two competing domains of operator networks and OTT networks. Both domains offer video conferencing to end users, but they follow different architectures. The study shows that OTT services can perform much better if they utilize support of the underlying network. This will also bring the user base back to the network operator. The proposal is to turn the competition into cooperation between both parties.

Assessments are done from both technical as well as business perspectives to assert that such cooperative agreements are possible and should be experimented in real life.

KEYWORDS: Video Conference, Over The Top, Media Processing

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ACKNOWLEDGEMENTS

I would like to express my gratitude to Allah and then to all the people who supported me during my thesis work: my wife, my parents, my friends at Aalto University, and my colleagues at Ericsson.

A special thanks to Professor Jörg Ott for his useful insights and Kaisa Kettunen for her constant guidance throughout my thesis work.

At last I would like to thank Kari Pekka Perttula, who pointed me in the right direction and helped define the research topic.

Espoo, April 05, 2012.

Syed Safi Ali Shah

ABBREVIATIONS

3G	Third Generation
ADPCM	Adaptive Differential Pulse Code Modulation
ADSL	Asymmetric Digital Subscriber Line
API	Application Programming Interface
AS	Application Server
AVC	Advanced Video Codec
B2BUA	Back to Back User Agent
BFCP	Binary Floor Control Protocol
CPU	Central Processing Unit
CSCF	Call Session Control Function
CSRC	Contributing Source
DHT	Distributed Hash Table
DNS	Domain Name System
DSP	Digital Signal Processing
DVD	Digital Versatile Disk
FQDN	Fully Qualified Domain Name
GSM	Global System for Mobile Communication
HD	High Definition
HLR	Home Location Register
HSPA	High Speed Packet Access
HSS	Home Subscriber Server
HTTP	Hypertext Transfer Protocol
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol

ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ITU	International Telecommunication Union
ITU-T	International Telecommunication Union - Telecommunication
LAN	Local Area Network
LTE	Long Term Evolution
MC	Multipoint Controller
MCU	Multipoint Control Unit
MP	Multipoint Processor
MPEG-2	Moving Pictures Experts Group - 2nd standard
MPEG-4	Moving Pictures Experts Group - 4th standard
MRFP	Media Resource Function Processor
NAT	Network Address Translator
NNI	Network Network Interface
OTT	Over The Top
P-CSCF	Proxy Call Session Control Function
P2P	Peer to Peer
P2PSIP	Peer to Peer Session Initiation Protocol
PSNR	Peak Signal to Noise Ratio
QoS	Quality of Service
RTCP	RTP Control Protocol
RTP	Realtime Transport Protocol
RTT	Round Trip Time
SAP	Session Announcement Protocol
SBC	Session Border Controller
SCCP	Simple Conference Control Protocol
SD	Standard Definition

SDES	Source Description
SDP	Session Description Protocol
SIM	Subscriber Identity Module
SIP	Session Initiation Protocol
SOAP	Simple Object Access Protocol
SRTP	Secure Realtime Transport Protocol
Telco	Telecommunications Company
TCS	Terminal Capability Set
TLS	Transport Layer Security
TV	Television
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
UNI	User Network Interface
URI	Universal Resource Identifier
VBR	Variable Bitrate
VOIP	Voice over Internet Protocol
WLAN	Wireless Local Area Network
WiFi	Wireless Fidelity

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TERMINOLOGY

MCU

Multipoint control unit. Although originally defined in the H.323 standard, this document uses this term to generally refer to an element in a conference or multi-party call, which is vested with the responsibility of maintaining signaling dialogues with multiple participants in a conference. An MCU may also provide support for media processing, such as mixing multiple streams into one stream or transcoding streams by modifying their media attributes.

OTT

Over the top (OTT) services are services which use the underlying carrier network as a bit pipe, while placing all the intelligence and decision logic on the end client devices. This type of services are seen as a risk to the Internet Service Providers (ISPs) and telecom operators since they use the network freely and openly without much respect for the operator boundaries. Examples of such OTT service providers are Skype [37], GoogleTalk [69] etc.

Operator Network

A network which is commissioned and administered by a well-established telecom operator or an ISP. The services are hosted on servers residing inside the network and are made available to the users registered in this network against a certain charge. Operator networks are also often referred as “carrier-networks” from the OTT service’s point of view.

Walled-garden

A network divided into separate and distinct operator domains through extensive use of firewalls and Network Address Translators (NATs). The operator determines which users get access to which of the services and applications. This concept goes opposite to the open internet architecture.

1 INTRODUCTION

Communication networks have seen a very fast and large scale development in the last decade. End users have upgraded their connections from a few kilobits per second to many megabits per second of bandwidth. This increase in speed and capacity has re-shaped the way people communicate over long distances. We have seen a shift from simple text based communication to voice based applications [1]. The latest addition to this ever expanding domain of communication services is video based communication. Today people do not just want to hear the person on the other side of the network, but they also want to see who they are communicating with. This makes communications a much richer interactive process. These evolving networks have enabled another dimension of communication; video conferencing. The idea that many people situated at geographically far off locations can simultaneously see and talk to each other has now become a reality.

At the same time, organizations and individuals would like to have high definition (HD) video support in video conferences, since the improved video quality can add great value and much broader applicability to video conferencing [2]. A few of the use cases where HD video conferencing is expected to be helpful include medical procedures which could be carried out on patients by different medical experts located at different geographic locations. Also, employees of an organization no longer need to travel all the way to different offices to meet and work with other people. Instead, they can simply work together in teams over HD video conferences thus reducing the need for physical travel. For collaborative research, participants can simply draw something on a piece of paper and show it to the person on the other side. That is to say, the experience becomes much more realistic and comfortable, and thus it promises to save a lot of cost in terms of money and time that is otherwise spent on traveling.

However just like voice, video communication has also seen the tussle between the two main competing players in the industry [4]: Over the top (OTT) service providers and the network operators. Both market players tend to offer similar services to the end user, while the underlying technicalities of how these services are delivered are very different. Network operators tend to host services inside their networks and offer them as Value Added Services (VAS) to end users. However OTT services are typically installed on the end users' terminals while they use the operator's network as a bit pipe for transferring data. End users simply want to be able to access their favorite service from any place and any terminal they have, irrespective of the underlying technology or network dynamics being used to deliver that service.

The competition between the OTT industry and network operators has continued since the early days of Voice over Internet Protocol (VoIP) or IP telephony. This competition with the OTT service providers lead to a general fear in the network operator domain about prospects of a slowly decaying business. The rationale behind the competition is straight forward. Operators tend to host the services within their networks [5] and thus boast of the concept of an "intelligent network". The terminals in this case need to take minimum amount of load, merely accessing the service from the network, while the network with its reliable and powerful servers does everything for the end user. To make the network "intelligent" and capable enough, operators invest heavily on their infrastructure. This in turn means, that the end user is charged a considerable amount of fee for the services he or she accesses. The OTT industry, on the other hand simply uses the underlying network as a bit pipe to route data packets through to end terminals, while all the logic is hosted on the end client devices. The OTT services are in some cases unreliable but appear to be generally more attractive to the consumer due to their minimal cost. The operators have for many years now, blamed the OTT industry for using their networks without compensating the operator for the services that are being offered through the use of its network. Since the OTT applications only use the network to transmit bits and bytes of information, the operator can only charge them for the use of bandwidth and not on the basis of the provided service (data, voice, video etc).

Also, increasing competition from alternate access providers is forcing operators to offer flat rate data subscription plans. Flat fee Internet access causes increase in traffic volumes flowing through the network, a large part of which is P2P or OTT traffic [6]. This increased demand and traffic on the network forces the operator to invest in additional network capacity. This trend may lead to the decoupling of traffic and revenue [7]. Therefore being a mere bit pipe is generally not seen as a profitable business by operators as the revenue generated may not be enough to cover the costs of carrying the OTT traffic [8]. Some circles of telecom operators have gone the distance of trying to block or firewall the OTT applications from accessing their network.

A network operator thus has to consider whether investing in expensive hardware, software and maintaining a reliable service is a profitable business scenario anymore. Or should the industry just accept that end devices today are capable of handling their own loads and requirements, and thus network should in fact be just a dumb bit pipe? This debate seems to put the OTT industry at an advantage when considering IP telephony as the service of contention. With the introduction of broadband and 3G networks, the end users got ample bandwidth at their disposal to allow voice traffic to flow without major hiccups over the best effort IP infrastructure, even in the absence of any particular QoS guarantees in the network. Also as the client devices (including mobile handsets, desktops and laptops) continuously evolved in terms of processing power and memory, they could fulfill the needs of audio processing themselves. Thus the OTT applications got more and more self reliant and required less or no support from the network hosted intelligence.

However, video conferencing, which is going to be our focus in the rest of the thesis, differs from VoIP scenario in two major ways. Firstly, it requires a considerable amount of video processing which is more complex and computationally intensive process as compared to audio processing. Secondly, video packets are bulkier and need a fair amount of network bandwidth to make sure they are delivered on time. It is primarily for these reasons that despite the considerable amount of time since its introduction, video conferencing and network collaborative environments still suffer from lack of quality [9]. This is especially common in scenarios where conferencing applications run over third party networks. The underlying network in such cases, does not provide much assistance such as QoS (Quality of Service) guarantees to the conference application. Participants on different sides of the video conference can see and hear each other, but due to the lack of video quality the experience remains unpleasant and the participants do not get the sense of really sitting in front of each other in the same room. As we discuss further on in the thesis that the network may be able to lend a helping hand in such scenarios by fulfilling the requirements of the OTT applications.

Thus the question remains: is there still a way for the network operators and OTT industry to cooperate with each other in order to bring better services to end consumer and to turn this operator fear into profit?

1.1 PURPOSE OF THE THESIS

Specific to video conferencing, this thesis aims to evaluate the requirement of high-end multipoint control units inside the network which are capable of handling video mixing and video transcoding. The study aims to understand whether the OTT services today have advanced to the point where they can handle everything within the end systems without any support from the network. If this is true then network operators can avoid heavy investments that go into provisioning these services in their networks. On the other hand, if support for media processing is still needed within the network, we investigate how the OTT service providers can cooperate with the operator based networks to bring better services to the end user. We also try to identify the benefits for the OTT industry and network operators in case of cooperative agreements between both of the market players.

1.2 METHODOLOGY

We start by reviewing the current literature available on the topic of video conferencing. The literature review identifies some key research questions, which we further investigate by taking measurements and well defined statistics. These statistics clearly identify some room for improvement in the system. We then highlight different possible solutions during brain storming sessions and collaborative discussions. At the end, we formulate a concrete proposal. The applicability of the proposal is evaluated in light of both technical and business related demands. In the end, we identify some areas for future research.

1.3 THESIS STRUCTURE

The thesis is structured as follows. We begin in chapter 2 by introducing the basic concepts of media conferencing. In chapter 3, we point out the various topologies used in media conferencing architectures and present their mutual comparisons. In chapter 4, we continue to study the current requirements posed at the Multipoint Control Unit (MCU) for multiparty conference calls involving video. Then we see how OTT services tend to accomplish multi-party control tasks such as media mixing and transcoding. We then look into the dimension of cooperation between operator networks and OTT service providers in chapter 5, and also discuss what benefits it holds for both parties. Chapter 6 then briefly points out some security considerations in case of cooperative agreements between OTT service providers and network operators. We then conclude in chapter 7 by summing up the findings of the research and identifying some future research areas.

2 VIDEO CONFERENCING CONCEPTS

To ensure that multiple participants in a video conference are connected and are able to talk and see each other simultaneously in realtime, while maintaining control of shared resources among the different participants, the communication application has to go through various steps such as conference and media setup, conference policy manipulation, media control, and floor control [10] as shown in Figure 2.1.

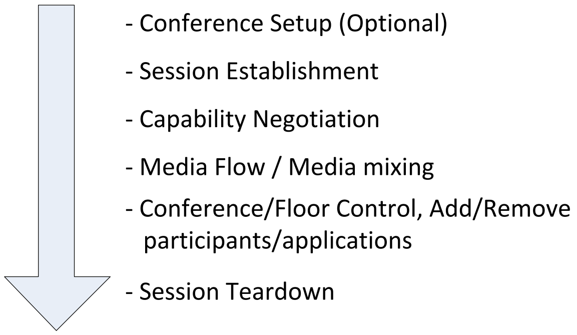
- Conference Setup (Optional)
- Session Establishment
- Capability Negotiation
- Media Flow / Media mixing
- Conference/Floor Control, Add/Remove participants/applications
- Session Teardown

Figure 2.1: Stages of a conference session

Video conference scenarios are generally complex and can often pose a lot of challenges for the network, which inherently does not guarantee a fixed quality of service. The situation becomes tougher as the number of participants in a conference increases and when they belong to different networks. To remedy these challenges, a collection of intelligent protocols, network topologies, media compression schemes, and specialized network elements have been proposed.

There are two main and often competing standards that are in active existence and have seen a wide deployment globally. These are:

1. Internet Engineering Task Force (IETF) standards track [13]
2. International Telecommunications Union (ITU) standards track [14]

In the following text, we give a general overview of the various methods and services that set the foundation of video conferencing systems. Some of the available protocols categorized according to their functions are summarized in

Figure 2.2. We will highlight both of the above mentioned standards wherever applicable.

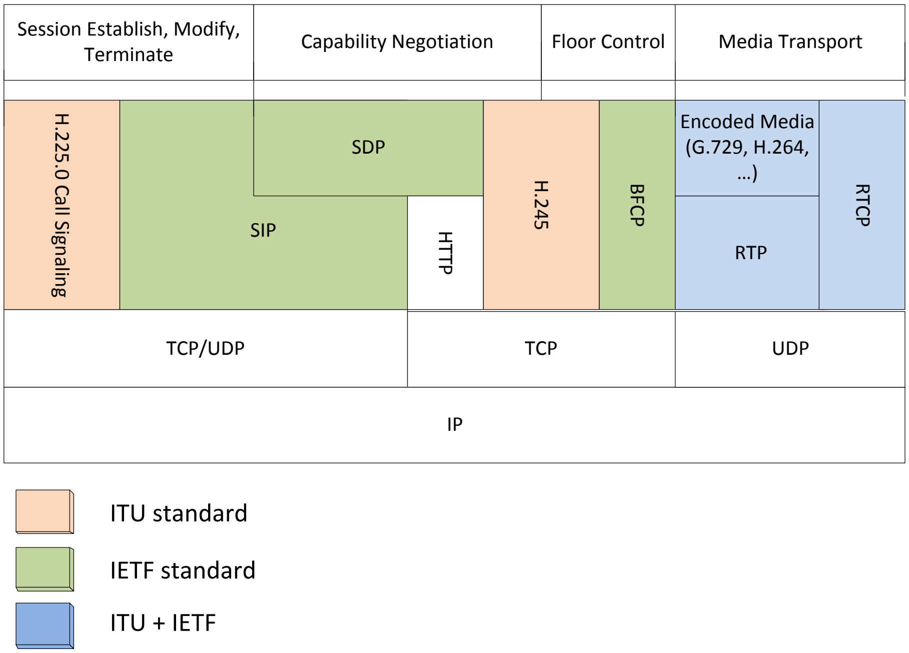

Figure 2.2: Protocols used during conference sessions, categorized according to their functionality

2.1 SESSION ESTABLISHMENT/TEARDOWN

Before any kind of media can start to flow between end parties in a conference, connections must be established between them. In some conference topologies, participants must connect to servers which provide functions such as media conversion, mixing and other such applications. This phase, where the devices are connected to each other, is referred to as session establishment. At the end of the conversation, when participants wish to leave the conference, they must close these connections. This is the session teardown phase.

IETF proposes the use of Session Initiation Protocol (SIP) [75] for Session Establishment, Modification and Teardown. SIP is a lightweight internet friendly protocol. A basic session setup with SIP is depicted in Figure 2.3. After this initial session establishment, media can start to flow between the end points.

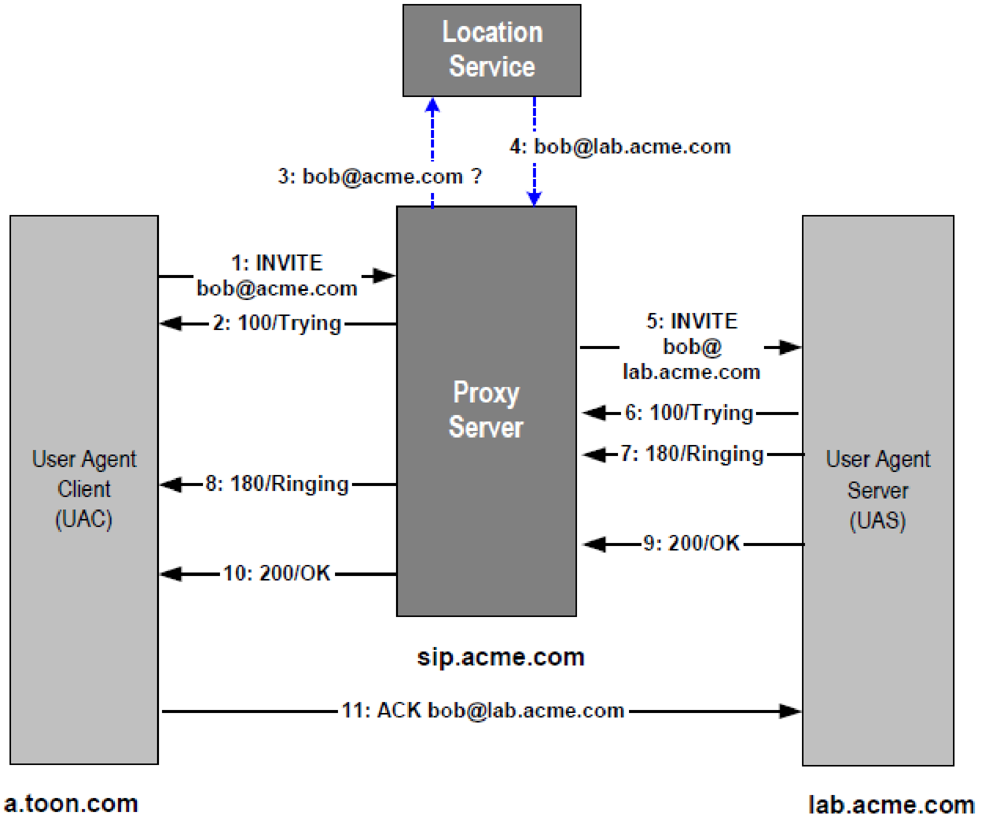

Figure 2.3: Session Initiation Protocol (SIP) call setup procedure [16]

ITU proposes the H.323 protocol suite [17]. This standard contains a family of protocols, each specialized for specific functions to enable realtime communications in todays networks, such as the internet. The H.323 inherits some of its mechanics from the Q.931 protocol [18] used for ISDN signaling. It tries to follow a more traditional circuit switched approach even though it is deployed on packet based networks. A simple call setup scenario using H.225.0 call signaling protocol [19], part of the H.323 standard, is depicted in Figure 2.4.

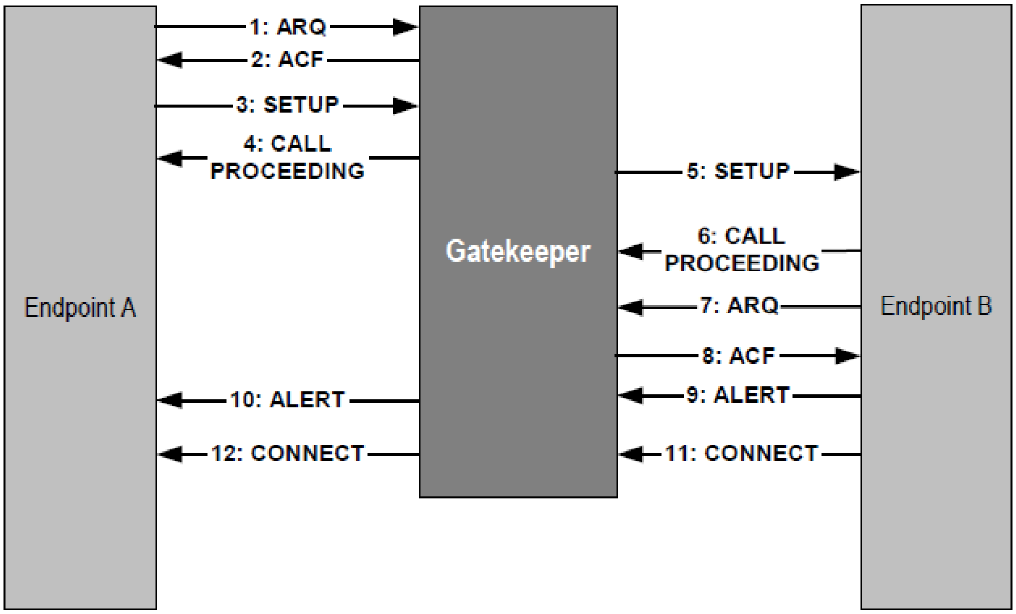

Figure 2.4: H.225.0 call signaling protocol call setup procedure [16]

For a more detailed comparison of both protocol suites, refer to [3].

2.2 CAPABILITY NEGOTIATION

The signaling protocols are coupled with capability negotiation methods, where all members of a realtime session exchange their capability sets during the connection establishment phase. This is to make sure that end points are compatible with each other, and that they can exchange media in formats understandable to each other. SIP messages can include a body containing Session Description Protocol (SDP) [22] based session descriptions, which defines the media capabilities of an end point. The process of negotiating compatible media formats and attributes is referred to as the SDP offer/answer model. H.323 uses the H.245 control protocol [23] which enables the exchange of Terminal Capability Sets (TCS) between end points to allow them to choose matching media formats for a session.

2.3 MEDIA TRANSPORT

Once the initial handshake is complete and matching media capability sets are exchanged, media streams can start to flow between the end points. A specialized application layer protocol named Realtime Transport Protocol (RTP) [24] has been proposed by IETF and is by far the most widely deployed protocol for transport of media streams in IP based networks. Both SIP architecture and the H.323 protocol suite recommend the use of RTP for media exchange. RTP by nature is independent of the underlying transport layer protocol, but it is generally deployed over UDP in order to maintain steady throughput by avoiding unnecessary re-transmissions of lost or delayed packets. Through the use of sequence number and timestamps, RTP maintains orderly and synchronized playback of realtime data. RTP is often (but not always) accompanied by the RTP Control Protocol (RTCP) [24]. The purpose of this control protocol is to exchange useful statistics about realtime packets between the communicating parties, such as the number of packets lost, and thus to estimate the condition of the link and the measures needed to improve the quality of the realtime session.

2.4 INTER STREAM SYNCHRONIZATION

The use of RTP for media transport means that in case of video conferences, audio and video packets are transmitted as separate media streams. The motivation for de-coupling audio and video in RTP are given in [24]. These media streams flow independent of each other through the entire network, as shown in Figure 2.5, before reaching the end destination, where they are played back to the end user. Because the streams flow independently in the network, audio and video packets may arrive with differing delays at the receiver. In addition to this, the video conferencing applications at the sender, receiver or an intermediate server might have separate processing pipelines for audio and video packets. Thus the audio and video streams need to be re-synchronized at the receiver before they are played back to the user. This synchronization can be achieved by using the timing information inside the RTCP sender reports [24][15]. RTCP packets from a specific sender can be used to map the timestamps contained inside the RTP headers of the independent media streams to a common sender based reference timestamp.

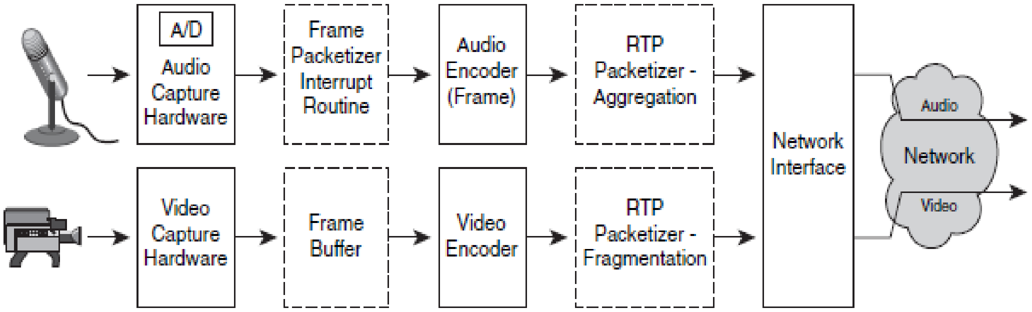

Figure 2.5: Audio and video are recorded and transmitted as separate streams [15]

2.5 CONFERENCING SUPPORT

When two nodes are connected in a call with each other, they have well defined point-to-point associations both in terms of signaling and media flow. However when a third node joins in the call, making it a multi-party call, things become a little complicated. Usually some node should take the additional responsibility of maintaining the signaling associations between all the nodes and to make sure that they all receive media streams from each other. As the number of participants in a multi-party call keeps increasing, so does the requirement for managing multiple connections and media streams.

Multi-party calls or conferences are supported by both standards, i.e SIP and H.323. To fulfill the above mentioned requirements posed by conference sessions, H.323 standard defines a central network element called Multipoint Control Unit (MCU). It consists of two distinct elements: Multipoint Controller

(MC) which is responsible for maintaining signaling associations among different conference participants and optionally Multipoint Processors (MP) which offers media processing support such as mixing and transcoding of the media streams from different participants. Hence in a conference session, usually all participants establish connections with the MCU which then makes sure that all participants receive the mixed stream from each other on their desired addresses.

SIP does not define a logically distinct element such as an MCU, and any user agent can act as a focus for a multi-party session [40]. The focus acts as the center of the conference and its responsibility is to maintain signaling relationships with all conference participants while maintaining full control over the conference. In general, the focus can be any user agent with B2BUA (Back to Back User Agent) functionality. The media processing requirements in SIP conferences are handled by mixers, which are logically disjoint elements and can be controlled by the focus using third party protocols. Physically, a mixer may or may not be a part of the focus. The focus uses third party call control mechanisms to instruct all conference members to direct their media streams to the mixer.

Both standards can support different architectures and topologies in which the conferencing nodes are connected with each other and how the responsibilities are shared among them. Details on conference architectures are discussed in Section 3.

2.6 CONFERENCE FLOOR CONTROL

Another aspect particular to multi-party calls is that of floor control. Floor control basically implies controlling access to shared resources in a conference. For example the mixer can be instructed to choose only part of the incoming streams to be mixed together and sent to all participants. In other words, a floor control protocol can be used to administer a conference and to allow only certain members of the conference to actively speak and be heard by all members, while the passive members only watch/listen to the active members. Methods which allow members of a conference to request floor and then allow a controlling user/server to grant the floor to a member based on certain policy need to be in place. Such conference control and conference management is defined as part of the H.245 protocol within the H.323 protocol suite. SIP, however, does not provide a standardized method of implementing such functionality, but instead leaves room for various protocols to be plugged into serve this purpose. Many protocols have been proposed, such as the Simple conference control protocol (SCCP) [20] which mainly deals with tightly coupled conferences and assumes a reliable transport infrastructure or using Simple Object Access Protocol (SOAP) [12] for implementing floor control. The Binary Floor Control Protocol (BFCP) [11] can also be used for this purpose in a conference session.

2.7 MEDIA CODING

In the telecommunication world, extra bandwidth means extra cost. It is infeasible to transport uncompressed audio/video streams over networks which are already low on bandwidth and are being shared by many users simultaneously. For this reason media streaming applications generally deploy intelligent and effective audio and video coding algorithms, which give maximum quality to the end user at minimal bit rates.

There are many audio and video compression standards, referred to as “codecs”. One can choose which codec to use depending upon the application, such as digital TV broadcast, DVD movies, media streaming over the internet etc. Most of these codecs employ lossy coding schemes. This means that during compression, some information from the audio or video packets has to be dropped out thus causing a degradation of user perceived quality at the receiving end.

In video conference systems, multiple audio and video streams have to be sent and received in realtime among many participants. The available bandwidth on the network links thus becomes a severe bottleneck. An intelligent choice of a media codec can greatly help tackle the problem of scarce bandwidth by compressing media into smaller size while maintaining good quality.

ITU has specified a number of audio and video coding standards that can be used in video conference applications. These include:

Video

- H.261, originally specified in 1988 for video transmission over ISDN lines.
- H.263, specified in 1995, as a replacement of H.261 for low bitrate applications
- H.264, specified in 2003. Has a lot of improvement in compression ratio over its predecessor standard.

Audio

- G.711, 64 Kbps, Comes in two flavors: A-law and mu-law
- G.722, 48/56/64 Kbps ADPCM 7Khz audio bandwidth
- G.728, 16 Kbps
- G.723.1, 5.3/6.3 Kbps, 30ms frame size
- G.729, 8 Kbps, 10ms frame size

2.7.1 H.264 VIDEO CODEC OVERVIEW

The ITU-T specified H.264 is today’s video codec of choice for nearly all applications. It is also referred to as the Advanced Video Coding (AVC) or the MPEG-4 part 10. The main motivation behind developing this coding standard was better coding efficiency as compared to its predecessor codecs. Better and more complex compression algorithms are employed which guarantee smaller bitrates without the compromise on video quality. This makes H.264 one of the forerunners in network friendly codecs, using minimal bandwidth for better quality videos which are used for both conversational (video telephony, video conferencing) as well as non-conversational (Video on demand, TV broadcasts, media streaming) applications. However, this efficiency in terms of bitrate comes at a tradeoff for processing demands. The H.264 encoding and decoding algorithms require fairly complex prediction and transforms thus making the process computationally much more intense than previous coding standards.

To understand the bandwidth and computational characteristics of H.264 video codec, studies have been conducted with different encoders and decoders. Alvarez et al. [27] have performed detailed tests of different sample video sequences using both MPEG-2 video codec (also known as the H.262), and the H-264 encoding. The results are quite elaborate but we summarize them in the tables below. The referred decoder are the following:

- H.264 = FFMPEG highly optimized decoder
- MPEG4 = XviD MPEG-4 decoder
- MPEG2 = libmpeg2 MPEG-2 decoder

A. Network Bandwidth consumption

H.264 is designed to be a bandwidth conserving codec. It aims at providing the same visual experience to the audience that its predecessor codecs would deliver at much higher bitrates. In general, the H.264 is said to provide as much as 50% of bitrate saving in video streaming applications [30]. To reduce the bandwidth demands, it uses a VBR (Variable bit rate) profile, which allows less bits per frame to be used when there is less motion in the video.

The bandwidth depends on a lot of other parameters apart from the resolution of the video. The below mentioned bitrates are for the same sample videos encoded with different codecs to give almost same PSNR in the encoded video stream.

That is to say that the bitrates differ depending on the codec used, when the encoded videos have similar visual quality.

Video Resolution	H.264 (kbps)	MPEG4 (kbps)	MPEG2 (kbps)
729x576	2033	2236	6318
1280x720	3471	4050	10010
1920x1088	6724	8064	17723

Table 2.1: Bit rates of video encoded with different codec (in Kbps) [27]

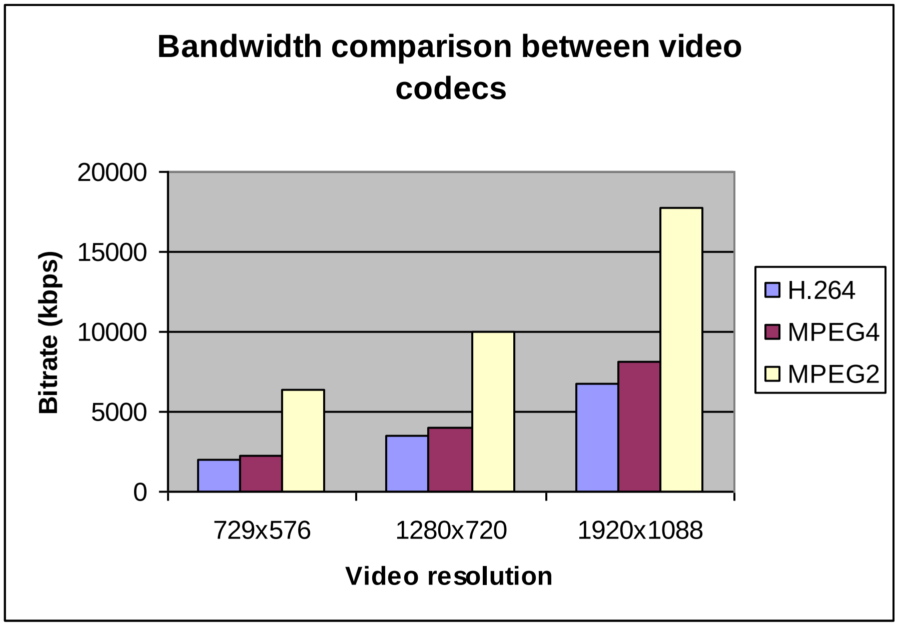

Figure 2.6: Comparison of bit rates between different video codecs [27]

B. CPU cycles utilization

Similar to the case above, the CPU utilization is plotted for decoding videos with different codecs having similar PSNR i.e. visual quality. The data presented in Table 2.2 serves as a good illustration of the fact that H.264 is much more processor intensive as compared to its predecessor codecs. It is worth while to note that decoding isn't the only task requiring considerable amount of CPU cycles, but as we will see later in Section 4.3.1, encoding is generally tougher than decoding in terms of CPU requirements. Thus encoding must also be considered when dimensioning systems.

Resolution	H.264	MPEG4	MPEG2
729x576	48	34	7,3
1280x720	99	73	14
1920x1088	213	165	31,8

Table 2.2: CPU cycles (x10^6) per second per frame for decoding video streams [27]

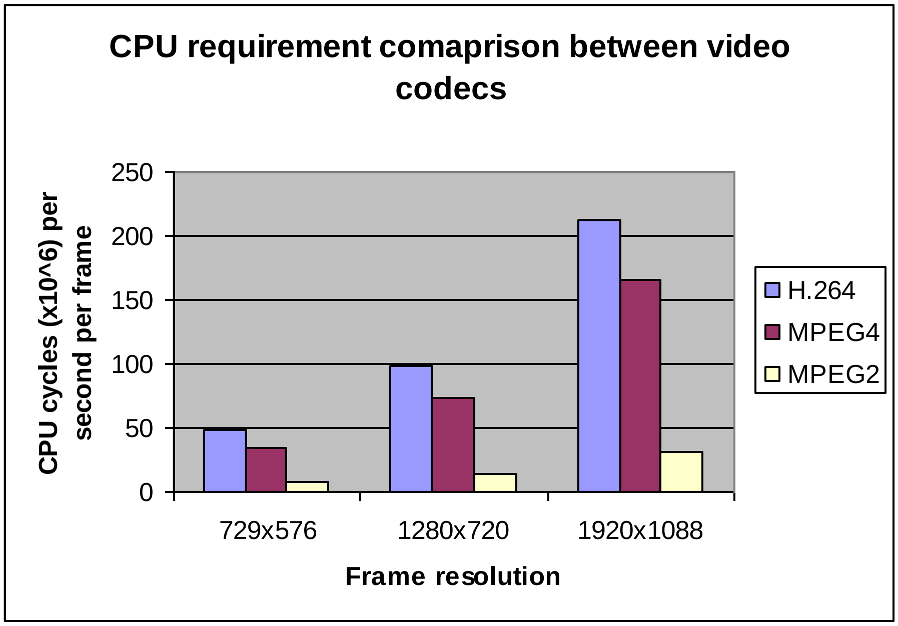

Figure 2.7: CPU utilization during decoding process [27]

C. Conclusions

An analysis of the results shows that H.264 on average offers 64% bandwidth saving as compared to the MPEG-2 codec. This high compression is, however, achieved at the cost of complex algorithms, which take more CPU cycles and instructions to evaluate. On average, the H.264 codec takes about 7 times more CPU cycles per frame as compared to MPEG-2 for the same video sequence.

There are two vital resources at hand which we try to conserve in internet based streaming or conferencing; bandwidth and the computational power. Trying to compress the video stream to make it more bandwidth friendly will require the use of more complex algorithms, which in turn means more processing cycles. On the other hand, saving on the processor power thus evading complex compression algorithms will adversely affect the bandwidth utilization.

2.8 SUMMARY:

To summarize, we see that video conferences involve elaborate procedures and must follow certain protocols in order to make sure that all participants can see and hear each other. Generally a conference proceeds through a series of stages such as the conference setup (optional), session establishment, capability negotiation, media transfer, floor control (optional) and finally session teardown. There are two prominent standardization bodies namely ITU and IETF who have been actively involved in proposing protocols and standards governing the realtime communications over packet switched networks. These standards are also applicable to multi party calls or conference sessions.

Another aspect that plays an important role in video conferences is the choice of video codecs. We highlight some of the key properties of the currently well known and widely used video codec H.264 and compare it to its predecessor codecs. The findings suggest that H.264 gives much better bandwidth efficiency compared to its predecessor codec but at the cost of increased processing power for decoding and encoding.

Having covered these key concepts that are fundamental to all conferences over packet switched networks, we will delve deeper into the various conference architectures and topologies in the next chapter.

3 MEDIA CONFERENCING ARCHITECTURES

Media conferencing by nature is an elaborate and complex communication scenario. As the number of participant nodes in a session increases, there are more ways in which they can be arranged in the network [41]. These different possibilities are summarized in the chart below.

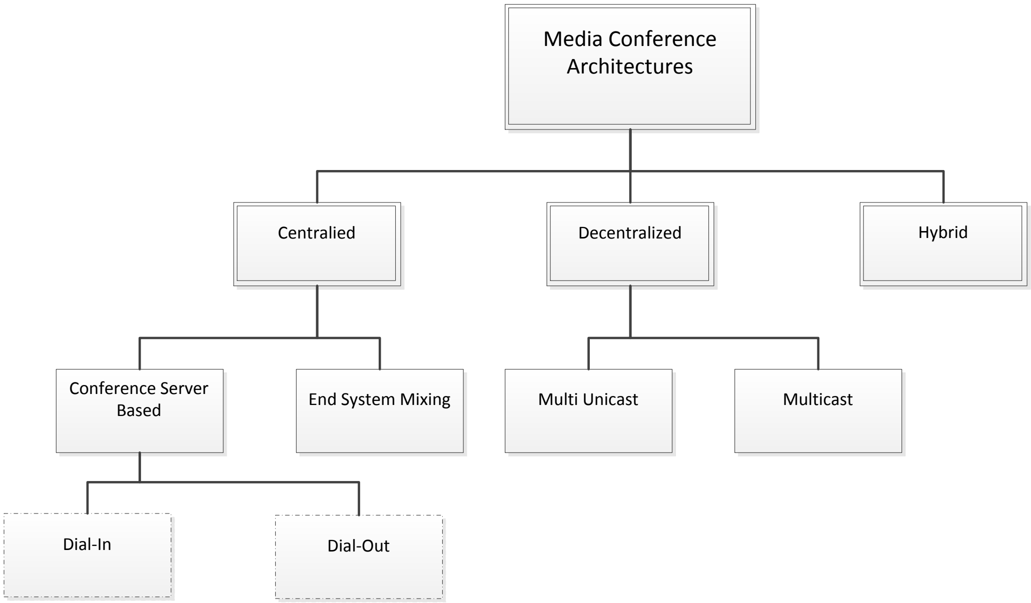

Figure 3.1: Classification of media conference architectures

In this chapter we will take a closer look at each one of these conference architectures. Although the following architectures can be implemented using any signaling protocol of choice, we will primarily discuss examples from SIP wherever applicable [41]. Similar examples for other signaling protocols, such as H.323, can be found in literature.

3.1 CENTRALIZED

In the centralized model, the conference participants are tied together through a central node. There are two variants of this as described below.

3.1.1 CENTRAL CONFERENCE SERVER

In the central conference server configuration, all the participants connect to a server individually using the signaling protocol of choice. This creates a star topology, where the center of the star is a powerful conference server (also referred to as an MCU, or a Focus). The server maintains the conference state and may also host the logic for conference management and floor control during the conference. The end systems just send their individual media streams to this central server, which processes them, mixes them together and sends the mixed stream out to each and every participant.

For each end system, the conference appears to be a point to point call where each participant sends one stream and receives one stream in response. The central server takes care of most of the load and plays an active role in the scalability of the conference in terms of number of participants supported.

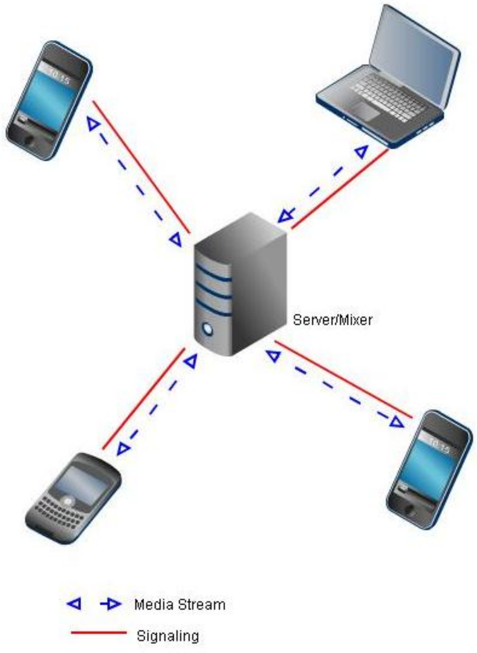

Figure 3.2: Central conference server based conference architecture

The central conference server architecture can still be implemented in two different ways depending on how the conference is set-up and how participants are added to it.

A) Dial-in Conferences

In the Dial-In type, the URI or address of the conference is published, and all users can establish a connection to the server individually. Since there can be simultaneously any number of conferences hosted by the conference server, it is important for the central server to know which users belong to which conference. This can be done for example by keeping the address specific to each conference. A conference can have an ID number and that can be reflected in the URI (for example conf-id@service-provider.com). More users can be added to the conference later by providing them the address of the conference server. This can be achieved for example by sending a SIP REFER message containing the URI of the conference to the user.

B) Dial-out Conferences

In Dial-out type of conference, the central server or the focus of the conference initiates connections to each participant asking them to join the conference. In a practical scenario, one user would first establish a connection with the server and then provide it a list of rest of the participants which should be invited to the conference. In SIP, this can be achieved by including the recipient-list [35] in the body of the first INVITE message sent to the server. The server on receipt of the INVITE can then process the recipient-list and in turn send INVITE messages to all of the URIs listed in the recipient list. More users can be added later on by sending a REFER message to the central server, which in turn can send INVITE to the requested user.

3.1.2 END-SYSTEM MIXING

End system mixing is another type of centralized conference architecture, even though it does not involve any specialized central server to manage the conference. In this architecture, one of the participant nodes is nominated to be the conference focus. All other users have single signaling relationship with this central focus. They unicast their media streams to the focus, which in turn acts as a media mixer and transcoder. After processing all the streams, it sends them out to the participants.

The nomination of the central media mixing/transcoding node depends on the processing power and the network bandwidth of the node. As a general rule and as seen in some of the popular peer to peer conferencing systems such as Skype [37], the peer with the best resources both in terms of CPU and network connectivity are vested with the responsibility of being the central media processor of the conference [46].

It is possible to keep the signaling and the media streams disjoint such that the system handling the signaling associations with all the conference member nodes does not necessarily have to process the media streams. This can be done through third party call control functionality in SIP, where the central node maintaining the signaling relationships instructs all participants to send and receive media from another node. For example in a scenario where a new participant with better capabilities joins the conference, the media transcoding and mixing responsibilities can dynamically be handed over to this new participant. One way in which this can be accomplished is by sending new SDP session description containing the address of the new participant as the media source and sink. The new session description can be propagated through the conference participants by sending a SIP REINVITE to modify the session parameters.

If the participant acting as the focus or the mixer for the conference leaves the conference, the whole conference ends.

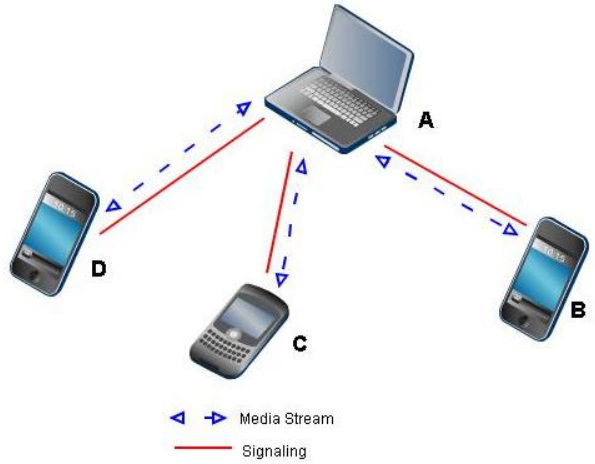

Figure 3.3: End system mixing model for media conferences

3.2 DECENTRALIZED

In decentralized architecture of media conferences, there is no central authority vested with the responsibility of keeping the participants tied together. All participants are considered equal and the conferencing tasks, such as signaling, media mixing etc are shared amongst them.

3.2.1 MESH NETWORK (MULTI-UNICAST)

In a mesh network, each and every participant maintains a connection with every other member. Every user unicasts his stream to all other members in the conference and receives the streams from all other active members. This is the reason why this scheme is also referred to as multi-unicasting. Every participant performs the media stream mixing individually for itself and it does not forward the mixed stream to anyone. Compared to the end system mixing model, this reduces the processing load (of encoding mixed media stream) on the end systems, but the bandwidth demands remain still quite high [36]. For N participant conference, where all users are actively sending media, an end point will need to send N-1 streams and receive the equal amount as well.

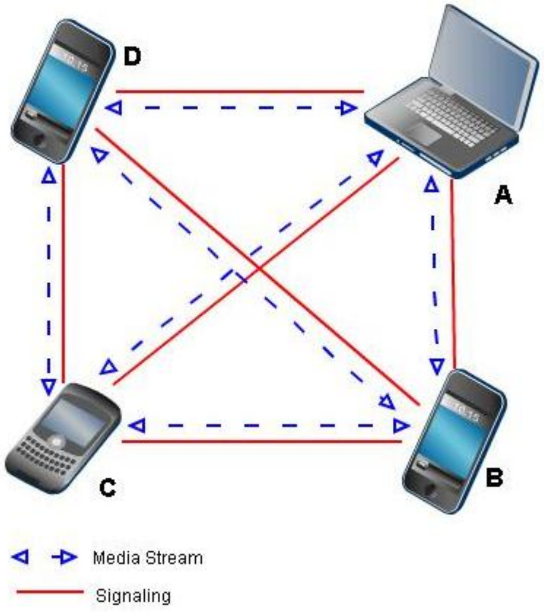

Figure 3.4: Mesh (multi-unicast) model of media conferences

3.2.2 MULTICAST

In the multicast model of conference, the initial signaling procedure is responsible for announcing the multicast address and the ports which will be used to send and receive media streams. This can be done for example through SIP INVITE messages or through SAP (Session Announcement Protocol) [38]. Once all the participants know about the multicast group address where the conference is taking place, they can open their ports and simply start sending/receiving on the multicast address to participate in the conference. It should be noted that signaling is still point to point. It is only the media streams that are multicasted. The model relies on the deployment of multicast on the lower layers such as the IP layer.

It has been seen that multicast deployment remains limited to local area networks, and the internet still does not allow large scale multicasting. Thus the applicability of this model is limited to local networks only.

Once again, the mixing process is a responsibility of the end systems, since no central mixer is present. However, there is a saving in upstream bandwidth as compared to the multi-unicast architecture. Each participant only has to send out his media stream once to the multicast group (as opposed to unicasting multiple copies of the same media stream for each participant), after which it becomes the responsibility of the multicast network to deliver it to all nodes part of that group.

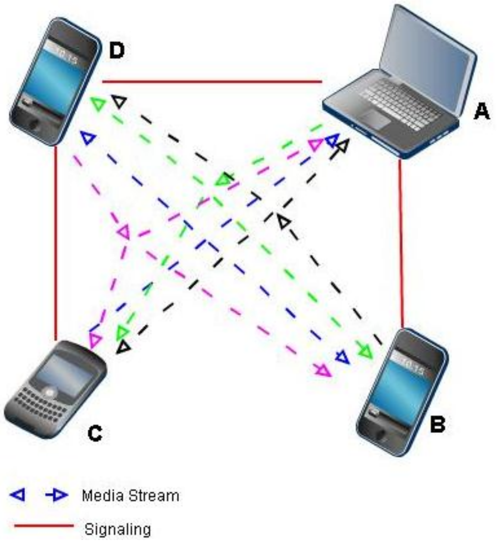

Figure 3.5: Multicast conference architecture

3.3 HYBRID

The hybrid model is a combination of the centralized and distributed models. The central conference server only handles the signaling and thus maintains control over the conference state. The media streams flow directly between the conference members either through multiple unicast streams or through multicast. The central server can use third party call control mechanisms [39] to allow new participants to send/receive media to all other participants of the conference. "*As a result, if there are N participants in the conference, there will be a single dialog between each participant and the focus, but the session description associated with that dialog will be constructed to allow media to be* *distributed amongst the participants*” [40]. The motivation of using such a model for conference can be that the central conference server does not have enough resources to handle the media manipulation/mixing processes, so it prefers the end systems to handle the media without involving the central server in it. At the same time, conference control is maintained by staying inside the signaling path. The central server can always remove any participant from the conference, add a new one and maintain useful statistics about the conference.

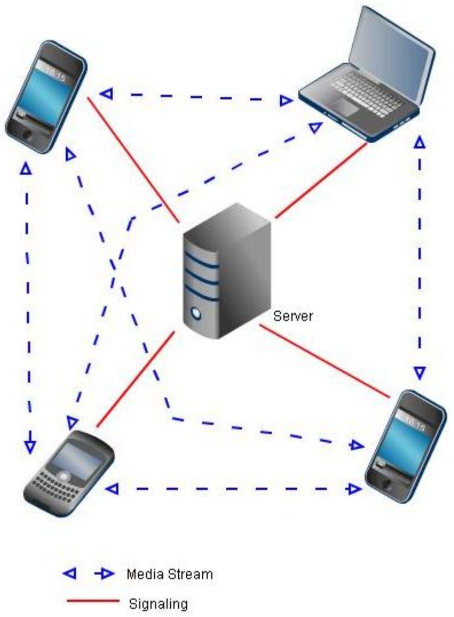

Figure 3.6: Hybrid conference architecture. Signaling associations are centralized, media flow is de-centralized.

3.4 COMPARISON OF CENTRALIZED AND DE-CENTRALIZED ARCHITECTURES

All of the architectures discussed above have their pros and cons, yet they remain in active use. We will now summarize the key differences in the above mentioned conference architectures.

- **Load Distribution:** In the centralized architecture, the load is concentrated on one central entity which is responsible for the transcoding and mixing of all media streams. This takes the load away from all end systems and thus makes it possible for lower end devices, having less computational or bandwidth capacity, to participate in the conference. In the de-centralized architecture, the load is distributed to end systems. While this removes the requirement of one powerful central system, but it does pose a certain minimum amount of requirements on end devices to join the conference. As the size of the conference increases, end nodes might need to handle (receive, decode, encode and send) more media streams. This can result in some nodes exceeding their available resources, either in terms of computational power or network bandwidth and thus will not be able to participate fully in the conference.

In table 3.1, we make a more formal comparison of different architectures and their complexity in terms of bandwidth as well as processor demands.

Topology	**Centralized**	**Mesh**	**Multicast**
Server/mixing-endsystem CPU	O(N)	N/A	N/A
Server/mixing-endsystem BW downstream	O(N)	N/A	N/A
Server/mixing-endsystem BW upstream	O(N)	N/A	N/A
Endsystem CPU	O(1)	O(N)	O(N)
Endsystem BW downstream	O(1)	O(N)	O(N)
Endsystem BW upstream	O(1)	O(N)	O(1)

Table 3.1: Complexity analysis and comparison between different conference models

- **Conference control and administration:** The centralized architecture is better suited for conference control than the de-centralized architecture. In de-centralized topology, all conference participants establish individual connections with each other. Consequently, it becomes difficult to manage the conference (for example accepting and removing participants) and maintain realtime conference state. This includes participants’ presence information, conference floor control state and other useful statistics concerning the conference. In contrast, in centralized conferences the central server/node is always inside the signaling/media path connecting different participants and thus it governs the state of the conference at any time. Thus, it is easy for an administrator to enforce conference policy.

- **Identification of conference participants:** De-centralized conferences have an edge over their centralized counterparts in terms of identifying conference participants. In centralized architectures, all end nodes have only one signaling association with the central server/node which sends them the mixed media stream. In such cases, the identities of all the participants must be explicitly mentioned in the media stream or through some other conference control protocol. For example, RTP headers can contain the identities of all the nodes contributing to the mixed media stream, inside the contributing source (CSRC) header field which when used with RTCP Source Description (SDES) reports can announce a list of conference participants. But it is solely the responsibility of the central server/node to add this information to the RTP headers or RTCP reports while it is mixing various media streams. In contrast, in de-centralized architectures all the participants are receiving individual media streams from all other members of the conference and thus identifying the conference participants at any time is not an issue. No separate means need to be put into place to announce the identities of the conference participants.
- **Robustness:** The centralized architecture, due to its central processing is more susceptible to threats pertaining to a single point of failure. If the central node handling all signaling and media streams is attacked or dis-connected from the network, the whole conference simply terminates. De-centralized architectures are more resilient to infrequent node disconnections.

3.5 SUMMARY

There are different architectures in which conferences can be setup. Mainly these can be organized in two broad categories namely centralized and de-centralized. Choosing a specific architecture generally means deciding how responsibilities will be shared among different nodes in a conference. Availability of resources at participant nodes or in the network also dictates which architecture will be suitable for a certain conference session.

In general it is seen that centralized architectures give better control over the conference, and concentrate the load (both in terms of CPU and network bandwidth) on one central node whether that is a conference server or a resourceful end system. While their de-centralized counterparts distribute the load on participant nodes thus reducing the need for one powerful node but consequently end nodes have to deal with their share of the load. At the same time we can argue that this makes the system more robust by eliminating a single point of failure.

Hence, we observe that both architectures have their advantages and disadvantages. In following sections we will see how and under what circumstances these both architectures are taken into use in today's networks.

4 MEDIA PROCESSING IN VIDEO CONFERENCES

A vital segment of the whole conferencing architecture is media processing. Once the signaling has established required connections and all the conference members have joined the conference in the required topology, it is time to distribute the media streams in a manner that everyone can listen/see the desired participants simultaneously. This means that the individual media streams originating at each conference participant need to be mixed or (if needed) transformed in some way with other media streams. As discussed in chapter 3.1, this task is handled by an MCU in centralized conferences and is pushed to end systems in de-centralized architectures. In the following sections, we will primarily be focusing on centralized architecture as it remains to be more popular with large scale conferences.

4.1 MULTIPOINT CONTROL UNIT DESIGN AND ARCHITECTURE

An MCU acts as the central node both in signaling and media planes in the centralized conference architecture. All participants of the conference are tightly connected with the MCU. That is, each participant establishes a point to point association with the MCU and all media traffic for all participants flows through it. Due to the huge media processing demands on the MCU, it is generally built on high performance media processing DSP chips with realtime media handling capabilities. The hardware and software capabilities of an MCU, although necessary for multiparty conference calls, can also be used in point to point calls where both end systems of the call do not have compatible media capabilities. Such cases are generally resolved within the capability negotiation procedures that take place during the session establishment time. However, in some cases, due to the acute difference in the devices, even the initial capability negotiation phase might not resolve into successful matching media attributes. In such a scenario there is a need for a media transcoder which can convert the media into compatible formats for both end points and make the communication possible.

4.2 RESPONSIBILITIES OF AN MCU IN A CONFERENCE

In the signaling plane, MCU is assigned the responsibility of managing multiple dialogues, one for each conference participant. In the control plane, the MCU can be used in conjunction with floor control protocols, where different participants can dictate their right to speak or use other resources in the conference. As one MCU generally hosts multiple conferences simultaneously, mechanisms to maintain and to control different simultaneous and disjoint conferences must be implemented.

In the media plane, which will be the focus of the rest of the chapter, the MCU has the following main responsibilities:

- Receive media streams from all active participants of the conference. Active participants are defined as those, which are generating media streams in realtime. Such participants can be distinguished from the passive participants which are in a listen-only state, i.e. receive media streams from the active participants but do not generate any media streams of their own. For example, in a class room conference between the students and the teacher, the teacher is the active participant for most of the time while the students are the passive participants.
- Decode the media streams received from active participants.
- Apply transformations as necessary to the decoded media streams (such as re-size, change color depth, add a textual layer over video, or re-shape video)
- Mix together the media streams in realtime to generate a composite mixed stream. This in turn might require the MCU to generate one separate mixed stream destined for each participant. For example in

Figure 4.1, node A receives a mixture of all streams except the one originating from A itself.

- Encode the mixed media stream for delivery to all participants of the conference. This step can vary depending upon the capabilities of the end client devices and their access network characteristics. If all devices have (almost) similar capabilities (such a screen size, video resolution display, video/audio codec support, network bandwidth or capacity), the same encoding can be applied for all participants. However, in case of differences between the end devices, a separate media stream with varying frame rate or bit rate values will be encoded depending upon the requirements of each client and then sent out to each participant.

4.2.1 MCU STRUCTURAL ARCHITECTURE

Taking the above mentioned responsibilities of an MCU into account, one can draw a simple structure diagram which represents the steps taken by the MCU on the media plane. Let us consider for example that we have four participants in a conference, and each of these four is an active participant. The diagram below shows the stages through which the media streams pass until they emerge out as different versions of the mixed media stream.

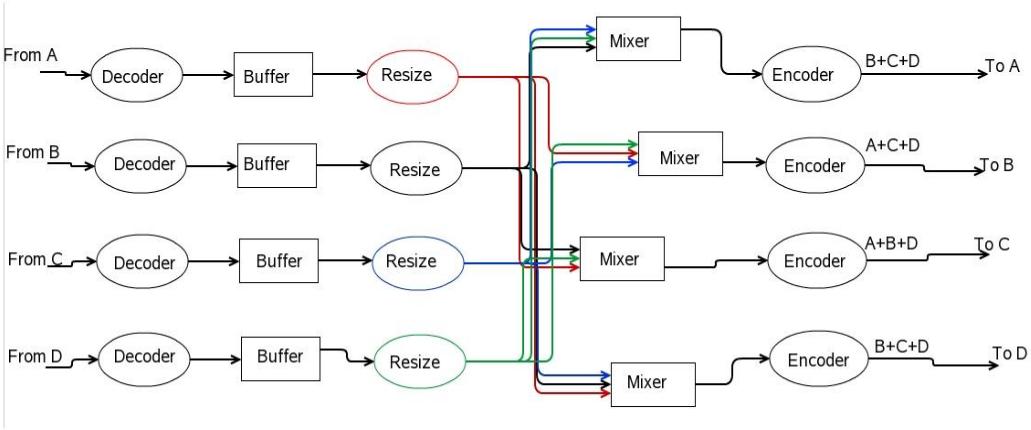

Figure 4.1: Internal logical structure of an MCU

Looking at this diagram, we can easily see that it consists of a number of steps, many of which require active processing from the CPU and pose huge demands on the bandwidth.

4.3 MCU PROCESSOR AND BANDWIDTH REQUIREMENTS

Fulfilling these responsibilities discussed in previous section adds processor and bandwidth load on the MCU. Additionally there are a few factors which play an important role in defining how much CPU and bandwidth is required for media transcoding/mixing. These are:

- Video resolution: number of pixels in one video frame
- Video Frame Rate: number of frames per second that are throttled through to and from the MCU
- Video Codec: defines the compression and other algorithms that can affect the bit rate of the video stream and also the computational complexity involved in the encoding and decoding processes
- Audio codec and bit rate accompanying the video: audio and video go as separate streams over the IP network, and thus will need to be mixed separately by the MCU.

4.3.1 PROCESSING DEMANDS

We will now look at the requirements posed on the processor by the transcoding/mixing tasks that an MCU must perform in realtime.

A. Standard Definition Video

First we take the case of standard definition video with a resolution of 640 × 360 pixels. It is also known as nHD resolution. There are other commonly used resolutions for video conferencing such as CIF (352 × 288) and 4CIF (704 × 576), but nHD (640 × 360) was chosen for the ease of comparison with HD (1280 × 720) since in terms of pixels its frame is exactly one fourth the size of a

720p HD frame. We take a frame rate of 25fps. Experiments were performed over a number of video sequences, and results were averaged out. The experiments reveal that at the said resolution and frame rate, the required CPU frequency to decode and encode the media in realtime is:

- H264 decoding SD video 640×360 @ 25fps = 400 Mega cycles/second
- H264 encoding SD video 640×360 @ 25fps = 550 Mega cycles/second

If this many CPU cycles are not available, the frames will need to wait in queue in the buffer and the realtime performance will severely degrade. Such a degradation affects the perceived quality of service for the end user.

Using the values for required processing cycles we can approximate to some extent the required CPU cycles at the MCU in a multiparty conference.

Number of Participants	Encoding (Cycles per sec)	Decoding (Cycles per sec)	Total (Cycles per second)
3	1650	1200	2850
4	2200	1600	3800
5	2750	2000	4750
6	3300	2400	5700

Table 4.1: Required processor cycles for encoding and decoding H.264 Standard Definition Video

The above calculations are a safe estimate, since they only take into account the processing capacity required for decoding and encoding steps. The processing demands for mixing the decoded streams, resizing the video frames or for any other modifications can in fact add to the processor cycles demand. In conferences video is generally also accompanied by an audio stream, which adds to the mixing load on the processor. Each participant sends and receives infact two separate streams: one for video (counted for in the above calculations) and one for audio (neglected for brevity of calculations).

Comparing this to the CPU capabilities of today's smart phones and handheld devices described in Appendix A, we see that even the high-end devices can only barely support a three party conference. Anything above that is beyond the processing capability of a single device.

There has been some research work done on the subject of cooperative mixing. Cooperative mixing means that if a single peer can not handle the demands of media processing in a conference, multiple peers can pool in their resources to process the media streams together. This scheme is still under development and faces a lot of challenges. It has been covered in more detail in chapter 4.6.2.

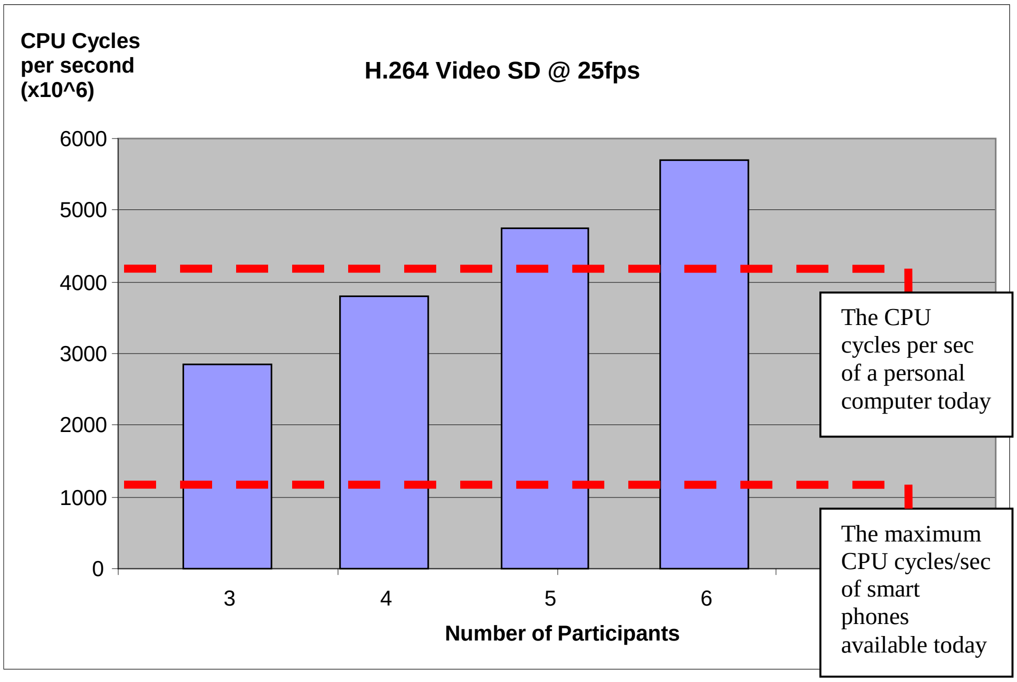

Figure 4.2: Processor cycles requirements for SD video conferences

B. High definition video:

The increasingly popular HD or high definition video uses more pixels per frame, which makes the images in the video look more sharp, crisp and realistic. Of course the trend of going high definition comes with the cost of more bandwidth, memory and CPU requirements. Just as for SD video, we make calculations for the High Definition Video scenario. We consider the 720p variant of the HD video. This has a resolution of 1280 x 720 pixels on screen per frame. For realistic video, we assume a 25 frames per second video stream. Similar to the case of SD video, decoding and encoding of a variety of HD video sequences was performed and the resulting values are given below:

- H264 decoding HD video 1280×720 @ 25fps = 900 Mega cycles/second
- H264 encoding HD video 1280×720 @ 25fps = 2000 Mega cycles/second

Number of Participants	Encoding (Cycles per sec)	Decoding (Cycles per sec)	Total (Cycles per second)
3	6000	2700	8700
4	8000	3600	11600
5	10000	4500	14500
6	12000	5400	17400

Table 4.2: Required processor cycles for encoding and decoding H.264 High Definition Video

As seen from the results in Table 4.1 and Table 4.2, switching from SD to HD video increases the processing requirements at the mixer manifolds. Even for a three party conference, we need a system which has at least 2.8 x 10^9 cycles available per second for this task. Such a processing capability is unheard of in the handheld devices industry, but if we take into account personal computers, we may find that the latest personal computers can have as much as a dual core 3 GHz processor, which makes it a 6 x 10^9 cycles per second system. Nevertheless, the normal operating system specific tasks should also be counted for. All in all, a very high-end and well maintained personal computer might be able to maintain an HD conference between 5 to 6 participants.

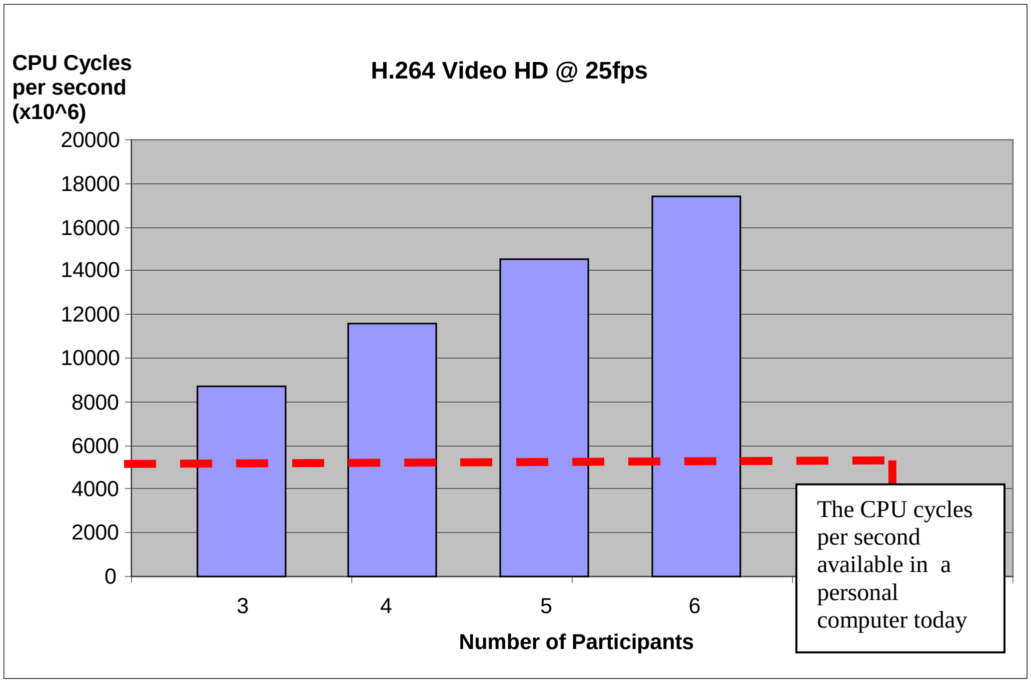

Figure 4.3: Processor cycles requirements for HD video conferences

As a further case study, CPU load while streaming videos from one of the most popular online streaming websites [28] was measured for different video resolutions. The details of the used client machine for these test runs are:

- Operating System: Linux SUSE 10 with SMP (multi-core) support
- Processor: Intel 2.2 GHz dual core
- Main Memory: 2 Gbytes

Following are the used CPU cycles for streaming one video at a time in the absence of any other processor intensive tasks running simultaneously. The major portion of these CPU cycles go into decoding the received video stream which is encoded by YouTube in H.264 format [29].

- 320p = 660 cycles per second
- 720p = 1672 cycles per second

- 1080p = 2420 cycles per second

4.3.2 BANDWIDTH UTILIZATION

Network bandwidth is the other scarce resource that needs to be taken care of during video conferencing. While an end client has to receive a single mixed stream for video (and one stream for audio), the MCU has to receive one video stream for each active participant. In the upstream direction, the MCU may need to send a different version of the media stream for each participant. This increases the bandwidth load on the MCU manifolds compared to an end system.

YouTube is currently one of the most popular online video streaming websites, and it is reported to deliver 30fps HD 720p video at around 2mbps datarate. Amazon, which is another popular video hosting and streaming portal, streams HD videos in 720p at 30fps at an average datarate 2.5mbps.

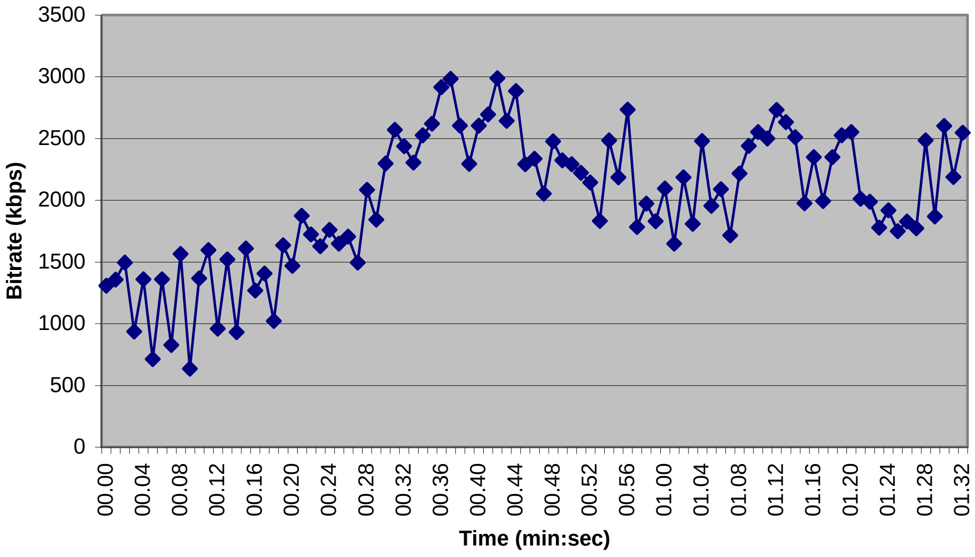

Figure 4.4: Bitrate variation in a high definition 720p video on YouTube [29]

For conference applications, after using some further compression, further reduced bitrates can be achieved.

Average datarates during video conference using H.264 at 25 frames per second are summarized as follows:

Resolution	**Bitrate**
HD 720p (1280×720)	1.5 Mbps
SD (640×360)	512 Kbps
CIF (352×288)	256 Kbps

Table 4.3: Average bit rates of video in media conference applications

Since an MCU has to receive and send out multiple video streams simultaneously, the required network capacity would be much higher. Figure 4.5 shows the average datarates in downstream as well as in upstream direction at an MCU for different number of conference participants.

Participants	**CIF (kbps)**	**SD (kbps)**	**HD (kbps)**
3	768	1536	4500
4	1024	2048	6000
5	1280	2560	7500
6	1536	3072	9000

Table 4.4: Average video bit rates at an MCU in media conferences

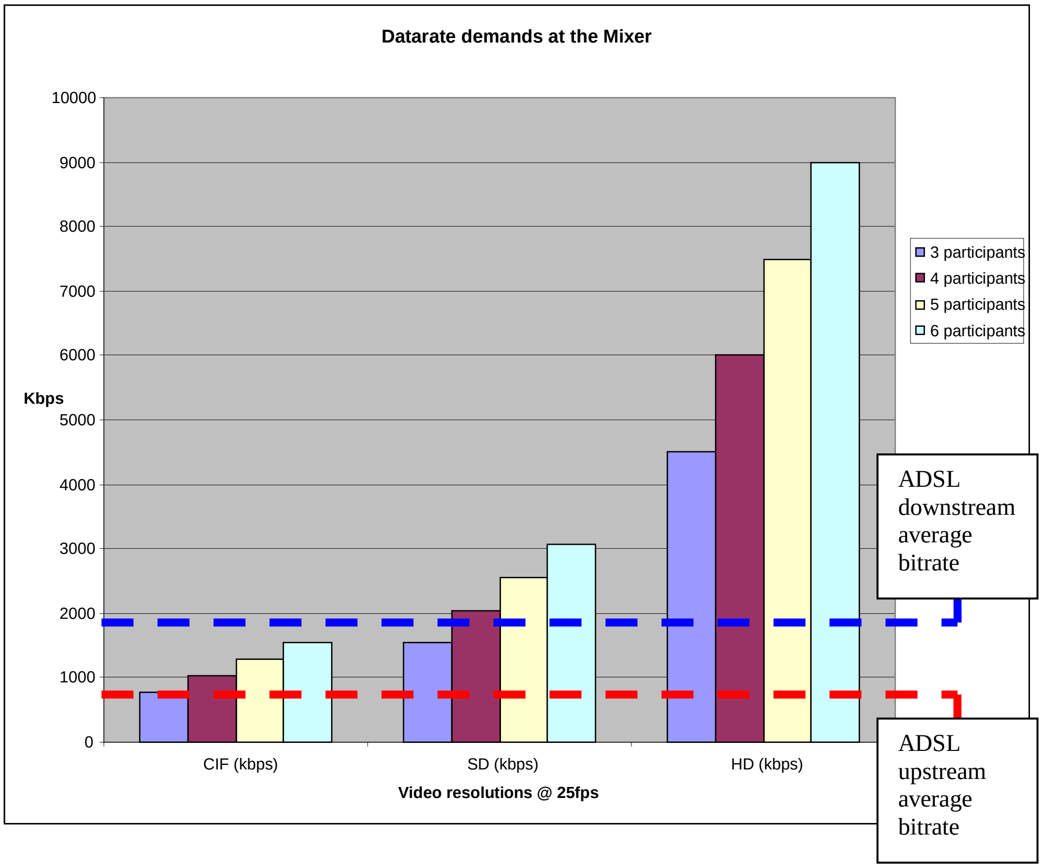

Figure 4.5: Required network bandwidth in video conferences

This shows the huge difference in bandwidth demands as we increase the video resolutions. It is important to note that this same bitrate is required by the network both in downstream as well as upstream direction. Most Internet connections have ample downstream bandwidth but when it comes to upstream, many end points lack even the minimum requirement for efficient and smooth video upstreaming.

It is possible to encode videos with the same resolution and frame rate but at lower bitrates, but this will reduce the quality in terms of SNR. For a viewer, this phenomenon will result in visible compression artifacts on the screen.

Another aspect to look at from the bandwidth utilization point of view is to make sure that where ever possible, the conference participants and MCUs are arranged in a topology which aims to reduce undue traffic on costly network links. Cross-ISP traffic generally means greater operational cost for the ISP.

OTT client applications generally do not have regard for such underlying network topology information when establishing connections with each other [33]. Instead they simply rely on measuring Round Trip Times (RTT) to estimate the quality of links. However ISPs and Telecom operators by virtue of owning the network have much better and detailed information about how the network is built and how links are dimensioned, and can thus select and configure the network links according to the traffic demands.

Bandwidth capacities of client devices today

The general home users with broadband or ADSL connections today reach internet connectivity rates as shown in Figure 4.6 below. Refer to Appendix B for more detailed values of network datarates. These of course vary with the distance from the central office.

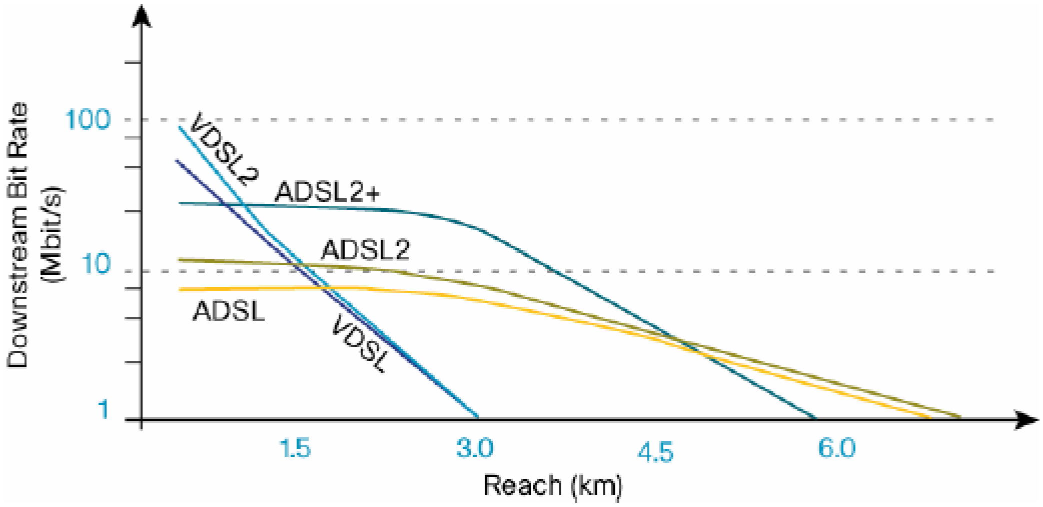

Figure 4.6: Internet datarates available to average end user today [34].

For a mobile user the access network generally is the bottle neck for the speed with which he or she can upload/download information from the Internet. The different access network technologies that exist today are compiled below along with their approximate theoretical (per cell) as well as practical (per user) datarates. The practical per user datarate is always less than the theoretical datarate because of many factors, some of which as listed below:

a- The available bandwidth in a cell is shared among all the users present in that cell. This means that as the number of simultaneous users in the cell increase, the per user datarate decreases.

b- Distance from the base station or access point also plays an important role in determining the strength of signals reaching the user terminal and thus determines the experienced datarate.

c- Interference from other devices in the neighborhood might also cause the datarates to suffer. This is especially true in case of WiFi, which works in the ISM (industrial, scientific and medical) band. Many other devices and technologies can also work in the same frequency band.

	Theoretical maximum datarates		**Practical per user datarates**	
Technology	**downstream**	**upstream**	**downstream**	**upstream**
UMTS	177Kbps	118Kbps	70-130 Kbps	70-130 Kbps
HSPA	14Mbps	5.8Mbps	1 to 4 Mbps	0.5 to 2 Mbps
LTE	100Mbps	50Mbps	13Mbps	3.8Mbps
Wireless LAN, WiFi	54Mbps	54Mbps	2 Mbps	512Kbps

Table 4.5: Average bit rates for mobile users using different access technologies [31][32]

4.4 SUMMARY

In summary, we see that the tasks of mixing and transcoding media streams from multiple participants in media conferences place stringent requirements on the computational and bandwidth capabilities of the MCU. This is particularly true for video conferences which support SD or HD video content, since this adds a lot more data to be processed and transmitted in realtime. When compared against the potentials of end user terminals available today, it is seen that even in best case scenarios, client devices struggle to meet the required processor and bandwidth demands even for small to medium scale conferences.

5 MEDIA PROCESSING IN COMMERCIAL COMMUNICATION NETWORKS

Having gone through the fundamental aspects governing media conferences in packet switched networks, we now go into details of how media conferences are really implemented in large scale commercial networks today. The most challenging aspect in media conferences is to handle multiple media streams simultaneously in realtime. The introduction of video makes it an even greater challenge due to the large size and complexity of video packets. We will now focus on the media processing methods used in two major sectors of communication networks widely seen today:

- Operator Networks
- Over the Top (OTT) Networks

5.1 MEDIA PROCESSING IN THE OPERATOR NETWORKS

Operator networks rely on the intelligence and capabilities hosted by the network. Building on powerful, reliable and redundant systems, these networks can provide services to large populations of subscribers on a continuous basis. The same is true for media processing tasks including media transcoding and media mixing. It is thus logical for operators to follow the centralized media conferencing architecture, where the entire media processing load is taken up by specialized servers in the network.

It is also possible that a session spans over multiple operator networks, such as a conference call where some participants of the conference belongs to network A and other participants belong to network B. In such cases typically the media processing and mixing is handled by the network where the conference call originates.

In the following sections, we give an overview of some of the common network elements in telecom operator networks which are good candidates for providing media processing support in small to large scale conferences [43].

5.1.1 MEDIA GATEWAYS

Media gateways reside in the network's media plane, where they act as intermediate network elements. All media is routed through them. Thus, they naturally pose a promising place for media transcoding in the network. The conversions performed by media gateways between different networks range from voice streams transformations, such as insertion of tones or changing of the used codecs, to data stream manipulation. Hence, they already have the needed DSP requirements and processing power that can guarantee realtime media transcoding and mixing.

5.1.2 SESSION BORDER CONTROLLERS

Session border controllers (SBC) are network elements deployed at the network edge (user-network interface (UNI) or network-network interface (NNI)). They primarily serve the purpose of ensuring network security. They monitor both signaling and media flows into and out of the network and perform a variety of security related functions, such as network address translation, firewalling or bandwidth monitoring, to avoid possible bandwidth theft and to enable QoS provisioning. To fulfill these stringent requirements, the SBCs have typically fast processors to dissect the packets quickly and rewrite their headers in realtime.

Since these devices lie at the edge of the network, they also provide a convinient location for media transcoding/mixing.

5.1.3 APPLICATION SERVERS

Application Servers (AS) are standalone processor intensive systems installed in an operator's core network specialized for providing specific services such as presence/location services, Network Address Translator (NAT) traversal services, Web portals etc. Considering their processing power and the fact that they can be built specific to the demands of the application, they are optimum for media processing. One drawback is that application servers are typically located in the network core and not at the edge of the network. Thus, even if two users are topologically close to each other in the network, their media streams will need to be routed all the way to the network core to undergo the required transformations. We see that in most cases, the location of application servers in the network plagues them with backhauling problems.

An example of AS is Media Resource Function Processor (MRFP). It is an application server in IMS domain which sits on the media plane to provide media mixing and tone generation services to IMS user agents.

5.2 MEDIA PROCESSING IN OTT NETWORKS

The majority of the Over The Top (OTT) services use peer to peer (P2P) networking as their underlying communication framework. Peer to peer networks are by definition those which do not have any dedicated clients and servers in the network. Instead several end systems (or peers) share their resources together to form an overlay network which can provide the intended services [42]. This inherently makes the peer to peer systems less reliable as compared to their centralized (dedicated server oriented) counterparts. The reason for this stems from the fact that the P2P systems have no control over the nodes that make up the network. Any node can leave the network without prior notification, and the P2P system has to adapt to such abrupt changes dynamically. Also, such systems are impeded by lack of dedicated resources, and instead all the requirements (CPU cycles, memory or bandwidth requirements) must be taken care of by the end systems themselves.

We take for example Skype [37], a widely used P2P telephony application. The Skype overlay network consists of two tiers of peers which can be referred to as nodes and super nodes.

When a skype node A wishes to make a voice or video call to another node say B, it establishes a TCP signaling connection with B, and then the media can flow over UDP or TCP. However often the scenario is not that simple, and either one or both the nodes are behind a NAT. In such scenarios, the caller A and callee B have to rely on other nodes in the skype network to act as relay nodes, which would then forward the packets from A to B and vice versa [46]. If during the call, the relay node disconnects from the skype network, a new relay node must be dynamically chosen. It is also possible that one node is being used as a relay by multiple other nodes in same or different simultaneous call sessions. This means that the relay node would see a lot of random traffic flowing through it.

Thus we see that the whole network relies on a certain minimum number of nodes to be ideally available and online simultaneously within the overlay for it to function properly. More number of online nodes guarantee a better service to all users while a decrease in nodes in the overlay would result in a lack of available resources and hence a degradation of service. More detail on the Skype P2P network is given in section 6.4.2.

Another example is that of internet based P2P Content Distribution Networks (CDN), such as PPlive [47]. Instead of streaming video or live television broadcast from some central streaming server(s), the users of the P2P service distribute the media streams to each other. This means that one peer will act as a streaming server for many other nodes and will upload the media stream(s) to them. It has been observed that there can be long delays before streams reach a peer in such a P2P CDN. The delays have been observed to range from a few seconds to even minutes [48]. Such delays might be acceptable in normal broadcast viewing, but serve as a challenge when considering interactive TV.

Realtime applications such as video conferencing pose stringent requirements on end to end delay and processing power of the participating nodes. This makes it a challenge for peer to peer networks to host such services.

In the following subsections we take a look at the different possibilities and limitations for video mixing and transcoding with respect to conference applications in today's OTT networks.

5.2.1 MESH NETWORK (MULTI-UNICAST)

The simplest solution in terms of architecture of the conference in the absence of any central conference server is a full mesh network. As discussed previously in section 3.2.1, in the mesh network every conference participant has a direct connection with every other participant. Considering the media flow, we see that a node has to unicast its media stream(s) to all the other conference participants. This also implies that a node will receive multiple media streams, generally one from every active participant. Receiving and sending out many media streams simultaneously is a bandwidth exhaustive process, and considering that video packets are quite bulky, the bandwidth utilization may hit the upper limit of many end user connections as the conference grows larger.

Nodes will also have to decode all of the received media streams to be able to play them. This also adds to the CPU load on the end devices.

5.2.2 SINGLE PEER

Another straight forward media conferencing architecture in the absence of a central server is that of the end system mixing, where one of the end systems is nominated to take care of all the transcoding and mixing processing. The end system mixing model is explained in more detail in section 3.1.2. This model works for smaller conferences, but is not very scalable. Based on our analysis on the design and requirements of an MCU in chapter 4.3, many of today's end systems and user terminals are incapable of handling the task of mixing and transcoding for good quality video from more than three participants.

Many of the client applications running on peer devices that make up OTT networks have certain criteria against which an end device is chosen to handle the media mixing tasks. [46] suggests that Skype, a well-known P2P client, nominates the mixer on the basis of its downstream/upstream bandwidth and its processing capabilities as compared to the rest of the peers in the conference. In addition, elaborate client protocols may enable the role of the media mixing to be shifted from one peer to another in case a peer with better bandwidth and processor capabilities joins the conference. The signaling associations may however remain coupled with one central peer throughout the conference, as shifting the signaling association from peer to peer might be complex. This makes such an architecture unreliable and susceptible to complete breakdown in case the peer acting as the center for the conference leaves the overlay or gets disconnected [36].

5.2.3 MULTIPLE PEERS (COOPERATIVE MIXING)

The idea of cooperative mixing has been proposed in some publications [40][57]. The model aims at distributing the task of mixing and transcoding, to more than one device in the conference. This way multiple peers can pool in their resources to cumulatively perform the tasks of an MCU. This model can be seen as a hybrid between completely centralized architecture (where one system is responsible for all mixing/transcoding tasks), and a completely distributed architecture (full mesh architecture as described in section 3.2.1). Even though multiple peers may be contributing their resources for media processing tasks, the signaling might still be handled by one central peer in the network, which in turn would use third party call control functionality to instruct other participants to make use of multiple mixers in the network.

The whole peer to peer network can be seen as arranged into two separate layers. One layer is made up of computationally capable peers with good network connections constituting the “MCU cloud”, while the rest of the peers attach to this layer or cloud. The idea is somewhat similar to the concept of super nodes, which make up a backbone for the peer to peer network. In case of conference, this backbone will also be responsible for providing MCU related functions. Such an architecture can further have two variants: one in which the different MCU’s are arranged in a chain, also referred to as cascaded mixers, and the other in which they make up a mesh between themselves. These two possible arrangements of peers inside the MCU clouds are demonstrated in Figure 5.1 and Figure 5.2. The arrows represent media flows, and the numbers on the arrows depict the peers which contribute to the mixed media stream flowing on this link. Each participant not acting as an MCU will receive a mixture of media streams from all other participants except his own media stream.

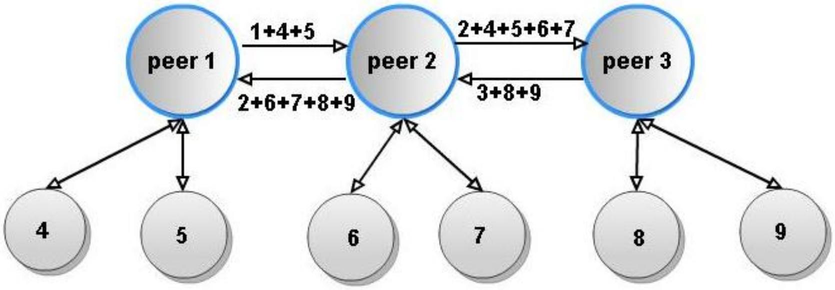

Figure 5.1: Cascaded MCUs

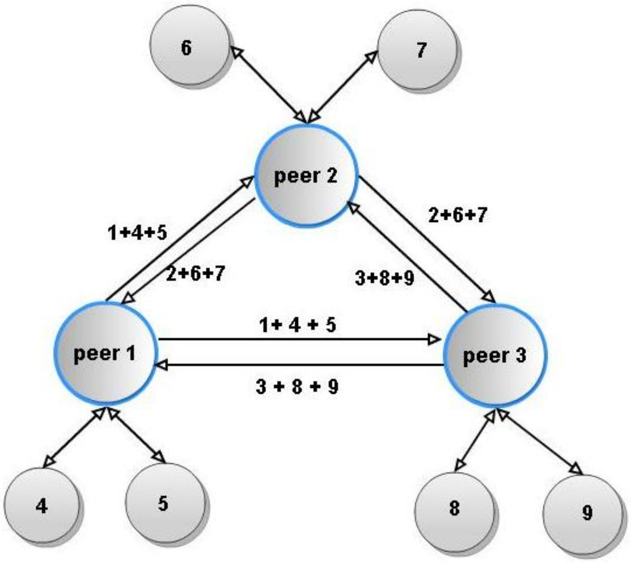

Figure 5.2: MCUs arranged in a mesh

Limitations of cooperative mixing

Both the architectural models have certain limitations. The mesh model adds to the computational load of the overall conference system. As the number of cooperating mixing peers increases, they each in turn act as a media source for the others and thus the system does not scale too well. For the cascaded mixers architecture, the mixing and transcoding delay becomes the limiting factor, as each mixer adds a certain latency between the input streams and the output mixed stream. Thus only a limited number of mixers can be chained together if the communication needs to be realtime. We will now take a deeper look at these shortcomings on such cooperating MCU architectures.

Limitations when MCUs are arranged in a mesh

We will first discuss distributing the media mixing/transcoding tasks to multiple peers arranged in the form of a mesh, as shown in Figure 5.2. This scheme distributes the load onto multiple peers, but in general this does not mean that the more peers we add to the so-called "MCU cloud", the better the performance is going to be. First, it depends on how many capable peers we have in the network, and even we have ample amount of such peers which can handle the load of media mixing and transcoding, it does not mean that adding all of them to the cloud would be the best solution. To understand this concept, we introduce a metric called “load-per-MCU”, which quantifies how many other peers are connected to an MCU in a network. Of course the more peers an MCU has to handle, the more processing cycles it needs.

It should be taken into account that if there are multiple MCUs in the network, the output of one will act as an input stream to the other so that media streams from all participants reach every participant. Now consider we have 12 peers participating in a conference. Considering that we have ample number of capable peers among them, the following combinations depicted in Figure 5.3 are possible and many more until the system becomes a full mesh network. One thing to keep in mind is that MCU with more capability can take up large number of peers, while less capable MCUs will serve smaller number of peers.

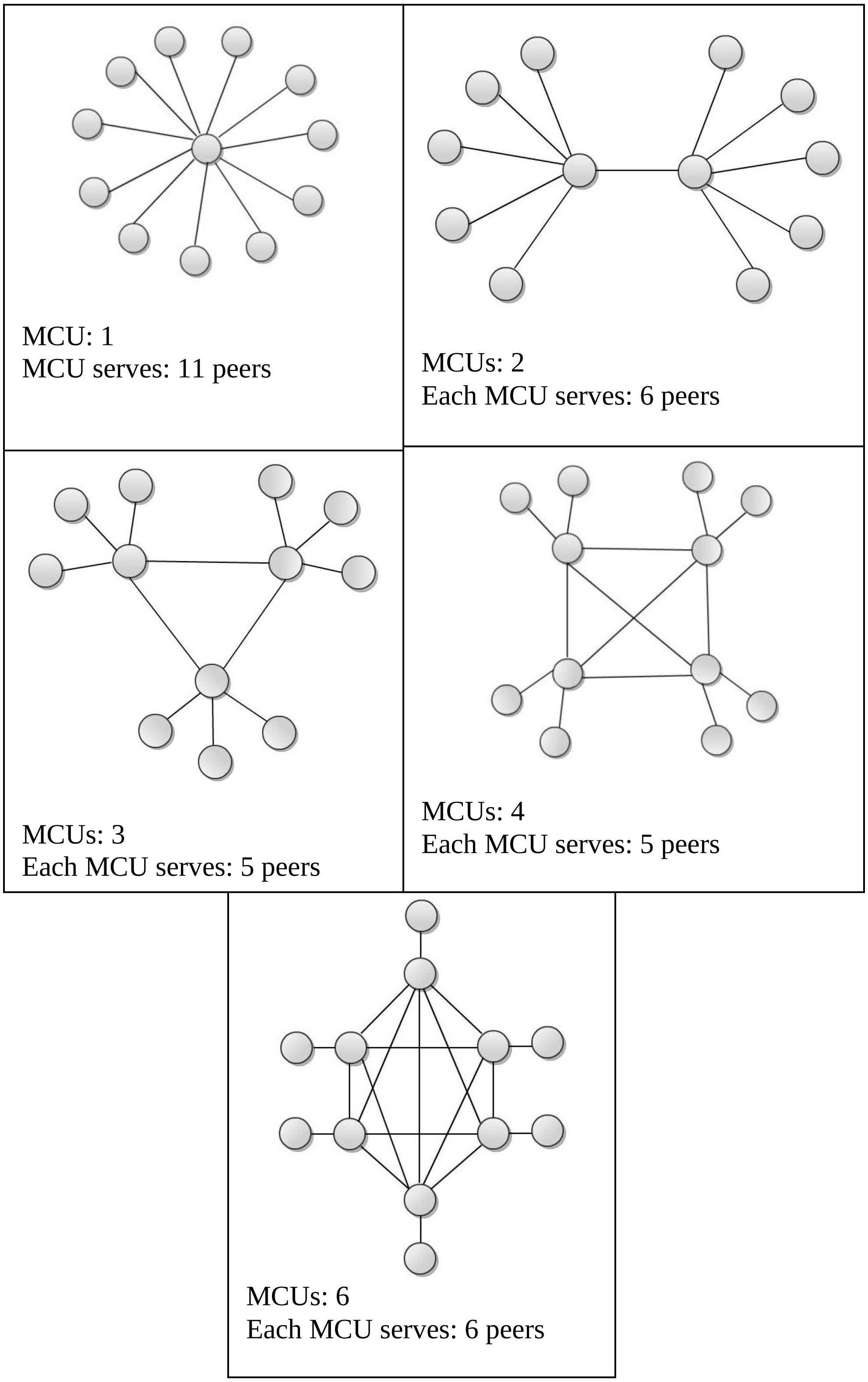

Figure 5.3: Possible scenarios in which MCUs can be arranged in a mesh

As can be seen from Figure 5.3, the most optimum load distribution is achieved with 3 MCUs in the network. Each MCU has to handle 5 peers (out of which 2 are other MCUs). When the number of MCUs increases to four, although the number of peers served by each MCU remains the same, the number of media streams flowing through the network increases. This effect is depicted in Figure 5.3. As more MCUs are added, the number of peers served by each MCU starts to increase.

The lower bound for the number of peers handled by a mixing peer, is therefore 5 peers in 12 member conference. For a normal end system today, handling 5 peers is not an easy task.

Using simple numerical analysis methods, an equation can be derived for calculating load-per-MCU. The load-per-MCU *y*, in a conference with *N* participants, amongst which *x* are nominated as cooperating MCUs can be expressed as

$$y = \frac{N}{x} + x - 2$$

This represents a parabola. Our intention is to find out the number of MCUs *x* that need to cooperate in order to attain the smallest load-per-MCU *y* in a given conference. We thus need to calculate the minima of this equation. Although *x* and *y* represent sets of discrete numbers (whole numbers), but if we for the moment assume them to be continuous, then taking derivative of both sides

$$\frac{dy}{dx} = \frac{d\left\{\frac{N}{x} + x - 2\right\}}{dx}$$

$$\Rightarrow \quad \frac{dy}{dx} = -\frac{N}{x^2} + 1$$

and setting the derivative equal to zero will give us the minima of our parabola

$$-\frac{N}{x^2}+1=0$$

$$\Rightarrow \quad x=\sqrt{N}$$

Since *x* can not be a decimal number, so we only keep the integer portion of *x* (equivalent of flooring *x*), which suffices as a solution to our problem scenario. If we put this value of x back into our original equation, it will give us the minimum value of load-per-MCU (number of peers served by each MCU) for a conference with *N* participants as:

$$y=2\sqrt{N}-2$$

Where we again keep only the integer part of *y*. To summarize, for a conference involving N participants, where multiple participants can act as cooperating MCUs arranged in a mesh, the load-per-MCU can not be less than $2\sqrt{N}-2$ in ideal cases. This shows that as the conferences grow bigger, so does the minimum bound of peers per MCU and this minimum limit is still quite high to cope with for the average client devices today.

Limitations of Cascaded MCUs:

We will now highlight some limitations of cascaded mixing model [40]. Media mixing involves complex processes like decoding, resizing and encoding the video frames and it normally takes some time for the MCU to accomplish these tasks. This is especially true for complex codecs such as the H.264, which provide high network efficiency but in turn depend on elaborate mathematical computations during the decoding and encoding process. Depending on the CPU power and capability, the delay can vary from a few milliseconds to couple of hundred milliseconds. Values from Texas Instruments [63] suggest that current media processing DSP boards add about 25ms of delay during decoding process and 30ms of delay during encoding 720p H264 video stream. This means that neglecting all other transforms applied to the video during mixing and transcoding process, just the encoding and decoding at one mixer would add about 55ms of delay.

As we have already pointed out, video conferencing is a highly delay sensitive application, and while other variants of video streaming such as TV broadcasts can tolerate delays in terms of seconds (due to receive buffer), video conferencing can only tolerate up to 200ms of one way mouth-to-ear delay to ensure satisfactory end user experience [64]. A delay bigger than this renders the interactivity in the video stream suffering badly and the user experience can drop.

A delay constraint of around 200ms means that we could have at maximum four MCUs in one chain, and adding a fifth one will render the video conference unusable for realtime communication purposes. This is because the input to one cascaded mixer depends on the output from a preceding mixer, thus the mixers make up a queue where the delays are simply added from end to end.

One way to mitigate this problem would be to select a codec which requires minimal processing time while decoding and encoding, but this in turn means that the codec will not be bandwidth efficient, and thus can not be used over typical networks of today.

5.2.4 CONCLUSIONS ON MEDIA PROCESSING IN OTT NETWORKS

We therefore see that in OTT networks, having a full mesh network has its limitations in terms of bandwidth and CPU requirements on end devices. The case where a single peer is used as an MCU suffers from unreliability issues. Similar limitations apply to both architectures of cooperative transcoding/mixing. In some conferences, the peers might be arranged as a combination of both architectures, i.e. some portion of the conference makes a mesh, while the other portion will have the MCUs chained together, but the optimization bounds are still narrow.

In addition to the above mentioned limitations of cooperative mixing and transcoding, peer to peer overlay networks also suffer from a number of other drawbacks [65] which make them unsuitable for such cooperative models where everything has to be accomplished in realtime. Some of these additional limitations of P2P networks are mentioned below in brief:

Managing peer dynamicity:

In P2P networks, peers can join and leave the overlay dynamically and without any prior warnings, or their bandwidth might suddenly change. Such behavior is even more common with mobile peers. A general problem with P2P streaming applications is that several intermediate nodes might be used to forward a stream from the source to the destination nodes. In case any of the intermediate node disconnects, the network must adapt quickly to construct a new delivery path. Several time critical packets might get delayed, lost or dropped during such a path re-construction.

Incentives for participation:

P2P systems are built of the concept of sharing resources. However it has been observed that peers tend to get greedy, and while they use a lot of resources from other peers, they do not so much want to let others use their resources. Such peers are termed as “free riders” [66]. In our scenario of cooperative mixing, there might be many peers who would intentionally avoid being selected as the mixer or transcoder. If the number of such peers grows, the system becomes more like a client server architecture where the number of clients simply outweighs the server capacity. Several proposals have been suggested [66] for providing incentives to peers in P2P networks such as Virtual Payments, Reputation based systems, Reciprocity based systems etc but it still remains an area open for research.

Security Challenges:

In P2P systems, who to trust with private information is always a question. In the case of conversational P2P systems, if the processing of media streams remains within the participants of the on-going session, it is deemed secure up to some level. But in cases where the processing (mixing/transcoding) demands can not be met by the nodes participating in the session, the processing demands might get pushed to a peer outside the (conference) session, and that opens a lot of trust issues. We cover the security issues in more detail in section 7.

This means that some level of support for media mixing and transcoding should ideally exist inside the network, and the idea of using end devices to handle such an intensive task does not scale too well. However, as discussed before, P2P networks are built on the idea that the individual peers would pool in their resources handle all the tasks, and as such do not have any in-network hosted resources.

This opens grounds for a possible cooperation between OTT service providers and telecom operators, where OTT users utilize the resources from operator network, and, the telco operators in turn get a better user base and coverage by expanding their services into P2P networks.

5.3 SUMMARY

OTT service providers and conventional telecom operators, both have different approaches in how media conferences are setup in their networks. Different conference architectures are preferred depending upon the resources available. But in general it is seen that operator networks prefer to host conferences on specialized conference servers inside the network. This corresponds to centralized conference architectures. The signaling, control and media processing requirements are handled by reliable and powerful nodes which have very strict availability criteria. Media processing demands can be handled at the edge of the network by nodes such as media gateways, session border controllers or in the core by specialized application servers. However heavy investment is required to build up and maintain such networks.

On the other hand OTT networks require little or no investment as they are based on P2P concepts. However generally this means that there is a lack of powerful nodes in the network. Thus media conferences are often established such that responsibilities are shared or distributed between various peers. This can involve multicasting, multi-unicasting or cooperative mixing models. However such models have their own limitations and in the absence of capable peers these models don't scale too well for large conference sessions. In addition there are other inherent limitations in P2P networks such as managing peer dynamicity, offering incentives to participate and share resources in the network and ensuring security of information.

Thus we see that both these domains are fundamentally different and these differences are also quite prominent when it comes to media conferences. In the next section, we shall see if it's possible for these both domains to benefit by cooperating with each other and how such cooperation can be achieved.

6 COOPERATION BETWEEN OTT AND OPERATOR NETWORKS

Traditionally Over The Top (OTT) service providers and carrier network operators have been on competing terms [4]. The network operators invest heavily in planning, developing and maintaining their networks and thus they expect that any service provided to an end user through their network, should generate direct revenue for the operator. Telecom operators have even gone the distance to “lock” the user terminal if the user wishes to install or use a service other than the ones provided by the operator itself. In short, the operators believe in “customer ownership”, meaning that their customers will use only their services. The operator can then use methods such as service aware charging to bill the customer for the service rather than the network bandwidth used. A good network means bigger customer base which generates more revenue.

But this ideal network ecosystem is disturbed when third party companies such as Google, Amazon, Skype etc start providing end users with multitudes of services which are not hosted by the network operator but instead by the free internet. The operator’s customer ownership is lost. The operator can no longer bill the customers for the services they use but instead can only charge for the bandwidth utilization in the network.

While the telecom operators come up with schemes to restrict access to third party services on their networks, the OTT industry fights for net neutrality, meaning that all end users should have the right to access the internet with freedom. Clearly this tussle for customer ownership will continue unless avenues for cooperation are sought. Telco 2.0 [25][4] is one such effort to highlight possible terms on which network operators and over the top service providers can co-exist, while offering value for each other.

Another aspect that also plays a role in shaping the ecosystem of the communications industry is the competition between various telecom operators.

So far we have only referred to the operators as one united entity, however operators also compete with each other. This competition is driven by the notion that better services and lower rates attract a larger customer base. For example some operator might offer network based video recording service during video conference sessions at little extra cost thus attracting more customers to its network. The question thus arises how this inter operator competition will affect their cooperative agreements with OTT service providers. We believe that operators will continue to compete with each other and indeed this competition will also be a major driving factor in deciding which operator enters cooperation with which OTT provider. Having a partnering OTT service provider can be seen as an additional service that the operator can offer to its customers (such add-on services are also sometimes referred to as Value Added Services (VAS) [51] in teleco domains). In fact such cooperations are already beginning to take shape such as:

a) Spotify [52] online music service is being budled with 3 UK [53] service plans [51].

b) Alhough in its early stages, another example is the initiative of cooperation between Ericsson [55] and Akamai [54][26]. Ericsson being a mobile network vendor and Akamai being a well known content delivery service provider can together target content providers with optimized content delivery (video, web content, audio etc) to mobile users. The content provider will pay Akamai for improved user experience. Revenue from content providers will then be shared among three players including Akamai, Ericsson and the mobile operator whose network is used for mobile content delivery.

c) Japanese mobile operator NTT Docomo [56] launched its i-mode service to enable collaboration with content providers. It also features a billing system based on revenue sharing between Docomo and the listed content providers. Docomo claims about the i-mode service: "*We have been promoting beneficial alliances with a variety of international partners,*

> *including content providers, overseas operators, ISPs, software developers and manufacturers*” [56].

In the chapters below, we assess why and more importantly how such cooperative agreements between network operators and OTT service providers can be established. We present video conferencing and media transcoding as example applications where telecom operators and OTT service providers can cooperate.

6.1 MOTIVATION FOR COOPERATION

Before any cooperative agreements can be reached, it must be clarified what business value do such agreements hold for both the market players and also what benefits should the end user expect. The following factors are seen to be a driving force behind ensuring cooperation between the OTT industry and network operators.

- By interworking with OTT service providers, operators can increase their coverage to parts of the network which do not directly fall under the operator’s domain, such as private LANs or mobile ad-hoc networks. In such cases, a P2P network may be used as an access network to connect a user to the operator’s core. This will allow the subscriber to use his or her home network’s services (which are otherwise buried inside the operator’s “walled-garden”) such as voice-mail while not within direct coverage of the operator’s network. Also any incoming calls to his URI can be routed through to him or her using the overlay P2P network.
- Peers registered in the OTT domain can benefit from services hosted in the operator network on need basis, such as using relay servers or transcoding servers.

- OTT applications involving real time delivery of packets can get better QoS guarantees from the underlying operator network, thus considerably improving the end user experience.
- Allowing users of different domains to share data, messages and converse with each other will create a ubiquitous communication environment. The operator network can also act as a bridge between different OTT networks, which otherwise do not have direct interfaces with each other.
- OTT applications running on client devices generally do not have regard for detailed network topology, and they usually calculate their network routes based on application layer methods such as taking Round Trip Time (RTT) measurements. Such complete isolation of application layer from network layer routing logic can cause un-optimized application layer routing. This leads to congestion or use of network links which are not dimensioned to carry large amounts of traffic. Since network operators dimension their networks, they have first hand capacity planning and network routing information. Unfortunately, operators do not know how the OTT applications will use their network and react to delay or congestion. If the network layer information is made available to the client application, the use of network bandwidth can be greatly optimized [70]. This also calls for a cooperation between the network operators and OTT service providers [71].

6.2 TECHNICAL REQUIREMENTS FOR INTERWORKING BETWEEN OTT AND OPERATOR NETWORKS

The first step before any cooperation between operators and OTT service providers can flourish is to make sure that both domains can communicate and understand each other. The P2P networks generally consist of small user terminals, which are privately connected over some client protocol to other peers. Thus, they construct an overlay network, where they can easily route and exchange information from other peers within the overlay, but seldom communicate with external systems. To integrate these P2P networks with operator networks is not very straight forward. Certain conditions need to be met before messages can start to flow between both domains and applications can work normally [44][45]. We take a look at these requirements for inter-working in the subsections below.

6.2.1 PROXY PEERS

As discussed earlier, most OTT networks are based on P2P overlay concepts. Various peers can communicate with each other using identification and routing information stored locally, centrally or in a distributed fashion inside the overlay [66]. This information allows peer within the overlay to communicate with each other. However, very few of these peers have publically routable IP addresses. This means that a user who is a part of an overlay network cannot be directly accessed from external users unless some method is devised by which packets can be routed to and from this user and external users. One proposed solution to this problem is to use proxy peers [44].

Some peer to peer networks (such as Skype) have the concept of distributing the routing and lookup responsibilities unequally amongst peers [46]. The more dynamic peers, which attach and detach frequently from the overlay, are called nodes, while other peers, which are considerably more stable and have better CPU and network capacity, are called super-nodes. These super-nodes act as the backbone of the overlay to which all other nodes are connected.

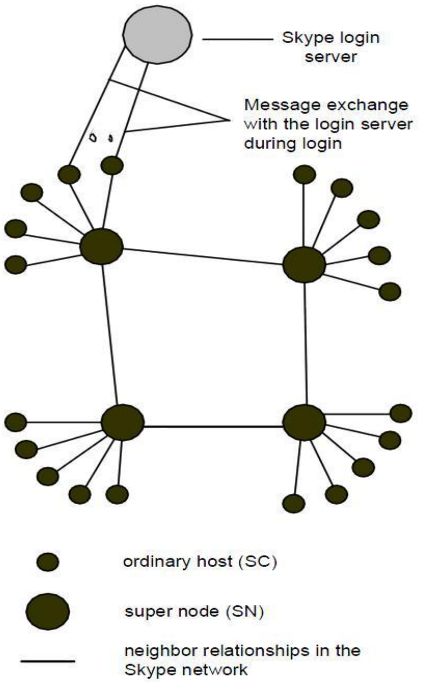
Skype login server
Message exchange with the login server during login
ordinary host (SC)
super node (SN)
neighbor relationships in the Skype network

Figure 6.1: Example Skype network topology [46]

In a similar way, some of these super-nodes, which have publically routable IP addresses, can act as proxy peers for the overlay. The proxy peer will act as a gateway for inter-domain communication, i.e. when a session has to be established between an internal peer and an external peer. The external peer might belong to some other overlay network or some conventional operator based network. Any information that has to come from an external node to a node inside the overlay will pass through the proxy peer, and the same route can be followed in the reverse direction. Ideally a number of proxy peers can work together to make up a proxy layer inside the overlay with different proxy peers distributed in different portions of the network.

The criteria on the basis of which a proxy peer gets chosen or elected from within a P2P network include these:

- Online time i.e. the time for which the node has been online and part of the overlay

- Network bandwidth
- CPU capability
- Publically accessible IP address

Once a proxy peer or collection of proxy peers is chosen, they will also be assigned a Fully Qualified Domain Name (FQDN) in the Domain Name System (DNS), which would represent the overlay network to which they belong. Due to the dynamic nature of overlay networks, the proxy peer might also detach from the network and instead a new proxy peer might need to be nominated. Hence Dynamic DNS should be used to enable the records to be dynamically updated to point to the current proxy peer(s).

Now lets look at an example scenario where a user of an external network (sip:alice@operator.com) wants to establish a session with a user inside an overlay (sip:bob@p2p.org). In this case, Alice follows the procedures for establishing sessions between two distinct network domains. Alice first formulates a session establishment request with the destination as the URI of Bob. This request is routed to the outbound proxy in the “operator.com” network (sip:proxy@operator.com). This proxy server needs to find where to forward the session establishment request. Thus it first follows the DNS procedures to locate the address corresponding to the domain of the destination URI. In effect it queries the DNS for the IP address corresponding to the FQDN of the overlay (p2p.org). The DNS response contains the IP address of one of the proxy peers belonging to the overlay network (the DNS can also load balance between multiple proxy peers). The proxy server in the “operator.com” network then forwards the request to the address it received from the DNS. This request gets routed to the overlay’s proxy peer (sip:proxy-peer@p2p.org). The proxy peer on receipt of this request extracts the identifier of the exact peer (Bob), for which the request is intended. It then uses the overlay client protocol to locate this peer, and once found, it forwards the request to the target peer. The receiving peer then either replies directly to the originating external node if possible, or otherwise, routes the reply through the proxy peer as well. The session establishment call flow is depicted in Figure 6.2 and Figure 6.3. Whether the proxy peer would be stateful or stateless is a design issue and is left out of scope of this work.

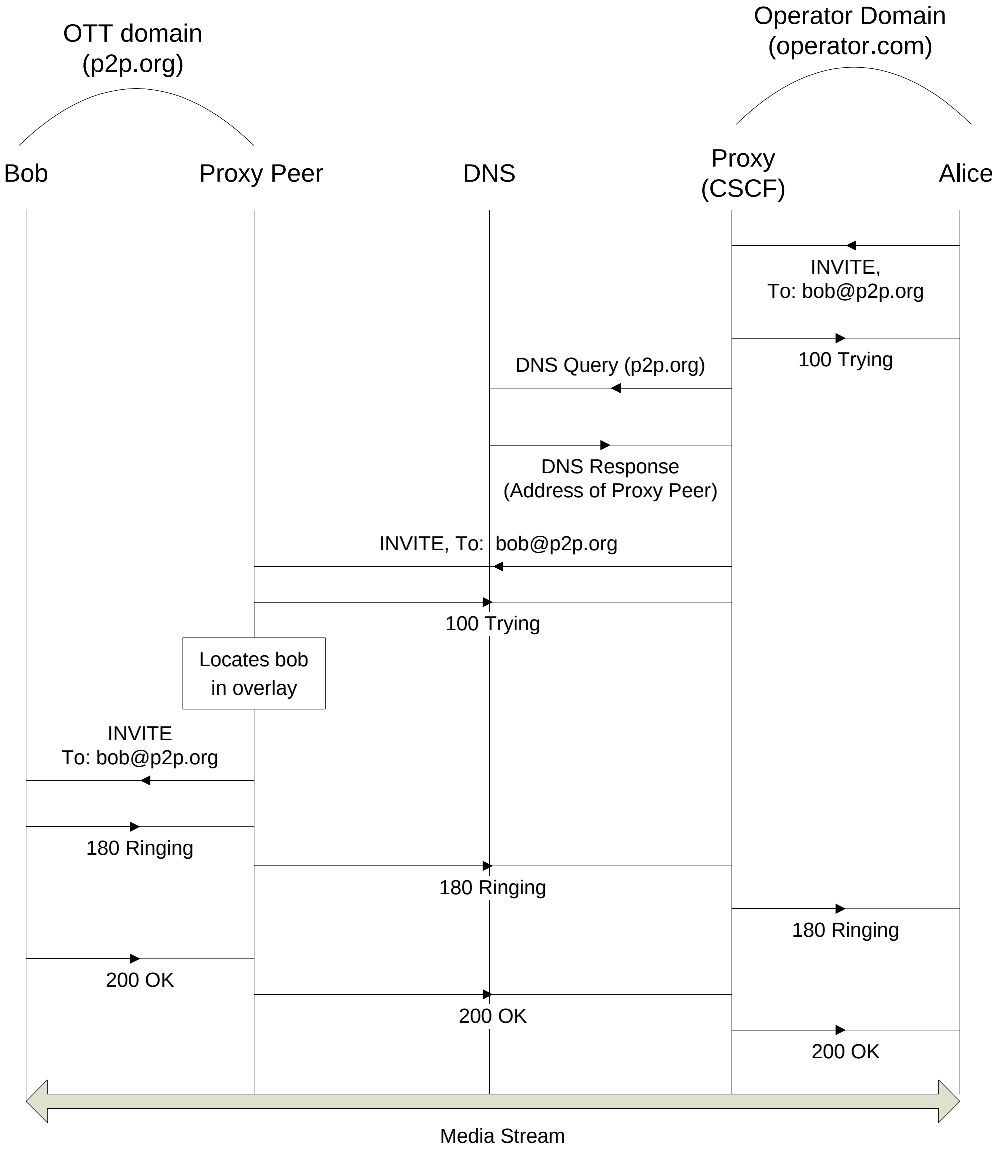

Figure 6.2: Proxy Peer example [49]

Using this methodology, different overlays can be connected with each other. Further more, the P2P overlay would be able to interwork with conventional networks as well. This is especially straight forward for the case where the P2P network nodes use P2PSIP as the signaling protocol, and the conventional network uses SIP as the signaling protocol (such as IMS). In case of incoming requests, location of nodes is made possible by the proxy peer using the overlay client protocol, after which the session proceeds like a normal SIP based session through a SIP proxy.

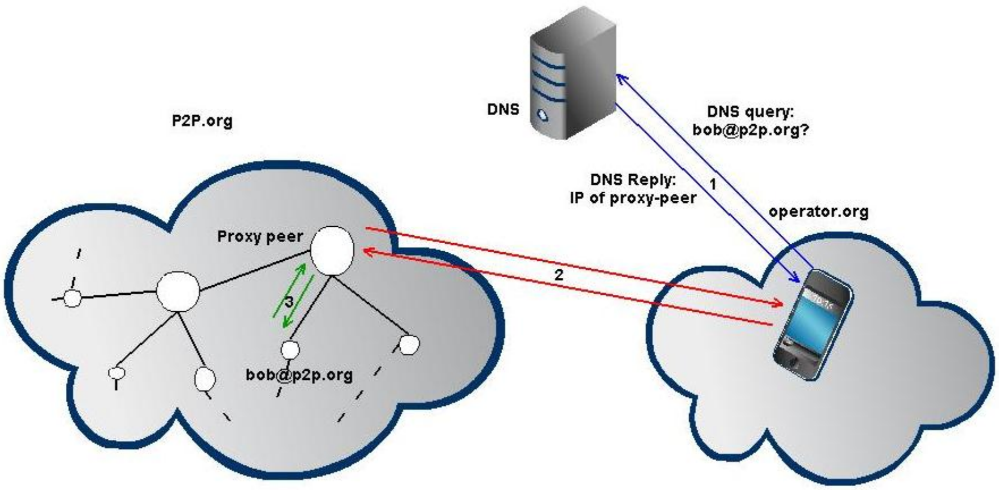

Figure 6.3: Proxy peer receives external call and routes to internal peer

6.2.2 SERVICE DISCOVERY MECHANISMS

P2P networks are based on the idea of pooling together resources from multiple peers and then working in collaboration to achieve the tasks that the immediately local peers might not be able to accomplish. Different peers offering different services are highly distributed inside the overlay. The first step inherent to every P2P network is the resource discovery within the overlay.

The generic mechanism for service discovery is to use the a service discovery protocol to send out queries for a certain service. The service location information may be stored centrally on some directory node(s), it may exist in a distributed fashion within the overlay or it may exist only locally on the nodes offering a specific service. If a directory or registrar node exists in the network, the peers can simply unicast their service queries to this node. This ofcourse is possible provided that the querying node knows about the directory node, otherwise this may involve a service discovery for the directory node itself. In the absence of such a directory node, the service query may be looked up in the overlay's DHT, multicasted or flooded through the network using some other scheme. On receipt of such a query, if a node hosts the required service or knows a location where the service is hosted, it will respond to the querying node. Otherwise it may discard or forward the original query depending upon the protocol specifications. An example of such a service discovery protocol is the Service Location Protocol (SLP) [50].

For the purpose of explanation, we assume that the video mixing and transcoding service is identified by the name 'mcu' in the overlay. A peer wishing to establish a conference or in need of transcoding services will send out the *mcuQuery* request in the network. This request will be routed through the overlay over many peers until some peers, that provide such a service, answer with their identities in the *mcuResponse(peer identifier)* message. The originator of the request will then determine which peer to choose from the available responses according to metrics discussed below. Once chosen, the request originator will then establish a session with this peer.

Now consider the scenario where an *mcuQuery* reaches a proxy peer. Since the overlay's client protocol is not necessarily understood by nodes that lie outside the overlay, this *mcuQuery* will not be forwarded out from the proxy peer. Even if the proxy peer itself does not provide the mcu service, it has information (address/URI) about the media processing server(s) in an external domain (which can be an operator based network). The proxy peer will then reply with the *mcuResponse(URI of the external mcu)* to the originator of the request. Thus the proxy peer acts as an interface between the overlay and the external networks also during the service discovery phase.

If the originator of the request receives multiple response messages to its query (which it will in case of a large enough overlay network), there must be some mechanism and criteria, on the basis of which the best one among the various responses is chosen for the session. To facilitate the choice, metrics such as the ones defined below could be included in the response.

- available CPU cycles
- available network bandwidth
- cost of usage

The above defined scenario assumes that the proxy-peers will have pre-shared knowledge about the addresses of media processing units in external networks. The question still remains, how such information will reach the proxy peers. One of the requirements for enabling such an external use of services hosted inside the conventional operator network is to announce or publish their addresses/URIs in some directory, from where peers can look up this service and then try to contact the specified network hosting this service. There can be multiple models depending upon what level of availability is required. The operator hosting MCU services can provide the address where the MCU nodes can be accessed to the OTT service provider under agreed contracts. The OTT service provider can then float this information in the overlay. Proxy peers can then cache this information if required.

Another model based on more open availability can be to add records corresponding to MCUs and conference servers in the public DNS, for example sip:mcu@ims.org and sip:relay@ims.org. An external user such as a proxy peer from a P2P network querying the DNS for mcu@ims.org would then receive the publically routable address of the network proxy (which is the Call Session Control Function (CSCF) in case of IMS). The user can then send the request for MCU to this address, where the CSCF after checking the network's service access policy will forward it to the available MCU inside the network.

The proxy peer can also cache the DNS records for MCUs temporarily to avoid having to send frequent requests to the DNS. It can periodically verify that the records are up to date with the DNS.

6.2.3 SECURITY POLICIES IN OPERATOR NETWORK

The interworking with external P2P networks and OTT applications would require commercial telecommunication network operators to modify the security policies governing their networks, so that external users can access for example relay servers or MCU servers inside their network. This can also be thought of as the operator opening external network interfaces or Application Programmable Interfaces (APIs) that can be used by third party applications to gain access to the network's internal assets [51]. One way to do this would be to add the URIs of certain known proxy-peers in the operator network firewall's whitelist. In such cases, the local MCU of an operator would act as a back-to-back user agent in terms of SIP. This means that it would accept connections from outside the network and create further connections to other users (inside the network or outside). Certain authentication and authorization schemes to allow users not only registered within the home network but also in external networks to make use of the intelligence hosted within the network have to be devised. Such schemes will be covered in Chapter 7 on security considerations.

6.2.4 SIGNALING GATEWAY

In cases where the two interoperating networks or domains use different signaling protocols, some protocol translation mechanism would need to be used at their adjoining interfaces. Such functionality is usually provided by signaling gateways. Examples of such a signaling protocol conversion can be H.323 to SIP and vice versa.

6.3 CALL CASES

The following call scenarios serve as example cases with all signaling based on SIP. One of the reasons for choosing this protocol here is that it can be implemented both in P2P based OTT networks, in which case it is referred to as P2PSIP, and in the conventional operator networks such as IMS. However, any other signaling protocol, which has the general capability to establish multiparty calls, should work in call scenarios similar to the ones highlighted below.

In the following scenarios, there are two network domains. The OTT network has a domain name of *p2psip.org*, while the operator network is named as *ims.org*. The OTT domain has a proxy peer reachable via Sip URI *sip:proxy@p2psip.com*. Alice (*sip:alice@p2psip.org*) and Bob (*sip:bob@p2psip.org*) are two users in the OTT network. Darth (*sip:darth@p2psip-b.org*) is a user in another OTT network (*p2psip-b.org*) with proxy peer reachable via *sip:proxy@p2psip-b.org*. Carol (*sip:carol@ims.org*) is a user registered in the operator domain. Since it is an IMS network, Carol uses a CSCF node (*sip:cscf@ims.org*) as a proxy server to route her signaling messages to other users or servers. The IMS network also has a conference server (*sip:mcu@ims.org*), which can handle both signaling and media planes for conference sessions.

6.3.1 ESTABLISHING A CONFERENCE

Alice wants to establish a video conference with her friends. Her friends are distributed in different OTT networks (logged into their different client applications), and some are in the operator network.

1. Alice uses her overlay’s service discovery mechanisms to locate an MCU, which can handle the conference’s signaling and media mixing/processing requirements.

As discussed in chapter 6.2.2, at the end of the service discovery phase, Alice receives some replies and chooses to use the MCU hosted inside the operator network (sip:*mcu@ims.org*).

2. Alice dials into the MCU (through the proxy peer in her overlay). The SIP Invite from Alice contains the URIs of all participants she wishes to invite to the conference session. This can be done using the recipient-list procedure as defined in [35]. This invite message will also contain the SDP which defines the media stream parameters from the session initiator that is Alice.

3. The proxy peer in Alice's domain resolves the URI of the MCU which then points to the IMS CSCF node. The proxy peer forwards the Invite request to the CSCF.

4. The CSCF will act as a proxy, and may choose to authenticate Alice. This can be based upon the SIP usage of HTTP digest authentication mechanism [75]. The use and distribution of authentication credentials is discussed further in chapter 6.4.2.

5. Once the authentication is complete, the CSCF forwards the Invite to the MCU. On receipt of the Invite message the MCU will start inviting the requested participants to the conference session. During invitation phase, the media attributes will be negotiated with each participant separately.

6. To invite a user outside its domain, for example Darth, the MCU will send a query to DNS through its local proxy (P-CSCF) to resolve the domain name of *p2psip-b.org*.

In response, the DNS will provide the public address of the *proxy@p2psip-b.org*.

7. The MCU will then send the SIP invite request to the proxy peer which will use the P2P client protocol to locate Darth within the overlay network.

8. Once the proxy peer has located Darth, it would forward the SIP Invite to the him. Otherwise, if Darth can not be located or is offline/disconnected, a 404 (Not Found) response will be sent back from the proxy peer to the MCU.

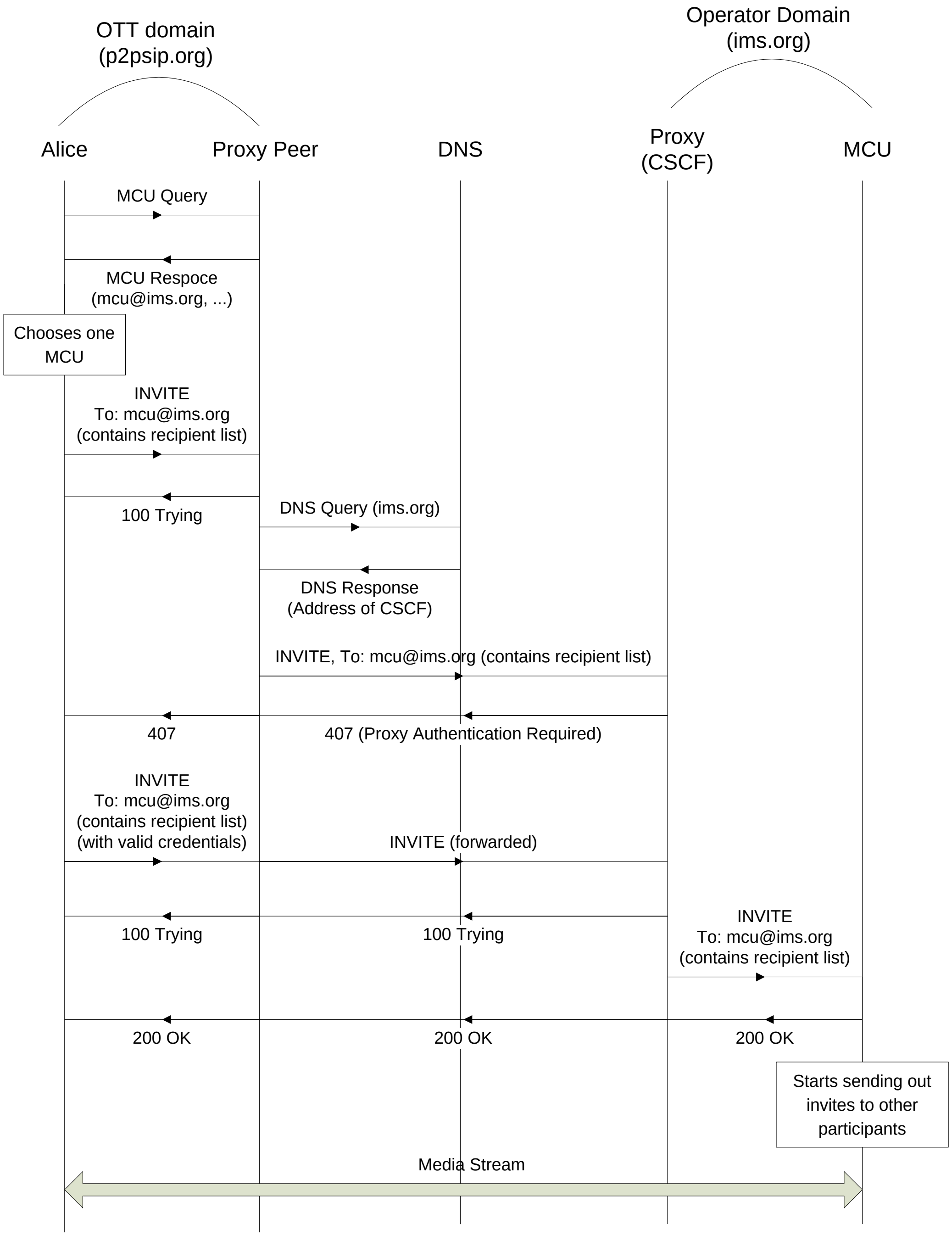

Figure 6.4: OTT domain user establishes a conference session using MCU in the operator domain

This call scenario is depicted in the Figure 6.4 and Figure 6.5. Figure 6.4 shows the first phase of the conference setup, where Alice locates and connects to the operator's MCU. Figure 6.5 shows the second phase, where on the basis of the recipient list included in the INVITE message from Alice, the MCU starts locating and inviting other participants to the conference.

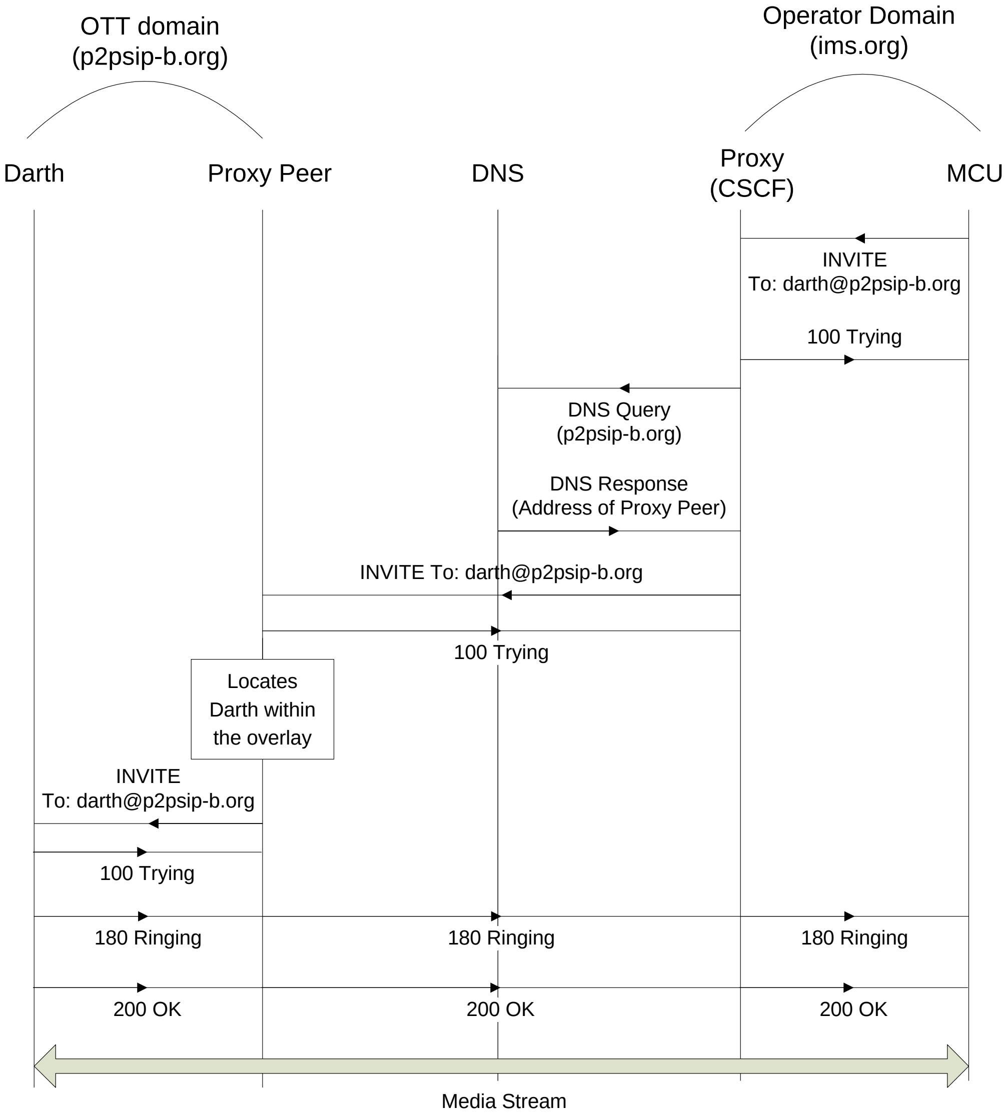

Figure 6.5: MCU in the operator network invites OTT users in a conference

6.3.2 REQUESTING A TRANSCODER IN A POINT TO POINT CALL

In some cases, it might be desired to modify the media stream in some way before it is played back to the user. For example, in cases involving users with hearing impairments, it might be necessary to add text overlay on the video stream containing textual subtitles of the speech [76].

In SIP, there are two prominent models for adding a transcoder [77] into a session, namely third party call control [79] and Conference bridge transcoding model [78]. Here we discuss the latter.

Consider a peer Bob (*sip:bob@p2psip.org*) tries to establish a session with another peer Alice (*sip:alice@p2psip.org*) in a P2P network. The callee (Alice), on receipt of the SDP, discovers that it would like to receive the media in a modified format (due to network, device or other limitations as discussed above). Alice would send out a request into the overlay to locate any node offering the desired transcoding services. Using service discovery mechanisms as described in [57], Alice would send out the *TranscoderQuery(Service)* messages, which will be answered by *TranscoderQueryHit(bandwidth, CPU)* by the peers willing to offer this service. In order to allow discovery of services that lie outside the overlay, the proxy peer will act as an intermediate party. When the *TranscoderQuery(Service)* message reaches the proxy peer, it will look into its cache and reply with a *TranscoderQueryHit* pointing to the transcoder(s) hosted in an external domain, such as an operator network (the cache in proxy peer can be populated gradually with time). Alice would then use her selection algorithm to choose one of the available transcoders. Once Alice has made the decision to add this external transcoder into the call path, she would send back a 302 (Temporarily Moved) response to Bob and include the chosen transcoder’s address in the Contact header field of the response message. Bob would then establish a connection with the transcoder and in the INVITE request include the desired callee’s (Alice) address as the recipient-list. The transcoder can use the same authentication methods as discussed in chapter 6.3.1. Once authentication is complete, the transcoder would send a separate INVITE to Alice.

For the rest of the session, the transcoder will act as a B2BUA (back-to-back user agent). As shown in Figure 6.6, the media stream from Alice would terminate on the transcoder and after required modifications will be forwarded to Bob. Same would happen in the other direction.

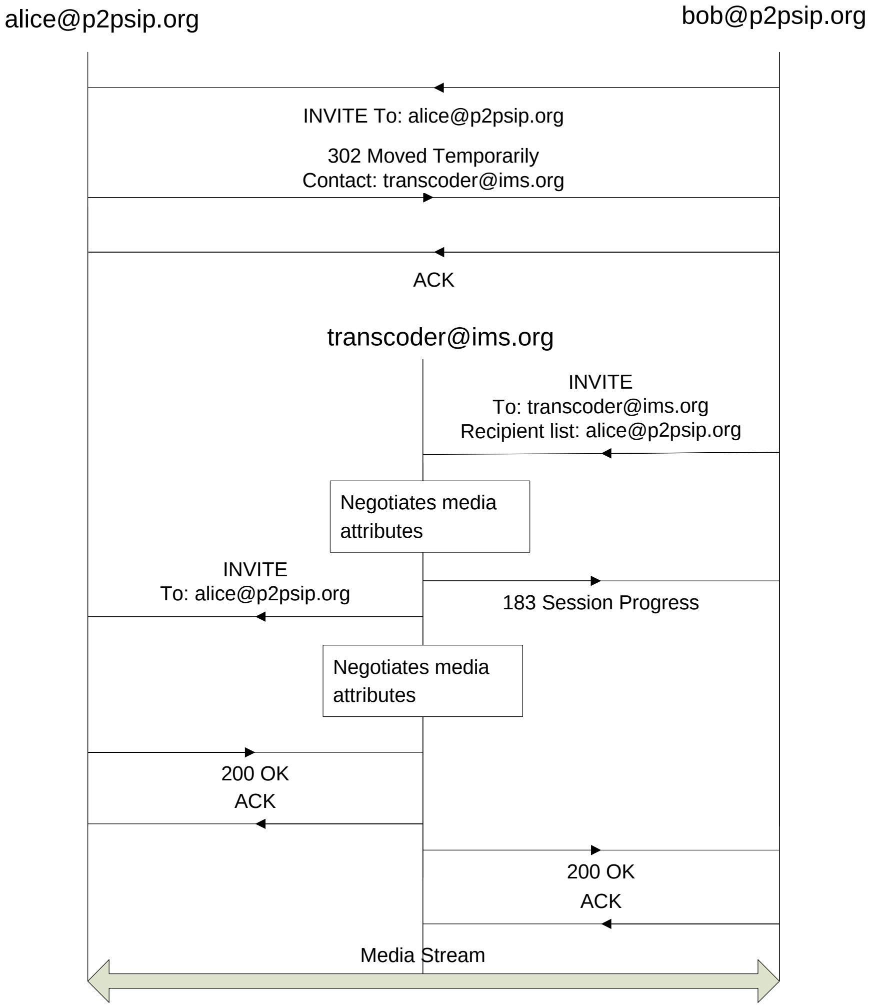

Figure 6.6: Transcoder from an operator domain is used to resolve media incompatibilities in a point to point call between two OTT users

6.3.3 REQUESTING A TRANSCODER IN A MEDIA STREAMING SESSION:

Samanta et. Al. [58] list some of the possible scenarios where transcoding may be required in video streaming sessions. These scenarios include switching client devices during a media session, wireless network bandwidth fluctuations and varying content access patterns.

Media streaming sessions such as mobile TV generally involve many clients connecting simultaneously to the same content source, such as a media streaming server. These clients may have heterogeneous media capabilities, for example in terms of video, some clients may want to view video in a larger resolution while other clients might not be able to display a large resolution video and thus require a down scaled version of the same video content. Thus the content may need to be adapted according to the needs to various clients. Content Distribution Networks (CDNs) typically deal with such client heterogeneity problems in different architectures which can be classified into two broad categories [59]: 1) Static content adaptation, and 2) Dynamic content adaptation. Static adaptation means that different versions of the same content are prepared and placed on the server. When a device requests the content, the version which matches the device requirements most closely is delivered to it. However, practical experience shows that such architectures come with some memory and I/O overhead [59] and given the growing number of different client devices it might not be practically possible to maintain a pre-adapted version of the content for each type of device. Thus in many cases, CDNs have to resort to dynamic adaptation, which means that a transcoder converts the media to the required format at run time when a demand arrives.

Also in mobile communication scenarios, the network conditions are often hard to predict. Due to the mobile nature of the user terminal, the network conditions can vary in terms of bandwidth, end to end delay etc. Mostly such changes are associated with the variance in the radio access network being used by the user terminal. In such cases one possible solution is to utilize a transrating function which can adapt the media stream according to the network conditions on the fly.

Dynamic content adaptation (we refer to it as transcoding from here onwards) can be accomplished at the media server but it may add considerable computational burden on the streaming server because mobile clients often require individually customized transcoding [60].

CDNs based on P2P networking concepts often distribute transcoding and content adaptation responsibilities to other peers in the network which are willing to share their resources for such tasks [60]. However the success of such schemes depends on the availability of resourceful peers in the network, and since transcoding in general and specially video transcoding is a CPU intensive task and random peers in the P2P network might not be willing to offer their resources without proper incentives.

Thus whether the OTT media streaming service is P2P based or server oriented, mobile clients can benefit from transcoding support from the mobile network. In such cases, a transcoding function can be activated on a proxy node which connects the mobile client to the streaming server. The proxy node will receive unmodified content from the server and will adapt it to the requirements of the mobile client at run time. Such a transcoder, if placed at the edge of the radio network and mobile core network will also serve as an ideal location for transrating functions to counter the bandwidth and speed fluctuations in the wireless network. This is depicted in figure 6.7, which shows that mobile clients stream the video from a media server on the internet. This media stream is delivered over the top of the mobile network. The mobile network is only used to enable transport of packets between the server and the client device. However since the video packets are going to traverse through proxies and gateways in the mobile network anyways, these proxy nodes can serve as good candidates for providing transcoding support if required.

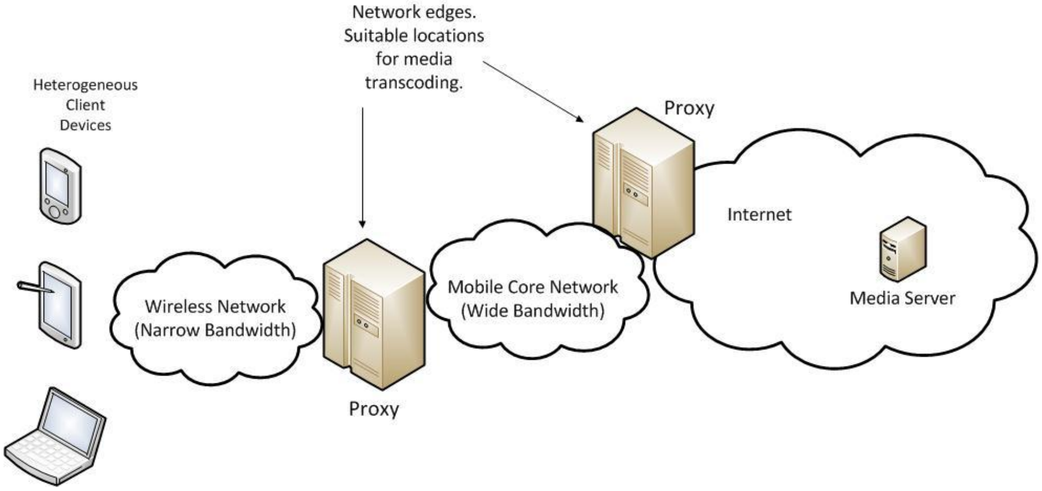

Figure 6.7: Example architecture for providing network hosted transcoding support in media streaming sessions involving mobile clients [61]

Figure 6.8 shows how such transcoding functions can be invoked in a media streaming session between a mobile client and a media streaming server on the internet. We make use of SIP as the signaling protocol for our example. The call flow is based on SIP third party call control functionality [39][79]. Suppose the client wants to receive one media stream from the server. The client initially sends an INVITE request to the server without any SDP offer. This results in the server generating a 200 OK response with an SDP offer (SDP S). The client on discovering that it can not support the media attributes offered by the server, decides to include a transcoder in the session. It then sends an ACK to the server containing 0.0.0.0 as the connection address, thus black holing the media stream [62]. The client will then generate an INVITE request to the transcoder. This INVITE will have an SDP message (SDP S+C) containing the earlier offer from the server and the clients own SDP offer. The transcoder will accept both media streams choosing one of the offered media formats from the server, and one from the client. The transcoder will send back a 200 OK response with an SDP answer (SDP TS+TC). This SDP among other things will also contain the transport address where the transcoder wants to receive media stream from the server. From this SDP, the client will tear away the portion which concerns with the server (SDP TS) and send it to the server in a RE-INVITE message. The server will accept the SDP offer and respond with a 200 OK containing an SDP answer (SDP S). Once the negotiation is complete, the client will send an ACK to the server and the transcoder. After this the media can start to flow from the server to the transcoder, where it will be adapted for the client device and then forwarded to the client.

Since this scenario involves unidirectional media flows, the SDP from client will contain attributes of the type ‘a=recvonly’ and the server’s SDP will contain attribute ‘a=sendonly’ [62].

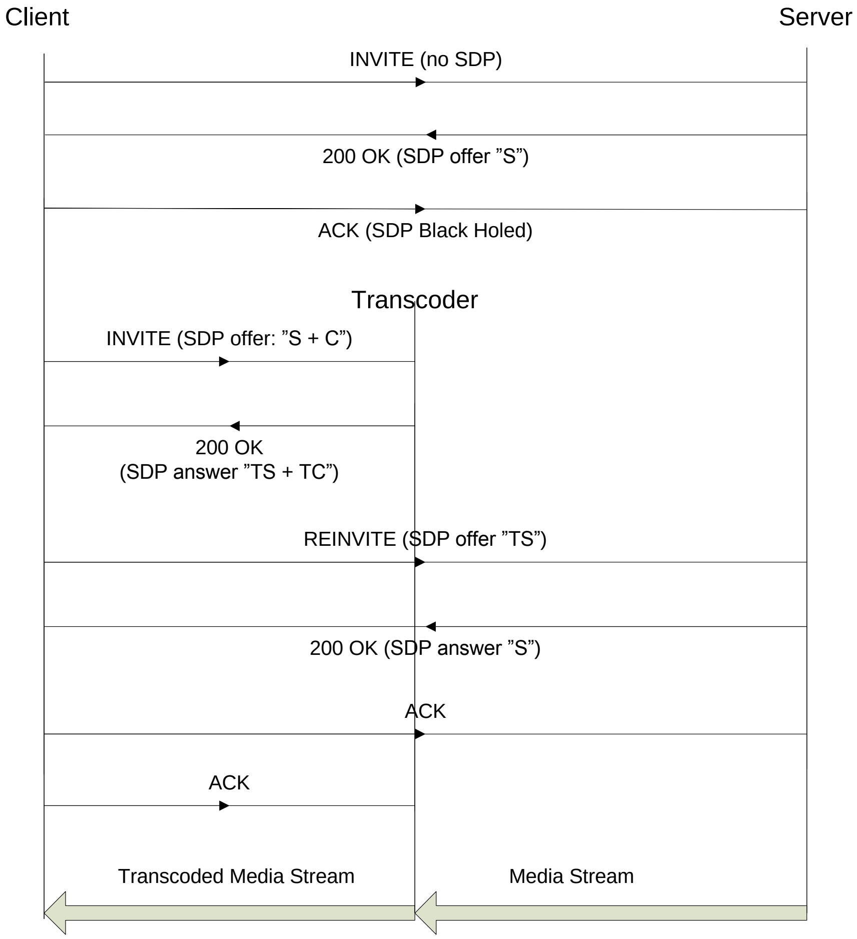

Figure 6.8: Invoking a transcoder in a media streaming session using SIP third party call control functions.

The discussed calls scenarios present only a few frontiers where we can enable cooperation between the operator domains and the OTT services. This cooperation can be extended to cover many other scenarios as well given the proper technical and business ecosystem.

6.4 CONTRACTUAL CHALLENGES IN CASE OF INTER OPERATION BETWEEN OTT AND OPERATOR DOMAINS

Earlier we pointed out the technical feasibility of a cooperation and inter-working of over the top networks with operator based networks for providing better services to end users. However, it must be kept in mind that these two network domains differ greatly in their business models. While the operator networks aim at charging end users for the services they use from the network, the OTT service providers generally provide services free of charge or at a nominal cost to the end user. Thus, enabling inter-working between these two differing domains is not just a technical issue but a business feat as well.

In collaborative agreements and business ventures, both players have to come up with some revenue sharing model which promises sustainable business cooperation. Telecom operators invest heavily in their networks and thus expect every user benefiting from the services hosted in their network to contribute to the generation of revenue in one form or the other. The operator keeps track of the services used by the users registered in the own domain by maintaining databases coupled with the registration servers. For example in GSM networks, this information is kept inside the Service Data Function, which is a part of the Intelligent Network. Charging logic is then applied to the service usage records maintained in such databases.

The existence of standard user registration procedures and pre-defined interfaces allows different operators to interwork. While the user might be roaming in any network, the service usage data is being collected by the visited network against a temporary identity assigned to the visiting user. The visited network then charges the home network for the services used by its user.

OTT services are generally provided at minimal or no cost to the end user. This is so because OTT service providers generally do not invest to maintain elaborate networks. With minimal investment, OTT services do not have to rely on extensive billing of their users to generate profit, but instead employ advertisement based revenue generation models.

The first step before a user can be charged or billed for service usage is to authenticate the user against some valid credentials. It is easy for a telecom network operator to bill its users because every user is registered against a valid identity and is authenticated and authorized by the network. However, OTT applications generally do not have such stringent authentication mechanisms. Thus, to enable inter-operation, there must be mechanisms in place inside the operator domain to authenticate an external user so he or she can be charged for service usage. Technically, there are solutions to such authentication paradigms but whether such solutions fit into the business models or not is a question we explore in the following sub sections.

To understand the potential avenues for authentication of users, we first study the authentication mechanisms employed by operator networks and those adopted by the OTT service providers, then highlight areas where possible inter-operation is possible.

6.4.1 AUTHENTICATION OF USERS IN OPERATOR NETWORKS

The authentication mechanism in most of the operator based networks (GSM, 3G, IMS etc) is tightly coupled with the Subscriber Identity Module (SIM) that is present inside the user equipment. Each SIM contains a unique key, which is also stored in the user’s home network in a server (such as HLR in GSM, HSS in IMS etc) dedicated to maintaining user’s authentication data. When a user turns his device on and wishes to use any of the services from his network or in a visited network, the first thing that takes place is the exchange of key pairs with the home network.

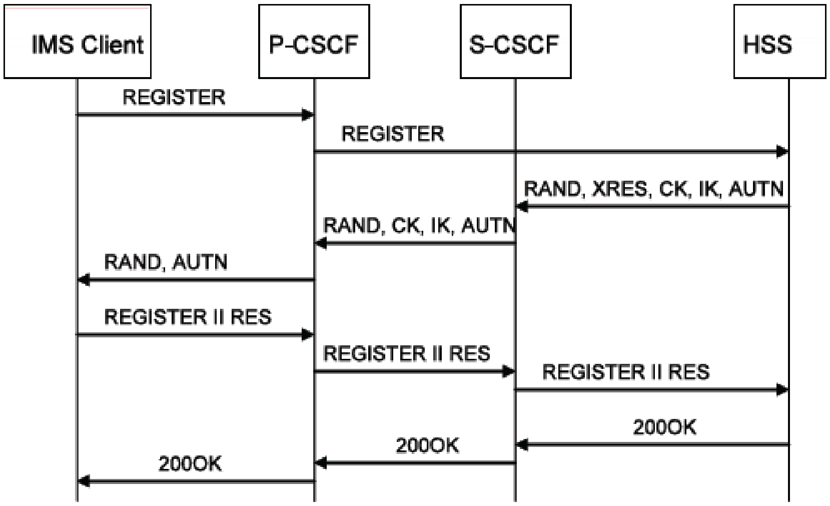

Figure 6.7: IMS ISIM based authentication [72]

With the introduction of IMS, operator networks have seen a shift in their consumer market. Today, operator networks do not just have to cater to telephony but also services like television broadcast, video on demand, internet access etc. Today, operators thrive to provide their users with a ubiquitous communication environment, and that means that devices, which necessarily do not have SIMs will also be accessing services from the network [72].

In such scenarios, methods such as the digest based authentication implemented within the signaling protocol, can be useful where the user or his device uses a username and password pair to authenticate on the network. There are a number of security proposals for SIP [75], including digest based authentication, Transport Layer Security (TLS) and the use of S/MIME. For the purpose of this document, we will take the digest based authentication method to allow users without a SIM card to authenticate and connect to servers hosted inside the operator domain.

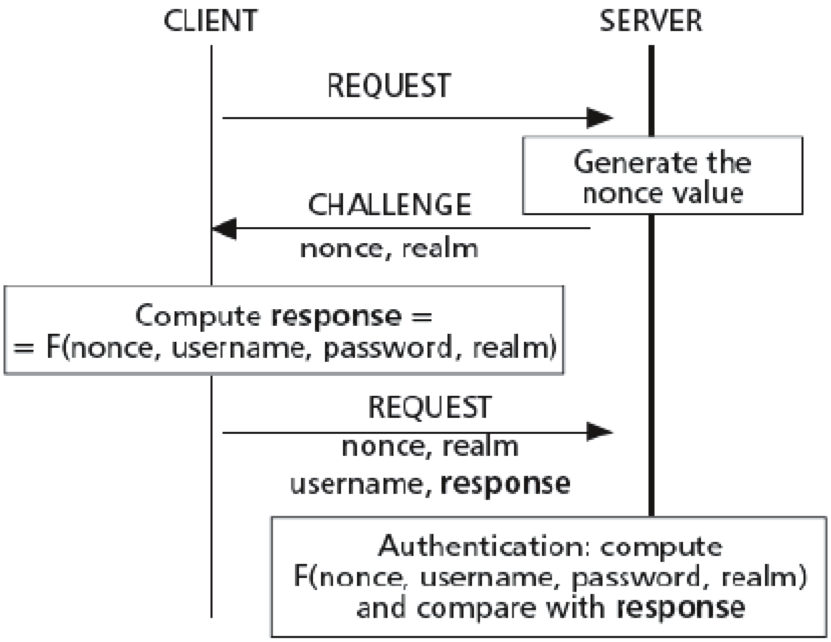

Figure 6.8: HTTP digest based authentication in SIP [73]

However the digest based authentication mechanism is limited by the efficiency of the nonce generation mechanism at the server. Most implementations use a pseudo-random function based on timestamps to seed the nonce generator. This makes the system prone to replay attacks if an attacker is recording packets over the network and replays the transaction from the client to server within the interval when the recorded nonce is still valid at the server [74]. One way to ensure better security and prevent possibility of replay attacks is to use the digest based authentication mechanism over TLS. TLS encrypts all the packets on the transport layer, thus preventing any attacker from listening to the messages being exchanged.

6.4.2 USER AUTHENTICATION MODELS IN OVER THE TOP NETWORKS

OTT networks can be differentiated into two main classes based on the way user profiling and authentication is maintained. These are explained below:

Traditional P2P networks

The traditional peer to peer model means that there is no central server or centralized authority in the network. All user information and identities are kept in the overlay, inside the DHT. Any user can choose his own identity and can add this into the DHT, without the need for any party to verify if the identity really belongs to the user registering it.

This technique works well in P2P file sharing services or media streaming applications, where multiple copies of the same information can be stored by multiple peers. Each peer then in a way contributes to the huge database of files which can be text, audio, video or other programs. Probably the most popular example for such P2P networks ate those based on Bit Torrent [80]. In such cases the DHTs are constructed based on the names/hashes of the files hosted by different peers. When a user queries for a certain file or media stream, the DHT is queried to locate all the possible peers that have a copy of the desired file/stream. In such use cases, the user or client identities are not deemed to be important, as long as they are hosting/contributing the necessary file or piece of information. The DHT is searched against the name of the file, rather than the identity of a specific user hosting it. In order to verify the integrity of the files, their hash is cross checked with a known hash.

P2P based VoIP telephony services can also be built in such a fashion, although it would find less promise to use such services over large networks. The reason being that in telephony services, the user sitting behind the client is the key being queried, and while the integrity of a file can be verified by computing a hash on it, the integrity of a user can not be verified unless there is a trusted authentication mechanism which verifies the user logged into his client.

[81] claims that in the absence of any central authority for identity assertion in a distributed system, malicious nodes can create virtual or fake identities. Thus authentication of participating nodes in truly distributed systems is a problem in general. In P2P based telephony systems, authentication of user identities

(URIs) in the absence of any trusted central authority is an even more difficult challenge to tackle [82].

Hybrid P2P networks

[83] uses the term "hybrid P2P networks" for networks which have centralized indexing such as Napster [84]. In such networks, all the search queries are directed towards the central indexing server while the data or services reside in the P2P cloud. It is due to this mix of centralized and distributed architectures that such networks are referred as being hybrid.

Having a closer look at some of the more widely used conversational P2P services today, we find that they also exhibit hybrid properties. These services are based on P2P concepts where all the call signaling and media streams flow in a peer to peer fashion, but the user authentication and authorization is done in a centralized manner. This is so because in conversational services a lot of emphasis is placed on identifying a user correctly, and if the user profiles are maintained within the DHT, there is a high risk of the information being corrupt or completely wrong. Some of the most widely used conversational OTT services are discussed below briefly for reference.

1. Skype:

Skype is one of the most widely used voice and video client over the internet. It provides the facilities of instant messaging, file sharing, group voice chat, video chat and lately group video chat as well. Although the protocol and operation of Skype is confidential and is not released for the general public, studies on Skype [46] reveal that the Skype registration and authentication process takes place through central servers called as the Skype "login servers". These servers keep the login information of all skype users. The client keeps the login servers' addresses inside the windows registry (when run on the MS Windows platform). When a user wishes to start using the Skype services, she first has to authenticate herself and only then is she allowed to connect to the Skype P2P network. The rest of all mechanisms take place in a P2P fashion, where the network is made up of normal end clients called nodes, and some faster and more reliable peers which are nominated as super nodes. The super nodes maintain a more robust backbone of the P2P network and the rest of the nodes attach to these super-nodes.

2. Google Talk:

Google has also launched its chat client called Google Talk, which supports instant messaging, file sharing, voice chat, and video chat capabilities. To use GoogleTalk, one must have a registered account with Google. The use of the GoogleTalk services begin by logging in to the Google account. This happens by contacting the centrally located Google servers. Once authenticated, the call signaling and media flow can take place in a peer to peer fashion using the Libjingle library [68]. The same peer to peer method is employed when transferring large files.

While within local networks, GoogleTalk operation remains strictly peer to peer after the initial authentication is complete, but in larger networks, with firewall and accessibility issues or where reliable peers can not be found, GoogleTalk can quickly shift from P2P to a more server oriented architecture. Owing to the large number of servers maintained and owned by google in the Internet, it has the capability of readily providing relay-server capabilities to its users if they cannot directly access each other due to the presence of firewalls and/or NATs. In such cases all signaling and media streams flow through the relay server [69]. This differs from other conventional peer to peer systems and also Skype, where the relay services are also provided by other peers (by super nodes in Skype network [46]).

3. Apple FaceTime:

FaceTime [86] is another peer to peer voice and video chat service built by Apple for Apple devices. It also works on the same principle as we discussed above. The authentication is performed in centrally located FaceTime servers, which are maintained by Apple itself. A user who wishes to use FaceTime either has to register his phone number or his email address with the FaceTime service. This establishes a binding between the device’s IP address and the user’s email/phone number. Once the registration and authentication is complete, the session signaling and media flow directly between the end devices in a peer to peer fashion [67].

6.4.3 CHARGING MODELS FOR COOPERATING OTT AND OPERATOR NETWORKS

To be able to charge a user, the operator networks must be able to authenticate OTT users against a valid identity. For both types of authentication models in OTT networks explained above, we need to come up with two corresponding methods to authenticate OTT users in operator networks. Once the operator has a valid identity of the user, he/she can be billed against a suitable and agreed charging metric. In case of a video conference server such as MCU or a transcoding service, the charging metric can be 1) utilized processing cycles, 2) used time period, 3) utilized network bandwidth or a combination of all of these metrics.

Charging users from OTT networks with centrally maintained registration

Because of the centrally administered user accounts and registrations, charging/billing users of hybrid P2P networks is easier as compared to traditional P2P networks discussed in section 6.4.2. The OTT service provider can make billing agreements with the telco operator much like the roaming agreements between different mobile operators. Standard AAA interfaces could be opened on the login servers in the OTT domain, which would allow the telco operator to access user authorization and account data from the OTT domain. Simply put, when a user of the OTT service will request to use a service such as an MCU from the telco operator network, the MCU will respond by asking for authentication credentials from the user. For example in SIP this can be accomplished by sending the 407 “proxy authorization required” response code. In reply, the user’s client application will use the username and password combination that had been used by the user to login initially to the OTT domain. Once these credentials are sent to the operator’s MCU, it will authenticate these from the OTT logon server using the standard AAA protocols such as Diameter which should be implemented at both ends i.e. the operator’s billing/accounting server and the OTT login server. If the logon server responds with a successful response, a temporary identity will be created in the local operators authentication servers, and the user will be allowed to use the desired service. The operator may in turn send billing information to the OTT provider about the service being used and how much should the user be charged.

The OTT service provider in turn can decide whether to charge the users directly from their credit or by generating revenue through advertisements played during the call.

We see that this model, while being a bit more complex in terms of contractual arrangements, connects an OTT user seamlessly to the operator network by re-using the same credentials as used to login to his OTT network through the client application.

Charging of conventional P2P users

Although we see in the section 6.4.2 that conventional P2P networks are generally not preferred for conversational services but for such type of P2P networks which do not have any central authority or service provider based model, the contract with an operator is not straight forward. Instead, the telecom network operator will have to come up with simpler models to charge the users directly. One such model is proposed below.

The operator hosting the MCU service will open an HTTP-based web interface for external users, who can pay online using credit cards and reserve capacity in the MCU for hosting video conferences. Once the reservation is made, the user is issued a username and key pair. The same can also be done by selling vouchers to users in open market, much like credit recharge vouchers sold for mobile pre-paid customers. Once the user tries to establish a connection with the MCU, it would respond with an “authentication required” message. In reply, the user would provide the username and key obtained earlier through the operators web shop or through the voucher purchase. This can be done for example through a GUI that pops up on the users client device. The MCU would verify the credentials with the database in the local network, and once verified, allow the user to use the services for the specified capacity. The proposal is demonstrated in Figure 6.10.

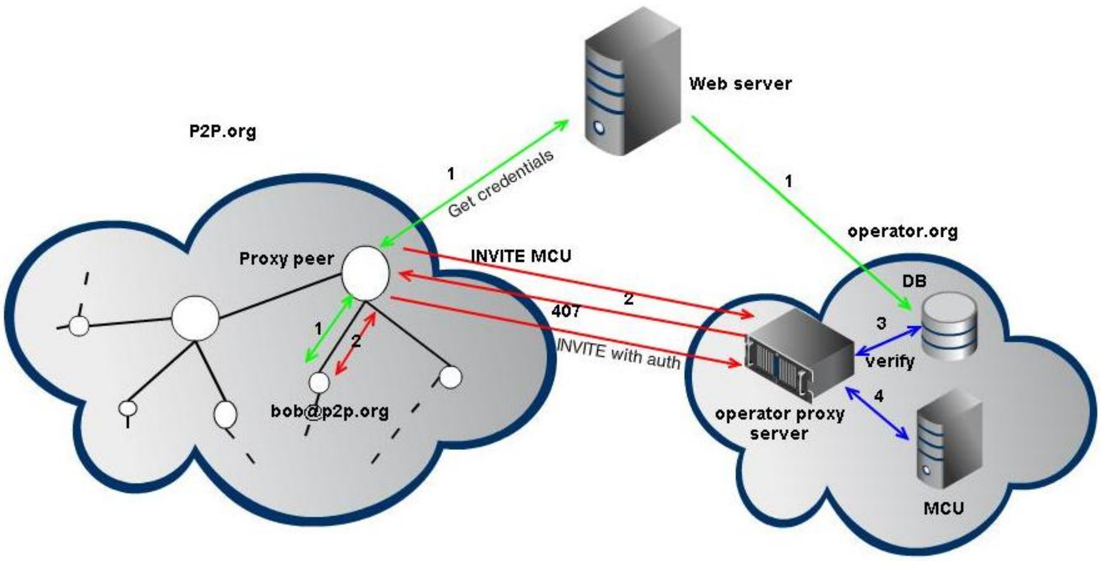

Figure 6.10: Charging model for inter operation between conventional P2P networks and operator networks

6.5 SUMMARY

Cooperation between telecom operators and OTT service providers can bring mutual benefits to both sides. While telecom operators have the promise to get their user base back and still maintain profitability while offering new services, OTT providers have the chance to offer more reliable services hosted inside the network where otherwise users typically only rely on other peers to get services. But such cooperation is not easy to establish and has certain pre-requisites both in technical and business domains.

In terms of technical requirements, the OTT overlay network will need to have some method in place to allow communication with external networks. One way to accomplish this is to have some peers acting as proxy-peers. These peers will serve as interfaces between the OTT network and external (operator based) networks. In addition certain service discovery protocols need to be in place to locate services in external networks. The operator network on the other hand will have to modify its security and access policies to allow external users to access services hosted inside the network. In cases where the inter-working networks use different signaling protocols, some signaling gateway needs to be in place to translate the protocols from one domain to another.

Meeting these technical requirements serves as the bare minimum to allow two different domains to inter-work. They also need to have some mutual billing logic in place so a domain can charge the users of another domain for using its services. We see that many OTT providers offering voice and video communication services today such as Skype, Google Talk and FaceTime have centrally managed user registration and authentication information. If standard AAA interfaces are opened on these central registrar/login servers, these same credentials can be used by an operator network to identify and charge the OTT users for service usage, provided the telco and OTT service provider have an agreement beforehand. On the other hand, more trivial methods can be used to charge users from P2P networks which do not have any centrally managed login information. Distributing one time pass-codes in the form of vouchers can be one possible way to allow users from such tradition P2P networks to use services from the operator networks.

But such inter domain communication of course opens up new security challenges for both the network and the users. We address these security challenges in the next section.

7 SECURITY CONSIDERATIONS

Multimedia conferences in general impose a number of security requirements on the system. Such requirements can be arranged into the domains of authenticating a user before he or she can join the conference, authorize participants of a conference to use certain features of the conference, privacy of the users registered inside the conference as well as maintaining the confidentiality of private information as it flows through the network.

7.1 TRUSTED NETWORK

The need to establish trust about the network becomes even more of a concern when multiple domains or network players are involved. An example of this is our proposed scheme, where media processing can be pushed to a third party network while the conference members themselves belong to a different network domain. The participants of the conference do not necessarily have a direct association or registration with this third party network.

Earlier we discussed how the network can authenticate and authorize a user accessing its services. However, the discussed solutions only cater for one way authentication; only the network authenticates the user. This is also referred to as client authentication. There should be ideally a two way authentication, where the user is also capable of making sure that the network is legitimate and trustworthy (referred to as network authentication or server authentication). This is called bi-directional trust or mutual authentication [85]. This can be done for example by using the well known public key of the operator network. The network is asked to present its signature or certificate, which can then be verified with the certificate authority. Once this verification is complete, the user can be sure that he or she is connecting to the legitimate network and not a spoofed network.

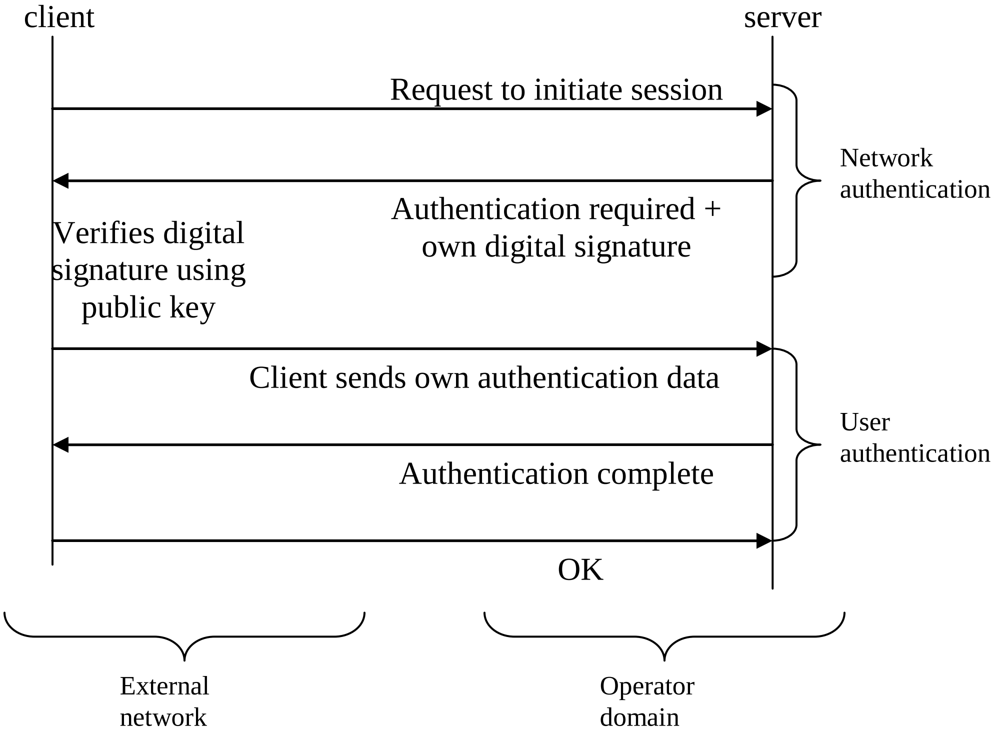
client
server
Request to initiate session
Authentication required + own digital signature
Verifies digital signature using public key
Client sends own authentication data
Authentication complete
OK
Network authentication
User authentication
External network
Operator domain

Figure 7.1: Establishing bi-directional trust

7.2 CONFIDENTIALITY

Maintaining true confidentiality, where no one except the source and the intended destination of the information can read the information, is a challenge when media processing is required from a third party. Confidentiality generally means data encryption to make sure the information being exchanged does not fall into the hands of any adversary listening on the way between the source and destination nodes. This is normally accomplished by encrypting the data being sent at the source, and only the destination node knows the key to decrypt it. This makes it immune to any modification or sniffing by a man in the middle. RTP, which is the most commonly used protocol for realtime media transport, can secure the data using its variant called SRTP [87] which uses encryption to preserve confidentiality of the media streams. ZRTP [88] is another variation of this protocol to accomplish the same objective.

When it comes to a network element such as a mixer or a transcoder, it inevitably needs to decode the incoming streams in order to modify them and then re-encodes them. This process of decoding, modifying and encoding requires the MCU to be able to read the streams in unencrypted form. Thus, we see that end to end confidentiality may be breached incase a third party node is used for media processing.

Consider that a conference session in on going. We call the users attending the conference as conference participants. Apart from the conference participants, many other nodes may also be present in the network. In such a scenario, the media mixing responsibilities can be handled in three different schemes:

7.2.1 MEDIA PROCESSING BY CONFERENCE PARTICIPANTS

In conventional peer to peer networks, enough information about another peer is generally not available to establish complete trust. It is therefore not possible without risking the loss of confidentiality, to delegate the responsibility of media processing to any random peer which is ready to allocate its resources for the task. To avoid this problem, most peer to peer networks restrict delegation of media processing tasks to formal participants of a conference or session. In this way, the media streams do not fall into a hands of any third party but remain within the confines of the conference participants.

This has also been one of the hurdles impeding the usage of coordinated transcoding/mixing in peer to peer networks. The choice of peer for media processing tasks is only thus limited to the members of the current conference.

This scheme maintains the best confidentiality guarantees as the media streams do not need to be propagated to any node other than the direct conference participants.

7.2.2 MEDIA PROCESSING BY NETWORK HOSTED SERVERS

As long as there are capable peers within the conference who can effectively handle media processing tasks, confidentiality and privacy of information is maintained. But what happens when a capable peer is not present inside the conference session.

A second way to handle media mixing tasks in within the conference is to delegate the mixing responsibilities to the network, or specifically to servers administered by trusted network operators. Media processing in the network should not largely act as an obstacle since well known network operators can not risk damaging their reputation in the market by illegally intercepting the media streams and using them for purposes other than the intended media transformations. Most network operators have strict confidentiality policies that govern the use of their subscribers' information. In case of any breach of conduct, the network operator could be held responsible legally. Taking these guarantees into account and assuming that bi-directional trust as discussed in section 7.1 is enforced prior to media flow, it may be safe to some extent to use the network for media processing.

7.2.3 MEDIA PROCESSING BY PEERS OUTSIDE THE CONFERENCE

The more common way for an OTT network to handle media processing demands in the absence of capable peers within the conference is to look for any other capable peer in the overlay which may not be a part of the conference session but is available to lend its resources for media mixing/transcoding. In such a case the privacy or confidentiality is violated as all (unencrypted) media streams will now flow through a peer which was otherwise not present in the conference.

This renders such schemes as the worst in terms of providing confidentiality guarantees due to the fact that any random peer in the overlay network can not be trusted.

One may argue that some level of confidentiality can be achieved if media processing tasks are broken down and distributed to different peers such that no peer has access to the complete information. For example in a video conference, a very trivial way to divide the media processing tasks among multiple nodes would be to make sure that audio streams are mixed by one peer, say peer A, while the video streams are being mixed by another peer, say peer B. B does not have access to the audio streams and likewise A does not have access to the corresponding video streams. More complex algorithms for dividing media processing tasks can be developed. However one can not rule out the possibility of a collusion attack [89]. In our example scenario, if A colludes with B and allows B to access the audio streams, then B gets access to complete media flow. Worse still, one adversary node can masquerade as multiple identities, thus fooling other nodes to believe that they are distributing their media streams to different nodes, while in practice, it's the same node.

Thus in terms of maintaining confidentiality and privacy of information, we rate the media processing schemes in the following order of decreasing preference:

1. By conference participants themselves
2. By servers hosted in a trusted network
3. By random peers in the overlay network willing to lend their resources for media processing.

7.3 USER PRIVACY

Privacy means that a user can hide its name or contact information while participating in a conference. In this case, other conference members will see this user as anonymous. This can be useful in cases where a user would not want to actively participate in a conference or discussion but just to listen to the other participants (such as an online training session or classroom).

User privacy guarantees are as such provided by the signaling protocol in use. SIP provides mechanism for maintaining user privacy as given in [90][91]. This makes privacy a subject of merely choosing the correct signaling protocol rather than asserting any strict dimensioning limitations on the conference architecture.

8 CONCLUSIONS

OTT service providers and network operators have been competing with each other now for a decade to establish their ownership over the services that the end users receive. While the tussle continues, network operators see an ever increasing “revenue-less” traffic on their networks. This traffic utilizing operator’s network bandwidth and capacity does not bring with it an equally growing revenue due to the fact that the services are provided by third party over the top organizations, while the operator network only gets paid for transfer of bits of data from one end to the other. This by proportion is a very small portion of the revenue that the network operators hope to generate with their network resources.

At one end this forces operators to try out dire measures such as blocking off OTT traffic from their networks At the other end, the prospects of decaying traffic and demand for network hosted services is causing operators to cut down on their investments to host high-end servers inside their networks.

Taking this as a background for the research, the first question we aimed to answer was whether there is still a requirement for high-end and reliable servers inside the network, or should the operators just succumb to the threat imposed by the OTT business and let the client devices handle their own loads through the use of OTT Peer to Peer networks. In the past OTT service providers have used the operators' network infrastructure as a bit pipe without harnessing any intelligence or support from the underlying network. While this has worked for the OTT industry to some degree for voice based services over IP, however with the increasing demand for video based applications such as video streaming and video conferencing, the OTT applications can benefit from the support of the underlying network considering that video poses more stringent requirements on bandwidth, CPU and energy. We presented statistics suggesting that for good quality average to large scale video conferences, support from the network is required for video mixing and transcoding. Thus in contrast to IP telephony, when it comes to video based services we see that even if the OTT industry has come up with better and more reliable protocols, end devices especially hand held devices today are not capable to handle such processor and in turn energy requirements themselves. Support from the underlying network can be used to the advantage in such situations. For applications with strict bandwidth requirements, the underlying network can also provide QoS guarantees. We argue that OTT applications can benefit more by using the underlying network as a smart pipe or intelligent pipe rather than a dumb bit pipe.

With this as a motivation, this thesis looked into some scenarios where telco operators and OTT service providers could mutually benefit from each other through collaborative contracts. In this way, the business fear for operators could be turned to a profit by bringing the OTT user base back to the operator networks for using services such as video mixing and transcoding in case of video conferencing. Some scenarios for such a collaboration were evaluated and requirements to practically implement such cooperation were discussed. It is seen that the technical components for bridging the gap between OTT and operator based networks already exist. This is especially straight forward in cases where the two domains use the same signaling protocols such as P2P-SIP in OTT network and SIP in IMS networks. But even in scenarios with differing protocols, certain signaling gateway functions can be deployed at the edge of the networks to allow interworking. We see that the major hurdle obstructing the cooperation between OTT and mobile operator networks is finding the right business models.

Once the OTT service providers realize the benefit in smart pipes over dumb bit pipes, they can tailor their applications to make use of the network hosted intelligence to offer better and more reliable services. The network operators on the other hand realizing the potential in OTT applications can transform their core network offerings from the one-size-fits-all flat rate data plans to multi tiered services customized for different types of users having different requirements. Such a model also bridges the gap between traffic and revenue growth. The operators can open up interfaces to their networks which can be used by third party OTT applications. Such interfaces can range from requesting network QoS, to invoking network hosted media processing services, video recording services, NAT traversal services, voice/video mail services or even accessing the user location data and subscriber profiling data for more tailored service experience.

The business models associated with such partnering OTT and operator based services will be most likely based on revenue sharing, however the exact specifics of how the business ecosystem will evolve are hard to predict. A lot depends on how various players in the industry react to such an evolving ecosystem. How openly will operators allow third party OTT applications to leverage their network hosted intelligence and what terms will govern such revenue sharing agreements? And last but not the least, various telecom operators also tend to compete with each other. Such competition between operators will also play a vital role in determining which operators partner with which OTT service providers.

8.1 FUTURE RESEARCH:

As with every research work, there is room for improvement in this thesis. Many topics within the scope of the broader research area still remain open for further discussion and research.

Specific to video conference, there remains a need to explore in detail the impact of having large scale conference sessions that involve peers from multiple cooperating networks. Video conferences can range from small scale sessions involving cooperation between one operator network and one OTT network to large scale more complex sessions involving many nodes belonging to different networks where some networks might be cooperating while others may be competing at the same time. Depending on the exact dynamics of a conference session, a purely centralized or purely distributed architecture may not scale well or may not be possible at all. Since such scenarios are often hard to analyze due to the unavailability of large scale test networks, network simulations can be performed to see how the network behaves when subjected to varying traffic and when nodes are arranged in different topologies. Network traffic patterns may be analyzed to calculate the most suitable architecture.

Operators also need to clearly identify which interfaces are they willing to open for external users, and how will this impact the overall security and performance of their network assets. In the presence of both external and internal users, a network may have to give higher preference to one class of users, especially when the network is low on resources.

Business collaboration between different players is also an intensely debated topic which was partially covered by this thesis. Even though we believe and motivate that there are good business prospects in cooperation between telecom operators and OTT service providers yet, moving from competition towards cooperative contracts will effect the business ecosystem in the communications industry in general. It is yet to be researched how the market will react to diminishing competition in favor of such cooperation. Although it is quite clear that such cooperation will be based on contracts involving revenue sharing models but the exact pricing and billing strategies for OTT users who use network hosted services and vice versa need to be planned in more detail.

User behavior is also another aspect that will play a vital role in determining whether such cooperation is successful in accomplishing what it promises. For example, whether the general community will react positively to more reliable but comparatively costly services or whether users will generally prefer a less reliable service simply because it is less costly or free. It should also be kept in mind that the user of a service may not be an individual in all cases but may also be one or many enterprises and organizations. More appropriate applications that target a specific user base can be used to statistically analyze user behavior patterns. Cooperative models may need to be tailored according to user expectations.

As part of any future research work, other avenues may also be sought where the underlay and overlay networks can mutually share information and resources to provide better services.

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APPENDIX 1

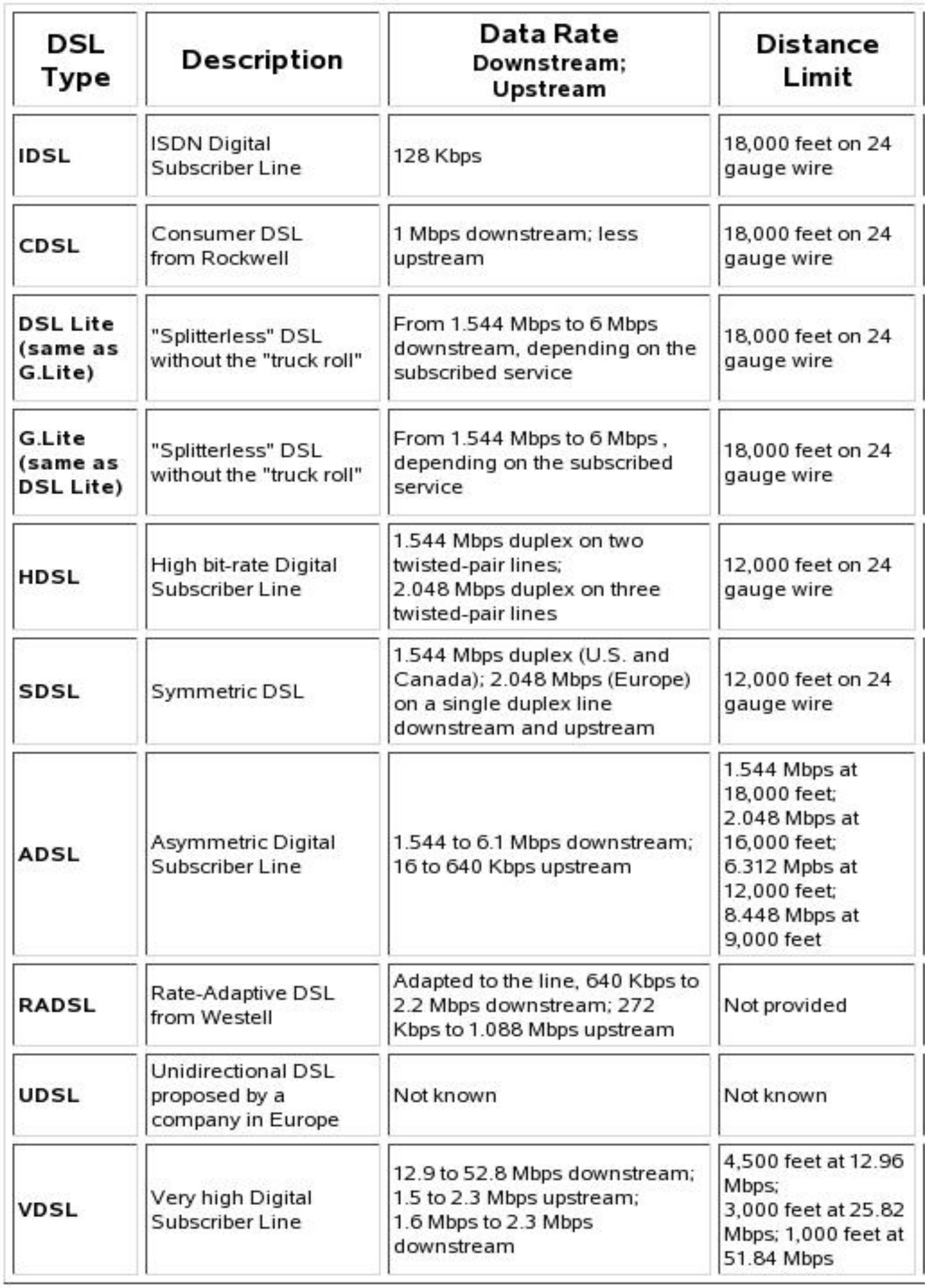
SMART PHONES PROCESSING AND DISPLAY CAPABILITIES

Phone	Display resolution	CPU
Samsung i997 Infuse 4G	480x800	1.2GHz
Apple iPad	768 x 1024	1 GHz
HTC Desire S	480x800	1GHz
Motorola PRO	320x480	1GHz
HTC HD7	480x800	1GHz
Sony Ericsson BRAVIA S004	480x854	1GHz
Acer Stream	480x800	1GHz
Apple iPhone 4 16GB	640x960	1GHz
Samsung I9100 Galaxy S II	480x800	1GHz Dual-core
Nokia E71	320 x 240	369 MHz
Nokia X6	360 x 640	434 MHz
Motorola XT301	240 x 320	528 MHz
HTC Pure	480 x 800	528 MHz
Sony Ericsson XPERIA X8	320 x 480	600 MHz
Nokia N8	360 x 640	680 MHz
Nokia C7	360 x 640	680 MHz
Sony Ericsson Vivaz pro	360 x 640	720 MHz
Acer Liquid E	480x800	768MHz
Samsung Galaxy Ace S5830	320x480	800MHz
Nokia N95	240x320	Dual 332 MHz

Smart phones features [93]

APPENDIX 2

DSL TYPES AND THEIR DATARATES



DSL Type	Description	Data Rate Downstream; Upstream	Distance Limit
IDSL	ISDN Digital Subscriber Line	128 Kbps	18,000 feet on 24 gauge wire
CDSL	Consumer DSL from Rockwell	1 Mbps downstream; less upstream	18,000 feet on 24 gauge wire
DSL Lite (same as G.Lite)	"Splitterless" DSL without the "truck roll"	From 1.544 Mbps to 6 Mbps downstream, depending on the subscribed service	18,000 feet on 24 gauge wire
G.Lite (same as DSL Lite)	"Splitterless" DSL without the "truck roll"	From 1.544 Mbps to 6 Mbps , depending on the subscribed service	18,000 feet on 24 gauge wire
HDSL	High bit-rate Digital Subscriber Line	1.544 Mbps duplex on two twisted-pair lines; 2.048 Mbps duplex on three twisted-pair lines	12,000 feet on 24 gauge wire
SDSL	Symmetric DSL	1.544 Mbps duplex (U.S. and Canada); 2.048 Mbps (Europe) on a single duplex line downstream and upstream	12,000 feet on 24 gauge wire

ADSL	Asymmetric Digital Subscriber Line	1.544 to 6.1 Mbps downstream;
		16 to 640 Kbps upstream
			1.544 Mbps at 18,000 feet;
			2.048 Mbps at 16,000 feet;
			6.312 Mpbs at 12,000 feet;
			8.448 Mbps at 9,000 feet

RADSL	Rate-Adaptive DSL from Westell	Adapted to the line, 640 Kbps to 2.2 Mbps downstream; 272 Kbps to 1.088 Mbps upstream	Not provided
UDSL	Unidirectional DSL proposed by a company in Europe	Not known	Not known

VDSL	Very high Digital Subscriber Line	12.9 to 52.8 Mbps downstream;
		1.5 to 2.3 Mbps upstream;
		1.6 Mbps to 2.3 Mbps downstream
			4,500 feet at 12.96 Mbps;
			3,000 feet at 25.82 Mbps; 1,000 feet at 51.84 Mbps

Summary Table listing various DSL types [92]