

KEMI-TORNIO UNIVERSITY OF APPLIED  
SCIENCES

Analysis of the architecture of VoIP and SIP to design a secured  
browser based web phone

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Bachelor's thesis of the Degree Programme in Business Information Technology  
Bachelor of Business Administration

TORNIO 2011

## ABSTRACT

Mahmood, Syed 2011. Research on analysis of the architecture of VoIP and SIP to design a secured browser based web phone. Kemi-Tornio University of Applied Sciences. Business and Culture. 2 Appendices. Pages 59.

The objectives of the research were to analyze fundamental architecture of Voice over Internet Protocol (VoIP) and the working mechanism and architecture of Session Initiation Protocol (SIP) to design secured browser based web phone. To accomplish the objectives of the research, exploratory research method was used to gather information and solid knowledge on VoIP and SIP whereas constructive research method was used to design the architecture of web phone. Literature review and documentation analysis were supplementary research techniques to get acquainted with the research area which was reflected on the research outcome.

The fundamental architecture of VoIP and the architecture and working mechanism of SIP is described in detail in chapter 4 and 5 in the thesis. Documentation and literature analysis was used to design the web phone architecture which was final objective of the research project.

Unified Modelling Language (UML) was used to design the visual diagrams and justified by the Meta-object Facility (MOF) method which is an ISO/IEC standard for meta modelling. The design process of the web phone is also summarized in the discussions and conclusion with further research suggestions.

The analysis of documentation and literature on VoIP and SIP architecture as well as the reason to choose SIP for the project is summarised in the result in discussion and conclusion.

Keywords: VoIP, SIP, UML, PSTN, H.323.

## ABBREVIATIONS LIST

VoIP	Voice over Internet Protocol
SIP	Session Initiation Protocol
PSTN	Public Switched Telephone Network
UML	Unified Modeling Language
MOF	Meta Object Facility
SS7	Signaling System #7
ASN.1	Abstract Syntax Notation One

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## 1 INTRODUCTION

This thesis outlines the requirements, benefits and issues involved in migrating to a Next-generation Network using Voice over Internet Protocol (VoIP) in web technology. The thesis represents a high-level network architecture that acts as a starting point from which PSTN equivalent VoIP networks can be built for websites which enables the client to get the benefits of VoIP without the restriction of installing and meeting access right criteria on any system. The thesis describes the architecture and functionalities of VoIP and Session Initiation Protocol (SIP) which was eventually used to understand and design the browser-based web phone.

The objective of designing the browser based web-phone is to create a new dimension in web technology. Today, VoIP plays a significant role in the telecommunication industry. A study shows that the annual growth rate of voice market is approximately 3 to 4 percent and the data transmission of voice is approximately 30 percent per year (Bud & Regis 2002, 528). The development of VoIP has played an important role in reducing the cost of telecommunication and made the telecommunication system accessible in an easy way with an ordinary computer or voice communication device which is connected with the Internet. However, in the field of communication technology which is evolving with a faster pace demands to continue the development process. The VoIP system can be more users accessible and also enable the users to use by making the system without going through the complexity of software installation process. Nevertheless, software installation process by the ordinary users requires access right to install software on a system. However, this installation access right is an issue of security. Besides that if the VoIP software is installed on a specific client machine which is always not portable, restricts the clients to use the VoIP software from different machine because clients do not have the privileges to install software. This restriction is reasonable since security issues are involved with installing software and giving access right to anonymous users. Furthermore, buying a computer or a voice communication device with Internet connection is not affordable to the low earners. Hence users without access to their own computers can benefit from using any computer from the local cyber café.

Therefore, the design of browser based web phone hence is the proposed solution of all the barriers of VoIP which restricts clients' accessibility on certain systems. Hence the browser based web phone creates a new dimension to the VoIP system with its flexibility and accessibility to the clients, regardless the access right clients have, to use any system as long as clients can browse the Internet. Hence the browser based web phone will enable the clients to use their VoIP accounts from anywhere in the world to establish voice communication.

The process and architecture of voice communication is described in the chapter four and five in the thesis where I explain the VoIP and SIP architecture in details. To design the system, I used the Unified Modelling Language (UML) which is defined by the Meta-model Object Facility (MOF).

### 1.1 Motivation and Background

Information Society has been an important factor for the progress of dynamic development of the world. Hence information plays a significant role in every society to manage events related with socio economic factors by adopting innovative technologies. The most important sectors that Information Society is composed are computing, telecommunication and media broadcasting. Though these sectors were separate earlier but by the faster evolve of digitization convergence them continuously. (Servas2003, 5.) Now a day, people are connected with technology and aware of equipments which people use to communicate with each other. Mobile phone, land phone, voice chat and net to phone technologies are the mostly used among the modern communication technologies. The collaboration of different technologies has made the tools easier to the end users who are not so much familiar with the complexity behind the scenario of modern technologies. However, still the complexity of voice communication has been an issue which cannot be ignored and should be considered for the sake of millions of telecommunication service users. Users still need the binary file to download on their system to install and run the program. This makes portability of the voice communication software over the internet facility complex and inconvenient to the end users. Studying and analyzing the process of development in the telecommunication industry, I was motivated to contribute my knowledge and idea to develop a convenient solution for the end users where the end users will be able to call from any part of the world through an ordinary webpage without installing any external software. There are



similar existing technologies (ex. c2call) developed recently which also motivated me to develop the design since the idea is relatively contemporary and new.

Additionally, the idea has potential business orientation which motivated me to do my practical training to get hands-on experience on communication technology and continue my work as the bachelor thesis project to design the system in order to carry further to develop business around the idea in future.

## 1.2 Structure of the Thesis

Chapter 2 discusses the objectives of research project including in its scope, research questions, research methodology, theoretical framework and expected result of the research project. The third chapter focuses on the review literature related to the brief history of telecommunication system and VoIP. The third chapter also includes the history VoIP and VoIP components and Public Switched Telephone Network (PSTN).

In chapter four, I analyzed the architectural requirements and functionality of VoIP and the protocol families of VoIP by reviewing a number of literatures on the area. This chapter is also pre-requisite to read chapter five.

Chapter five describes the development phases of the SIP and its network architecture. In this chapter, I also described the working mechanism of the SIP and its entities and functionalities and interactions with Request for Comments (RFC) based on the knowledge of reviewing literature on SIP and RFC.

After analyzing the architecture of SIP network, I designed the browser based soft-phone in chapter six. In chapter six, I used literature on software engineering, requirement engineering, UML and telecommunication. This chapter specifies the modelling of the web phone, design patterns and deployment recommendations of the software. The security issues of browser-based web phone are also discussed in chapter six. Security issues include the risk classification of vulnerabilities and policies to manage the risks. Chapter seven concludes the thesis with discussions on result, limitations and challenges of the research project and directions for further research.

## 2 OBJECTIVES, RESEARCH QUESTIONS AND METHODOLOGY

The first objective of my research project was to analyze the architecture of the VoIP and the SIP to acquire knowledge on the technologies. Based on the analyzed result, my second objective was to design a browser based web phone which enables the VoIP services available on websites without requiring any software installation process. This work includes contemporary technology analysis and design of the web phone followed by risk identification and security management strategy.

The scope of the research was limited to the voice communication web phone associated with the Internet, PSTN and Internet Protocol (IP) phone. Hence the architecture and functional aspects of hardware and network devices are not part of the thesis.

Due to the fact that the development of VoIP for website was a complex process and time was limited for implementation phase, this research focused only on the designing phase of the system to carry out the further analysis and development for web technology in future. The testing and implementation result of the research project was beyond the scope of thesis work. Nonetheless, I gave recommendations on implementing the project using an open source voice communication server and real-time voice communication system.

### 2.1 Research Questions

**RQ1.** What is the fundamental architecture of Voice over Internet Protocol?

The first step to design the browser-based web phone for the objective of the research was to analyze the fundamental architecture of VoIP. The architecture of VoIP is described based on the authentic Internet sources and printed literatures. Clear understanding of the VoIP architecture created a baseline for preparing the requirement specifications of the final objective of the research which was to design the secured browser based web-phone.

**RQ2.** What is the architecture and working mechanism of Session Initiation Protocol?

One of the objectives of the research was to analyze the architecture and responding mechanisms of SIP. SIP is a standard Request for Comments (RFC) 3261 defined by the Internet Engineering Task Force (IETF). As the design model requires specification of SIP and the responding methods of SIP, it was important to analyze its architecture and working mechanisms. The analyzed result helped to design the web-phone in compliance with the requirement specifications. Since there are other signaling protocols available, the analyzed result gave a clear explanation of using SIP in designing web phone. Finally, the outcome was used in the core design part of the browser based web-phone and UML documentation of this research project.

**RQ3.** How can the architecture of the secured browser-based web phone application be designed?

Designing the architecture was the ultimate objective of the research project which was to develop a design model of browser based web-phone. The design model was based on the analysis of the architecture of VoIP and SIP and the correspondence between RFCs with VoIP and SIP. Finally, UML 2.0 was used to design the visual model of functional components of the system

## 2.2 Research Methodology

The research methods used for this work were exploratory and constructive research. The decision to choose these two methods was logical because the objective of this research was to construct the design of browser based web-phone which will enable the VoIP services accessible to the users on any computer or communication device connected with the Internet based on self-educative exploratory research result on VoIP and SIP.

Constructive research method was chosen because the research objectives to develop a new realistic solution for a practical problem which was built based on the past research since the objective of constructive research is to provide solution to real world problem and build the solution on previous research. In other words, Järvinen (2001, 88) suggests that “It is typical for constructive research to build a new innovation and this

process is based on existing (research) knowledge and/or new technical, organisational, etc advancement.” which guided the research work. Furthermore, my final objective of the research was to design the system and suggest how to implement the plan. Hence Järvinen’s (2001, 88) suggestion on constructive research method which is, “The constructive research is applied research, but instead of the final product it is possible to accept the prototype or even a plan” relied on in this work.

Carrying out the research project, several literature on telecommunication technology and documentations on entities of RFC, VoIP and SIP were used to reach the target state of the research project. The literature and documentation were written based on previous research on telecommunication technologies. In other words, Järvinen’s (2001, 88) suggestion that “the purpose of the construction process is to achieve a movement from initial state to the target state” is relied on in this thesis.

Exploratory research was chosen due the fact that the knowledge and experience to build the system was based on analyzing and exploring information on necessary area which were closely related to build the design of the web phone system. Since the process of system design was depended on the explored data hence the relationship among methodologies was not discrete in terms of dependency. In other words, it means that the information and background knowledge on the system development were needed to be explored for self-education which eventually helped to design the system. However, if the background knowledge on the system development would be strong and substantial, then the exploratory method would not be used in the research since this would be revised explanation of technology.

The complexity of the research work assigned me to work and collaborate closely with experts in communication and real-time system development field and review literatures to collect background information and acquire knowledge on telecommunication system. Hence carrying out the research project, I worked closely with IT specialists and my supervisor to collect background information on VoIP and SIP. The collaboration aimed to gather information and go through the experimental learning process to explore the complex technology and prepare the base of the design. Panneerselvam (2004,6) defines exploratory research as follows, “Exploratory research is an initial research which analyzes the data and explores the possibility of obtaining as

many relationships as possible between different variables without knowing their end-application.” This definition by Panneerselvam (2004, 6) justifies the research work.

Data collections for requirement specifications through existing software documentation analysis and literature review were the supplementary research techniques which were used to carry out the research project. Additionally, the data collection and documents analysis provided a strong theoretical foundation as well as the baseline for preparing requirement specification for the research project.

The data collection for system design was done through analysis of literatures on data communication, software engineering, information security, VoIP and SIP from established sources which are listed as references. This phase was important in accomplishing the research project outcome since I used the literatures when need arose for solid information. Having the theoretical knowledge on the fundamental architecture of VoIP and SIP, I intended to acquire hands-on experience on the practicalities of their (VoIP and SIP) architecture. Therefore during the set-up process of voice communication server, recommended as part of implementation in chapter 6, to get practical experience on how a VoIP server works, I did existing documentation analysis on both VoIP and SIP. I also analyzed the documentations the related entities of VoIP and SIP such as hardware and network device set-up plan which helped in my research work by giving clear knowledge on VoIP server.

### 2.3 Research Outcome

The first objective of the research work was to analyze documentation on the architecture and mechanisms of VoIP and SIP. Both VoIP and SIP are technologically complicated. Therefore it was important to acquire solid knowledge on the architecture and mechanisms of VoIP and SIP to work with the technologies. Furthermore, the result was tightly connected with the final objective which needed substantial knowledge on the technologies to accomplish.

The final outcome of the research was to design a secured browser-based web phone which should be able to work as with the benefit of independent soft-phone attributes on websites. The design model will be the blueprint of the system on implementation. The design model will assist to develop a relatively contemporary system based on the analysis and system specifications out of the research. Necessary changes to the web

phone requirement specification and the system design modification can be done by re-assessing the documentation and the design model. Since this is a design model of a complex system, therefore the documentation will be important for further research and development in the area.

## 2.4 Theoretical framework

To verify the research output with an existing established standard, a theoretical frame needed to choose which to be used for comparing the standard of the design. In my thesis project, I used Unified Modelling Language (UML) to design the software functions, attributes and process entities for clients end and communication with sever. Therefore, it was important to choose software modelling standard which is used in software engineering to verify the validity and standard of the design. Since UML itself is not a method, it was needed to be defined by a standard method of visual modelling architecture to specify the diagrams requirements of UML. This was done by MetaObject Facility (MOF). MOF is a meta-modelling standard for software design developed by Object Management Group (OMG) to define the UML (OMG, Inc. 2011).

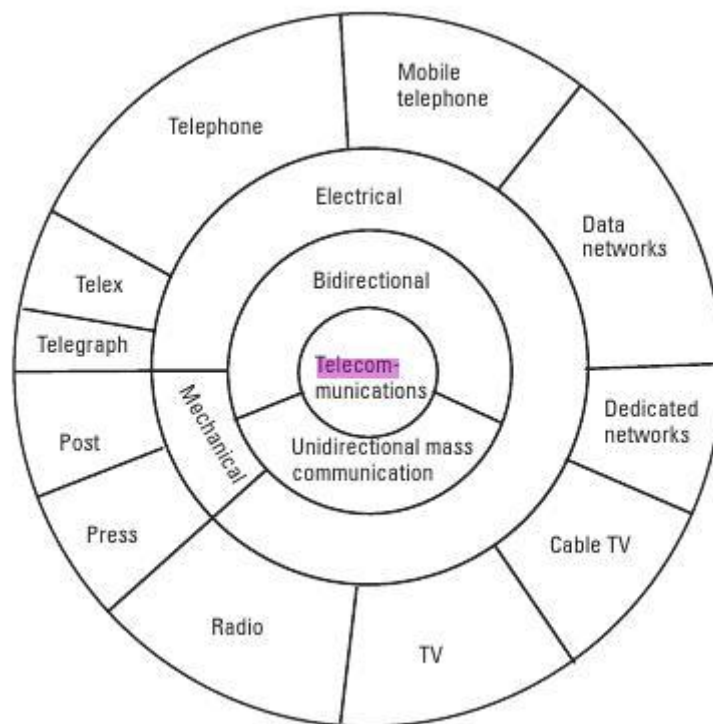
Meta-Objective Facility was used as standard for model driven engineering in software design process. MOF is an international standard (ISO/IEC 19502:2005) to write meta models. (ISO 2011.) During designing the UML diagrams on the web phone, MOF was used as standard to be able to verify and define the UML diagrams and validate the diagrams requirement specification.

To verify the server side design, the functions of call processing described from RFC-3880 documentation by IETF Network Working Group was used. The call processing language is described in the recommended implementation part in the thesis in detail.

### 3 TELECOMMUNICATION SYSTEM AND VOIP

The research project was entirely related to Voice over IP (VoIP) technology and telecommunication technology. Therefore, it was important to acquire the basic conception and background of the core research entities which are telecommunication and VoIP to accomplish the goal of research project and flow of information.

Telecommunication technology is defined as a technology which deals with the issues of communicating from distance, and can be categorized in various ways. It includes different communication technologies such as mechanical communication and electrical communication because it has evolved from a mechanical to an electrical form using sophisticated electrical systems. (Anttalainen, February 2003, 1.) Figure 1 demonstrates the dimensions of telecommunication.



**Figure 1.** Telecommunication dimensions (Anttalainen 2003, 2)

In my research project, I worked specifically on voice communication which is part of bidirectional communication in the upper part of the above figure.

Considering the telephone network which includes more than 2 billion fixed and cellular telephone network globally one can understand the idea of the complexity of telecommunication technology. No other technology in the world exceeds the complexity of telecommunication networks. (Anttalainen 2003.)

Regardless the complexity of the technology, telecommunication is consistently evolving the development phases by developing innovative ideas and introducing new features which has great impact on social, national and global economy.

Telecommunication services have an essential impact on the development of a community. If we consider the telephone density of a region, we can estimate the level of technical and economical development of that region. *Teledensity* which is fixed telephone density is lower than 10 telephones per 1000 inhabitants in developing countries. In comparison, the density is significantly high in developed countries. In comparison with the developing countries, there are about 500 to 600 fixed telephones per 1000 inhabitants in North America and Europe which shows the relation between economic progress and development of telecommunication technology. (Anttalainen 2003, 3.) Therefore, it is clear that the development of region is related with the development of communication technology.

There are different entities of communication system. It has achieved the current phase which is the most sophisticated phase so far by passing many development phases. The development of communication system started with PSTN and developed consistently to wireless and IP (Internet Protocol) technology to reach billions of users around the world.

### 3.1 Brief history of PSTN

PSTN stands for *public switched telephone network* which usually used to refer to fixed phone network before modern communication technologies such as wireless communication or mobile phone technology. This PSTN is a circuit-switched network where users set up the world wide connections by dialling a series of digits to establish communication. A series of circuits together switched by the network assign them for the unshared use of session. The circuit returns to the common pool when any of the



parties hangs up. The Internet is the data counterpart of the PSTN which enables broadband facilities to maximize the user's time share. (Green 2000, 21-22.)

The PSTN network is a complicated technology which is built on circuit switching. The PSTN carries significant amount of data traffic and contains hardware such as circuits, switches, radio, digital and analogue signalling devices, signal inter-transformation devices, fibre-optics and many other instruments related to telephone. Besides, the current telephone network is based on the most sophisticated and prominent signalling protocol named SS7. This SS7 uses an out-of-band signalling to establish a call which reflects the depth of complexity in PSTN. (Ganguly & Bhatnagar 2008, 3-7.)

The development phases however have not been stopped since the PSTN invented. Now a day one of the most important inventions of last century "Internet networking" is being used to transfer human voice over the Internet protocol. This is what we know as "Voice over Internet Protocol."

### 3.2 Fundamental idea of Voice over Internet Protocol (VoIP)

VoIP can be defined as the transportation of sounds mainly human speech through the Internet protocol (IP) packet in a digital form over the Internet and the data network gateway. One of the most important advantages of using VoIP technology is to save tariff. The Internet technology is cheap to use and hence the voice communication over the Internet protocol is widely used and accessed by the telephone users who are aware of the technology and familiar with the existing IP phone products and services by the vendors. Therefore it is clear that the most immediate attraction to VoIP is to save money on long-distance communication. (Ohrtman 2002).

Since the VoIP technology deals with transmission of voice over a packed switched IP network such as the Internet, therefore there are certain types of communication models which need to be supported by generic VoIP architecture. (Ganguly & Bhatnagar 2008.) These communication models can be listed as follows:

- Internet-to-Internet: The concept of this type of call is to connect both the originated and terminated phone to the Internet and reside the whole router inside the Internet.

- Internet-to-PSTN: Here callers have the phone connected to the Internet where receiver is connected to the PSTN.
- PSTN-to-Internet: In this system, caller is connected to the PSTN but the receiver is connected to the Internet.
- PSTN-to-PSTN: Here the whole call routing is done over the Internet protocol where the caller's and receiver's devices are connected to PSTN system.
- Internet-to-Internet via PSTN: Finally, there could be a situation that the callers' and receivers' devices are connected to Internet but a part of the call routing is done over the PSTN system.(Ganguly & Bhatnagar 2008.)

The above communication models are defined based on the contemporary and past technologies used so far in the telecommunication industry. Hence, the next chapter corresponds with this chapter to describe the generic functionality and design of VoIP on the basis of above communication models. Therefore in chapter 4, the architectural design and the functional components of VoIP are described in detail with the reference of above communication models which deal with IP telephone technology.

## 4 ARCHITECTURE AND FUNCTIONALITIES OF VOIP

This chapter deals with the architectural design phases of VoIP including the “VoIP functionalities” and “call processing mechanism”. It was important to know the architectural design and functionalities of VoIP in my research since the web phone interacts with technology to establish VoIP calls.

### 4.1 Overview of VoIP architecture

The minimum requirement to establish a VoIP phone call is to have two listening parties and each has to have a calling device equipped with VoIP codec and connected over the Internet protocol network (Ganguly & Bhatnagar 2008). However, to establish phone calls under different communication models (described in chapter 3) for instance PSTN enabled services many other functional components and features are being introduced in the basic architecture of VoIP. Although there is not any standard VoIP architecture which can deal with all the needed functionalities what users need but there are basic requirements of recognized architecture which are widely used in VoIP industry. Ganguly and Bhatnagar’s (2008, 18) suggestion on the area is that, “There exists no standardized VoIP architecture that can cover all the possible deployment scenarios and functionalities”, which validates the fact.

Therefore, different VoIP vendors have designed and implemented their own unique architecture to be able to compete in the VoIP industry by making them distinguished. Nevertheless it is possible to refer to root architecture for revealing the basic functional requirements and common functional components of a modern contemporary VoIP architecture.

### 4.2 Analysis of architectural requirements of VoIP

The basic architectural model by Ganguly & Bhatnagar (2008) of VoIP (described in chapter 3) requires meeting the following functional requirements:

Address specification: The initiation phase of a call requires figuring out the specific address location of destination which could be Uniform Resources Identifier (URI) or an Internet protocol or specific user ID in peer to peer application request. It is also possible to initiate a call to the PSTN destination by using PSTN phone numbers.

Inter-operatability of devices: Despite the fact of being architecturally different in VoIP devices by different vendors, devices should be able to interact with each other. For instance device made by vendor A should be able to establish connection with the device made by vendor B.

Association with PSTN: VoIP should be capable to establish phone call to and from PSTN. Therefore the architecture should facilitate the protocol level translation and data level transcoding.

Control of Session-level: The functionalities of sessions in different levels of using VoIP are important. For instance authentication, authorization, billing etc are part of it.

Functionalities in Media-level: This level is responsible for transporting voice over the media transport protocol.

Inter-operatability of components: The VoIP architecture is a complex architecture. Hence the modelling of the VoIP components during designing and the fundamental architecture of VoIP should contain the standard components. This is called Inter-operatability. For instance transportation protocols SIP and H.323 are inter-operatable regardless the fact that H.323 is a binary code based protocol and SIP is a script based protocol. The functionalities of SIP protocol are described in detail in chapter 5.

The functionalities mentioned above in the VoIP architecture are not complete. However, they still are well recognized and used to design VoIP architecture in different levels.

### 4.3 VoIP Functional Components

The data transmissions are done in different layers of data communication technology which are core background of VoIP. In an Internet model the layers are

1. Application layer.
2. Transport layer.
3. Internet layer.
4. Network interface layer. (Carne 2004.)

Based on the Internet model, functional components of VoIP are formulated. As the generic architecture of VoIP is described above, the functional components of VoIP which consist of VoIP calling devices, Gateway, media server and session control server has been described based on the generic model concept. (Ganguly & Bhatnagar 2008, 21-22.) Though in my research, I mostly focused on topic related with media server and session control server but the basic idea of the components has been discussed briefly below to acquire clear understanding of VoIP functions.

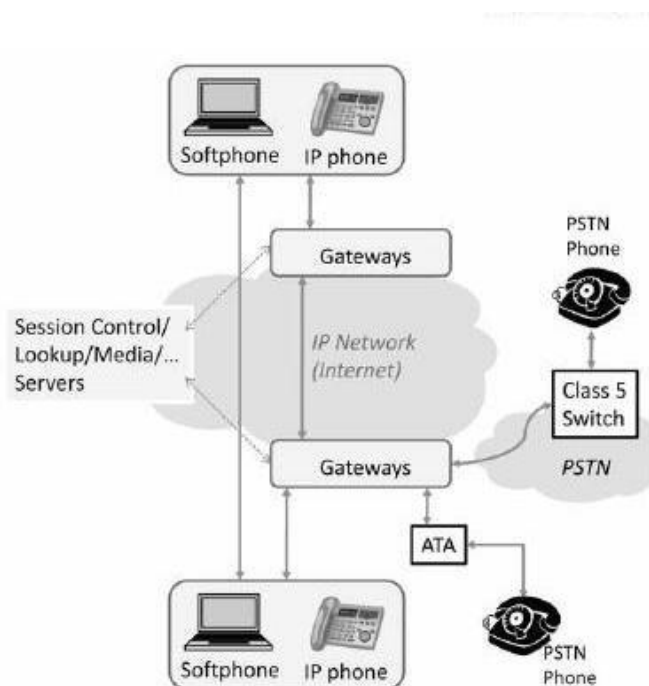
Calling devices: There are required devices to initiate a VoIP call what end user requires. These are

1. IP phone.
2. Soft-phone.
3. Analogue telephone.
4. Analogue Telephone Adapter. (Ganguly & Bhatnagar 2008, 21.)

VoIP Gateway: Gateway plays the active role to connect two dissimilar networks. A Gateway consists of two important components which are media gateway and gateway controller. The main task of gateway is to establish communication between two networks. This task is done by translating information into a common format and makes the information be able to understand by the networks. (Ganguly & Bhatnagar 2008, 21.) The component is important in base architecture of VoIP. In other words, Ohrtman's (December 2002) suggestion on gateway is that "To date, it has been impractical to route VoIP "desktop to desktop" meaning Internetworking between PSTN and IP networks must be facilitated. This is done with a gateway."

Media server: There are certain phases of architecture deployment where the media gateway functionalities can be extended by media server. The media server basically processes the VoIP Real-Time Transportation Protocol (RTP) to be able to decode the Dual-Tone Multi Frequency (DTMF), Mix multiple media streams, process VoiceXML scripts, record audio, recognize speech etc. (Ganguly & Bhatnagar 2008, 22.)

Session control server: The role of session control server is to provide session level functionalities. For instance, Session control server enables authentication, authorization and admission of VoIP calls. This control server's role can also support call routing and forwarding to another networks or service providers and maintain states about outgoing calls (Ganguly & Bhatnagar 2008, 22.) Figure 2 demonstrates the generic architecture of VoIP.



**Figure 2.** VoIP architecture (Ganguly & Bhatnagar 2008)

Protocols of VoIP: VoIP requires different protocols to transmit information. Hence protocols are important to be analyzed to go further with the idea of web-phone design. Practically protocol defines the session of communication mechanism which must be followed. From the VoIP perspective, protocols are used to establish end to end voice

call. The categories of protocols according to Ganguly & Bhatnagar (2008, 60) are defined and described briefly as follows:

- Address discovery protocols
- Signalling protocols
- Gateway control protocol
- Media transport protocol

From the perspective of VoIP, two types of protocols are used to accomplish core tasks. They are Call Signalling Protocol and Media Transport Protocol

Call Signalling Protocol: These protocol family controls the functions related to set up and maintain a VoIP call (Ganguly & Bhatnagar 2008). The tasks are following:

- Determine the receiver's location
- Determine the availability of receiver
- Agree upon the session parameters
- Session modification
- Termination of session. (Ganguly & Bhatnagar 2008.)

Media Transport Protocols: These protocols family carries the task of actual voice transmission. The voice is nothing but packets of information which are carried to both parties using these protocols family. Hence voice transmission, encoding and decoding of voice signals and packetization of speech are done by using these protocols. (Ganguly & Bhatnagar 2008.)

In my bachelor thesis, I analyzed the specific type of call signalling protocol to design the soft-phone which enables the end users to make phone call over the Internet by using Internet browser only. Therefore, I focused more on call signalling protocol especially on Session Initiation Protocol (SIP) in the following chapter in detail with the architecture of SIP and its interaction with Request for Comments (RFC).

## 5 SIP NETWORK ARCHITECTURE ANALYSES AND RESPONDING MECHANISM

This chapter is the base of designing the browser based web phone. In previous chapters, I formulated the background information in a way that the chapters are interconnected so that readers without prior knowledge on telecommunication technology can also build the basic concept on voice communication technology and clearly understand my research intention. Nevertheless, all the chapters are independent and intended to reveal certain information on specific areas of telecommunication technologies. It is difficult to go further without having solid background of SIP and RFC mechanism. Therefore, in this chapter I analyzed the signaling protocol what I proposed to be used for the implementation for the idea and also designed the software model on the basis of analyzed result on SIP network architecture and mechanisms of SIP in next chapter. To understand the concept of the protocol and mechanism, I describe the entities of SIP and RFC with solid background information.

### 5.1 Overview of SIP

SIP was developed by Internet Engineering Task Force (IETF) as merger of two different protocols named SIP and Simple Conference Invitation Protocol (SCIP) which proposed to accomplish similar goals of data communication. Johnston (2003) lists the main functionalities of SIP as follows

- End point location define
- Contact an end point to validate the willingness to establish a session.
- Media information interchange to allowed session to be established
- Media session modify.



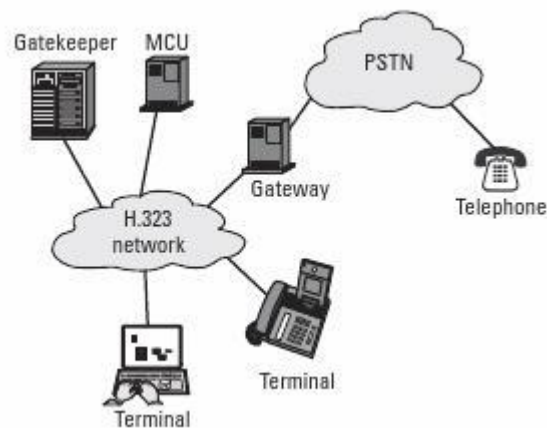
## 5.2 Comparison of SIP with H.323 protocol

There are different signaling protocols existing in voice and video data communication in telecommunication technology. In my research project, I chose the most prominent and established protocol used in the area of telecommunication technology which is H.323 to compare with SIP. (Ganguly & Bhatnaga 2008.) The comparison verifies the advantages of using SIP to design the web phone.

H.323, developed by International Telecommunication Union Standardization Sector (ITU-T) has been a widely used protocol in the data transmission area in telecommunication technology. However, I found SIP as the rich featured protocol for my research project after the feature and flexibility comparison analysis with H.323. To understand the difference between SIP and H.323 and decision to choose SIP as the protocol, it is important to have the basic concept on H.323 and the focused area of H.323 in telecommunication system.

The H.323 protocol is a member of the H.32x protocol family which intended to cover the general functionalities of multimedia communication over Integrated Service Digital Network (ISDN), broadband, PSTN and packet over IP network (Packetizer Inc. 2011). According to Joston, (2003, 181), “H.323 is an umbrella recommendation that covers all the aspect of communication over packet networks.”

The basic elements of a H.323 are gatekeeper, multi point control unit (MCU), PSTN, gateway, telephone, H.323 network and the terminal (Johston 2003, 182). The following figure demonstrates the elements of H.323.



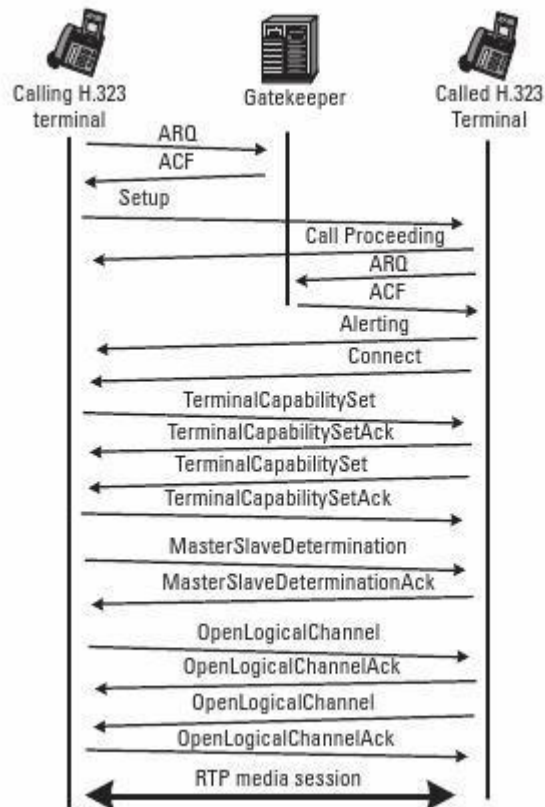
**Figure 3.** H.323 network (Johston 2003, 182)

The basic call flow involves a gatekeeper and two terminals in H.323 network. The call flow also involves many elements interacting with each other using different other protocols for establishing a call session. The call starts with exchanging H.225.0 Registration Admission Status (RAS) between the gatekeeper and terminals. The process also includes the terminal registration with gatekeepers. Here Registration Request (RRQ) message is used by the terminals and gatekeeper. The call initiating terminal requests to the gatekeeper an Admission Request (ARQ) message with the address of receiver (ex. E.164 telephone number, URL or e-mail, Unique ID of H.323 alias) and expected session type. Hence the gatekeeper has all the information about both caller and also address of receiver. (Jones 2007.) The gatekeeper plays important role by checking following things lists below:

- Identity of Calling terminal (checking authorization to make call)
- Sufficiency of band-width
- Availability of resources

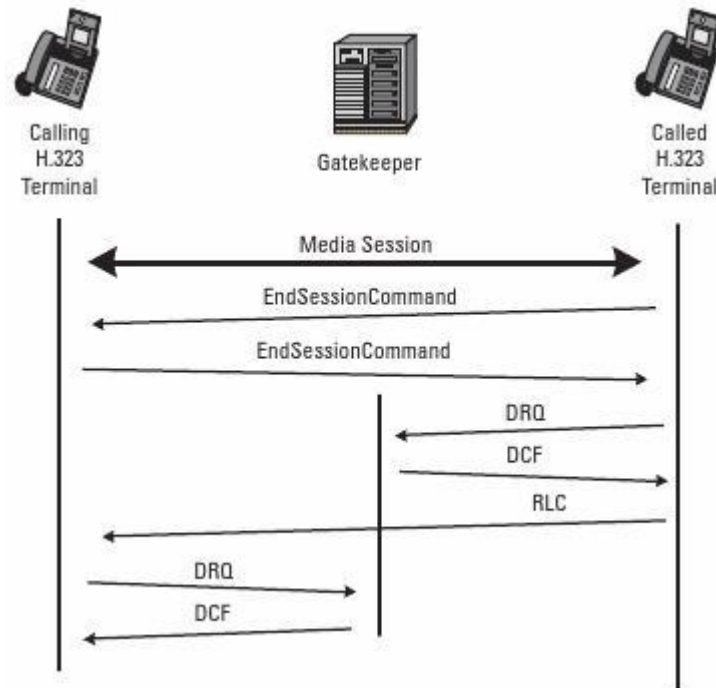
After checking the status of above criteria, the gatekeeper then forwards the call. It uses Admission Conformation (ACF) to the receiving terminal. The gatekeeper also translates the receiver's terminal ARQ into transport address received in ACF. The calling terminal then opens the Transmission Control Protocol (TCP) to the receiver terminal. Then it precedes the call by sending Q.931 setup message to the destination terminal. The destination terminal returns with a call proceeding response to the caller terminal. During this message exchange process, the gatekeeper is asked to permit to accept the call by destination terminal. After receiving the ACF from gatekeeper, the called terminal sends an alert message to the users. Then comes the connection negotiation phase where H.245 handles the control signal messages. Following the control signal messages, the calling terminal opens the port selected by destination terminal and sends connect message. The calling terminal has to open the TCP connection between two terminals to do that. Then the media capability is checked by the both terminals by sending TerminalCapabilitySetmessage. Then the MasterSlaveDetermination message needs to be exchanged by the both terminals to specify the master and slave terminal since it is required by the H.323 protocol. The hierarchical number contains by the terminals is responsible to set master and slave terminal but if the terminals type is same, a random number which is also contain by the

terminals set the master and slave terminals. The final stage of setting up call is to open the logical channels between calling and destination terminal by sending H.245 OpenLogicalChannelmessgae which are used to set and control the media channels in H.323 networking. Finally the RTP and RTCP control packets are sent over the IP and using the opened port exchanged by the OpenLogicalChannel messages. (Johston 2003, 183-187.) Figure 4 visualizes the call flow system in H.323 protocol.



**Figure 4.** Call flow in H.323 (Joston 2003)

After the call session, it requires to end. The call session ends by sending EndSessionCommand message sent by either terminal. Figure 5 demonstrates the call tears down sequence in H.323.



**Figure 5.** Call tear down sequence in H.323 (Joston 2003)

Difference with SIP: Unlike H.323 protocol which uses complex binary encoded ASN.1 message, SIP is a simple text based protocol. It is easier to implement than H.323 since it is possible to script it easily. Hence SIP has made the application development for the Internet and web technology more flexible than H.323. Furthermore, it has the instant message capability where H.323 is exclusively a signaling protocol. (Johston 2003, 187.) Moreover, SIP uses less CPU than H.323 in voice server for its smaller message size amount (Packetizer Inc. 2011).

The main objective of my research project is completely related with telecommunication and web technology which establishes a bridge between two parties over Internet Protocol (IP). These two parties might be using different protocol suits for communicating as their mean but at least one party must use web technology exclusively to make the phone call, whereas other party might use PSTN or web technology. In view of transmitting data over the network, the way of dealing with the media and signaling protocol affects on the whole communication system. The protocols are assumed to be inter-operatable and compatible. Hence, it is easier to use

SIP than H.323 in web related software development. Therefore I decided to go with SIP in designing the browser-based web phone.

### 5.3 SIP Architecture: Entities and Functionalities

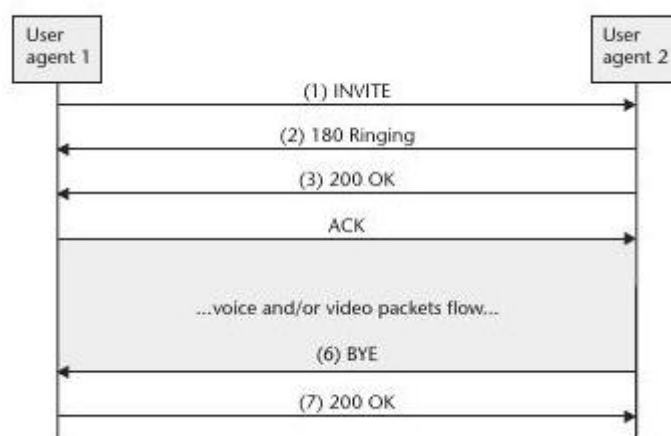
SIP is composed of two basic components. They are the user agent and the network server. The user agent component is responsible for the call network and the server manages the signaling issues related with the call. (Johston 2003.) The user agent is divided into two segments that are User Agent Client (UAC) and User Agent Server (UAS).

The UAC is responsible for initiating a call and UAS is responsible for answer a call. This enables peer to peer call by using the client server protocol in a SIP network.

Unlike the UAC, the UAS is responsible for session set up and media control in SIP network architecture. Hence the main functions include

- Location identification of receiver. (The location identification is important since caller is unable to specify the called end location)
- Passing messages to other network servers.

The Figure 6 demonstrates the call signaling flow in SIP.



**Figure 6.** A call signaling flow in SIP (Stafford 2004)

There are two different types of modes exist which operate SIP server (Johston 2003). The modes are Stateful and Stateless.

The difference between these two modes is straight forward. Under the stateful mode, a server is able to store the incoming and outgoing call's responses. Unlike stateful mode, stateless mode is more flexible since it does not store the responses of call after ending the session. The stateless server works with the SIP signaling infrastructure whereas stateful server is closer to the local devices of user agents and domain control. (Johston 2003.)

Media tools: SIP delivers description of session to its user agents. If it is a voice session, user agents are responsible to provide audio tools or if it is a video session or audio and video session then both video and audio tools have to be available. Without the media tool, session establishment is not possible. (Camarillo 2001, 100.) Since SIP is widely used protocol and run on many different systems, it is highly likely that the device types are different and vary from system to system. Nevertheless, device type doesn't affect on SIP behavior at all.

SIP session establishment process: Russel (2008) defines session as “A session is a logical connection between two entities wishing to communicate.” Establishing a session in SIP requires different stages and message flow since it is not only used for a voice session but also for e-mail, text message or in video conference. Therefore, session are established after several steps of framework.

The frame work steps which should be done first listed below and discussed further:

1. Network access
2. Dialog initiation
3. Client request
4. Response by server
5. Routing
6. Session Modification
7. Session termination

Network access: This is the very first step what is needed to establish a session in SIP. The devices intended to participate in SIP session first look forward to network availability.

Dialog initiation: This dialog initiation step is done after checking the network connection. At this phase, devices are ready to communicate over the IP or PSTN network cloud. The dialog initiation phase is also known as virtual connection. The mechanism used by SIP is “offer and accept” mechanism to establish a session. Therefore one of the entities of UAC has to initiate an offer to UAS which contains the parameter to establish a connection. If the destination party is ready to accept the call, UAS sends an answer to UAC. This is how the UAS knows that both client entities are agreed to initiate dialogue. (Russel 2008, 93-94.)

Client request: UAC has to initiate all the requests to establish a session. The mechanism is to simply offer and response in the client request phase. The request initiated by UAC is supposed to be responded by UAS. There are certain methods used to identify the type requests. (Russel 2008, 97-98.) The methods are

- INVITE method intends to establish virtual connection to transmit data over the network between the entities. The data could be voice calls, e-mails, files etc.
- BYE method is used to release the established session. This method can only be used after establishing a dialogue and session.
- ACK is ‘Acknowledgement’ in data transmission. ACK method is used to complete the process of establishing a session by sending “packet received” message.
- CANCEL method is used to cancel a session or cancel a previous request which was sent to establish a session. It works similar like BYE method but it does not require a session to be established.
- REGISTER method is used to register the device location used by the UAC with the server.
- OPTIONS method checks the resources availability on a server.
- MESSAGE method is used to transmit simple text message

Response by server: There are six different responses done by a SIP server. The codecs are used by level wise for each type of response (Russel 2008, 99). The message families correspond to SIP codecs level are given in Table 1.

**Table1.** SIP server response

Server response type	Codec
Provisional message	1xx
Successful status	2xx
Redirection status	3xx
Client failure message	4xx
Server failure message	5xx
Global failure message	6xx

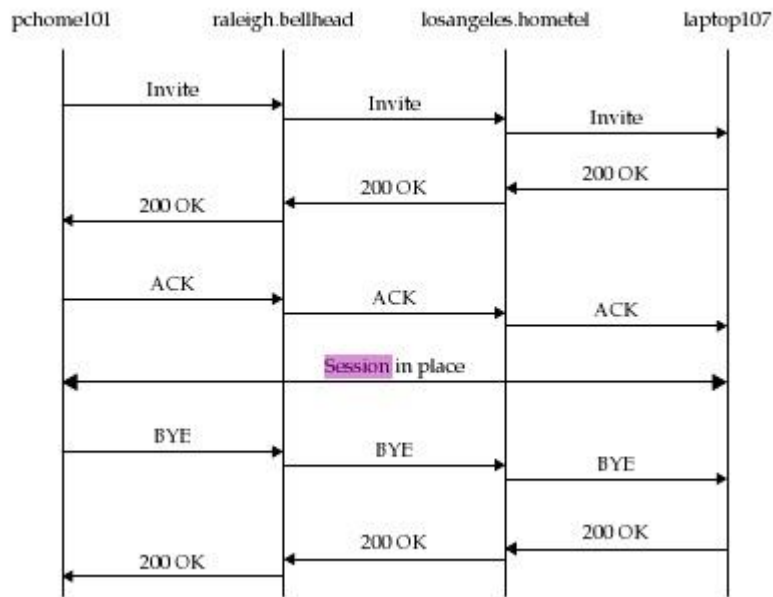
The messages given by server have also specific codes. It is important to have knowledge on the message code to handle server operational events. One example code of message of each type is also given in a table below. The whole list is included at the end of the report in appendix 2.

**Table 2.** Message code of SIP request

Code	Message
180	Ringing
200	OK
301	Moved permanently
400	Bad request
500	Server internal error
606	Not acceptable

Routing: Router is used to locate the devices by checking their identity in a network. All the devices which are connected to the network have specific identity to be able to resolve the exact address. SIP uses Universal Resource Identifier (URI) to resolve the exact IP address used by Domain Name Server (DNS). (Russel 2008, 102-105.) The following figure demonstrates the routing process of SIP.





**Figure 7.** SIP routing example (Russel 2008, 104)

Session modification: Session can be modified as per the need of clients at any point of session. If a virtual conference is going on and some of the participants want to leave or join, the session can be added, deleted or modified according to client's need. (Russel 2008, 111-112.)

Session termination: Termination of session can be done by any of the UA. Session can be terminated by external entities other than UA. For instance, global error might occur and unavailability of the resources might cause the termination of a session. It is important to note that establishment of a session requires to take place prior to terminate a session. (Russel 2008, 112-113.)

#### 5.4 SIP & RFCs

RFC stands for Request for Comments in data transmission technology. The Internet Engineering task force (IETF) publishes documentation on the Internet or Internet connected system to describe their methods, behavior, protocols or research which is formally known as RFC.

SIP is RFC 3261. The request mechanism described in SIP was formerly known as RFC 2543. (Rosenberg & Schulzrinne & Camarillo & Johnston & Peterson & Sparks & Handley & Schooler 2002.) In practice, there are different protocols used to collaborate

with SIP although SIP is not dependent on others. Some important SIP related RFCs are given in the following table:

**Table 3:** SIP related RFCs

RFC	Description
2327	Session Development Protocol
2486	Network Access Identifier
2778	Model for Presence & Instance Messaging
2779	Instant Messaging or Presence Protocol requirements
2848	PSTN and Internet Networking (PINT)
3050	SIP Common Gateway Interface
3262	SIP Reliability of Provisional Responses
3263	SIP Locating Server
3329	SIP Security Mechanism Agreement
3824	Using SIP with E.164 number
3487	Requirements for Resources Priority Mechanism for SIP

## 6 DESIGN AND IMPLEMENTATION OF THE BROWSER-BASED WEB-PHONE

In this chapter, I describe the design process of the web phone and provide recommendation for implementation of the web phone based on software design standard and analyzed result on previous study.

The chapter focuses and formulates the general idea of web phone, system design requirement analysis and specifications, hardware and software requirement analysis and specifications of SIP telephony, required components of web-phone, UML diagram specifications, interface design, calls processing and finally provides recommendation for implementation.

### 6.1 General idea of web-phone

The web phone is a website embedded browser based phone where user can login to a specific website using Internet browser and make phone call over the Internet to either a personal computer (peer-to-peer) or to PSTN. The idea requires meeting a number of specific criteria which distinguish the web phone from existing technologies. The criteria are

- End-users must not require installing any external software to their system to make or receive a call.
- End-user should be able to make phone call over the website to PSTN
- Web-phone System embedded website should be available to the user on any voice communication device which is connected to the network

In general, existing available VoIP software require downloading and installing a binary execution file on client's system to be able to operate with the network server. The software download and installation process might face the following problems:

- End-users might not have the right to download and install any external software if the system is different than users' personal system.

- End-users might not have the knowledge on installation process thus might find the set up process complex.
- Lack of knowledge on security and installation might cause unsecured software installation which eventually cause serious harm to the clients system.

The web phone solves the above problems by dealing with them practically with no complex installation process. The web phone shows the user interface-the dial pad on a website and operates the phone call requests over the website server.

## 6.2 System design: requirement analysis and specification

Requirements are predefine set of system functions, attributes and development stages as a specification of what should be accomplished during the development process. They describe the system behavior and implementation phases of a system. (Sommerville & Sawyer 1997, 4.) Therefore it is important to predefine the system requirement to continue the designing process. The system requirements have been divided into two fundamental segments. They segments are hardware requirements and software requirements.

The system is based on both software engineering and telecommunication engineering. Therefore, there had to be a common ground where the two different technologies meet. The common ground is web technology because the idea of web phone comes from web technology which is part of software engineering and voice over the Internet protocol to PSTN which is part of telecommunication technology. Hence the requirements come from the analysis of web technology requirements on communication technology.

The web-phone is a web browser based telephone which is able to be embedded in the website and communicate with server to establish communication between the clients over the Internet protocol. The communication should not restrict the users to make phone call from PSTN and require installing any external software. Therefore, SIP device telephony requirements have to be specified prior to design the interface and set up the voice communication as well as web server.

### 6.2.1 Hardware Requirement analysis and specification of SIP devices

The draft RFC-4504 published by IETF describes the requirements of SIP telephony devices. Since SIP is a powerful protocol which is able to establish voice communication, video streaming, instant messaging, text messaging, its device has the requirements based on the different media.

The draft RFC-4504 was chosen to specify the SIP device requirements because the research scope was voice communication using SIP and did not cover the requirements of non-voice media. The documentation explains the SIP device requirements in details. However, the generic requirements type and requests which must be met by all SIP telephony devices are described in brief and the address of the document is added in the source list in the end to for further reading on RFC-4504. (Sinnreich & Lass & Stredicke 2006)

- SIP Telephony Devices
- DNS and ENUM support
- SIP device Resident Telephony Features
- Support for SIP services
- Basic Telephony and presence Information Support
- Emergence and Resource Priority Support
- Multi-Line requirements
- User Mobility
- International Requirements
- Web based feature management
- Other related protocols
- Media Requirements
- Voice Codecs. (Sinnreich et al. 2006.)

### 6.2.2 Software requirements

The SIP technology requires a number of software requirements to be executed. They are divided into two different ends. The ends are Client Side and Server Side.

Client side software requirement: An operating system with GUI feature is needed to be able to run the web phone. Since the web phone is supposed to be embedded inside a website, hence the following resources should be available on client's system

- Network connection (Ex. LAN or wireless)
- Access to the web phone embedded Web page

Server-side software requirement: A voice communication server (ex. asterisk) has to be installed on server side. The SIP embedded website should be able to communicate over network or mutually if they are on same machine.

### 6.3 Required component of web-phone

The web-phone requires following component to be executed:

1. A website where the dial pad is embedded
2. Log in field to validate user's identification
- 3 An interface of dial pad
4. Web server with voice communication server enabled.

### 6.4 UML diagram specification

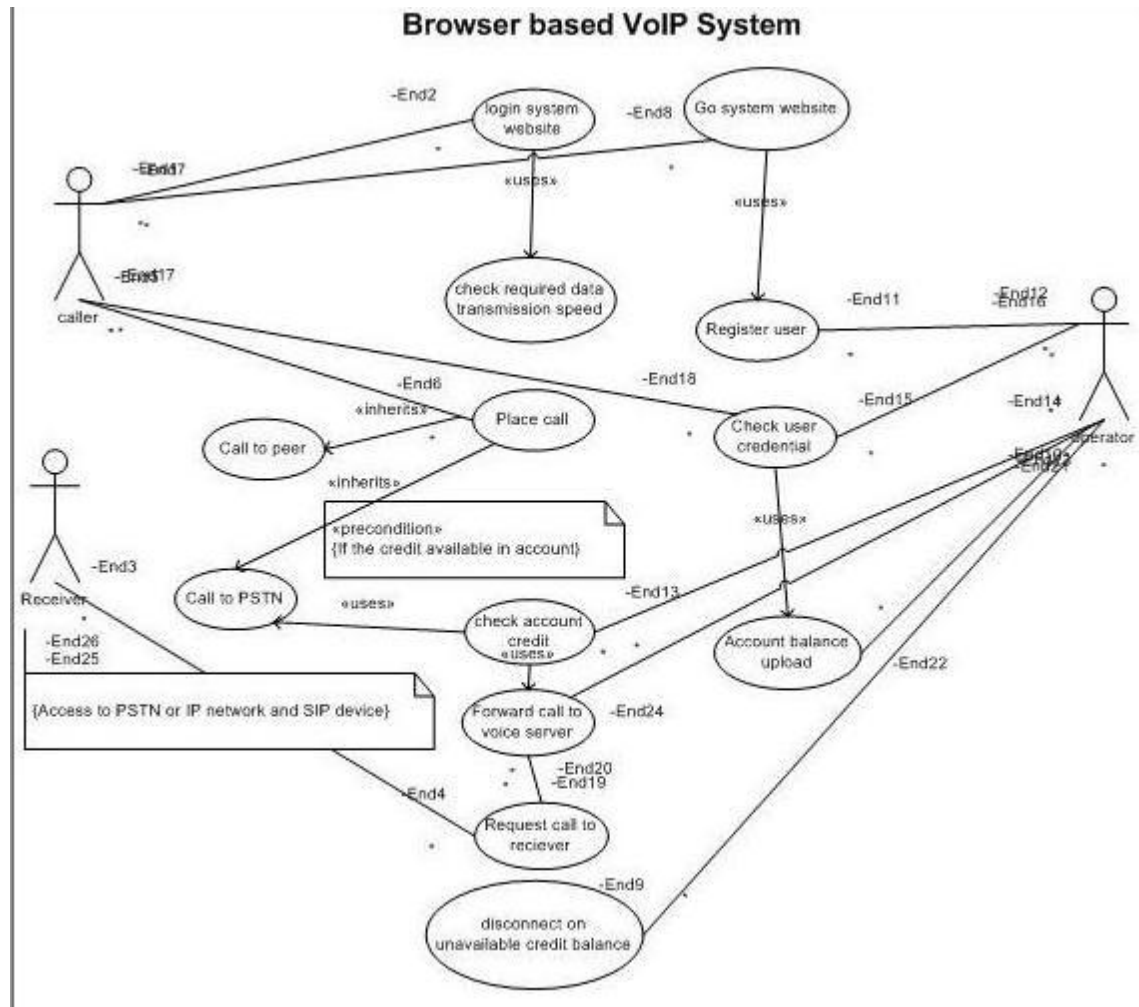
The visual representation of the design was done by using UML. The UML modeling language is used widely in software design as standard. The UML has several versions available since it was released by Object Management Group (OMG). OMG defined the method of UML components in Meta Object Facility (MOF) which was used as standard during design the UML diagrams (OMG Inc. 2011). Since the design is contemporary and new, therefore UML version 2.0 was used to design the system.

Use case specification: Use case diagrams visually represent the abstraction of a system with the specification of the users' expectations from the system. Use cases specify the behavior and attributes of a system and the system's entities. In other words, Pender's (2003, 59), suggestion about use case is that "The use case diagram models the users' expectations for using the system. The people and system that interact with target system are called actors. The features of the system that the actors use are called use cases."

Therefore, specifications of Use Cases are important since they formulate the following actions related to the user and the system

[illegible]

The above meta model diagram specifies the requirements of use cases in an information system. Therefore, the diagram was used as standard for the purpose of requirements specification of the use cases of web phone in the designing process. Hence, compliance with the MOF standard I designed the Use Case diagram of the web phone. The figure 9 demonstrates the use cases of the web phone.



**Figure 9:** Use case diagram of web phone

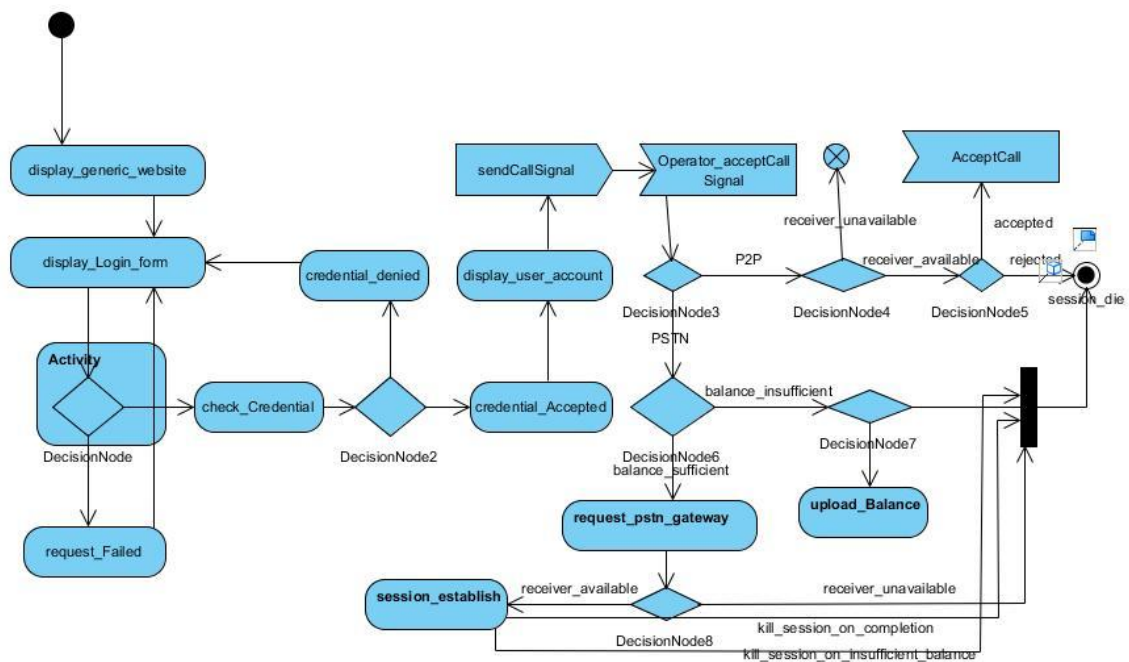
**Activity diagram:** The activity diagram shows the scenario of actions in a system. Activity diagram is often considered as functional description of processes that connect logically to each other. In earlier software engineering approach, the logical connections between the functions were showed in flow chart which was part of functional modeling.

Although functional modeling approach compare to object oriented (OO) modeling was less efficient, but OO development methods do not omit the necessity of functional modeling. It rather combines the functional, which is still the base of application design, and OO modeling and represents the method in a sophisticated way. Hence UML represents the functional modeling as the activity diagram (Pender 2003, 447.)



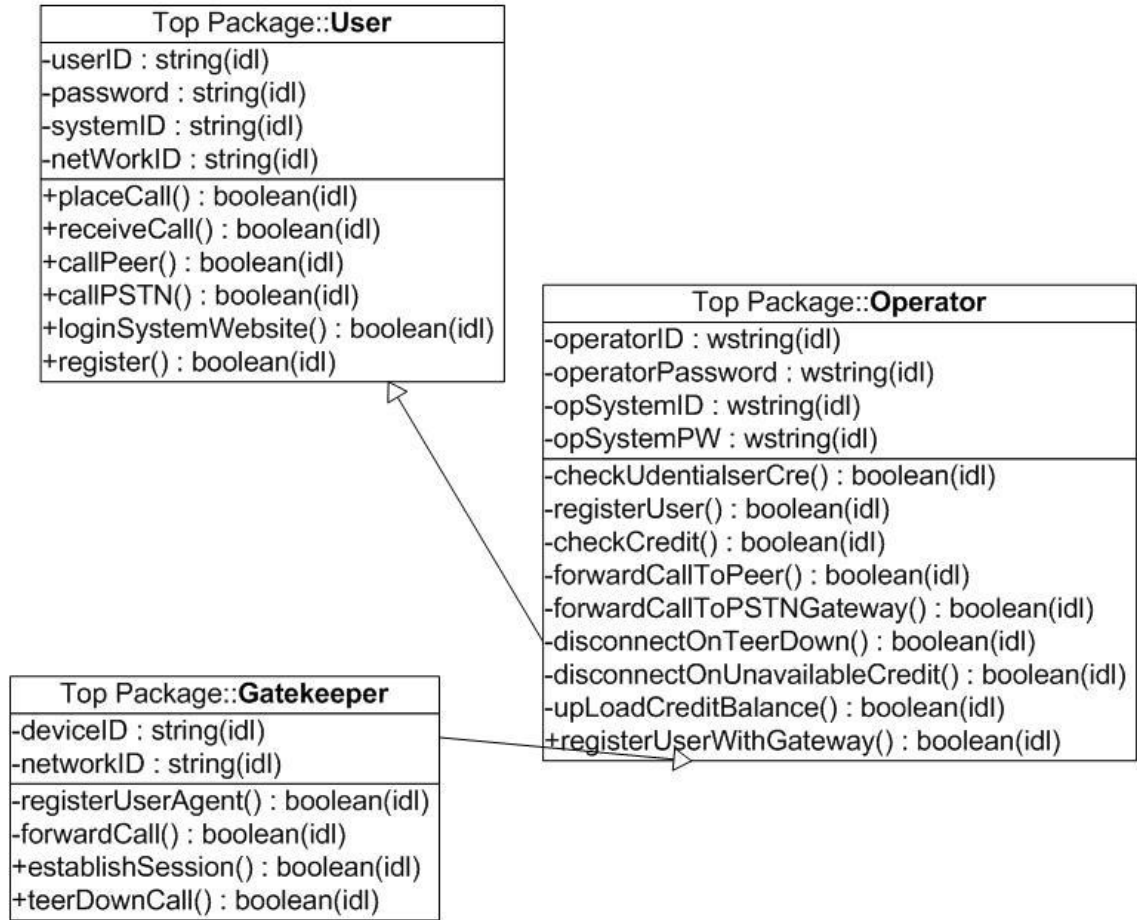
The activity diagram modeling with UML 1.4 and UML 2.0 is different. In software design where most other UML diagrams are used to assign and provided notations for object responsibility, the activity diagram in UML 2.0 sets the responsibility in packages to focus on the behavior of the object to accomplish the tasks (Pender 2003, 465).

The activity diagram in Figure 10 describes the activities and functional behavior of web phone.



**Figure 10:** Web phone activity diagram

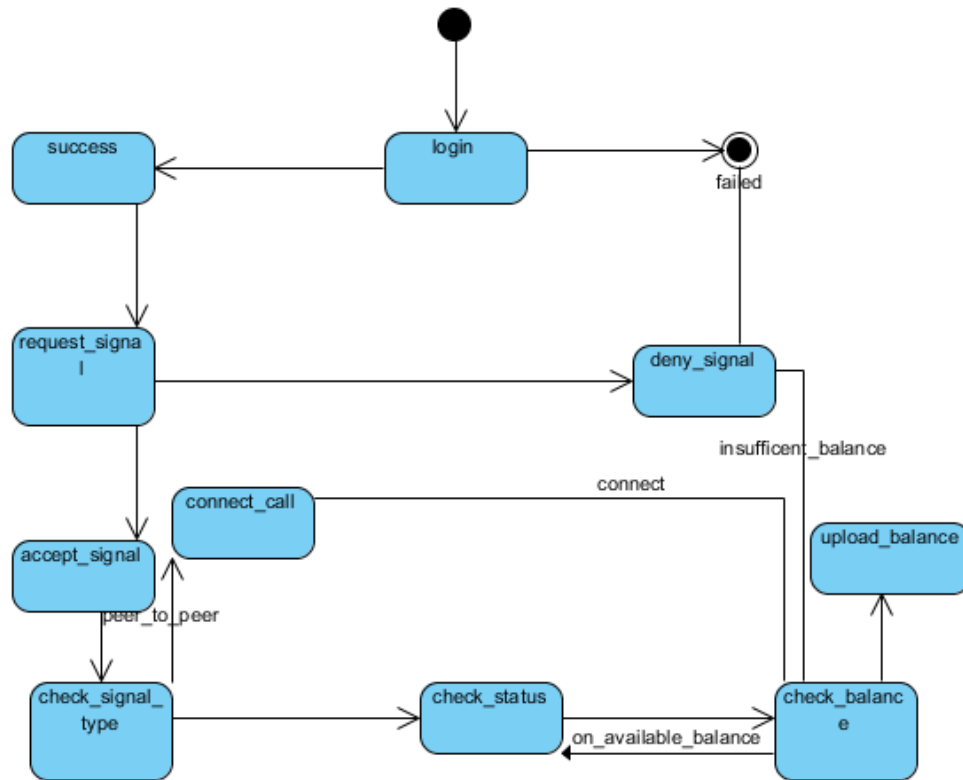
**Class diagram:** The class diagram formulates the object attributes and functions and represents them as a package. From the OO point of view, a class acts as a blue print of object. A class diagram is the center of all UML diagram modeling since it feeds data from all the fields and work as a mean of both forward, model to code, and reverse ,code to model, engineering (Pender 2003, 105). Figure 11 demonstrates the class diagram of the web phone.



**Figure 11:** Class diagram of web phone

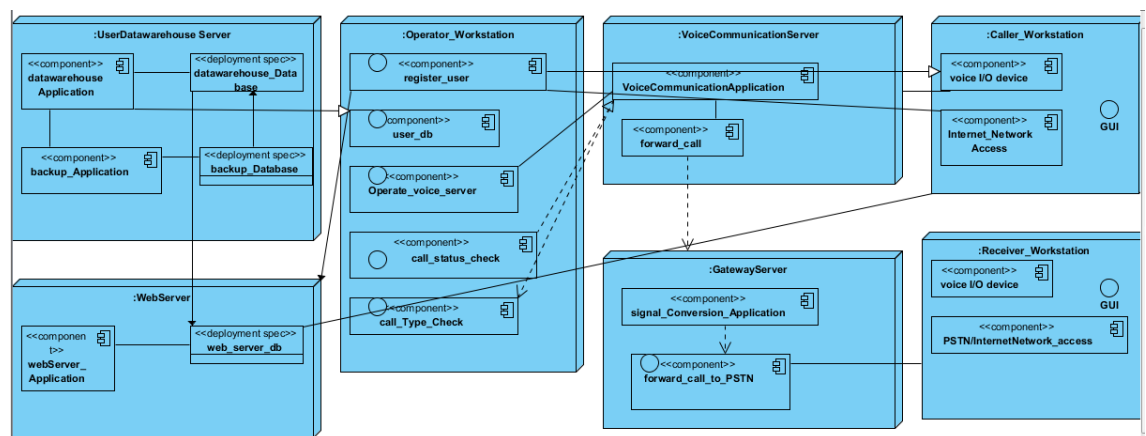
Interaction diagrams: The interaction diagrams represent the web phone objects communication to accomplish tasks. In UML 2.0, the interactions diagrams consist of sequence diagram, communication diagram and timing diagram. Each of these diagrams provides the object communication in a system (Pender, 2003, 61). However, timing diagram of the system is depended on the hardware compatibility, voice server and PSTN gateway. Therefore, timing diagram was not considered in the design of web phone. Hence the sequence diagram and communication diagram are part of the thesis and included in as Appendix 1.

State Machine diagrams: State machine diagram, state chart before UML 2, represents the system status after every specific event. Figure 12 represents the state machine diagram of the web phone.



**Figure 12.** State machine diagram of web phone

Deployment diagram: Deployment diagram represents the run time configuration of the web-phone system. It shows the hardware, software and middleware relationship on run time environment. Figure 13 demonstrates the deployment diagram of the web phone.



**Figure 13.** Deployment diagram of web phone

## 6.5 Interface design

The web-phone interface needs to meet certain requirements to establish call. Since the web phone design is intended to communicate with both ordinary networked computer and PSTN devices, the requirements are separated into two segments according to the need. The segments are PSTN device requirements for interface and Networked Computer requirements for interface

PSTN device requirement for interface: To make a phone call over the Internet network to PSTN network (ex. E.164) the interface must have following features

1. A field to input telephone numbers manually
2. A dial pad to input the numbers by using GUI
3. A call/Receiver button

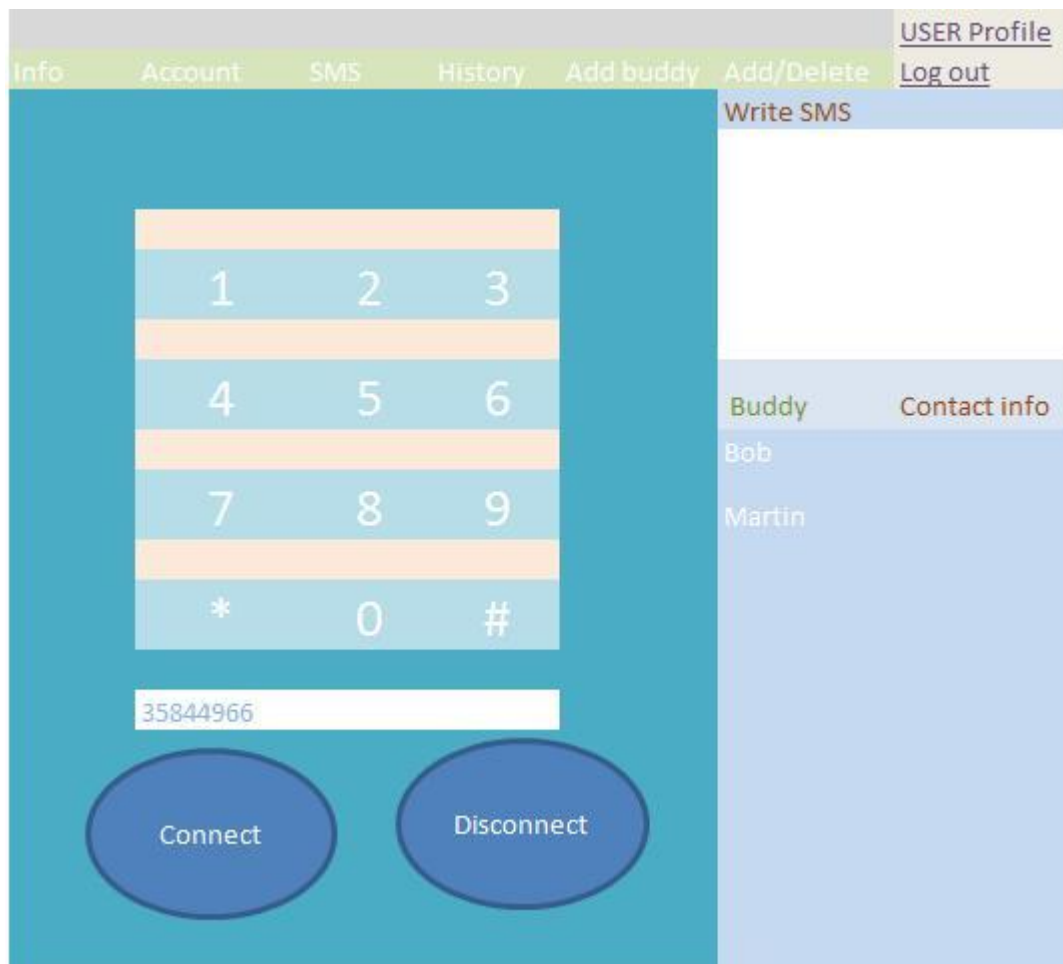
Networked Computer requirements for interface: Establishing pc to pc phone call (Peer to Peer) is different than PSTN. In this communication phase, both the caller and the receiver have to agreed beforehand to establish a phone call. The task can be accomplished by adding each other in a list in web phone account. Therefore the interface requires the following criteria to meet in the web phone features.

1. A search field to search users
2. A add button to send the add request
3. A call button- active only on click the intended user's name

A disconnect button on ending call and a volume controller on the interface are common features on the web-phone.

However, Optional features like call history, information tab and other symbols which represent the same action can also be integrated with the interface.

Based on the above requirements, a horizontal prototype of the web phone interface has been designed. Figure 14 demonstrates the interface prototype of web phone:



**Figure 14:** Website embedded Web phone interface prototype

The given interface is a horizontal prototype of the web phone and can be modified according to the requirements of the website where the web phone will be used. My recommendation is to design the web phone interface in compliance with Nielsen's heuristics of user interface.

## 6.6 Call Processing Language Overview

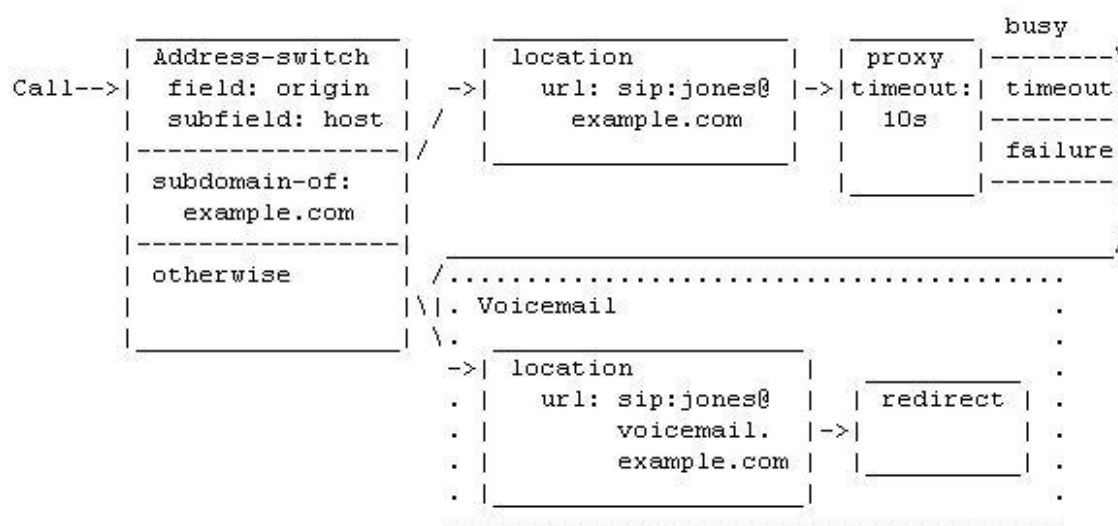
Call Processing Language (CPL) describes and controls the Internet telephone services. CPL is a protocol architecture independent language which can tie up with any protocol. Unlike other application programming languages, CPL does not allow to write loop or recursion or any other external source code by user for establishing secure call. To avoid the complexity and do the parsing easily, the language is designed graphically editable and written on Extensible Markup Language (XML). (Lennox & Wu & Schulzrinne 2004.) A typical CPL code example taken from CPL documentation by IETF Network Working Group is shown below:

```

<?xml version="1.0" encoding="UTF-8"?>
<cplxmlns="urn:ietf:params:xml:ns:cpl"
xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
xsi:schemaLocation="urn:ietf:params:xml:ns:cpl cpl.xsd ">
<subaction id="voicemail">
<location url="sip:jones@voicemail.webphone.com">
<redirect />
</location>
</subaction>
<incoming>
<address-switch field="origin" subfield="host">
<address subdomain-of="example.com">
<location url="sip:jones@example.com">
<proxy timeout="10">
<busy><sub ref="voicemail" /></busy>
</proxy>
</location>

```

Figure 15 visually demonstrates the CPL actions to make the concept clear on it



**Figure 15.** Sample CPL action (Lennox et al, 2004)

## 6.7 Recommendation for implementation

The implementation of web phone has to meet the following requirements:

1. Access to web server
2. Access to voice communication server
3. A soft phone embedded inside the website
4. Gateway to PSTN

5. An IP phone

6. A network card for voice communication

Since the actual implementation was beyond the thesis scope, I worked on setting up a voice communication server only to get hands on experience on voice communication technology. My recommendation is to use ‘Asterisk’ as communication server since this is a free communication server. Asterisk is an open source voice communication server which enables a computer to become a VoIP server (Digium Inc. 2010). Asterisk is based on Linux operating system. Since it is a server by default, it did not require any Graphical User Interface (GUI). However, the latest released ‘asterisknow’ has the GUI set up facility to make the configuration process easier. The installation documentation can be found on asterisk’s official web site of ‘Asterisk’.

After setting up the voice communication server, my recommendations for further steps are

- A SIP enabled web phone should be embedded to the desired website.
- Then the website should establish connection with voice communication server
- Voice communication server has to have the access to PSTN gateway to establish call to E.164 numbers

The recommended suggestions for implementation should also consider certain security procedure to mitigate the risk since it is intended for website. Next sub chapter will describe the risk, threats and vulnerabilities of information security over the Internet in the area of VoIP and SIP.

#### 6.8 Security policies and risk management of web phone

The sub chapter describes the security risk of web phone and VoIP signaling protocol-SIP since the phone was designed based on both technology and embedded inside a website and also discusses briefly the policies of mitigate risk. To understand the overall security risk of the web phone is complicated. To make it clear and understandable, I break down the security and risk management of the system in two

sub-sections to focus on the core concept of security policies and management of the risks that are security policies of VoIP and security risk and mitigation policies of SIP

#### 6.8.1 Security policies of VoIP

There are security differences between PSTN and VoIP as PSTN is secured due to its' perimeter and physical security whereas VoIP eliminates the barriers since VoIP is operated over the Internet protocol. Therefore VoIP needs more security level than PSTN. Since the whole VoIP operates over the Internet, all the security issues imply on Internet data security also imply on VoIP also. The common security requirements in the Internet technology set the following security policies. (Johston & Piscitello 2006, 3, 73-84.)

- Authentication- Validating users credential
- Authorization- Controlling the access list
- Auditing- Analysis the unauthorized execution or attempt of execution of programs
- Monitoring log files- monitoring users log files to keep track of executed and visited programs

The mentioned security policies should be applied to the following area of a VoIP network:

- Signaling protocol security
- Media Security
- PSTN Gateway Security

In my thesis, I analyzed only signaling protocol security since the Media Security and PSTN gateway security is substantially related with hardware technology which was out of the research scope.

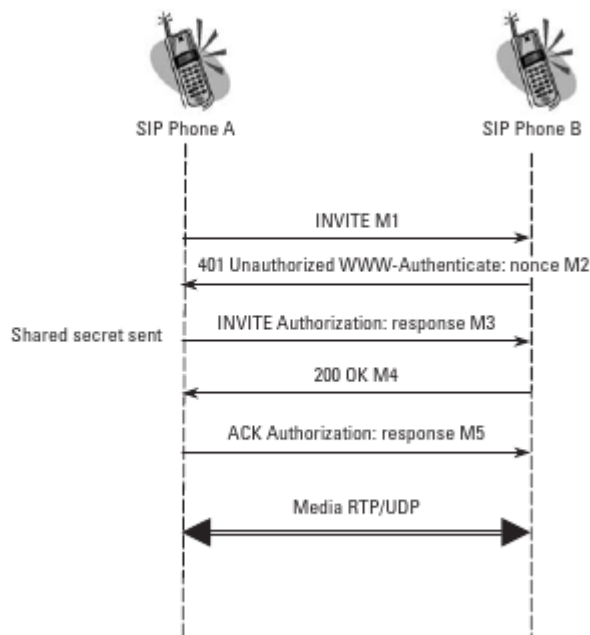
#### 6.8.2 Security and risk mitigation of SIP- signaling protocol security

Security of signaling protocol is important in VoIP technology since it handles a large number of sensitive information of users credential, bill resources and other important



data. In signaling, a call can be attacked in two ways. The attacks include impersonation which means that attacker imitates the actual user and eaves-dropping which means listening conversation secretly. Therefore, SIP signaling protocol contains certain security policies to protect signaling traffic. (Johston & Piscitello 2006, 145-159.) They are as follows:

- Basic Authentication. Base64 is an encoding method of username and password. Base64 uses 6 bits for each printable character. This Base64 can also use 65 possible symbols as well to encode the character. However, basic authentication cannot protect users credential if the attack is replied by the attacker.
- Digest Authentication. The digest authentication method is a borrowed authentication method by SIP. This digest authentication method is actually used in HTTP and known as RFC 2917 proxy Authorization since it is defined in RFC 2917. The Figure 16 depicts the SIP digest authentication:



**Figure 16.** SIP digest authentication between 2 users agent (Johston & Piscitello 2006, 147)

- Pretty Good Privacy (PGP). The PGP is an encryption scheme which originally was used in RFC 2543. The PGP encryption method was invented by Phil

Zimmermann to secure e-mail message. However the PGP was extended to secure signaling protocol also in later versions.

- S/MIME. This S/MIME uses Digital signature to all the parts of the messages to secure. RSA algorithm support is a prerequisite to use S/MIME for SIP UAs.
- Transport Layer Security (TLS). The TLS is a hop by hop basis security mechanism for SIP. A hop is a separate session of TLS. In a SIP signaling session, a message goes through three different hops. The hops for the signaling message are UA to proxy, proxy to proxy and finally proxy to UA. The negotiations are done over secured TLS tunnel independently.
- Secure SIP (SIPS). The SIPS is URI scheme to secure hop. This SIPS secured every hop in the signaling way from caller to receiver. ( Johston & Piscitello 2006, 145-159.)

The given security methods are widely used in securing SIP signaling messages. Since the VoIP system is relatively new technology, the VoIP protocols are still an easier field for attackers. Therefore, apart the above security policies, there are other security issues which also have to be taken care of during setting up an IP telephony using SIP signaling method. In general the security issues are media security, identity, PSTN gateway security and Spam and Spitt. ( Johston & Piscitello 2006, 163-237.)

## 7 DISCUSSIONS AND CONCLUSION

The chapter contains the results of the thesis, limitations and challenges of the research project, directions for further research and conclusion on my thesis work.

The objectives of my thesis were to analyze the architecture of VoIP and SIP mechanism and design the browser based web phone based on the analyzed result. During my research on exploring VoIP technology I acquired the core concept on the following components with what a whole VoIP system is built on:

- Voice Communication Server: Voice Communication Server is server which processes the call requests over the Internet protocol. Although there are many communication server exist by different vendor in VoIP industry, I recommend to use ‘Asterisk’ since it is a free and open source communication server which requires no external hardware to set up. Furthermore, Asterisk enables to modify its kernel and set up according to the clients need.
- Media Sever: The media server processes the VoIP RTP to be able to decode the DTMF, process VoiceXML scripts, record audio, recognize speech etc.
- Session Control Server: The session control server manages and provides session level functionalities. This session control server enables authentication, authorization and admission of VoIP calls. This role is also to support call routing and forwarding to another networks or service providers and maintain status about outgoing calls using necessary protocol.
- Gateway: A gateway is the entry and exit point of requested calls. When a call requests to forward to the PSTN network, it first request to the PSTN gateway to seek permission to the PSTN network. In other word, a gateway is a transition point of different network.

- Calling devices: Calling devices are tangible components of VoIP. Calling devices can be audio input devices (ex. computer), network card, IP phone and gateway devices (Ex. to PSTN).

After analyzing the fundamental architecture of VoIP, I analyzed the architecture and working mechanism of one of VoIP's protocol SIP. Since there are other protocols existing also, I had to differentiate and clarify why to use SIP and what its working mechanism and advantages.

SIP as a data transfer protocol differs many ways from the H.323 protocol. The H.323 protocol uses complex binary encoded ASN.1 message to establish session whereas SIP is a simple text based protocol. Therefore it is easier to implement than H.323 since it is possible to script it easily. This simple text based language makes SIP a powerful and flexible protocol to make the application development for the Internet and web technology than H.323. Furthermore, SIP has the instant message capability whereas H.323 is an exclusive signaling protocol. Moreover, SIP uses less CPU than H.323 in voice server for its smaller amount of message size.

Based on the findings, I designed the web phone which completed my research by producing the expected result from the research.

I designed the web phone by using UML exclusively based on the knowledge acquired on VoIP and SIP in the research findings in chapter six. Since UML itself is not a method, I had to validate the diagrams and their requirements with a standard method which defines UML. Therefore I used Meta-modeling Object Facility as standard which was designed by OMG to define the UML.

I also designed a horizontal prototype of the web phone interface which demonstrates the desired implementation from the idea.

Despite the fact that I accomplished my objectives of the thesis and successfully answered the research questions, the research work has its limitation. The field of telecommunication technology is complicated, expensive and broad. Therefore, I had to limit my scope of the research and work in a given boundary. Hence, further comprehensive research on VoIP and SIP security should be carried out during implementation of web phone. Moreover, the time frame for this research project was

tight and I had to carry out the project within limited resources which was difficult to manage. Additionally, exploring complicated technologies like VoIP and SIP was time consuming since they are relatively more complicated than other technical area. In spite of having all the limitations, I managed to accomplish my objectives which will help to do further research on the area and implement the idea in future.

Due to the limitations and challenges, working on hardware was beyond my research scope. Developing hardware has an important role to develop the telecommunication industry. I suggest doing comprehensive research on SIP and VoIP devices to develop the technology more flexible and available to deploy than the current available technologies. Furthermore, the initial complexity of setting up communication server with PSTN gateway may not attract the young students to enter the industry. Therefore, comprehensive research on the industry with the result of less complicated deployment facility may motivate the young students to enter the industry and become entrepreneur which has abundant of opportunity to become successful.

To conclude my thesis, I suggest that comprehensive research work on VoIP and SIP helped me to understand the fundamental concept to enter telecommunication industry. My idea to develop the web phone reached the state of visual structure which is more understandable and clear to develop further. Therefore, implementation of the design will have the platform to acquire necessary information on related technologies which may provide a new service to telecommunication industry in near future.

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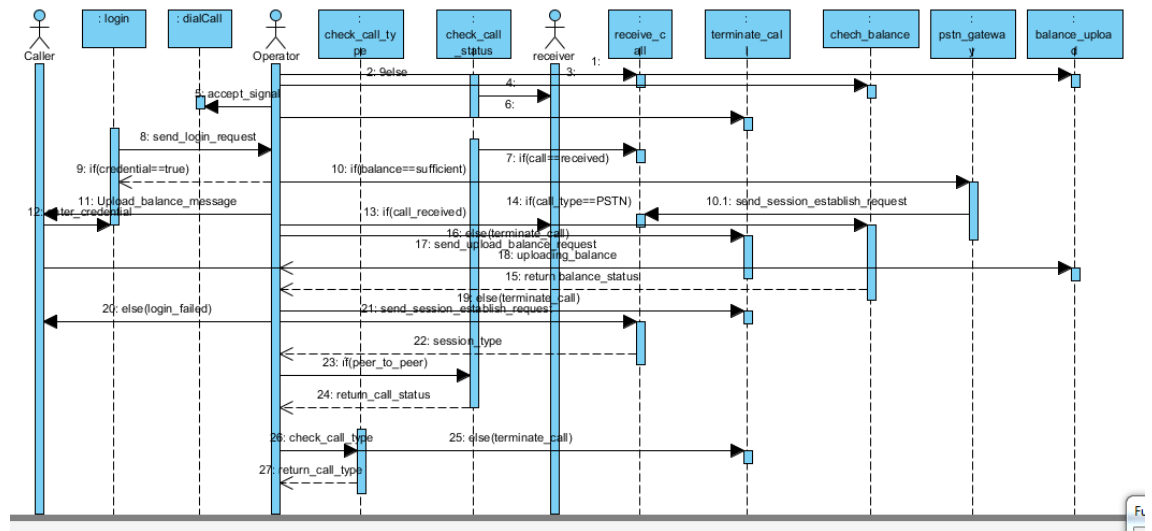
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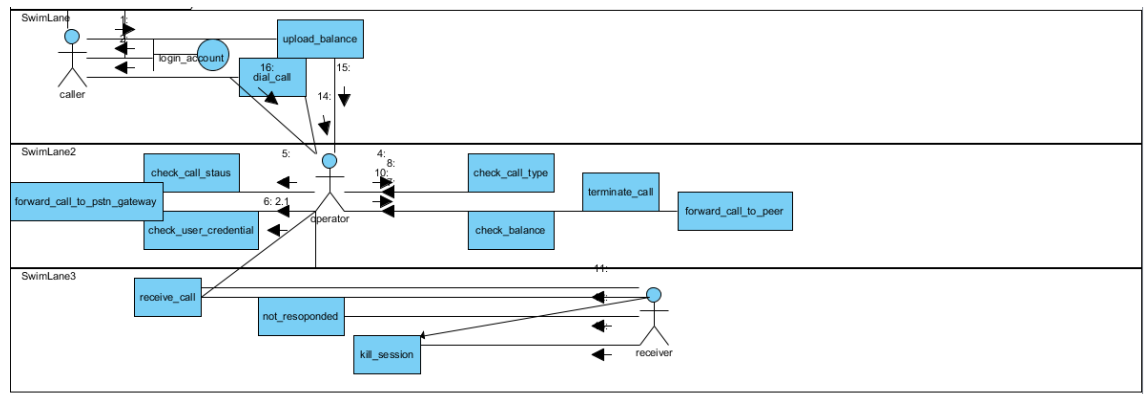
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**Figure 17.**web phone sequence diagram



**Figure 18.** Web phone communication diagram

**Table 3:** SIP code and message

<b>Code</b>	<b>Message</b>
100	Trying
180	Ringing
181	Call is being forwarded
182	Queued
183	Session progress
200	OK
202	Accepted
300	Multiple choices
301	Moved permanently
302	Temporarily
305	Use proxy
380	Alternative service
400	Bad request
401	Unauthorized
402	Payment required
403	Forbidden
404	Not found
405	Method not allowed
406	Not acceptable
407	Proxy authentication required
408	Request timeout
410	Gone
413	Request entity too large
414	Request URI too long
415	Unsupported media type
416	Unsupported URI scheme
420	Bad extension
421	Extension required
423	Interval too brief
480	Temporarily unavailable
481	Call/transaction does not exist
482	Loop detected
483	Too many hops
484	Address incomplete
485	Ambiguous
486	Busy here
487	Request terminated
488	Not acceptable here
489	Bad event
491	Request pending
493	Undecipherable
500	Server internal error
501	Not implemented
502	Bad gateway

<b>Code</b>	<b>Message</b>
503	Service unavailable
504	Server timeout
505	Version not supported
513	Message too large
600	Busy everywhere
603	Decline
604	Does not exist anywhere
606	Not acceptable