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"Comparison between VoIP clients"

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ABSTRACT

IES Italia service utilizes multicasting technology to deliver large amounts of content to many ships at once, overcoming the prohibitively high satellite communications costs typically charged to deliver files for individual use. The new service (patent pending) is notable for numerous technological advancements: Content is delivered over the top of the network so there is no charge for the delivery, only for the content itself; the multicasting transmission does not affect the vessel's mini-VSAT Broadband onboard data speed; the service ensures digital rights management (DRM) of copyrighted material, such as Hollywood movies and television programs; and the content is delivered using forward error correction to minimize burden on the mini-VSAT Broadband.

Voice over IP (VoIP) and wireless are revolutionary technologies by all means of modern time which change the attributes of communications dramatically. VoIP has been established as potential alternative to tradition public switched telephone network (PSTN) technology whereas Wireless communication is the most widely used access method where fixed or remote access to network resources is important. Since both technologies have shown their existence in today's market individually, merger of these technologies was necessary and hence both technologies are being deployed but the question is whether these newly merged combination of technologies will be able to serve up to the same level of expectations and survive in current economic downturn and helps the companies to cut their cost drastically or will be finish with the time.

IES Italia comes up with the VOIP application named as VAVE, IES created VAVE Application through which passenger can make a VoIP calls using a VoIP client called C sip simple an open source VoIP client, In this thesis I will be comparing different VOIP clients which we experienced while configuring the VoIP Application(VAVE). This VoIP application is available on the Android store, so passenger can easily download and install on their smart phones, laptop and Pads after installing they are free to make VoIP calls.

CHAPTER 1:

1) INTRODUCTION TO IES ITALIA:

IES Italia is an SRL Founded in 2005, they believes in new technologies as a solution to business support. Integration of cutting edge tools, featuring great application versatility and able to interact with users in real time, allows producing customized communication and marketing solutions. Digital Signage, NFC, Wi-Fi, kiosk, VoIP, Bluetooth, beacons, are all tools that allow us to develop high-quality and high-impact solutions value added, that can make any business more competitive with success guaranteed by the meeting of creative, technological and planning know-how.

1.1) IES PRODUCT PLATFORM SOLUTIONS

IES Italia has been continually investing in making owner protocol language and innovative platforms where Wi-Fi, real streaming, beacons, RFID, cloud, mobile apps, digital Signage can converge to offer relevant answers to the specific needs of the Maritime, Hotel and Hospital industries). These platforms allow the delivery of content through IP, apps and web portals, all manageable from a single central location either in remote or in local. Flexible, scalable, customizable, built around the client's needs, IES system can be modulated according to the requests. We work with a team of software developers, using technology that's leader in the “media” field, believing that offering unique content and services creates appeal, localizes the client and differentiates each offering.

1.2) Maritime (MCP)

Multimedia content and Services for the Passengers, Communication solutions and Business opportunities for navigation Companies, Maritime Communications was our Customer for VAVE Project. We Devolved a VoIP client based Application name VAVE through which a Passenger can make a VoIP call when internet is around.

1.3) INTERNET SURFING ON THE CONNECTED CRUISE

Marketing is satisfied customers and the crew onboard your cruise ships. The urge to share great experiences needs to be satisfied, and IES enables you to connect via the Internet to a highly receptacle audience, and increase revenue at the same time.

1.4) Adding Values to Voyage

Passengers demand to use their mobile devices on board. By enabling them to connect, we not only generate revenue, but can also deliver real-time marketing campaigns and allow passengers to promote their experiences onboard your vessels

1.5) Increasing Revenue

The business benefits are increased revenue, better passenger service and a far better solution to interact with everyone on board. Additional services are developing, and where there are great experiences, great opportunities for growth in revenue follow.

1.6) Strategy and Solutions

Supports your business strategies. Your offshore communications strategy connecting with the onshore world in a seamless manner that intensifies experiences onboard telling the story of the voyage in real time. Making an effort to create great opportunities for both passengers and crew will make business thrive.

1.7) HOSPITAL-IES

A power full tool to manage digital content and provide and Medical Applications, In order to improve communication comfort and Efficiency, Hospital-IES is a project in which we deliver a multimedia contents to the patients in a Hospital.

1.8) IES-WEB

Advanced Web Platform to develop Website and portals targeted to content delivery streaming widget and Apps.

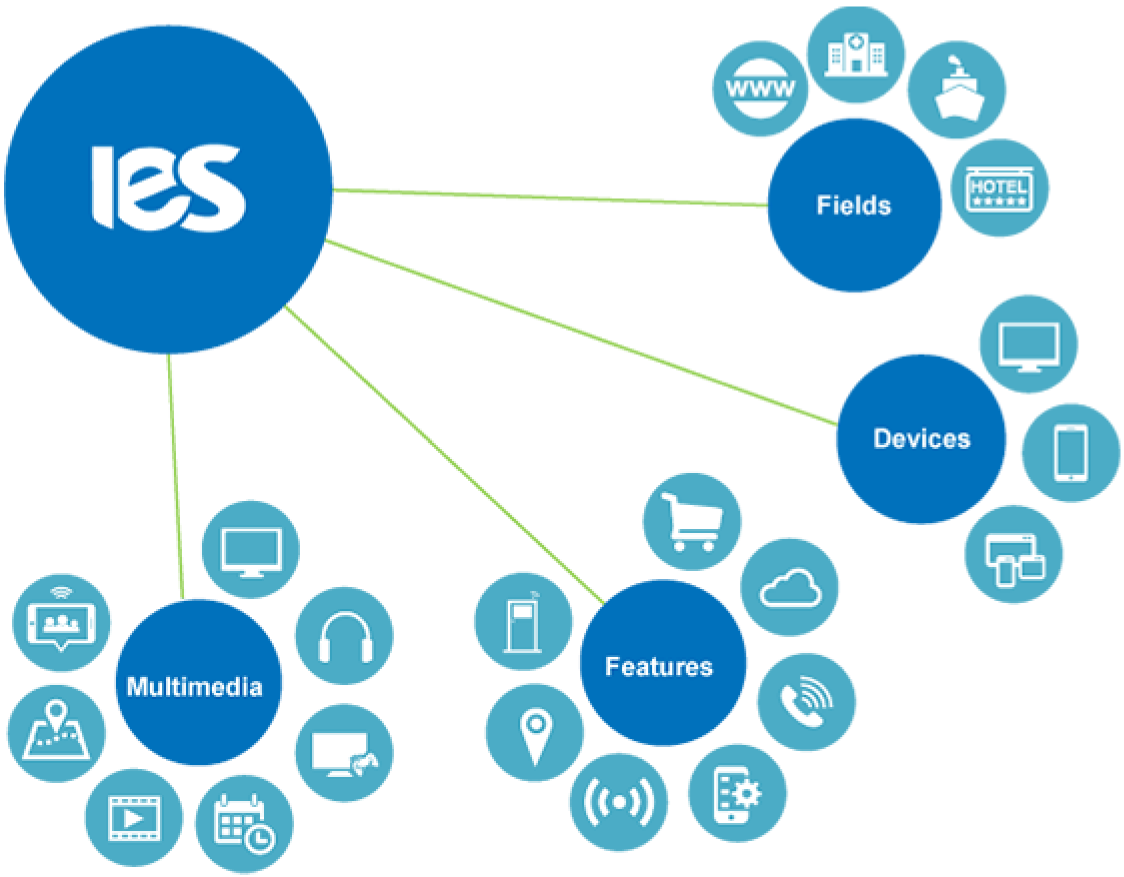


Fig 1.1 Features of IES

1.9) FOCUS ON NEW TECHNOLOGY

IES Italia believes in new technologies as a solution to business support. Integration of cutting edge tools, featuring great application versatility and able to interact with users in real time, allows producing customized communication and marketing solutions. Digital Signage, NFC, Wi-Fi, kiosk, VoIP, Bluetooth, beacons, are all tools that allow us to develop high-quality and high-impact solutions value added, that can make any business more competitive with success guaranteed by the meeting of creative, technological and planning know-how.

1.9a) Technologies Provided by IES

- Wi-Fi
- Digital signage
- APP
- Streaming
- Kiosk
- Location based
- NFC
- Beacons
- Streaming
  - Elemental
  - Appear TV
- Cloud
- E-Commerce

1.9.1) WIFI:

Management and Configuration of Wireless Architecture, It enables to connect the main server to the shore and from the shore to the ferry or cruise, Wi-Fi is the backbone to provide the contents on the ship to ensure the fast and compact communication.

1.9.2) Digital Signage

Real time Broadcast of video content, Interactive touch screen Kiosk configuration. Digital signage use Technologies like LCD, LED, and Projector to display Digital images, Video, Streaming Media and information can be used in public places. Digital signage displays use content management system and Digital Media Distribution systems which can either be run from Personal computer and servers or regional /national media hosting providers.

With the Help of Digital signage Passenger can easily view all the Application on the LCD and can be entertained by the application which he wants to use.

1.9.3) Applications (APP)

Android and IOS apps, Enlargement of the circulation of content and services. They are many apps which is provided by the IES Italia Team which can make any person's journey enjoyable, currently the main focus on the VAVE App which gives you an opportunity to connect to anyone using the IP based network. VAVE App is based on VoIP and we configure a VoIP client named C sip simple for the smart Phones and make this available on Android and IOS.

So any passenger can Download this directly from App store and the play store just they have to get the login id from the Hostess and then he can Call to anybody using the VoIP .Just you need to have an internet around and you can make a call up to the time you want, VoIP make a solution easier and cheaper instead of the tradition TDM based calls which are Costly and usually some connectivity issue during on board services.

1.9.4) IP TV

Video content Broadcast through Web streaming and playlist setting. Internet Protocol Television (IPTV) is a system through which television services are delivered using the internet Protocol suite over a Packet switched network such as a LAN or the Internet, Instead of being delivered through traditional satellite signal and cable television formats.

IPTV services Classified into three main groups.

- Live television with or without interactivity related to the current TV show.
- Time shifted television catch up TV (replays a TV show that was broadcast hours or days ago) start over TV (replays the current TV Show From its beginning).
- Video on Demand (VOD) a catalog of videos not related to TV programming.

IP TV is Defined as Multimedia services such as television /video/audio/text/graphics/data delivered over IP based networks managed to provide the required level of quality of service and experience ,security and reliability.

IPTV is defined as the secure delivery of multimedia contents and related services.IES Italia used the IP TV technology in order to deliver the multimedia contents to the passenger so that they can feel at home while they are on board travelling with different other passenger.

1.9.5) KIOSK

Multimedia touch screen totems, info and promo content, digital signposting. Kiosk is the technology now a day's bringing to many devices in smart phones pads, Laptops and TV this enables you directly touch the screen and access the application you want to explore.IES use Kiosk technology in order to provide more reliable and comfortable solution to the passenger to have a better and fastest vision of the applications provided by our young and energetic team.

1.9.6) Location Based Service

Geo location through smart phone beacons configuration NFC services. Location based services are a general class of computer program-level services that use location data to control features. LBS is an information service and has a number of use in social networking today as an entertainment service, which is accessible with mobile devices through the mobile network an which uses the information on the geographical position on the mobile device.

1.9.7) NFC

Access control and mobile payment interactive kiosk and tags configuration. NFC is a form of short-range wireless communication where the antenna used is much smaller than the wavelength of the carrier signal, antenna can produce either electric field or magnetic field but not the electromagnetic field, so the NFC communication either the electric modulated or magnetic modulated.

NFC is builds upon RFID systems by allowing two way communications between the endpoints.AS NFC is RFID based system so we have tags which can be read by the NFC devices, NFC devices can be used in contactless payment system its similar to those currently used in credit cards and electronic ticket smart cards and allow mobile payment to replace or supplement these systems.

NFC application is developed by the IES team so that if any passenger wants to do some payment, so they can easily access the APP NFC and while during their journey they can make the important payments.

1.9.8) Beacons

Bluetooth based micro-localization, beacon is an intentionally conspicuous device designed to attract attention to a specific location.

Beacon can be used for many purposes.

- Navigation
- Defensive communication
- On vehicles

Beacon help guide navigators to their destination, navigational beacons include radar reflector radio beacons sonic and visual beacon, while knowing about the Knowledge of beacon IES team able to provide a one handy application to the passenger.

Beacon help to Provide navigation in the Cruise if some passenger is lost in the cruise using the navigation he can find his seat also with the help of navigation they could know about where they are and how long they have to go, navigation also help to catch the hostess and amazing places of the cruise which involve the basement, club, common sitting rooms and coffee places these kind of application make the journey easier for the passenger also exciting as he is exploring different parts of the cruise which cannot be easy without these kind of Applications.

1.9.8. A) Streaming

IES Italia team provide one other application called live streaming which gives the passenger Live and on demand video Broadcast, Live streaming refers to the content deliver live over the internet requires a form of source media (e.g. a video cameraman audio interface, screen capture software) an encoder to digitalize the content and a media publisher and a content delivery network to distribute and deliver the content.

IES chooses the best encoder called elemental which is used for encoding.

1.9.8. B) Elemental

Elemental is a video processing Platform component that provides real time video and audio encoding for linear pay TV Broadcast and live streaming to new media platforms. The software based solution performs simultaneous processing of multiple video outputs delivering the High efficiency performance required for formatting live video for any device. Elemental live is designed to integrate seamlessly into an end real time video delivery workflow, evolve as technology requires and maximize revenue opportunities.

1.9.8. i) HIGH PERFORMANCE

Deliver content via Apple HLS, Adobe Primetime (HDS and RTMP) Microsoft smooth streaming MPEG-DASH or transport streams. Alternatively mezzanine deliverables for wrapping with a separate packager as Elemental stream to reduce network bandwidth.

1.9.8. J) Software Defined Architecture

Address today's opportunities while laying the foundation for future video requirements. A highly upgradable video platform provides adherence to regulations such as the CALM Act and rules for IP closed captioning, a seamless transition to new standards such as HEVC and Ultra HD, and ongoing improvements to features, functionality and quality with each software release.

1.9.8. K) Versatile Deployment

Control the Linux-based software through an intuitive web interface or REST/XML APIs for quick and simple integration in a turnkey, cloud-based or virtualized deployment. Unified control and management with Elemental Conductor reduces setup time, simplifies maintenance tasks and allows for centralized upgrades of multiple deployments.

1.9.8. L) Seamless Cloud integration.

Expand video processing as needed to flex with variable demand. Integration with Elemental Cloud replicates the profiles, capabilities and formats used on premise so that video outputs are identical regardless of where they are processed.

IES Italia uses an APPEAT TV an Encoder/Transcoder which is used to transcode the data from the satellite.

1.10) Appear TV

1.10. A) Power and flexibility in signal acquisition and distribution.

Appear TV offers high-performance equipment designed with the needs of satellite operators in mind. Our product portfolio can be used in a number of applications. Flexible, high functionality and powerful solutions help enable operators to build tomorrow’s solutions.

The XC5000 can receive signals from a variety of sources including satellite (DVB-S/S2), terrestrial (DVB-T/T2), cable (QAM), IP networks or local ASI feeds. Selected services are then descrambled (and scrambled) and multiplexed before being fed to high capacity streaming output (IP or ASI) modules to feed third party multiplexers and modulators.

1.10. B) Streaming between signal acquisition and uplink locations

The modularity of Appear TV solutions makes them ideal for operators who need to perform signal acquisition at a different location before multiplexing and up linking, thereby requiring transmission between sites. Using Appear TV equipment, operators can acquire, descramble, scramble and multiplex content over QPSK in one location, and stream it over IP or ASI to the uplink location

1.10. c) IP content Acquisition

Appear TV offers the most complete, flexible and scalable play out solutions. Our head-ends enable operators to acquire content over IP (for instance directly from studios) over a high capacity IP link. Content is descrambled and selected services scrambled if required. Streams can then be distributed via IP or ASI to IRDs for decoding and re-encoding, or directly to a multiplex.

1.10. D) Flexible scrambling solution

Appear TV's powerful scrambling solution enables encryption (CSA and AES scrambling) and streaming. The streaming solutions support MPEG-2 and MPEG-4 for HD and SD services.

1.10. E) Service Monitoring

Appear TV head-ends offer superior service monitoring solution. They enable high density of audio-visual outputs (up to 30 per chassis). In addition, the intuitive web management interface offers simple input statistics as well as alarm monitoring.

1.11) CLOUD

Virtualization of data, hardware and software resources on web IES team uses Cloud Application through which an IES can have an access of hardware and software, also all the documents, video, Pictures can be shared through Cloud.

Passenger has the opportunity to upload their pictures documents and video during their journey to their loved ones.

IES team uses the Cloud to share the important documents, Presentation and new projects that can be easily access by all the member of IES.

1.12) E-Shop

E-commerce integration, front end and back end configuration. E-commerce is trading in products or service using computer networks such as the internet, using e-commerce following functions can be performed such as electronic funds transfer, internet marketing, online transaction processing electronic data interchange and automated data collection.

IES team give one very interesting application to the passengers named e-commerce through which a passenger can do online shopping using the network and make their journey more exciting.

Chapter 2

2.1) ARCHITECTURE OF IES ITALIA

The Architecture of IES is to design and implementation of network infrastructure solution to design the digital radio, optimization of existing network infrastructure solutions to support digital radio.

Optimization of existing Networks to improve connectivity, solutions for data center and e-commerce, open source software development and integration creation of IOS/Android Applications, Planning cable networks of licensed radio systems, integration of Virtualization technologies, cloud computing.

Setup and activation of cable and wireless network connections, Production of GSM, UMTS, 4G data networks, Testing roll out, commissioning, operators training, Planning and set-up off radio networks: Hyperlink, Mesh, radio links PDH and SDH, Efficiency Enhancing Attainment, Project and realization of broadcast networks for audio-video signal diffusion, project and realization of security system, CCTV system for video and traffic control

These days Virtualization is the process which can make the networking easier reliable faster and with a lot of capacity, instead of having so many wires and big servers rooms with so many computers, virtualization is the process which can control many devices and machines without existing physically just to create the virtual machines which can control by virtual machine manager, can be monitor easily through this manager can be easily ON OFF and configured delete through this manager and can easily assign a big storage to the machines. IES Italia have to handle so many devices and application in order to maintain the quality and make the life easier for the IES team, We concentrated on the LAMP(Linux APPACHE, MYSQL, PHPMY ADMIN), With the help of this project it was easy to control devices and machines.

2.1) LAMP (LINUX, Apache, MySQL, PHP)

Lamp is an acronym for an archetypal model of web service solution stack, originally consisting of largely interchangeable components if instead of using Linux as an operating system we can use the window as an operating system is known as WAMP. If you are not using Apache as a web server than Ngix server is another opportunity to use and then project become LEMP (Linux Ngix MySQL and PHP)

2.1. A) Linux

Linux is the operating system assembled under the model of free and open source software development and distribution Most Linux distribution as collections of software based around the Linux kernel and often around the package management system, Provide complete LAMP setup through their packages. IES team uses Linux to install the base server and other machines which can be control through virtual machine manager.

Development of Linux is one of the most Prominent example of free and open source software collaborations, IES uses Linux as platform to install their servers and apache as a web servers in order to launch their virtual hosts (dashboard, ads, and manager), and MySQL is the data base management that can be managed by the Perl/python language.

2.1. B) Apache Server

Apache is HTTP server Project to develop and maintain HTTP server for modern Operating System including windows Unix Linux and open UMS, Goal of the Provide a secure efficient and extensible server that provides HTTP service in sync with the current HTTP standard.

Apache is software that performs the functions of information transport, Http.conf allows access to one or more sites, mapping different security features and being able to accommodate diff extension for active page as PHP or Jakarta/tomcat.

Apache web server whenever gets a request of the Client every module which it composed as independent unit. Each module deals with functionality and control is handled by the core.

Apache servers perform the functions in the following way explained through the Diagram as seen below.

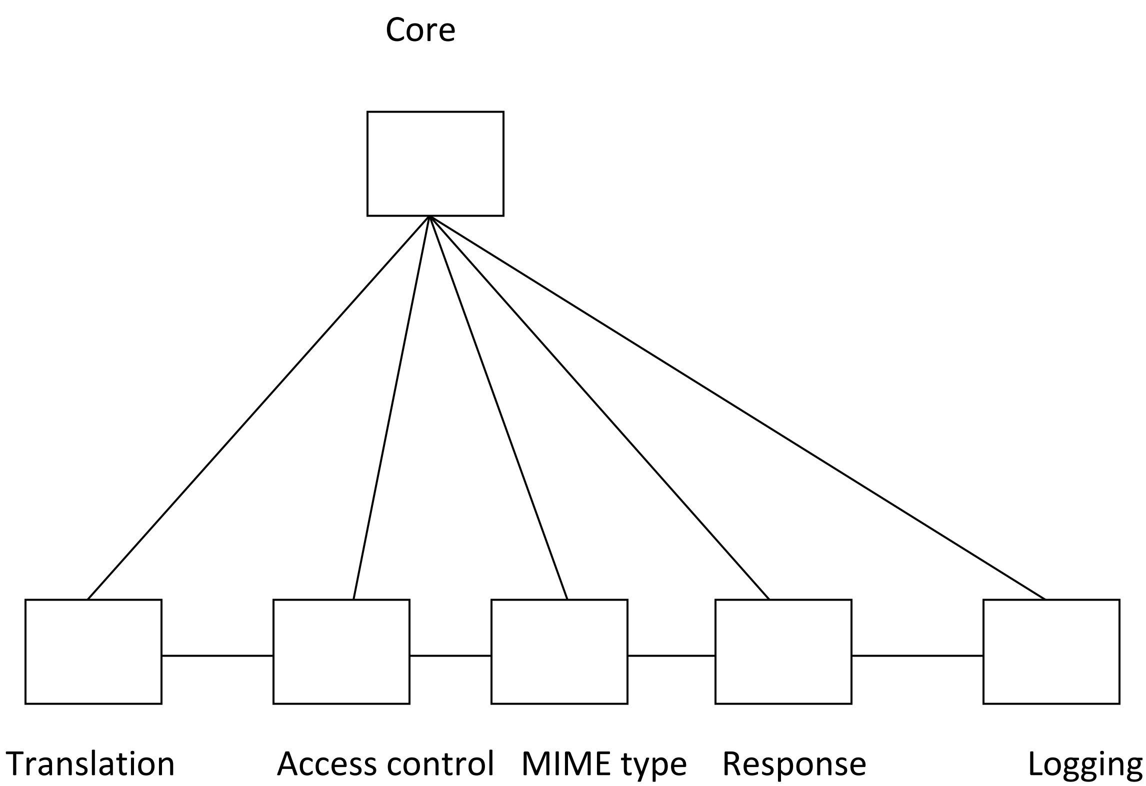


Fig 2.1 Core of LAMP

Core: Main program consists of series of sequential calls

Translation: Translate the call request

Access control: Control any malicious request

MIME Type: Check the type of contents

Response: send the response back to the client and active procedure.

Logging: Keep tracking of everything that has been done.

2.1. C) MYSQL

Open source data base use in web Application central component of the wording use LAMP, data management system MySQL, MySQL is a relational data base management system with no GUI tools, MySQL is written in c and C++.

2.1. C. a) Limitations

MySQL does not currently comply with the full standard for some of the implemented functionalities.

2.1. C. b) Deployment

On most Linux distribution the package management system can downloaded and install MySQL with minimum effort through further is often required to adjust security and optimization settings, MySQL can also be run on cloud computing platform, Clod user can upload a machine image of their own with MySQL installed.

2.1. D) PHP My Admin

PHP My admin is a free web base front end widely installed by web hosts it is developed by PHP, PHP is an open source tool written in PHP intended to handle the administration of MySQL with the use of web browser, it can perform various tasks such as creating, modifying, deleting, tables, fields and rows, IES team working under LAMP to provide a secure transmission and handling the complex infrastructure.

AS to explain the architecture of IES Italia how it works it will be explained by another figure which explains the different virtual machines which are controlled by the virtual machine manager the figure you can see below.

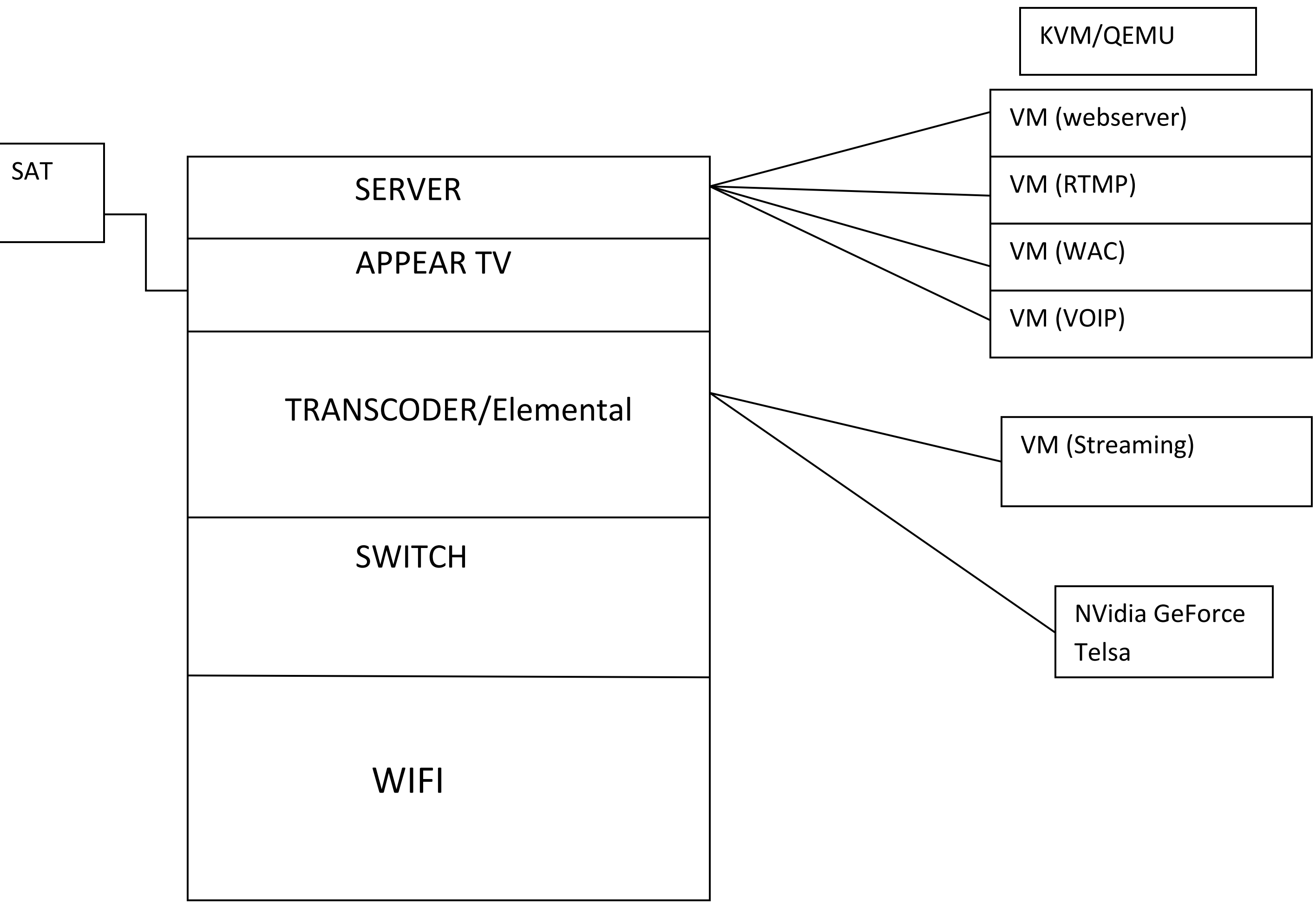


Fig 2.2 Architecture of Different virtual machines

Following are the Virtual machines which are explained below.

- WEB server
- RTMP
- Streaming

- WAC(wireless access control)
- VOIP

2.2) KVM/QEMU

KVM is a kernel based virtual machine is a full virtualization for Linux on x86 hardware containing virtualization extensions (Intel VT or AMD-V)

2.2. A) WEB SERVER (VM)

Web server VM (virtual machine) is created to control the main web server using the Raid server which have a lot of storage and virtual machine is helpful in controlling the main server and have a lot of storage to upload the data using the FTP (file transfer Protocol) as some time it is difficult to control the server, with the help of virtualization it's easy to on and off the main server using the Virtual machine manager.

2.2. B) RTMP

Real time messaging protocol is used to deliver an audio video and data over the internet, RTMP is a TCP based Protocol, It uses to deliver streams smoothly and transmit as much information as possible in this way it splits the stream into the fragments and size is negotiated between the client and server for audio data the size is 64 bytes while for the video fragment size is 128 byte, RTMP encapsulate MP3 and AAC audio and FLV1 video.

RTMP Virtual machine used to have an access virtually to control the contents delivered its directly in contact with the elemental which is used to transcode the data from satellite and RTMP is the Protocol which is used to deliver an audio and video data coming from satellite.

In RTMP data is encapsulated and exchanged via HTTP and message from the clients are addressed to port 80, to start a video stream the clients send a create stream innovation followed by a ping message, Followed by a Play invocation with the file name as argument, the server will then reply with a series of on status commands followed by the video data as encapsulated within RTMP message.

The Flash player which is used to display the audio and video content is Adobe Flash Player.

2.2 .C) Streaming:

Virtual machine available for streaming from web server, the stream computing separate the Applications computational kernels from communication streams, Matching the structure and performance constraint of modern multiprocessor.

Streaming virtual machine is called a peer to peer machine as we have Void to present to the passenger by creating the Virtual machine it's easy to deliver content which usually passenger requested, P2P computing is process in which the task is divided between the peers, peers are very good in arranging resources such as power disk storage and network bandwidth, peers are both supplier and consumer of resources.

With the help of P2P machine it was very much easy to handle the large amount of data as P2P network along with streaming Servers to stream audio and video to their clients.

2.2. C) WAC (Wireless Access control)

Wireless access control virtual machine minimizes the hardware cost while optimizing wireless network performance with centralized control. The virtual wireless controller optimize the performance of local and branch wireless network, virtual wireless controller optimize the business and also good in serving many access points.

Virtual wireless controller offers:

- Centralized wireless network visibility and control for up to 200 branch locations.
- Ability for IT managers to configure manages and troubleshoots up to 200 access points and 300 clients.
- Protection of access points connected to remote controllers from branch WAN failures, Wireless clients remain connected with access to local resources.

IES team built a Wireless access control virtual machine in order to have a secure and efficient control of the wireless access on the cruise so all the passenger will remain connected to the internet and keep on enjoying the amazing Apps of the IES without any interruption.

2.2. D) VOIP VM:

Despite challenges VoIP VM is able to achieve great performance for VOIP applications; it facilitates the highly optimized networking stack and par virtualized device drivers to minimize the virtualization overhead adding little variance in packet delivery. The overhead is usually in order of tens of millisecond that are negligible, especially in VoIP applications where packets need to be delivered at intervals of tens of millisecond, VM gives a fair share of CPU ensuring the predictable processing of audio data even under high CPU contention when running multiple VM.

VoIP VM do not affect the quality of voice, It is maintained when the number of users and media server instance increased while fully utilizing the CPU still able to manage the better voice quality and it depends on the VM ware you are using the quality of service is very important for the Passenger to have a quality time during their journey.

IES team built all this architecture with all these virtual machine in order to control these devices and machines easily with less equipment and the big server rooms with so many wires and cables which are also very costly and not very efficient.

All these virtual machine are controlled through the virtual machine manager ,so that can time you can switched off and on the virtual machine also you can delete and configure again according to your need.

The clear and Better view of Architecture of the IES Italia in terms of virtual machines, and how they are controlling all the infrastructure in a real world of virtualization and networking. Complete Architecture diagram is seen as below.

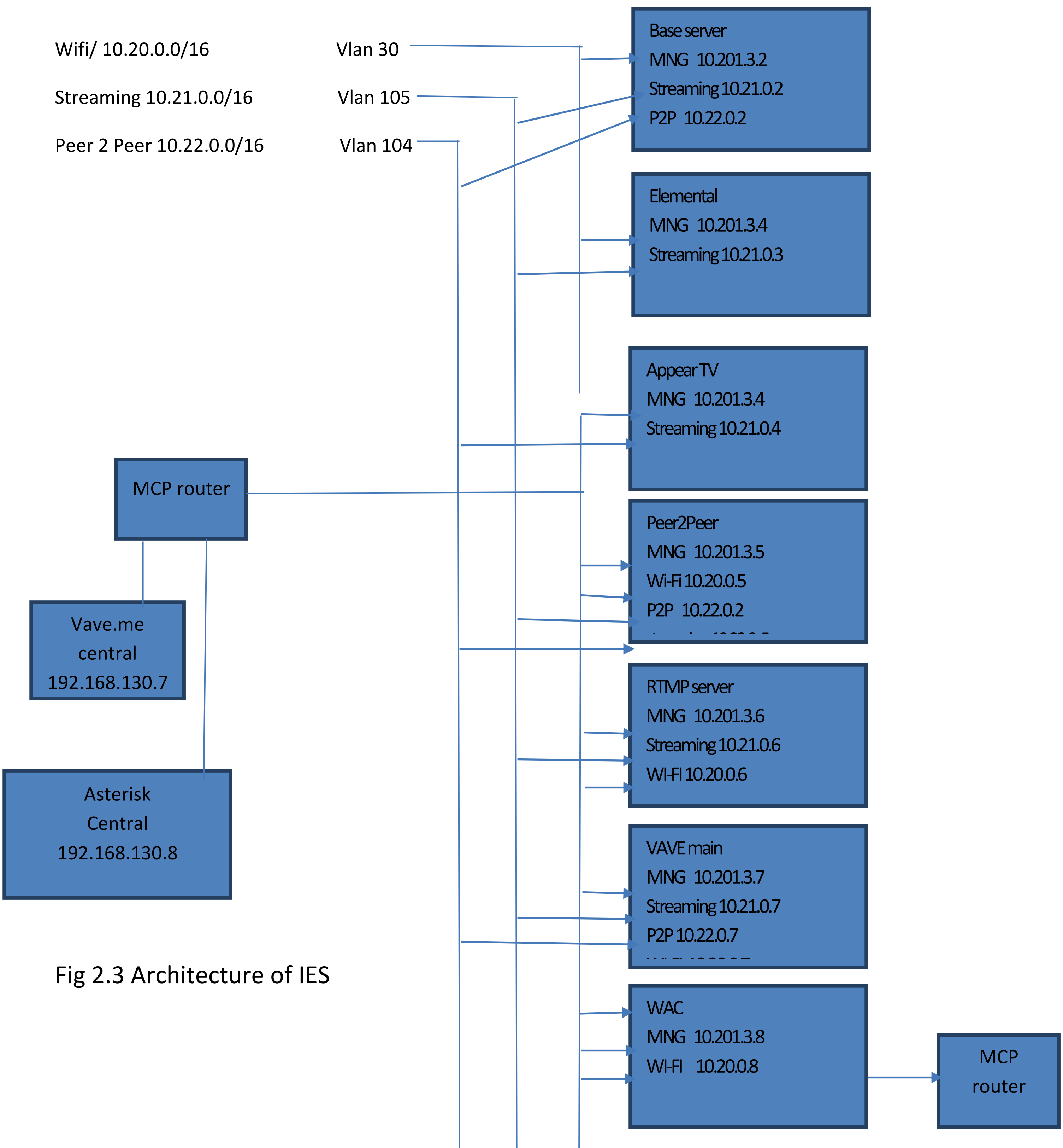


Fig 2.3 Architecture of IES

CHAPTER 3

3.1) VAVE (Value Added Voyage service)

VAVE is the application which is developed for the customer so that they can use IP based network in order to complete their calls which can be audio or video, configuring the VAVE application on their smart phones, as VAVE application is uploaded on the Apple and Mac store it's easy for the customer to download the application from the store and make a VOIP calls anywhere to anyone freely when they have the internet Around.

IES Italia ensures quality of service throughout the journey by providing so many exciting applications and out of these VAVE is one of the major and exciting App which makes the passenger to be in contact with their loved ones throughout the journey.

Fig 3.1 VAVE App

3.2) Introduction to VOIP

VOIP is called voice over internet protocol; this protocol is used to transfer audio and video data over IP network, commonly some terms associated with the IP telephony and internet telephony.

Internet telephony is the process in which voice, fax, SMS and voice messaging over internet instead on the traditional Public switched network. The steps and principle is same as in traditional digital telephony the only difference is that the digital data is converted into packetized form and routed over IP based network.

In VOIP we use codec's in order to deliver the audio and video stream over the Packet network, usually called as audio and video codec, various codec optimize the media stream based on application requirements and network bandwidth, implementations relay on narrow band and compressed speech, while other support high fidelity stereo codec, some popular codec includes G711 and G 722, they are usually referred as HD voice codec, only codec which uses 8 Kbit/s is g729.

VoIP is now available on smart phones tablets, laptop and PC; VoIP has been implemented in different ways by using different kinds of protocols.

Following are the Protocols we have as seen below.

- H.323
- Media gateway control protocol (media gateway control protocol)
- Session initiation protocol (SIP)
- Real time transport protocol (RTP)
- Real time Transport control Protocol (RTCP)
- Service real time transport control protocol (SRTP)
- Session Description Protocol(SDP)

Out of following Protocol we define some of them which commonly used in a standard VOIP calls.

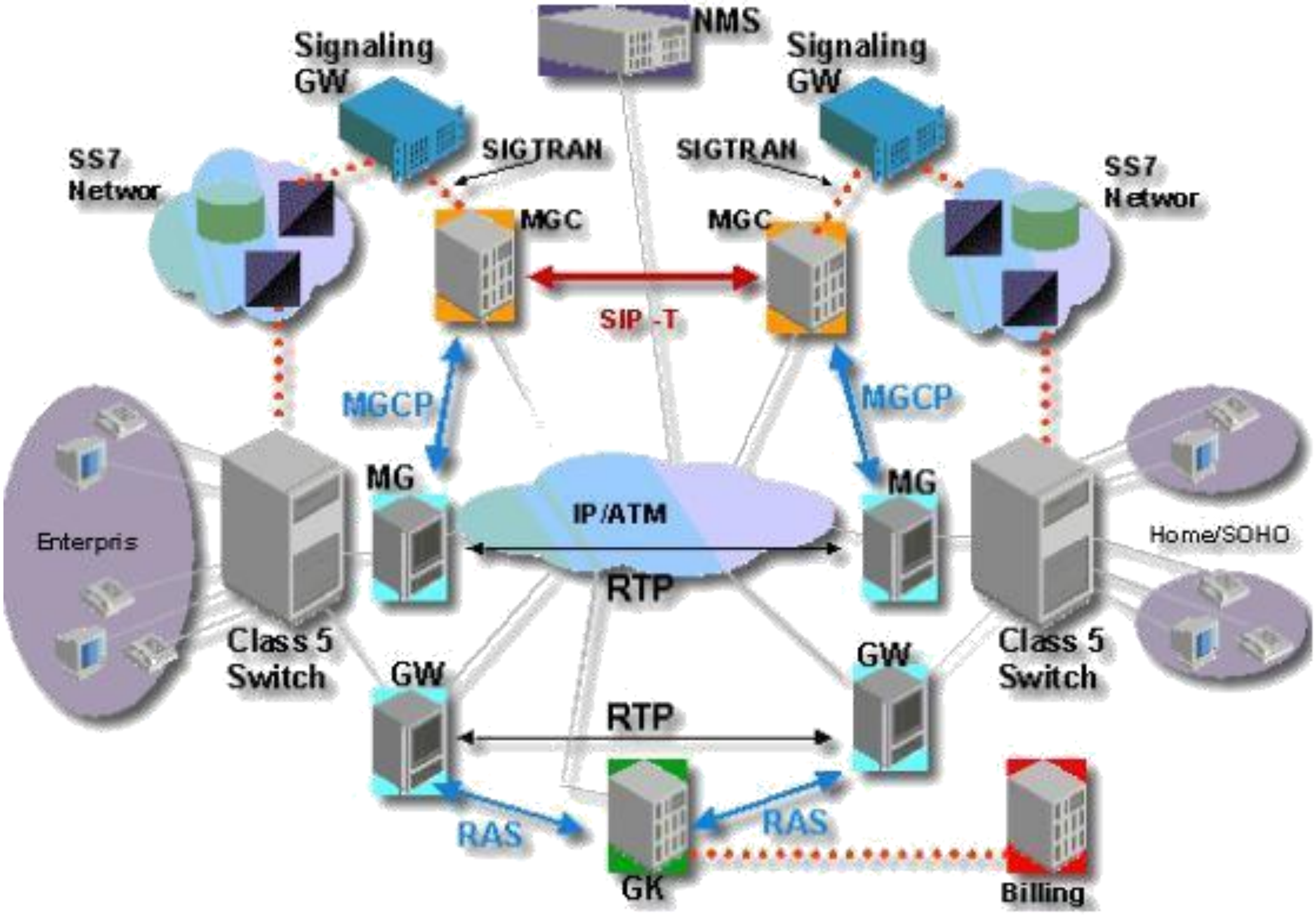


Fig 3.2 Basic Architecture of VOIP

3.3) SIP (Session Initiation Protocol)

Sip is a signaling protocol standardized by the IETF, it is used together with other protocols such as session description protocol (SDP), Real Time Streaming Protocol and session Announcement Protocol (SAP), SIP handles the setup, tear down and management of IP multimedia session.

Sip usually adopts RTP as a transport for media streams; sip signaling has a dedicated logical channel from the logical channels used for the transport of media on the user plane.

3.3. A) Sip defines two type of entity

Client

Called also user agent client, it is an application sending sip request.

Servers

Applications committed to the establishment of connections server respond to sip requests from client.

A sip call involves at least two remote entities.

User agent client

User agent server

A sip device can host both user agent client and server, this is typical case of user device.

There are four basic types of SIP server.

Proxy server

Redirect server

Registrar server

User agent server

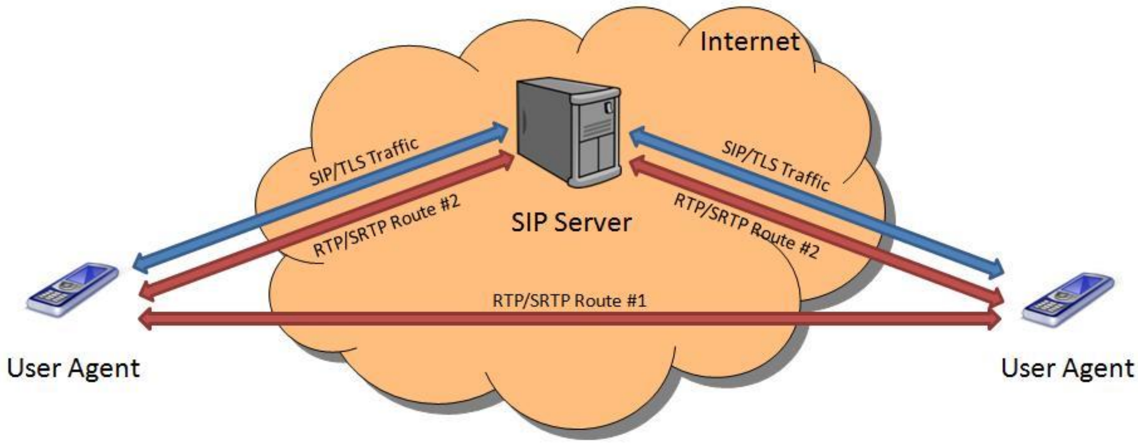


Fig 3.3 How Sip server works on IP based network

PROXY SERVER:

Proxy server can implement multiple types of sip server it can colloquia both with user device and other sip server in the network, through proxy servers it is possible to implement many value added service such as Call forwarding and time of day routing.

REDIRECT SERVER:

The redirect server can also provide the calling terminal with an alternative address to call instead of redirecting the call automatically.

USER AGENT SERVER

It is the server usually installed on user terminals, it host also user agent client, used to send signaling requests, user agent server receives incoming request and responds.

Registrar server

Registrar server handles sip register requests, sip register requests are used by devices to register into the sip network, after successful registration, sip device is operative.

3.4) Session Description Protocol

SDP is used for the description of the format of the media streams, For each media stream of a session, an SDP description is needed, SDP does not transport media it is used only for their description, SDP description are carried in the body of SIP messages.

STRUCTURE OF SDP DESCRIPTION

Session level information

Protocol version.

Originator and session ID

Session Name

Session Time

Media Description

Media name and transport

Connection information

SDP descriptions are included in the INVITE messages and in the following responses.

3.5) Real Time Protocol

The Protocol which is used to send real time stream of data across a network is simply called the Real Time Protocol. RTP has been originally defined by IETF in RFC 1889 and the up to date definition is in RFC 3550.

When transmitting the streams of data, Protocol needs to handle the following conditions in the network.

- The network can de-sequence packets
- Some packets can be lost
- Jitter is introduced

Out of these three RTP aims to solve only two issues, Packet de-sequencing and jitter (using sequence numbers and timestamps). When it comes to packet loss, the protocol prefers real-tameness to reliability. If some packet is lost, they get lost because RTP cares about the transmission in the real time, Because of this RTP works on top of UDP. TCP is not suitable for real time Protocols because of its retransmission scheme.

3.6) Real Time Control Protocol

RTCP accompanies RTP and is used to transmit control information about the RTP session, RTCP packets are send only from time to time since there is a recommendation that the RTCP traffic should consume less than the 5 Percent of the session bandwidth.

The most important content type carried in RTCP packets include:

Information about call participants (name and email address)

Statistics about the Quality of the transmission (For example inter-arrival and the number of lost packets) the report sent by the a participants who both sends and

Receive data is called a sender report, while report sent by participants who only receive RTP stream are called receiver report (RR).

There is a rule that RTP should use an even UDP port number (e.g. 5000) and the related RTCP should use next odd port (e.g. 5001)

3.7) Introduction to VoIP Clients

A VoIP client is a software application that is also called a soft phone. It is normally installed on a user's computer and allows the user to make VoIP calls. Through the VoIP client, can make free or cheap local and international calls and it gives a lot of features. These are the main reason many people install VoIP client, when installed on a computer, will need hardware devices that will allow the user to communicate, like a microphone, headsets, earphone and a webcam.

3.7. A) VoIP Service

A VoIP client cannot work alone. To be able to make the calls, it has to work with VoIP service or a server. A VoIP service is the subscription you have from a VoIP service provider to make the calls, a bit like your GSM service you use with your mobile phone. The difference is that you make the calls for very cheap with VoIP and if the other person you are calling is using the same VoIP service and VoIP client, the call is in many cases free and unlimited.

3.7. B) VoIP client feature

A VoIP client is software that carries many features. It many simply by a soft phone, where it would have a dialing interface, some contact memory, user ID and some other basic features. It may also be a complex VoIP application that only makes and receive calls but also contains functionality like network statistics, QOS support, voice security, video conferencing etc.

3.7. C) SIP VoIP clients

Sip is a technology that works on VoIP servers that offer calling service to machine that have a sip compatible VoIP client installed and registered, Sip VoIP clients are more generic and are not tied to any particular VoIP service. You can simply install one on your machine and configure it to be used with any service that offers sip-compatibility.

While developing the VAVE application we experience different VoIP clients but mainly focused on the two of the VoIP clients which is named as C sip simple and jitsi and finally we developed the VAVE application with C sip simple.

3.8) Introduction to C sip simple

C sip simple is a voice over internet protocol application for android operating system using the session initiation Protocol; it is open source and free software released under the GNU general public license.

The key features of this software are:

- Multi-codec support which contains G.711 G.729, GSM, ILBC, G729, G.722, and AMR.
- A plug in ads supports for codec2 G.726, G.722.1 and opus.
- Video calling with VP8, H264 and H263-1998 codec
- Compatibility with Android
- Security and encryption with SRTP, sip over TLS
- Sip messaging is available
- API for third party application is available
- Packet loss concealments (PLC)
- Support for IPV6

Two main codec’s we use explore the testing and configuration as explained below

- G729
- G711

3.8. A) G729

G.729 is an algorithm used for the voice compression, it compress the digital voice in packets of 10 milliseconds duration. It codec the speech at 8 Kbit/sec using code excited linear prediction speech coding, due to its low bandwidth requirement the most used in VoIP is G729 because in conference calls where the less bandwidth required G.729 is the best solution usually G729 is operate at 8 Kbit/sec there are some kind of extensions which are called Annex for better and worst speech Quality.

In G729 we have G729a and G729b they are not very much different from G729 but they have additional properties and they are useful and compatible in some conditions.

G729 original codec and uses high complexity algorithm.

G729a or Annex a medium complexity variant of G.729 and it is compatible with G.729, it is less complex but have the lower voice quality.

G.729 B or annex B gives silence compression and not compatible with the Previous one.

G 729 A silence compression and only compatible with the G.729 B

DTMF, fax transmission and high quality audio codec cannot be transported reliably with this codec. DTMF requires the use of RTP pay load for DTMF digits.

G. 729 are highly efficient in delivering the data requiring the less bandwidth but have some limitations in DTMF, Fax and HD voice.

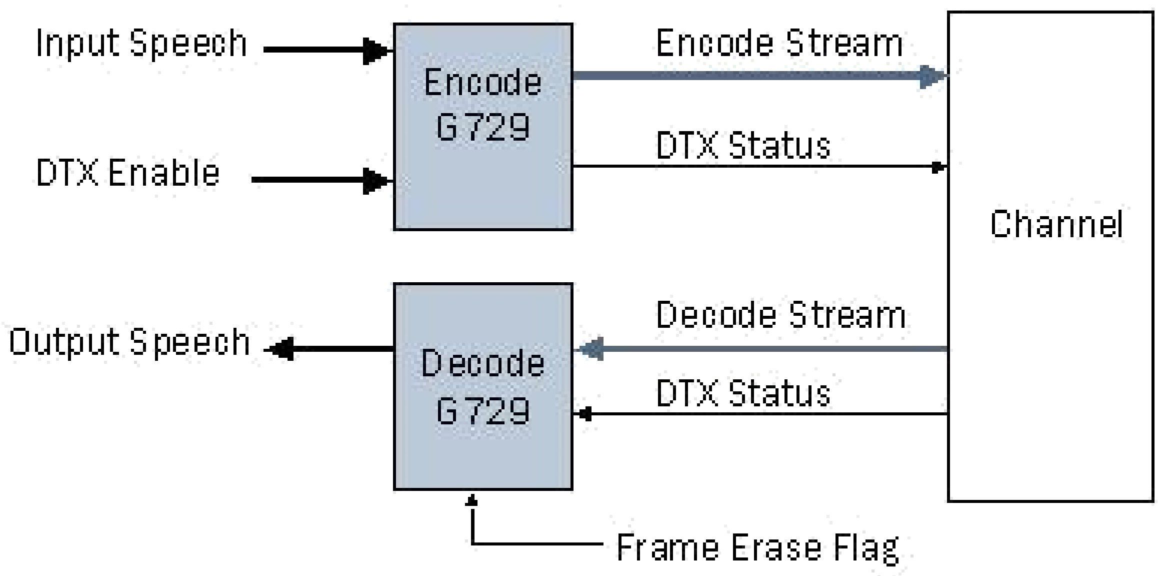


Fig 3.4 Block scheme of G.729

3.8. B) G.711

G.711 is a standard for audio companding, it is used in telephony it's another name is pulse code modulation for voice frequencies it is required for many technologies H.320 and H.323, it can also be used for fax communication over IP network defined as T.38 specification, its known as waveform codec, G.711 is a narrow band codec, it provides the quality at 64Kbit/second G.711 passes audio signals in the range of 300-3400HZ and samples them at the rate of 8,000 samples.

G.711.0 utilizes lossless data compression to reduce bandwidth usage and G.7.11.1 increasing bandwidth.

G.7.11 defines two main companding algorithms the U-law algorithm and A-law algorithm encode 14-bit and 13-bit signed linear PCM samples to logarithmic 8-bit samples. Thus G.711 encoder will create a 64 Kbit/s bit stream for a signal samples at 8 KHz.

G.711 U-law tends to give more resolution to higher range signals while G.711 A-law more quantization level at lower signal levels.

G.711 is good in providing a digital audio at 64kbit/s also allow the addition of narrowband and/or wideband enhancement.

G.711 is usually the first choice in local area networks (LANs), such as, within the same office and/or the same building. While G.729 is usually the first option for WAN, such as, between two branches in different cities, because its pay load requirement is only 8 kbps, while the requirement of G.711is 64 kbps [20]. Generally, engineers, who implement VoIP systems, under-stand that G.711 provides better voice quality than G.729. Therefore, in this study, these two codec's were focused. Details about G.711 and G.729 can be presented as normally, each VoIP application has a default codec.

While creating the VAVE Application we come across both the codec's as we work on the top level of the C sip simple it's an open source so we do not change the codec as we do not make changes in C sip simple we just configure for the VAVE app.

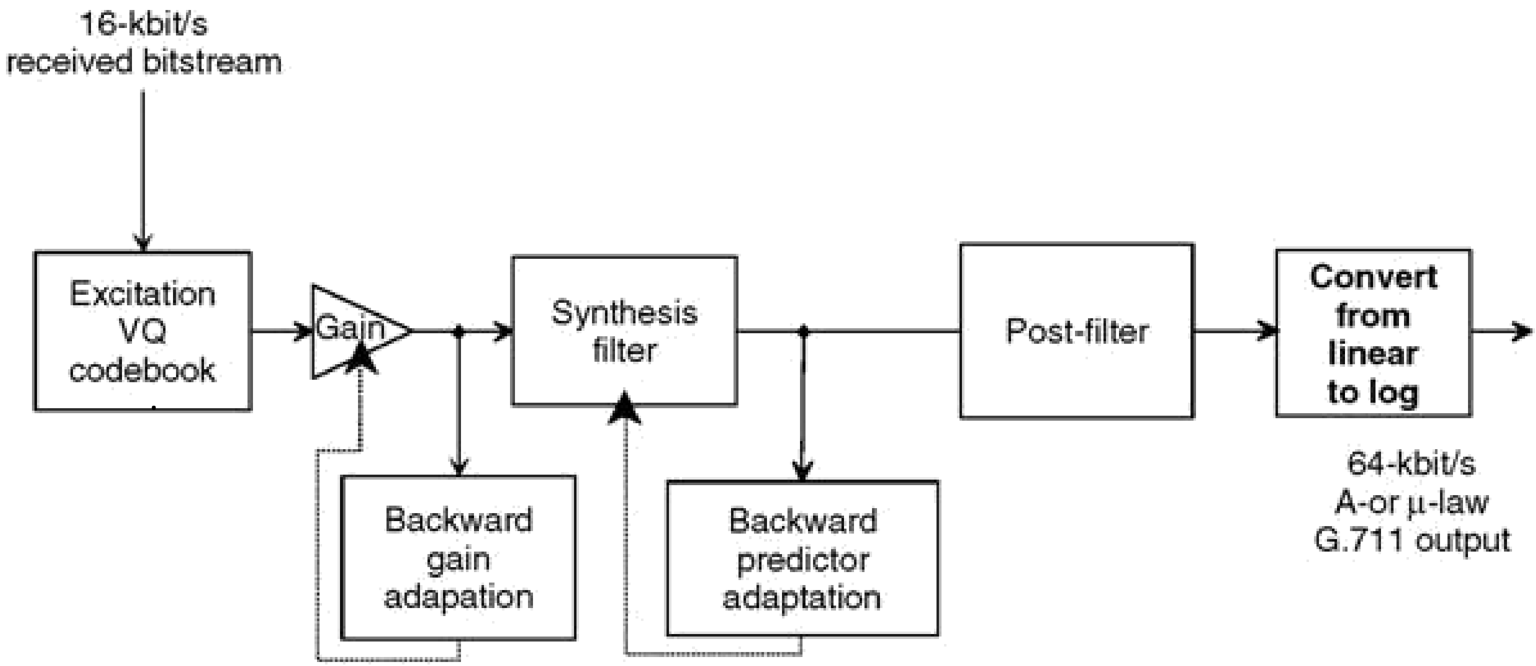


Fig 3.5 G.711 Block Scheme

3.9) INTRODUCTION TO JITSI

Jitsi is a free and open source multiplatform voice VoIP video conferencing and instant messaging Application for windows Linux and MAC OS it support several popular instant messaging and telephony protocols including open recognized encryption protocol for chat OTR and Voice/video/conferencing. Jitsi and its source code are released under the terms of the LPGL.

Jitsi supports multiple operating systems including windows as well as Unix MAC OS X and BSD.

3.9. A) ARCHITECTURE OF JITSI:

JITSI is written in Java which helps reuse most of the same code over various operating systems, it is GUI based system. Jitsi uses the native code for the implementation of platform specific task such as audio/video capturing, jitsi is compatible with the apache, among other jitsi uses the JAIN-SIP support

Jitsi can handle IPV6 it is especially interesting for direct PC-to-PC communication for instance, if both sides were trapped behind NAT routers, but could obtain a reachable IPV6 address via a tunnel broker.

3.9. B) FEATURES OF JITSI

- Attended and Blind call transfer.
- Auto away and auto reconnect auto answer and auto forward
- Call recording and call encryption with SRTP and ZRTP.
- Conference calls
- Desktop streaming
- IPV6 support for SIP and XMPP.
- Voice and video calls for sip and XMPP
- Message waiting indication
- DTMF Support with SIP info
- Group video support
- Packet Loss concealment

Following protocols are supported by Jitsi

- MNSP

- OSCAR
- SIP/simple
- XMPP/jingle
- YMSG

3.10) CODECS

AUDIO Codec's for Jitsi.

Following are the Codec's for the jitsi which are listed below.

- OPUS
- Silk
- G.722
- G.711
- G.729 Annex C

Video codec's for the Jitsi are listed below.

H.264

H.263

This study of thesis evolve to go into the deep evaluation of the codec's how they are used and what advantages and disadvantages we have in using these certain type of codec's. AS we are now going to evaluate the usage of these codec's in detail.

3.10. A) OPUS

Opus is a loss audio coding format developed by the Internet Engineering Task Force (IETF) that is particularly suitable for interactive real-time applications over the Internet. As an open format standardized through RFC 6716, a reference implementation audio codec called opus-tools is provided under the 3-clause BSD license. All known software patents which cover Opus are licensed under royalty-free terms.

Opus incorporates technology from two other audio coding formats: the speech-oriented SILK and the low-latency CELT. Opus can be adjusted seamlessly between high and low bitrates, and internally, it transitions between linear predictive coding at lower bitrates and transform coding at higher bitrates (as well as a hybrid for a short overlap). Opus has a very low algorithmic delay (26.5 ms by default), which is a necessity for use as part of a low audio latency communication link, which can permit natural conversation, networked music performances, or lip sync at live events. Opus permits trading-off quality or bit rate to achieve an even smaller algorithmic delay, down to 5 ms. Its delay is very low compared to well over 100 ms for popular music formats such as MP3, Ogg Vorbis and HE-AAC; yet Opus performs very competitively with these formats in terms of quality per bit rate. Unlike Ogg Vorbis, Opus does not require the definition of large codebooks for each individual file, making it preferable to Vorbis for short clips of audio.

3.10. B) SILK CODEC

SILK is an audio compression format and audio codec developed by Skype Limited. It was developed for use in Skype, as a replacement for the SVOPC codec. Since licensing out, it has also been used by others. It has been extended to the Internet standard Opus codec.

Skype Limited announced that SILK can use a sampling frequency of 8, 12, 16 or 24 k Hz and a bit rate from 6 to 40 Kbit/s. It can also use a low algorithmic delay of 25 ms (20 ms frame size + 5 ms look-ahead). The reference implementation is written in the C programming language. The codec technology is based on linear predictive coding (LPC). The SILK binary SDK is available.

3.10. C) G.722

The technological developments in digital communication systems increase the usable bandwidth of sound signals which results in increased intelligibility and naturalness of the signal. The emerging digital communication systems enable the use of wideband speech codec in a wide area of applications. Recognizing the need of high quality wideband speech codec.

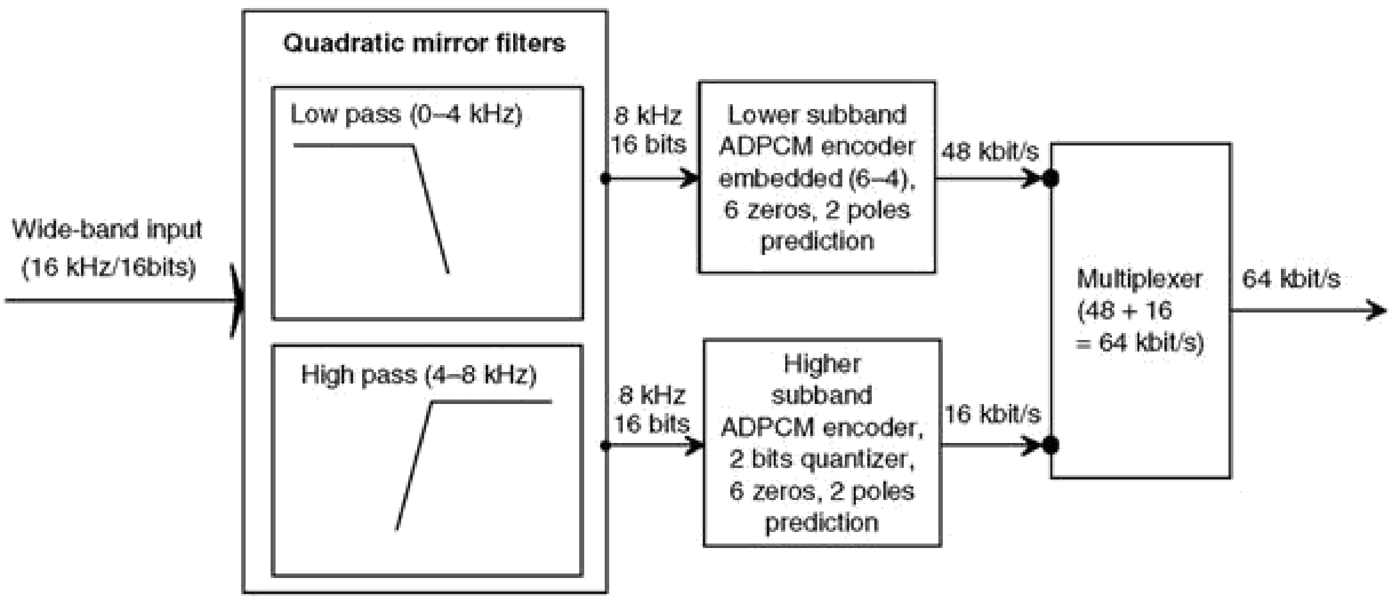


Fig 3.6 Block scheme of G.722

G.722is one of the first wideband speech codec standards implemented in the telecommunication systems. In this thesis, after examining the G.722 wideband speech codec theoretically, a real time implementation of G.722 wideband speech codec has been developed on BlackfinBF533 DSP development kit from Analog Devices. Also, G.722 Codec is optimized by using assembler for BF533. As a result to the thesis, very good speech performance is obtained in real time implementation of G.722 and optimization increases the efficiency of Black fin BF533 DSP drastically.

G.722 is an ITU standard codec that provides 7 k Hz wideband audio at data rates from 48, 56 and 64 Kbit/s. This is useful for voice over IP applications, such as on a local area network where network bandwidth is readily available, and offers a significant improvement in speech quality over older narrowband codec's such as G.711, without an excessive increase in implementation complexity. Environments where bandwidth is more constrained may prefer one of the more bit rate-efficient codec's, such as G.722.1 (Siren7) or G.722.2 (AMR-WB).

G.722 has also been widely used by radio broadcasters for sending commentary grade audio over a single 56 or 64 Kbit/s ISDN B channel (the least significant bit is dropped on 56kb circuits).

3.10. D) G.729 (ANNEX C)

The G.729 is considered as an optional coder due to its low bit-rate performance and high quality of speech. However, since the G.729 is patent protected, one of its disadvantages is that extra cost occurs since parts of the G.729 algorithm are patent protected in the implementation, the G.729 was replaced by the low-complexity version G.729A. Jitsi is the VoIP clients which provides G.729 Annex c Which usually requires compilation and licenses in order to use such codec G.729 is an important codec at low bit rate for higher performance.

3.10. A. a) Video Codec

H.264

Digital video can be found in familiar applications, such as the Digital Versatile Disc (DVD), Digital television, Internet video streaming, and digital high definition television. Digital video shares all the features of other digital formats, including lossless transmission, lossless storage, and ease of editing.

To ease storage / transmission requirements, compression is commonly performed on the video. Modern loss compression techniques allow tremendous storage savings with little visible degradation. The MPEG-2 video compression standard has allowed the success of DVD-video and digital high definition television. New advancements in digital video compression technology have led to the recently finalizedH.264 video compression standard, which is poised to follow the success of the highly accomplished MPEG-2 standard.

A digital format, such as DVD-video, allows lossless copies to be created without the inherent generation loss associated with analog formats, such as VHS tape. For example, a duplicate copy of a VHS tape will not be an exact reproduction of the original. On the other hand, a duplicate copy of a DVD will result in a exact reproduction of the original.

Digital video formats lend themselves more easily for transmission, especially over error-prone, or loss, channels. Traditional error prevention and recovery techniques, such as error-correcting codes can be employed. Digital video editing is easier to perform than analog video editing.

One of the major drawbacks of a digital representation of video is storage and transmission requirements. Uncompressed digital video can require large bandwidths, especially as spatial and temporal resolution increase.

3.10. A. b) H.263

Multimedia video communication system is a hot subject in communication application field. Video information processing needs a large amount of calculation. The video communication terminal based on DSP is an optimal media. With the development of multimedia digital processing and wireless network technology, Wireless Video Communication System has become a new application trend.

Currently, most video transmission application has founded on fixed public IP network interface, which is a generic single mode and makes a limited applicable range on video monitor etc. So research on wireless multimedia video communication system of diverse mode has value in application. The thesis use internet radio mode on ISM band as theory foundation, providing multi-point design of wireless video network, which emphasis on research of wireless process terminal. The thesis has designed a DSP video transmission plan both in Ethernet and RF mode.

H.263 is used for low-rate video transportation, which is used and optimized on DSP. RTP/RTCP real-time video transportation protocol could provide reliable and traffic control of real-time multimedia video streaming. Wireless video communication terminal is a crucial part of Wireless multimedia communication system. It has function of video gather, compression and communication interface.

H.263 was developed as an evolutionary improvement based on experience from H.261, the previous ITU-T standard for video compression, and the MPEG-1 and MPEG-2.

3.11) Comparison between the Features of C sip and Jitsi

Jitsi Features

- Calls
- Instant Messaging

- Security
- Codec's
- Miscellaneous.
- Sip specific
- XMPP specific

3.11. A) Calls:

While working on our project we came to know about the wide features regarding to the calls, which include audio and video calls, desktop streaming, audio conference calls, audio level display, call recording attended transfer blind transfer, call encryption noise suppression and echo cancellation. While with such a huge amount of features provided by jitsi made this VOIP client a better than all the others.

3.12. B) Instant Messaging

Instant messaging provided by jitsi includes, Presence one to one chat multi user chats file transfer and OTR encryption. These features made this Application fast and accurate also require less time to communicate.

3.12. C) Security

Jitsi provides a security level with encrypted password storage, Encrypted instant messaging with Off-the record messaging chat authentication with the socialist millionaire protocol over OTR, call encryption with SRTP and ZRTP for XMPP and sip, call encryption with SRTP and SEDS for XMPP and SIP,DNSSEC support.

3.12. D) Miscellaneous:

On line provisioning, Provisioning server discovery via DHCP and m DNS IPv6 fully supported by sip and XMPP, call history and missed call notifications, systray notifications, drag and drop support and file transfer integration with Microsoft and apple address book, support for LDAP directories support for Google contacts, cross protocol conference calls: call your contacts over different accounts and Protocols.

3.12. E) SIP Specific:

On-line contact list storage with XCAP Secure signaling with TLS DTMF (SIP INFO, RTP RFC 2833/4733, in band) Message Waiting Indication (RFC 3842) Certificate-based client authentication.

3.12. E) XMPP Specific

- DTM(RTPRFC2833/4733inband)
- Correctingpreviouslysentmessages(XEP-0308LastMessageCorrection)
- Certificate-based client authentication

3.13) Features for C sip Simple

- SIP for calls and instant messages
- Android integration with rewriting and filtering rules
- Codec's : PCMU/a (aka g711u/a); SPEEX; g722; gsm; ISAC; SILK; G729; AMR (depending on device) and as extra plugin : OPUS; g726; g722.1; codec2
- Echo cancellation (with various back ends : web RTC, SPEEX, simple)
- Auto speaker/earpiece option
- API provided for other apps plug-in.
- Conference, transfer.
- Secure call with TLS transport for SIP and SRTP or ZRTP for media

3.14) Major Difference between C sip and Jitsi.

C sip simple enrich with all the useful codec which consist of G.711 g.729 and G.722 while making the test we tried all this codec by enabling and disabling in order to know the voice quality as we know G.729 is used for low bit data rate of 8 Kbit/sec and G.711 is used for companding and for the HD voice which brings the high Quality and the noise suppression, Whereas G.722 is a wideband audio codec operating at 64 K bits. All of these codec are compatible with all kind of latest devices we have.

Echo is always a major problem in the communication during the call attempts for the echo cancellation the backend on c sip simple is web RTC SPEEX and simple, secure call provided by the c sip simple, Most of the smart phone laptops and Pads are built with android application and c sip simple has the operating system is android just in the starting phase we developed this application for the android user and on Mac it's just for experimental but it was not good on the Mac and windows, the major drawback of the c sip simple is not good for the other operating system but provide all the Compatible codec's for audio and video conversation even for the instant messaging.

Jitsi is highly enrich with the advanced features and codec but the code that used in jitsi are very advanced and not that much compatible with the recent devices. No G.729 A and B which are useful for low data at 8 Kbit/sec most widely used. it gives G.729 annex c which needs licenses and compilation, Jitsi mainly focuses on G.722 which is a wideband audio codec and also some advanced codec like opus and silk usually more compatible with Mac and advanced devices, the major advantage of the jitsi the operating system which are compatible with the jitsi include Linux MAC, Window OS and All java supported so with the help of jitsi application is enable on all of the operating system but currently VAVE application is configured through C sip simple but later on the best choice will be the jitsi because it can provide best features to the most of the operating system present today.

CHAPTER 4

4.1) Observation and Conclusion:

4.1. A) Advanced Technology:

As the development of smart mobile phone industry, there are more and more people connected to the internet through smart phones. The real time communication demands are from the traditional telephony network to IP network. There are many client applications provide real time communication service through the internet. There are two main different categories for this real time communication solution. One kind of application is like Google Hangout, it provides users a real time communication channel on the internet and requires user clients are both using browser to communicate. The other kind of application is like Skype1, it provides VoIP service and let different client users (application client and physical phone) to communicate with each other. The second type of service is the goal of unified communication as the development of smart mobile phone industry, there are more and more people connected to the internet through smart phones. The real time communication demands are from the traditional telephony network to IP network. There are many client applications provide real time communication service through the internet. There are two main different categories for this real time communication solution. One kind of application is like Google Hangout, it provides users a real time communication channel on the internet and requires user clients are both using browser to communicate. The other kind of application is like Skype1. It provides VoIP service and let different client users (application client and physical phone) to communicate with each other. The second type of service is the goal of unified communication.

However, the problem of the second category application is that users have to install some application client and it requests for some application credential to use the service. There are already many different applications installed on user's Smartphone and desktop computer. It is difficult for users to remember another application credential and install one more application for just calling.

The main goal of a unified communication solution with Web RTC is to integrate Web RTC technology with traditional telephony network. The term, unified communication, in this thesis means the unified solution for real time communication on the internet and on the traditional telephony network.

4.2) WEB RTC

Software or plug-in and not ask user to remember another new credential information either. The approach for that would be a web application service using user telephone number as credential and provide the user call any kind of other user no matter the other user is on his mobile phone or his computer through internet. This system will be an Over the Top (OTT) solution integrated with Web RTC network and VoIP network. The service can provide user a new real time communication way to reach other people in the world since everyone is on the internet or on the phone nowadays.

The prototype system implemented in this thesis will provide rich multimedia real-time communication service with Web RTC network and SIP network. Some basic real-time communication application functions will be achieved, like calling mobile phone, having video conferencing, instance messaging and file sharing. And normal telephony functions will be achieved by the prototype system as well, for instance, calling phone number, receiving call from other phone number, forwarding phone and Short Message Service (SMS) messaging

4.2. A) CHALLENGES:

On some specific programming languages and network knowledge. Challenges of this WEB RTC are mainly from two categories, research challenges and implementation challenges.

For research challenges, since Web RTC technology is a new web technology and not scandalized yet, there are a lot of articles about it but they are not all relevant as references because different browser has different implementation on Web RTC and the implementation keeps changing with the updates of the browser. Moreover, there are not many open sourced projects to support SIP on the web application. The objective is to integrate Web RTC technology with traditional telephony network, and then it is necessary to do the search about the similar implementing project or application in this scope. There are no such directly communication service between SIP and Web RTC in the commercial market. The research cases could be studied are mostly based on one of these technology.

For implementation challenges, there are no commercial products using the same technology as the prototype system of this thesis in the market yet. The combination of the technologies implemented in the prototype system is completely new in the field. There are not so many references and documentation could be helpful during the development. Student who developed the system has to understand the basic and fundamental knowledge about SIP protocol and Web RTC implementation in order to implement a unified communication service based on SIP and Web RTC. It requires a lot of time on programming demo prototype to evaluate the implementation solutions. Furthermore, there are many system design cases which need to be considered during the development because the target integration system is the traditional telephone. Network it requires high system stability. All the implementation source code of the prototype system to achieve the objective of unified communication solution with Web RTC is created by the student alone. Some of them are based on other third Party library with some changes by the student. These development process requires student have very high programming skills

4.3) Bridging between IP and telephony Network

In order to bridge the IP network and telephony network, a solution to create a real-time communication channel between IP network and VoIP network is the key factor since the VoIP network is the bridge to make IP network to talk with telephony network. In this Chapter, some introduction of Web RTC and SIP network will be covered. SIP is one of the VoIP signaling protocols widely used in current internet telephony service which is also the target telephony network in this thesis. There will be some studies of Web RTC business cases and prototype working scenario based on these Web RTC usage cases in this chapter. The prototype working scenario is designed by considering these different Web RTC usage cases.

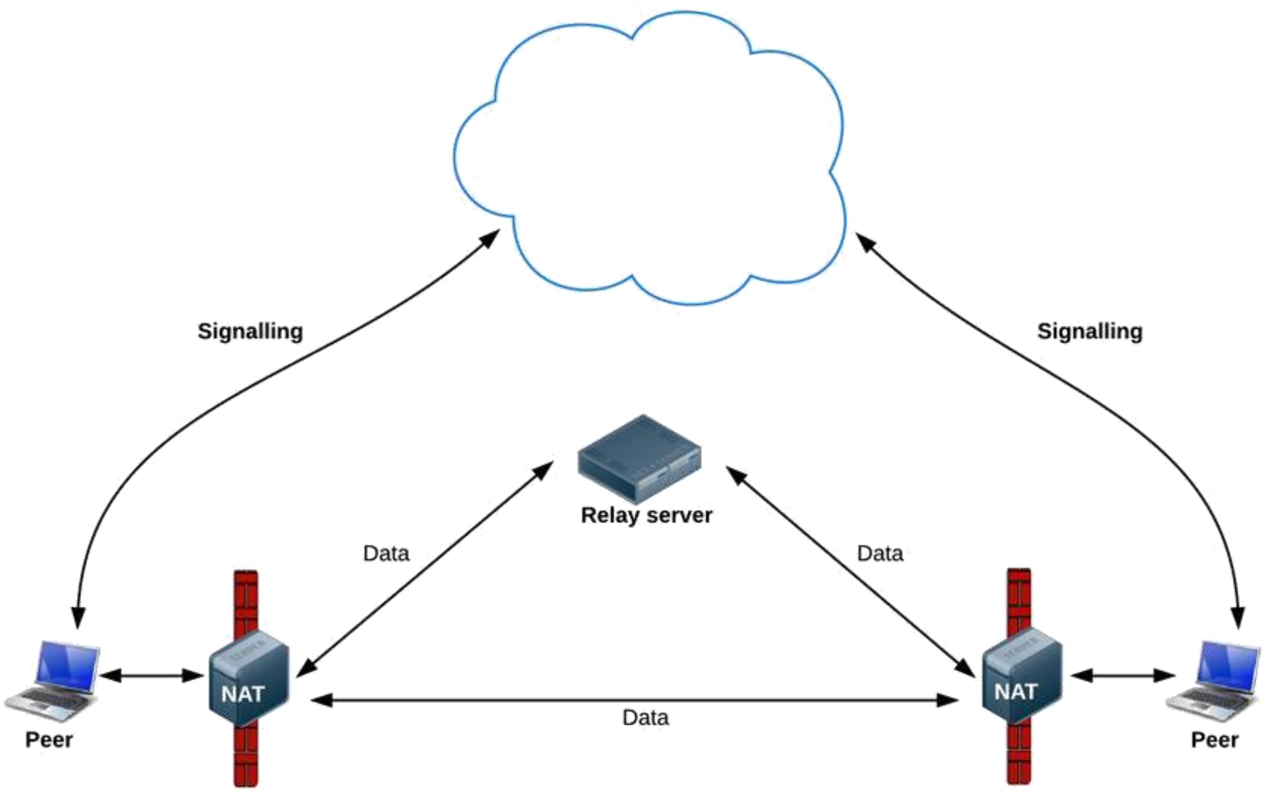
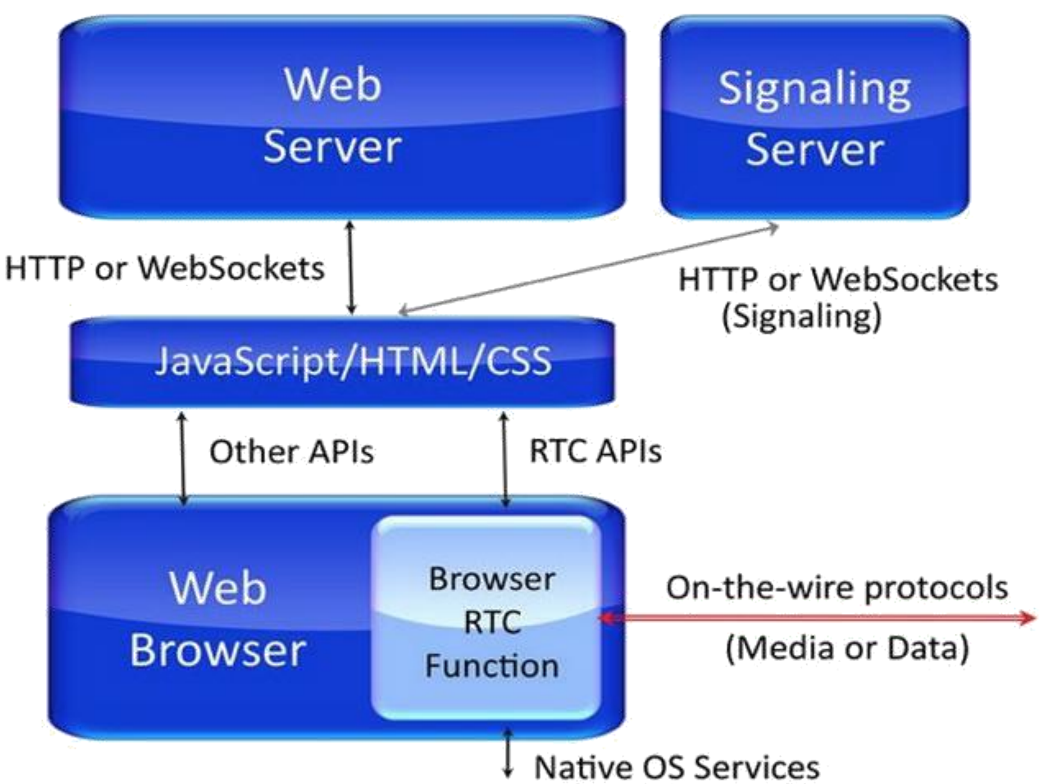


Fig 4.1 Bridging between IP and telephone network

4.4) Web RTC Implementation Steps:

There are four main steps to implement a Web RTC session shown in Figure. The browser client need to obtain local media first, then set up a connection between the browser and the other peer through some signaling, after that attach the media and data channels to the connection, afterwards exchange the session description from each other. Then the media stream will automatically exchange through the real-time peer to peer media channel. Each step shown in the Figure is implemented by some Web RTC APIs. The Web RTC architecture is shown in Figure 2.4, the main focus in this thesis will be Web API part and transport part because Web API is the tool to implement the Web RTC application and transport part is the key for Web RTC application to communicate with application server, media server and any other end peer in the system.



Besides Web RTC APIs, signaling is the other important factor in the system. Web RTC uses RTC Peer Connection to communicate streaming data between browsers, but also needs a mechanism to coordinate communication and to send control messages, a process known as signaling. Signaling methods and protocols are not specified by Web RTC by Google in purpose, then signaling is not part of the RTC Peer Connection API which can be decide how to implemented based on different project scenario.

Instead, Web RTC app developers can choose whatever messaging protocol they prefer, such as SIP or Extensible Messaging and Presence Protocol (XMPP), and any appropriate duplex (two-way) communication channel. The prototype application in this thesis will use WebSocket6 as signaling between Web RTC browser end point and keep use SIP as signaling for SIP end point (mobile/fixed phone based on PSTN in this case).

Signaling is used to exchange three types of information in Web RTC [Dut14]:

– Session control messages: to initialize or close communication and report errors.

– Network configuration: to the outside world, the computer's IP address and

Port.

– Media capabilities: the codec's and resolutions can be handled by the browser

And the browser it wants to communicate with.

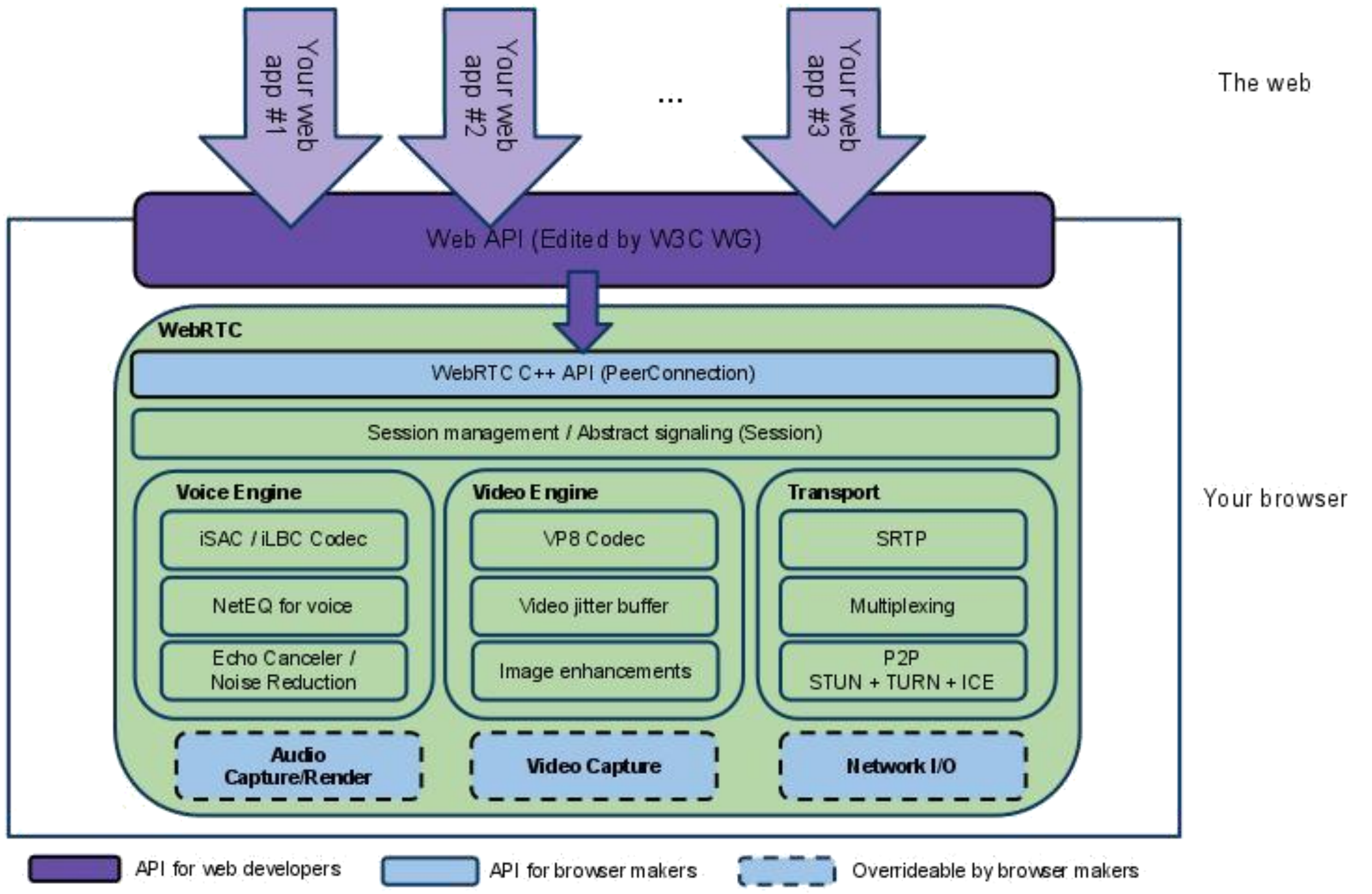


Fig 4.3 Architecture of Web RTC

4.4) WEB RTC USAGE:

After Google released the Web RTC as open source project. There are more and more web applications using it in different ways. Web RTC APIs includes three important APIs, shown below.

There are mainly two types of the Web RTC applications used them in separately or cooperatively way.

– RTC Peer Connection: audio or video calling, with facilities for encryption

And bandwidth management.

– Media Stream: get access to data streams, such as from the user's camera

And microphone.

– RTC Data Channel: peer-to-peer communication of generic data.

RTC Peer Connection is the foundation of all Web RTC application to establish the peer to peer connection. For showing remote peer media source content and exchange the local peer media source content, the web application need to get the user's camera view and microphone sound, the Media Stream API is used always in real-time communication application.

4.5) Prototype system working Flow

To connect with the traditional telephony network, the VoIP system bridges the PSTN and the IP network. VoIP systems employ session control and signaling protocols to control the signaling, set-up, and tear-down of calls. They transport audio streams over IP networks using special media delivery protocols that encode voice, audio, video with audio codec's, and video codec's as Digital audio by streaming media. In the prototype system, SIP signaling is used because of its widely usage and current target PSTN has SIP server support.

The Figure shows the basic working flow of the prototype system. The Web

Server/ Gateway are the application server in the proto type system, it mainly bridge.

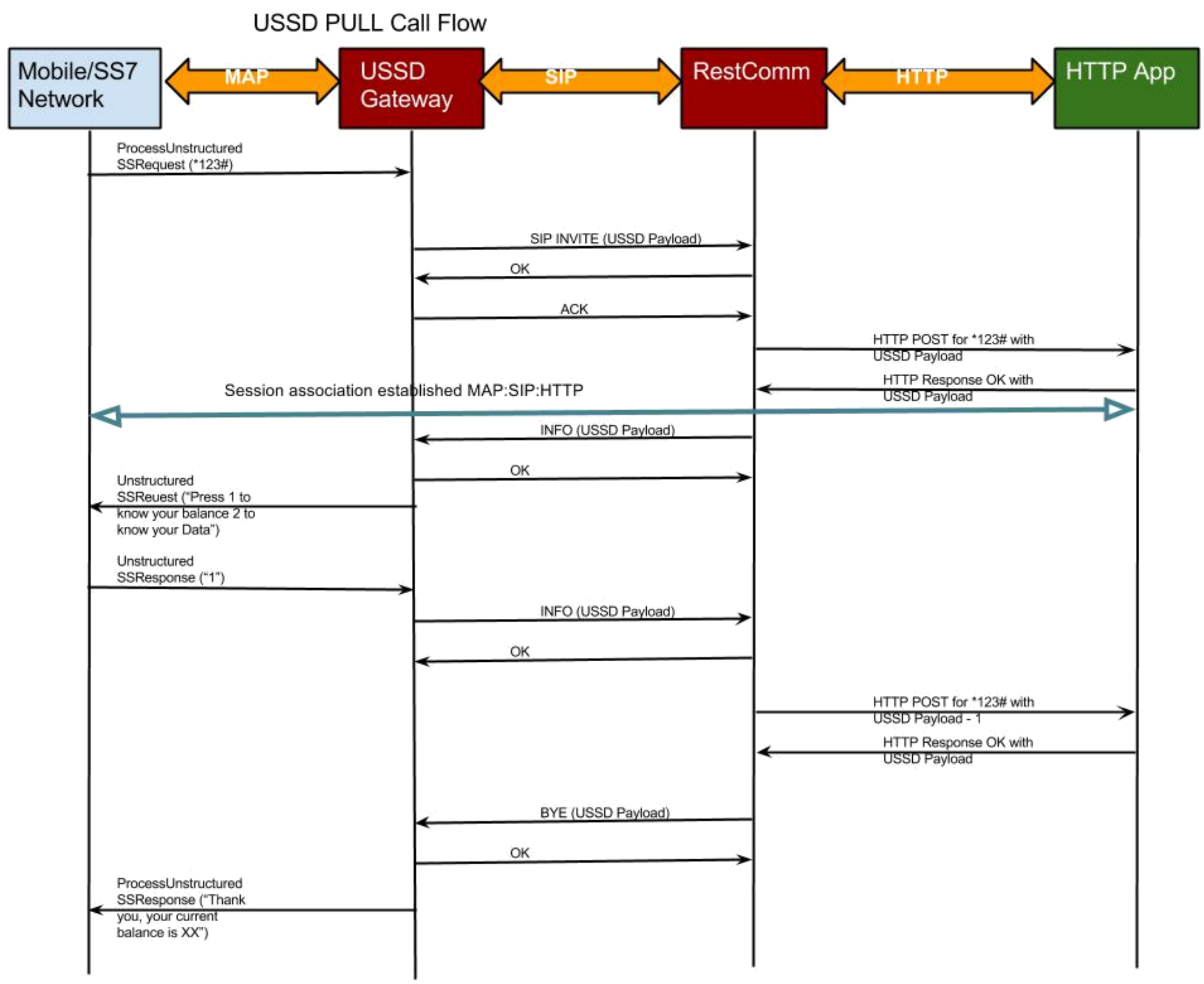


Fig 4.4 Prototype system working design

4.7)Web RTC conclusion:

The advantage of prototype system is that it does not require users to install any application client and it is no need for users to have another user credential for this service (prototype system uses telephone number as user credential). Moreover, this unified communication service is server centralized system, it will have more advanced real time communication functions can be implemented on both server side and client side. In this system architecture, there is more space for developers to add more advanced functions and it is easier for scaling for larger user base.

Because the prototype system is based on Web RTC, it means that it is highly dependent on web browser client. More advanced concept about unified communication service would be implementing OTT real time communication. It will either require the mobile browsers on the smart phones implemented for Web RTC standard or Web RTC can be implemented on different mobile operation platform as native API. Afterwards, there will be more devices can use prototype system to have rich real time communication service between mobile phone users and computer users. Therefore, current application client of the prototype system is based on browser client. The compatible devices which can use the application client are Web RTC supported browsers. Because there are not so many mobile browsers support Web RTC yet, then the user clients to use the prototype application are computer clients only, there are more potential users on the mobile platform.

Furthermore, there are no commercial products to provide unified communication service based on Web RTC and SIP. There are many potential usage of the proto type system integrated with other popular web service in different industry area. And the bridge to connect the web world and telephony work is the prototype system service. The unified communication service will be the big game changing for the web communication and telephony communication business.

4.8) Conclusion

It is clear that voice over internet protocol (VOIP) is overtaking a lot of single segment or voice-network-only voice applications (such as law enforcement radios).VOIP can also be seen in phone service as applications on some smart phones. Because of VOIP's rapid growth, there is a need to analyze VOIP clients and security they provided.

This thesis focused on the feasibility of using VOIP/SIP as a means of achieving True end-to-end communication with security also providing a fast and valuable communication means to the Passenger, by making a test and comparison between different clients in order to achieve the reliable application.

While making a Comparison between different clients of VOIP security QOS and end to end communication is very important while jitsi is providing more feature than the C sip but on the other side jisti is lacking in some kind of codec's which are now a day's compatible with most of the smart machines.

While looking on the advanced technology web RTC is the best choice for the end to end communication. There are no commercial products to provide unified communication service based on Web RTC and SIP. There are many potential usage of the prototype system integrated with other popular web service in different industry area. And the bridge to connect the web world and telephony work is the prototype system service. The unified communication service will be the big game changing for the web communication and telephony communication business.

REFRENCES.

1) *http://www.ies-italia.it/*
2) *http://www.mcp.com/*
3) *EMP Stack (Linux, Nginx, MySQL, PHP)". lemp.io. Retrieved 2014-10-15*
4) *http://en.wikipedia.org/wiki/MySQl*
5) *http://en.wikipedia.org/wiki/Linux*
6) *http://en.wikipedia.org/wiki/PhpMyAdmin*
7) *http://www.elementaltechnologies.com/*
8) *http://www.appeartv.com/*
9) *http://en.wikipedia.org/wiki/Voice_over_IP*

10)*The Stream Control Transmission Protocol (SCTP) as a Transport for the Session Initiation Protocol (SIP)*, IETF, The Internet Society (2005)

11) *Zurawski, Richard (2004). "RTP, RTCP and RTSP protocols".* The industrial information technology handbook. *CRC Press. pp. 28–7. ISBN 978-0-8493-1985-3*.

12) *Real Time Control Protocol (RTCP) attribute in Session Description Protocol (SDP), C. Huitema, Microsoft (October 2003)*

13) *International Telecommunications Union, Standardization Sector (ITU-T), Study Group 15 (1993-1996), Recommendation G.729, March 1996.*

14) *G.711: Pulse code modulation (PCM) of voice frequencies; ITU-T Recommendation (11/1988)*

15) *"H.262: Information technology — Generic coding of moving pictures and associated audio information: Video"*

16) *ETSI (2009-01) ETSI TS 126 140 V8.0.0 (2009-01); 3GPP TS 26.140; Multimedia Messaging Service (MMS); Media formats and codes*

17) *"RTP Payload Format for Opus Speech and Audio Codec, draft-ietf-payload-rtp-opus-03"*

18) *ean-Marc Valin, Koen Vos and Timothy B. Terriberry (September 2012). "Definition of the Opus Audio Codec".* RFC 6716. *IETF. Retrieved 2013-08-19*.

19) *ohansson, Leif (April 18, 2005). "Greater NOrdic MIddleware Symposium (GNOMIS)". Oslo: University of Stockholm.*

*20) Alan B Johnston and Daniel C Burnett. WebRTC: APIs and RTCWEB Protocols
of the HTML5 Real-Time Web.*
*21) Alan B Johnston and Daniel C Burnett. WebRTC: APIs and RTCWEB Protocols
of the HTML5 Real-Time Web, Introduction to Web Real-Time
Communications, page 15. Digital Codex LLC, second edition, 201*
22) Muaz Khan. Translator.js, 2014. [Online; accessed 27-May-2014].
*23) The Internet Society. Sip: Session initiation protocol — rfc 3261, 2002. [Online;
accessed 7-May-2014].*
*24) Peer5. Sharefest — web based p2p file sharing built on webrtc data channels api,
2014. [Online; accessed 27-May-2014*