Choice - Casting In IP Multimedia Subsystem Based Push to Multimedia Service

MS Research Dissertation

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IN THE NAME OF ALLAH

THE MOST BENIFICIENT

THE MOST MERCIFUL

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Declaration

We hereby declare that this work, neither as a whole nor as a part has been copied out from any source. It is further declared that we have conducted this research and have accomplished this thesis entirely on the basis of our personal efforts and under the sincere guidance of our supervisor Prof. Dr Muhammad Sher and our Co-Supervisor Mr. Zeeshan Shafi Khan. If any part of this project is proved to be copied out from any source or found to be reproduction of some other project, we shall stand by the consequences. No portion of the work presented in his dissertation has been submitted in support of any application for any other degree or qualification of this or any other university or institute of learning.

Nazish Munawar

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Project in Brief

Project Title:	**Choice Casting in IP Multimedia Subsystem Based Push To Multimedia Services**
Undertaken By:	**Nazish Munawar**
Supervised By:	**Prof. Dr Muhammad Sher**
Co-Supervised By:	**Mr. Zeeshan Shafi Khan**
Start Date:	**July, 2009**
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Documentation Tools:	**MS word, EDraw, MS Excel**
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Abbreviation Used

Abbreviations	Acronyms
IMS	IP Multimedia Subsystem
TCP	Transmission Control Protocol
FTP	File Transfer Protocol
SMTP	Simple Mail Transfer Protocol
TELNET	Telecommunication Network
3G UMTS	3rd Generation Universal Mobile Telecom.
PTM	Push To Multimedia
ADSL	Asymmetric Digital Subscriber Line
UTRAN	UMTS Terrestrial Radio Access Network
NGN	Next Generation Network
RAN	Radio Access Network
GPRS	General Packet Radio Service
SGSN	Serving GPRS Support Node
GGSN	Gateway GPRS Support Node
ETSI	European Telecommunication Standard Institute
SIP	Session Initiation Protocol
VB.NET	Visual basic . Net
POC	Push To Talk Over Cellular
PTT	Push To Talk

Abstract

Service Delivery Platform converged the different services and networks to one IP based platform. This convergence allows us to create services independent from the network architecture. In this regard different architectural frameworks have been developed i.e. IP Multimedia Subsystem (IMS). Push to Multimedia (PTM) is one of the main services provided by the IMS based Next Generation Networks (NGN). It is half duplex service that is controlled by a centralized server and allows the users to communicate by using the push technology. Currently PTM service is using uni-multicasting to transfer data. A message sent by one user will be received by all the other members of the group. The sender has no authority send his information to the selected members of the PTM group. This scenario limits the scope of the PTM service in daily life and puts unnecessary load on the network. Moreover in PTM all the members of the group are allowed to send and receive any type of data i.e. audio, video, text (triple play). There is no mechanism to limit the rights of a user in PTM session. This mechanism gives inefficient results, increases the network load and reduces the scalability of the PTM service. We proposed a new casting technique to deliver the information to selected users only. We named it as choice casting. Moreover we also proposed a mechanism to allocate different rights to the members of the same group. This solution will reduce the network load and will result in efficient utilization of resources. Moreover this solution will also increase the applicability of the PTM service in daily life. The proposed solution will also enhance the scalability of the PTM group.

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1. Introduction

Computer networks uses packet switching that divide the message into packets before they sent on the network. These packets uses different route to reach from source to destination. It depends on the type of switching that packet follow different route. Packets contain header information to reach the destination. At the destination packets are reassemble into original message. For fast transmission and to prevent delay each packet has given a maximum length. Messages are divided in to packets before transmission at transport layer. Packet switching technology can transmit data more efficiently. Packet switching is ideal for e-mail messages and web pages.

The directly connected networks face two types of disadvantages. Firstly it limits the number of hosts to be connected. For example in point -to -point link only two host can be attached and 1,024 hosts can be attached in Ethernet specification. Secondly it covers a small geographical area in which a single network can work. For example Ethernet can cover only 2,500 wireless networks, and there is a long point to point link but it does not serve between the two ends of area. There is also a communication problem between indirectly connected hosts. So packet switching enables the communication between indirectly connected hosts. Packets are travel from one host to another host by using packet switch network.

The host can be interconnected with the packet switch device has many input and output. The switch accepts the packet that reaches on its input, read the address and forward the packet to the output. In this way the packet reaches to its destination.

Packet switching networks are of two types. These are

1. Virtual Circuit Packet switching Network
2. Datagram Packet Switching Network

In virtual circuit network hosts are connected from source to destination before data transmission. It is called connection oriented network. During the connection setup phase a route is established between the source and destination nodes. All the packets use this route to reach the destination. The virtual circuit table contain the entry of intermediate route in each switch. Each packet contains header information that identifies the route by using virtual circuit identifier. ATM and frame relay are example of virtual circuit packet switching network.

Datagram Packet Switching Network is connectionless network. Each packet's header contains complete information about the destination. The intermediate nodes read the packet's header and send it to the next hop. Internet which uses IP network protocol is the main implementation of datagram packet switching.

The Protocol which is used for communication across the network using packet switched technologies is referred as TCP/IP. It is a primary protocol of network layer and it has the ability to transmit packets from the source to the destination host. TCP is designed to fit in layered approach and to support multiple network applications. It provides a secure communication between the source hosts to the destination host. It is called a connection oriented protocol. In different networks internet provide the TCP addresses of source and destination. Internet Protocol (IP) is responsible for partitioning or reassembling the TCP segments to transfer across the multiple networks and interconnection gateways. Internet Protocol (IP) contain information about the security arrangement, preference and partitioning of the TCP segments. In this way messages transfer from one end to another end across the multiple networks.

During the communication all the lost, damaged and out of order packets must be recover by TCP. TCP assign unique sequence number to each segment before transmission and need acknowledgement from the receiver. If it does not receive acknowledgement within the time interval, it retransmit the data. Receiver reassembles the segments in to packets by using the sequence number that may be in different sequence, sort out the duplicate packets and remove the damaged packets.

TCP/IP has four layer architecture in which data passes from application layer to the physical layer. At each layer delivery information is attach with the packet's header that guarantee the correct delivery. This delivery information is called data encapsulation. At the receiving end each layer takes off the header and passes it to the upper layer.

Operating System presumes TCP as a module. User has right to use the TCP like a file system. TCP manage the data structure by calling other operating system functions. Device drivers controlled the actual network interface. TCP indirectly communicate with the network device driver that is firstly it communicates with the datagram protocol and then these protocols communicate with the device drivers. The user made a call by using TCP interface for open or close TCP connection, for sending or receiving data and for connection status. This call is similar to other calls that user made on the operating system.

TCP is independent of the physical hardware network that is why TCP can use different kinds of network. It can run on any transmission media for example run on DSL, Ethernet, Dial up line, optical network etc. TCP device can communicate with any other device of the network by using its unique addressing scheme. TCP/IP are able to create open protocol various networks that does not depend upon the operating system and different architecture. Everyone can make products by using TCP/IP specification. TCP uses TELNET, FTP and SMTP applications.

A Next Generation Network (NGN) based on packet switching networks. In this network telecommunication services are able to use multiple broadband and transport technologies enabled by the QOS. In this type of network service related functions does not depend upon the transport technology. In an NGN, transport layer is separately defined and services run on the top of the transport layer. The transport layer is based on IP and services are operated by the session principles to control the requested services. SIP protocol is largely used at the application layer. NGN architecture is defined by European Telecommunication Standard Institute **(ETSI)** based on IMS for internet media-services capability. It is called a 3rd Generation Partnership Project (3GPP).

Next Generation Networks (NGN) aim is to provide services of any type for example to deliver internet, voice and multimedia services to end users, any type of media is used to access the network. Now there is a chance for service providers to separate network infrastructure. Thus we can say the NGN is a union of different type of data communication over IP that is voice, video, data and multimedia. It is a union of fixed, wireless and mobile network. NGN make the user able to access different services available on different networks such as ADSL, UTRAN, WiFi, WiMAX etc. Conferencing, instant messaging, video on demand play an important role in the NGN network.

NGN network was first implemented on 3G UMTS mobile network. It provide multimedia contents on the network. It supports an environment for user mobility testing. 3GPP consider different issues belonging NGN and one of the most important issue is to provide guaranteed end to end QOS. Now a day's 3G UMTS release 5 reduce the gap in the services of heterogeneous IP network such as user identification, authentication and QOS. 3GPP combine all this work on IP Multimedia Subsystem (IMS). Multimedia services like conferencing, video on demand, instant messaging play an important role in the success of NGN network.

To initiate multimedia session such as session invitation and session announcement session description protocol (SDP) is used. SDP describes session description format. It describes the format of initialization parameter of streaming media. Transport protocol does not include in SDP because SDP intentionally use different suitable transport protocol such as session initiation protocol (SIP), Real Time streaming Protocol, Hypertext transfer protocol (HTTP), session announcement protocol. SDP is a general purpose protocol that's why large environments of network and applications can use it than multicast session directories.

This general purpose protocol has larger impact on wider network environments and applications. It does not support media encoding and session contents. To advertise the multimedia conference on internet multicast backbone a session directory tool is used. It communicates with the conference addresses and conference tools. These are the necessary tools for participation. SDP provide sufficient information to enable participation and communicate with the continuation of the session.

SDP use multicasting to send messages. These messages use UDP packets to multicast address and port using SAP (Session announcement protocol) header and text payload. E-mail or WWW are also used to send SDP messages.

1.1 IMS (IP Multimedia Subsystem)

The IP Multimedia Subsystem defines a standard that provide a platform for Voice over IP (VOIP) and multimedia services. 3GPP/3GPP2 (Third Generation Partnership Project) firstly specified IMS standard and now other standard like ETSI/TISPAN specified it. GSM, IMS standard can access WCDMA, CDMA2000, wireless broadband access and WLAN.

IMS serve different functions such as to provide a global system for IP based connectivity with independent access for end users to be able to use internet based protocol with different multimedia services. IMS also provides the multimedia session control in packet switched domain and bring circuit switched functionality in the packet switched domain. IMS provide variety of services such as voice, text, pictures, video or combination of all these in a very controlled environment.

IMS provide advance communication system that will take the communication to the next level. During single communication session of IMS user can easily mix and match IP based services. User can add or drop services on their own choice and can mix voice , video and text. For example if a voice session is started by two people and later on they add video component in the same session.

IMS require a device that provides IP connectivity to access it. Home or visited networks are used to obtain IP connectivity. In the visited network user equipment can obtain IP address and UMTS network contain RAN (Radio Access Network), SGSN (Serving GPRS Support Node), GGSN (Gateway GPRS Support Node) are placed in visited network when user is roaming in the visited network. In the home network user equipment can obtain IP address and UMTS network contain RAN and SGSN are placed in the visited network when a user is roaming in the visited network. In the home network a user obtain all necessary elements and IP connectivity from that network.

IMS ensure to provide end to end quality of service (QoS) while on internet there are chances of delay, out of order packets, lost or discard of packets. In the IMS architecture user equipment use SIP protocol for the session setup and to express QoS requirement.

IMS provide secure communication system by maintaining its own authentication and authorization methods between user equipment and IMS network. IMS offer the equal level of security as GPRS and circuit switched network. Before starting services IMS guarantee authentication. When session is occupied user are capable to demand for privacy

There are two types of charging models which IMS architecture are used. These are offline and online charging models. Offline charging model is like a post paid in which charging information is collected for a whole month and at the end of the month a bill is posted to the customer. It is a traditional method. Online charging model is like a prepaid user account can be checked by the operator before engaging the session and when all credit are consumed operator stop the session.

There are two roaming instances. These are

1. GPRS roaming
2. IMS roaming

In GPRS roaming to access IMS, visited network provide RAN and SGSN and home network provide GGSN and IMS. In IMS roaming visited network provide IP connectivity and remaining IMS functionality provided by the home network. IMS roaming model provide better usage of resources benefits over GPRS roaming. There is inter domain roaming exist between IMS and CS. Both of them have their own services and they cannot interfere to one another domain. For example if user is unable register himself in one domain he can be routed to another domain.

The visited service control model is used in 2G mobile network. In the visited network entities provide services and control traffic when user is roaming. This 2G mobile entity is called visited mobile service switching centre. Internet Engineering Task Force (IETF) protocol support home and visited service control model and help in registration and session flows. 3GPP is a standard that provide service facilities but it is not a service itself. IMS architecture provide a scalable service platform that provide different services like speech, video, multimedia, messaging, file sharing, data transfer, gaming etc.

3GPP make a decision for IMS architecture to use layered approach. IMS signalling network and session management separate the transport layer and services. On the top of IMS signalling network services are run. This layered approach minimizes the dependencies between the layers. As services run independent of the access network, this increase the importance of application layer. IMS act as bridge between the services and application layer that fill the gap between them. These services need different constraint such as bandwidth, latency and processing power in the device. To execute these services properly network contain service logic and access aware control for multimedia services. IMS architecture contains the functionality of multiple accesses that provide a platform for fixed and mobile network.

1.1.1 IMS Entities

IMS entities can be divided in to six categories. These are

1. IMS Session Management
2. Databases
3. Service Function
4. Internetworking Function

5. Support Function
6. Charging

1.1.2 IMS Session Management

During the registration and session establishment call session control function play an important role and maintain SIP routing. Offline charging function are used to send charging data. There are three types of CSCF. These are

1. Proxy CSCF (P-CSCF)
2. Serving CSCF (S-CSCF)
3. Interrogating – CSCF (I-CSCF)

1.1.2.1 Proxy Call Session Control Function

In IMS architecture the first contact point for user is Proxy Call Session Control Function (P-CSCF). All SIP signals will be sent from UE to the P- CSCF. Similarly all the SIP terminating signals in the network is sent from P-CSCF to the UE. P-CSCF performs four unique tasks. These are SIP compression, IP security association, interaction with Policy Decision Function (PDF) and emergency session detection.

The text based SIP protocol contains large number of header and header parameter which increase the message size than the binary coded protocol. If UE want to receive a compress message P-CSCF compress it.

To maintain security P-CSCF is responsible for it. It helps to provide protection and applying integrity for SIP signalling. UE and P-CSCF agree for IPSEC Security association during SIP registration.

To interact with the policy decision Function (PDF) P-CSCF depend on the session and media related information. PDF make IMS able to send IMS charging information to the GPRS network and receive GPRS charging information from the GPRS network.

Although emergency detection of IMS is not fully precise but to detect emergency session IMS network use P-CSCF.

1.1.2.2 Interrogating Call Session Control Function

To subscribe all network connection Interrogating Call Session Control Function is designed as a contact point within the operator's network. Interrogating Call Session Control Function Perform four unique task. These are

1. From Home Subscriber Server I-CSCF obtain the name of next hop that is S-CSCF or application server.
2. Based on received capabilities from the Home Subscriber Server assignment of S-CSCF is take place. The user registration takes place with the network after assignment of S-CSCF or when user unregistered from the network he/she receive a SIP request.
3. S-CSCF or application server contain the information of incoming request.
4. I-CSCF also give the functionality of THIG (Topology Hiding Inter-network Gateway).

1.1.2.3 Serving Call Session Control Function (S-CSCF)

To handle registration processes, making routing decision and maintaining session and storing service profile Serving Call Session Control function play an important role. User registration request will be routed to the S-CSCF and authenticated data download from HSS. S-CSCF verifying the registration and supervising the registration process. At this stage user is capable to start and get IMS services. During registration process S-CSCF obtain service profile from HSS. HSS contain service profile which permanently store the user information. To register user identity in the IMS S-CSCF download the service profile. It is the responsibility of S-CSCF to take routing decision that when UE initiate and finish the session and transaction.

1.1.3 Databases

IMS architecture contains two types of databases. These are

1. Home Subscriber Server (HSS)
2. Subscription Locator Function (SLF)

1.1.3.1 Home Subscriber Server (HSS)

All subscribe and service related data of IMS stored in the HSS that's why it is called main data storage. HSS contain user identities, registration information service triggering information and access parameters as a stored data. There are public and private type of user identities. Home network operator assigned private user identity for registration and authorization purpose while user use public user identity to send communication request with the end user. S-CSCF takes user specification from the HSS and I-CSCF use this information to select suitable S-CSCF for the user. Packet switch and circuit switch domain download Home Location register and authentication from HSS. Packet switch domain is supported by Home locator Register functionality such as SGSN and GGSN. Packet switch domain services are accessed by the subscriber. Circuit Switch domain entities are also supported by HLR and make the subscriber able to access CS domain services and GSM (Global System For mobile Communication) support roaming. In the home network if there are more than one HSS it means that there are more mobile subscriber, more capacity equipment and more network organization.

1.1.3.2 Subscription Locator Function

SLF is the main component of IP multimedia Subsystem (IMS). It is database that contain the subscribe data in reaction to queries from I-CSCF or Application server. SLF enable I-CSCF and S-CSCF and act as resolution mechanism. When the network operator contain multiple and separately addressable HSS then Application Server are used to find the addresses of HSS.

1.1.3.3 Service Function

Multimedia Resource Function Controller (MRFC), Multimedia Resource Function Processor (MRFP) and Application Server are three service related functions in the IMS. Application Server is a part of IMS function. In the IMS architecture Application Server entities provide multimedia services such as presence and push to talk over cellular. Application server process incoming SIP session received from IMS. It initiates the SIP request. Charging function receive account information due to Application server. In the IMS architecture MRFC and MRFP jointly supply bearer related services. MRFC is used to handle SIP communication and MRFP. On the MRFC request and instruction MRFP provide user plane resources. Following are the functions performed by the MRFP.

1. Responsible for media mixing stream.
2. It provide a source for media stream.
3. It process media stream such as media analysis and audio transcoding.

1.1.3.4 Internetworking Function

To exchange media and signalling between IMS and CS CN Internetworking functions are used. When breakout occur in CS domain then Breakout Gateway Control function (BGCF) receive SIP session request from the S-CSCF. There is a selection process that check either the breakout occur in the network where BGCF located or it occurs in the different network. If the breakout occur in the same network then Media Gateway Control functions (MGCF) are used to handle session. If breakout occur in different network then BGCF send session to another BGCF of the particular network.

MGCF start the conversion of protocol between SIP protocol and ISDN user part when it receive SIP session request. This conversion request routed through Signalling Gateway to the CS CN. Signalling gateway convert IP based transport of Signalling and Signalling System No 7 protocol in to SIP protocol. At the transport layer conversion of protocol is performed by the Signalling Gateway. MGCF is also control all the calls that came from CS user to IMS user. It performs protocol conversion and for session termination it send SIP session request to the I-CSCF.

1.1.3.5 Support Function

PDF stands for policy decision function. All the session and media related information that is obtained from P-CSCF is used to make policy decision. SIP and SDP are used to exchange end to end messages during session establishment in the IMS. UEs agree with a set of media characteristics during the exchange of message. If operator want to implement SBLP then SDP information is send to the PDF with the indication of originator received by the P-CSCF. Authorized token is allocated by the PDF and P-CSCF will send this token to the UE. To authorize IP QoS parameter, SDP parameters are used to compare with the authorize IP flow of the chosen media component and PDF notes that transfer that comparison to the access network. GGSN use PDF context activation information or authorization to query about the authorization information from PDF. PDF perform comparison between received binding information and stored authorization information and then send back the authorization decision. PDF start communication with the media authorization that is decided by the GGSN if it receives correct binding information.

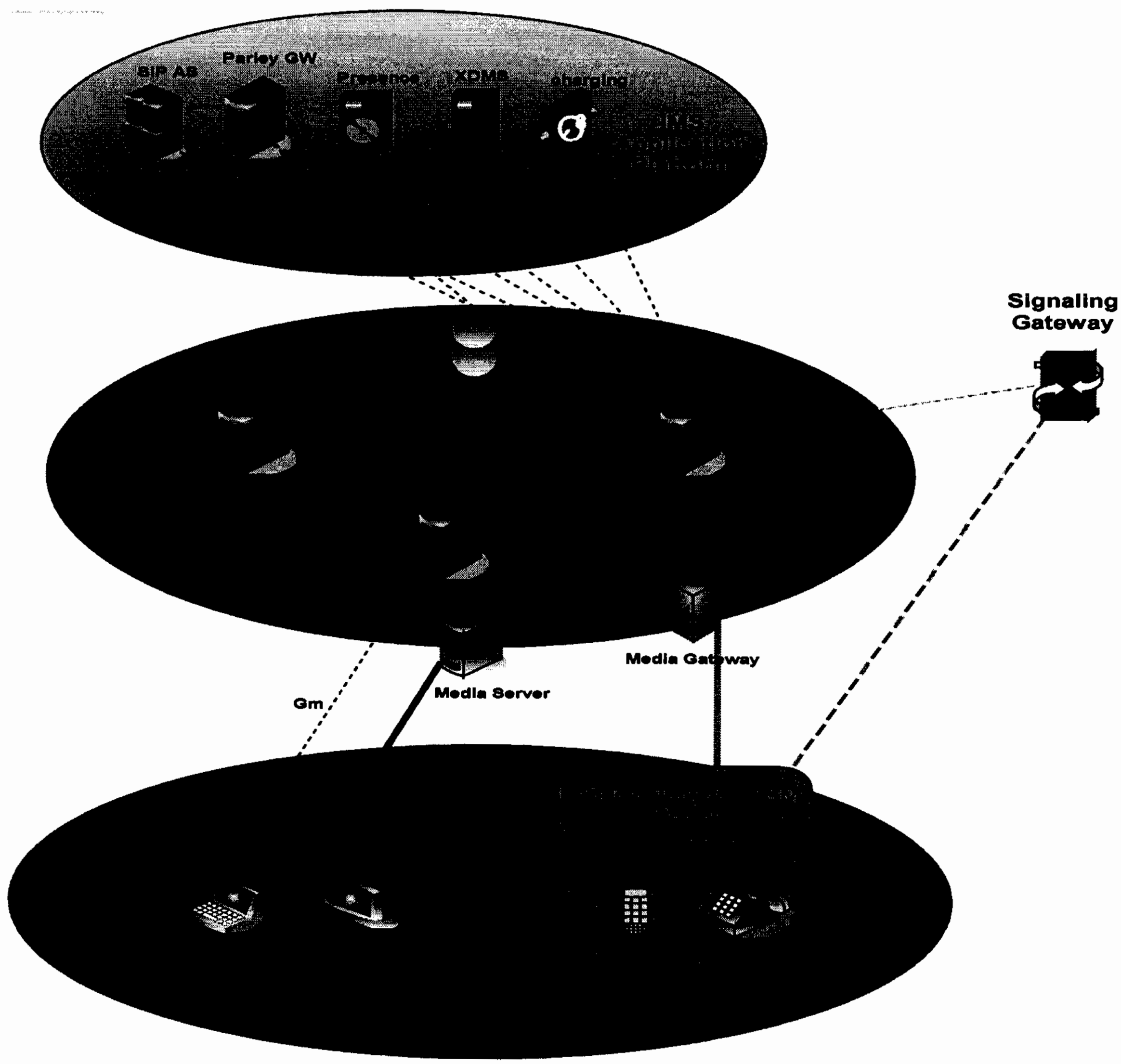


Figure 1: IMS Architecture

1.2 IMS Services

The upper most layers of the IMS consist of application plane. Application plane contains various types of application server those are providing different types of services to the users. Few main services are:

1.2.1 Conferencing

A conversation between multiple participants is called conferencing. Conferencing is not restricted to only audio but video and text conferencing is also used. Conferencing is popular due to its face to face meeting ability. Conferencing has different types. These are.

1. Loosely Coupled Conference
2. Fully Distributed Multiparty Conference
3. Tightly Coupled conference

In tightly coupled conference participants are connected with the central point of control. Varity of services are provided by the central point such as media mixing, transcending and list of notification. Session Initiation Protocol User Agent is a central point and it is also called Focus. Conference uniform resource identifier addresses the central point and there exist signalling dialogue setup between each participant. There are several set of rules associated with conference. These rules increase the life of conference. These rules define the availability of conference, responsibilities and policies that describe who is allowed to use these rules. Media policy and mixing characteristics is a policies of conference.

To create conference different ways are used. One way is used to create adhoc conference by using SIP method. Adhoc conference is without scheduling and it has short live. Another way is to create scheduled conference using conference control protocol. User can create and manipulate conference and its policies due to conference control protocol. Conference's membership is also managed by this protocol. Adhoc conference use SIP protocol to allocate and publish conference factory URI. SIP event framework provides a conference state event package. This event package allows the user to learn and send notification who can join or left the conference. It also describes the status of participant in the conference. To subscribe the user SIP SUBSCRIBE request is send to the conference URI. It identifies the conference. User status provides two types of information. It describes the current status of user in conference and also describes the method that which participant entered or leaves the conference. Connected, disconnected or on hold describe one of the following activity of the status.

1.2.2 Messaging

Messaging is used to send message from one person to another person. In messaging different types of data can be send and that can be delivered through different ways. Multimedia as well as text messaging is used and these messaging can be delivered like instant messaging or like email in the mailbox. There are three types of messaging.

1. Immediate Messaging
2. Session Based Messaging
3. Deferred Delivery Messaging

Every type has its own characteristics. All forms of messaging have ability to send message from A to B.

1.2.2.1 Immediate Messaging

In the IMS architecture immediate messaging is the familiar form of instant messaging. This type of messaging use session initiation protocol (SIP). In this framework message request is generated by the user equipment and message is equipped with desired contents such as text, images, sound and URI request with recipient address. Immediate message is routed through IMS architecture to find its way to the recipient UE. If IMS subscriber is offline or unregistered then message is routed to the Application Server (AS). The message is stored in the AS and when the user registers himself it will deliver to the user. In the IMS a single message can be send to a number of recipients using a list of server extension. An alias is created by the IMS user and Public Service Identifier (PSI) is formed using SIP address. Alias contain SIP URIs of the intended membership and send message to the PSI related to the list, list server receive the request. List server is actually an Application server which interprets the message and new request for each member of the list is generated.

1.2.2.2 Session Based Messaging

Session based messaging is similar to the internet. This type of messaging contain media component which have short message that take part in a session. The session of messaging started when user want to start and stop when user want to close the session. Session setup is formed between participant using SIP and SDP that flow the media from peer to peer. Message Session Relay protocol is actual protocol that send message with in the session. On the top of transmission Control Protocol MSRP layer is present that forward Multipurpose Internet Mail Extension encapsulated data. Messages are in random size and these complete messages can be send in the form of small chunks that are reassembled at the destination.

1.2.2.3 Deferred Delivery Messaging

Multimedia messaging service is known as deferred delivery messaging. Now third generation partnership project (3GPP) merge with MMS. It means that in IMS architecture MMS used deferred delivery mode.

1.2.3 Presence

Presence is a new way of telephone to function. It is a best way of communication that enhance messaging and other services. It provides good business opportunities to operators and service providers. To share information presence provides a user profile that is used to represent the user status and other control services. User status means that user personal or device status, user location, terminal capabilities etc. Presence services like audio, video, messaging and gaming are used by the user for communication with others. Presence provides user personal information. It provides the list that which user initiates the communication, who is available and wants to communicate. It also provides information about those users who are available and willing to communicate. Presence enhances the security and private issues. Presence user use SIP protocol to control their presence information.

The basic idea behind presence services is "Publish receive" presence information. Operators fetch these attributes related to different infrastructure from communication network. User uses these attributes by subscribing the operators. Presence information use new services in committed area of application. It provides a good chance to different companies who want to spread out their services. It also facilitate the user by providing advertising and information sharing channel. It is efficiently provide a way of working in team for sharing information among the team members such as future plans and meeting location.

Presence is a combination of two things that is it provide my availability to others and others availability to me. Event packages are created to extend the SIP protocol for presence. These events are subscribed by the presence token. Presentity and watcher are used to define the subscriber and notifier. Presentity means to provide information about presence to the presence services. Watcher means that it provides requested information of resources. Presence defines two SIP entities. These are

1. Presence Agent (PA) that is used to store subscribes data and produce notification.
2. Presence User Agent (PUA) is used for Presentity that uploads and publish presence information.

We can obtain user's presence information from IMS which contain multiple entities such as PUA sited in foreign network, at terminal or located as an entity in the network. Presence sever, watcher presence proxy, Presentity presence proxy and PUA are the entities of IMS network.

1.2.4 Push to Talk Over Cellular

Push to Talk over Cellular (POC) is a one to one and one to much communication system. It is half duplex that it provides one way communication in which one person talk and other listen. POC provides uni multi-casting communication. Each client sends data to the push to talk over application server and server send traffic to all the recipients. In core and access network multicasting is not performed and radio network perform mobility management. So POC works behind the fixed and cellular networks. SIP protocol is used to control POC session and signalling and Real Time Transport Protocol (RTP) is used to control voice traffic. Radio resources and cellular access is more efficiently used by POC user. One way network resources are reserved for talk burst duration.

1.2.4.1 POC Architecture

POC architecture is based on POC client, POC application server and XML document management server. POC server also used presence service that shows the presence of POC user. POC. Talk burst control and session control is handling by POC application server. IMS service control reference point is used to connect the POC server with IMS. In IMS architecture authentication, session routing and charging is controlled by SIP protocol. POC Application server provides services to the user in the IMS architecture. Session setup procedure, its policies and group user information are defined by the POC application server. The server handle setup POC session, enforce policy, information about group user, media distribution and also control the talk burst. POC server control the services associated with the control plane and user plane traffic. So server uses ISC and Mb IMS reference points.

Two different POC server roles are defined by OMA these are

1. Participating POC Function
2. Controlling POC Function.

During the session setup there is one controlling function and two participating function in the POC server. POC client firstly send SIP signal to the participating POC function and then these signals are sends towards the controlling function. There is a direct media and signalling connection of POC client to the controlling function. On the UE POC client is act as a functional entity that use POC features to register itself on the IMS.

1.2.4.2 POC Features

For group communication different communication models are used. The only difference between these models is related to the group policy and session setup. There are two groups for POC communication. These are

1. Dial Out Group
2. Join –in Group

To participate in group communication user invites group of users in dial out group communication. Incoming session indication is received by the invited user and these invited user join the session either manually or automatically. Dial out group communication is may be prearranged or ad hoc POC group.

In pre-arranged group communication individual member of the same prearranged group establish the session by inviting the other members of the group to join the session. To start the communication firstly prearranged group member accept the invitation. Controlling POC server granted the media permission for the prearranged group initiator. Only predefined members are allowed to participate in prearranged group.

Ad hoc POC group also follow some rules as well. To create ad hoc group one or more users are invited by the POC user for POC session. Only invited users are allowed to use the ad hoc POC session. Controlling POC server use local policy that allows initiator of the ad hoc POC group to add more users.

The participant themselves join the communication session in Join –in group communication. This type of communication makes the user able to fully control the participating group. After joining the group they receive all the traffic. Join in POC group is further subdivided into unrestricted chat POC group and restricted chat POC group. Unrestricted chat POC group is open to all those people who knows identification group. On chat room or group portal, group identification is found. For open discussion on general and specific topic unrestricted chat group is suitable. In restricted chat group pre defined users are allowed to access the group. To join the group user needs group identification and needs a right to join the group. Restricted cha group is best suitable for business users because it provide secure communication.

To establish POC session two different model exists. These are on demand and pre-established session. SIP method is used in on demand model. Media parameters are negotiating after receiving POC session request made by user. Participating POC function receive a session establishment request from user in pre-established model and media parameters negotiate in advance by making request of POC session for POC user. In this model POC client can be invited by receiving POC session without negotiating in advance with media parameters again.

POC has defined two different answering mode. Auto answer and manual answer mode. In auto answer mode incoming POC session request is accepted by the POC device without any action from the user that is immediately played media stream. In manual answer mode incoming POC session request is accepted by the user after confirming acceptance of POC Session by the POC device. POC server receives signals from the answer mode performed by controlling POC. SIP protocol is used to send confirmed or unconfirmed indication to the originating user function. User receive confirmed indication when manual answer mode is used by the terminating POC device and new POC session is accepted by sending SIP 200 ok. Terminating participating POC server function generates unconfirmed indication.

POC uses Instant personal alert message when caller does not able to reach the recipient. Instant alert messaging uses SIP METHOD. There are two types of instant alert that is SIP instant message and POC Instant Personal alert. The only difference is that POC instant alert creates automatic operation at the POC client. For this purpose accept contact header is included in the originating POC client tag feature. Instant personal Alert Barring is activated if user does not want to receive personal alert. To activate this service participating POC server block the delivery and send SIP 480 error message to the user.

Participant information and their status is provided to the POC user request. User can also be informed that who has joined or leave the conference through notification. Conference state can be subscribed by the SIP SUBSCRIBE request that recognize the controlling POC server.

1.2.4.3 User Plane

There are three components in user plane layer. These are media flow, Talk Burst Control and quality feedback.

1.2.4.4 Talk Burst

In Talk Burst control system controlling POC server receive single burst of media from participant. The server multicast the burst to all participants in the session. This media flow is received by the participant and provide sound system and loudspeaker play this sound. POC session contain a number of participants on a single floor and floor functions are control by the moderator that permit who is allowed to transmit media at any given instant. Floor request may be queued or granted in a succession. To control the floor Talk Burst Control Protocol (TBCP) is used. This protocol is used to accept, reject and release session. Controlling POC function contain TBCP server.

1.2.4.5 Quality Feedback

Feedback report processed by the RTCP can be optionally created and send to the POC server and client. Reception quality feedback report can be generated at Receiver end. There are two types of reports sender report (SR) and Receiver Report. The difference between these report is that the SR contain the data send information. These two reports also contain the information received media packets like total number of RTP packets, total octet received and time of the media packets transmission. In the session server act as RTP translator that depend on the RTCP report between participants.

1.2.5 Push to Multimedia

Multimedia is real time-critical computing application. Video conferencing, real-time image communication is the key for success of network multimedia system. Since the emergence of the Internet and World Wide Web, multimedia applications have greater impact on manufacturing industry. PTT in a cellular network are extended by using media capabilities like Instant Messaging (IM), real-time video transfer and file transfer and is known as Push to Multimedia (PTM) [14]. PTM introduce Packet switched functionality in a cellular network that brings good opportunity for business. The basic principle of communication is very simple - just push a button and communicate to users from your list. It uses instant messaging of client in whom call can be started from both individual and talk group and the buddy list. Preference functionality shows the online, busy, and away users. It is 3rd generation UMTS/WCDMA network technology for Push to Multimedia on VoIP. At the IMS Playground at Fraunhofer FOKUS the service is deployed for the IP Multimedia Subsystem (IMS) using the OSA/Parlay API, which enables the service as a network independent solution, utilising the abstraction layer of the Parlay interfaces for different target networks Signalling is done by session Initiation protocol (SIP). It has three components [14].

1. PTM Application Server
2. Media Server
3. PTM client

Application server act as a host of Parlay service, implementing the logic needed. Media server takes care of media stream. PTM client is the "FOKUS Mobile SIP Client", which is also capable of SIP SIMPLE Presence functionality.

Outline of the Thesis:

The chapter 1 is the introduction of this thesis, which described overview of PTM using IMS architecture. Chapter 2 belongs to the Literature Review, which describe the prior work of the researchers who have contributed in this domain. Chapter 3 is associated with research objectives and problem definition, which describes the aim and scope of the research as well as problem of the research. This chapter explain all the requirements to meet with this problem on which focus research is being studied. Chapter 4 is related to proposed solution and methodology, Chapter 5 is related to results, Chapter 6 describes the conclusions, in this chapter we conclude our work with some future work. Chapter 7 is References; in this chapter links of list is provided which have been confer with while conducting this research

2. Literature Survey

Literature Review

In Research field there is no boundaries and limit, everyone can participate in this field who theorist and is inventive. Pioneers work cannot be ignored and neglected by the emergence of new researches. Similarly before conducting our this research we have studied books, thesis, different articles, research publications and other related material to determine what has been done in this domain earlier and to find out the gaps that are present in this specific area from which we have drawn our problem.

Jenq-Muh Hsu et.al. [15] described that Push to Talk (PTT) is a very easy, simple and one way voice group communication. To join the PTT session talker presses the talk button and all the session members hear the talk of the talker at the same time. It is a half duplex communication system. PTT is practically used by military, rescue, security, police and emergency radio communication. Push to talk service is introduced in mobile telecommunication called push to talk over cellular. To obtain present status of the user context awareness mechanism is used. Presence service makes the availability of user status to other and other status to user. Presence services and application are broadly presented in the future. Newly subscribe context information can be responded by the presence service. Context awareness allows the user to cooperate with their surroundings through touchable and insubstantial interactivities around them.

To make PTT conversation with multiple participants mobile user use POC service but there exist some disadvantages for example current status of participant partner is unknown, bandwidth wasting, and unnecessary transmission. So Open Mobile Alliance (OMA) combines a presence server with its defined POC architecture. It works together with IP multimedia subsystem which is the central part of all IP 3G/4G network. SIP protocol is used to help the routing in IMS architecture. Each POC entity can obtain the context information of the related POC server, POC users and POC session through presence service. Context awareness is mechanism in which PTT session will dynamically choose which members those are available to join the PTT talk according to the context status of group members. It can also save some unnecessary floor controls and PTT voice media forwarding among disabled user during a PTT session. OMA POC architecture and 3GPP IMS are used to implement the context aware PTT service.

PTT session is not frequently used while presence service is a continuous online service. In PTT we need presence information only when we want to establish PTT session. But PTT members continuously receive presence service whether they need or not. In this way there will be load on the network. All the members continuously receive all the type of media like video, images and voice. There will be wastage of bandwidth. PTT members will also disturb due to continuously receive irrelevant information. Author did not mention the specific presence attributes which will be used to decide about inviting a user.

Niklas et. al. [14] described that for next generation network IMS is used as service delivery architecture. PTT used IMS based services to move VoIP application for mobile devices. OMA specifies PTT as an IMS based service to exchange and use information between different operator domains. PTT will extend with different media types like voice, video communication, file transfer and service subscription for context push service. So push to Multimedia is an enabler to provide IMS applications with advanced multimedia communication facilities. 3GPP define IMS and provide platform for service delivery of IP multimedia services with an emerging mobile all IP network environments. It provides a platform for both circuit and packet switch network. This help to access mobile application and services from different networks. PTT services are specified by the Motorola, Nokia, Siemens and bring IMS applications in market called Push to Talk over Cellular (POC).

Next Generation Network at Fraunhofer FOKUS has employed Push To Multimedia services using voice and video application. It facilitates the user to permit different communication channel. These networks enable high level application programs interface of the parley group and Java application for integrated network. These applications help developers to generate services that are free from network equipment. Internet developers now knows the telecommunication world and their essential technologies and provide secure network to operators by using OSE policy. An independent network for open application is offered by OSA/ Parlay interface that is presented to 3rd party application host.

This paper illustrate that it is possible to extend the Push functionality to a Push to Multimedia architecture. The basic purpose of author in this paper is to introduce multimedia services using voice and video applications.

Menget.al [16] proposed that for mobile network Push to Talk over cellular service is designed. It is just like walkie talkie service. This architecture provides a platform in which predefined members participate in POC session. In this session one member speaks and other members listen simultaneously. POC applications are installed in mobile terminal called POC client.

Talk burst control provide mechanism for service permission. In one POC session predefined group members participate. Talk Burst Control specify that at both client and server side finite state machines are implemented by OMA. TBC and FSMS for POC server (called FSMG) and POC client (called FSMU) where "U" represent "user" and "G" represent "general". "S" and "R" prefix of the transition represent "send" and "receive" respectively. SIP and SDP are used for session establishment. "tb_grant" is added as a new parameter in the SDP's field to support the POC service. If "tb_grant = 1" then POC client is permitted to speak and if "tb_grant =0" then POC client is not permitted to talk. This paper indicates that as the request rate increases, group members that support the POC service decreases.

The solution of this paper is only for pre arranged push to talk session. The solution is not scalable and only member can be the part of session because as the request rate increases, group members that support the POC service decreases.

Balazs et. al [17] described that POC services are analyzed by implementing GPRS Network in terms of bandwidth and delay. It concludes that service will be implemented using enhanced GPRS and UMTS networks. GPRS network uses Packet switch domain. Implementation of POC service with GPRS network in term of quality and performance is a difficult task. POC service does not yet support real time voice component. There is no guaranteed of bandwidth when IP packet transferred with a limited delay. This paper concludes that If GPRS parameters are selected carefully and GEARN cell are configured appropriately then we can implement POC service design in the current GPRS network.

Although end user delay, flow control and voice quality required for media transmission are not recommended in a GPR network. The quality of service can be achieved by using extended GPRS and UMTS network where it supports real time traffic.

Tran et.al described that [18] POC traffic consists of traffic modelling and group modelling. Simulation is used for the investigation of POC traffic. Group communication of POC system is effected by speech activity of joining members and different features such as timer and counter. This paper describes the effect of different processes, basic group model of POC traffic and behaviour of user in speech activity. The results of most appropriate features are presented by simulation. POC services consider two invitation methods that is one by one invitation and batch invitation. The result of simulation in batch invitation methods shows the maximum size of group. In batch invitation floor blocking is higher than one by one invitation. The request behaviour of early floor is less than floor blocking. POC traffic consists of two types that is termination of calls by inactivity timer and normal completed calls.

Lan Wang describes [20] that for group and individual communication cost effective, interactive and global research service are provided by push to talk over cellular. This paper presents an approach towards the enhancement of session setup procedure of POC group communication using rich presence information. POC send invitation only the specific members of pre-arranged session but after getting presence information of group member's from XDMS. 20% session setup delayed can be reduced if there is a POC setting of each member in a POC group and XDMS has latest presence attributes.

To reduce SIP signalling this paper present POC service over the air interface. When POC service activated it initiate POC contact list of presence subscription. On UE RLS subscribes presence information and save it on shared XDMS. This fresh presence information reduces session setup time.

Alam et.al. in 2006 purpose the idea of narrowcasting [21]. According to them instead of sending their communication to all the participants of the conference, each participant can configure his preferences that to whom he want to send his voice and to whom he don't want to. Same is the case with receiver.

The targeted service of the paper is conference and the major focus is on audio communication. Moreover all the users have the same rights.

Shafi et.al. in 2009 described that in conference service instead of allowing only one user to refer any one, an election should be arranged. Everyone will poll his vote and at the end if the total vote exceeds a certain threshold the referred user will be allowed to join conference otherwise the request will be discarded. Moreover authors also specified that the referred user's right can be reduced.

First of all main focus of the author is on conference service and secondly authors authorized a single user to reduce the rights of the referred user. Moreover in the election algorithm every participant has same power of voting [22].

2.1 Limitation in literature Surveyed

The discrepancies which are seen in this literature survey, which would be addressed in the survey are as follows.

1. There is no criterion for PTT members to avoid receiving all the services including video, voice and text which they don't want to use. This type of communication not only waste the bandwidth but irrelevant information disturbs the users.
2. In talk burst control mechanism pre arranged group members use permission service but when the request rate increases the members in the session decreases. So service permission mechanism is not scalable.
3. In GPRS network user delay, flow control and voice quality are not recommended for media transmission. The quality of service can be achieved by using extended GPRS and UMTS network where it supports real time traffic.
4. Conference service used election algorithm to select the members but in the election algorithm every participants has same voting power. There is a single user who reduces the rights of the referred user. The main focus on the conference service

Summary of the Chapter:

PTM are popular and rising services in daily life. Their importance and applications are increasing rapidly. Due to their wide usage especially in the situations when there are lot of load on the network this is the most efficient system to be build. But this type of services requires a strong and effective measure to reduce network load and increase the scalability of the PTM services. The earliest research papers just present PTM functionality and its applicability. But PTM services require an efficient way that reduces the network load and make the architecture more scalable.

3. Requirement Analysis

3. Requirement Analysis

For Next Generation Network IP Multimedia act as service delivery architecture. Push To talk introduce IMS based services that move VoIP applications for mobile devices. The aim of IMS is the convergence of fixed or mobile networks at network level. This makes it possible to access mobile services and application from different end system but the key importance is open service delivery platform. Fraunhofer Institutes FOKUS employ PTM applications such as voice, video communication, file transfer or service subscription etc by using IMS architecture.

Due to limited protocol support and the other limitation of device capabilities, a majority of legacy devices (e.g. mobile phone, PDA, laptop, etc) today and in the near future lack coherent support for those services. There are different issues which PTM cannot satisfied. Firstly PTM does not provide a scalable system. There is a lot of load on the network and as a result there is wastage of resources. Secondly it does not compromise on real life scenarios and irrelevant information disturbs the users.

3.1 Problem Statement

There two different PTM servers that play an important role. These are controlling server and participating server. There is only one server that performs the controlling function and two or more server performing participating function depends upon the number of session participants. There are two types of sessions. These are

1. One to one and Adhoc session
2. Chat session

In case of one to one and adhoc session controlling function act as a PTM server and in case of chat session participating function act as server.

In PTM client always send message to a controlling server by using Real Time Protocol (RTP) and server forward that message to all members of the group as shown in the figure below.

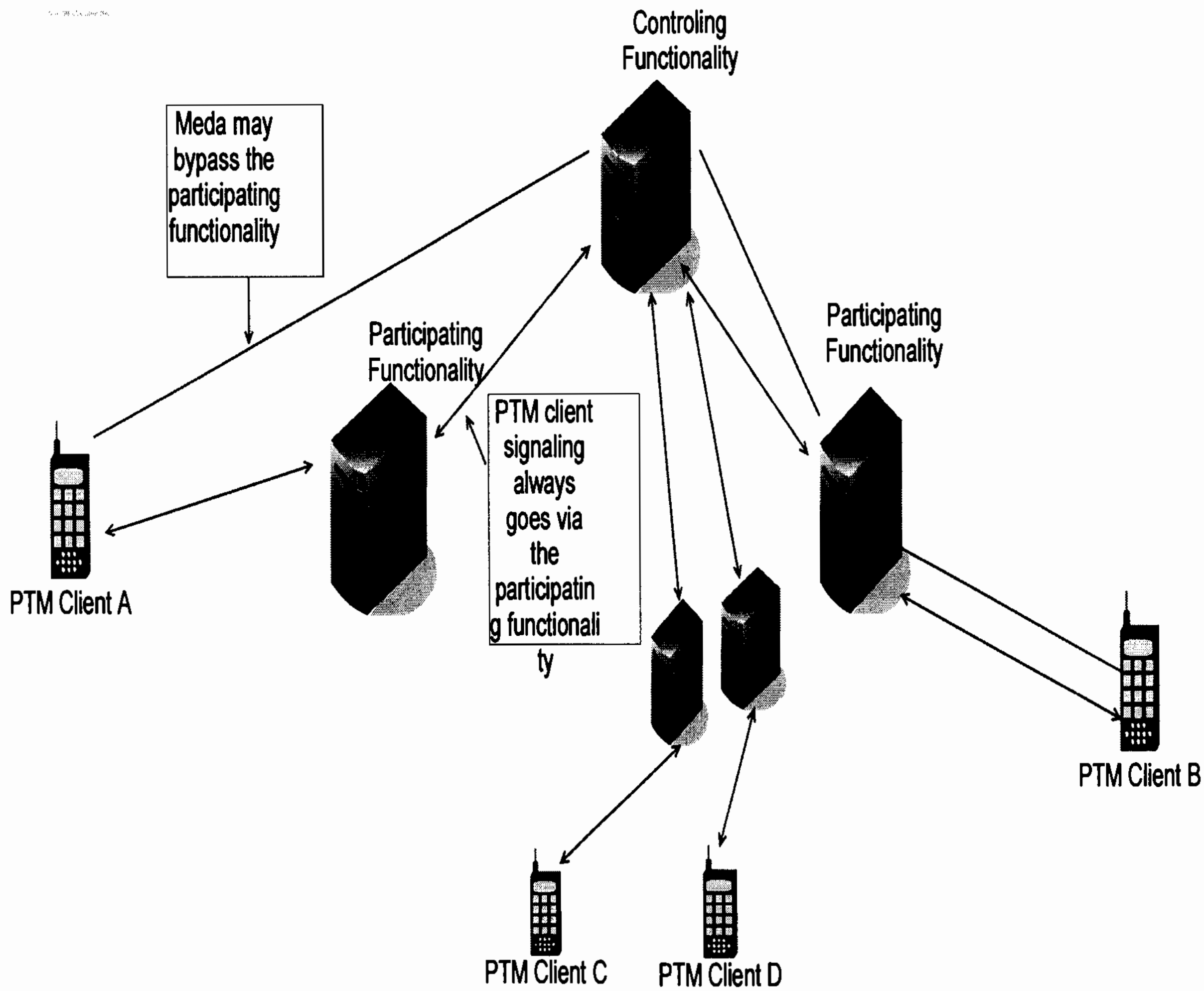


Figure 2: PTM Architecture

In this mechanism sender's information will be received by all the members of that PTM session. The sender cannot restrict the information to be delivered to the selected members of the group instead of all [2]. It might be possible that information is relevant to only few members of the group and sending it to all members not only results in wastage of network resources but also disturbs the irrelevant users. Moreover this mechanism also not allowed the sender to share some secrets with only few members of the PTM group. So on one side current mechanism of the PTM service reduces the quality of service and on the other side reduces the applicability of the PTM service in daily life scenarios [14].

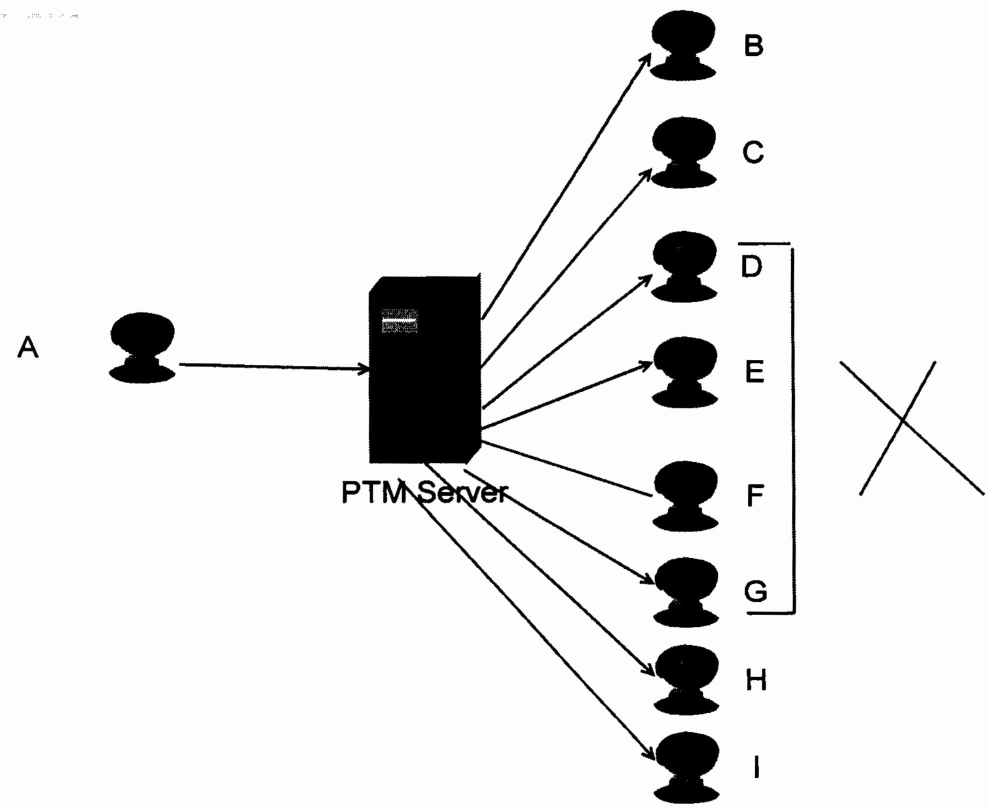


Figure 3: PTM Related Information

This figure shows that a PTM group is created. In this group eight participants participate. Client A want to communicate to only client B, C , H, I and does not want to communicate with the client D,E, F, G. but PTM using uni- multicasting technique in which if one client send the data to the server and server multicast this data to all of the participants of the session. There is no such type of mechanism in which client only send information to the specific clients. All the members of the session receive this information. Whether they want this information or not. As we know PTM provide all type of services to the participant including text, graphics, sound and video. If a client A use all these services to send this information and all the irrelevant participant's receive this information, it increase a load on the network and more bandwidth utilized. Client cannot share his/her secret information to only the selected members.

Secondly when a member joins a PTM group, it becomes a fully authorized member and allowed to speak, listen and watch. In many real life scenarios this mechanism does not work. For example in university a PTM based meeting is organized and the vice chancellor, Deans and Chairmen participated in it. Since the meeting is held to discuss the issues related to students so students are also invited to attend this PTM session. Since there are thousands of students in the university so the scalability issues does not allow inviting all the students. However if few students are invited then the first come first serve allocation of media burst control can results in talkative students and silent administration. Therefore students should not be allowed to speak but there is no mechanism to do this. Therefore we conclude that providing full rights to all the members of the PTM group on one side reduces the scalability of the PTM service and the other side does not suit well in many real life scenarios. This existing mechanism also increases the contention for the media burst control [19].

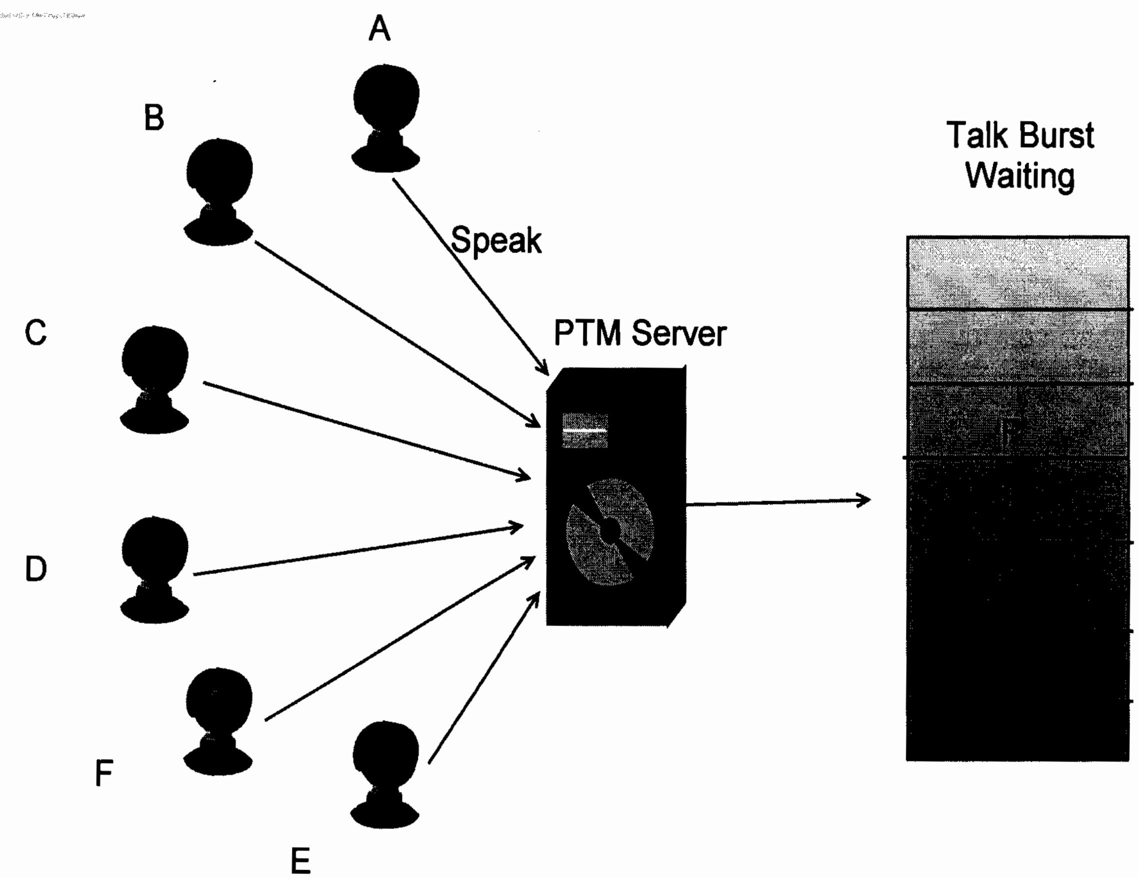


Figure 4a: waiting Queue when all user speak

Figure 4a shows a PTM group in which there is six participants. These participants talk in the session on first come and first serve basis. First participant A press the button and he is allowed to speak and remaining participant speaks on his/her own turn. Suppose every participant is allowed to speak only one minutes in the session. If participant A speaks for one minute on his turn then after six minute there will be a turn of participant E.

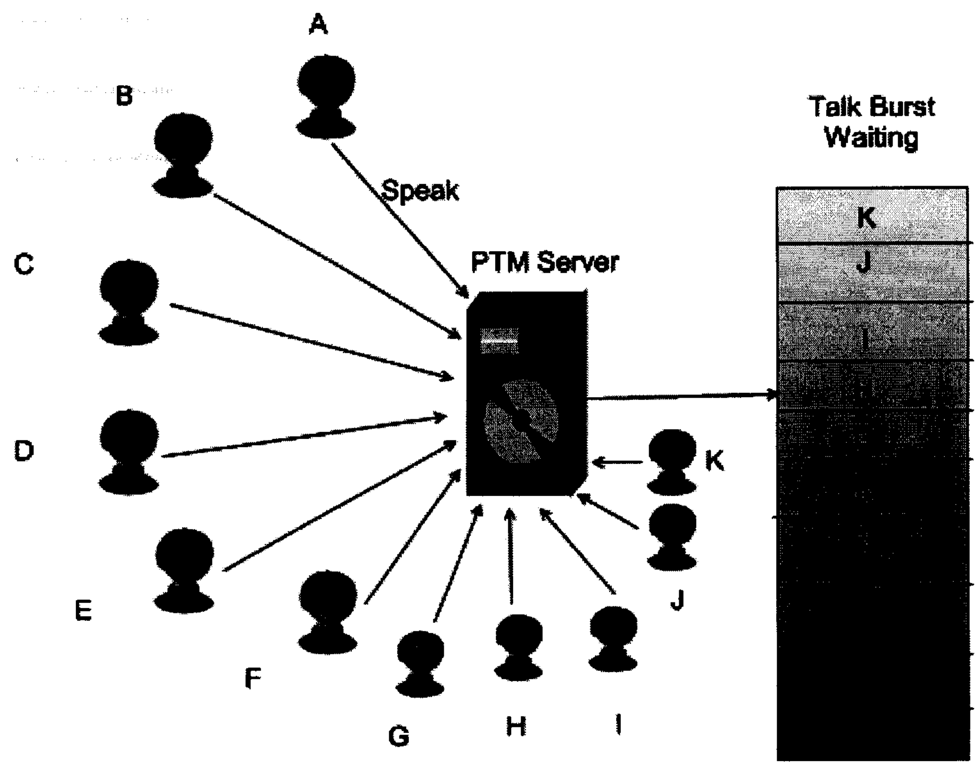


Figure 3b: waiting Queue when all user speak

Figure 4b shows that if we increase the number of participants. 12 participants participate in the group. Every participant speaks in his/her own turn. The speaking time for every participant is one minute. The participant waits for their own turn in the waiting queue. Participant A talk for one minute and participant K is allowed to talk after 12 minutes. If we add more participants for example 50 participants the last participant speaks after 50 minutes. This happens only if we fix the time for each participant for one minute. But if there is no time fixing and each participant is allowed to speak for unlimited time on his/her turn. Then waiting time will be more increased and nobody wants to use this type of group. This shows that the PTM does not provide a scalable system.

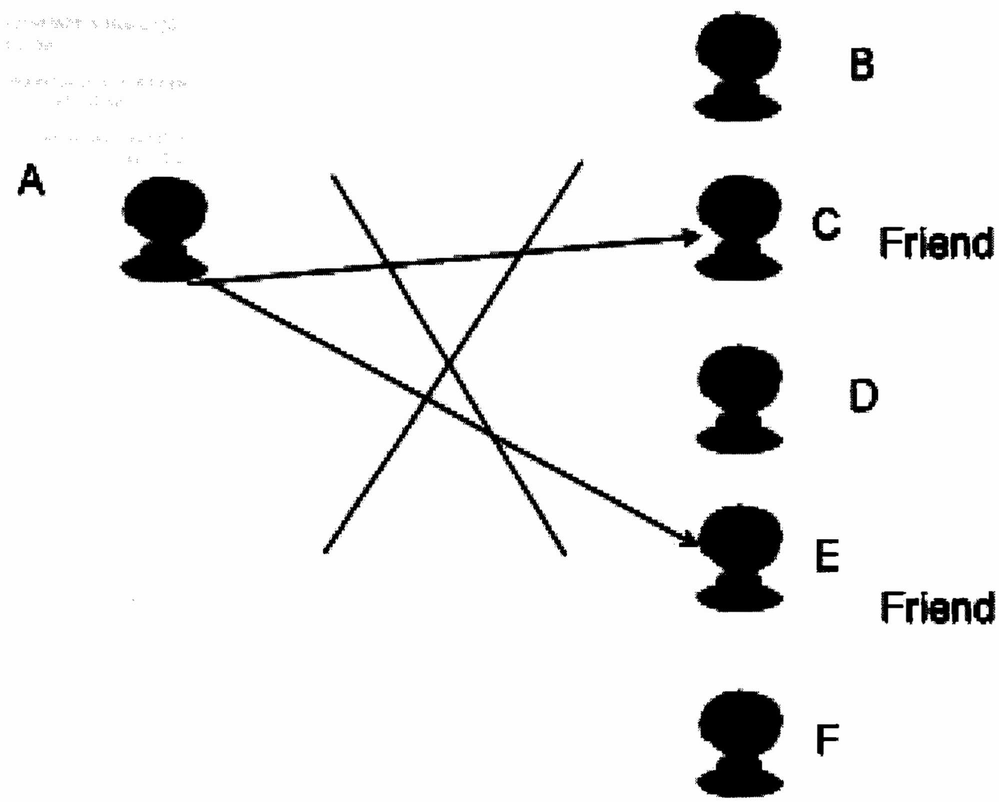


Figure 5:

In this figure client A want to communicate with his friends client C and Client E. But PTM does not provide such mechanism in which participants share its secret information to their friends only. PTM server multicast the data/information to all of the participants who join the group. Most of the participants do not want to these information because it is irrelevant to them. But they continuously receive this information from the server. This irrelevant information disturb the participants. Participants cannot share their secret or private information to the relevant users only.

3.2 Objectives of Research

According to the different problematic situations as it is discussed, we have set certain goals of this research which have built in solutions for the problems already discussed. The targets and objectives of this research with underlined methodologies are as follows.

3.2.1 Improve Scalability

To improve scalability we propose a new information delivery mechanism. In which we restrict the participant of receive all type of media. Due to our strategy participant use only those media for which they will be selected. More and more participants will join the group and made the group more scalable.

3.2.2 Reduce Network Load

To reduce network load our new information delivery mechanism will restrict the participant's to receive all type of media. Our solution will provide in such a manner in which we distribute the media services according to the status of the participant. We distribute the media services as a speaker, as a listener, as a speaker or listener, as a watcher or listener or as a listener, watcher or speaker.

3.2.3 Reduce Wastage of Resources

Due to fairly distribute the media services between the participants, there will be reduction of resources usage. Participants join the group as a listener, as a watcher or as a speaker according to their status. Due to this strategy participant use only those services which they will be registered for. Our strategy will overcome the wastage of media resources.

3.2.4 Maintain User Privacy

Our new information delivery mechanism also maintains the privacy of the participants. Our strategy will allow the participants to communicate with those selected participants to which they want to communicate. Participant will share their secret information to the relevant participants only.

3.2.5 Minimize the Chances of Disturbance

Our strategy will allow the participants to communicate with only the selected participant or to whom they want to communicate. So participants do not receive the information which is irrelevant to them. The receiver participants will not disturb due to irrelevant information. So our solution will minimize the chances to disturb the participants.

3.2.6 Covers Real Life Scenario

Our strategy will provide a solution that will be helpful in many real life scenario. As we discussed above the university example that previous PTM session does not cover real life scenario. But our solution will distribute the media services between the participants in such a way that some participants will be listener, some will be watcher and some will be viewer or some will use all these services.

3.2.7 Cost effective and Enhanced Applicability:

The proposed solution is more cost effective. It enhances the applicability of the PTM services in the daily life. It provides the control information to the sender and also enhances the scalability of PTM services by allocating different rights to the participants.

3.2.8 Evaluation and Formulation:

Each of the strategy or technique is formulated and evaluated against.

Summary of the Chapter

PTM client send message to the server and server multicast this message to all of the participant using real time protocol. But due to this strategy PTM has to face the following problems. As all the participants receive all type of media services so this type of group is not a scalable. If more and more participants join this group, waiting time of the participants will increase. Nobody wants to join this group.

Participants join the group as a listener, watcher or talker. There is no such type of mechanism which restricts the participants to use only those multimedia services which they require. This type of PTM services increases the load on the network as well as unnecessary use of multimedia services increase the wastage of media resources.

This type of services utilizes a more bandwidth because there is usage of more multimedia services. There is no such type of mechanism in which participant can share their secret information to the relevant users only. Privacy of the participants disturbed by using these PTM. Participants disturbed by receiving irrelevant information.

4. Proposed Solution

&

Implementation

4.0 Proposed Solution

In the prior chapters we had explained our problem which is the applicability of push to multimedia services using IMS architecture in daily life scenario. PTM is susceptible to use services because of hereditary built in disadvantages from which they experience like low bandwidth, less scalability, load on the network, disturb privacy, wastage of media resources. Our provided solution will manage the solution while taking into consideration these issues.

By considering the above mentioned problems we have designed our proposed solution that increase the applicability of PTM service and improve the scalability in daily life scenario. Our proposed solution provides a new information delivery mechanism and its name is "Choice Casting". In choice casting sender is allowed to send his information to selected members within PTM session. When a sender joins the group he configures rules that at what time what information will be delivered to which members of the group. After that whenever sender sends some information, the controlling server verifies the registered information delivery rules and deliver the information accordingly. This solution allows the users to share their secrets with the selected members and provide the information to the relevant users only.

In our thesis we formulate information delivery mechanism which allocates different rights to the user that makes the PTM service more applicable and more scalable. Till to date all the work that has been done on the PTM is not concerning to this discipline which we have focussed on. All the existing work describe the push functionality in push to multimedia architecture, in which participants use the PTM services with all rights that increase the load on the network and utilizes more bandwidth. Our value able resources are wasted by using the existing PTM architecture. In contrast to this mechanism our proposed solution does not give the full rights to all of the participants. Our allocation rights mechanism makes sure to reduce the network load and reduce the wastage of value able resources.

In the real network the above scenario can be more simple as there is no such type of mechanism that control the media between the participants. As PTM use SIP protocol, so server use uni- multicasting technique in which participant sends data to the server and server multicast this data to all of the participants. As participants receives all type of media and wastage of resources causes starvation and can also even paralyze the whole network.

One simple and straight forward solution that comes to mind is to stop using all types of media. In the case of PTM choice casting is the only option for obtaining optimal solution in this type network because of the reason specified in chapter 1.

An important task is here that through these choices casting mechanism we would be introducing equality along with it we would attempt to balance the load originating within the network mainly due to the talk burst. The server use different Choice Casting strategies such listener, watcher and viewer for attaining the scalability of the group. We just want to provide fairness among each of the participants in the network with the desire media distribution mechanism.

Our mechanisms for allocating different rights are not allowing all the users to speak, listen and view all the time. Some users will be invited with the full rights and some will be invited with limited rights. In the case of university examples the students can be invited as listener only. Since students are not allowed to speak and send data to the controlling server so it eliminates them from the contention of media burst control. Elimination of students from the media burst control session allows inviting hundreds of students. Secondly since students are not allowed to watch the PTM session so it also reduces the burden from the network and results in efficient utilization of resources.

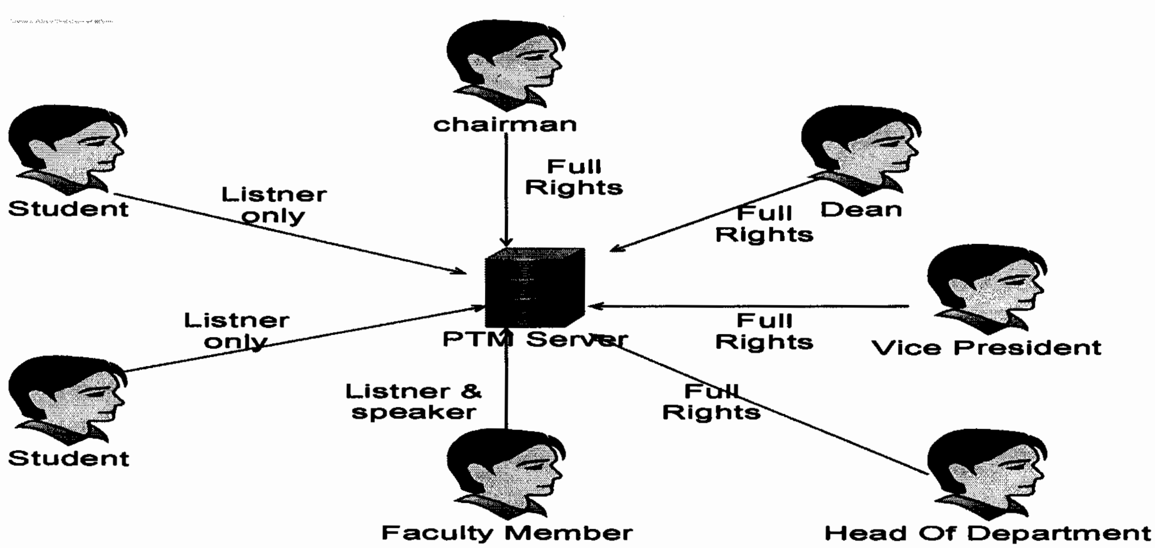


Figure 5: university example

We have just explained the concept of our solution by the simple study of fractional division. Later in this chapter we would study our mechanism for distributing media services, which server distribute between the participants. Sometimes these distribution mechanism which server follow may reject the participants if he/she is not the authorized participants of the PTM or if the participant want to join the PTM and existing participants don't allow that participant to join the group based on their vote. So to become the member of the PTM this mechanism help the user to join the PTM.

With a more general example we can understand the working of this process which is as follows. In order to restrict the media distribution between the participants it is necessary to provide him only a small portion of media resources. If the resources are distributed fairly among the participants of PTM than the load on the network is reduced and is not able to occupy a major share of the network resources.

To achieve this target we in this project formulate our new information delivery mechanism that fairly distributes the media resources among the participants of the PTM. The rights will be allocated to the students at the time of session joining. Whenever a new student joins a session a polling process starts to allocate rights to that user. The polling process is controlled by the controlling server. All the existing members of the PTM session cast their vote but every vote does not have the same value. Vote of the users with full rights has more value as compared to the vote of the user with limited rights. At the end votes are counted and rights are assigned to the invited user. After that if the invited user is allowed to send the media then he configures the information delivery rules at the controlling server which are checked at the time of information delivery.

Figure shows the detail architecture that how Media distribution strategies work.

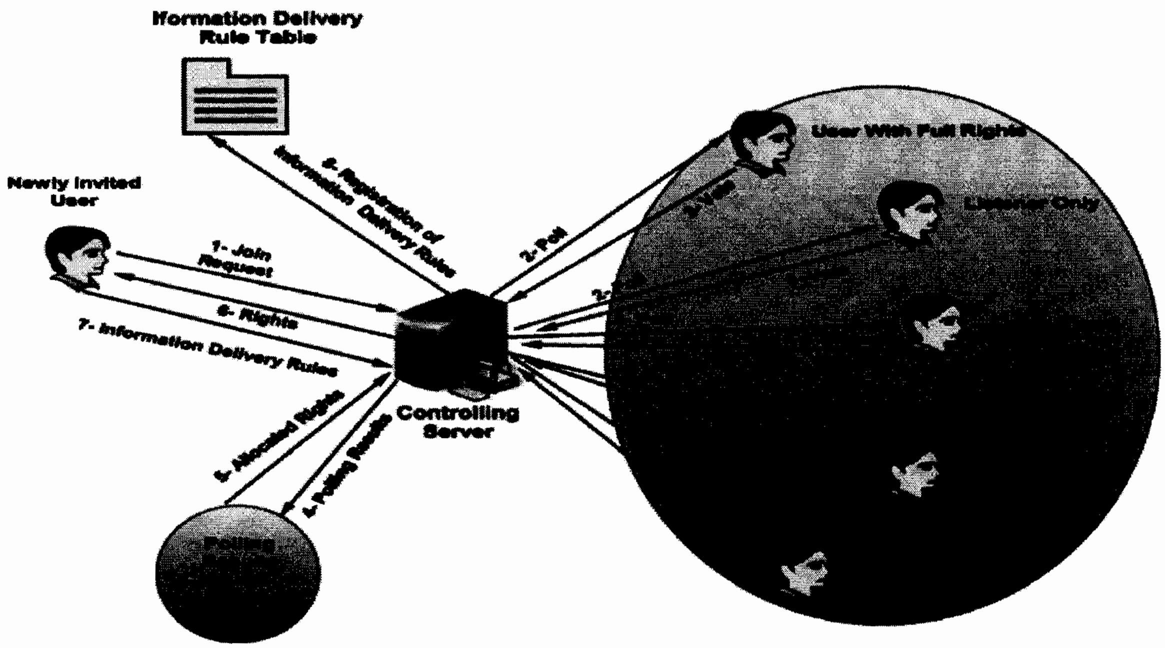


Figure 5: university example

4.1 "Election Based Mechanism" New information Delivery Mechanism

In this section we formulate various choice casting strategies with an aim to mitigate the load on the network. Election Based Mechanism manages the Talk Burst in the network. The main objective which is to be achieved with this technique is that it increases the scalability of PTM and decrease the load on the network. To solve the above mentioned problems and to make PTM service successful in many real life scenarios we, in this paper, proposed a new service mechanism for PTM.

4.2 Membership and Rights Allocation

Whenever a user is invited in a PTM session or a user sends a join request, an election is started among the existing members of that PTM session. Each of the existing members is notified by the PTM controlling server that a new user with ID <ZZZ> wants to join this PTM session. Each of the existing members casts its vote that whether she want to allow the new users to become part of the PTM session or not. If the new user is allowed to become member of the session then what rights should be allowed to that users. In our proposed mechanism all the users do not have full rights. Few users can have full rights and few can have limited rights. For example a user can have full rights <Speaker, Listener, Watcher>, another user can have limited rights <Watcher, Listener>. Moreover voting power of each of the user is not equal. A user who has more rights has more voting power as compared to the user who has lesser rights.

4.2.1 User Sends Join Request

To join the session two types of users exists.

1. Existing users
2. Newly Invited Users.

4.2.2 Existing User

When existing user send the join request to the server, the server verifies the username and password from the database. If the user is authorized the server allows him to join the session and if the user name and password is not verified then server reject the join request.

The algorithm used for sending the request to the server that check user name and password are.

```
Private Sub btnconnect_Click(ByVal sender As System.Object, ByVal e As

   System.EventArgs) Handles Buttonl.Click

  If loginform.ShowDialog <> Windows.Forms.DialogResult.OK Then Exit Sub
      a = loginform.UsernameTextBox.Text
      b = loginform.PasswordTextBox.Text

       VoiCln.OpenSession("127.0.0.1", 2222, a, b)

  End Sub
```

```
Private Sub VoiCln_OnSessionOpen() Handles VoiCln.OnSessionOpen

        Button1.Enabled = False
        Button5.Enabled = True
        BtnUser.Enabled = False
        BtnAlert.Enabled = True
    End Sub


Private Sub VoiCln_OnPeerConnected(ByVal aHandle As Integer) Handles
VoiCln.OnPeerConnected

        LI = lvPeers.Items.Add(VoiCln.GetPeerName(aHandle))
        LI.Tag = aHandle
        LI.SubItems.Add(Now.ToLongTimeString)
        LI.SubItems.Add("Connected")
        LI.SubItems.Add(aHandle)

       End Sub
```

If user name and password are not matched then it will reject the session by using this algorithm.

```
Private Sub VoiCln_OnSessionRejected(ByVal aCode As Integer) Handles
VoiCln.OnSessionRejected
        If ustatus = 1 Then
                  MsgBox("       User Name already exist        ",
      MsgBoxStyle.Exclamation, "Server Alert Message")
            ustatus = 0
        Else
                  MsgBox("    User Name or Password may be wrong       ",
      MsgBoxStyle.Exclamation, "Server Alert Message")
        End If
       End Sub
```

4.2.3 Newly Invited User

When new user send join request to the server, the server sends this request to the existing members. The existing members cast their vote that allow the newly invited user to join the session. So it depends on the existing members vote that allow the new user or reject the new user.

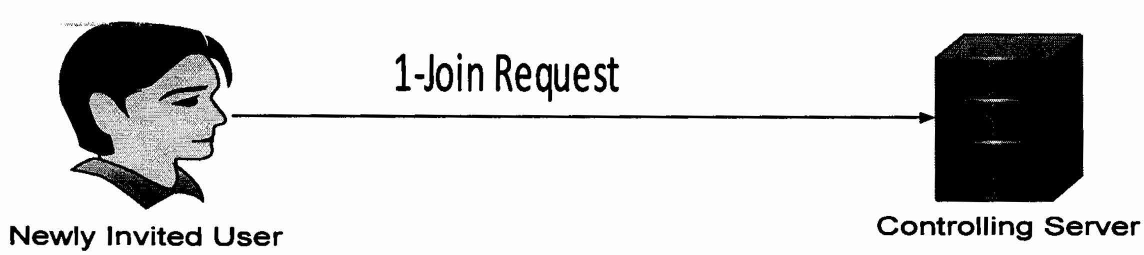


Figure 1: server send join request

The algorithm for the newly invited user request to the controlling server.

```
  Private Sub btnaddnew_Click(ByVal sender As System.Object, ByVal e As
System.EventArgs) Handles BtnUser.Click

    If FUser.ShowDialog <> DialogResult.OK Then Exit Sub
        UName = FUser.TextBox1.Text
        UPassword = FUser.TextBox2.Text
        ustatus = 1
  VoiCln.OpenSession("127.0.0.1", 2222, UName & ":" & UPassword, UPassword)


    End Sub
```

4.3 Voting Power

As we have discussed above that voting power of a user depends upon the rights allocated to the users so we placed a table that define the rights of each of the user.

This voting power is allocated firstly on the basis of media contention. Since PTM is a half duplex service and at a time only one user can send media so a user who is allowed to send media has more rights as compared to others. So all the possibilities in which user has the rights to speak have more voting power. Scenario in which a user is receiver only (Listener or Watcher), she will not participate in media contention so that user has lesser voting power as compared to others.

Table 1: Rights and Voting Power

Rights	**Voting Power**
Full Rights- Speaking, Listening and Watching	**4**
Speaking and Listening	**3**
Speaking only	**2.5**
Listening and Watching	**2**
Listening only	**1**

4.3.1 Server Sends Vote Request To The Existing User.

When the controlling server receives a connect request from the newly user, the controlling server sends the vote poll request to all the existing users of the session. As we know, in this session user have different status. Users join this group with different status categories. Such as:

1. Listener, Watcher and Speaker (Full Rights)
2. Speaking and Listening
3. Listening and Watching
4. Speaking only
5. Listening only

In our solution we set the vote value for the users. We set the vote value 4 for the user who joins the session with full rights; vote value 3 for the user who joins the session as a Speaking and a Listening, vote value 2.5 for the user who joins the session as a speaker only, vote value 2 who join the session as a listener and watcher and value of the vote is 1 who join the session as a listener only. All of the existing users who are online receive this vote poll request from the controlling server. It's depend on the existing user vote that allow the newly invited user to join the group.

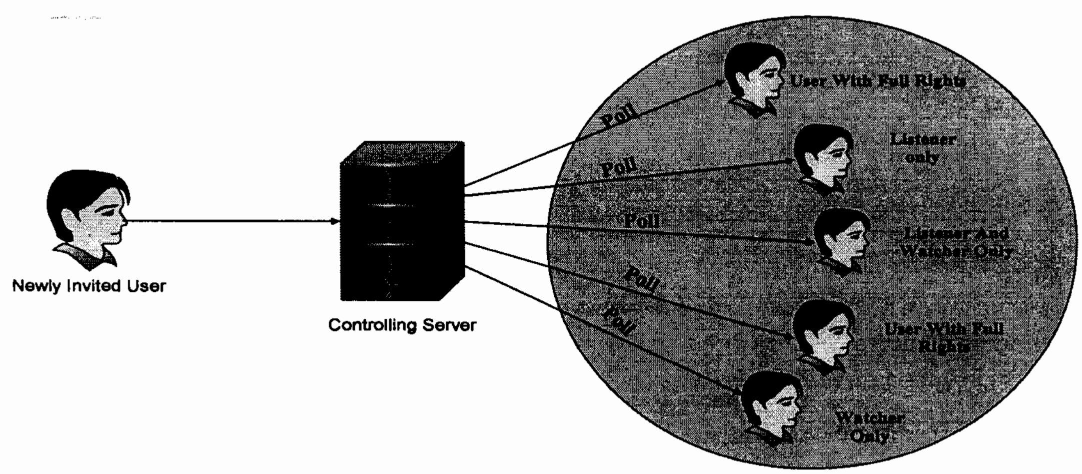


Figure 2: Controlling Server send Poll request

The algorithm use to send the new user request to the existing user.

```
LI = lvPeers.Items.Add(VoiCln1.GetPeerName(aHandle))

        LI.Tag = aHandle
        LI.SubItems.Add(Now.ToLongTimeString)
        LI.SubItems.Add("Connected")
        LI.SubItems.Add(aHandle)
        If userid = 1 Then LI.SubItems.Add("1")

        If userid = 1 Then

            For ab As Integer = 0 To lvPeers.Items.Count
                If lvPeers.Items(ab).Text <> VoiCln1.GetPeerName(aHandle)
Then
                    VoiCln1.SendAlertMessage(lvPeers.Items(ab).Tag, "Case1"
& newuser)
                End If

            Next

        End If
        userid = 0
```

4.4 Voting Option

Whenever a user casts her vote, she can select any of the 6 available options. Each option shows that what rights this user wants to allocate to the new incoming user. Each of the picked option has different weight as compared to the other options. Available options and their weights are shown in Table II.

Table II: Available Options to Cast Vote and their Weights

Available Options	**Weight**
Full Rights	4
Speaking and Listening	3
Speaking Only	2.5
Watching and Listening	2
Listening only	1
Not Allowed	0

Each participant has to cast its vote within a specified period of time. After timeout a user who does not cast its vote will not be counted in election process.

4.4.1 Sever Receive Vote result from the Existing Users

At this stage the controlling server start collecting the vote from the existing users. All of the existing members cast their vote but we know that each vote value is not same because member joins this group with different allocated rights. At this stage we set a timer for receiving the vote. We set the timer value for 1 minute. When existing members receive request, server wait for receiving the vote from the members only for 1 minute. During this time server send alarming message to the existing members after every 1 minute to cast their vote. If the members do not cast their vote during this time, then server will not collect the vote from that member who does not use their right to cast the vote.

So all of the members who receive vote poll request must cast their vote. Because the fate of newly invited user is based on their vote. After collecting all votes from the existing members, server will decide whether he is eligible to join the group or not.

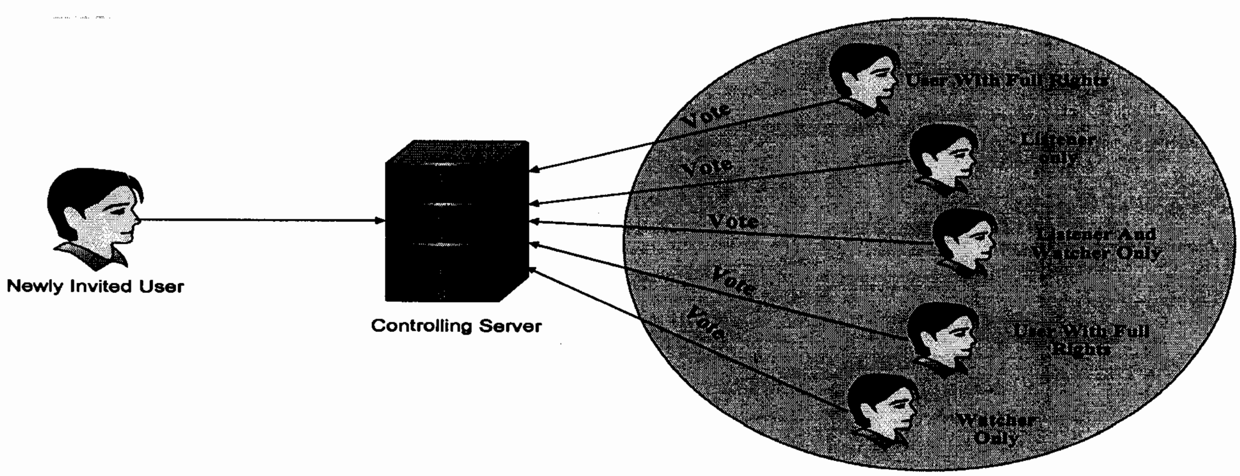


Figure 1: Members Cast their Vote

4.5 Votes Calculation

PTM controlling server after receiving votes from all the users or at timeout calculates the votes obtained by the new user. Obtained votes are calculated by using the formula:

$$\text{Votes Obtained} = \sum_{i=0}^{n} (\text{Voting power of user } i * \text{Selected option by user } i) \rightarrow \text{eq } 1$$

n is the total number of users who are casting their votes.

After that total votes are calculated by using the formula:

$$\text{Total Votes} = \sum_{i=0}^{n} (\text{Voting power of user } i * 4) \rightarrow \text{eq } 2$$

After that percentage of obtained votes is calculated by using the equation:

Percentage of obtained votes = (Votes obtained / Total votes) * 100 → eq 3

.

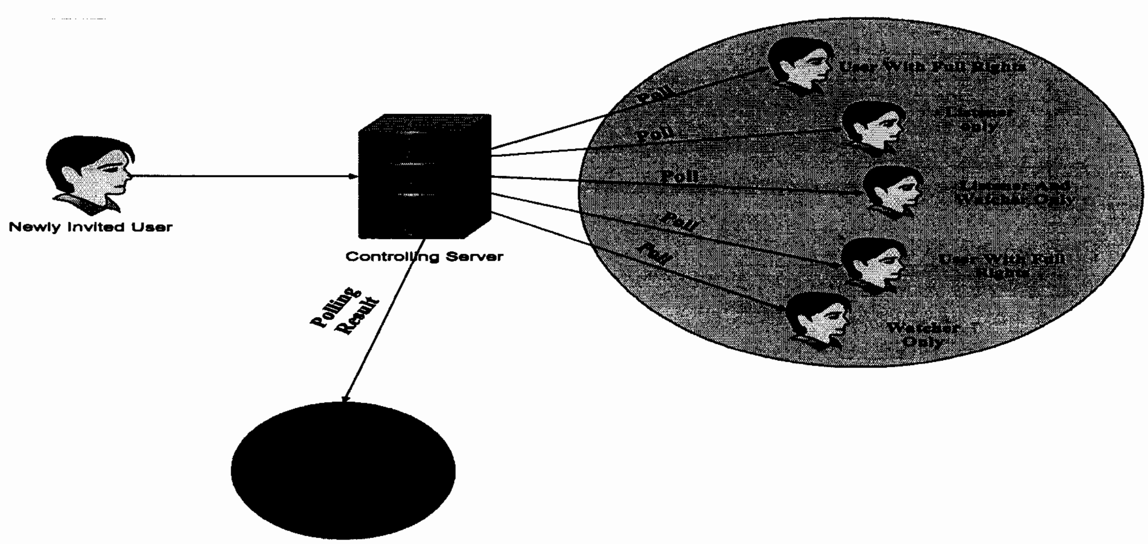


Figure 4: server calculating the vote

The algorithm used by server to calculate the votes by using above formulas are

```
str1 = "select * from vote where id = " & Uid2 & ""

rss.Open(str1, con1, ADODB.CursorTypeEnum.adOpenDynamic,
ADODB.LockTypeEnum.adLockOptimistic)
          Do Until rss.EOF
                    ' If rss.Fields("id")
  If rss.Fields("userpower").Value = 4 Then
                    U4 = U4 + 1

    VObt4 = (rss.Fields("userpower").Value * rss.Fields("vote").Value) * U4
End If

 If rss.Fields("userpower").Value = 3 Then
                        U3 = U3 + 1
    VObt3 = (rss.Fields("userpower").Value * rss.Fields("vote").Value) * U3

                    End If

                    If rss.Fields("userpower").Value = 2.5 Then
                        U21 = U21 + 1
                        VObt21 = (rss.Fields("userpower").Value *
rss.Fields("vote").Value) * U21
                    End If

                    If rss.Fields("userpower").Value = 2 Then
                        U2 = U2 + 1
                        VObt2 = (rss.Fields("userpower").Value *
rss.Fields("vote").Value) * U2
                    End If

                    If rss.Fields("userpower").Value = 1 Then
                        U1 = U1 + 1
                        VObt1 = (rss.Fields("userpower").Value *
rss.Fields("vote").Value) * U1
                    End If

                    ' VoteingObt = (UserValue * rvote)
                    'End If

                    rss.MoveNext()

                Loop

                rss.Close()

     VObt = VObt1 + VObt2 + VObt21 + VObt3 + VObt4
U = U1 * (1 * 4) + U2 * (2 * 4) + U21 * (2.5 * 4) + U3 * (3 * 4) + U4 * (4
* 4)
                PR = VObt / U * 100

        End Select
```

4.6 Server Decides Allocated Rights

After calculating votes and percentage PTM controlling server decides whether to allow this user to become member of the PTM session or not. If user is allowed to become member then what rights should be allocated to that user. For this purpose threshold values are configured and rights are allocated according to threshold values. Threshold values that we used in our experiments are listed in table III.

Table III- Votes and Decisions

Percentage of Obtained Value	Decision
< 20 %	Not Allowed
>20%	Listener Only
>35%	Listener and watcher
>50%	Speaker Only
>60 %	Speaker and Listener
>70%	Full Rights

Since all the users are not allowed to send media so it first reduces the media contention. Reducing the media contention solves the scalability problem. More and more users can be added as listener and watcher only.

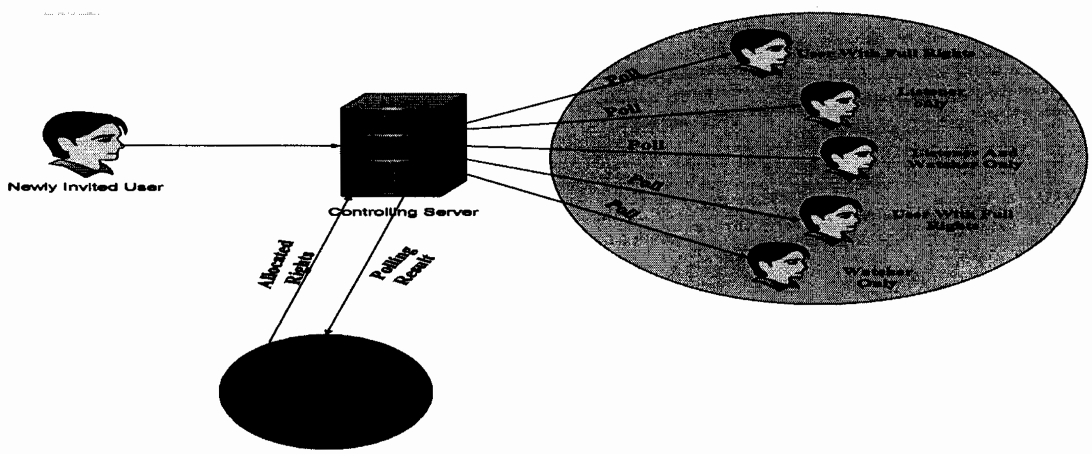
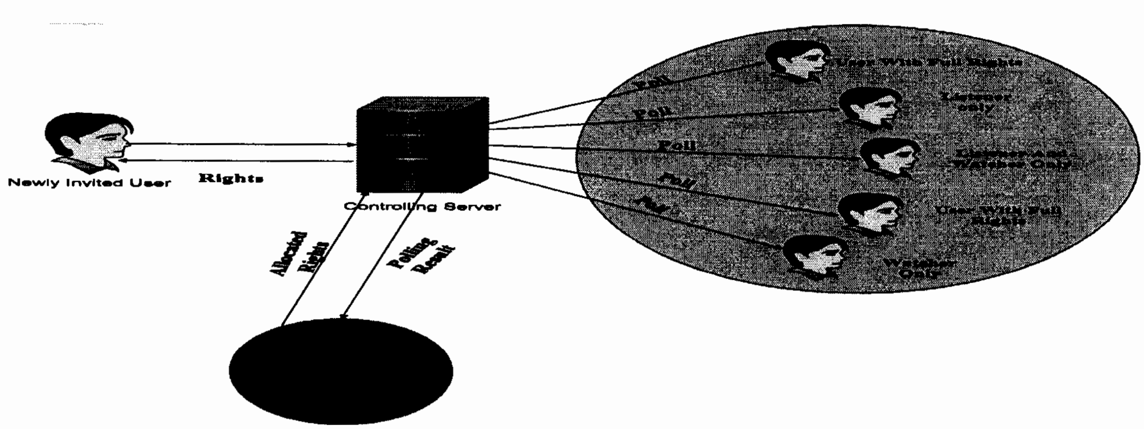


Figure 1: server decides for allocating rights

4.7 Server Allocated Rights to the New user

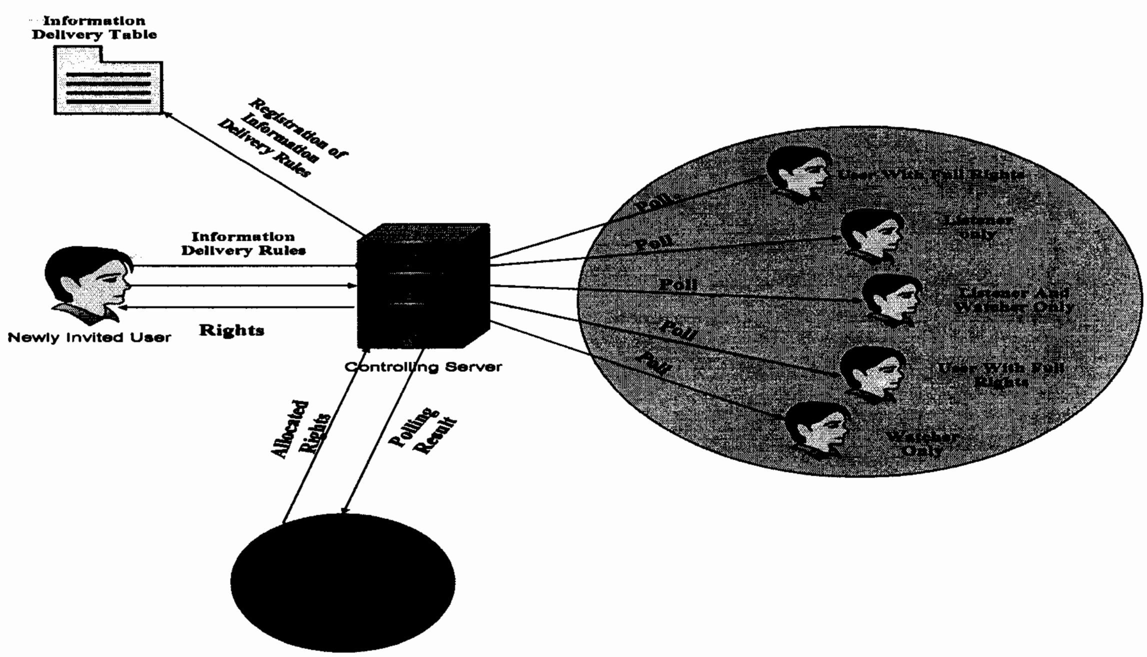
At this stage server send information to the newly invited user that whether he is selected as a member of the group or not. There are two type of possibilities at this stage. Firstly if existing members does not want that new user join the group with any status. Then polling process starts, calculates the values and server send the sorry message to the new user that you cannot join this group.

Secondly if existing user allow the new user to join this group then server send congratulation message to the newly invited user that you are allowed to join this group. Server will also inform about the status of the newly invited user that he join the group with full rights or with limited rights.



4.8 Newly Invited User Configure Information Delivery Rules.

At this stage when newly invited user receives congratulation message from the server. Then new user send message to the server to configure the information delivery rules.



Example-Election Based Rights Allocation

We take a scenario in which there are already 50 registered members and a new member sends request to join that session. Out of these 50 users 12 has the full rights, 8 are speaker and listener, 6 are speaker only, 12 are watcher and listener and 12 are listener only.

After getting new join request from user N an election is initiated. All the 50 members cast their votes. Controlling PTM server receives the following votes:

- 6 users out of which 2 have full rights, 1 speaker only, 2 watcher and listener and one listener only, agrees to grant full rights to user N.
- 9 users out of which 2 have full rights, 2 speakers only, 4 watchers and listener and one listener only, agrees to invite user N as speaker and listener only.

- 5 users out of which 1 with speak only rights and 4 with listener only rights, agree to invite user N as speaker only.
- 11 users out of which, 3 users having full rights, 5 have the rights of speaker and listener, 2 with watcher and listener and one with listener only rights, agree to invite user N as watcher and listener.
- 16 users out of which 3 have full rights, 3 have speaker and listener rights, 2 have speaker only rights, 4 have watcher and listener rights and 4 with listener only rights, agree to allow user N rights of a listener only.
- 3 users out of which 2 having full rights and one having rights of listener only, are not willing to accept user N as group member.

Now controlling server calculates the vote by using eq 1.

$A = 2(4*4) + 1(2.5*4) + 2(2*4) + 1(1*4) = 62$

$B = 2(4*3) + 2(2.5*3) + 4(2*3) + 1(1*3) = 66$

$C = 1(2.5*2.5) + 4(1*2.5) = 16.25$

$D = 3(4*2) + 5(3*2) + 2(2*2) + 1(1*2) = 54$

$E = 3(4*1) + 3(3*1) + 2(2.5*1) + 4(2*1) + 4(1*1) = 38$

$F = 0$

Total votes obtain are A+B+C+D+E = 236.25

Total votes are calculated by using eq 2.

$= 12 (4*4) + 8 (3*4) + 6 (2.5*4) + 12(2*4) +12(1*4) = 492$

Percentage of obtained votes is calculated by using eq 3.

$= (236.25/492)*100 = 48.01\%$

So as shown in the table III user N is allowed to join the PTM session as a listener and

only.

4.9 Media Mixing

To overcome disturbance problem, privacy problem and resource utilization problem we proposed idea of media mixing. Each of the users after becoming member of the session specifies send/receive rules at PTM controlling server. Sending rules of user A means to whom user A wants to send her multimedia data and to whom she does not want to deliver her data. Receiving rules of user A means from whom user A wants to receive multimedia data and from whom she does not want to receive. Now conflicts can arise in send/receive rules. For example user A want to send her data to user Z but user Z does not want to receive that data. In case of any conflict priority will be given to the receiving rules.

Summary of the Chapter

In this chapter we have provided the solution that makes the PTM services more applicable and scalable. We have developed an election based algorithm that decides who become a member of the PTM session with different rights. The election based algorithm decides that the participant join the session with full rights, medium rights or only with one right. After becoming a member of the session the user specifies the rules form the PTM controlling server. This election based algorithms makes the PTM session more scalable.

5. Validation & Results

To validate the efficiency of the proposed solution we conducted number of experiments. Each of the experiment gives results better than the existing values.

5.1. Scenario 1

In this scenario Alice (a user) initiates a PTM session. After a while Bob sends a join request, an election is initiated and Alice allowed him to join with full rights. Carel was the third who wants to become member of this PTM session, the election results also allowed him with full rights. Duminy was the fourth to request and he is allowed as watcher and listener only. After 10 minutes of session establishment 50 users are allowed to become member of the PTM session with different rights. 10 members are allowed with full rights including the session initiator, 6 members as speaker and listener, 2 as speaker only, 12 members join the PTM session as listener and watcher and 20 members are the listener only.

In the first experiment we calculate the average waiting time for media burst control. Every user was allowed to speak maximum for 2 minutes in one turn. We run this experiment for 5 hours. First we take the results on the current delivery mechanism. We found that when all the users are allowed to speak the average waiting time per user is about 76 minutes. We repeat the same experiment by activating our election based right allocation mechanism where all the users are not allowed to request for media burst control. Only 18 from the 50 members were allowed to send media. We found that at this time average waiting time per user for media burst control is only 31 minutes. Even though the total member are still 50 but since all of them are not allowed to send media so the contention and average waiting time per user reduces significantly. We repeat the same experiments with different number of users and the results are shown in the figure below.

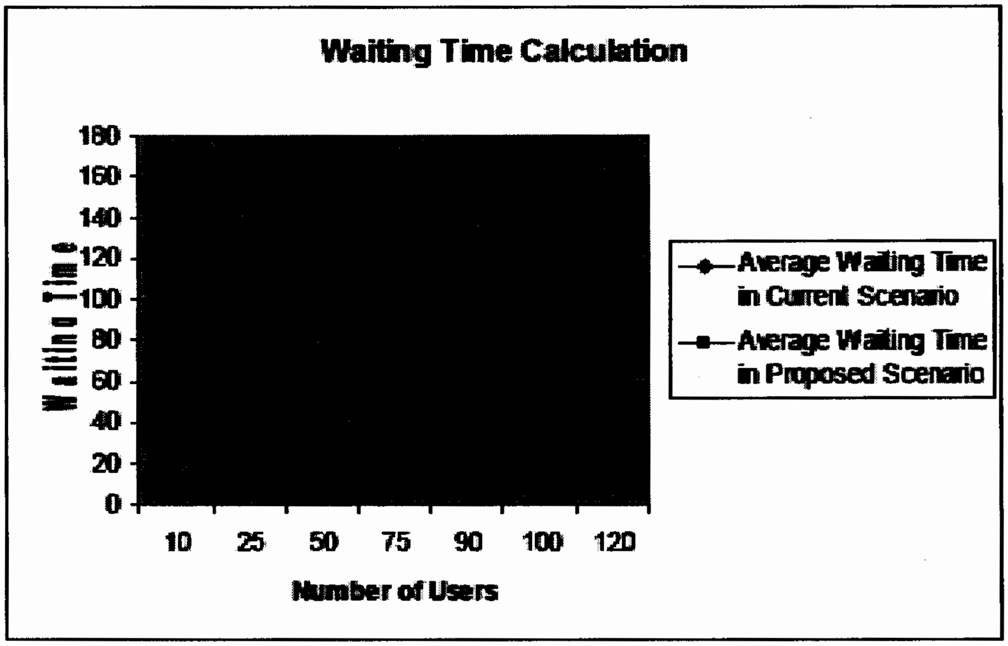


Figure 5: Average waiting Time

In the figure 5, it is shown that average waiting time increases with increasing the number of users. But the ratio of increasing in our proposed scenario is far lower as compared to existing scenario. If there are 100 users in a PTM session average waiting time 150 minutes in case of current scenario while it is 62 minutes in case of proposed scenario. So this solution makes the PTM service more scalable and a large PTM group can be established with different rights to different users.

5.2. Scenario 2

In the 2nd scenario we take 10 members all with full rights. These 10 members include Alice, Bob, Carel, Duminy, Ellen, Freddy, Gllen, Quetra, Paul, and Shan. We assumed that Alice has some secretes to share with Carel, Dumminy, Freddy, Paul and Shan therefore he does not want to send his media to Bob, Ellen, Gllen and Quetra. However Alice is willing to receive media from all the members of the session. Similarly other members also have some conflict with one another and want to share some information only with selected members of the session. Table IV shows the detail that who wants to send and receive media to whom.

Table IV: Preference Rules

User	Send To	Receive From
Alice	Carel, Duminy, Freddy, Paul, and Shan.	Bob, Carel, Duminy, Ellen, Freddy, Gllen, Quetra, Paul, and Shan.
Bob	Alice, Carel, Paul, and Shan.	Alice, Carel, Paul, and Shan.
Carel	Alice, Bob, Duminy, Freddy, Paul, and Shan.	Alice, Bob, Duminy, Ellen, Freddy, Gllen, Quetra, Paul, and Shan.
Duminy	Alice, Bob, Carel, Ellen, Freddy, Gllen, Quetra, Paul, and Shan.	Alice, Bob, Carel, Freddy, and Shan.
Ellen	Alice, Bob, Gllen, Paul, and Shan.	Alice, Bob, Gllen, Paul, and Shan.

Freddy	Bob, Carel, Duminy, Gllen and Quetra,	Alice, Bob, Duminy, Ellen, Gllen and Quetra
Gllen	Alice, Bob, Carel, Duminy, Ellen, Freddy, Quetra, Paul, and Shan.	Alice, Bob, Carel, Duminy, Ellen, Freddy, Quetra, Paul, and Shan.
Quetra	Alice, Bob, Carel, Duminy, Ellen, Freddy, Gllen, Paul, and Shan.	Alice, Bob, Carel and Duminy
Paul	Bob, Carel, Duminy, Freddy and Gllen	Bob, Carel, Duminy, Freddy, Gllen and Quetra
Shan	Alice, Bob, Carel, Duminy, Ellen, Freddy, Gllen, Quetra, and Paul	Carel, Freddy, Gllen and Quetra

Now when all the users specified their rules, Controlling PTM server mix these rules and devise a new table consisting of the exact entries that who's information should be delivered to whom. Table V shows the summery of information delivery rules. Table V is constructed by eliminating the conflicts of table IV. Preference is given to the receiver.

Table V: Information Delivery and Receipt

User	Send To	Receive From
Alice	Carel, Duminy, Freddy	Bob, Carel, Duminy, Ellen, Gllen, Quetra, Shan.
Bob	Alice, Carel, Paul	Carel, Paul, Shan.
Carel	Alice, Bob, Duminy, Paul, Shan.	Alice, Bob, Duminy, Freddy, Gllen, Quetra, Paul, Shan.
Duminy	Alice, Carel, Freddy, Gllen, Quetra, Paul	Alice, Carel, Freddy, and Shan.
Ellen	Alice, Gllen	Gllen
Freddy	Carel, Duminy, Gllen	Alice, Duminy, Gllen, Quetra
Gllen	Alice, Carel, Ellen, Freddy, Paul, Shan.	Ellen, Freddy, Quetra, Paul, Shan.

Quetra	Alice, Carel, Freddy, Gllen, Paul, Shan.	Duminy
Paul	Bob, Carel,Gllen	Bob, Carel, Duminy, Gllen, Quetra
Shan	Alice, Bob, Carel, Duminy, Ellen, Gllen	Carel, Gllen, Quetra

After summarizing send/receive information we measured the number of messages sent by controlling PTM server towards receivers. Each of the user sent 10 messages towards PTM server. So total 100 messages are received by the controlling PTM server. Then the controlling PTM server forwards these messages towards intended receivers. First we measure the number of messages exchanged by using the current service mechanism (without considering send/receive rules). After that message are delivered by PTM server by inspecting send/receive rules. Results are summarized in figure 6.

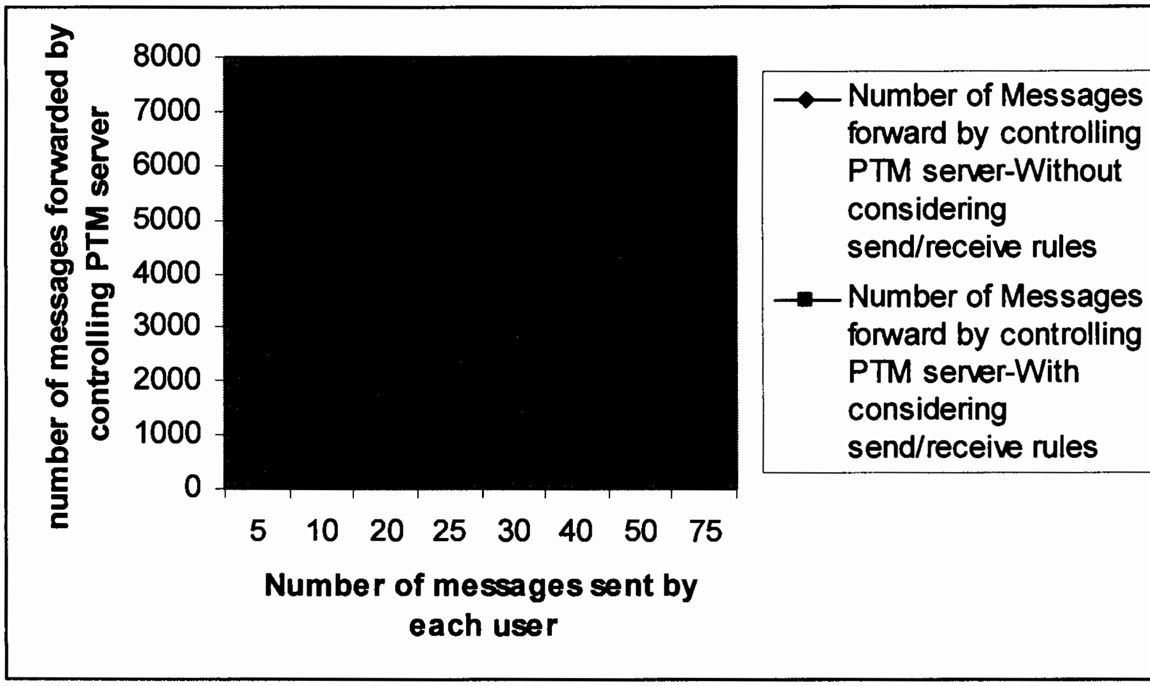


Figure 6: Number of messages Forwarded by controlling PTM server

In the figure 6 we plot the graph between the number of messages forwarded by the controlling PTM server in case of current scenario and proposed scenario. In current scenario a message sent by one user in forwarded to all other 9 users so if total 10 messages are sent by one user and there are total 10 users then total messages forwarded by controlling PTM server are 900. While in our proposed scenario we have seen that due to specified rules each user's sent media is forwarded to selected members only. So if user A sends a message it will be forwarded to only 3 users as shown in table 5. So if every user sent 50 messages total number of messages sent by controlling server in current scenario is 4500 and total number of sent messages by controlling server in case of proposed scenario is 2150. So total number of messages sent in proposed scenario are almost half as compared to that in existing scenario.

5.3 Achievements

After applying election based algorithm we achieve the following results.

5.3.1 Scalability

The election based algorithm improves the scalability of the PTM session. Participants can be selected with different rights such as a listener, a watcher and a speaker, as a speaker and a watcher, as a listener, as a speaker. Due to this strategy our results proves that more and more people can participate in the PTM session.

5.3.2 Resource Utilization

Our election based algorithm proves that the distribution of rights between the participants fairly utilizes the resources.

5.3.3 Privacy

The participants selected as a member of the PTM session can restrict the other participants for sending and receiving media. Our strategy proves that participants can easily share their secretes with those participants for which they want to share.

5.3.4 Disturbance

As we know that participants can restrict the other participants for sending and receiving media. Our strategy proves that participants cannot receive any information from the irrelevant participants. so participants can not disturb by the other participants.

5.3.5 Network Load

Our strategy proves by distributing the media services between the participants reduces the load on the network.

5.3.6 Wastage of Resources

In our technique participants join the session with full rights and limited rights. This strategy proves reduction of resource wastage.

5.3.7 Covers Real Life Scenario

Our strategy proves that we can implement this solution in real life scenario. Our solution fairly distributes the media services between the participants.

6. Conclusion and Future Work

6.1 Conclusions

As we know from the last few years among IMS has gained much more popularity. But this type of network face several kinds of applicability problems such as load on the network, low level of scalability, less privacy. To minimize these disadvantages is itself challenging due to its active and less transportation nature. In our thesis we have to deal with the applicability of PTM by using IMS. This type of networks ceases the entire network due to load on the network and reduce the scalability.

PTM, a one of the main services provided by the IMS, allows the user to form a group and share information in half duplex mode. But by using this type of services, participants have to face various kinds of problems. Sender is not authorized to send his information to selected members and all the members of the group have same rights. These two mechanisms reduce the applicability of the PTM in daily life, increase the network load, reduce the scalability etc.

We can use different type of process to reduce the load on the network and increase the scalability. But we know these types of techniques might not be perfect and may be unsuccessful to avoid such kinds of problems which the PTM face. So in this case we need such type of mechanism in which we choose the type of multimedia services which we want to use. Such type of techniques increase the applicability of PTM in daily life, decrease the load on the network and increase the scalability of the group.

So in order to increase the effect of applicability of PTM in daily life we have to introduce the idea of choice casting techniques to deliver information to the relevant users only. These techniques also introduce the fair distribution of PTM services between the participants i.e it allocate the services on merit basis.

Using the Choice Casting techniques the multimedia services are distributed among the participants on merit basis and this increase the scalability of the PTM session. These techniques are also used to reduce the load on the network and also maintain the privacy of the participants.

We have formulated polling result calculation algorithms. These algorithms give different rights to different users according to their status in PTM session. The algorithm chooses different type of users. These are the user with full rights (watcher, listener and viewer), user with half rights (watcher and listener, listener and talker). And only one rights like listener, watcher and viewer.

All these algorithms are based on the simple mathematical formulas which require simple processing capabilities, thus distribute different rights to the different participants according to their status. Due to these algorithm server can allow that he is eligible to join the PTM session or not.

To achieve this target we have evaluated these different strategies in VB.net in order to calculate the algorithm that calculate the different rights and assign these rights to the participants in most appropriate way and under what situation.

6.2 Future Work

In this thesis we have focused on applicability of PTM by using IMS. Enhancement in any area is always welcomed. Much of the work can be still done regarding the applicability of PTM, and distribution of PTM services can be more refined as some points are discussed as below.

In our work we have tested the choice casting technique to distribute the media between the participants. This choice casting technique can be tested in an environment where presence services exist, i.e. It provide the list that which user initiate the communication, who is available and wants to communicate. We could then determine the performance of our these techniques with presence service and also analyze this that whether all of these techniques could efficiently cater with this or need some modifications to cope with this situation.

In our work when a new participant want to join this session and send the join request with full rights to the server. But a new participant can send the join request with limited rights. This is an open area for future considerations.

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