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SCHOOL OF INFORMATION TECHNOLOGY

DEPARTMENT OF SOFTWARE ENGINEERING

TITLE:

Research and Development of a mobile voice over IP application which is hosted by open source software running on inexpensive backend servers

Thesis presented in partial fulfilment of the requirements for the degree of Master of Information Technology at the Polytechnic of Namibia.

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March 2013

DECLARATION

I, **Veiko Mpareke Muronga**, hereby declare that the work contained in the mini–thesis, entitled: "**Research and Development of a mobile voice over IP application which is hosted by open source software running on inexpensive backend servers"**, is my own original work and that I have not previously in its entirety or in part submitted it at any University or other higher education institution for the award of a degree.

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ABSTRACT

Voice over IP has been driven by the convergence of voice and data on a single transport medium. This means cheaper service usage fees for the user if the offering is costed properly. But for a service provider there is still an increase in expenditure, solely related to spending on infrastructure such as call servers, phones and gateways. This study is aimed at analysing a different strategy of deploying VoIP, by starting at a lower base, meaning hosting the VoIP application on inexpensive hardware and open source Software. Secondly, the study looks at how to extend the service to mobile subscriber by introducing a mobile VoIP (mVoIP) application. The study's main goal is to introduce telecommunication services at the lowest cost possible while considering reduced complexity, optimisation, manageability and flexibility. Barriers to deploying VoIP such as reliability and quality of service are also addressed by the study.

The mVoIP infrastructure is made out of the following main components: Mobile VoIP app, Wireless Access Point and VoIP Server. Each main component's functionality and characteristic are carefully explored in order to understand its contribution to the solution and the study as a whole. The research also looks at other projects where similar approaches were used to deploy inexpensive VoIP services. Factors leading to acceptance or rejection of the technology have served as guidance to the study.

During the design, the research compares different open source VoIP server software's functionalities and features in order to make a selection of the software to use for the projects pilot network deployment. A development platform for the mobile application is also selected based on characteristics such as: the range of devices it runs on, the market share of the platform and the learning curve of the platform. The development of the application follows the six steps from the Eri Mobile's "Mobile Development Process": Concept Sketching, Research/Strategies, Wire-Framing, User Interface Design, Development and Testing.

The test scenarios for the research are all lab-based, but it is highlighted that implementing the solution on broadband, 3G and 4G networks can improve the service drastically in the areas of network coverage, Quality of Service and Mobility.

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Last but most importantly I would like to say thanks to the Mighty Father GOD for giving me the strength and endurance through my studies and beyond.

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LIST OF ACRONYMS

3G	--	3^{rd} Generation
4G	--	4^{th} Generation
AES	--	Advanced Encryption Standard
API	–	Application Programming Interface
B2BUAS	–	Back-to-back user agent server
BICC	–	Bearer Independent Call Control
BLF	--	Busy Lamp Field
BSS	--	Basic Service Set
BTS	–	Billing and Traffic System
CAPEX	--	Capital Expenditures
CDMA	--	Code Division Multiple Access
CLRF	--	Carriage return line feed
CMSS	–	Call Management Server Signalling
DSL	--	Digital Subscriber Line
DNS	–	Domain Name Server
DoS	--	Denial of Service
DTMF	--	Dual-Tone Multi-Frequency
EDGE	--	Enhanced Data for Global Evolution
EMEA	--	Europe, the Middle East and Africa
ESS	--	Extended Service Set
ETSI	--	European Telecommunications Standards Institute
EWSD	--	Electronic Worldwide Switch Digital
FHSS	--	Frequency Hoping Spread Spectrum

GPRS	--	General Packet Radio Service
GSM	--	Global System for Mobile Communications
HCL	--	Hardware Compatibility List
HTTP	–	Hypertext Transfer Protocol
IAX2	--	Inter Asterisk Exchange Version 2
IBC	–	IP Border Controller
ICS	--	IP Call Server
IETF	--	Internet Engineering Task Force
IMS	–	IP Multimedia Subsystem
IP	--	Internet Protocol
iOS	–	iPhones Operating System
ITU	--	International Telecommunication Union
IVR	--	Interactive Voice Response
JAVA ME	--	JAVA Micro Edition
LAN	--	Local Area Network
LGPL	--	Lesser General Public Licence
LTE	--	Long Term Evolution
MGC	–	Media Gateway Controller
MGCP	--	Media Gateway Control Protocol
MGW	–	Media Gateway
MRF	–	Media Resource Function
NAT	--	Network Address Translation
NCS	–	Networked-based Call Signalling
OPEX	--	Operational Expenditures

PABX	--	Private Automatic Branch Exchange
PBX	–	Private Branch Exchange
PC	--	Personal Computer
PCI	--	Peripheral Component Interconnect
POTS	–	Plain Old Telephone System
PRNG	–	Pseudo Random Number Generator
PSTN	–	Public Switched Telephone System
QoS	–	Quality of Service
RAID	--	Redundant Array of Independent Disks
RC4	--	Rivest Cipher 4
RTCP	--	Real Time Control Protocol
RTP	–	Real Time Protocol
SBC	--	Session Border Controller
SCCP	–	Skinny Client Control Protocol
SCSI	--	Small Computer System Interface
SDP	–	Session Description Protocol
SIP	–	Session Initiation Protocol
SIP-T	--	Session Initiation Protocol for Telephones
SS7	--	Signalling Systems No. 7
SLA	--	Service Level Agreement
ToS	--	Theft of Service
UA	–	User Agent
UAS	–	User Agent Server
UDP	--	User Datagram Protocol

UMTS	--	Universal Mobile Telecommunications System
URI	–	Uniform Resource Identifier
URL	–	Uniform Resource Locator
WACS	--	West African Cable System
WAP	--	Wireless Access Protocol
WAN	--	Wide Area Network
WEP	–	Wired Equivalent Privacy
WPA	--	Wi-Fi Protected Access
WPA2	--	Wi-Fi Protected Access 2

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1. INTRODUCTION

The following reading introduces the topic that this work is based on. The motivation for undertaking this research is described with the scope of the research and the objectives that needs to be executed in order to satisfy the scope. The chapter also defines all the main terms used in the dissertation. The chapter ends off by highlighting the structure of the rest of the dissertation.

1.1. DEFINITION OF THE PROBLEM

Telephony is an essential service of our day to day living. It is highly utilised by the public making it a daily commodity similar to services such as electricity or water. But lately, telecommunication (e.g. fixed and mobile) service providers have seen negative responses from the public. Affordability of the service has become a great concern to citizens, but due to lack of other options for telecommunication services, both the poor and rich must make use of the expensive data service while the service providers are reaping the benefits. In traditional methods such as PSTN and GSM, call tariffs are calculated according to predefined parameters: 1) The called party Zones (e.g. Local, National and International), 2) The called party technology (e.g. Fixed or Mobile), 3) The duration of the call in minutes, and 4) The time period (peak or off-peak). Different international/national destinations are also charged at different rates. This project therefore seeks to deploy a solution where users can enjoy telecommunication services whilst bypassing the traditional method based service provider. With the rapid deployment of mobile access networks combined with IP-based access to content, cheaper solutions to telecommunication should be feasible.

1.2. MOTIVATION

On the other side of communication, there is a vast growth in broadband and 3G technology where data is charged using flat/fixed fees which takes fewer of the tariff parameters in consideration. Service providers do not have different tariff tables for different destinations connected to the IP-based data network. Therefore utilizing a single network infrastructure to carry both voice and data reduces costs and introduces improved data management and communication efficiency within an organization. The convergence of voice and data gives rise to the term VoIP. By adopting VoIP, both customers and service providers will benefit.

Customers making use of VoIP can also benefit from uncapped offers provided by the service providers as seen in a comparison made based on the year 2011 packages shown it Table 1-1 from Telecom Namibia:

Package	Entry (192k)	256k	384k	512k
CPE (Ports)	1 x Data, 2 x Voice Ports	1 x Data, 2 x Voice Ports	4 x Data, 3 x Voice Ports, Wi-Fi	4 x Data, 3 x Voice Ports, Wi-Fi
Data Volume	Unlimited	Unlimited	Unlimited	Unlimited
Monthly Charge (1 year contract)	N$ 499	N$ 749	N$ 899	N$ 999
Monthly Charge (2 year contract)	N$ 399	N$ 599	N$ 699	N$ 849

Table 1-1 -- Broadband Home Packages Telecom Namibia

By looking at the table above, it’s evident that if customers had the VoIP option then they would benefit from the unlimited data usages, because they can make VoIP calls without any additional charges. On the other hand the operators would make a high return on their investment by offering low packages and attracting more customers.

By making use of Wi-Fi access that the broadband packages has to offer, the mVoIP services can be implemented without acquiring additional spectrum. Wi-Fi operates in unlicensed spectrum of 2.4 GHz and therefore legal issues with the regulators can be avoided. In some cases, Wi-Fi access points may be integrated into small cells that complement or supplement an operator’s data capability.

The goal of the project is to produce a cost-effective mobile voice over IP system. The main objective is to devise methods that will extend the VoIP service capabilities to mobile devices and using open source software and standard computers to host such a service at organisation premises. Therefore the reader must note that the terms VoIP and mVoIP are very closely interlinked because the term VoIP will be used when referring to the service on the backbone network and mVoIP will be used when referring to the service on the mobile phone.

According to Thompson (2011), Cross operating system (OS) platforms for mobile development as a whole is not yet completed perfectly and are provided on trial basis (Thompson, 2011). Therefore the researcher has made a few changes to this project and will not create a cross-platform mobile application through this project. Other factors to these changes are cost and time required to develop applications on cross development platform. But the success of this project will lay the foundation for easily extending the application to other platforms and also for easily adopting the service on to new fourth generation networks (e.g. LTE).

By designing and implementing an end-to-end mVoIP network prototype which consists of; low-cost VoIP servers and a free designed mVoIP application, the project will be able to lay the blueprint for implementing complete inexpensive mVoIP solutions for service providers with available documentations for future source code modifications or upgrades.

However, the question might arise: Why not just download existing mobile VoIP apps and integrate to existing VoIP systems? The mVoIP project introduces a sense of flexibility and ownership. It will be easier to make changes on the system if the code, the interfaces and the protocols are understood from the ground up. It is easy to administrate and make any new changes on the network, if it's fully owned by the developer or a group of users than paying for support on a system that make use of proprietary protocols that are only upgradeable by the vendor. Another aspect is the future opportunities that can be created by the knowledge gained from this project. Importantly the research introduces a system that addresses the cost aspect of telecommunication by hosting the service on off-the-shelf equipment.

With a complete wide mVoIP network, users can put their mobile's broadband capabilities to more use rather than just browsing the Internet. Telcos can make fast return on investments if they reduce their Capex by investing in low-cost systems and will also be able to attract more customers if they start bundling services and also reduce service costs. This will enable Telcos to sell more services to many subscribers for very low prices. A cheap mVoIP solution can also be extended to areas where telecommunication services cannot be afforded at all (e.g. rural areas which includes farms and schools).

1.3. SCOPE

The research primarily explores a cost effective approach to implement and provide mobile VoIP services. This involves swotting of existing commercialised or open source voice over IP systems and applications in order to learn from examples of previous implementations and also to gain understanding in which systems to use for the project; understand the underlying network of the VoIP network inclusive of protocols. This is to help understand the communication flow involved in this technology. Supplementary, the research seeks to understand the type of network elements required to implement an end-to-end VoIP network in order to understand what is necessary for an experimental deployment and what needs to be used in an actual wide network deployment.

The research further pursues to compare different development platforms in order to understand what the most suitable platform for the mVoIP application is. The scope of the research will then be narrowed into understanding the aspects of one development platform once the choice have been made then explore the approach of developing a mobile VoIP application. Additionally open source software will be implemented on inexpensive hardware to develop VoIP backend servers will host the service. The research therefor explores all the requirements and factors that have an impact on developing the VoIP network and mVoIP application and provides detail steps towards achieving the targeted network. The envisaged end result is an end-to-end mVoIP network producing mVoIP services.

1.4. PROJECT OBJECTIVES – DELIVERABLES

The main objective of this project is to develop a mobile voice over IP application for Wi-Fi enabled phones which targets the widest range of devices and promoting high CAPEX reductions on the VoIP network Implementation. For all parties in the VoIP business (e.g. customers and Service Providers) the end objective is affordability and simplicity that this project intends to deliver. In order to achieve this, the following milestones must be satisfied:

	Project Tasks	**Project Deliverables**
Sub-Objectives	Key Significant Milestones	Resulting Project Deliverables
1	Investigate related Software and design Specifications that will best suit the project requirements.	1) Software Specification. 2) Obtain all the project required Software
2	Investigate related Hardware Requirements based on the Software Specifications.	1) Hardware specification. 2) Acquire and build the required backend servers.
3	Analyse and design the high and low level network diagrams.	1) Network Specification 2) Acquire all required network elements. 3) Build the target network.
4	Study and chose the Mobile Application Development Platform	1) Install the best applicable Software Development Kit.

	that best fits the project objective.	2) Obtain the target device and kick-start development 3) Produce a mobile application
5	1) Integrate all network elements 2) Component Testing	1) Produce an mVoIP network 2) Produce a final design document
6	Carry out comprehensive testing for: 1) mVoIP application Services 2) Network Features (e.g. system monitoring) 3) Transport and Signalling 4) Performance Management	1. Produce functionality test document

1.5. DEFINITION OF TERMS

1.5.1. CORE NETWORK TERMINOLOGY

The following are some of the terms used in a VoIP system's control network. The control network involves the systems responsible for registering the user and controlling the call through its entirety:

Back to back user agent (B2BUA) – Robomax (2010) defines the back to back user agent as a SIP call controlling component. And unlike a SIP Proxy server, which only maintains a transaction state, the B2BUA maintains the complete call state and participates in all call requests which enables it to perform a number of functions (such as accurate call accounting, prepaid rating and billing, failover call routing and so on) which are not possible with a proxy server (Robomax, 2010).

Private Branch Exchange (PBX) – Wikipedia (2012) defines a PBX as a telephone exchange that serves a particular business or office. It is also referred to as a PABX (Private Automatic Branch Exchange) (Wikpedia, 2012).

Proxy Server – Wikipedia (2012) defines a proxy server as a server that acts as an intermediary for requests from clients seeking resources from other servers. A client connects to the proxy server, requesting some service, such as a file, connection, web page, or other resource available from a different server and the proxy server evaluates as a way to control its complexity (Wikipedia, Proxy server, 2012).

Session Border Controller (SBC) – Wikipedia (2012) defines a SBC as a device regularly deployed in VoIP networks exert control over the signalling and usually also the media streams involved in setting up, conducting and tearing down telephone calls or other interactive media communications (Wikipedia, Session Border Controller, 2012).

User Agent Server (UAS) – Javvin (2012) describes UAS as a term used in the Session Initiation Protocol (SIP) based VOIP system which is a server application that contacts the user when a SIP request is received and then returns a response on behalf of the user (Javvin, User Agent Server, 2012).

1.5.2. TRANSPORT NETWORK TERMS

The following are some of the terms used in a VoIP system's transport network. The transport network is responsible for the transportation of the VoIP packets.

Multiprotocol Label Switching (MPLS) – Wikipedia (2012) defines MPLS as a mechanism in high-performance telecommunications networks that directs data from one network data to the next based on short path labels rather than long network addresses, avoiding complex lookups in a routing table (Wikipedia, Multiprotocol Label Switching, 2012).

Nat Transversal – excITingip (2009) defines a NAT application (e.g. router, firewall) as an entity which is responsible for changing source addresses (Private IP addresses) on every outgoing packet from the internal computers in to the single public IP Address (excITingip, 2009).

Quality of Service – Dictionary.com (1998) defines QoS as the performance properties of a network service, possibly including throughput, transit delay and priority (Dictionary.com, 1998).

1.5.3. ACCESS NETWORK TERMS

The following are some of the terms used in a VoIP system's access network. The access network is responsible for providing users with access to the VoIP services.

3rd Generation (3G) – Wikipedia (2012) defines 3G as a generation of standards for mobile phones and mobile telecommunication services. Application services include wide-area wireless voice telephone, mobile Internet access, video calls and mobile T.V (Wikipedia, 3G, 2012) .

4th Generation (4G) – Wikipedia (2012) defines 4G as the fourth generation of cell phone mobile communications standards. It is a successor of the 3rd generations of standards. Applications include IP Telephony, gaming services, video conferencing and 3D Television (Wikipedia, 4G, 2012).

Asymmetric Digital Subscriber Line (ADSL) – Bradley (2012) defines ADSL as a digital subscriber line Internet service mostly used in homes which provides more bandwidth for download than for uploads. ADSL characteristics are: “High-speed” service (up to 6 Mbps), an “always on” combination of voice and data support and availability and performance that is only limited by physical distance (Bradley, 2012).

Code Division Multiple Access (CDMA) – Wikipedia (2012) defines CDMA as a channel access method used by radio communication technologies. CDMA employs spread-spectrum technology and a special coding scheme (where each transmitter is assigned a code) to allow multiple users to be multiplexed over the same physical channel (Wikipedia, CDMA, 2012).

Enhanced Data for Global Evolution (EDGE) – The GSM Arena (2012) defines EDGE as a data system used on top of GSM networks, which provides nearly three times faster speeds than GPRS. The theoretical maximum speed is 473 kbps for 8 timeslots but it is typically limited to 135 kbps in order to conserve spectrum resources (GSMArena, EDGE, 2012).

General Packet Radio System (GPRS) – The GSM Arena (2012) defines GPRS as a packet-switching technology that enables data transfers through cellular networks which is used for mobile Internet, MMS and other data communications. In theory the speed limit of GPRS is 115 kbps, but in most networks it is around 35 kbps (GSMArena, GPRS, 2012).

Global System for Mobile Communications (GSM) – Wikipedia (2012) defines GSM as a standard set developed by the European Telecommunications Standards Institute (ETSI) to describe protocols for second generation (2G) digital cellular networks used by mobile phones. GSM originally described a digital, circuit switched network optimised for full duplex voice telephone which was later expanded to data communications such as GPRS (General Packet Radio Service) and EDGE (Enhanced data rates for GSM evolution) (Wikipedia, GSM, 2012).

Universal Mobile Telecommunications System (UMTS) -- 3GPP (2012) defines UMTS as an umbrella term for the third generation radio technologies developed within 3GPP. The specification provides for Frequency Division Duplex (FDD) and Time Division Duplex (TDD) variants and it also allows for the re-use of bands currently assigned to 2G services (3GPP, 2012).

Worldwide Interoperability for Microwave Access (WiMAX) – Wikipedia (2012) defines WiMAX as a wireless communications standard designed to provide 30 to 40 megabit-per-second data rates, with the 2011 update providing up to 1 Gbit/s for fixed stations. The name “WiMAX” was created by the WiMAX Forum, which was formed in June 2001 to promote conformity and interoperability of the standard (Wikipedia, WiMAX, 2012).

1.5.4. SECURITY TERMS

The following are some of the security terms used in a VoIP network.

Advanced Encryption Standard (AES) – Rose (2011) defines AES as an encryption algorithm for securing sensitive but unclassified material by U.S. Government Agencies which would eventually become the de facto encryption standard for commercial transactions in the private sector. The specification makes use of a symmetric algorithm using block encryption of 128 bits in size, supporting key sizes of 128,192 and 256 bits, as a minimum (Rose, AES, 2011).

Block Cipher – Rose (2006) defines the term block cipher as a method of encrypting text (to produce ciphertext) in which a cryptographic key and algorithm are applied to a block of data at once as a group rather than to one bit at a time (Rose, block cipher, 2006).

Stream Cipher – Rose (2005) defines stream cipher as a method of encrypting text in which a cryptographic key and algorithm are applied to each binary digit in a data stream, one bit at a time (Rose, Stream Cipher, 2005).

Wi-Fi Protected Access (WPA) – Webopedia (2012) defines WPA2 as a Wi-Fi standard designed to improve upon the security features of WEP. The standard has improved data encryption through the temporal key integrity protocol (TKIP) which scrambles the keys using a hashing algorithm and by adding an integrity checking feature ensures that the keys haven't been tempered with (webopedia, 2012).

Wi-Fi Protected Access 2 (WPA2) – Wikipedia (2012) defines WPA2 as a replacement of WPA, which introduces CCMP (Counter Cipher Mode with Block Chaining Message Authentication Code Protocol), an enhanced data cryptographic encapsulation mechanism designed for data confidentiality and based upon the counter mode with CBC-MAC (CCM) of the AES standard (Wikipedia, Wi-Fi Protected Access, 2012).

Wired Equivalent Privacy (WEP) – Webopedia (2012) defines WEP as a security protocol for wireless local area networks (WLANs). WEP aims to provide security by encrypting data over radio waves so that it is protected as it is transmitted from one end point to the another (Webopedia, WEP, 2012).

1.5.5. *PROTOCOLS*

The following are definitions of some of the protocols used in a VoIP system's network.

Bearer Independent Call Control (BICC) – ITU-T (2000) species BICC in the recommendation Q.1901 as a signalling protocol based on narrowband ISDN user part (ISUP), used for supporting narrowband Integrated Service Digital Network (ISDN) service independent of the bearer technology and signalling message transport technology used (ITU-T-Q.1901, 2000).

Distance Vector Protocol – Cisco (2010) defines Distance Vector Protocols as routing protocols that uses distance and direction (vector) to find paths to destinations. Routers running these protocols learn who their neighbours are by listening for routing broadcasts on their interfaces and in turn pass routing table updates to their immediate neighbours in all directions (Cisco, Introduction to Distance Vector Routing Protocol , 2010).

Direct Sequence Spread Spectrum (DSSS) – Is defined by the Telecom dictionary (Telecom ABC) (2005) as a spread spectrum technique whereby the original data signal is multiplied with a pseudo random noise spreading code. The spreading code has a higher chip rate which results in a wideband time continuous scrambled signal (ABC, 2005).

Frequency Hoping Spread Spectrum (FHSS) – Wikipedia (2012) defines FHSS as a method of transmitting radio signals by rapidly switching a carrier among many frequency channels, using a pseudorandom sequence known to both transmitter and receiver (Wikipedia, Frequency-Hoping Spread Spectrum, 2012).

H.248 – Wikipedia (2012) defines H.248 or Megaco or Gateway Control protocol is a recommendation from ITU Telecommunication Standardisation Sector (ITU-T) which defines protocols that are used between elements of a physically decomposed multimedia gateway (Wikipedia, H.248, 2012).

H.323 – Webopedia (2012) defines H.323 as an international Telecommunications Union (ITU) standard that provides specification for computers, equipment and services for multimedia communication over packet based networks that defines how real-time audio, video and data information is transmitted (Webopedia, H.323, 2012).

Link State Routing Protocols – Cisco (2012) defines Link State Routing Protocols as routing protocols which calculates their network routes by building a complete map of the entire network area and then calculating the best path from this map of all the interconnected networks (Cisco, Link State Routing Protocols, 2012).

Orthogonal Frequency Division Multiplexing (OFDM) – Franzel (2009) defines OFDM as a broadband multicarrier modulation method that offers superior performance and benefits over older, more traditional single-carrier modulation methods because it is a better fit with today's high-speed data requirements and operation in the UHF and microwave spectrum (Franzel, 2009).

Session Description Protocol (SDP) – Wikipedia (2012) defines SDP as a format for describing multimedia communication sessions for the purposes of session announcement, session invitation, and parameter negotiation. SDP does not deliver media itself but is used for negotiation between end points and media type, format and all associated properties (Wikipedia, Session Description Protocol, 2012).

Session Initiation Protocol (SIP) – Henning (2008) defines SIP as a signalling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. SIP was developed within the IETF MMUSIC (Multiparty Multimedia Session Control) Working Group, with work proceeding since September 1999 in the IETF SIP working group. (Henning, 2008).

Signalling System Number 7 (SS7) – Hewett (2004) defines the SS7 suite as a set of telephony signalling protocols which are used to set up most of the world's public switched telephone network telephone calls. The main purpose is to set up and tear down telephone calls but it is also responsible for functions such as number translation, local number portability, prepaid billing mechanisms, short message service (SMS) and other market services (Hewett J, 2004).

Voice over Internet Protocol (VoIP) – Wikipedia (2012) defines VoIP as the communication protocols, technologies, methodologies and transmission techniques involved in the delivering of voice over IP networks (Wikipedia, Voice Over IP, 2012) .

Voice over Wi-Fi (VoWi-Fi) – Wikipedia (2012) defines the term VoWi-Fi as the use of wireless broadband network according to the IEEE 802.11 standards for the purpose of vocal conversation. In essence, it's VoIP over a Wi-Fi network (Wikipedia, Voice over WLAN, 2012).

1.5.6. STANDARDS

The following are some of the definition of the standards used in VoIP systems.

Application Programming Interface – 3Scale (2011) defines An API as a particular set of rules and specifications that a software program can follow to access and make use of the services and resources provided by another particular software program that implements that API (3Scale, 2011).

IEEE 802.11 – Wikipedia (2012) defines IEEE 802.11 as a family of specifications developed by the IEEE for wireless LAN technology. 802.11 applies to wireless LANs and provides 1 or 2 Mbps transmission in the 2.4GHz band using either frequency hopping spread spectrum (FHSS) or direct sequence spread spectrum (Wikipedia, IEEE 802.11, 2012).

Long Term Evolution (LTE) – Wikipedia (2012) defines the term LTE as a standard for wireless communication of high-speed data for mobile phones and data terminals. It is based on in the GSM/EDGE and UMTS/HSPA network technologies, increasing the capacity and speed using a different radio interface together with core network improvements (Wikipedia, LTE, 2012) .

1.5.7. OTHER TERMS

Beacon Frame – Brenner (1997) defines Beacon frame as a periodic frame sent by the Access Point with synchronisation information (Brenner, 1997).

Broadband – Rose (2007) defines Broadband as telecommunication in which a wide band of frequencies is available to transmit information. And due to the availability of a wide band of frequencies, information can be multiplexed and sent on many different frequencies or channels within the band concurrently, allowing more information to be transmitted in a given amount of time (Rose, Broadband, 2007).

Failover Cluster – Lunik (2009) defines failover cluster as a group of independent computers that work together to increase the availability of applications and services. The clustered servers (called nodes) are connected by physical cables and by software. Therefore if one of the cluster nodes fails, another node begins to provide service, a process known as failover (Lunik, 2009).

1.6. HYPOTHESIS AND RESEARCH QUESTIONS TO BE INVESTIGATED

If the mVoIP solution is implemented then it will be able to provide the users with similar services as those provided by the traditional voice network at less cost and effort. With reference to the assumptions above, the following questions should be addressed:

- To which extent can the mVoIP System provide similar services as the traditional voice network?
    - Does the solution cater for the following essential functionalities of a Voice network:
        1. User Security
        2. System Security
        3. High availability
        4. System Management
        5. User Management
        6. Call Detailed Record
        7. Visual and Audible Alarms
        8. Network Monitoring
    - What are the CAPEX and OPEX components of the solution?
    - Is implementation cost drastically reduced by the solution?

- How will the user experience be improved?
    - Will there be any improvement in the QoS?
    - How will handover be realised?
    - Can roaming be realised by the mVoIP solution?
    - Are there any new features for the users?
    - Will the mVoIP application be easy to use?
- What criteria's will be used to choose a development platform?
- How about the regulatory framework, how will spectrum licensing be handled?

The research seeks to address these questions and therefore deep insight has to be gained on all the topics covered.

1.7. THESIS STRUCTURE

This document is organised to follow the life cycle of research paper. It begins with the idea that launches a project and ends with the termination of the project. Before undertaking the journey of reading this document, it is better to first understand the navigation routes of the document. Following this introductory chapter, **Chapter 2** is a literature review where the research analyses text written by other authors to consider the critical points of current knowledge including substantive findings as well as theoretical and methodological contributions to the topic of mobile VoIP. The chapter reviews topics of (mobile) VoIP implementations, systems and applications. **Chapter 3** is the methodology chapter which starts with an evaluation of existing voice system to understand their advantages and disadvantages. The chapter then looks at the process of designing the mVoIP system through to the implementation of the system. **Chapter 4** looks at the development of the mVoIP application. It analyses mobile development process in order to understand the best model to adapt. The chapter also deals with testing of the mVoIP application. **Chapter 5** is the findings and discussions chapter. The chapter revises the findings of the research and answers the research questions. **Chapter 6** is the conclusion and recommendations. The chapter also recommends future research works.

1.8. SUMMARY

This chapter introduced the subject of mobile VoIP and discusses the motivation factors. It defines the problem that needs to be addressed and provides motivation of why this research project should be undertaken. The chapter defines the scope of the project and also lists the project deliverables. This chapter also defines all the main terms used in this paper.

2. LITERATURE REVIEW

The chapter review literature which provides information on the research subject. It reviews the requirements, components, protocols and networks involved in the VoIP system to help guide the research. The chapter also looks at the adoption factors which outline how similar implementations have survived. The chapter ends off by describing the existing open source systems.

2.1. INTRODUCTION

The following sections aim to review the critical points of current knowledge including substantive technology findings as well as theoretical and methodological contributions to the topic of (Mobile) Voice over IP. VoIP is the groundwork of this research and therefore a comprehensive review on the topic is crucial. The literature review was conducted to ascertain the researcher of the intended research path and also to provide more information on the research field. The review looks at relevant topics and in turn conducts an in-depth evaluation of the topics which may guide the research in its particular approach.

2.2. VOICE OVER IP

Technological innovation is launching itself upon us once again. This time, it is coming in the form of improving the way we communicate, bringing with it new capabilities that change the meaning of the phrase “telephone call”. VoIP which stands for ***voice over Internet protocol*** basically means voice transmitted over a digital network. What is necessary for VoIP is not necessarily the Internet but the use of the same protocol suite and the same packet switched connection that the Internet uses. Thus voice over Internet protocol means voice that travels by way of the same protocols used on the Internet.

VoIP is the foundation of this research and therefore understanding the VoIP terminology and what differentiates it from the other technologies used in telecommunications will help set the technical scope limits of this research. The insightful understanding of VoIP will also help guide the researcher in knowing what the requirements of the VoIP network are and which areas of the technology are of most important and relevant to this study.

With reference to Figure 2.1, JDSU[1] (2010) also explains that VoIP makes use of the Internet Protocol as its transport layer. The same report further explains that both the User Datagram Protocol (UDP) and the Transmission control protocol (TCP) are utilised by VoIP where UDP delivers the Voice protocols (such as RTP) while TCP is responsible for the signalling protocols (such as H323 or SIP). UDP is used for voice hence it does not support retransmission of packets which can results into a conversation having delays (JDSU, voipterm, 2010).

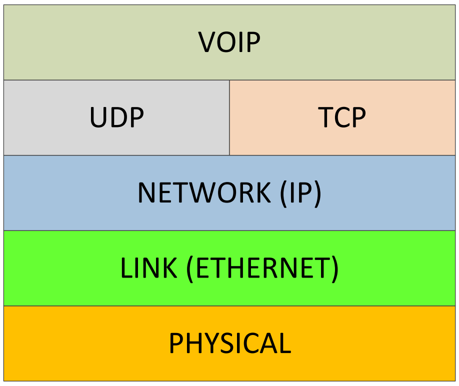

Figure 2.1: VoIP Protocol Stack (JDSU, Voipterm, 2010)

Kelly (2005) recalls that before the introduction of digital networking, telecommunications services were based on the old POTS (Plain Old Telephone System) which was hosted by the Public switched Telephone Network (PSTN). Kelly further states that without a replacement service for the POTS system, service costs for POTS were inflated much higher than need be. But with the introduction of VoIP as an alternative for POTS, voice service costs can be made more affordable or can totally be removed (Kelly, 2005).

The researcher also agrees that one of the key drivers of converging voice and data is the cost saving factor. With the deployment of the right VoIP solution, money can be saved in many areas of the telecommunications service: for example the implementation of the system, the operation and maintenance of the system and the ongoing usage charges.

Shoretel (2007) also complements on the benefits of VoIP by stating that "VoIP can also help an organisation gain a competitive advantage, boost employee productivity and enhance customer service" (Shoretel, 2007). Kelly (2005) also supports the advantages of VoIP by stating that by converging voice and data using the underlying IP network, organisations can also connect their different dispersed locations (including mobile workers) onto a single converged communications

[1] An international corporation that specialises in fibre networks.

network with enhanced services and features, which can also lead to enhanced productivity if the communications barriers are eliminated in an organisation (Kelly, 2005).

2.2.1. Requirements of a VoIP Network

There are several requirements that need to be addressed in order to provide a toll-quality, VoIP Network. These include:

- **Service set to be offered, and the types of end user terminal supported** – Paul (2003) states that this could range from basic call services (e.g. simple user-to-user voice services) to offering possibly value added services (e.g. multi-party conference). He also adds that another important part of service design is the choice of end user terminals that are supported by the service offering, possibly choices include (Paul D. , 2003):
    - POTS “black phones”;
    - IP Phones;
    - PBXs and key systems;
    - PC soft-clients (including web-based applications).

- **Choice of signalling protocols** – Paul (2003) states that the type of signalling to be used must be applicable to a VoIP solution. They include (Paul D. , 2003):
    - Device controlled protocols such as H.248 (Megaco), MGCP, NCS, etc.
    - Access services signalling protocols such as SIP, H.323, etc.
    - Network service signalling protocols such as SIP, SIP-T, BICC, CMSS, etc.

- **Security** – Paul (2003) explains that the Public Switched Telephone Network is immune to user attacks due to the type of signalling method (out-of band SS7) used. He further states that similar to any IP network a VoIP network is much more susceptible to security attacks and therefore security issues such as the ones below must be addressed prior to implementation (Paul D. , 2003):
    - Denial of Service – Writer (2000) explains that a DoS attack floods a system such as a web server by saturating it with external communication requests, such that it cannot attend to genuine users requests and therefore becomes unavailable to its intended users (Writer, 2000).

- Theft of Service – Wikipedia (2012) explains that this attack is executed when the attacker makes use of services illegally while bypassing the service provider and not paying for the services (Wikipedia, Theft of Services, 2012).
- Invasion of Privacy – Paul Drew (2003) explains that this attack is committed when the attacker or third party secretly listens in to someone else telephonic conversation, without the calling or called party's consent. (Paul Drew, 2003).

- **Quality of Service (QoS)** – Lokua (2011) describes QoS as the methods introduced into a network to manage the requirements of a connection such as delay, delay variation, echo and cross-talk on a network (Lokua, 2011).

- **Reliability/Availability** – Paul (2003) explains that in order for VoIP systems to perform and maintain its functions in hostile conditions, the systems must be implemented in redundant and load sharing configurations. Paul further states that a VoIP network should maintain it integrity in the presence of impact or fault and to achieve this, the following functionalities need to be in place (Paul Drew, 2003):
    - Redundant Hardware;
    - Redundant Network Connections;
    - Hot-Swap Capabilities

Reliability/Availability is also available on Windows systems and according to Microsoft (2012), it will be easier to implement high availability on systems running windows server then on the non-server systems. This, Microsoft (2012) explains is due to the fact that Windows server operating systems software (e.g. Windows Server 2003, Windows Server 2008) comes pre-equipped with a copy of the cluster administration software which can make the implementation of a VoIP system on Windows backend systems less complicated. Some of the requirements for a cluster server installation on Windows systems are as below (microsoft, 2012):

- A boot disk with Windows 2000 - 2012 Advanced Server or Windows 2000 – 2012 Datacentre Server installed. The boot disk cannot be on the shared storage bus described below.
- A separate PCI storage host adapter (SCSI or Fibre Channel) for the shared disks.
- Two PCI network adapters on each machine in the cluster.
- An HCL-approved external disk storage unit that connects to all computers. This will be used as the clustered disk. A redundant array of independent disks (RAID) is recommended.

- Storage cables to attach the shared storage device to all computers.
- All hardware should be identical, slot for slot, card for card, for all nodes. This will make configuration easier and eliminate potential compatibility problems.

- **Firewall and Nat Transversal** – Paul (2003) explains that in a VoIP network where the customer premises equipment are also IP based it is best to introduce security functions such as firewall and Network Address Translation (NAT) in order to separate the service provider's private internal network from the users or public network (Paul Drew, 2003).

- **OSS Support** – Paul (2003) explains that similar to a PSTN network Operational Support Systems (OSS) are vital to a VoIP network and should provide functions such as (Paul D. , 2003):

  - Operations and Maintenance;
  - Alarms;
  - Performance Monitoring;
  - Fault Isolation;

- **Bandwidth Utilization** – For this requirement, Paul (2003) explains that the bandwidth utilised for transporting voice should be considered and reduced wherever necessary by implementing compression methods for the protocols involved. He adds that with real-time protocol (RTP) as the transport mechanism for VoIP, a voice payload size gets increased due to the addition of the headers. This in turn Paul (2003) explains can reduce the number of voice channels available especially for the lower-bandwidth access systems such as DSL or Cable with higher header overhead (Paul Drew, 2003).

2.2.2. Protocols Related To VoIP

VoIP is best understood as a collection of the protocols that make up its mechanisms. Three categories of protocols are relevant to VoIP:

- Signalling
- Routing
- Transport

Ohrtman (2004) clarifies that Signalling Protocols (H.323 and SIP) are responsible for functions that facilitate calls such as: Call control (setting up and tearing down calls), Identification of a connection state and Provisioning of call information. The same report adds that Gateway control protocols such as MGCP and MEGACO are also signalling protocols that control calls via external elements called media and signalling gateways. Routing protocols such as RIP (Routing Information Protocol) are used to determine which path the traffic should follow, while transport protocols such as RTP (Real-Time Protocols) defines the standardized packet for delivering the audio and video of the IP network (Ohrtman F. , Voice over 802.11, 2004).

The VoIP Protocol stack can be seen in Figure 2.2 which is also called a multimedia protocol stack due to the fact that these protocols are responsible for audio and video.

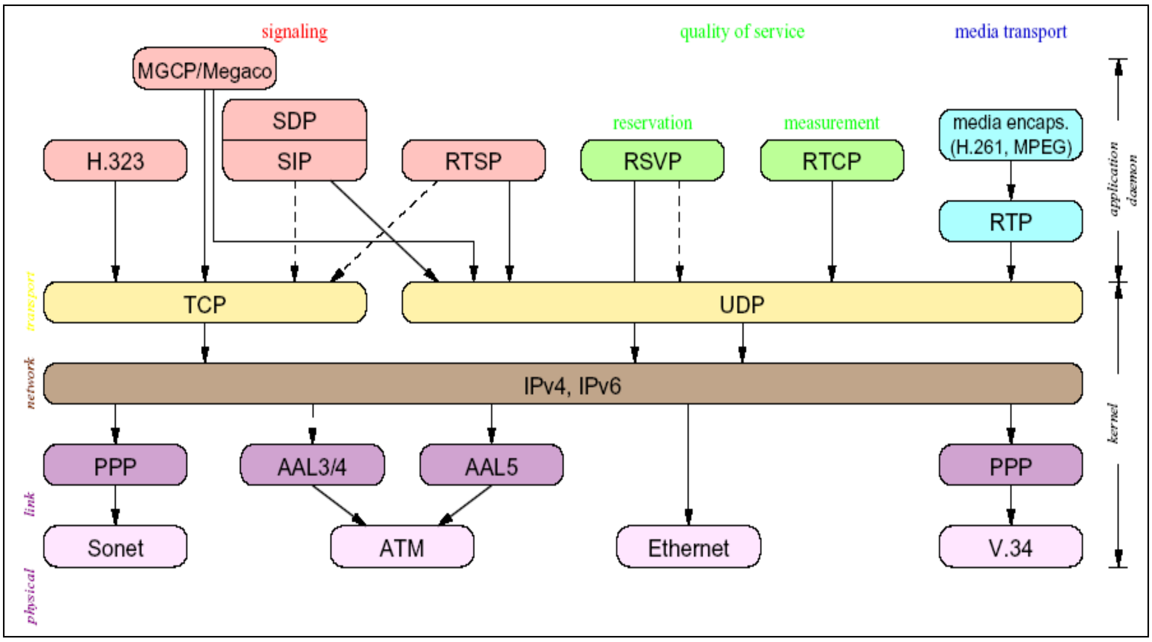

Figure 2.2: Multimedia Protocol Stack (Schulzrinne, Session Initiation Protocol, 2001)

2.2.2.1. SIGNALLING PROTOCOLS

A signalling protocol is a type of protocol used to identify signalling encapsulation. Signalling is used to identify the state of connection between telephones or VoIP terminals. When the calling and called parties agree on how to communicate and the signalling criteria is established, the media stream over which the packetized voice conversation will flow is established. Signalling establishes the virtual circuit over the network for that media stream. Signalling is independent of the media flow. It determines the type of media to be used in a call. Currently, two types of signalling are popular in VoIP: H323 and SIP (Douskalis, 2000).

H.323 is the International Telecommunication Union (ITU-T) recommendation for packet-based multimedia communication. H.323 was developed before the emergence of VoIP. Because it was not specifically designed for VoIP, it has faced a good deal of competition from a competing protocol, the Session Initiation Protocol (SIP), which was specifically designed for VoIP (Frank, 2004).

2.2.2.1.1. H.323

As described in the ITU-T recommendation H.323 (2009), H.323 is a protocol that provides the foundation for audio, video and data communications across IP-based networks. The recommendation also describes the characteristics of H.323 based systems such as gateways, terminals and multipoint controllers units used in multimedia communications over packet-based networks (such as LANs) that do not provide a guaranteed Quality of Service. The document further explains that the H.323 protocol includes parts of H.225.0 – RAS, Q.931, H.245 RTP/RTCP, audio codecs (such as G.711, G.723.1, G.728, etc) and video codecs (such as H.261 and H.263) (H.323, 2009).

Figure 2.3 depicts the H.323 protocol stack.

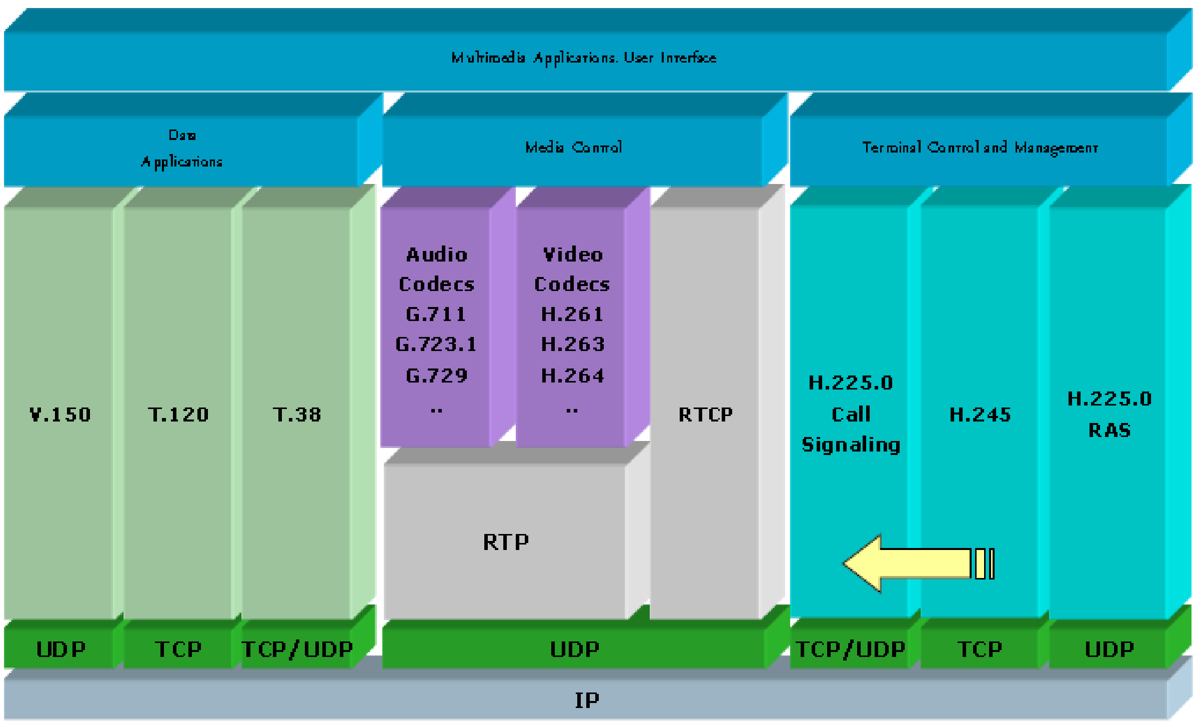

Figure 2.3: H.323 Protocol Stack (Paul E. , 2007)

2.2.2.1.2. SESSION INITIATION PROTOCOL

RFC 3261 defines the SIP protocol as an application layer control protocol or signalling protocol used for controlling communication sessions such as voice and video calls over the packet-based networks. The document further explains that SIP protocols can be used for control functions such as creating, modifying and tearing down two party or multi party sessions independently of the transport protocol, meaning that it can run on Transmission Control Protocol (TCP), User Datagram Protocol (UDP) or Stream Control Transmission Protocol (SCTP). SIP supports session management capabilities such as adding or removing participants to/from an already existing session (3261, 2002).

Frank (2004) explains that SIP is a text based protocol which makes use of many elements of the Hypertext Transfer Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP) and therefore the programs that are designed for analysing HTTP can be adapted easily for use with SIP. With reference to Figure 2.4 and Figure 2.5, Frank explains further that SIP addresses known as SIP uniform resource locators (URLs) are made out of the same structures as web addresses and therefore a web address can be the equivalent of a telephone number in the SIP environment. And since proxy servers also form part of the SIP network, mobility is realised because the user can make calls using his SIP number from anywhere in the world, similar to accessing one’s email address (Frank, 2004).

```
Message Header
Via: SIP/2.0/UDP 192.168.0.100:5060;rport;branch=z9hG4bK6464641000000007343c52679000020a600000e45
Content-Length: 0
Call-ID: 911D32E5-EEDF-4572-B0B2-61B294636E88@192.168.0.100
CSeq: 1 ACK
From: "Prueba"<sip:20000@miasterisk.com>;tag=8922404614682
Max-Forwards: 70
Route: <sip:20001@192.168.0.1>
To: <sip:20001@miasterisk.com>;tag=as0a27b928
User-Agent: SJphone/1.60.289a (SJ Labs)
Contact: <sip:20100@192.168.0.100:5060>;expires=3600
```

Figure 2.4: SIP Header (Think, 2012)

```
GET /tutorials/other/top-20-mysql-best-practices/ HTTP/1.1
Host: net.tutsplus.com
User-Agent: Mozilla/5.0 (Windows; U; Windows NT 6.1; en-
US; rv:1.9.1.5) Gecko/20091102 Firefox/3.5.5 (.NET CLR 3.5.30729)
Accept: text/html,application/xhtml+xml,application/xml;q=0.9,*/*;q=0.8
Accept-Language: en-us,en;q=0.5
Accept-Encoding: gzip,deflate
Accept-Charset: ISO-8859-1,utf-8;q=0.7,*;q=0.7
Keep-Alive: 300
Connection: keep-alive
Cookie: PHPSESSID=r2t5uvjq435r4q7ib3vtdjq120
Pragma: no-cache
Cache-Control: no-cache
```

Figure 2.5: HTTP Header (Guzel, 2009)

Frank (2004) explains that even though SIP borrows a lot of protocol concepts from internet protocols such as SMTP and HTTP there are still many differences between this protocols because HTTP is used integrate and provides access to content (text, audio, and video) on internet Server (client server architecture), while SIP is used to set up calls between peers (peer-to-peer architecture) (Frank, 2004).

The signalling protocols discussed in the section above are not vertically integrated communication system but should rather be used with other IETF protocols to build a complete multimedia architecture. These architectures include protocols such as:

- Media Gateway Control Protocols
- Transport Protocols
- Routing Protocols

2.2.2.2. MEDIA GATEWAY CONTROL PROTOCOLS

As Radvision (2012) explains, the need for Media Gateway Control Protocols aroused from the requirement to have old telecommunication system such as PSTN interwork with the newer IP based telecommunication systems such as VoIP and also to enable access systems independency. The same document further explains that Media Gateway Control Protocols are responsible for the control of media gateways which provides the bridge for media streams as they travel between IP and PSTN networks. Radvision further explains that in signalling protocols such as SIP, the entities involved are in a peer-to-peer configuration while in Media Gateway Control Protocol configuration the entities are in a Master/Slave Configuration where the Media Gateway Controller is the Master and the Media Gateway is the slave. Radvision further elaborates that the difference between signalling protocols and media gateways control protocols is that, signalling protocols establish and manage calls while media gateway control protocols define and creates media paths between IP and other transport protocols or networks. The well-known media gateways control protocols are MGCP and Megaco/H.248 (Radvision, Implementing Media Gateways Control Protocols, 2012).

2.2.2.2.1. MGCP

IETF defines MGCP in the RFC 3435 document as a protocol used by external call control elements called gateway controllers or call agents to control media gateways which are network elements whose sole function is to convert the media carried on telephone circuits into data carried over packet networks and vice versa. The document describes the different types of gateways used in VoIP network as follows (IETF, RFC 3435 MGCP, 2003):

- Trunking Gateways – A media conversion device between a TDM and a Voice over IP network.
- Voice over ATM Gateways – Operates in a similarly to Trunking Gateways but they convert media between ATM and IP networks.
- Residential Gateways – Are VoIP gateways that reside at customer premises and can provide an RJ11 interface for a normal POTS phone.
- Access Gateways – are gateways that provide traditional PSTN interfaces (e.g. analogue or E1's) to a VoIP network.
- Business Gateways – Provides traditional analogue or ISDN PBX interfaces to a VoIP network.

2.2.2.2.2. MEGACO/H.248

Javvin (2012) defines Megaco/H.248 as a media gateway control protocol which is a joint effort between IETF and ITU-T, of which the Megaco is the IETF naming convention while H.248 is an ITU-T naming. Javvin (2012) explains that similar to Mgcp, Megaco controls media gateways which are responsible for media conversion between different transport protocols. From a protocol architecture point of view, Megaco is very similar to Mgcp but Megaco supports a wide range of networks including ATM (Javvin, Megaco/H.248, 2012).

2.2.2.3. TRANSPORT PROTOCOLS

Jonathan (2006) defines the two transportation protocols residing at the transport layer of the IP protocol stack: UDP (User Datagram Protocol) which is connectionless and unreliable protocol and TCP (Transport Control Protocol) which is a reliable and a connection-oriented protocol. Jonathan further explains that due to the fact that voice is a real-time application, UDP/IP became the preferred protocol for voice transport. UDP could not provide all the information (e.g. timestamp and payload type (PT)) required in a voice packet as shown in Figure 2.6. The IETF (Internet Engineering Task Force) implemented RTP (Real-Time Protocol) as a transport protocol on top of UDP for all time sensitive application as shown Figure 2.7 (Jonathan Davidson, 2006).

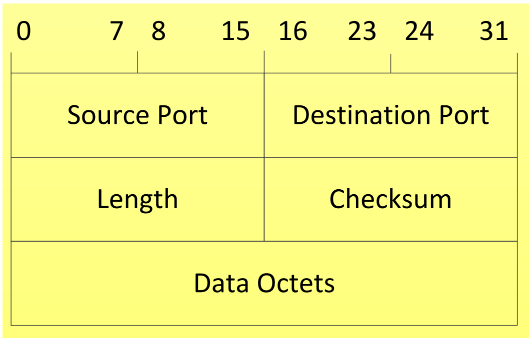

Figure 2.6: User Datagram Header Format (Postel, 1980)

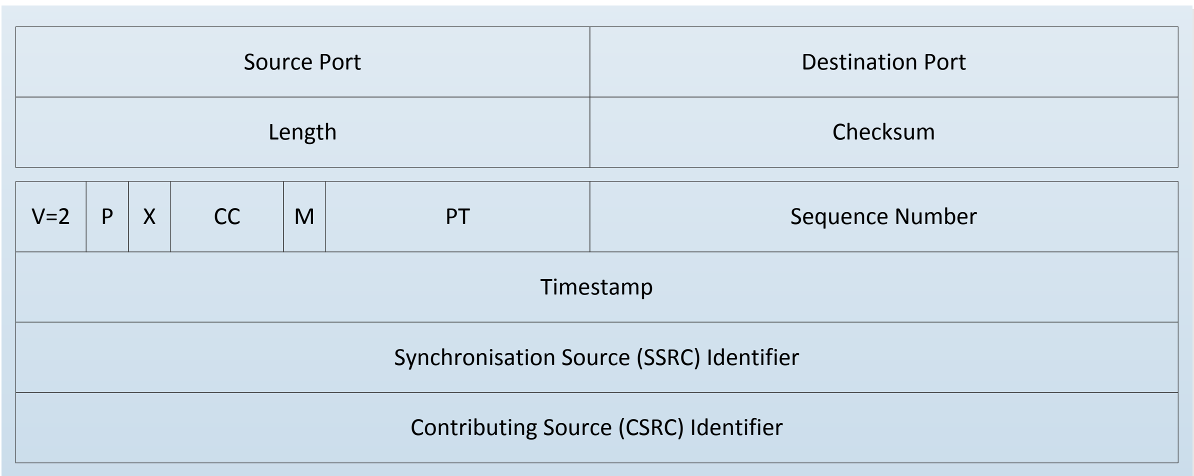

Figure 2.7: Real-Time Transport Header (Jonathan Davidson, 2006)

2.2.2.3.1. *RTP*

Shulzrinne (1996) defines RTP in RFC 1889 as a protocol used for communication systems that require transmitting real-time data such as telephony, conferencing and broadcasting services such as television which can tolerate delays that are introduced by TCP. The RFC document further explains that RTP is used in aggregation with RTCP (Real-Time Control Protocol), while RTP carries the media streams, RTCP is used to monitor the quality of service and to provide control such synchronisation of data (Shulzrinne, 1996).

According to Johnston (2001) SIP handles the signalling functions of multimedia sessions including voice and although it works with most transport protocols, its ideal transport protocol for real-time applications is RTP (Johnston, 2001). Figure 2.8 shows how SIP functions as a signalling protocol, with RTP as the transport protocol for a voice conversation.

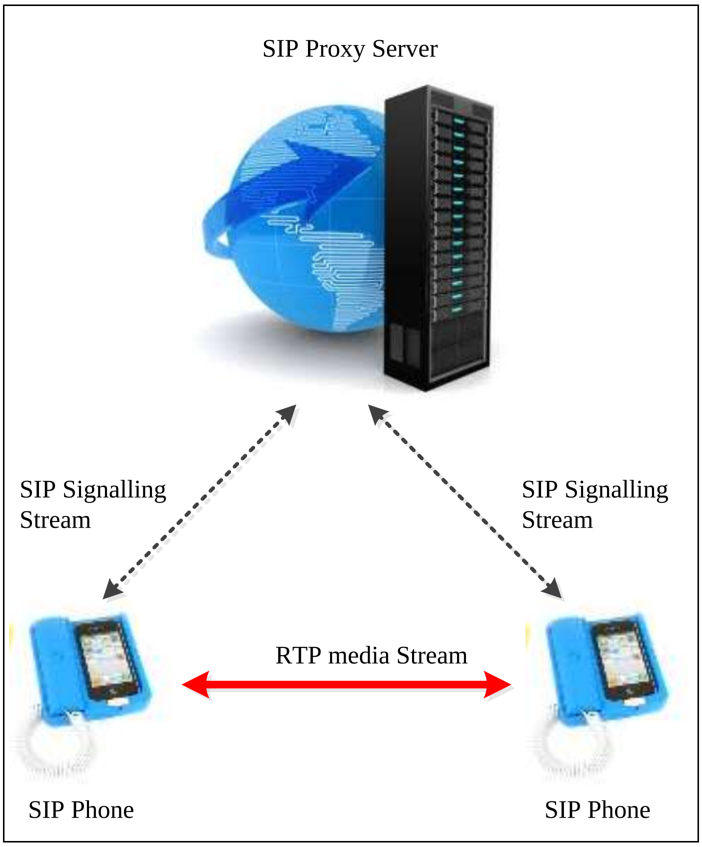

Figure 2.8: SIP is a Signalling Protocol & RTP transports the conversation (Johnston, 2001)

Camarillo (2002) explains that even though SIP works with RTP as its transport protocol, it still works in conjunction with other protocols such SDP (Session Description Protocol) in order to achieve end-to-end functionality of applications (Camarillo, 2002). SDP is defined in RFC 2327 document as the format for describing the characteristics and parameters of multimedia sessions, which makes it an essential protocol in VoIP Communication (Hanley M, 1998).

The RTP part of the multimedia protocol stack is represented in Figure 2.9.

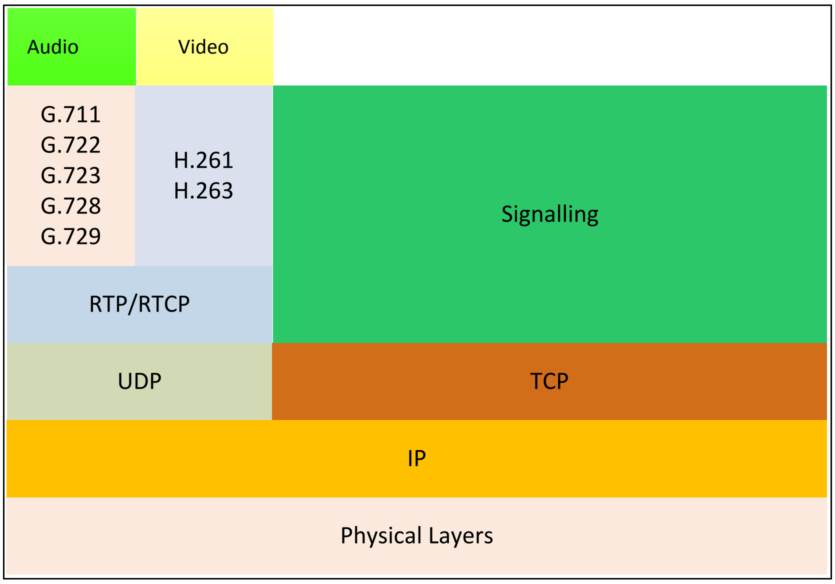

Figure 2.9: RTP Protocol Stack (Zeuch, 2008)

2.2.2.4. ROUTING PROTOCOLS

Faraz (2004) defines routing protocols as the softwares that specifies how routing equipment communicate with each other to advertise and learn routes and also to discover available routes (Faraz, 2002). According to Cooper (2012), BGP (Border Gateway Protocol) is the protocol which makes principal routing decisions over the internet. But Cooper continues to explain that even though this is the case on the internet, the routing systems on other networks is decided on by the administrator of the network.

Cooper (2012) adds that over fixed line data networks, the commonly used routing protocol are the Routing Information Protocol (RIP), Interior Gateway Routing Protocol (IGRP), Enhanced Interior Gateway Routing Protocol (EIGRP), Open Shortest Path First (OSPF) and the Intermediate System to Intermediate System (IS-IS). Some of the factors that decide which routing protocol to implement are the size and complexity of the network but RIP is the preferred routing protocol for VoIP since it is designed to work with UDP, while IS-IS and OSPF works directly with the Internet Protocol (Cooper, 2012).

2.2.2.4.1. ROUTING INTERNET PROTOCOL (RIP)

Malkin (1998) describes RIP in the RFC 2453 document as a protocol which dynamically makes use of distance calculations plus an outgoing network interface called a vector to choose a best path to a destination network. This type of functionality as RFC 2453 explains qualifies RIP as a distance-vector routing protocol. The document further explain that since RIP only makes use of hop counts to calculate the best route to a destination, it can become inefficient in a network making use of more than one LAN protocol because the shortest path might be the slowest (Malkin, 1998).

The RFC 2453 document further explains that the RIP protocol was mainly meant for smaller size networks hence it is limited to networks with a longest path of 15 hops and makes the protocol inappropriate for larger networks implementations because destinations becomes unreachable once the hop count has been reached. The document therefore adds that making use of fixed metrics to determine alternative routes can be inappropriate in cases where other parameters such as real-time parameters has to be considered (Malkin, 1998).

2.2.2.4.2. INTERIOR GATEWAY ROUTING PROTOCOL (IGRP)

According to Faraz (2002), IGRP is another distance vector protocol similar to RIP, but was designed to overcome the limitations of RIP by considering different variables (e.g. bandwidth and delay) when calculating its combined metric. Faraz adds further that IGRP makes use of a calculation demonstrated in Equation 2-1to calculate its metrics, where the lower the metric the more favourable the route become. By using these variables, IGRP provides a better method for routing packets then RIP (Faraz, 2002).

$$\boxed{\text{IGRP Metric} = \left[K1 * BW + \frac{(K2 * BW)}{(256 - \text{Load})} + K3 * \text{Delay}\right] * \frac{K5}{(\text{Reli} + K4)}}$$

Equation 2-1: IGRP Metric Equation (Faraz, 2002)

The different parameters of the equation are defined below (Faraz, 2002):

- K1,K2,K3,K4,K5 = constants;
- Default values: K1 = K3 = 1, K2 = K4 = K5 = 0;
- BW = 10^7 /(min bandwidth along paths in kilobits per second);
- Delay = (Sum of delays along paths in milliseconds)/10;
- Load = load of interface;
- Reli = Reliability of the interface.

With reference to Equation 2-1, the EIGRP metric makes use of constant numbers K1 to K5, which if inserted in the equation, reduces the equation to IGRP metric = BW + Delay, meaning that IGRP takes into account only the bandwidth and delay to calculate its metrics (Faraz, 2002).

2.2.2.4.3. ENHANCED INTERIOR GATEWAY ROUTING PROTOCOL (EIGRP)

Rouse (2007) defines EIGRP is a routing protocol that enhances the exchange of information between routers. Rouse further explains that EIGRP uses the Diffusing-Update Algorithm to determine the least cost route to a destination, this possible because a DUAL enabled system contains decision information to determine the more efficient route by considering distance and whether the route is loop free which is not accomplished by IGRP (Rouse, 2007).

Faraz also explains that EIGRP also overcome the problems caused by RIP and IGRP such as “Full periodic routing tables that consumes bandwidth” by only providing updates when a network has been changed. Therefore EIGRP is better choice when implementing large networks because besides consuming too much bandwidth, it also supports route summarisation which reduces the numbers of routes that a router must maintain and therefore saves on memory and CPU resources (Faraz, 2002).

2.2.2.4.4. OPEN SHORTEST PATH FIRST (OSPF)

IETF defines OSPF in RFC 2328 as a protocol based on link-state technology, meaning that each router making use of the protocol maintains a database referred to as a link-state database which describes the whole network topology. The RFC 2328 document further explain that a similar database is kept on all the routers with information pertaining information such as the router’s usable interfaces and its neighbours. According to TFC 2328 the router uses the method of flooding, to announce its state throughout the whole network (e.g. internet). OSPF has the ability to automatically detect topological changes in the network and calculates new loop-free paths (IETF, OSPF, 1998).

2.2.2.4.5. INTERMEDIATE SYSTEM TO INTERMEDIATE SYSTEMS (IS-IS)

IETF defines IS-IS in the RFC 1142 document as link-state routing protocol, which operates by flooding link state information throughout a network of routers. The document further explains that each router making use of the IS-IS protocol builds its own database of the network topology by incorporating the flooded network information from other routers. IS-IS and OSPF makes of the Dijktra algorithm which calculates the shortest path between a node and any other node and then forwards packets based on the calculated low cost path, through the network to the destination (IETF, IS-IS, 1990).

2.2.3. SIP ARCHITECTURAL COMPONENTS

Ohrtman (2002) explains the SIP network as network made out of two basic components which are the SIP server and the SIP client or SIP user agent. According to Ohrtman, SIP clients are peer components that initiates and answer calls while SIP Servers are responsible for the setup of the call. Therefore a computer installed with a telephone application can also be defined as a SIP client (Ohrtman, 2002).

The SIP Architecture defines the following functional elements (Sip Architecture, 2012):

- **User Agent Client (UAC)** – an end device or application which can originate and receive SIP calls. E.g. Cell phone, PDA or Laptop.
- **User Agent Server (UAS)** – a Server that receives and responds to SIP requests on behalf of the clients: Accepts redirects and refuse calls.
- **Back-to-Back UA (B2BUA)** – Is a SIP agent acting both as a Server and Client.
- **Redirect Server** – a redirect server is a user agent server that generates 3xx redirect responses to requests received, directing the client to contact an alternate set of URI’s.
- **Location Server** – Is a network node that determines the location of the SIP terminal
- **Proxy Server** – Is a server that acts as an intermediary for requests from clients seeking for services from other servers.

An example of a SIP architecture is shown in Figure 2.10:

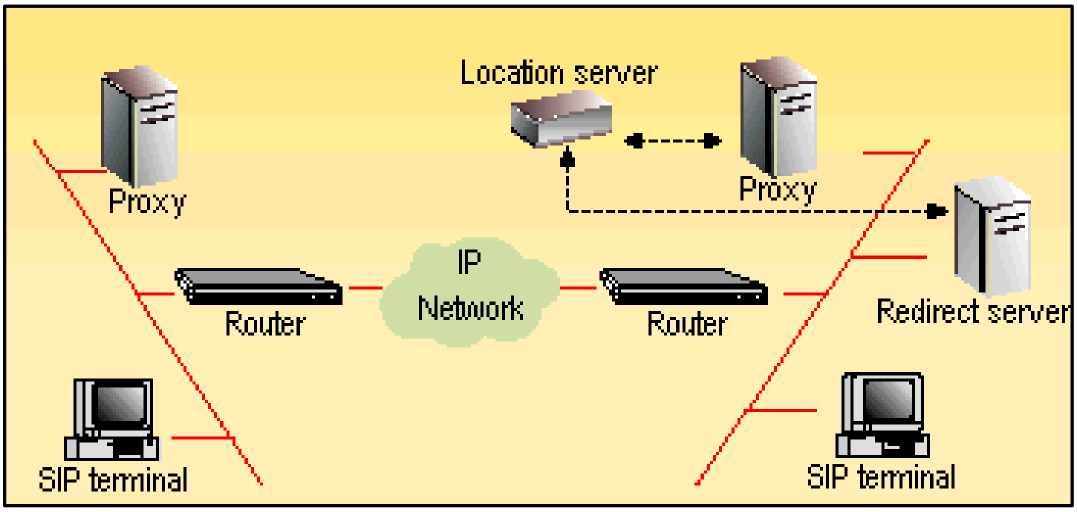

Figure 2.10: SIP Architecture (Sip Architecture, 2012)

2.2.4. SIP MESSAGES

Radvision (2001) defines SIP signalling as the setting up, modification and tearing down of communication and collaboration sessions which is realised through the exchange of messages: requests and responses of which the requests are sent from a client to a server whereas the responses are sent from server to the client (Radvision, SIP:Protocol Overview, 2001).

Edward (2007) explains that the requests and responses share a common message format which consists of a start-line, one or more header fields, and an optional message body. The SIP message structure is illustrated in Figure 2.11 (Edward Oguejior, 2007).

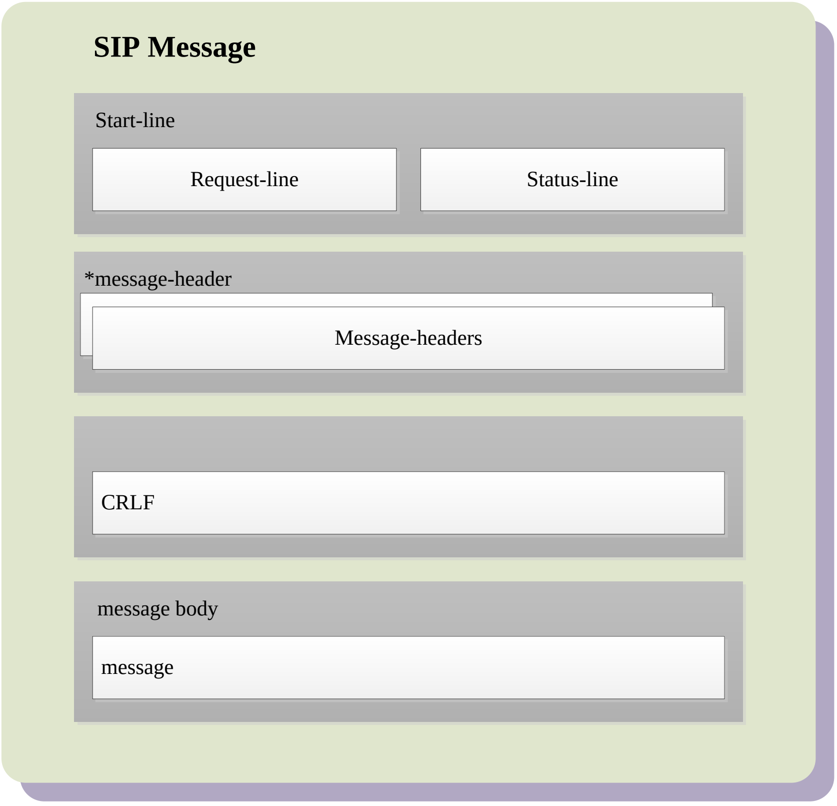

Figure 2.11: Structure of SIP messages (Edward Oguejior, 2007)

Edward explains the structure of the SIP message as follows (Edward Oguejior, 2007):

- Start Line – Every SIP message begins with a Start Line, The Start Line conveys the message type (requests and responses) and the protocol version;
- Headers – Used to convey message attributes and to modify message meaning;
- Carriage Return Line Feed (CRLF) – Required by SDP (Session Description Protocol) that there should be an CLRF terminating in each line of a SIP transaction in order to distinguish between message boundaries;
- Body – Used to describe the session to be initiated (for example, in a multimedia session this may include audio and video codec types, sampling rates etc.)

2.2.4.1. SIP Requests

Camarillo (2002) explains that the SIP requests have a Request-line which consists of three fields that are separated by a single space (SP) character. The parameters are explained as below (Camarillo, 2002):

- **Method** – Indicates the method to be performed;
- **Request-URI** – Holds a SIP URI. Indicates the user to which the request is address;
- **SIP-version** – Identifies the version of SIP protocol that is in use.

Table 2-1 illustrates the SIP Requests Methods:

Method	Description
INVITE	Initiates a call, changes call parameters (re-INVITE)
ACK	Confirms a final response for INVITE
BYE	Terminates a call
CANCEL	Cancels searches and "ringing"
OPTIONS	Queries the capabilities of the other side
REGISTER	Registers with the location service
INFO	Sends mid-session information that does not modify the session state

Table 2-1 : SIP Request Methods (Camarillo, 2002)

A SIP INVITE request is demonstrated in Figure 2.12:

```
INVITE sip:user2@server2.com SIP/2.0
Via: SIP/2.0/UDP pc33.server1.com;branch=z9hG4bK776asdhds Max-Forwards: 70
To: user2 <sip:user2@server2.com>
From: user1 <sip:user1@server1.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.server1.com
CSeq: 314159 INVITE
Contact: <sip:user1@pc33.server1.com>
Content-Type: application/sdp
Content-Length: 142
```

Figure 2.12: SIP INVITE request (Banerjee, 2006)

2.2.4.2. SIP Responses

Edward (2007) explains that the SIP response messages contain response codes that are made out of numbers which are almost similar to HTTP response codes. The document further explains that there are two types of responses and six classes as shown below (Edward Oguejior, 2007):

Response Types:

- Provisional (1xx class) – provisional responses are used by the server to indicate progress, but they do not terminate SIP transactions;
- Final (2xx, 3xx, 4xx, 5xx, 6xx classes) – final responses terminate SIP transactions.

Classes:

- 1xx = provisional, searching, ringing, queuing etc;
- 2xx = success;
- 3xx = redirection;
- 4xx = request failure (client mistakes);
- 5xx = server failures;
- 6xx = global failure (busy, refusal, not available anywhere).

2.2.5. SIP TRANSACTIONS

Edward (2007) defines a SIP transaction as a set of messages exchanged by SIP components which are involved in a session, where clients sent requests and servers sent responses to those requests. A transaction can therefore be an Invite or a Non-invite, where Invite transactions can establish a long-running session while a Non-invite transaction is based on a timer function which monitors if request are responded to in time or otherwise a timeout is reached (Edward Oguejior, 2007). A SIP Invite transaction[2] is illustrated in Figure 2.13.

[2] In VoIP a SIP transaction can also be referred to as a SIP Call

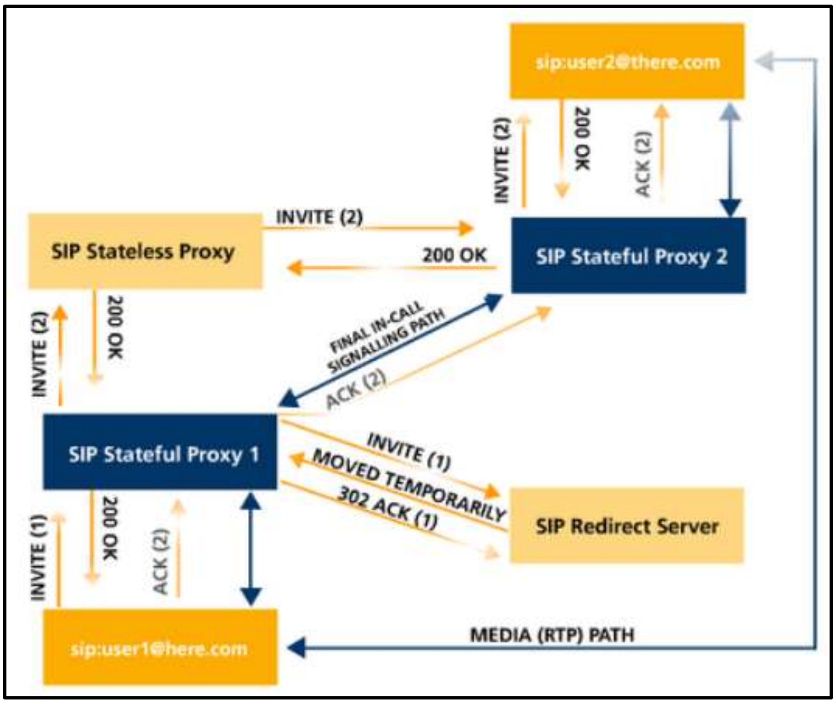

Figure 2.13: SIP Invite Transaction using a Proxy Server (Johnston, 2001)

With the knowledge gained from SIP Transactions, the call in Figure 2.13 can be analysed as follows:

1) The caller sends an invite message to its own proxy with the called party's information in the "To" field;
2) The location server (which can co-exist on the proxy server) determines the location of the called party;
3) The INVITE request is forwarded to the right called party;
4) A 200 OK is send back by the called party to indicate that the INVITE request was processed successfully. Note that this is a positive final response meaning that this is the only response the caller will receive as an indication by the called party that a session invitation has been accepted.
5) The caller then acknowledges (ACK) the response and once the ACK is delivered to the called party, a media session is established.

2.2.6. Conclusion

This section highlights the protocols used in the VoIP network and as the section describes, the SIP protocol is the control or signalling protocol responsible for establishing, modifying and terminating multimedia IP sessions. In the VoIP environment, this is an application level protocol which needs to be involved in the communication between applications exchange messages. Without the SIP protocol communication cannot be initiated. Therefore understanding the SIP protocol can assist the researcher with fault finding during VoIP implementations. Understanding the SIP messages can pinpoint exactly the communication status if it's successful or not. This might require tools to do the exact decoding but knowing the SIP messages and their related codes will be of assistance in case when communication must be re-established in a very short time. Understanding the SIP transaction will also assist the researcher in structuring the flow of the application code, hence the element executing the code will be required to register with another element such as a proxy server which handle the signalling and control functions of the VoIP network.

2.3. Adoption Factors of VoIP

The business environment has changed dramatically within the last decade. Globalization and market liberalization has altered the way a firm competes within this environment and how the firm interacts both with its customers and suppliers. For example:

- Both customers and competition have become global. To cut cost and to ensure easy access to customers, production and sourcing have shifted overseas.
- Technology has become complex and sophisticated.
- The use of communication networks is widely available in many parts of the world.
- To manage customer expectations and needs, firms have begun to form alliances and partnerships to manage their supply chain (Mathiyalakan, 2006).

To compete in this new economy, firms are looking at many strategic options. Recent events noted below suggest that firms, in particular large ones, are exploring the use of Voice over Internet Protocol (VoIP) as means to cut costs, to improve productivity, and the firm's strategic position:

- Bank of America is deploying more than 180,000 Cisco VoIP phones across its branches.
- Boeing has announced plans to equip its 150,000 workers with VoIP

- BT, the major telecommunications player in UK has announced that it plans to convert its infrastructure to VoIP by 2009 (Mathiyalakan, 2006).

VoIP is a relatively new technology. As with any new technology, some firms quickly adopt technology while other firms wait or ignore it. Moore developed a model for categorising new technology adopters. According to Moore, based on the time of adoption, firms may be placed in one of five categories: innovators, early adopters, late majority and laggards[3] (Moore G. , 1991).

The researcher argues that in many cases, firms may not be ready for another investment if they made a recent significant investment in mobile and next generation networks. Organisation readiness can be a combination of many factors, for example the organisation skill set can also contributes to the type of strategic decisions that are made in technology choices. An Organisation can be a laggard towards a certain technology trend because there is no suitable organisational structure in place to support such a technology. With no proper structure in place, companies must initially spend a lot of money on external support.

But still in some parts of the world (especially in the USA) it's reported that investment in VoIP is accelerating. For example, telecommunications giants such as Avaya (http://www/avaya.com) and Cisco (http://www.cisco.com) report successful implementations in hundreds diverse firms.

The view of the researcher is that it is easier to adopt VoIP in countries where free market policies are practiced. Therefore in USA, an individual firm make the decision to adopt the VoIP technology, whereas in many third world countries, state enterprises often operates the telecommunications networks and this enterprises may not be quick to adopt any technology that can have a negative impact on their profitability. The enterprises are also filled with many decision makers in the adaptation matter and an agreement might take years to reach.

Additionally, it is also difficult for the decision makers to give a green light to have the existing networks replaced with VoIP if the return on investment cannot be demonstrated. Security on the VoIP networks is also a big concern hence the IP network can easily be penetrated externally compared to the PSTN networks. Therefore if decision makers are not convinced than an organisational innovation adoption might not be supported.

But besides the fear of adopting the VoIP services, one should also consider the benefits of such an adoption. The justification of implementing such a network includes:

- **Increasing flexibility** – easy to provision VoIP users, because they are not fixed to physical ports.

[3] Slow responders

- **Simplified network structures** – the voice network moves from the traditional distributed core control network to a centralised core control network, which makes it easier to manage and deploy staff.
- **Enabling new features** – In the traditional PSTN networks it is very expensive and complicated to add new features to the network due to the fact that the operating systems are supplier proprietary. But in VoIP 3rd party applications can easily be developed and adopted.
- **Lower voice call costs** – All calls are seen as local calls and no additional zones (e.g. national and international) are introduced.

VoIP as it has been highlighted in this section is a new technology which is easily accepted by some and highly repelled by others. The section highlights many factors why this technology be adopted, hence it's a game changer on how we communicate. There are also many benefits in deploying VoIP networks, example OPEX reduction. Therefore the section highlights the factors that can help motivate the deployment of the VoIP technology and therefore inspires why research is required for this technology. There are also negative aspects to the introduction of such a technology, for example VoIP means a convergence of Voice and data, which at the end of the day can lead to the reduction of staff in a telecommunications organisation. Therefore there are also many areas that needs to be researched for example the social impacts of VoIP and ways to mitigate the risks.

2.4. MOBILE VOIP

IEEE (2012) defines the standard 802.11 as a definition for one medium access control (MAC) and various physical layer (PHY) specifications for implementing wireless local area networks (WLAN) in the 2.4, 3.6 and 5GHz frequency bands. The IEEE standard 802.11 has received a series of amendments over the years which are specifications for over the air modulation techniques that use the same basic protocol (IEEE, Wireless LAN Medium and Access Control and Physical Layer Specification, 2012). Table 2-2 defines some of the 802.11 amendments which need to be understood in order to know the specifications of elements required for a certain WLAN solution:

IEEE 802.11 Specifications	Descriptions
802.11a	The standard operates in the 5GHz band and uses a 53-subcarrier Orthogonal Frequency Division Multiplexing (OFDM) encoding scheme with a maximum raw data rate of 54 Mbit/s. Used mainly for backhauling in overcrowded areas.
802.11b	A standard with an extended throughput of up to 11 Mbit/s in the 2.4GHz band. It is widely implemented around the world under the marketing name Wi-Fi. The standard also allows wireless functionality comparable to Ethernet.
802.11c	Is an amendment to the IEEE 802.1D MAC bridging standard to incorporate bridging in wireless bridges or access points. 802.11c is a supplement to IEEE 802.1D that adds requirements associated with bridging 802.11 wireless client devices.
802.11d	Is a specification that adds support for "additionally regulatory domains". This supports includes the addition of a country information element to beacons, probe requests and probe responses.
802.11e	A wireless draft standard that defines the Quality of Service (QoS) support for LANs, and is an enhancement to the 802.11a and 802.11b wireless LAN (WLAN) specifications. 802.11e adds QoS features and multimedia support to the existing IEEE 802.11b and IEEE 802.11a wireless standards, while maintaining full backward compatibility with these standards.
82.11F	Also referred to as inter-Access Point Protocol is a recommendation that describes an optional extension to IEEE 802.11 that provides wireless access point communications among multivendor systems.
802.11g	Applies to wireless LANs and is used for transmission over short distances at up to 54-Mbps in the 2.4GHz bands. Also marketed as Wi-Fi and also makes use of OFDM transmission.
802.11h	It refers to the amendment added the IEEE 802.11 standard for spectrum and transmit power management extensions. It solves problems like interference with satellites and radar using the same 5 GHz frequency band.
802.11i	802.11i supersedes the previous security specification, Wired Equivalent Privacy (WEP), which was later discovered to have severe security vulnerabilities. Wi-Fi protected access (WPA) was first introduced by the Wi-Fi alliance as an intermediate solution to WEP insecurities. WPA

	implemented a subset of a draft of 802.11i. The Wi-Fi Alliance refers to their approved, interoperable implementation of the full 802.11i as WPA2. 802.11i makes use of the Advanced Encryption Standard (AES) block cipher, whereas WEP and WPA use the RC4 Stream cipher
802.11n	802.11n builds upon previous 802.11 standards by adding multiple-input multiple-output (MIMO). The additional transmitter and receiver antennas allow for increased data throughput through spatial multiplexing and increased range by exploiting the spatial diversity through coding schemes like Alamouti coding. The real speed would be 100 Mbit/s (even 250 Mbit/s in PHY level), and so up to 4-5 times faster than 802.11g. Also marketed as Wi-Fi.
802.11r	802.11r, also called Fast Basic Service Set (BSS) Transition, supports VoWi-Fi handoff between access points to enable VoIP roaming on a Wi-Fi network with 802.1x authentication.
802.1X	Not to be confused with 802.11x (which is the term for the family of 802.11 standards) 802.1X is an IEEE standard for port based Network Access Control that allows network administrators to restrict the use of IEEE 802 LAN service access points to secure communication between authenticated and authorised devices.

Table 2-2: 802.11 IEEE Specifications (IEEE, 802.11, 2012)

According to Frank (2004), the emergence of voice over 802.11 (Vo802.11) was made possible by simple moving VoIP over 802.11 as an access mechanism. Once the VoIP stream reaches the wired part of such a network via a wireless access point, it is transported on an IP network (Frank, 2004).

Mobile VoIP only caters for the wireless access side of the VoIP network, whilst the servers discussed in the VoIP section are also active on the wired side of the network. Besides identifying the differences in the 802.11 specifications, Table 2-2 also assist the research in answering the following questions:

- How do I deploy my mobile VoIP network (which standards must the devices support)?
- How do I optimise my mobile VoIP network (e.g. throughput)?
- Which wireless networking standards should I use for my mobile VoIP network?
- Which network elements can I use to secure my network?
- Which network elements can cater for the required functionalities (e.g. mobility and roaming)?

Therefore to upgrade a fixed VoIP network to a mobile VoIP, one only needs to add a VoIP-compatible wireless access point (WAP) which can support handoff. As per Table 2-3 by Cisco, WAP devices supporting Wi-Fi have a limited range, so they should be added to the network in a manner that is useful for the users. Note from Table 4 that the antennas used for 802.11b and 802.11g are similar since they both separate in the same frequency range. Careful Radio network planning must be executed in order to deploy reliable coverage. Finally users need a wireless IP telephone that can connect to the wireless network.

Data Rate (Mbps)	802.11a (40 mW with 6 dBi gain diversity patch antenna)	802.11g (30 mW with 2.2 dBi gain diversity dipole antenna)	802.11b (100 mW with 2.2 dBi gain diversity dipole antenna)
54	13 m	27 m	--
48	15 m	29 m	--
36	19 m	30 m	--
24	26 m	42 m	--
18	33 m	54 m	--
12	39 m	64 m	--
11	--	48 m	48 m
9	45 m	76 m	--
6	50 m	91 m	--
5.5	--	67 m	67 m
2	--	82 m	82 m
1	--	124 m	124 m

Table 2-3: Comparative Ranges in an open indoor office environment through cubicle walls (Cisco, Capacity, Coverage and Deployment considerations for IEEE 802.11g, 2005)

What is also exhilarating about 802.11 is that it allows the transmission of voice over an unlicensed spectrum. That is, for the cost of a radio and antenna, a service provider can offer voice services similar to that of a fixed line or mobile telecommunications operator and avoid the expense of copper wires, spectrum licences issues and wireless spectrum issues. Therefore Wi-Fi is an enabling technology that allows the local telephone companies to be bypassed. This section therefore clearly highlights the different 802.11 specifications used by the access points and therefore provides direction in which access point to use for which deployment.

2.5. IEEE 802.11 ARCHITECTURE

2.5.1. ARCHITECTURE COMPONENTS

Babbar (2005) explains that an 802.11 LAN is based on a cellular architecture where the system is subdivided into cells or Basic Service Set (BSS). Babbar explains that a BSS is formed when two or more stations communicate with each other via a single Base Station called an access point. Babbar further explains that a BSS can standalone without being connected to a base and the stations within are only able to communicate peer to peer, which makes it an Independent BSS or an Ad-hoc network as shown in Figure 2.14 (Babbar, 2005).

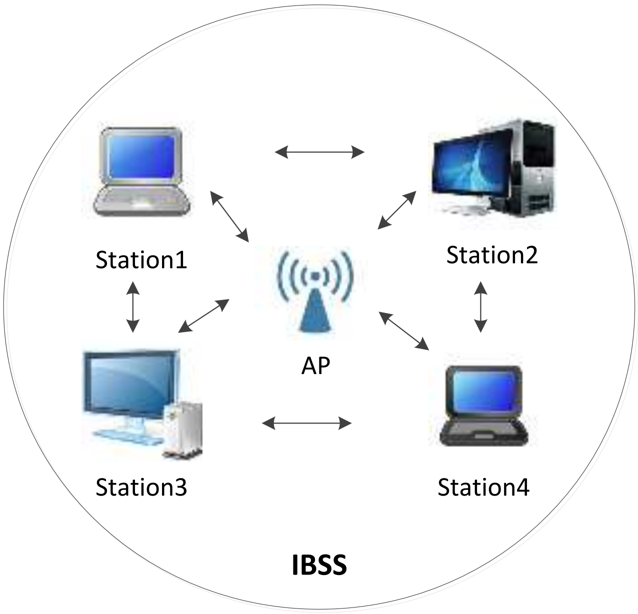

Figure 2.14: an Independent Basic Set Service (Babbar, 2005)

Babbar (2005) adds that most installations are implemented by several cells or Basic Service Sets, where the Access Points are connected through a backbone[4] network, which can be Ethernet or wireless itself. The interconnect Wireless LAN including one or several cells, their respective Access Points and the Distribution Systems, is seen to the upper layers of the OSI model as a single 802 network, and is called an Extended Service Set (ESS) as seen in Figure 2.15 (Babbar, 2005).

[4] Also called Distribution System or DS

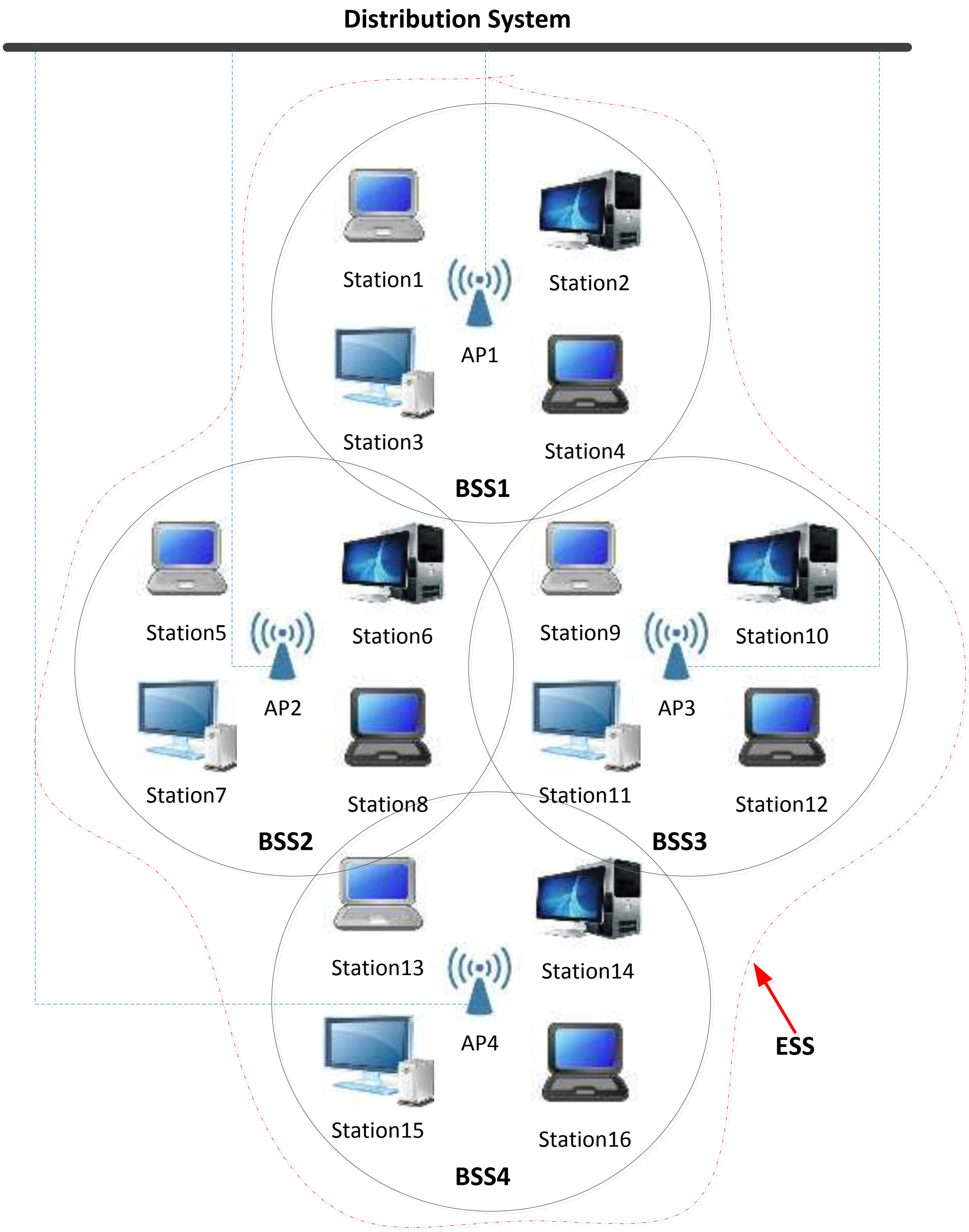

Figure 2.15: an Extended Service Set (Babbar, 2005)

Babbar (2005), explains that the beauty of creating extended service sets is that the entire ESS network looks like an independent basic service set to the Logical Link Control (LLC) layer of the 802.11. This means that stations within an ESS can communicate or even move between BSS's transparently to the LLC (Babbar, 2005). Figure 2.16 clearly shows that IEEE 802 standards operate in the lower layers of the OSI reference model.

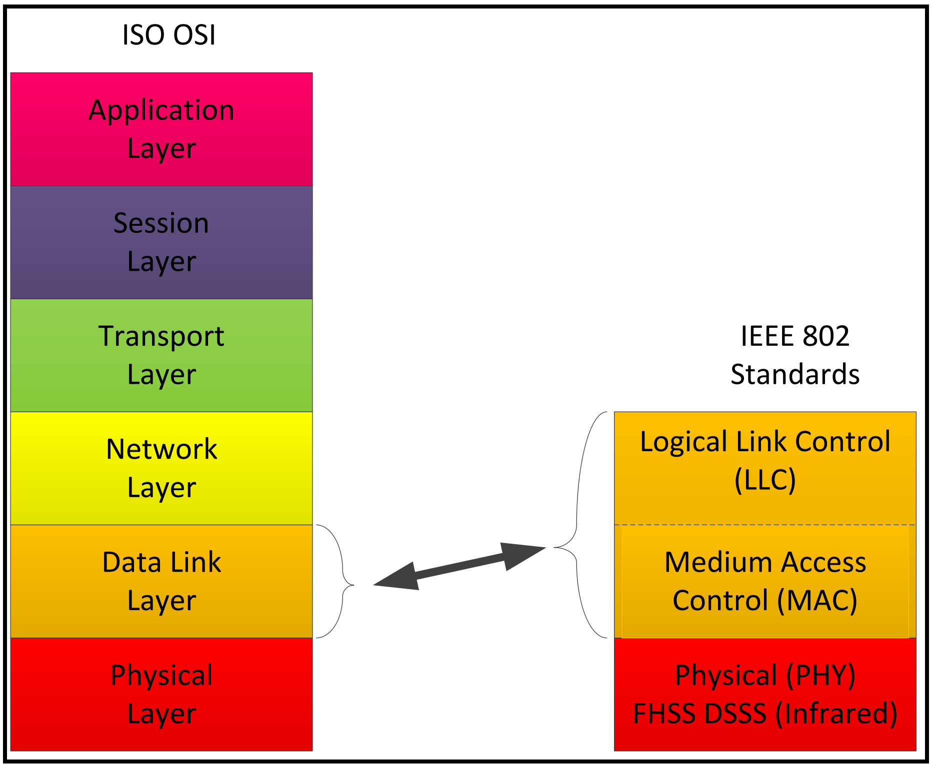

Figure 2.16: IEEE 802 and the OSI reference model (Babbar, 2005)

IEEE 802.11 does not specify how to implement the Distribution System and therefore a DS can be implemented from existing technologies. But nevertheless, IEEE 802.11(2008) does specify the services that a DS must support. These services are divided into two sections which are Station Services (SS) and Distribution System Services (DSS) (IEEE, IEEE 802.11, 2008).

2.5.1.1. Distribution System Services

The five services provides by the DSS and they are as follows (IEEE, IEEE 802.11, 2008):

- Association;
- Reassociation;
- Disassociation;
- Distribution;
- Integration.

Babbar (2005) explains that the first three services deals with station mobility. The following terminology is used to describe station mobility (Babbar, 2005):

- No-Transition – When the station is not moving or it is moving in its own BSS;
- BSS-Transition – When the station is moving between BSS's within the same ESS;
- ESS-Transition – The station moves between BSS's belonging to different ESS's.

Brenner (1997) adds that when a station enters a BSS it must first affiliate itself with the BSS infrastructure before it will be able to make use of the LAN. This is achieved by associating itself with one AP in order to get the synchronisation information by one or two means (Brenner, 1997):

- Passive Scanning: In this case the station just waits to receive a Beacon Frame from the AP.
- Active Scanning: In this case the station tries to find an Access Point by transmitting Probe request frames, and waiting for Probe Response from the Access Point.

Babbar (2005) further explains that association only supports services within a BSS (such as No-Transition) but does not support BSS-transition. Therefore Babbar (2005) explains that Reassociation is required to allow a station to switch its association from one AP in one BSS to another AP in another BSS. Babar (2005) explains Disassociation as the termination of an association between the station and an AP, which prevents a station from sending or receiving data (Babbar, 2005).

Babbar (2005) explains the Distribution service as primarily getting the data from the sender to the intended receiver, when the data is sent to the local AP (input AP) then transported by the Distribution System to the remote AP (output AP) which the receiver is associated with. Babbar (2005) explains that Integration is when the output AP is set up as a portal to link the wireless LAN to a DS.

2.5.1.2. STATION SERVICES

Station Services are described below (Babbar, 2005):

- Authentication
- Privacy
- Deauthentication
- MAC Service Data Unit (MSDU) Delivery

Babbar (2005) explains that the medium is not exactly confined with a wireless system which is not the case in a wired system and therefore in order to control access to the network, stations must first establish their identity, which is the process of authentication. Babbar (2005) further that there are two types of authentication services: the Open System Authentication and the Shared Key Authentication. With Open System Authentication, anyone who attempts to receive authentication is authenticated while with the Shared Key Authentication one need a password (Babbar, 2005).

Shamir (2010) defines Privacy as encryption algorithms implemented to prevent other WLAN users from eavesdropping on your LAN traffic. Below are descriptions of the algorithms used to achieve Privacy (Adi Shamir, 2010):

- Static WEP (Wired Equivalent Privacy) provides simple authentication and encryption based on static 40-bit or 104-bit encryption key that must be manually entered on all access points and stations. But the algorithm can easily be broken in by utilising tools such as Aircrack and Airsnort.
- WPA (Wi-Fi Protected Access) was introduced as the transitional measure to take the place of WEP. The Temporal Key Integrity Protocol (TKIP) which was adopted for WPA makes use of a per-packet key, meaning that it dynamically generates a new 128-bit key for each packet and thus prevents the types of attacks used to crack WEP;
- WPA2 (Wi-Fi Protected Access 2) is based on the final IEEE 802.11i amendment which provides government level type of security by implementing the National Institute of Standards and Technology (NIST) FIPS 140-2compliant AES (advanced encryption standard) algorithm. There are Personal and Enterprise version WPA2 available: In the personal mode, a password is used for authentication on a client device and access point, while in the Enterprise mode authentication is achieved via 802.1X which was described earlier in Table 2-2 typically requires an authentication server on the enterprise network.

Babbar (2005) explains that Deauthentication is achieved when either the station or AP wishes to terminate a stations authentication which is followed automatically by the Disassociation of the station (Babbar, 2005).

2.5.2. KEEPING SYNCHRONIZATION

Ye (2003) explains that time synchronization is an important design issue of wireless communication for time sensitive applications such as voice, in which the order of data is of great interest. Ye (2003) further explain that the IEEE 802.11 standard for wireless LAN specifies a time synchronization function (TSF) to fulfil timing synchronisation among users. Ye (2003) further explains that TSF keeps the timers for all stations in the same Basic Service Set (BSS) synchronised and this is achieved by stations periodically exchanging timing information through beacon frames (Ye, 2003).

Bill (2003) explains that clock synchronisation is needed for power management, synchronisation of frequency hopping and medium reservation. Bill (2003) further explains that synchronisation is not an issue in BSS (Basic Service Set) infrastructures since the access point is responsible for the timely transmission of the beacon frame (Bill, 2003). The paper of Ye (2003) adds that In an IBSS (Independent Basic Service Set) architecture the timer synchronisation function mechanism is distributed among the mobile stations and is therefore more complicated because more than one transmission may simultaneously occur causing a collision which in turn cause timer synchronisation problems (Ye, 2003).

The paper by Bianchi (2000) explains that the IEEE 802.11specifies a distributed coordination function (DCF) mode designed for IBSS communication. Bianchi (2000) explains that DCF is a protocol that does carrier sensing to prevent packet collision. The basic functionality is explained as follows (Bianchi, 2000):

- Whenever a station wants to transmit, it first wait for a random backoff time;
- If the station senses that another station is using the channel, it will pause its timer until the other station has finished transmitting;
- When the backoff time expires, the station senses the channel again to see if there is any other transmission;
- If the channel is clear, it will wait a short time and sense the channel again;
- If the channel is still free, it will transmit a request to send (RTS) to the destination upon which the destination replies with a clear to send (CTS) if it is able to receive data;
- The source station can then transmit data once it receives the CTS.

The researcher reasons that the packet network was originally designed for delivering time-insensitive data such as e-mail or static web traffic. Therefore with no end-to-end delay bounds, the packet networks best effort service is unsuitable for delivering time sensitive data and for interactive applications such as VoIP. Although the approaches to optimise the quality of service for the (m)VoIP application will not be explored in this paper, it is still important to the researcher to point out that a (m)VoIP system must make use of synchronised time to combine the useful characteristics of both fixed and adaptive buffer strategies, thereby improving VoIP quality of service which will prevents calls from being dropped. This can be achieved by using a combination of Global Positioning System and the network time protocol (NTP) which enables hosts to learn the precise end-to-end delay of each packet. It is therefore pertinent to know all the factors that might degrade the quality of Voice before the testing and implementation.

2.5.3. POWER SAVING

A key characteristic in mobile stations is the ability to operate using a limited energy resource, such as a battery. Because battery capacity is quite restricted, the issue of reducing power consumption in all aspects of wireless host design has been a challenge for many years of research in order to lengthen battery life. This section explores some of the past works that was conducted to address energy conservation.

Wi-Fi Alliance (2012) defines how WMM (Wi-Fi multimedia) works by stating that it increases the efficiency and flexibility of data transmission. According to Wi-Fi alliance (2012), WMM power save this is achieved by allowing the station to go into sleep mode between packet transmissions in order to save power, while the access point buffers downlink frames. Wi-Fi alliance (2012) further explains that with WMM power save the application chooses the time to wake up and receiver data to maximise power conservation without sacrificing quality of service. Wi-Fi alliance (2012) estimates a power savings improvement of 15% to 40% in battery life depending on the application characteristics (Alliance, 2012).

Brenner (1997) also explained that battery power is a scarce resource for mobile stations in a WLAN and therefore the 802.11had to address the power issue by defining a mechanism which could help station save power. Brenner (1997) explains that the approach was to have the AP maintain a database of all the stations working in Power Saving Mode and will keep buffering the packets addressed to this stations until such time that the stations is in an active mode and sends a polling request (Brenner, 1997).

Brenner (1997) explains that on the AP side, information about stations that are in sleep mode but has frames buffered at the AP side is sporadically transmitted as part of the beacon frames so that these stations can be forced to wake up and receive the frames. Brenner (1997) further explains that if there is an indication of frames buffered by the AP for a specific station, then that station should stay awake and send a polling request (Brenner, 1997).

This section clearly clarifies how communication is kept alive during power saving mode. Besides polling requests when the station is active, the AP also sends periodic updates that force the station to wake up when there is a packet for it which is essential in real-time applications such as VoIP. The section also highlights terminologies used in mobile VoIP networks and also explains the relevant processes involved in the mobile VoIP communications that needs to be understand while considering deployment. For example roaming, synchronisation and security are very important aspects of mobile VoIP design and deployment and therefore should be understood intensively during the research. This aspects also has an impact on the application design and therefore the researcher needs to understand all the processes involved on the network side (e.g. the authentication and association processes).

2.6. ADOPTION FACTORS OF MVOIP

There are many aspects that will determine the success of mobile VoIP in the telecommunications industry. According to the paper by Chrestin and Woyczechski (2005), there are more factors that determine the success of mobile VoIP, which are (Chrestin & Woyczechoski, 2005):

- **Quality of Service** – Quality and efficiency of the digital voice coding;
- **Security aspects** – prevention of fraud and illegal interception;
- **Regulatory aspects –** emergency functions and lawful interception;
- **User friendliness** – use of the available telephone infrastructure, “Plug & Play” of new infrastructure, availability, simple and transparent billing.

The researcher’s view is that these challenges are especially experienced on large scale mVoIP implementations because, the radio transmission capacities are shared by many users, which means that additional resources (e.g. bandwidth) needs to be provisioned in order to ensure quality of service and efficient transmission. This can solve problems that currently exist as high latencies, varying signals quality and interferences. In order for mobile VoIP to be adopted successfully, handover complications that might exist due to protocol mismatch must be resolved too.

Mark Milliman (2007) is a Principal consultant at Inphotonics Research[5] and his view on mobile VoIP is that it is more of a technology than a service just like VoIP is to fixedlines which will disrupt the

[5] Is a company that does planning and building access networks for service providers and carriers worldwide

mobile market much faster than VoIP is disrupting the fixedlines. He goes on saying that the latest communication trend is the convergence of all service on to IP because data transport is much cheaper than the service dedicated traditional methods of transport (e.g. circuit switched networks) used by services such as voice or mobile voice. The convergence of voice and data over one network reduces the cost of transport and also introduces opportunities for service providers to implemented services that bundles voice and data (Milliman, 2007).

The researcher also reasons that many local underdog operators and the ones without any landline infrastructure to protect will embrace mobile VoIP as a way to attract customers with added advantages like reduced monthly charges. The market leaders might resist the migration from traditional landlines, hence this is a huge investment which in some cases the return on investment is not evident but the resistance will fail soon as the regulators require them to open up their networks and allow the market to adopt mobile VoIP. Mobile VoIP can also be used as a tool to enhance productivity and therefore business users can easily adopt mobile VoIP. The market leaders will also be left with no choice but to reduce their telephonic costs but to maintain their profit they will need to rid of purpose networks (e.g. PSTN networks) and combine all services onto IP transport networks. This move will also help them to reduce operating stuff and power usage expenses.

According to a report by Research and Markets (2012) it was highlighted that the increase in NGN (next generations) implementations has been the key contributor to the growth of mobile VoIP technology. The report by Research and Markets (2012) also indicates that the mobile VoIP market in the EMEA (Europe, the Middle East and Africa) region has also been witnessing increasing usage of social networking sites but the slow deployment of mobile broadband services could limit the growth of the mobile VoIP market. The key vendors dominating this market space include Skype, Fring Ltd., Vonage Holdings Corp., and Truphone Ltd (markets, 2012).

Adding to the statements above, an analyst from TechNavio's Telecom team stated in the same report by Research and Markets (2012) that the convergence of mobile VoIP with social media will open up a lot of opportunities in the year 2013, which is one of the mega-trends of the mobile VoIP market in the EMEA region. Several providers are already developing applications to provide free voice calls between social networks such as Facebook but due to complications that are introduced by the service in to the regulatory domain many service providers are reluctant to add the service to their packages (markets, 2012).

Campbell (2011) notes that Voice over IP adoption rates are rapidly growing which is a result of strong subscriber rates and revenue growth rates in the business sector. He goes on stating that Comcast[6] and other cable companies have also implemented VoIP services so that they can expand

[6] Is an American provider of entertainment, information and communications products and services.

their customer reach and respond to VoIP service demand and as consumers and business users are increasing in the mobile space, the demand for mobile VoIP is also growing, creating new opportunities in the mobile space (Campbell, 2011).

The report by Campbell (2011) also state that to help satisfy the demand and create new opportunities in the mobile VoIP business, some networks providers are starting to build converged business communications solutions which bundles the Short Message Service with VoIP. Campbell further states that the increase in Wi-Fi capable handsets usage will also be an enabler for the growth of mobile VoIP. Campbell also states that another contributor to the growth of mobile VoIP is that upgrades from 3G and WiMAX to LTE which is on the drawing boards of some of the major service providers. Adding to this opportunities presented in mobile VoIP, Campbell states that competition and innovation will also contribute to the growth of mVoIP (Campbell, 2011).

Devin Moore (2007) is an IT-consultant who also has many years of experience in telecommunication systems. According to him, mobile VoIP service is restricted by outdoor public coverage and until such time that full cell coverage is possible, good quality implementations of mVoIP will not be possible. Devin adds that even though there are public broadband initiatives sponsored by cities, mobile VoIP is still unlikely to ever reach this coverage level because of the lack of handover capabilities on most phones He states that this technology should be implemented in the future because it will help the users save money whenever they are connected to a VoIP network (Moore D. , 2007).

Devin (2007) adds that total mobile VoIP coverage also will require tremendous bandwidth in order to provide service to all the mobile phones that are connected to the service in high traffic areas such as cities. This he adds may also limit the availability of mobile VoIP for the next few years, until such time that multi-gigabit wireless internet is commonly adopted. Devin adds that mobile VoIP is the future trend and therefore wireless providers should be early adaptors of the service if they want to retain their customers. Devin states that he expect that services such as voice, Internet, video, and all other streams will become a bundled offering to mobile device with a global and unlimited coverage for a single flat-rate (Moore D. , 2007).

Alan Williamson (2007) who is a java developer’s insight to mobile VoIP is that having experienced with mobile VoIP for a couple of weeks with the 3SkypePhone, he can only testify that it’s nice in theory but in reality it still has a long way to go before it becomes the must have feature. Williamson feels that the SMS (short message service) is still the killer application for a mobile phone because its cheaper for the user compared to the voice call and the delay introduced by the service is also acceptable. He continues stating that the network carriers also love it because they can make money from the service without worrying about the network being overloaded (Williamson, 2007).

Williamson (2007) also adds that all these benefits introduced by SMS is not available to mobile telephony, because telephony is a real-time service which requires unshared bandwidth during the whole duration of the call. He adds that because of these requirements of mobile telephony, the service quality is highly degraded once a user of the service (example the 3GSkypephone) steps out of the service area such as a fully covered 3G area (Williamson, 2007).

Servaas Schrama's (2007) (a freelance IT consultant) insight to mobile VoIP is that with the global availability of the SIP protocol, VoIP has become available to a wider range of equipment other than computers, such as phones and mobile devices. Schrama's adds that the main focus is on mobile devices, since they can be used in VoIP networks to provide either cheap or free communication. Bust as Schrama stresses, no matter which VoIP application (e.g. Skype) or software package you might have on a mobile device, you will still need a data connection to your mobile device. And this Schrama explains can be accomplished by either making use of data packages over GPRS (General Packet Radio Service) or 3G as provided by the service provider or one can make use of a Wi-Fi connection. Schrama further explains that GPRS provides a very slow connectivity and the data packages provided with are not always cheap unless if provided as flat free data plans (Schrama, 2007).

Schrama continues by posting a question: "So how will the providers reach the general public?" Schrama explains that service providers will not advertise their ability to provide VoIP as their core product, because the more users start making use of voice, the higher the bandwidth utilisation which will cost the service provider money. Schrama adds that service providers have the fear that they will make losses once they offer VoIP, while overlooking the fact that there are other service providers who are making a business out of this service. Schrama further adds that one advantage of offering mobile device is that support easy because there is not much difference between fixed and mobile VoIP if based on common standards. Schrama ends off by saying that the general public will adopt mobile VoIP once the service providers start offering it with products such as VoIP only without a GSM contract or a combination of both (Schrama, 2007).

In support with the above analyses from the Telco business, the researcher also feels that the penetration of next generation networks has been very slow, which is evident in our country Namibia, where fixed line voice calls (local, national and international) mostly originates from old traditional POTS networks. But this might change because most Telecom system vendors have discontinued developments in POTS networks and are focusing mostly in the direction of IMS/LTE. The introduction of such technologies will be able to boost the Mobile VoIP networks since operators will be able to interconnect on IP basis. In addition there are already international partners of our local fixed network operators who are requesting to interconnect on IP basis, which the backbone of mobile VoIP. Other projects like the WACS[7] will also enhance partnerships based on VoIP.

The adoption factors of mobile VoIP are therefore also important to this research hence they pin point what the researcher must consider while undertaking this research. For example the adoption factors highlight that there is a place for mobile VoIP in the world. Service providers are busy with projects to advance their networks that in turn can provide better transportation systems for mobile VoIP. Other researchers also provide ideas how users can benefit from mobile VoIP. Schrama (2007) stated that "At the moment, mobile VoIP is an added value, but imagine buying a pda with mobile connectivity, but without a GSM contract".

2.7. FACTORS OF MOBILE APPLICATION DEVELOPMENT

2.7.1. INTRODUCTION

Building a mobile application requires getting your application to yield high usability by the target audience. This means that the developer must identify the app development challenges beforehand, find solutions for the challenges. The developer must also plan and reduce costs before even considering attracting users.

Vision Mobile (2010) lends a sense of what launching a mobile application truly involves through the information highlighted below. These are six most crucial factors of mobile development that Vision Mobile highlights, based on its Developer Economics 2010 and beyond report (Vision, 2010):

- **Platform Selection**. The choice of platform to develop for is often a commercial choice which looks at the amount of users that the application can target instead of the technical specifications of a platform.
- **"Installed base" versus "available apps."** This factor looks at the comparison between how many users the platform has to how many applications are already available for the platform. Meaning that the platform might have the most users but fewer available applications.
- **Learning Curve.** This has to do with how easy it is to learn how to develop an app for a given platform. The platform must also have a large developer community and a good documentation support.

[7] West African Cable System

- **Best platform aspects vs. top pain points.** This factor looks at how quickly a developer can code and create a prototype for the platform versus which platforms provides better usability.
- **Go-to-market.** This factor looks at which platform provides the best marketing channels for getting the application out to the users.
- **Time-to-market vs. time-to-payment**. This factor looks at how soon can payment be received once the application has been marketed.

By considering the factors above, a developer will be able to make a great mobile idea even better because (s)he understands the technology and market knowledge to project costs, time allocations, and the benefits and setbacks of a given platform. The sub-sections below elaborate on some of these aspects.

2.7.2. Platform Selection

Before a decision can be made on which is the best platform to develop for, one must also look at the standings on how the different mobile platforms compares with each other. A report by Warren (20111) discusses the 5 common platforms that defined the mobile space in the year 2010. According to the report, Android has grown bigger and bigger, chipping away at the market shares that were previously held by RIM (Research in Motion), Apple and Symbian. The report also mentions that still the mobile platform space was hardly monopolised by one company, because the OS doesn't matter anymore since it is less about the underlying structure and more about the user experience and development tools (Warren, 2011).

Below is Warren's (2011) list of 3 of the 5 Mobile platforms that changed the mobile industry in 2010. The other two platforms compared with are unity and Appcelerator, which are not actual platforms but toolkits instead:

Rating	Mobile platform	Comment
1	Android	The iPhone dominated technology news in 2007, 2008 and 2009. It's hard to argue that no any other device, software program or piece of technology had more of an impact and an industry as each android version launched through the years. In 2010, Android displaced the iPhone as the best-selling smartphone platform in the U.S., powered many of the hottest smartphones including the EVO 4G, Droid x and

		Samsung Galaxy S. The android market grew by leaps and bounds and more and more developers indicated that they see Android as the long-term path to success. But the real news with Android wasn't just phones. E-book readers, laptops, tablets and slate computers, Google TV set-top boxes, car systems, television sets – you name it, an Android-based variation is either out or probably in the works. Android's rise from second or third-tier mobile platform to mobile superstar and embedded system of the future is certainly one of the biggest stories of 2010.
2	iOS	Apple may have faced some tough competition in 2010, but the company didn't let iOS sit idle. The fourth generation iPhone, the introduction of iOS and of course, the iPad still showed that apple is in the game to play. As a platform, iOS continues to enjoy the largest application store (200,000 apps and counting) and is the commercial platform of choice for many developers large and small. With iOS 4, the company added some features to bring the OS to parity with some of the competition, features like folders and multitasking and better notifications, while still introducing its own special features like FaceTime, Game Center and iBook store. Still the biggest thing to happen to iOS was the iPad. The iPad is not just one of the biggest technologies of the year, it's one of the most successful product launches of all time. Millions of units have sold in the last six months with the supply levels finally reaching the point that the device can be sold from outlets like Target, Walmart and Amazon.com. The iPad is helping transform the publishing industry, is being used in education, and is appealing to users and buyers of all stripes. iOS faces more competition than ever but the platform continues to remain strong and for many, is still the undisputed champion when it comes to consistent, usable interface.
3	Windows Phone 7	Microsoft isn't a company that can be described as the underdog in any arena. In mobile, however, it's a pretty fair assessment. After ditching it's Windows Mobile Platform (now dubbed Windows

	phone classic) Microsoft formally announced Windows Phone 7 in February of 2010. With Windows phone 7, Microsoft did a very un-Microsoft thing by cutting all ties from the its legacy Windows Mobile platform. Windows 7 was started from the ground up and it took another approach to interface and smartphone user motifs. The Windows Phone 7 which is partly Zune[8], portable Xbox and minicomputer is taking a bit of different path than its competitors like Android, iOS and BlackBerry. With the different approach, Microsoft hoped to distinguish itself in the marketplace.

Table 2-4: Mobile Platforms that defined the mobile space in 2010 (Warren, 2011)

It is also important to know, on which devices is a platform installed in order for readers to understand the impact a certain platform is having on the market in cases where reports are only reflecting the devices that are shipped or sold per year. A comparison was done by DevX[9] (2010) in Table 2-5. This includes the devices that a certain platform runs on and who the target audiences are (DevX, 2010):

Platform	Devices	Target Audience
Android	Smartphones and PDA's from: Motorola, LG, HTC, Sony, Sony Ericsson, Samsung, Alcatel, Bluelans Communication, NEC, CD-R King, Cherry Mobile, CSL, Dell, Garmin, GeeksPhone, General Mobile, Huawei, T-Mobile, Kyocera, Sanyo, Lenovo, ZTE, Acer inc.	Consumers for personal use
Blackberry	Blackberry Smartphones in a variety of models	Primarily business users
iPhone (iOS)	~iPhone 3G ~iPhone 4	~Primarily early developers, young adults ~Enterprise users have been reluctant to jump on board

[8] Is a digital media brand owned by Microsoft which includes a line of portable media players, a digital media player software for Windows machines, a music subscription service known as a 'Zune music Pass', music and video streaming for the Xbox 360 via the Zune Software, music, TV and movie sales, and the media software for Windows Phone.
[9] Is the leading provider of technical information, tools and services for IT professionals developing corporate applications.

Windows Phone 7	Smartphones from: HTC, Samsung	Business and average consumers
Palm WebOS	~Mostly Palm smartphones.	Business and consumer users
Symbian	~Nearly all devices using Symbian are Nokia phones. ~A few Sony Ericsson phones and one from Arima.	Business and consumer users

Table 2-5: Mobile Development Platform Comparison Matrix (DevX, 2010)

Another important factor that needs to be considered in mobile application development is the growth (or shrinkage) of a platform in the smartphone market share or the position that a certain platform is holding at the moment. This factor will determine if a certain platform is worth developing for (it has a huge customer base) or not (if it's slowly approaching its end).

According to the report by Gartner (2012), In 2006, 64 million smartphones were sold while platforms such as Android, iOS, Windows Phone, and Bada did not yet exist. The report adds that by 2012 the market has grown nearly 10 times and the top smartphones by market share are Android, Symbian, Apple iOS, RIM Blackberry, MeeGo, Windows Phone, and Bada. Nokia's S40 which is a non-smartphone OS-base is not included in the report. The chart in Figure 2.17 (Gartner, 2012) depicts the world-wide market share in percentage:

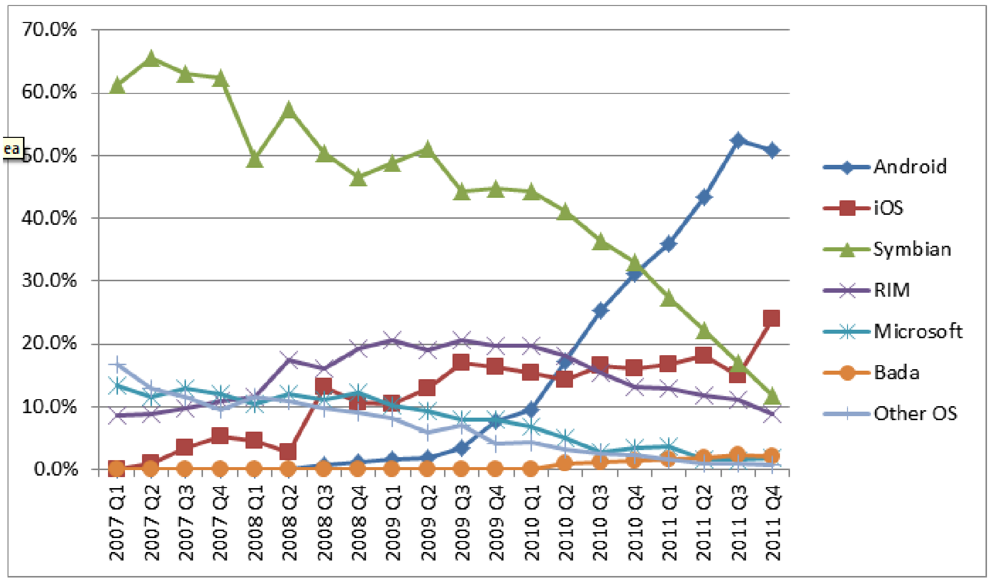

Figure 2.17: World-wide Smartphone Mobile OS Marketshare % (Gartner, 2012)

The statistics show that smartphone sales to end users reached 115 million units in the third quarter of 2011, up 42% from the third quarter of 2010. Sequentially, smartphone slowed to 7% growth from the second quarter of 2011 to the third quarter of 2011. Smartphone sales accounted for 26% of all mobile phone sales, growing only marginally from 25% in the previous quarter. The report by Pettey (2011) also noticed how China and Russia as opposed to the Western countries helped increased the overall sales (Pettey, 2011).

According to Pettey (2011), it was noticed by Gartner analysts that the slowdown of consumption in the 3rd quarter of 2011 was caused by consumers waiting on promotions on smartphones that are normally launched in the 4th quarter holiday season. According to the report another factor was consumers who held off from upgrading due to a rumoured new iPhone and associated price cuts on older iPhone models which affected U.S. sales (Pettey, 2011).

According to Pettey's (2011), despite a drop in market share, Nokia continued to be the worldwide leader in mobile devices sales as it accounted for 23.9 percent of global sales, as shown in Table 2-6 which compares the hardware mobile device sales for the 3rd quarters of 2010 and 2011. The report acknowledges that the second quarter of 2011 was the low point for Nokia, and the third quarter brought signs of improvement and the main reason analysed for this improvement was Dual-SIM phones and feature phones which maintained Nokia's momentum in emerging markets (Pettey, 2011).

Vendor	2011 Q3 (1,000 Units)	2011 Q3 Market Share (%)	2010 Q3 (1,000 Units)	3Q10 Q3Market Share (%)
Nokia	105,353.5	23.9	117,461.0	28.2
Samsung	78,612.2	17.8	71,671.8	17.2
LG Electronics	21,014.6	4.8	27,478.7	6.6
Apple	17,295.3	3.9	13,484.4	3.2
ZTE	14,107.8	3.2	7,817.2	1.9
RIM	12,701.1	2.9	12,508.3	3.0
HTC	12,099.9	2.7	6,494.3	1.6
Motorola	11,182.7	2.5	8,691.4	2.1
Huawei Device	10,668.2	2.4	5,478.1	1.3
Sony Ericsson	8,475.9	1.9	10,346.5	2.5
Others	148,990.9	33.8	135,384.1	32.5

Total	440,502.2	100	417,085.7	100

Table 2-6: Worldwide Mobile Device Sales to End Users by Vendor in 3Q11 (Pettey, 2011)

The report by Pettey (2011) states that Samsung became the No. 1 smartphone manufacturer worldwide for the first time as sales to end users tripled year over year to reach 24 million. This according to the report placed Samsung ahead of Nokia in Western Europe and Asia. Pettey (2011) attributes this to the strong performance of Samsung Galaxy smartphones, which in 2011 already covered a broad range of prices, and a weaker competitive market (Pettey, 2011).

Another consideration for platform selection is what customers are looking for and does the choice of platform support the customer applications requirements. Understanding customer requirements also assists developers into developing apps that can be used and not remain idling on the devices. Table 2-7 lists the most popular features that consumers look for in a mobile phone.

Scenario	Description
Game Platform	A platform that can support gaming
Corporate Email Platform	Secure encryption-based client-server connection with automatic email synchronisation with enterprise email servers like Lotus Domino or Exchange, and automatic email "push" to mobile device. RIM offers capability built into the OS; support for iPhone, Windows Mobil, and Symbian available through Sybase iAnywhere Mobile Office, a third party application.
Deliveries Tool	This application refers to mapping ability, inventory control, and connection to headquarters IT. Good platforms include WinMo, Android, and OpenMoko.
Social Networking/ Messaging	Support for social networking and other types of messaging applications. iPhone, WinMo, RIM, Symbian, Android, Moblin and Palm WebOS provide the best experiences for this scenario.
Media Player	Build an application for streaming audio/video. Requires presence of a media player capable of streaming of some kind (MMS/RTSP/HTTP Live streaming).
Third-party applications	This refers to application developed by third parties, that is, not the OS or hardware developers or carrier. Considerations for this are support for developers and size of user base. Good Platforms are WinMo and iPhone.

Table 2-7: Scenarios for Mobile Platforms (Colin Chipman, 2009)

2.7.3. INSTALLED BASE VS. AVAILABLE APPS

There is a large discrepancy between the installed base or total devices shipped of a mobile platform and the corresponding applications. As seen from Figure 2.18, Java ME has the largest installed base (3 billions) but a relatively small number of apps (45,000), while iPhone has an installed base of about 110 millions, but 398,000 apps.

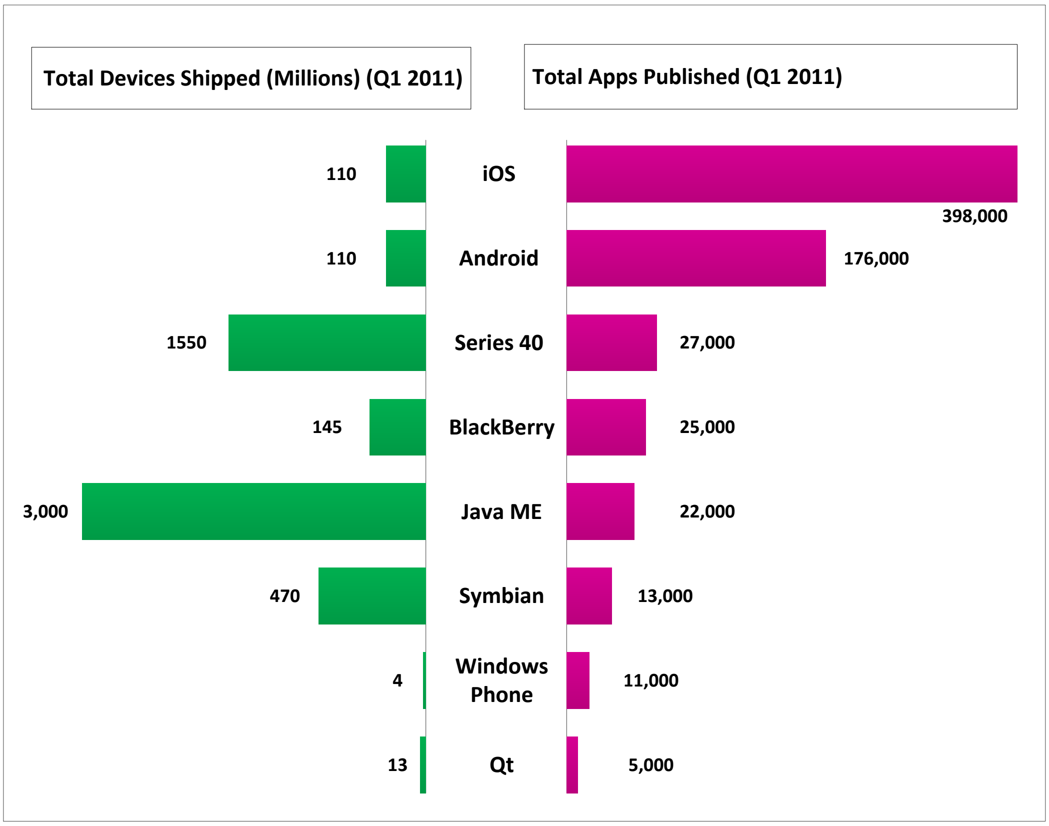

Figure 2.18: Installed base vs. number of Apps (Vision Mobile, 2011)

Figure 2.18 also shows that some earlier mobile platform has penetrated the market base but there is not much interest shown by application developers which is mainly the cause of limitations that comes with the platforms (e.g. developer access to the platform, or easy-to-use-tools). The figure shows that most application developments are taking place on android and iOS platforms.

2.7.4. LEARNING CURVE

Another approach of comparing mobile platforms is to look at what each platform can offer to a developer, by looking at the mobile platform and to distinguish if it's based on open source software, or have an open Application Program Interface (API) or open Software development Kits (SDK). This can further guide the developer in the choice of platform to develop for. Below are two tables that were compiled by a group of students from Carnegie Mellon University as part of their class project for the course "Open Source Software" (Colin Chipman, 2009).

The team identified 10 mobile platforms. Table 2-8 provides a quick comparison of the 10 mobile platforms (Colin Chipman, 2009):

Mobile Platform	Open Source	Open API/SDK	Underlying OS	Documentation	Programming Language	Support	Comments
Symbian	Yes	Yes	Symbian OS	Symbian Developer Network	C++	Symbian developer network	Most popular platform in the world
Android	Yes	Yes	Linux	Android Documentation	C,C++,Java	Development Community	Initially designed for touchscreen mobile devices.
LiMo (Linux Mobile)	Yes	Yes	Linux	API Documentation	Java	Developer Support	Foundation code available to founder and core members of LiMo Foundation
OpenMoko	Yes	Yes	Linux	OpenMoko Wiki	C, Java, Python, Ruby and Mono	OpenMoko Community	Platform is open source hardware OpenMoko
Palm webOS	No	Yes	Linux	API docs	C, C++, Pascal	Support	Latest and most viable threat to iPhone
iPhone	No	Yes	Darwin	Documentation	Ruby, Python, C#, JavaScript, C, C++ etc.	Support	The market leader in smartphones with respect to functionality and growing at a very fast clip
Maemo	Partially	Yes	Linux	Documentation	C, C++ and Mono C#	Developer Support	Developed by Nokia and improved by Maemo community for Smartphones
Qt	Yes	Yes	Embedded Linux, Linux, OS X, MS Windows,	Documentation	C++, Java, Python, Pascal,	Developer Support	Cross-platform application framework
RIM	No	Yes	Blackberry OS	Documentation	C++	Developer Support	Has the largest smartphone market share. iPhone is a

							real threat to RIM.
Windows Mobile	No	Yes (Win32 API	Win 32	Online Documentation	C, C++	Developer Support Site	Declining worldwide smartphone market share (14% in 2009 down from 23% in 2004). HTC makes 80% of all windows Mobile smartphone

Table 2-8: Comparison of Mobile Platforms (Colin Chipman, 2009)

Another interesting comparison was done by DevX (2010) in Table 2-9 which shows the cost factor of developing for a certain platform (DevX, 2010). This means that developers will also select a platform based on how it will enable them to reach the most users most cost-effectively.

Platform	Cost
Android	Free, open source; no upfront fees
Blackberry	~Free access to SDK and simulators ~$20 to sign an app ~$200 account fees for every 10 apps on BlackBerry App World ~Blackberry Alliance Program for ISVs (Independent Software Vendor) ranging from $2000 to $5000
iPhone	~$99 per year for Apple iPhone Developer Program ~Most developer tools are free, open source, others may vary from $99 to $400 per year
Windows Phone 7	~Prices not finalised yet ~Free SDK, most probably free development tool ~Windows marketplace for Mobile developer subscription -$99/year
Palm WebOS	For a limited time (as of 7/22/2010), HP Waiving $99 annual fee for developer account
Symbian	A few hundred dollars

Table 2-9: Mobile Development Platform Comparison Matrix (DevX, 2010)

The platforms have different learning curves: According to Figure 2.19: Average time required to master each platform (Mobile, Developer Economics 2010, 2010) Android has the shortest with 6 months, while Symbian has the longest with 15 months.

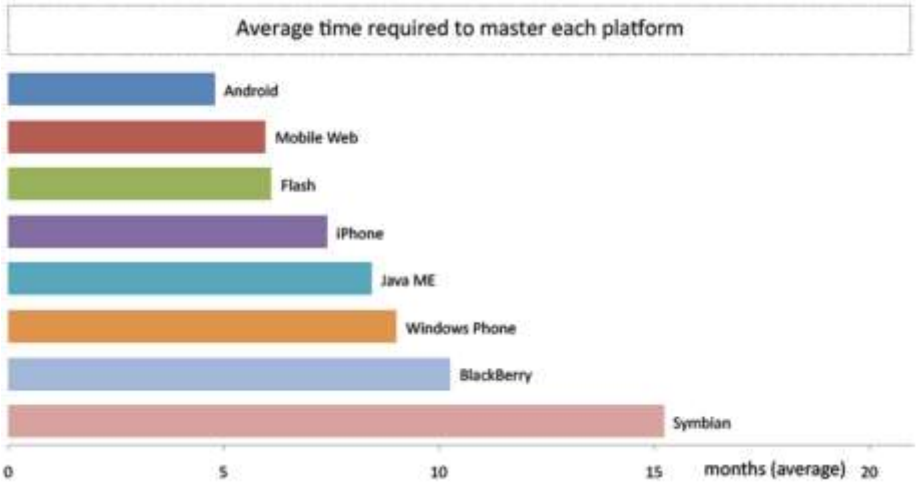

Figure 2.19: Average time required to master each platform (Mobile, Developer Economics 2010, 2010)

With reference to Figure 2.20, debugging is faster on Android and more than two times slower on Symbian. With the main factor "Learning curve" and the availability of documentation, Android seems to be the easiest platform to develop for.

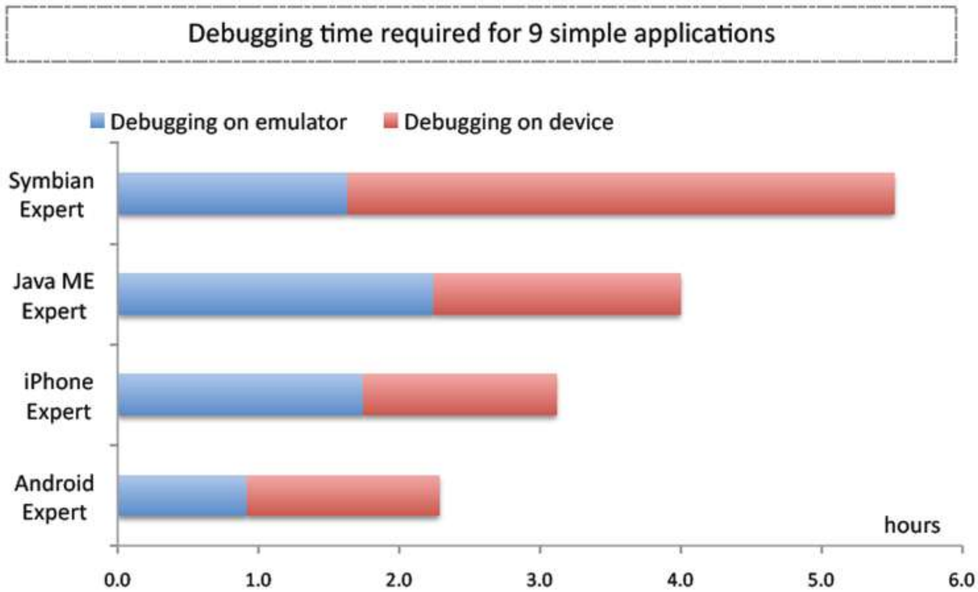

Figure 2.20: Debugging time required for 9 simple applications (Mobile, Developer Economics 2010, 2010)

2.7.5. *GO-TO-MARKET*

Vision Mobile (2012) reports that the most notable trend across 2011-2012 is the demotion of Telcos as distribution channels for mobile applications. As per the report this was evident by the 47% decrease in Telco portals usage by developers of different platforms. The report also adds that even though Telcos have lost control of service distribution, there is still one exception: China Mobile's app store which has seen a tremendous growth. China Mobile's app store has enrolled 22% of its subscribers, and has served over 600 million downloads taking advantage of the absence of Google Play in China (Mobile, Developers Ecomomic, 2012).

As per Vision Mobile (2012) report, App stores are constantly taking over the business of app distribution from traditional mobile distribution channels such as Telco portals and 3rd party aggregators. The report was based on information received from a Developers Economic survey that was undertaken by More than half of the 1,500+ developers, irrespective of platform who stated that they used an app store as their main channel. In other words, more developers use app stores as their primary channel than use all other channels combined – regardless of their primary platform (Mobile, Developers Ecomomic, 2012).

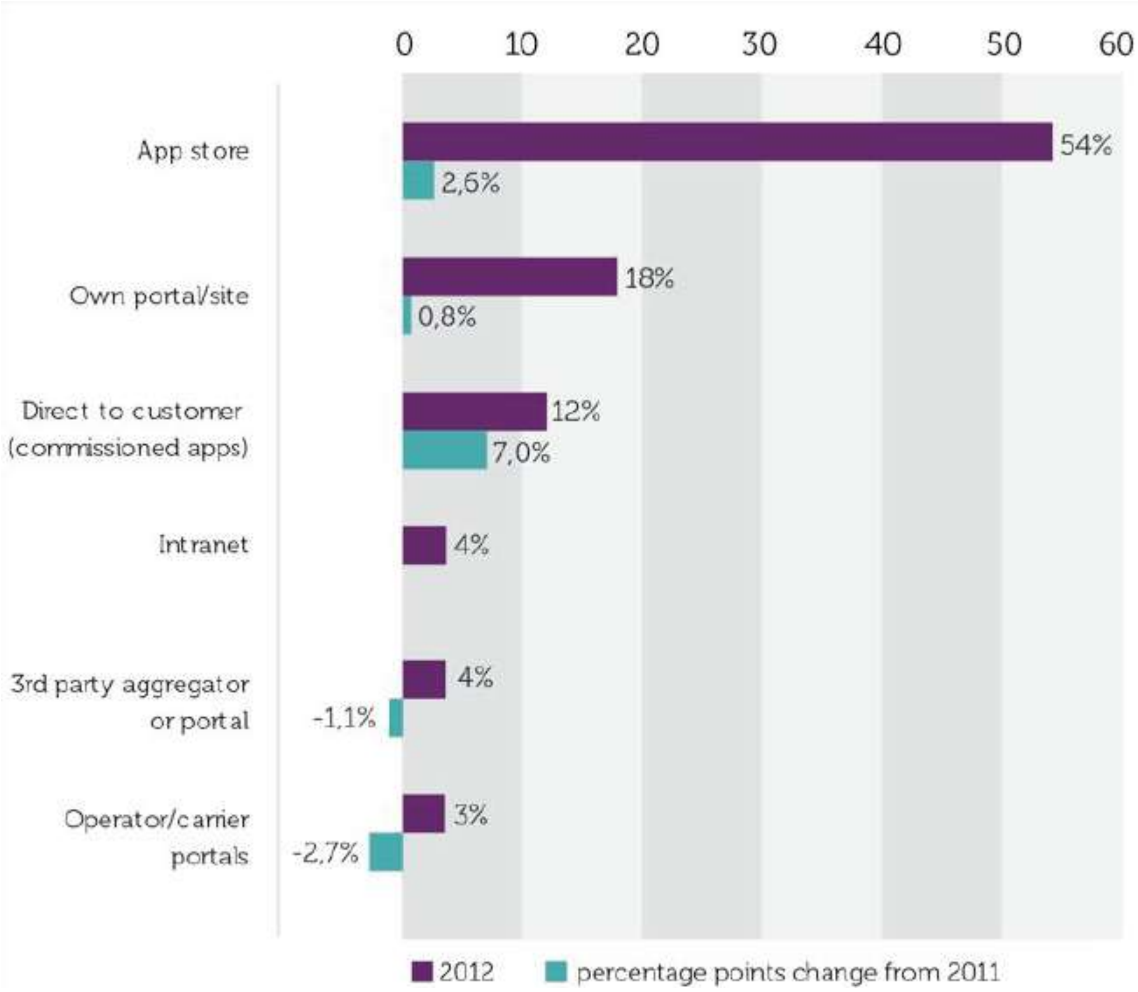

Figure 2.21: Mobile Apps distributions channels (Mobile, Developers Ecomomic, 2012)

As shown in Figure 2.21 (Mobile, Developers Ecomomic, 2012), it's clear that apps stores are real taking over as the preferred distribution channels compared to own web sites. There has also been a significant growth in the "direct to customer" channel, this as Figure 2.21 shows was due to the fact that more customised applications are being developed (Mobile, Developers Ecomomic, 2012). Figure 2.22 shows how popular the different platforms are across the top three distribution channels.

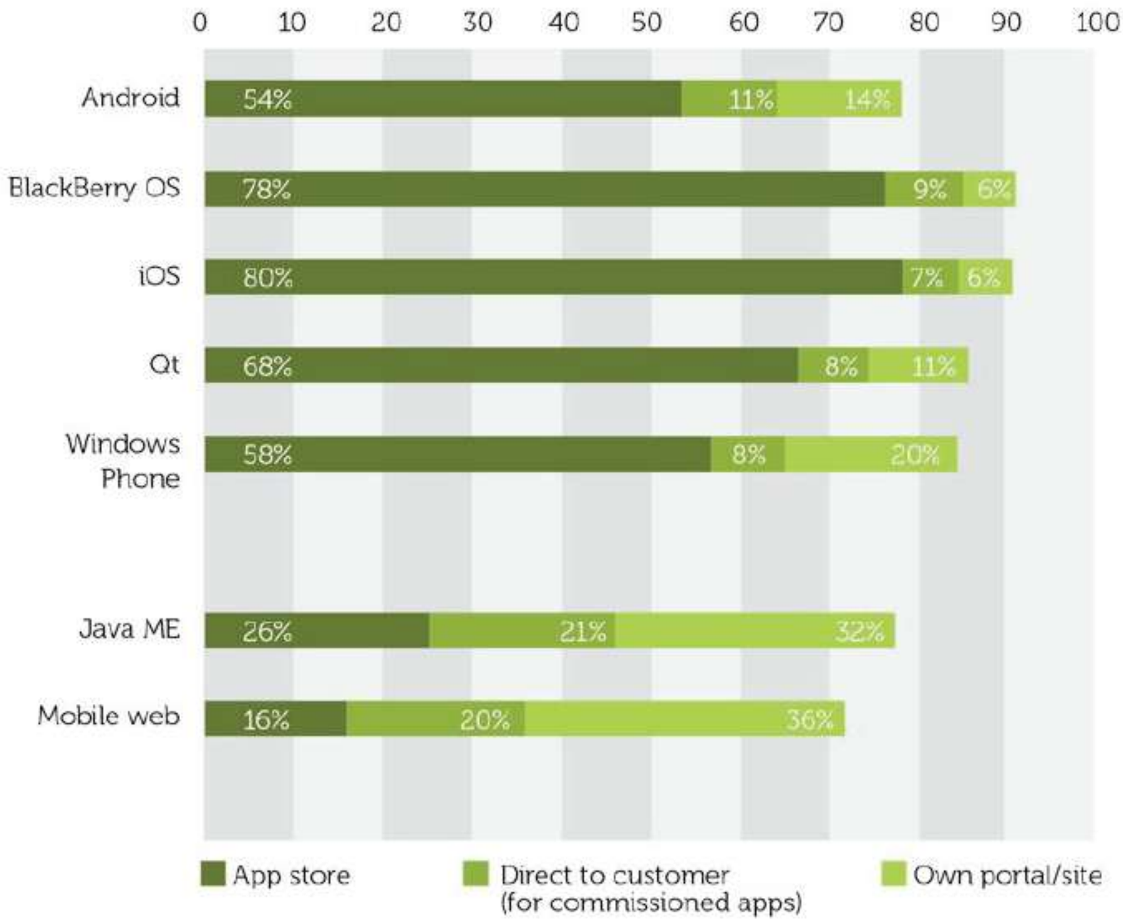

Figure 2.22: Platform popularity among apps stores (Mobile, Developers Ecomomic, 2012)

With reference to Figure 2.22, the Vision Mobile (2012) report clearly indicates shows that App store use is much more popular among iOS (80%) and Blackberry developers (78%), compared to developers of other platforms such as Java ME and Mobile Web. Android and Windows Phone developers also use app stores: 54% and 58%, respectively (Mobile, Developers Ecomomic, 2012).

According to a report by Sarah (2010), the preferred distribution channel is app stores which include Android Market, Nokia's Ovi Store, Microsoft's Windows Phone Marketplace and others. According to Sarah there is another distribution channel called app recommendation sites, which is excelling in pushing downloads (Sarah, 2010). Some of the app recommended sites are listed in Table 2-10 (Sarah, 2010):

Website	Comment
Appoke	A combo Android app store and social network
AppStoreHQ	A semantic search engine
Frenzapp	A cross-platform app and recommendation engine that looks at what your Facebook friends like
Sidebar	A recommendations platform for developers
Chorus	iPhone apps and recommendations platforms
Appsaurus	iPhone apps and recommendations platforms
Appitalism	A social network and app store for different platforms

Table 2-10: App recommended sites (Sarah, 2010)

According to Sarah (2010), there are also other methods that developers can use to get an application out to the customers. Below are three methods that can be used as mentioned by the report (Sarah, 2010):

- **Optimise search discovery and the channels you own** – Developers should optimise their websites through traditional search engines and then convert websites visitors to application downloaders. For example GetJar which is a download applications for mobile games and applications, can be pasted onto social network sites (e.g. Facebook page or twitter account) to point visitors to the hosted app at GetJar;
- **Specialist Press** – Have the application reviewed by Press, especially those who review apps exclusively, because news which begins on the Web spread fast through social channels such as Facebook;
- **Ad Networks –** Advertising networks can be useful for driving traffic to a developers website.

This review assists in answering the question: “Which factors were considered while deciding on a development platform?”. By considering these factors, the development time of an application can be reduced and will also prevent a developer by starting off with a project and realising while in the middle of the project that he is coding for the wrong platform. These factors or guidelines have assisted the research to easily narrow down to a single platform of choice, while considering the time it will take to understand the development platform and the size of the installed base for the platform.

2.8. OPEN SOURCE SYSTEMS

2.8.1. OPEN SOURCE VOIP SOFTWARE

The Section explores the top five open source VoIP backend software and their features. The VoIP Software systems explored are as follows:

- Asterisk
- SipX
- Elastix
- FreeSwitch
- Callweaver

These software packages are provided at no cost and enjoy support from users who provide product support. The choice of which VoIP application to use depends highly on the availability of resources (e.g. documentations, feature sets and stable releases).

2.8.1.1. Asterisk

Asterisk website (2012) defines the Asterisk platform as an open source PBX system which performs all the functions of a traditional telephone system. According to the website, with just Asterisk telephony software, an ordinary computer can be turned in to a VoIP Server which can be used by small and large businesses to integrate with analog phones and most standards-based IP telephone handsets and software. Asterix provides old and new features as shown in Appendix C (Asterisk feature list) and also help reduce operational and capital expenditures (Asterisk, 2012). According to the VoIP News report (2008), asterisk is the dominant open-source PBX solution on which many others are based (VoIP, 2008).

According to the document by Meggelen, Van Madsen and Smith (2007) Asterisk supports a wide range of protocols for the controlling and transmission of voice over traditional telephony interfaces including H.323, SIP, MGCP and Skinny Client Control Protocol (SCCP) (Meggelen, Van Madsen, & Smith, 2007).

Some of Asterisk features (Asterisk, 2012) are:

- voice mail;
- conference calling;
- interactive voice response (phone menus);
- and automatic call distribution.

Asterisk on a Windows PC has the following requirements (Asterisk@Home Handbook, 2005):

- Windows 2000/XP/2003/7
- Pentium 500MHz or above with at least 128MB RAM

Even though the review provides the lowest specifications per hardware system, it should be noted that a faster system will still support more simultaneous calls because the calls must still be processed.

Supported platforms for Asterisk are (Johanson, 2009):

- Linux
    - Linux Kernel 2.4: Debian, Fedora, Gentoo, Mandrake, Redhat, Redhat Enterprise clones
    - Linux Kernel 2.6: Gentoo, Debian, Fedora, Slackware, SuSE, CentOS, Mandrake
- Non-Linux: FreeBSD, NetBSD, OpenBSD, Mac OS X, Solaris, Windows, Windows Cygwin

2.8.1.2. SipX

Calivia (2012) defines SIPX as an open source voice over IP PBX software with voice mail and auto-attendant. According to Calivia feature of SipX includes voice mail, interactive voice response system and auto attendants but the main feature is the software implementation of SIP that makes an IP based communication system. The design of SipX deviates from the traditional Asterisk as it does not require any additional hardware as it can interoperate with any SIP compliant gateway, phone or application (Calivia, 2012).

Supported Platforms for SipX are (mlctrez, 2011):

- Linux distribution mostly used – Fedora Core or CentOS/RHEL;
- Open SuSE RPMs are built regularly;
- Other Linux distributions such as FreeBSD;
- 32 bit and 64 bit operating systems (release 3.11 and greater).

SipX features (Niculae, 2011)are:

- SIP Session Router, optionally geo-redundant and load sharing;
- Media server for unified messaging and IVR (auto-attendant) services;
- Conferencing server based on FreeSwitch;
- XMPP Instant Messaging (IM) and presence server (based on Openfire);
- Contact center (ACD) server;
- Call park / Music on Hold (MoH) server;
- Presence server (Broadsoft and IETF compliant resource list server for BLF);
- New: Shared Appearance Agent server to support shared lines (BLA);
- Group paging server;
- SIP trunking server (media anchoring and B2BUA for SIP trunking & remote worker support);
- Call Detail Record (CDR) collection & processing server;
- Third party call control (3PCC) server using REST interfaces;
- Management and configuration server;
- Process management server for centralized cluster management.

2.8.1.3. Elastix

Edgar (2012) defines Elastix as an open source Unified Communications Server software that is offered with IP PBX, email, faxing and collaboration functionality and supports SIP and IAX (Inter-Asterisk Exchange) (Edgar, 2012).

Elastix features (Edgar, 2012) are:

- Voicemail;
- Fax-to-email;
- Support for softphones;
- Web Interface Configuration;
- Virtual conference rooms;
- Call recording;
- Least Cost Routing;
- Extension Roaming;
- PBX Interconnection;

- Caller ID;
- CRM;
- Advance Reports.

Elastix Hardware Requirements (Elastix, 2008):

- Pentium IV, Opteron, or faster is recommended;
- 512MB memory or greater;
- Ethernet Interface.

Elastix supported OS platforms are:

- Linux: CentOS;
- Windows XP.

2.8.1.4. FreeSwitch

Collins (2012) defines FreeSwitch as a scalable open source cross-platform telephony platform designed to route and interconnects popular communication protocols such as SIP and H.323 using audio, video, text or any other form of media. He adds that FreeSwitch offers support to various stable telephony platform on which many telephony applications can be developed using a wide range of free tools. It was originally designed and implemented by focus on several design goals (collins, 2012) such as:

- Modularity;
- Cross-platform support;
- Scalability;
- Stability.

According to Collins (2012) FreeSwitch offers many advanced SIP features such as presence, Busy Lamp Field and shared line appearance as well as TCP Transport Layer Security and sRTP $secure Real Time Protocol. Collins further states that FreeSwitch can also be used as a transparent proxy with and without media in the path to act as a SBC (session border controller) and proxy T.38 for fax transmission and other end to end protocols (collins, 2012) such as:

- H.323
- SIP

FreeSwitch supported OS Platform (collins, 2012) are:

- Builds native on Windows;
- Builds on Mac OS X, Linux, Solaris and BSD.

Minimum/Recommended System Requirements (collins, 2012) are:

- 32-bit OS (64-bit recommended)
- 512MB RAM (1GB recommended)
- 50MB of Disk Space

2.8.1.5. CallWeaver (OpenPBX)

CallWeaver is a community-driven, vendor-independent, cross-platform, open source, PBX software project (formerly known as OpenPBX.org). It was originally derived from Asterisk (Boned, 2010).

The features include (Boned, 2010) are:

- Modular architecture;
- Cross-platform (Linux, FreeBSD, NetBSD, OpenBSD, MacOS X/Darwin, Open/Solaris);
- PSTN Connectivities (FXS/FXO, ISDN, PRI, E1, T1);
- Native support for Sangoma TDM cards;
- Integrated Faxing and Messaging;
- Supports (william, 2012):
    - analog and digital PSTN telephony;
    - Multi-protocol VoIP Telephony (h.323, IAX2, MGCP, SIP and SCCP);
    - Fax and Software Fax;
    - STUN for SIP communications;
    - T.38 fax over IP;
    - IVR;
    - Conferencing;
    - Callcenter queue management.

The CallWeaver supported platforms (Boned, 2010) are:

- ucLinux;

- Linux;
- FreeBSD;
- NetBSD;
- OpenBSD;
- MacOS X/Darwin;
- Solaris 10;
- HP-UX.
- OpenVMS

2.8.1.6. CONCLUSION

Table 2-11 shows the criteria's used to compare the Open Source VoIP Softwares.

Criteria	Asterisk	SipX	Elastix	FreeSwitch	CallWeaver
Open Source	Yes	Yes	Yes	Yes	Yes
Supported Protocols	Supported telephony protocols such as SIP, H.323& MGCP	Support SIP	SIP and IAX	H.323, SIP	Support SS7, SIP, ISDN, H.323, SIP, MGCP, IAX2,
Supported Platforms	**Linux Kernel**: Gentoo, Debian, Fedora, Slackware, SuSE, CentOS, Mandrake **Non-Linux**: FreeBSD, NetBSD, OpenBSD, Mac OS X, Solaris, Windows, Windows Cygwin	Linux distribution mostly used – Fedora Core or CentOS/RHEL; Open SuSE RPMs are built regularly; Other Linux distributions such as FreeBSD; 32 bit and 64 bit operating systems (release 3.11 and greater).	Linux: CentOS; Windows XP.	Builds native on Windows; Builds on Mac OS X, Linux, Solaris and BSD.	Linux, FreeBSD, NetBSD, OpenBSD, MacOS X/Darwin, Open/Solaris

Features	Old and new features including: voice mail, conference calling, Interactive voice response and automatic call distribution.	Unified messaging and IVR (auto-attendant) services; Conferencing; Instant Messaging (IM) and presence server; Contact center ; Call park / Music on Hold (MoH); Group paging; Call Detail Record (CDR) collection & processing; Third party call control; Management and configuration;	Voicemail; Fax-to-email; Support for softphones; Web Interface Configuration; Virtual conference rooms; Call recording; Least Cost Routing; Extension Roaming; PBX Interconnection; Caller ID; CRM; Advance Reports.	Integrated Faxing and Messaging; IVR; Conferencing; Callcenter queue management.

Table 2-11: A Comparison Chart of Open Source VoIP Software

With reference to Table 2-11, this section highlights some of the open source VoIP systems that are freely available for deployment. Emphasis on which one to deploy on the project can be based on the availability of support documentation, the operating system that it can be implemented on and the features that the system can support. Most of the open source VoIP systems highlighted here can be used to implement the VoIP network but by highlighting the features can assist in future modifications. The developer can emulate features of all the different systems on a single one.

Also looking back at the five VoIP platforms, the factors which were analysed mostly are:

- Features
- Supported OS platforms

All the five VoIP platforms are able to support the known OS platforms such as Windows and Linux. And all the five VoIP platforms has a very long feature list of which the main features are highlighted in the Comparison Chart (Table 2-11), but as it was stated by Asterisk website (Asterisk, 2012), Asterisk is able to support new and old features, making it the best replacement for a traditional PSTN based system. This is because a user can keep the legacy services and still take advantage of the newer VoIP features.

2.8.2. *WI-FI VOIP NETWORKS*

2.8.2.1. A Wi-Fi/VoIP Network in Laos

According to Eli (2011), one of the clearest cases of a cooperatively owned community network using new technologies was developed in Laos in the Hin Heup district. Eli adds that this solution was based on a bottom-up development principles which involves putting systems together to come up with a grand system rather than a traditional approach which are based on sequential steps such as requirement definition, solution building, testing and deployment (Eli, 2011).

Eli (2011) describes that the solution was implemented by the Jhai[10] foundation and the local people of Hin Heup district, which comprise of five small villages within a radius of a few kilometres and is home to 425 people. Eli adds that before the project, the villagers have requested for telecommunications services but to no luck. But as Eli further states, the Jhai foundation with the cooperation from the villagers was able to design and test a Wi-Fi and VoIP based solution. Eli adds, that the villagers also prepaid a detailed business plan based on parameters such as: a mix of voluntary and paid labour, the use of community resources, services selected by the community and tariff levels. The plan demonstrated that the service inclusive of internet use, local, national and international calls was indeed financially feasible (Eli, 2011).

2.8.2.2. Android Wi-Fi VoIP

According to Davis (2009), there's already a Skype client available in the Android market, but it doesn't offer true VoIP because it makes use of a client's minutes and data plan. Davis therefore adds that if customers would like to make use VoIP over Wi-Fi, it's better to wait until the Sipdroid project is completed. Davis explains that SIPdroid uses a Wi-Fi connection to make VoIP calls between devices and as seen in Figure 2.23 the second phone is called from SIPdroid running on the G1. The project started off as open source but has now been moved to close development (Davis, 2009).

[10] Is a group of consultants who are involved in development training especially for rural areas.

Figure 2.23: SIPdroid call using Wi-Fi (Davis, 2009)

Davis (2009) warns that besides the excitement, SIPdroid will fall victim to the same distribution agreement terms & conditions as tethering[11] apps have. Davis explains that just as Google pulled tethering software from the Android Market because it breaches T-Mobile's Term of Service, most carriers also have non-VoIP clauses in their contracts which could be enough reason for Google to pull SIPdroid off the shelves (Davis, 2009).

2.8.2.3. *Skype Offers Free Wi-Fi*

Maisto (2011) reports that Skype announced at the Mobile world congress show in Barcelona in 2011 that it has launched a partner program for Wi-Fi hotspot operators around the globe, including BT Openzone, Fon, Tomizone, Row 44 and Vex extending the number of hotspots that users can connect to using Skype Credit. Maisto adds that due to the partnerships, Skype users now access internet in more than 500,000 locations including airports, 30,000 hotels and a number of and a number of trains, planes, offices, cafes and convention centres (Maisto, 2011).

Maisto (2011) also adds that Skype also introduced a mobile partner program for mobile operators in markets with low 3G broadband penetration, which allows Skype to be extended to their customers. Maisto further explains that a client/server solution was implemented which lets users make a free

[11] Tethering turns your mobile phone into a USB Internet access point for your PC.

Skype-to-Skype calls or lower billed Skype-to-mobile and Skype-to-landline calls on more than 100 smartphones and a variety of operating systems (Maisto, 2011).

Eddy (2011) backs up Maisto' s report by stating that Skype, launched an application on Apple's app Store that lets users access Wi-Fi hotspots around the world and pay with Skype credit. Eddy adds that this application was intended for travellers abroad and constant internet users. The free app's connection rates start at $0.06 cents a minute and is available for the iPhone, iPad or iPod Touch with the Skype service accessible via more than 1 million Wi-Fi hotspots around the world, including hotels, airports, train stations, convention centres, bars and restaurants (Eddy, 2011).

Bouton (2010) adds that Skype made its services available to HTC and Motorola Smartphones running Google's android platform version 2.1 or later. Bouton further informs that users with Skype accounts who own such handsets as the Motorola droid, HTC droid incredible and HTC Evo 4G can now make free Skype-to-Skype calls and makes use of additional services such as instant messages from their phones to users' computers or phones. Skype doesn't guarantee the app will work outside of HTC and Motorola devices with Android 2.1 and Android 2.2 and later (Bouton, 2010).

2.8.2.4. CONCLUSION

The section highlights some of the (mobile) VoIP over Wi-Fi deployments used in the world. What is important in this section to the research is the types of deployments that VoIP is used for. This can also help the researcher learn from other projects on what different projects the mVoIP can be deployed on. As highlighted, some countries are deploying inexpensive VoIP for their rural communities. This concept can also be extended to our rural populations in order to help enhance communications services in the country.

2.8.3. MOBILE VOIP APPLICATIONS

2.8.3.1. MobileVOIP

According to MobiWeb (2012), MobileVOIP is a free calling application which lets users make free national or international mobile VoIP calls directly from their iPhone. MobiWeb explains that the free VoIP calls are made over Internet connections such as 3G, Wi-Fi, GPRS, EDGE, and UMTS. The application is easy to use and all that a user has to do is simply select any contact from their contact list, which is accessible in the app, and start using MobileVOIP anywhere, anytime over an Internet connection (MobiWeb, 2012).

Figure 2.24: MobileVOIP Screenshots (MobiWeb, 2012)

Some of the MobileVOIP application features are as follows (MobiWeb, 2012):

- Make Free calls to a selection of destinations using an iPhone;
- Send cheap text messages (SMS) worldwide;
- Make cheap calls to any international destination;
- Works all around the globe anytime, anywhere;
- Bypass VoIP blockades with MobileVOIP.

2.8.3.2. TiviPhone's

According to the TiVi website (2012), TiviPhone's mobile version is a VoIP application that works on wireless data connections such as 3G, Wi-Fi, UMTS and GPRS. It provides multimedia services such as VoIP and real-time video streaming for smartphone users. If you have a multi-mode telephone or PDA, TiviPhone will be able to automatically choose among several connectivity options (GSM coverage versus your local wireless LAN, which is also called backward/forward compatibility) (TiVi, 2012).

According to the TiVi website (2012) TiviPhone provides features such as voicemail, IM/text messaging, phonebook management. Besides updating and enriching the TiviPhone freeware, TiviPhone's programming team continues to create customised end-to-end VoIP solutions for customers ranging from end users to service providers in different countries around the world (TiVi, 2012).

TiviPhone mobile VoIP software enables (TiVi, 2012) the following features:

- Superior quality outbound and inbound calls with mobile and fixed telephone subscribers and PC users;
- Calling to PSTN or CS, SIP buddy or direct IP call;
- Free voice and video IP calls;
- Chat;
- Creating, sharing and maintaining of the user's phone book;
- Access to computer- based voicemail box via a Web Browser;
- Seamless roaming between Wi-Fi, GPRS(UMTS), GSM and 3G;
- Sending DTMF (full support, incl. SIP INFO DTMF and RFC 2833);
- Instant/automated redialling;
- Access to the account balance information.
- Is available on Symbian OS and Windows Mobile

2.8.3.3. Fring

Timor (2012) defines Fring as a mobile communication service that enables free mobile multi-conference video calls, one-to-one video calls, voice calls and instant messaging. The application is available on all major smartphones (including iPhone, iPod touch, Android and Nokia), on any mobile operator, and any wireless network (3G/4G, Wi-Fi, GPRS, EDGE) (Timor, 2012).

Figure 2.25: Fring application logo (Noah, 2008)

Timor (2012) adds that Fring has a real time contact availability feature which allows the user to see another Fring user's online status such as idle, busy or logged off before the call can be made. Fring was the first to bring mobile VoIP, video and multimedia conference over IP and it adds more than a million new Smartphones users every month (Timor, 2012).

Figure 2.26: Fring Network Architecture (Timor, 2012)

2.8.3.4. Talkonaut

According to the Talkonaut website (2010), Talkonaut is free mobile VoIP application for mobile phones which is available for many different mobile platforms: Android, Symbian S60, Windows Mobile 5/6, J2ME with MIDP-2.0, Google Android, iPhone and iPod touch. Talkonaut supports presence, instant messaging and also allows to make free VoIP calls over GPRS/EDGE,3G, or WI-FI (talkonaut, 2010).

Talkonaut Features (Talkonaut, 2010) are:

- Instant Message Chat;
- Group chat;
- Presence and status;
- Cheap VoIP calling using callback;
- File Sharing;
- Traffic Compression;
- SIP support;

- GPRS/WIFI auto-switching.

2.8.3.5. TringMe

According to the tringme website (2011), TringMe is defines as a converged solution for calls, SMS and scheduling conferences from anywhere in the world to anywhere in the world. Tringme can be used on a web-browser with any additional downloads, on a mobile or from instant messenger services such Gtalk (tringme, 2011).

Figure 2.27: TringMe Web-based calling (tringme, 2011)

TringMe features (tringme, 2011) are:

- Call, SMS & Conference worldwide from
  - Any web-browser
  - Mobile Phones
  - Instant Messengers (e.g. Google Talk)
  - SIP devices
- Free Peer-to-Peer VoIP calls over Wi-Fi/3G networks on
  - Blackberry
  - Android
  - Windows Phone 7
  - iPhone

- Symbian

2.8.3.6. CONCLUSION

Table 2-12 summarises the criteria used to compare the five mobile VoIP applications

Criteria	MobileVOIP	TiviPhone	Fring	Talkonaut	TringMe
Mobile VoIP Offering	Free	Free	Free	Free	Free
platform Support	iOS	Symbian, Windows Mobile	iOS, Symbian and Android	Android, Symbian and Windows Mobile	Blackberry, Android, Windows Phone, iOS and Symbian
VoIP Features	VoIP & SMS	VoIP, SMS, Voice Mail & Instant Messaging	VoIP, Video and Conference	VoIP, Instant Message, Presence and Group Chat	VoIP, SMS, Conference and Instant messaging
Wireless Connectivity	3G, Wi-Fi, GPRS & UMTS	3G, Wi-Fi, GPRS, GSM & UMTS	Wi-Fi, 3G and UMTS	GPRS, EDGE, 3G and Wi-Fi	Wi-Fi/3G

Table 2-12: Comparison Chart of Mobile Applications

The main highlights of this section are:

- **Free Mobile VoIP** -- All the mobile VoIP applications have one significant similarity which is enabling free calling over broadband wireless networks. This is important to mobile VoIP implementation on the user side because the trend has already been set by many service providers on how VoIP should be offered. But it should still be noted that there might be charges on international calls due to interconnect charges between operators. This criteria is therefore not real influenced by the time of application used but by the service provider and therefore in this category all the applications are at the same level.
- **Platform Support** -- most of these applications are able to operate on more than one OS platforms with the exception of Mobile VoIP. For a developer this is the main consideration

for any mobile application development project. Developers should target as many platforms as possible. A cross-platform development will reach more customers which will mean higher revenues for commercial application and free applications such as the ones used in communication can also help boost traffic revenues. TringMe is the leader in this category since it targets the most mobile platforms.

- **Key VoIP features** – All applications reviewed provides VoIP and key features such as Instant Messaging and SMS.

Therefore the way that the applications above were implemented can serve as a guideline to newer application developments.

2.9. EVALUATION OF COMMERCIAL VOICE SYSTEMS

2.9.1. SIEMENS EWSD

According to Siemens (2002), the electronic digital switching system is one of the most widely installed telephone exchange systems in the world. Siemens further describe that the EWSD (Electronic Worldwide Switch Digital) system can work as a local or an international gateway or the combination of both and it can be used to provide fixed and mobile phone services. Siemens, the main supplier of Siemens AG, claims that EWSD switches perform switching of over 160 million subscriber lines in more than 100 countries (Siemens, 2002).

Siemens (2002), defines the EWSD software as an APS (Application Program System) which is installed on a hard drive and includes the operating system developed by Siemens in cooperation with Bosch. The APS software is system specific and is responsible for functions such as traffic management, path search, and call charging (Siemens, 2002).

Seven EWSD systems are deployed to date in Namibia and serves up to 120, 000 residential subscribers. Figure 2.28 shows the EWSD system architecture.

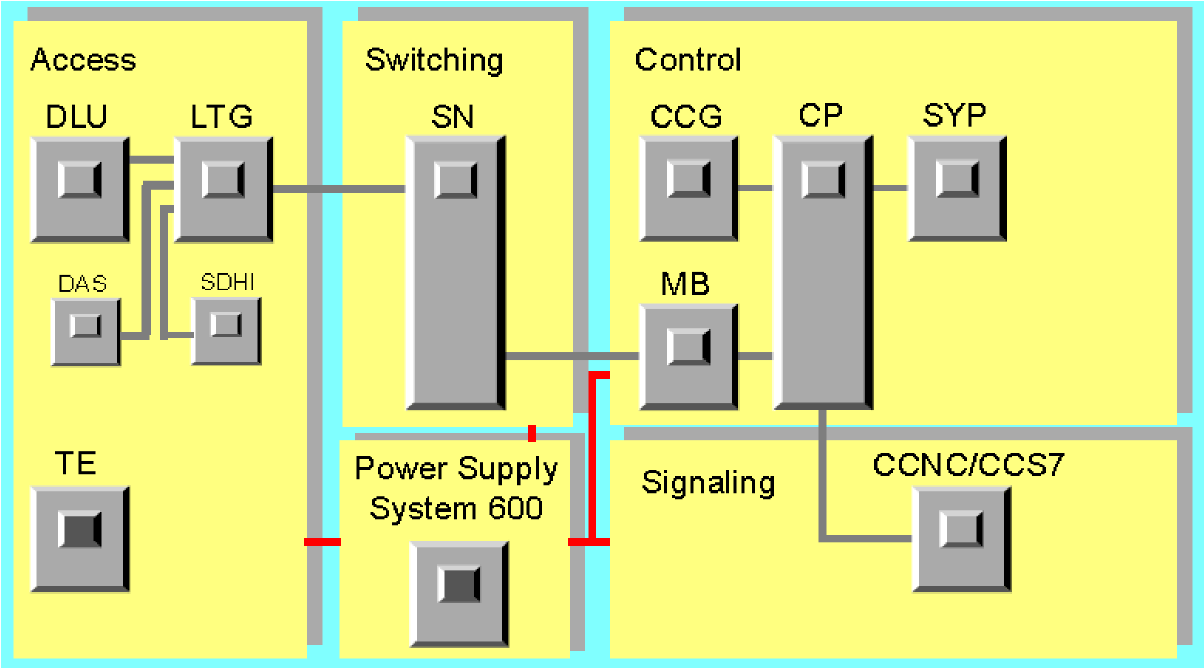

Figure 2.28: EWSD Block Diagram (picture from EWSD V.11 Edoc)

The Main EWSD Subsystems are:

- **Access:** LTG's (Line Trunk Groups) and DLU's (Digital Line Unit) Provides access for:
  - Subscribers Lines (Analog, ISDN BA (Basic Access), ISDN PA (Primary Access)
  - Trunk Lines (E1's) for interconnection
- **Switching:** the Switching Network (SN) is responsible for routing and switching of calls
- **Control:** Responsible for call Processing (CP), alarms on System Panel (SYP), Central Clock generation (CCG) & Call data Records (CDR)
- **Signalling:** The CCNC (Common Channel Network Controller) is responsible for the control of all interconnect signalling messages in the Signalling System Number 7 (SS7) network.

PERCEIVED ADVANTAGES OF THE EWSD ARE (Siemens, 2002)**:**

- **Very stable system:** only two total systems failures recorded to date during the 15 years of deployment in Namibia;
- **High Availability of Skilled Labour:** There are knowledgeable personnel deployed in every EWSD location in Namibia;
- **Support & Documentation:** Documentation exists for every failure code and national support is also available.

PERCEIVED DISATVANTAGES OF THE EWSD ARE (Siemens, 2002)**:**

- **Size:** EWSD is a very huge systems – a EWSD serving 10000 lines can take up 35 square meters of floor space, which also uses a lot of power;
- **EOL:** Many sub-systems and Software of the EWSD has already reached end of life and can only receive support from the supplier on Life Extender basis;
- **Cost:** These are very expensive systems a software upgrade from one SW version to another can cost up to 40 Million Namibian Dollars;
- **SLAs:** Yearly Service Level Agreements are also costly;.
- **Spares:** Due to the obsoleteness of the system, it sometimes becomes difficult to source spares;
- **Outdated Interfaces:** The EWSD makes use of TDM (Time division Multiplexing) & X.25 interfaces which is not compatible with newer IP based interfaces. Therefore any interface requirement can only be satisfied by deploying protocol converters or gateways.

2.9.2. SIEMENS SURPASS HIE 9200

Nokia Siemens (2007) describes its Surpass hiE9200 product as a media gateway control function (MGCF) or Softswitch which fits perfectly well into the ETSI (European telecommunications Standard Institute) Tispan architecture applied in Fixed Mobile Convergence (FMC) Networks. Nokia Siemens states that the hiE 9200 Softswitch is a solution that can work both in TDM and next generation network. The surpass offers all the necessary interfaces and features to connect PSTN and NGN subscriber/trunks, which makes it easy to migrate from old PSTN networks to new generations networks (Nokia Siemens, 2007).

One Surpass hiE9200 is currently deployed into Telecom Namibia's Network. Figure 2.29 shows the hiE9200 network architecture.

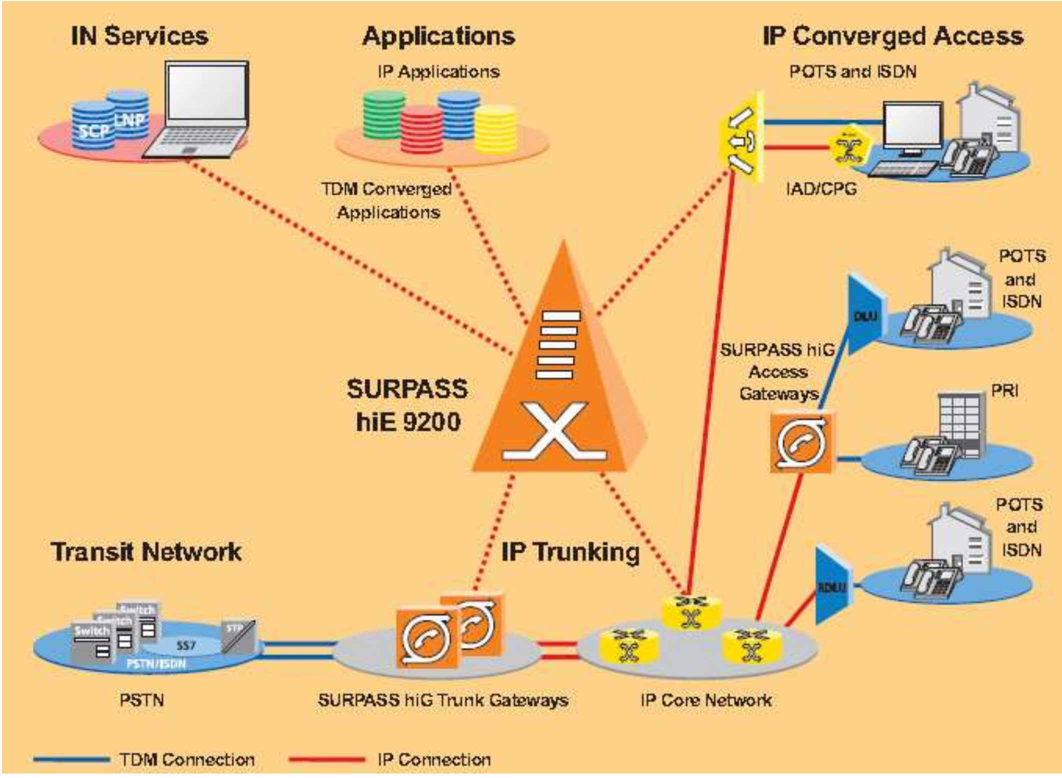

Figure 2.29: hiE 9200 Network Architecture (Nokia Siemens, 2007)

ADVANTAGES OF THE HIE9200 ARE (Nokia Siemens, 2007)**:**

- **Space:** Takes up little floor space – up to 80% space savings can be achieved by replacing the EWSD network with the Surpass hiE9200;
- **Protocol Compatibility:** It supports old and new protocols which make it easier to interface to most systems (e.g. 3rd party systems);
- **Documentation:** Operations and Maintenance manuals exists for the Surpass hiE9200.

DISADVANTAGES OF THE HIE9200 ARE (Nokia Siemens, 2007):

- **Skilled Labour:** There is a shortage of skills for these types of networks and Service Providers mostly depend on the supplier for operations and maintenance. SLA charges remain high;
- **Training:** Training on the system is also very expensive and can cost up to one hundred thousand Namibian dollars per individual;
- **Cost:** It costs Millions of dollars to implement such a system;
- **Spares:** Spares cannot be sourced locally and therefore is very expensive.

2.9.3. *HUAWEI U-SYS SOFTX3000*

Huawei (2003) describes the Softx3000 as a Softswitch which can be used on the Network control layer of NGN (Next Generations Networks). The Softx3000 implements call control, gateway control, connection management of voice, and multimedia services such as conferencing based on the IP network (Huawei, Product Catalogue For Fixed Network Solutions, 2003):

The Softx3000 can offer support for (Huawei, Product Catalogue For Fixed Network Solutions, 2003):

- Traditional PSTN signalling, such as SS7, R2, DSS1 and V5. It can be a voice office end office, tandem office or toll office;
- Black and white lists, call authentication, call interception and so on;
- MTP and M3UA, can be an integrated signalling gateway;
- INAP and INAP+, so it can be used as an SSP or IPSSP in IN system;
- H.323 protocol and can function as a Gatekeeper (GK) in the traditional Voice over IP (VoIP Network).

One such U-SYS system is currently deployed in the Telecom Namibia Network as a test system. Figure 2.30 shows the U-Sys network architecture.

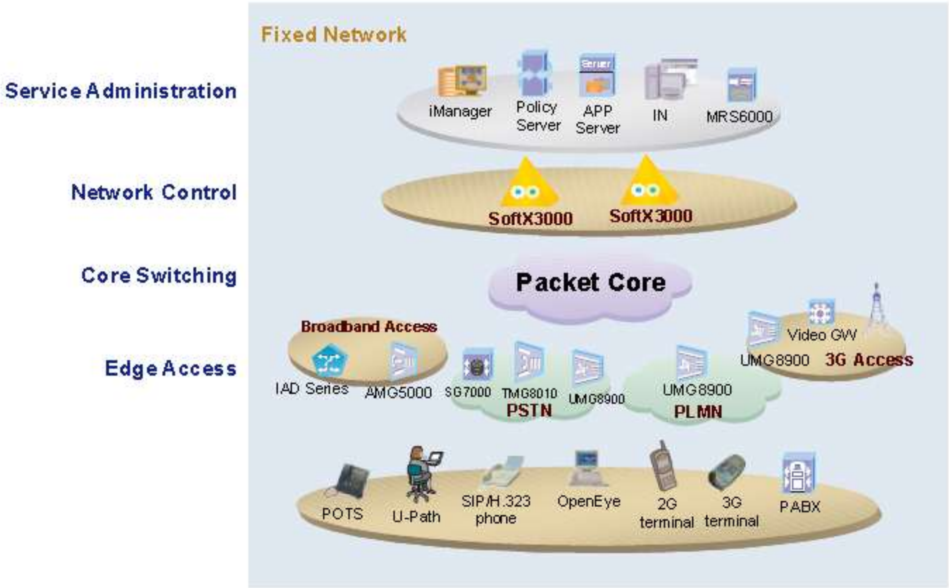

Figure 2.30: U-Sys Huawei NGN Solution (Huawei, Product Catalogue For Fixed Network Solutions, 2003)

ADVANTAGES OF THE SOFTX3000 ARE (Huawei, Product Catalogue For Fixed Network Solutions, 2003):

- **Space:** Takes up little floor space – up to 80% space savings can be achieved by replacing the EWSD network with the U-Sys;
- **Protocol Compatibility:** It supports old and new protocols which make it easier to interface to most systems (e.g. 3rd party systems);
- **Documentation:** Operations and Maintenance manuals exist for the U-Sys.

DISADVANTAGES OF THE SOFTX3000 ARE (Huawei, Product Catalogue For Fixed Network Solutions, 2003):

- **Skilled Labour:** No local knowledgeable personnel are available to implement the U-Sys network, especially on the underlying networks (e.g. Transport);
- **Cost:** It costs Millions of dollars to implement such a system;
- **Training:** Training on the system is also very expensive and can cost up to one hundred thousand Namibian dollars per individual;
- **Spares:** Spares cannot be sourced locally and therefore is very expensive.

2.9.4. HUAWEI CDMA2000 CSOFTX300

Huawei (2012) defines the CsoftX3000 as a mobile switching center used in the CDMA2000 network. The CDMA2000 (code division multiple access) network is a family of 2G/3G networks which uses CDMA channels to send voice, data and signalling data between mobile phones and cell sites (Huawei, CDMA2000 , 2012). Figure 2.31 shows the CDMA2000 network.

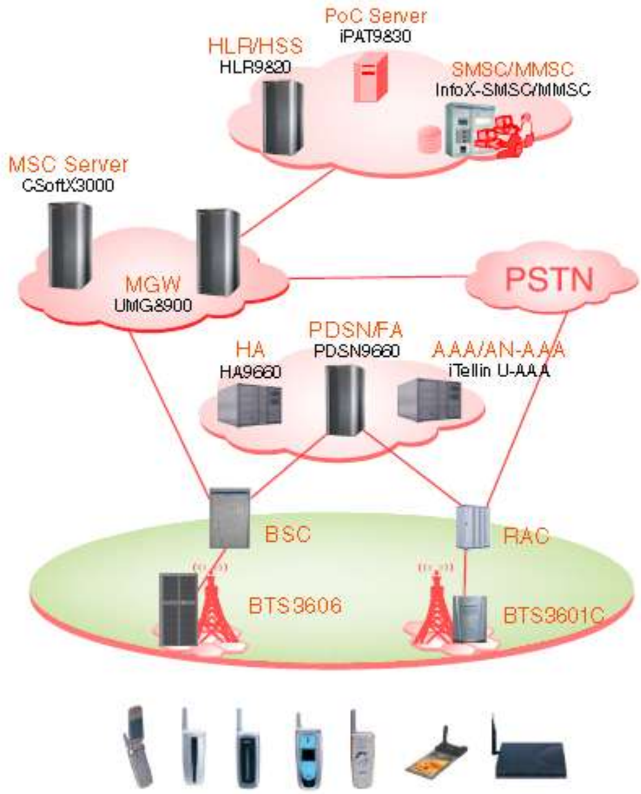

Figure 2.31: Huawei CDMA2000 Network Architecture (Huawei, Huawei CDMA Network Solution, 2005)

THE ADVANTAGES OF A CSOFTX3000 ARE (Huawei, CDMA2000 , 2012)**:**

- **Space:** Takes up little floor space – up to 80% space savings can be achieved by replacing the EWSD core network elements with the CSOFTX3000;
- **Protocol Compatibility:** It supports old and new protocols which make it easier to interface to most systems (e.g. 3rd party systems);
- **Documentation:** Operations and Maintenance manuals exists for the CSOFTX3000.

THE DISADVANTAGES OF A CSOFTX3000 ARE (Huawei, CDMA2000 , 2012):

- **Roaming Agreements:** CDMA is not widely deployed in the world and therefore it's difficult for subscribers to roam internationally;
- **Training:** Training on the system is also very expensive and can cost up to one hundred thousand Namibian dollars per individual;
- **SLAs:** Service Level Agreement costs are very high because service providers are still highly depended on the Supplier of the System;

- **Cost:** Cost still a high factor for the CDMA network, when it comes to implementation, training and support;
- **Spares:** Spares cannot be sourced locally and therefore is very expensive.

2.9.5. GILAT SKYEDGE II

Gilat (2012) defines the SkyEdge as a Next Generation Softswitch system which is used to support voice, broadband data and video communications via satellite. The SkyEdge is designed to support transaction-oriented applications (such as lottery, PoS (Pont of Sale) and SCADA (supervisory control and data acquisition)), legacy protocols such as SS7 and R2 signalling, rural telephony deployments for areas which cannot be reached by the terrestrial infrastructure and multi-topology networks configured mesh, star and multi-star topologies (Gilat, 2012).

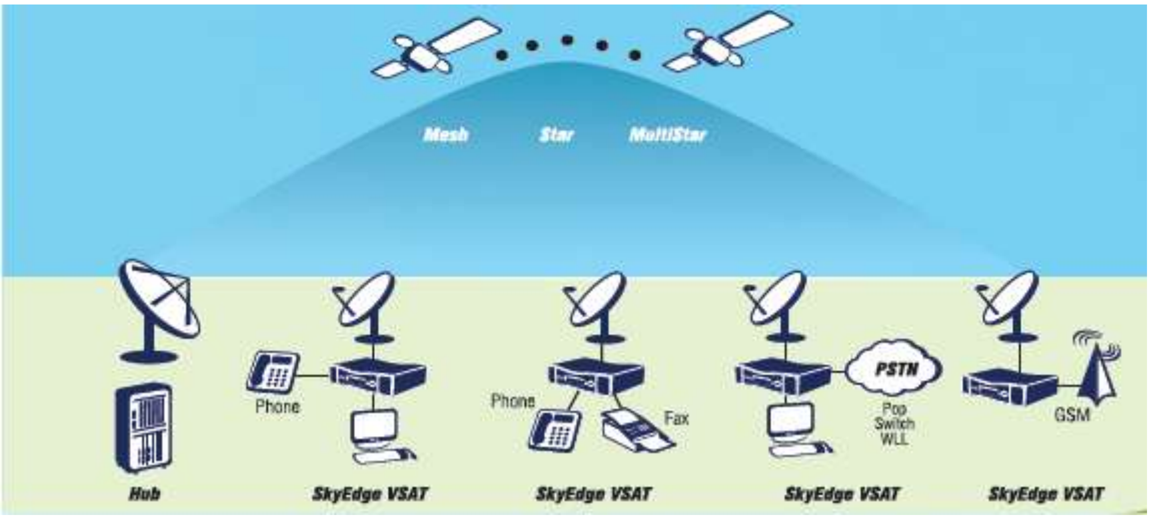

Figure 2.32: SkyEdge Application Diagram (Gilat, 2012)

THE ADVANTAGES OF SKYEDGE ARE (Gilat, 2012):

- **Space:** Takes up little floor space – up to 90% space savings can be achieved by replacing the EWSD network with the SkyEdge System, which automatically leads to power savings.
- **Protocol Compatibility:** It supports old and new protocols which make it easier to interface to most systems (e.g. 3rd party systems).
- **Documentation:** Operations and Maintenance manuals exists for the SkyEdge

THE DISADVANTAGES OF SKYEDGE ARE (Gilat, 2012):

- **Skilled Labour:** Few knowledgeable personnel available to operate and maintain the SkyEdge network and therefore the supplier is responsible to rectify all the high-level failures
- **Cost:** It costs Millions of dollars to implement such a system
- **Training:** Training on the system is also very expensive and can cost up to one hundred thousand Namibian dollars per individual.
- **Spares:** Spares cannot be sourced locally and therefore is very expensive.
- **SLAs:** SLA costs are high because of the supplier's high involvement.

2.9.6. Comparison of the Voice Systems

Table 2-13 is a comparison chart of the five commercial voice systems

Criteria	EWSD	hiE9200	Softx3000	CSoftX3000	SkyEdge II
Cost (Estimates)	More than N$300 Mil.	More than N$100 Mil	More than N$100 Mil	More Than N$300 Mil	More than N$100 Mil
Floor Space (Occupied by core elements)	35 Square Meters	8 Square Meters	8 Square Meters	8 Square Meters	4 Square Meters
Protocol Compatibility	Can only support old PSTN (SS7) and X.25 protocols	Can support PSTN and gateway control protocols (e.g. H.248)	Can support IP protocols (SIP) and PSTN interfaces through gateways	Can support IP protocols (SIP) and PSTN interfaces through gateways	Can support IP protocols (SIP) and PSTN interfaces through gateways
Documentation	O&M documents available	O&M documents available	O&M documents available	O&M documents available	O&M documents available
Service Level Agreements	SLA must always be in place	SLA must always be in place	SLA must always be in place	SLA must always be in place	SLA must always be in place

Table 2-13: Comparison Chart of Commercial Voice Systems

Most of these systems have one disadvantage in common, which is cost. The systems are difficult to implement and therefore implementation is usually carried out by the supplier. The complexity of the system requires that the service provider’s staffs are trained for some part of the system and the supplier must still remain with the 3rd level or specialist responsibilities. Most upgrades and modifications can only be done by the supplier. This means than that for the complete life cycle of the system the service provider must have an SLA with the supplier. There the total cost of implementing this systems, the SLA’s and the Operations and Maintenance of the system are the biggest setbacks in considering buying this systems. Therefore for an affordable solution this research explores the mVoIP System.

2.10. SUMMARY

The chapter highlights the tools that are freely and readily available for use in a VoIP network. Different open source software that can be used as backend servers and mobile VoIP applications are also described with detailed features that can be emulated to produce a more enhanced application. The chapter also assists the project by advising on the different factors to consider when embarking on an application development process. Besides the backend systems and the user applications, the chapter also discusses what is involved on the transport network that acts as the back bone of the project.

The mVoIP project therefore looks at the implementation of a complete network and establishes interoperability between different network elements. Besides the implementation the projects seeks to provide reference to a complete mVoIP network that can be modified, branded and enhanced by providing the low level architecture of the network and code involved. The mVoIP projects therefore pursues to bring all the knowledge gained from literature and previous experience into a combined project in order to provide a simplified mVoIP network with simplified code that can be used for purpose of academy or large-scale deployment. The following chapter discusses the methods of producing the targeted network.

3. DESIGN

This chapter provides the motivation behind selecting the specific PBX system and also defines the backend network elements used in the mVoIP system. The final section of the chapter explains the Android software and its architecture. The Section also highlights the motivation behind choosing the platforms that were used in this project.

3.1. DEFINING THE MVOIP SYSTEM

3.1.1. MOTIVATION

The telecommunication service is a commodity that is not affordable by every citizen. Calls from both mobile and fixed line service providers are billed per second or minute at rates that an average person can afford during the first two weeks after receiving his/her pay cheque. This due to the fact that telecommunications systems are very expensive[12] to implement and therefore Telcos tend to cost their services very high in order to gain enough profits to make a good return on their investments. Telcos have many cost affecting factors to consider (e.g. power, space and staff), since this systems are cumbersome and can take up a lot of room space and uses a lot of power. Access systems are also made out of expensive technologies which sometimes hinder Telcos from deploying their services in low-populated remote rural areas, which in-turn makes the services inaccessible to the many small communities leaving in these areas. With 62%[13] of the Namibian population leaving in rural areas, there is still a need for affordable and easily accessible telecommunications services for many.

With all this expensive methods of communication come data packages that Telcos give to their customer, at monthly flat rates but the monthly usage is not capped. Customers are keen to use the data services for downloading content or simple just serving the net.

Besides the hardware, this project aims to make use of free software and tools in the design of this solution. The project proposes a complete and affordable mobile telecommunication solution (mVoIP) which has the following characteristics:

- Inexpensive Backend servers based on standard computers (x86 processors);
- Computers can be fixed easily – Huge Knowledge base exists locally;
- Free and Easy to setup – makes use of open source software;

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[12] Telecommunication systems cost millions to implement, see Appendix A

[13] As per 2011 census http://www.indexmundi.com/namibia/demographics_profile.html

- Low power requirements;
- Easily scalable – upgrade computer HW;
- Makes use of data connections as the transportation media, easy to interface to other systems;
- Makes use of free app on the user end.

The solution is beneficiary for both the service provider and the customer. The service provider can implement it at low cost and the customer can make use of the service without the fear of hiking up his/her monthly telephone bill but just paying the monthly data flat rate. The mobile app that the project envisions comes with documentation so that the app can easily be customised for a specific service provider.

In addition, Telecommunications sector is perhaps the last major electronics industry that has not been penetrated by the open source revolution. Major telecommunications manufacturers are still building incredibly expensive, incompatible systems (due to proprietary interfaces and protocols), running complicated code. All of the manufacturers have a similar approach towards the products that they offer, they try their best to prevent the systems users to have flexibility or choice, but instead they are locked into the manufacturers product-cycles.

This project intends to change all or part of that. With open source technology, the limits of the telephone system (e.g. functionality and type of technology) can be overcome. Innovative developments are allowed and therefore how the system is implemented is up to the user's preference. The project takes advantage of the computers CPU's whose price/performance ratio continues to increase while turning voice into just another application on the data network.

3.1.2. THE MVOIP CONCEPTUAL DESIGN

The conceptual design covers the holistic view of a targeted mVoIP infrastructure as depicted in Figure 3.1 which also shows an extension of the service to the mobile network.

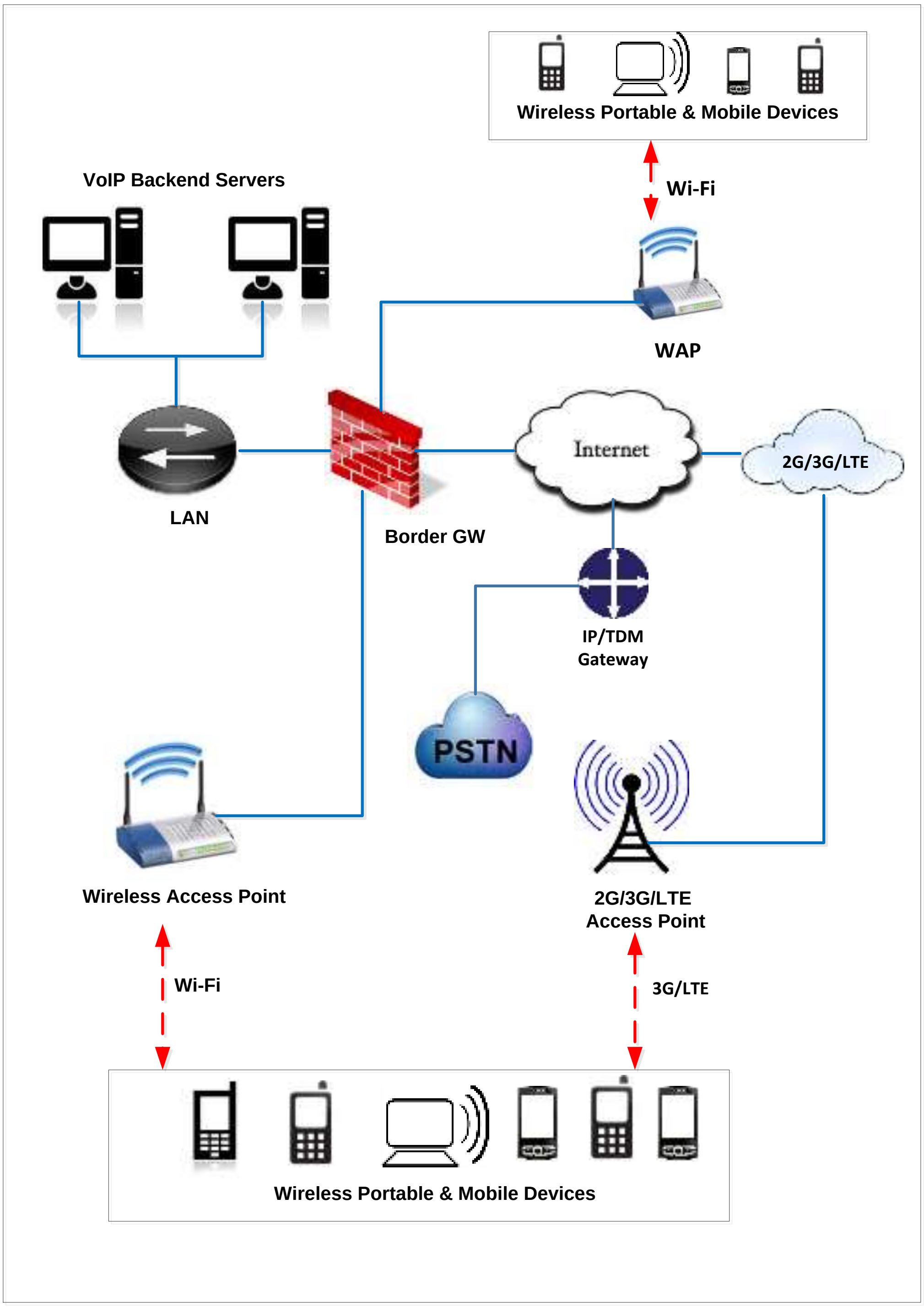

Figure 3.1: mVoIP Network Conceptual Diagram

The mVoIP network is made out of the following components:

- **VoIP Backend Servers (Registrar and SBC)** – with the following functions:
  - Private Branch Exchange (PBX);
  - Firewall & Security;

- VoIP Service Monitoring;
- System Monitoring.

- **Wireless Access Point (WAP)** – with the following functions:
  - Wi-Fi Protected Access;
  - WLAN IEEE 802.11 b/g/n support.

- **User Agents/devices (UA)** – with the following characteristics:
  - Android 2.3 or higher.

Figure 3.2 shows the protocols involved in the communication between the main network elements of the mVoIP Network.

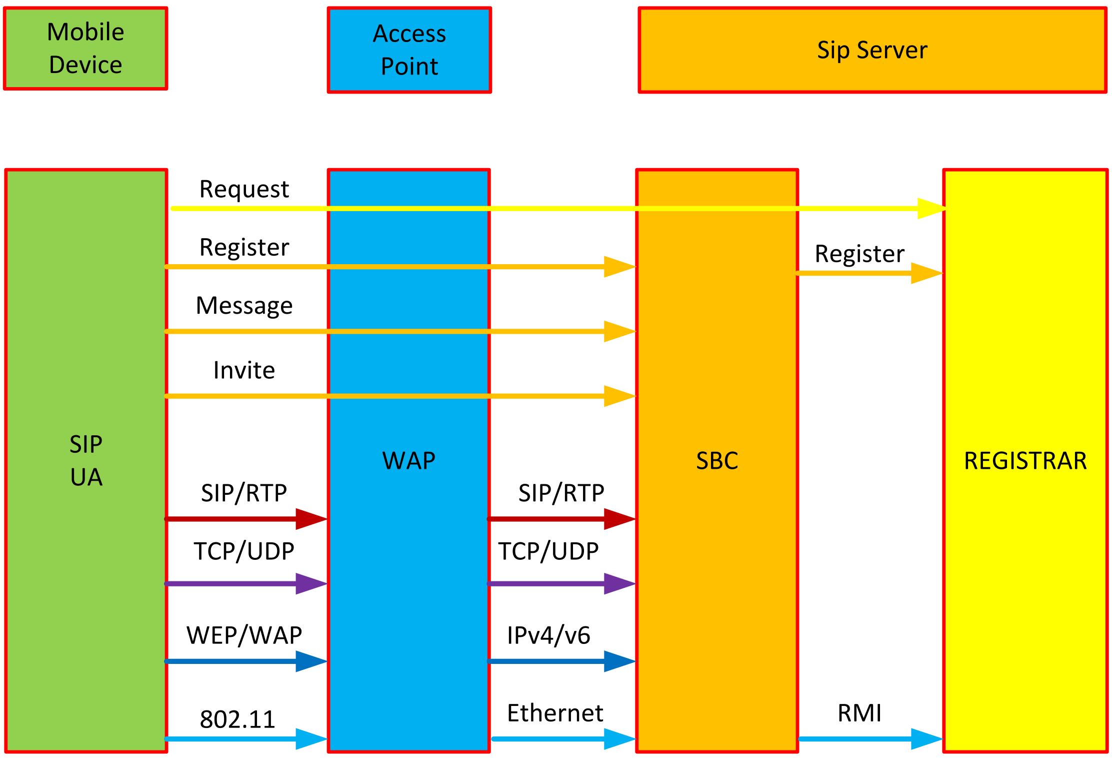

Figure 3.2: mVoIP Functional Diagram

The main point to be highlighted from Figure 3.2 is that the Proxy/SIP server is only involved in the signalling process of the call. The basic call flow is explained in the steps below:

- The SIP UA send a register request to the Register Server;
- Once the SIP UA is registered, it sends an INVITE message to another registered SIP UA via the SBC;

- Once the Invite is accepted then an RTP session is set up.

Note: that the other protocols such as (802.11 and WEP/WAP) are implemented at the beginning of the connection, during the processes of authentication and association.

3.1.2.1. VoIP Back-end Servers

The VoIP Back-end Servers or SIP Servers are two x86 computers running on Windows XP operating System, in a failover cluster setup in order to achieve high availability. With Asterisk 32 open source software, the Backend Servers Provide the following functionalities:

- **Proxy or Session Border Controller (SBC)** – These functionality is implemented for security reasons. The proxy is an intermediary entity that acts both as a server and client for the purpose of making requests on behalf of the other clients. It is responsible for routing and also borders between the backbone network and the access network in order to prevent the network from malicious attack.
- **Registrar** – Accept registrations request from user agents or Proxy. In case of a proxy RMI (Remote Method invocation) is used to communicate to the Registrar.
- **User Agent Server** – It's the logical entity that generates responses to user agent's SIP requests. The response, request, rejects or redirects the request.

3.1.3. The Asterisk PBX System (mVoIP Backend Server)

The mVoIP system on this project makes use of the Asterisk Private Branch exchange as its open source VoIP back end system. The main criteria of choosing the Asterisk system over other open VoIP systems is its ability to support old and new features as was highlighted in section 2.8.1.6. Asterisk is an open source framework for supporting and building communication applications which can turn an ordinary computer into a telecommunications server. Below are more features of Asterisk (Asterisk, 2012):

- Can be used to power IP PBXs, VoIP Gateways, Conference applications servers and Interactive Voice Response Servers;
- Is a free and open source system which can be used by small business, large business, Telcos, call centres and Government institutions;
- Can run on top of different Operating Systems.

Asterisk is also made out of a flexible software architecture which needs to be understood clearly in case the developer requires to do additional development or primarily to ensure that the required communications services are available on the Asterisk system. The following subsection highlights the Asterisk Software Framework

3.1.3.1. The Asterisk SCF Architecture

Malcolm (2012) describes The Asterisk Scalable Communications Framework (SCF) Architecture as a framework that allows developers to create real-time communications applications, which includes (Malcolm, 2012):

- Voice;
- Video;
- Text Messaging;
- Presence;
- Embedded applications for enterprise & carrier solutions.

Malcolm (2012) also states that the Asterisk SCF is designed to provide the highest levels of availability, scalability, extensibility, fault-tolerance and performance and is also delivered as a system of distributed components and can therefore be deployed on to different systems or on to one system (Malcolm, 2012). Figure 3.3 shows the Asterisk Scalable Communications Framework Architecture.

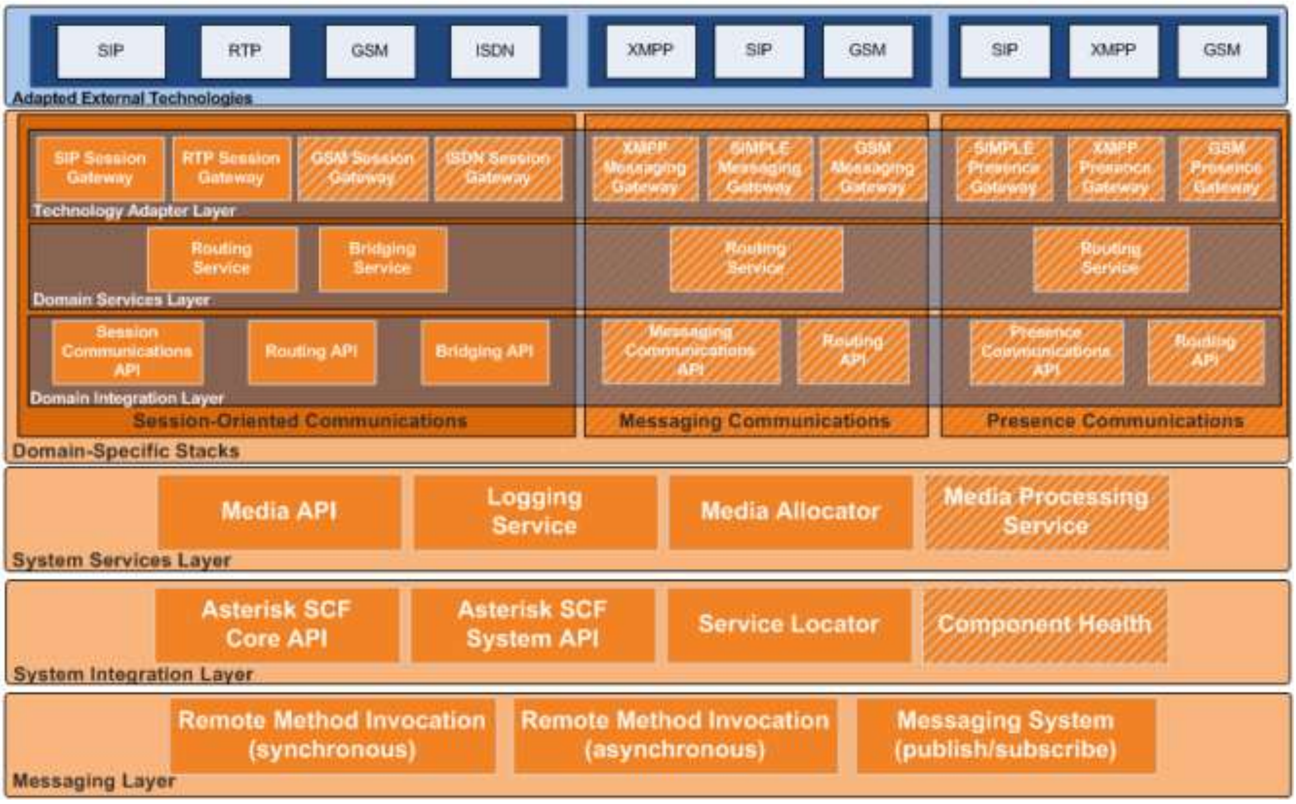

Figure 3.3: Asterisk Scalable Communication Framework Architecture (Alvarez, 2010)

The Asterisk SCF is made out of the following layers and as Alvarez (2010) explains, these layers are not always strictly dependent on each other and therefore components can consume and provide interfaces in multiple layers at once (Alvarez, 2010):

- **Messaging Layer** – Contains additional development patterns, libraries and other tools that provide functionality to be used in other Asterisk SCF components such as:
    - Messaging;
    - Cross-Platform and Cross-Language Data Marshalling and object binding;
    - Event Delivery System.
- **System Integration Layer** – The layer defines interfaces that determine how components will react with each other. The interfaces that exist in this layer provide administration functions to Asterisk SCF system and components. These functionalities includes:
    - Locating Running Services;
    - Monitoring the health of service components;
    - Signalling activation/deactivation of standby status for replicated components.
- **System Services Layer** – Is a collection of interfaces that can be used by many service components but are not required. The components that provide these interfaces sometimes do

it as a way to map external services into the Asterisk SCF system. For voice calls, these services will provide functionalities such as media format transcoding (converting between voice encoding schemes such as converting from G.711 codec to G.723 codec)

- **Domain-Specific Layer**
    - Domain Integration layer – Defines the set of constructs that define the type of communication in the domain. For example the Session Oriented Communications stack defines many objects and operations that apply to a Sip session of different media such as voice, video or text). All this session have common characteristics (such as establishing and tearing down sessions) and therefore the layer can define a set of operations used by routing API's without knowing the type of session involved.
    - Domain Service Layer – Defines services for achieving interoperability within their associated domain among multiple (dissimilar) service components. For example a bridging exists in the Session Oriented Communication stack which can interconnect media streams associated with multiple sessions.
    - Technology Adapter Layer – Adapts external technologies into Asterisk SCF system
- **External Technology Layer –** These layers consist of components that adapt the SCF to be able to interoperate with endpoints such as (users and devices).

3.1.3.2. The Asterisk Functional Architecture

Figure 3.4 shows the modules involved in the functionalities or operations of the Asterisk system. The modules include all the available interfaces that can be used for managing the system and also the available communications interfaces.

Madsen (2011) explains that the heart of any Asterisk system is the PBX core which is the important component that connects calls. Madsen adds that the core also executes other functions such as reading the configuration files such system configurations and loading the other modules such as resources and applications. These files are loaded during the initiation of Asterisk and their functionality is added to the system (Madsen, 2011).

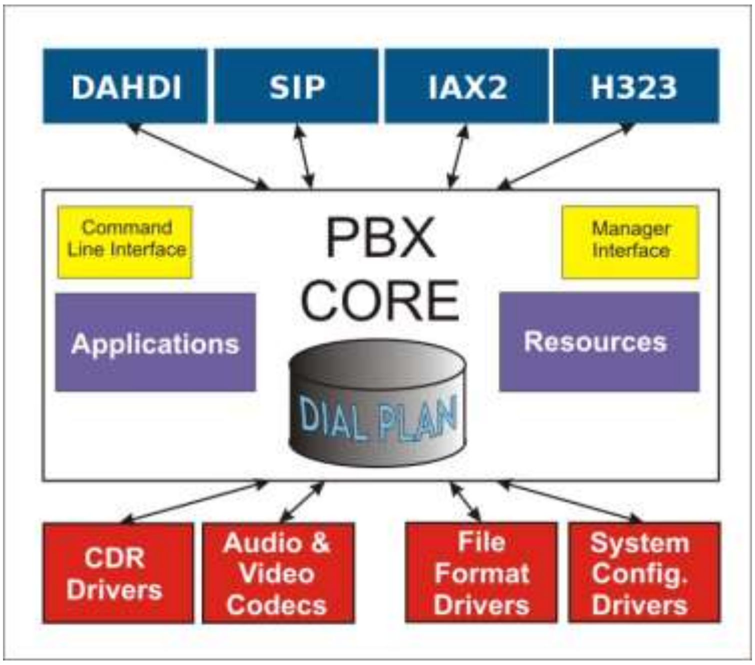

Figure 3.4: The Asterisk Functional Architecture (Madsen, 2011)

The Asterisk Functional Architecture is made out of modules of which some are described below (Madsen, 2011):

- Channels:
    - IAX2 – Used to connect asterisk serves with each other;
    - H.323 – Gateway;
    - SIP – Enables asterisk to communicate with SIP telephones;
    - DAHDI (Digium/Asterisk Hardware Device Interface) – used to control Digium and other legacy telephone interfaces.
- Command Line Interface – The Manager interface for command line administration
- Manager Interface – The PBX Manager Graphical User Interface used to configure users, mailboxes and routes;
- Resources – Includes resource for indications such as ring, busy, congestion and dial tone;
- Applications – Includes applications such as authentication, control playback, message waiting indicator etc.;
- Dial Plan – Database for user, mailboxes and route information

3.1.3.3. The Hardware & OS Selection Examples

The Operating system for hosting the Asterisk PBX is based on Windows 7. The choice was based on the availability and the familiarity of the OS platform to the researcher. There are hardware recommended considerations when planning and dimensioning an asterisk PBX system. Table 3-1 shows some sample hardware recommendations for asterisk system configuration that were used by other projects for home setup (Oej, 2011).

System	Processor	Memory	PSTN interface	SIP Client
HP Vectra PII	400 MHZ MMX	128Mb RAM	Digium X100P + TDM400P	Xten Lite on 4 PCs
AMD Athlon XP	2400+	512Mb RAM	X100P & S100U for local Phones	Xten Lite on 1 PC
AMD Athlon XP	2200	512MB RAM	ISDN Fritz PCI	Xten Lite

Table 3-1: Asterisk System Configuration Options (Oej, 2011)

3.1.3.4. The mVoIP Asterisk Hardware System

For the choice of a backend system the main criteria was availability of the system at no cost and the capability of the system to support the IEEE 802.11 throughput amendments (802.11b/g/n). The project's hardware system for hosting the Asterisk services is based on an Acer TravelMate 5742G which was the available system to the researcher at the moment and which satisfied the IEEE 802.11 requirements. Table 3-2 shows the Hardware specifications of the system:

Specification	Details
Processor	Intel Core i5-480M 2.66GHz with Turbo Boost up to 2.93kHz
Graphics Card	AMD Radeon HD 6370M Up to 2235 MB HyperMemory
Display	15.6" HD (1366x768) LED LCD
Memory	4 GB DDR3 Memory
Hard Disk	640 GB HDD

Wi-Fi	Acer Nplify 802.11b/g/n
Operating System	Windows 7 Enterprise 64-bit operating system

Table 3-2: Asterisk Server HW Specifications

The system configuration is efficient to host the asterisk system w.r.t Table 3-1. The Asterisk feature list for all features available in this project can be found under Appendix C: Asterisk Feature List.

3.1.3.5. The mVoIP Wireless Access Point

The choice for a Wireless Access point for this project highly depended on the availability of the WAP and to its compliance to the specifications below:

- IEEE 802.11b/g/n – for easy integration to other systems such as the backend server and user agents;
- IEEE 802.11i – to be able to provide WPA2 security;
- DHCP Capabilities;
- Ease-of-Use

The access point which was used in this project is a Cisco wireless router product, model number WAG120N which supports the 2.4GHz wireless band only, the other features are listed below in Table 3-3:

Feature	Description
SETUP	
DDNS (Dynamic DNS)	Allows you to assign a fixed host and domain name to a network computer that has been assigned a dynamic Internet IP address. This useful when hosting a server behind the device
MAC Address Clone	The MAC address clone screen lets you use the MAC address of a device that has already been registered with a service provider, by copying the MAC address to the device.
Advanced Routing	The advanced Routing Screen allows you to configure the device's advanced routing functions such as operating mode, dynamic mode and static routing.
WIRELESS	

Basic Wireless Settings	The Basic Wireless screen allows you to set up a network with Wi-Fi protected setup, change the radio band, select the network mode, change the wireless network name (SSID), change the wireless channel and disable the SSID broadcast.
Wireless Security	The wireless security scree allows you to configure the security of your wireless network(s). It allows you to select from a number of security modes such as WPA2.
Wireless MAC Filter	The wireless MAC filter screen allows you to control (block or allow) Internet access by individual devices on your wireless network by specifying their MAC addresses.
SECURITY	
Firewall	The firewall screen allows you to configure a firewall that can filter out various types of unwanted traffic on the device's local network.
VPN Passthrough	The VPN Passthrough screen allows you to configure the device's VPN Passthrough feature. This feature lets VPN tunnels using the IPSec, L2TP, or PPTP protocols to pass through the devices firewall.
ACCESS RESTRICTIONS	
Internet Access Policy	The Internet access policy screen allows you to define policies that are used to block or allow specific kinds of Internet Usage and traffic, such as Internet access, designated applications, websites, and inbound traffic during specific days and times, and for specified list of devices in your network.
ADMINISTRATION	
Reporting	The reporting screen allows you to configure system event logging and to configure sending of e-mail alerts when a Denial of service attack is detected.
Diagnostics	The Diagnostics screen allows you to perform a ping test to check the status of your Internet connection
STATUS	
Local Network	The local network screen displays the status of your Local Area Network
Wireless Network	The wireless network screen displays some basic information about the wireless network of the device

DSL Connection	The DSL Connection screen displays the status of your DSL connection.

Table 3-3: WAG120N Feature list (Cisco, Quick Installation Guide ADSL Modem outer, 2012)

DDNS is a very useful feature for the project especially when deploying the VoIP servers over a Telco network or behind the Internet. Users will be able to reach the server by using the same host name without any worries about DHCP services.

The Wireless MAC Filter feature can be used to disallow users who threaten the network. Therefore mVoIP users known for instigating security threats on the network can be black listed. The firewall feature can be used to only allow the traffic required for user communication, which in turn can improve QoS.

Table 3-4 lists the other specifications of the WAG120N which are relevant to the project:

Specifications	Details
STANDARDS	
WLAN	802.11n draft 2.0, 802.11g, 802.11b
ETHERNET	802.3, 802.3u
ADSL	T1.413i2, G.9921 (G.DMT), G992.2 (G.Lite), G992.3 (ADSL2), G992.5 (ADSL2+) for Annex A,B,M,L, U-R2 for Annex B
MODULATIONS	
802.11b	CCKJ/QPSK, BPSK
802.11g	OFDM/BPSK, QPSK, 16-QAM, 64-QAM
802.11n	OFDM/BPSK, QPSK, 16-QAM, 64-QAM
Security Features	WPA2/WPA Personal and Enterprise; 128, 64 bits WEP; MAC Address Filtering; SPI Firewall;
OS Requirements	Windows XP, Vista, or Vista 64-bit edition with latest updates

Table 3-4: WAG120N Specifications (Cisco, Quick Installation Guide ADSL Modem outer, 2012)

The WAG120N Supports 802.11b/g/n standards which are Wireless LAN standards that deals with the throughput of the device. Security is provided by Wi-Fi protected access and wired equivalent privacy protocols and additionally the SPI (stateful packet inspection) Firewall is also supported and it can be programmed to distinguish and allow legitimate packets for each connection.

3.1.4. Overview of Android

The characteristics of the Android Platform (e.g. availability of support Documentation, shortest learning curve and targeted devices) steered to it being the preferred development platform for this project. But before venturing into the designing and development processes of the platform, there are some concepts that need to be elucidated.

As seen in Figure 3.5, android is an open source software stack for mobile devices that includes an operating system, middleware and key applications. The Android SDK provides the tools and API's necessary to begin developing applications on the Android platform using the Java programming language (Consortium, 2010).

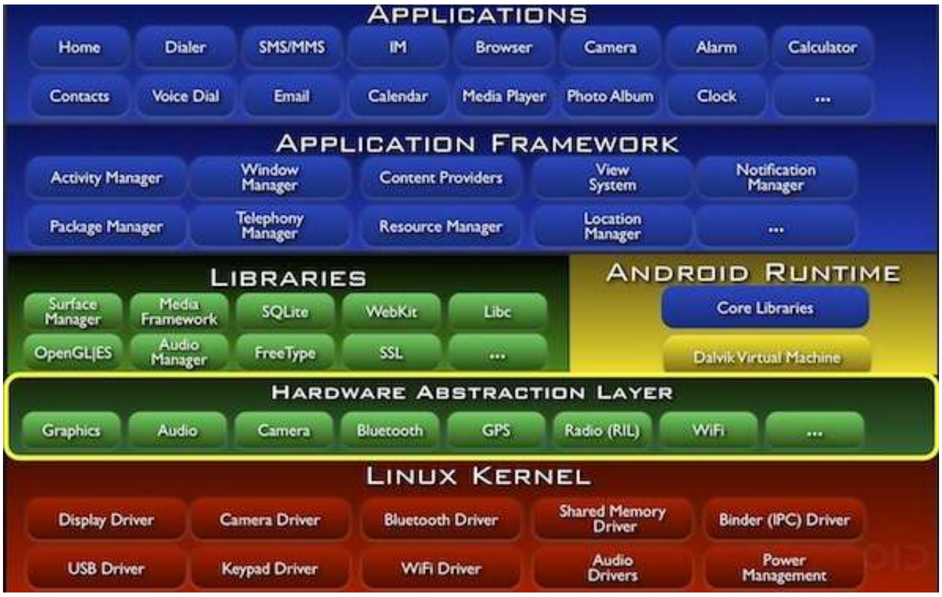

Figure 3.5: Android Software Architecture (Consortium, 2010)

The major components of Android are (Consortium, 2010):

- **Linux kernel.** Android relies on Linux 2.6. The kernel acts as an abstraction layer between the hardware and the rest of the software stack and provides core system services: security, memory management, process management, network stack and driver model.

- **Android Runtime.** Android includes some of the core libraries that provide most of the functionality available in the core libraries of the java programming language. Every android application runs in its own process, with its own of the **Dalvik virtual machine**[14]**.**

- **Libraries.** Android includes a set of C/C++ libraries used by various components of the Android system. These capabilities are exposed to developers through the Android application framework.

- **Application Framework.** Developers have full access to the same framework APIs used by core applications. The application framework was designed to simplify the reuse of components.

- **Applications.** Android ships with a set of core applications including an email client, SMS program, calendar, maps, browser, contacts and others.

3.2. SUMMARY

This chapter highlights some of the motivations why open source VoIP softwares should be a considered as a replacement for the traditional and the Next Generation Networks Voice Systems. The current deployed systems are costing service providers a lot of money to implement, operate and maintain. On a yearly basis there are also additional costs in the form of Service Level Agreements that stipulates how a Supplier can assist the customer who bought the system in case of an emergency.

This chapter therefore motivates a mVoIP solution that can be used to overcome this hurdles that the Service Providers are facing. The required backend systems for the solution are also highlighted in this chapter. The next chapter looks at the design of the user interface.

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[14] Dalvik has been written so that a device can run multiple VMs efficiently. The Dalvik VM executes files in the Dalvik Executable (.dex) format which is optimised for minimal memory footprint (Consortium, 2010).

4. THE MVOIP APPLICATION DEVELOPMENT & DEPLOYMENT

This chapter starts off by analysing the mobile application development processes that have been used by other developers. The chapter then goes into comparing the different development approaches in order to choose the most suitable one. The chapter also describes the development process of the mVoIP application. The final section descries the test scenarios of the project.

4.1. INTRODUCTION

Each mobile application has its own set of unique challenges, most projects follow the basic development processes which include: Planning, Design, Implementation and Deployment. For this projects application development, the focus is mostly on a development process that provides clear guidelines when designing a Mobile Application. The approach is therefore a review of process used by other developers.

4.2. MOBILE APPLICATION DEVELOPMENT PROCESSES

Mobile app development processes are sometime based on traditional software development processes such as the Waterfall model and in some cases the developers have customised the models to add flexibility to their projects. The approach to the development of this Mobile Application development was choosing the right development process. This involved swotting of mobile app development processes defined by different mobile application developers and comparing them to see which processes can be adapted for the development of this application.

4.2.1. SPUDD MOBILE DEVELOPMENT APPROACHES

Adam (2012) describes Spudd Mobile as provider for custom mobile solutions. Adam states that a number of their clients prefers the waterfall development model, which is made out the 5 phases:

Requirements, Design, Implementation, Verification and Maintenance. Adam further states that this model is only applied for fixed-price projects where a products requirements have been decided in advance because the model phases has to be executed sequentially (Adam, 2012).

Adam (2012) adds that It is also important to note that the waterfall development model is not right for every project and therefore some of their clients prefer a time and materials approach, which gives them the flexibility to work with agile development approaches. Adam further adds that in agile development the customer first pays a retainer fee and then development process start. This approach Adam explains ensures all working hours are covered financially, giving the developer the freedom to revisit any stage of the process at any point in time. Adam further explains that this is ideal in situations where a client requires the flexibility to change their requirements as the product develops. It is understandable that it may impossible for a client to pinpoint exactly the complete requirements without seeing it in action first – a time and materials approach allows for this (Adam, 2012).

4.2.2. ERI MOBILE DEVELOPMENT PROCESS

Aron (2011) describes Eri Mobile as a full service development company that provides its clientele with various options. No matter if the requirement is for the application simple or complicated, Eri mobile will take the customer's idea from concept to completion (Aron, 2011).

Below are the steps executed during EriMobile's mobile application development process (Aron, 2011):

- Concept Sketching – Bringing the idea to paper;
- Research Strategies – Conduct research on the concept;
- Wire-Framing – Design Mock-up views;
- User Interface Design – Convert mock-ups into clean interfaces;
- Development – Coding of application;
- Testing – Functionality testing;
- App Store Submission – Uploading of the application onto an app stores;
- Marketing – marketing the application

4.2.3. *MyFirstMobileApp Application Development Process*

MyFirstMobileApp is a company that specialises in mobile application development and makes use of the following models (MyFirstMobileApp, 2012):

- Fixed Cost Model. The fixed model consists of the following steps:
    - Customer sent an enquiry
    - Requirement is analysed
    - Provide quotation
    - Payment is received
    - Development Starts
    - Complete and Sign Off

- Hourly Basis Model
    - Customer sent an enquiry
    - Requirement is analysed
    - Time and Material model is suggested
    - Development is started
    - Payment is received
    - Project Monitoring
    - Complete and Sign off

- Hire Dedicated Model
    - Customer sent an enquiry
    - Requirement is analysed
    - Hire dedicated model is suggested
    - Payment is received
    - Assign Dedicated Resources
    - Assign Project Tasks
    - Resource Work on Tasks
    - Daily reporting
    - Monthly Contract Renewal

4.2.4. *EVOLUTIONATE APPLICATION DEVELOPMENT PROCESS*

Evolutionate is a mobile application development company and their process follows the following steps (Evolutionate, 2012):

- Discovery – Discussion and review of customer designs and requirements;
- Understanding – Provide customer a refined requirements documentation
- Improvement – Suggest improvements on customer requirements
- Document – Wireframing of the requirements
- Design – Design how the app will look
- Coding – Building of the applications
- Alpha Testing – Extensive testing to fix any tweaks in the code
- Beta Testing – Customer test the app
- Release – release to the app to the app store
- Warranty – if a bug got through with the release of the app, it will be fixed for free

4.2.5. *CONCLUSION*

The processes defined in this section are mostly adapted by the developers. What is most important to this project is to make use of one of the process that has clear defined steps and but is simple to implement. The approaches used by the other developers can be summarised as follow:

- **Spud Mobile Development Approaches** – The approaches summarised here is based on traditional development models such as waterfall and agile development but in the content of mobile application development, nothing was clearly highlighted.
- **Eri Mobile Development Process** – The approaches can be followed because it is sequential in its design and it clearly highlights all the required steps. Focuses on the important phases, which therefore simplifies the development process.
- **MyFirstMobileApp Development Process** – This are actually business process that pin points when development takes place but the development process is not defined.
- **Evolutionate Application Development Process** – This process also defines the steps for mobile application development very clearly and therefore it is easy to follow. The process is much similar to Eri's Mobile Development Process but with more steps introduced which can lengthen a development process.

4.3. MOBILE APPLICATION

The mVoIP application development is a customised approach which made use of the process that was defined by Eri Mobile (Aron, 2011). This process is very easy to follow due its sequential set steps but it is not attached to the complexity or cost of a project and does not require interaction with too many decision makers. The development process is also not too long like the process followed by Evolutionate which can save time for the purpose of this project. With reference to Figure 4.1, the mVoIP application development process only adopts phases one to six, while seven and eight has been omitted.

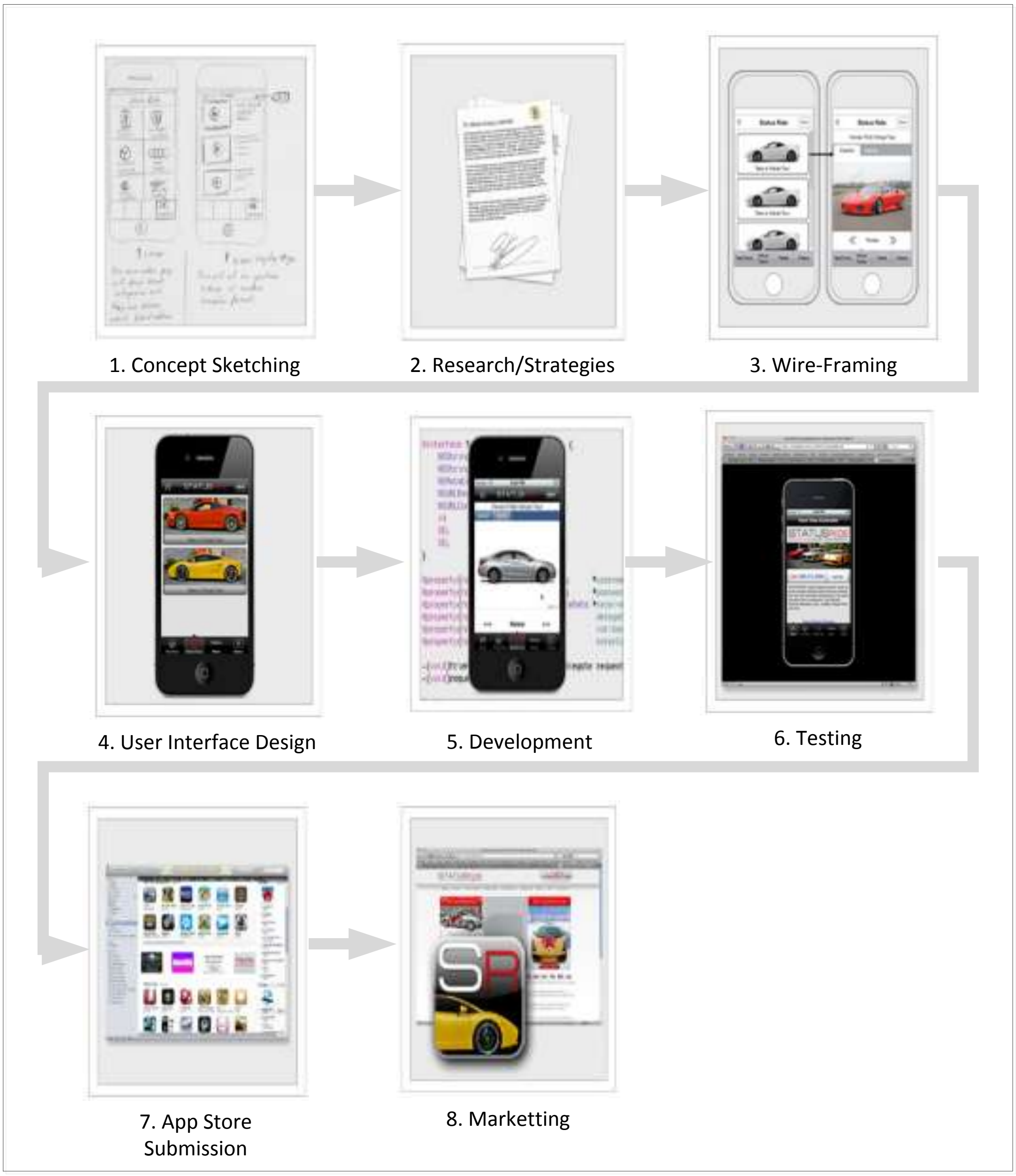


Figure 4.1: Mobile App Development Process (Aron, 2011)

4.3.1. CONCEPT SKETCHING

This step introduces an idea or concept of the application. The design highlights the basic functionality of the application as a user may describe it after the application has loaded. Figure 4.2 shows the mVoIP concept sketch.

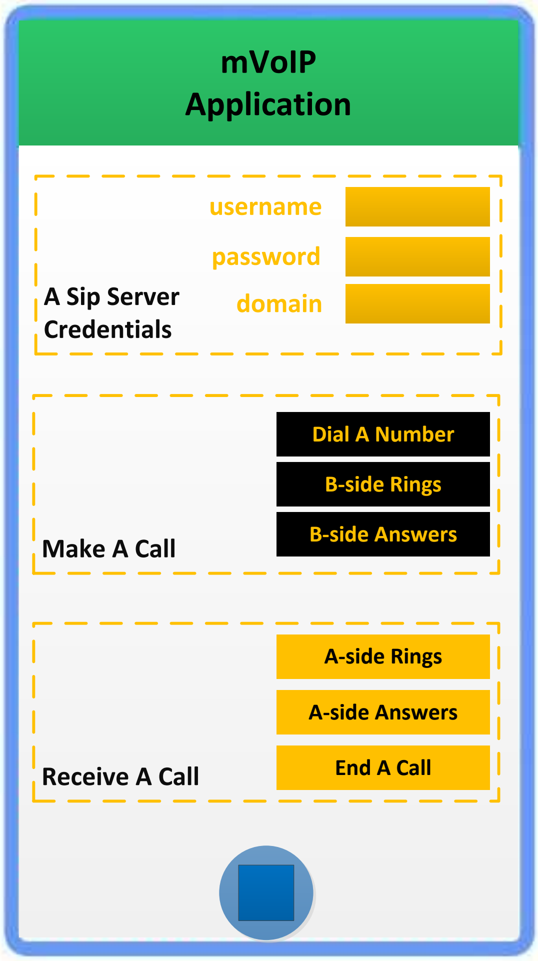


Figure 4.2: mVoIP Concept Sketch

The main functions of a telephone application are highlighted in the concept sketching, which are:

- **SIP Server Credentials** – This function requires the application to accept the user information that will be used to register with a backend server in order to make use of the VoIP service. This function is excluded from traditional telephone applications, since the service is activated once a SIM card is attained;
- **Make a Call** – Where the A-number (app user) or caller dials the B-number or dialled number;
- **Receive A Call** – Where the B-number in this case the app user or called receives a call from an A-number or caller.

4.3.2. RESEARCH & STRATEGIES

Aron (2011) explains that the R&S stage is where the concept starts taking shape and its steps contemplate on the specific approaches, classes and interfaces required to make the functions highlighted in the concept sketching work. In order to assist the research on how to create a mobile VoIP application, the researcher (after long research on how to develop android applications) turned to the android developer's site which highlights the material required to develop a SIP or VoIP application (Aron, 2011). The following sub-sections summarises the information provided on the development of a SIP application.

4.3.2.1. Introduction to SIP API

According to Android developer (2012) website, Android provides an API that supports the Session Initiation Protocol (SIP) which enables a developer to add SIP-based Internet telephony features to an application. Android includes a full SIP protocol stack and integrated call management services that lets applications easily set up IP telephony, without having to manage sessions, transport-level communication, or audio record or playback directly (Android D. , 2012).

According to the Android website (2012), the SIP API is available in the **android.net.SIP** package. The key class is **SIPManager**, which applications use to (Android D. , 2012):

- Create a SIP Session;
- Initiate and receive SIP calls (video or audio);
- Register and Unregister with a SIP service provider;
- Verify session connectivity.

Once a call is established, applications can mute calls, turn on speaker mode, send DTMF (Dial Tone Multi Frequency) tones, and more. Applications can also use the **SIPManager** to create generic SIP connections (Android D. , 2012).

Below are examples of the types of applications that might use the SIP API (Android D. , 2012):

- Video Conferencing;
- And Instant Messaging.

4.3.2.1.1. SIP API Requirements and Limitations

Here are the requirements for developing a SIP application (Android D. , 2012):

- Only mobile devices that are running Android 2.3 or higher can be used to run SIP applications. The SIP Protocol Stack and Framework API was only included in Android 2.3 to allow developers to build Internet telephony applications;
- SIP runs over a wireless data connection, so your device must have a data connection (with a mobile data service or Wi-Fi). This means that one cannot test an application using AVD (Android Virtual Device), but one can only test using a physical device.
- Each participant in the application's communication session must have a SIP account. There are many different SIP providers that offer SIP accounts. But for this project the SIP accounts will be offered by the SIP proxy that is implemented by the project.

4.3.2.1.2. SIP API Classes and Interfaces

Table 4-1represents a summary of the classes and interface (**SIPRegistrationListener**) that are included in the Android SIP API which the developers can use to develop the mVoIP application (Android D. , 2012):

Class/Interface	Description
SIPAudioCall	Handles an Audio call over SIP. It can be instantiated with SIPManager using **makeAudioCall ()** and **takeAudioCall ()**.
SIPAudioCall.Listener	Listener for events relating to a SIP call, such as when a call is being received or a call is made.
SIPErrorCode	Defines error codes received during SIP actions. For example **onRegistrationFailed (), onError ()** and **onCallChangeFailed ().**
SIPManager	Provides APIs for SIP tasks, such as initiating SIP connections, and provides access to related SIP services. This class is the starting point for any SIP actions. You can acquire an instance of it with **newInstance ()**.
SIPProfile	Defines a SIP profile, including a SIP account (username and password), domain and server information.
SIPProfile.Builder	Helper class for creating a SIP profile.
SIPSession	Represents a SIP session that is associated with a SIP dialog or a standalone transaction not within a dialog.
SIPSession.Listener	Listener for events relating to a SIP session, such as when a session is being registered ("on registering") or a call is outgoing ("on calling")

SIPSession.State	Defines SIP session states, such as "registering", "outgoing call" and "in call".
SIPRegistrationListener	An interface which is a listener for SIP registration events.

Table 4-1: SIP API Classes and Interfaces (Android D. , 2012)

4.3.3. WIRE-FRAMING

Aron (2011) explain that his step focuses on fine-tuning and modifying the user experience by designing the layout of each of the app's views. Aron adds that a developer should always aim to meet or optimally exceed the user needs when designing a mobile app. Wire-framing is one of the highest important steps and it is the foundation for providing a successful application development which requires that the user is involved in these rough sketches of the app's views (Aron, 2011).

The first step in the Wire-framing processes was to plan and design the application views. The application is made out of a series of views which are:

- **SIP Settings, and**
- **Make a Call**

Note: That "**receive a call**" function will only be invoked once there is an incoming call and is not part of the root view.

The Next step was to plan and design the navigation routes of the application, which are represented by the directional lines as shown in Figure 4.3 which illustrates the application views and navigation routes of the user SIP settings input process. For example in which view will the user end up in, if an OK button is pressed? Note that the wire-framing looks at designing the parts after the application have loaded and all the generic views are mocked in the wire-framing design.

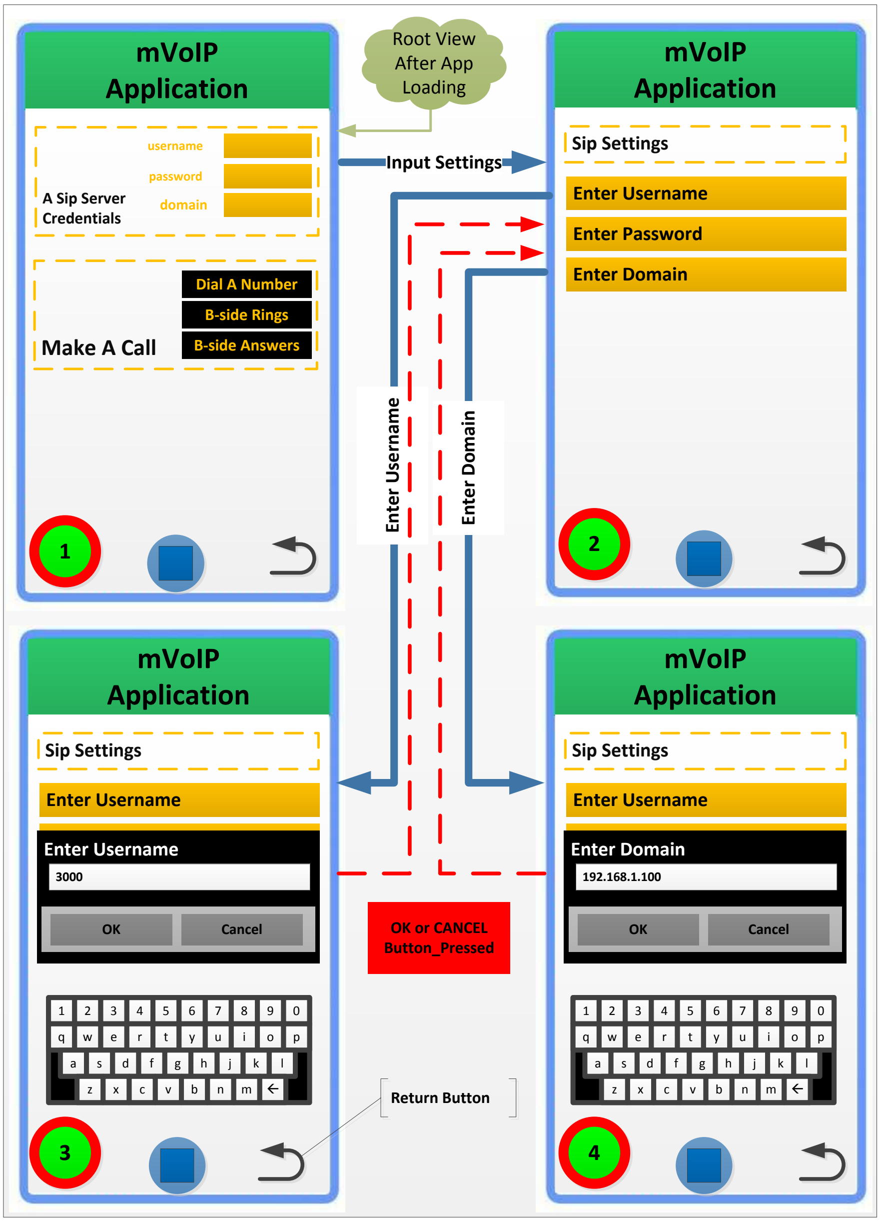


Figure 4.3: SIP Server Settings Functional views and Navigation routes

The Root View is the initial view that a user will see after the application has loaded. The root view contains all the functionalities of the application. If a user decides to input the SIP server settings then the following process takes place:

- **View 1** – Is the Root View from which the setting Button is clicked.
- **View 2** – Provides the user with SIP setting options
- **View 3** – If the user chose to enter a username for example then,
    - A Edit Text Box appears or pops up in a front of the SIP setting options
        - And an alphanumeric keypad also appears

- After the user enters a user name and completes the action with OK or Cancel, s(he) will be navigated back to the SIP Setting option view.

- **View 4** – Can also be invoked right after View 2 if the user opts to enter a Domain
- **View 5** – Is not shown here but it can be used to represent the action of choosing to enter a password, which is similar to views 3 & 4.
- The embedded return button of a phone can also be used to navigate back to a previous view.

The next step was to design the Make a call function views and navigation routes shown in Figure 4.4:

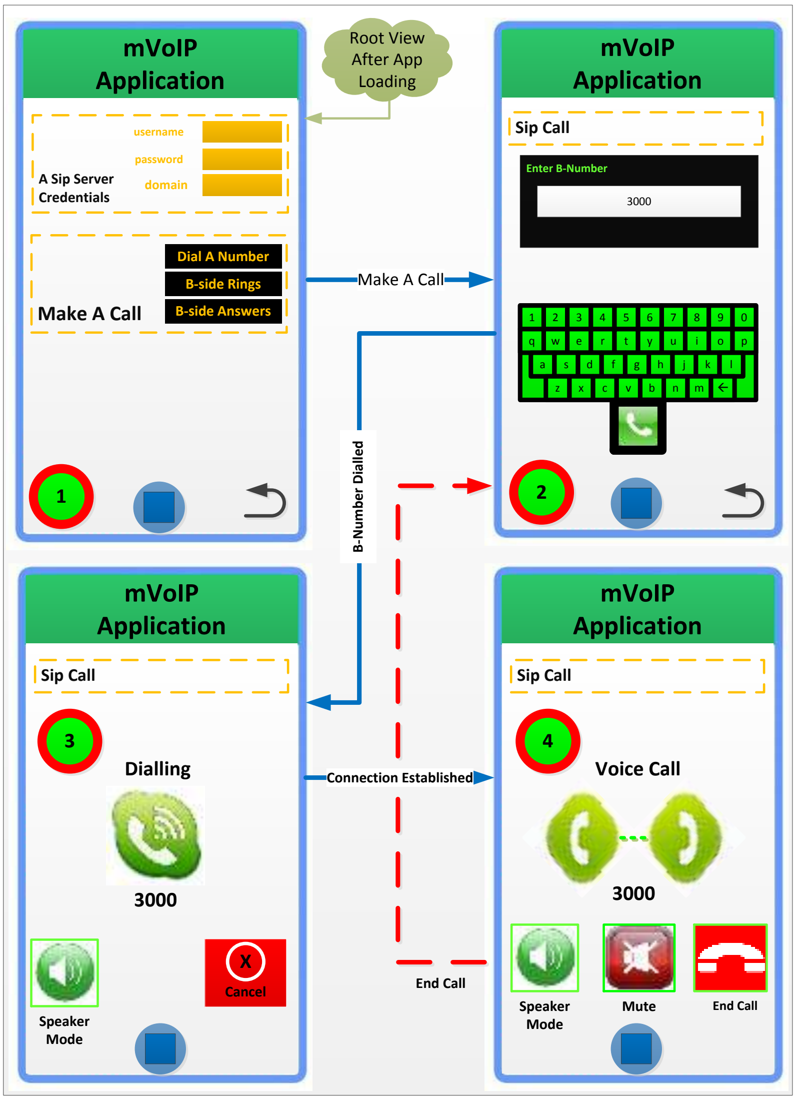


Figure 4.4:Make a Call Function Views and Navigation Routes

In the “Make a call Function” , the views designed can be explained as follows:

- **View 1** – Root of view of the application, which is the first view that the user sees after the application have loaded.
- **View 2** – If the user opts to make a SIP call, this view will be loaded, with a numeric keypad, edit text window and a call button. Once the number to be called (B-number) is entered in to the text window, then the call button can be pressed to make the call. The user can also opt to press the return button to go back to the root view.
- **View 3** – Is the view that the user gets when an outgoing call is made. The user has the following options:
    - Cancel the call – Navigates back to View 2
    - Switch to speaker mode – Will hear the ring tone on the phone’s speaker
- **View 4** – This view is presented to the user, once the called number (B-number) answers. The options on this view are as follows:
    - Switch to speaker mode – For free purposes and for amplifying the conversation
    - End call – ends the call and navigates the user back to view 2

The last Wire-Framing step was the design of the “Receive a call” function views and navigation routes as shown in Figure 4.5

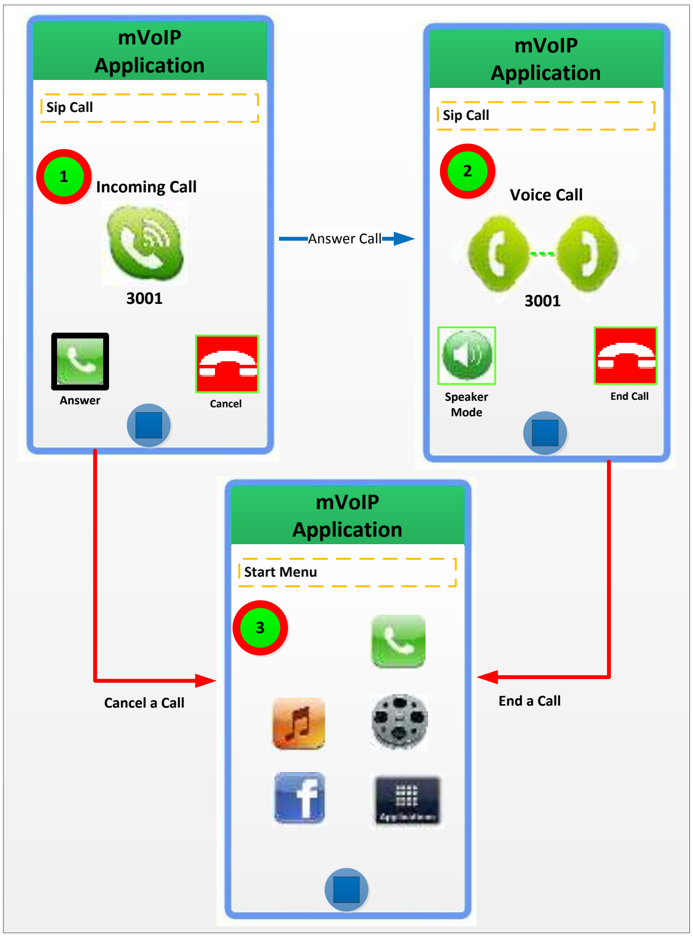


Figure 4.5: Receive a Call Function Views and Navigation Routes

Figure 4.5 depicts the views of the receive a call function. The three views can be explained as follows:

- **View 1** – The view is invoked once another user (B-number) dials the user (A-number). The options that comes with the view are:
  - Answer the call – navigates the user to view 2
  - Cancel the call – navigates the user to view 3 which is the applications view that is invoked at the normal start-up of the phone.
- **View 2** – The user is navigated to view 2 if he accepts or answer the call. The options of this view are:
  - Speaker Mode – Which amplifies the conversation
  - End Call – which exits the application and returns the user to View 3
- **View 3** – Applications view

4.3.4. USER INTERFACE DESIGN

After completing the Wire-Framing, the developer will have a good idea of how user engagement and screen flow will take place on the specific application. Aron (2011) explains that this step converts the wire-frames into clean and attractive interfaces and the important factors to remember here is the easy-of-use and quality of the UI design (Aron, 2011). The UI designs for the SIP setting process are used. Some of the UI designs are demonstrated in this section, in order to demonstrate the end product. Figure 4.6 depicts how the application will look like in real life once it is installed on the user device.

Figure 4.6: UI design -- mVoIP application installed on the phone

Figure 4.7 shows the screen that the user will see once the application is initiated. The screen provides the user with the options to:

- Make a call
- Insert/Change the SIP Settings
- End a call

Figure 4.7: mVoIP when initiated

Once the Settings button is clicked then the screen in Figure 4.8 appears with the options to enter/edit the username, domain or password of the user. The information entered in to these fields should be the information that exists on the backend VoIP server.

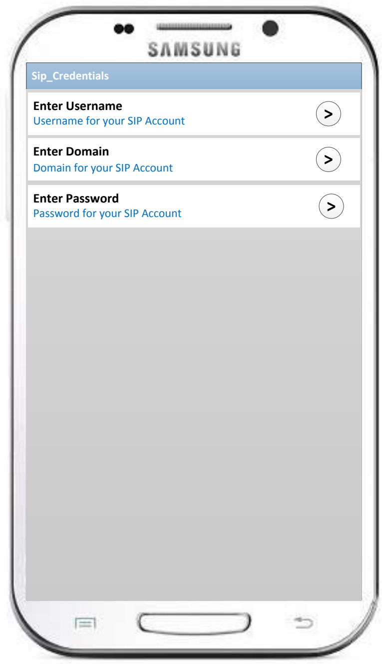


Figure 4.8: The mVoIP SIP Settings Menu

Once the Enter Domain option is chosen the screen in Figure 4.9 is fired to allow the user to modify or enter the URL of the SIP or proxy server.

Figure 4.9: mVoIP Enter Domain option

What should be noted from the UI design is that the receive a call, make a call and end a call functions makes use of the same layout depicted in Figure 4.7

4.3.5. DEVELOPMENT

The development step involves coding of the application. Server components will be taken into account during the process to streamline development time and to avoid any complications. The guidelines for developing an application that makes use of the SIP API are provided on the Session Initiation Protocol webpage (Android D. , 2012). Figure 4.10 shows the Lifecycle of the mVoIP application.

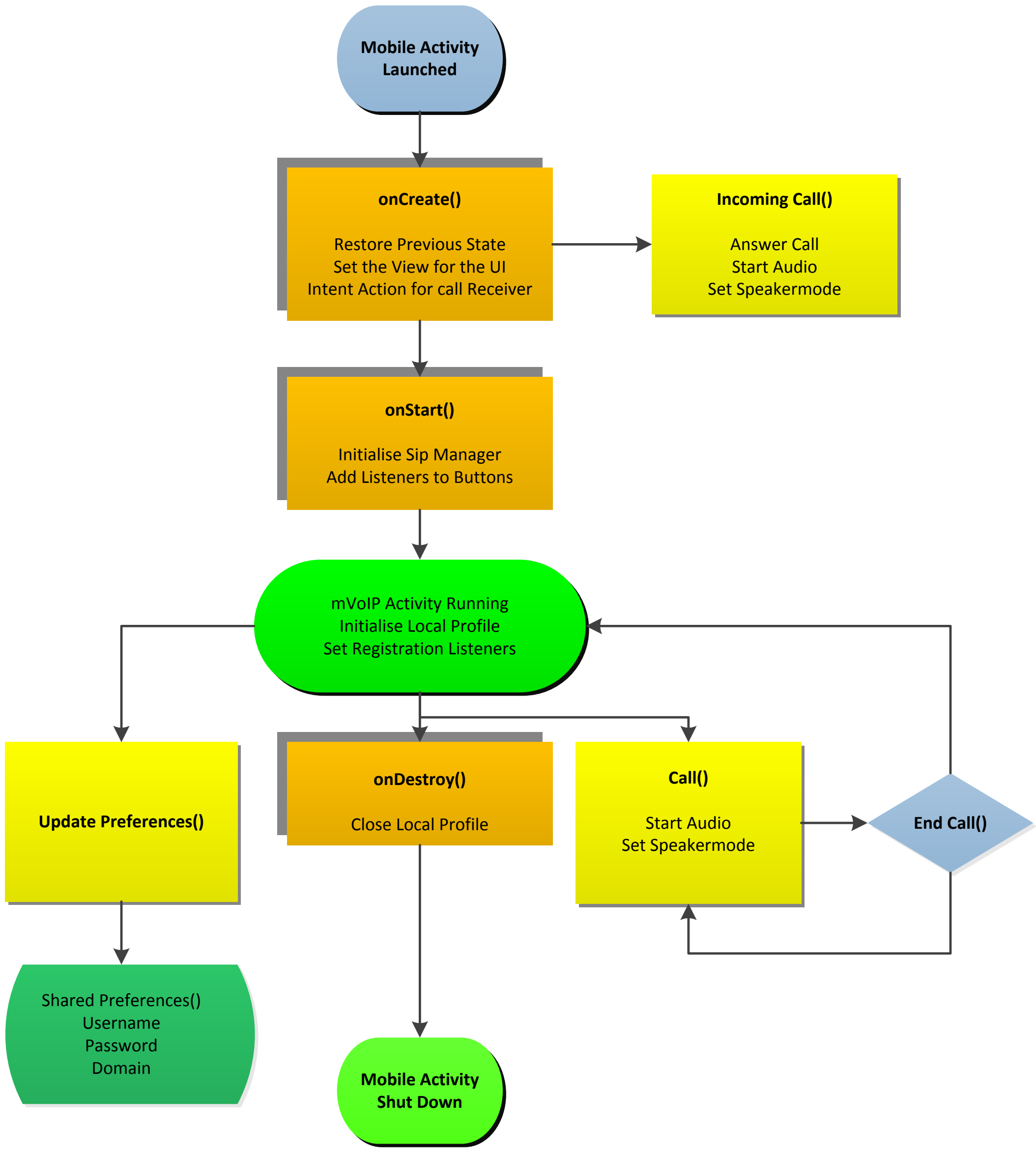


Figure 4.10: mVoIP Life Cycle

The initial steps in the developing of the mVoIP application were to create the manifest. The manifest file is the configuration file of the application, which tells the android platform about the parameters of the application. The file defines the following:

- Which SDK versions the application is being written for. In this case:
    - minSdkVersion = 9
    - targetedSdkVersion = 15
- The permissions that the application must be granted to function. In this case:
    - USE_SIP – Allows an application to use SIP Services
    - INTERNET – Allows an application to open network sockets

  - ACCESS_WIFI_STATE – Allows application to access information about WiFi networks
  - WAKE_LOCK – Allows using Power Manager to keep processor from sleeping or screen from dimming
  - GET_TASKS – Allows an application to get information about the currently or recently running tasks
  - RECORD_AUDIO – Allows an application to record audio
  - MODIFY_AUDIO_SETTINGS – Allows an application to modify global audio settings
- The Hardware features that the application must use. In this case:
  - Hardware.SIP
  - Hardware.SIP.voip
  - Hardware.wifi
  - Hardware.microphone

Below are the key loops that were included in the mVoIP application (activity):

- The **entire lifetime** of the activity happens between the first call to **onCreate()** through to a single final call **onDestroy().** The activity does all setup of "global" state in onCreate() and release all the remaining resources in onDestroy(). The following methods are also called in onStart():
  - **SetContentView()** – Calls the activity Main Layout or user interface on the screen.
  - **RegisterReceiver()** – Register a Broadcast receiver that monitors for changes in the User Interface. In this case the Broadcast receiver is registered to listens for incoming call by instantiating an incoming call activity. The receiver is unregistered in **onDestroy()** call. The receiver implements the following methods:
    - **answerCall()** – Answer calls and is initiated with a timeout period where the attempt will timeout within the given period if the call is not established
    - **startAudio()** – Starts the audio for the established call
    - **setSpeakerMode()** – If the method is set true then it puts the device to speaker mode.
    - **ToggleMute()** – the method checks if the call is muted and removes it out the mute state.

- The **onStart()** call is genuinely responsible for the **visible lifetime** of the activity. The Layout is presented and user interaction with the activity is enabled. The following methods are also called in **onStart()**:
    - **InitializeManager()** – Initialises the SIP Manager which provides the SIP APIs for SIP tasks, such as initiating SIP connections, and provides access to related SIP services.
    - **AddListenerOnButton()** – Click Listeners are added to the image buttons (call, settings and endcall:
        - The call button initiates the call() method through a Click event. The methods then executes the following:
            - It first implements a dialog called CALL_NUMBER. The dialog inflates an edit text box, which allows the user to enter a number to call. This can be accepted by an OK or rejected by a CANCEL Click events.
            - OK – Initiates the call – Implements methods:
                - **startAudio()**
                - **setSpeakerMode()**
                - **toggleMute()**
            - Cancel returns the user to the Main Layout
        - The settings buttons initiates the **updatePreference()** method, which allows the user to enter or modify three preferences: a username, password and a Domain. The method also calls for the **sharedPreference()** class that allows the user to save and retrieve persistent key-value pairs of persistent data types. This data will exist across user sessions even when the activity is destroyed. If all three preferences are returning null onStart() then the alert dialog "UPDATE_SIPSETTINGS_DIAOLOG" is called, which alerts the user to update the settings/preferences.
        - The endcall button initiates the **endcallsession()** method, which ends an active call (incoming or outgoing).

The complete Java Codes and activity layouts are appended in 0

4.3.6. TESTING (LABORATORY)

In this phase, the following questions needs to be answered:

- What will the test cases involve in each phase?
- Does the application fulfil the establish performance requirements?
- Which problems have occurred during testing?
- Which problems can occur during application service delivery?
- What will be the criteria's of testing (Virtual devices or real devices)?

The mVoIP (Laboratory) Network setup can be seen in Figure 4.11. The setup consists of the following elements:

VoIP server: The VoIP Server is a Windows PC running Asterisk Win32 PBX (which is an open source VoIP/SIP Server Software). Additionally, the same PC is running Wireshark which is a free and open source packet analyser for windows and Unix. It is used here for analysis of VoIP calls.

VoIP Client1: The PC that is used as a VoIP Server is also used as one of the VoIP clients. This was achieved by installing X-Lite, which is a freeware VoIP softphone which uses SIP.

VoIP Client2: The PC is used primarily to run X-Lite

mVoIP Client1: The mVoIP client app was installed on the Samsung mobile device.

WAG120N: The Cisco Wireless Access gateway was used as the Access Point for all the devices involved in the network setup.

Figure 4.11: mVoIP Laboratory Setup

4.3.6.1. Network Testing

TEST OBJECTIVE: The test was conducted to confirm that the end-to-end wireless VoIP communication could be established on the LAN and also to establish that network monitoring of packets to analyse if SIP transactions are being executed. The Wireshark outputs are also compared to SIP requests and responses in Appendix B to verify the functionality of the network analyser tool. The test also looks at alarm monitoring in cases of server unavailability.

TEST SETUP: Includes all the elements shown in Figure 4.11 but excludes the mobile device.

TEST CASE 1: Register with the SIP Server

The test involves having the VoIP clients register with the Asterisk VoIP Server. To achieve this all the clients must be provided with SIP credentials that exists on the Asterisk VoIP server.

Figure 4.12 shows a Wireshark capture for a client trying to register with a non-existing username. Response code 404 is received which means “user not found”.

Time	Source	Destination	Protocol	Length	Info
76.5335980	192.168.1.102	192.168.1.101	SIP	588	Request: REGISTER sip:192.168.1.101
76.5345980	192.168.1.101	192.168.1.102	SIP	483	Status: 404 Not found (0 bindings)
94.9720380	192.168.1.102	192.168.1.101	SIP	588	Request: REGISTER sip:192.168.1.101
94.9728110	192.168.1.101	192.168.1.102	SIP	483	Status: 404 Not found (0 bindings)
95.1053960	192.168.1.102	192.168.1.101	SIP	588	Request: REGISTER sip:192.168.1.101
95.1063190	192.168.1.101	192.168.1.102	SIP	483	Status: 404 Not found (0 bindings)

Figure 4.12: Registering with a non-existing username

Figure 4.13 shows a Wireshark capture for a client trying to register with a wrong username. Response codes 401 (unauthorised) and 403 (forbidden) are received

Time	Source	Destination	Protocol	Length	Info
2.35932800	192.168.1.102	192.168.1.101	SIP	588	Request: REGISTER sip:192.168.1.101
2.36023700	192.168.1.101	192.168.1.102	SIP	500	Status: 100 Trying (1 bindings)
2.36081000	192.168.1.101	192.168.1.102	SIP	562	Status: 401 Unauthorized (0 bindings)
2.41520400	192.168.1.102	192.168.1.101	SIP	743	Request: REGISTER sip:192.168.1.101
2.41611600	192.168.1.101	192.168.1.102	SIP	500	Status: 100 Trying (1 bindings)
2.41665200	192.168.1.101	192.168.1.102	SIP	526	Status: 403 Authentication user name does not match account name

Figure 4.13: Registering with a wrong username

Figure 4.14 shows a capture for a SIP client registering using the right credentials. The response 200 OK is sent to the client. This means that the registration was processed successfully.

Time	Source	Destination	Protocol	Length	Info
13.9738110	192.168.1.102	192.168.1.101	SIP	588	Request: REGISTER sip:192.168.1.101
13.9784990	192.168.1.101	192.168.1.102	SIP	500	Status: 100 Trying (1 bindings)
14.0283800	192.168.1.101	192.168.1.102	SIP	608	Status: 200 OK (1 bindings)
25.0440490	192.168.1.101	192.168.1.102	SIP	650	Request: NOTIFY sip:3001@192.168.1.102:5060;rinstance=c5eee1ac6ffc3a87
25.1005210	192.168.1.102	192.168.1.101	SIP	442	Status: 200 OK

Figure 4.14: Registering with the right credentials

TEST OUTCOME: This test has successfully proven that the VoIP network is active and SIP packets can be captured that shows registration activities between the client and the Server. The test cases also shows that using network analysing tools can easily help identify connections to service problems being experienced by the clients.

4.3.6.2. *Alarm Monitoring*

TEST OBJECTIVE: To establish that major alarms, for example server failure can be monitored from a network management system.

TEST SETUP: The test made use of the Asterisk PBX Server and the PBX Manager (as the NMS)

TEST CASE 1: Planned Out of Service

Figure 4.15 shows how the NMS receives a visual alarm when the Asterisk server is shut down by the administrator.

```
Beginning asterisk shutdown....
Executing last minute cleanups
 Nov 27 22:25:26 WARNING[9832]: res_musiconhold.c:1181 ast_moh_destroy: fd 11 Close handles
Nov 27 22:25:26 WARNING[9832]: res_musiconhold.c:1183 ast_moh_destroy: fd 11 handles closed
Asterisk cleanly ending (0).

Disconnected from Asterisk server
Attempting to reconnect ...
```

Figure 4.15: Asterisk Planned Out of Service

TEST CASE 2: Unplanned Out of Service

Figure 4.16 shows how the visual alarms are received by the NMS when a forced shutdown is implemented from an external source.

```
MurongaVLPT/3732/1.2.26.1
 Nov 27 22:28:53 NOTICE[3732]: win32_tapi.c:639 TapiEventThread: Terminating TAPI msg thread...

Nov 27 22:28:53 ERROR[3732]: chan_tapi.c:870 load_module: Unable initialize TAPI

Nov 27 22:28:53 WARNING[3732]: loader.c:416 __load_resource:  chan_tapi.so: load_module failed, returning -1
Nov 27 22:28:53 NOTICE[3732]: win32_tapi.c:237 telephonyShutdown: Closing Message Handler)

Nov 27 22:28:53 NOTICE[3732]: win32_tapi.c:244 telephonyShutdown: Closing lines.

 Nov 27 22:28:53 NOTICE[3732]: win32_tapi.c:273 telephonyShutdown: Shutting down TAPI.

Nov 27 22:28:53 ERROR[3732]: win32_tapi.c:276 telephonyShutdown: TAPI Error: 80000014 on lineShutdown.

Nov 27 22:28:53 WARNING[3732]: loader.c:556 load_modules:  Loading module chan_tapi.so failed!
 Beginning asterisk shutdown....
Executing last minute cleanups
 Nov 27 22:29:22 WARNING[3732]: res_musiconhold.c:1181 ast_moh_destroy: fd 11 Close handles
Nov 27 22:29:22 WARNING[3732]: res_musiconhold.c:1183 ast_moh_destroy: fd 11 handles closed
Asterisk cleanly ending (0).

Disconnected from Asterisk server
Attempting to reconnect ...
```

Figure 4.16: Asterisk unplanned OOS

TEST OUTCOME: Both test clearly shows that the server shutdown activity can be monitored on the NMS.

RECOMMENDATION: The PBX Manager and server can be implemented in a distributed setup instead of having them on one computer. But enhanced monitoring tools needs to be implemented in a case where the system is to be commercialised in order to also monitor the hardware system itself remotely and also to log the events. Logging can also be done on the system if implemented correctly by setting the options shown in Figure 4.17

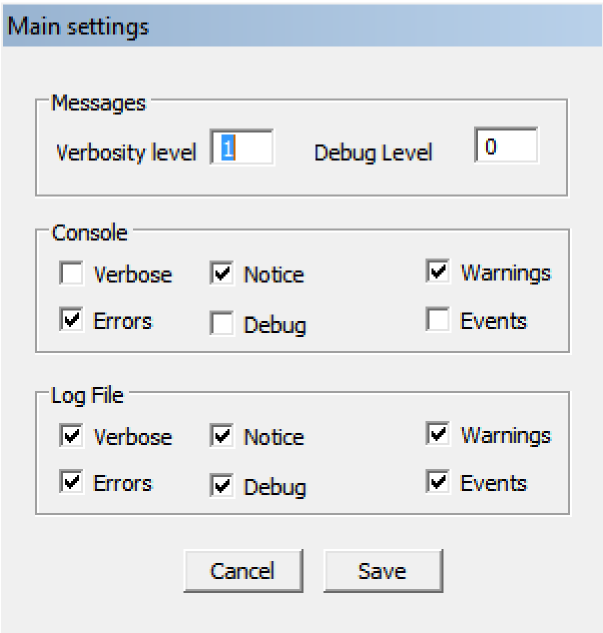


Figure 4.17: Asterisk Logging options

4.3.6.3. Quality of Service

The QoS setting on the WAG120N[15] router allows you to specify priorities for different types of traffic. Lower priority traffic will be slowed down to allow greater throughput or less delay for high priority traffic. The QoS Types available on the router are **UBR** (Unspecified Bit Rate), **CBR** (Constant Bit Rate) and **VBR** (Variable Bit Rate).

TEST OUTCOME: QoS (e.g. jitter or packet loss) could not be tested due to a lack of tools during the testing period. User experience when a call was establish was also very poor, but the cause could not be isolated because of echo caused by the different devices operating next to each other through speakers.

RECOMMENDATION: QoS is a huge factor in real-time communication and is therefore subject to proper testing. The QoS tests should therefore be conducted on a wider network with proper QoS implementation on the routers.

[15] The Linksys Wireless-N Home ADSL2+ Modem Router

4.3.6.4. *Security Testing*

TEST OBJECTIVE: The test was conducted to establish if Network Address Translation function provided by the VoIP server can be used as a security measure, that will be able the clients from accessing the host IP. This is a topology hiding mechanism that can be used to separate the backend network from the client/customer network while providing interworking functions. This prevents client devices from doing any modifications on the backend servers.

TEST SETUP: All devices

TEST CASE 1: Register using NAT on SIP Server

NAT was tested by inserting an external IP and changing the Global Nat field with the following options:

- Never
- Route
- Rfc3851
- true

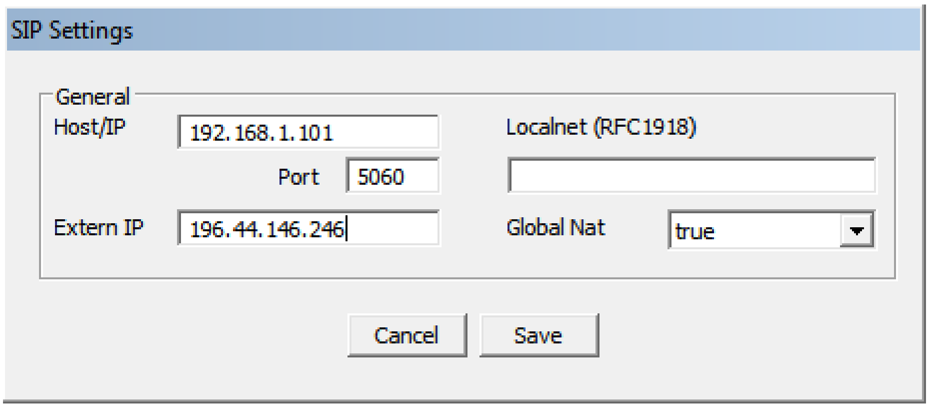


Figure 4.18:Asterisk NAT option

TEST OUTCOME: None of the options had an effect on the LAN. The assumption here is that the parameter is meant for devices behind a NAT router and not the local devices.

RECOMMENDATIONS: The test should be repeated on a wider network or make use of devices that connects to the Asterisk server via the Internet. Service access security is implemented already by providing the user with a username and password. Network access security can also be setup on the access points but it will become a pain to the user when the solution is implemented on the wider network and the user is expected to know the keys to all the access points.

4.3.6.5. Android Virtual Device Testing

As per guidelines from the Android developer site: SIP runs over a wireless data connection and therefore a device must have a data connection (with a mobile data service or Wi-Fi). This means that you can’t test on AVD – you can only test on a physical device (Session Initiation Protocol, 2012).

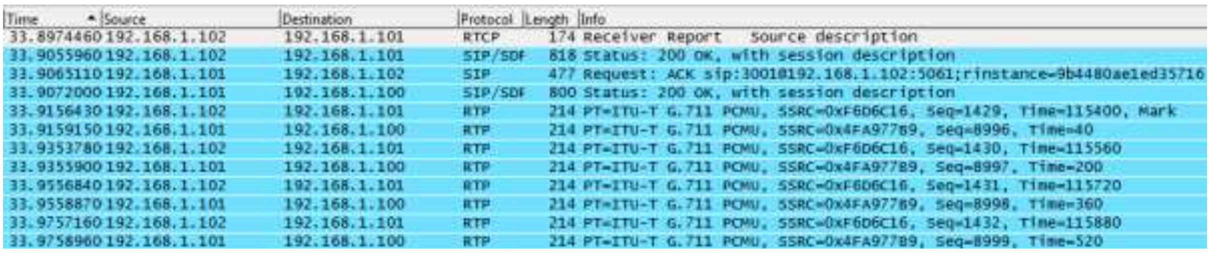
An AVD was still used to test other functions of the applications which are not network related. For example testing layouts, buttons “on click” functions and text edits and views.

4.3.6.6. Physical Android Device Testing

TEST OBJECTIVE: The test was conducted to establish if it was possible to establish one-to-one SIP communication between the mVoIP device and one of the SIP clients running on the laptops. This is necessary to confirm that the mVoIP application is functioning as it should.

TEST CASE 1: Making a Call

Figure 4.19 shows live RTP being exchanged between the mVoIP device and another SIP Client



Time	Source	Destination	Protocol	Length	Info
33.8974460	192.168.1.102	192.168.1.101	RTCP	174	Receiver Report Source description
33.9055960	192.168.1.102	192.168.1.101	SIP/SDF	818	Status: 200 OK, with session description
33.9065110	192.168.1.101	192.168.1.102	SIP	477	Request: ACK sip:3001@192.168.1.102:5061;rinstance=9b4480ae1ed35716
33.9072000	192.168.1.101	192.168.1.100	SIP/SDF	800	Status: 200 OK, with session description
33.9156430	192.168.1.102	192.168.1.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0xF6D6C16, Seq=1429, Time=115400, Mark
33.9159150	192.168.1.101	192.168.1.100	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x4FA977B9, Seq=8996, Time=40
33.9353780	192.168.1.102	192.168.1.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0xF6D6C16, Seq=1430, Time=115560
33.9355900	192.168.1.101	192.168.1.100	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x4FA977B9, Seq=8997, Time=200
33.9556840	192.168.1.102	192.168.1.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0xF6D6C16, Seq=1431, Time=115720
33.9558870	192.168.1.101	192.168.1.100	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x4FA977B9, Seq=8998, Time=360
33.9757160	192.168.1.102	192.168.1.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0xF6D6C16, Seq=1432, Time=115880
33.9758960	192.168.1.101	192.168.1.100	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x4FA977B9, Seq=8999, Time=520

Figure 4.19: Making a call from mVoIP device

TEST CASE 2: Receiving a Call

Figure 4.20 shows RTP packets being exchanged between the mVoIP and another SIP client.

Time	Source	Destination	Protocol	Length	Info
0.39748900	192.168.1.101	192.168.1.102	SIP/SDP	789	Status: 183 Session Progress, with session description
0.47053000	192.168.1.100	192.168.1.101	SIP	375	Status: 180 Ringing
0.49628200	192.168.1.102	192.168.1.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x7082CEB1, Seq=2415, Time=2315400, Mark
0.52026600	192.168.1.101	192.168.1.102	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x7C407014, Seq=30010, Time=160
0.54293100	192.168.1.102	192.168.1.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x7082CEB1, Seq=2416, Time=2315560
0.54323900	192.168.1.101	192.168.1.102	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x7C407014, Seq=30011, Time=320
0.55996600	192.168.1.100	192.168.1.101	SIP/SDP	654	Status: 200 OK, with session description
0.56272000	192.168.1.102	192.168.1.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x7082CEB1, Seq=2417, Time=2315720
0.56292600	192.168.1.101	192.168.1.102	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x7C407014, Seq=30012, Time=480
0.56898000	192.168.1.100	192.168.1.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x60D04439, Seq=7773, Time=3453895683
0.57575400	192.168.1.101	192.168.1.100	SIP	458	Request: ACK sip:30020192.168.1.100:48747;transport=udp

Figure 4.20: Making a Call to mVoIP device

TEST OUTCOME: Both test have proven that it possible to make and receive calls from the mVoIP client device. This tests also proves that the SIP protocol stack on the mobile device is fully operational as it can be seen by the message descriptions in the captures (e.g. 180 Ringing).

4.4. MVOIP DEPLOYMENT STRATEGIES

4.4.1. CURRENT (POTENTIAL) DEPLOYMENTS

Figure 4.21 depicts an mVoIP deployment on Telecom Namibia’s current Broadband network. The user can be connected to the network via Adsl or WiMax. The Wireless Access Point (WAP) is responsible for allowing the User Equipment (UE) to connect to a wired network using Wi-Fi. ADSL and WiMax are the available access technologies that can be used to connect residential users to the IP/MPLS transport network.

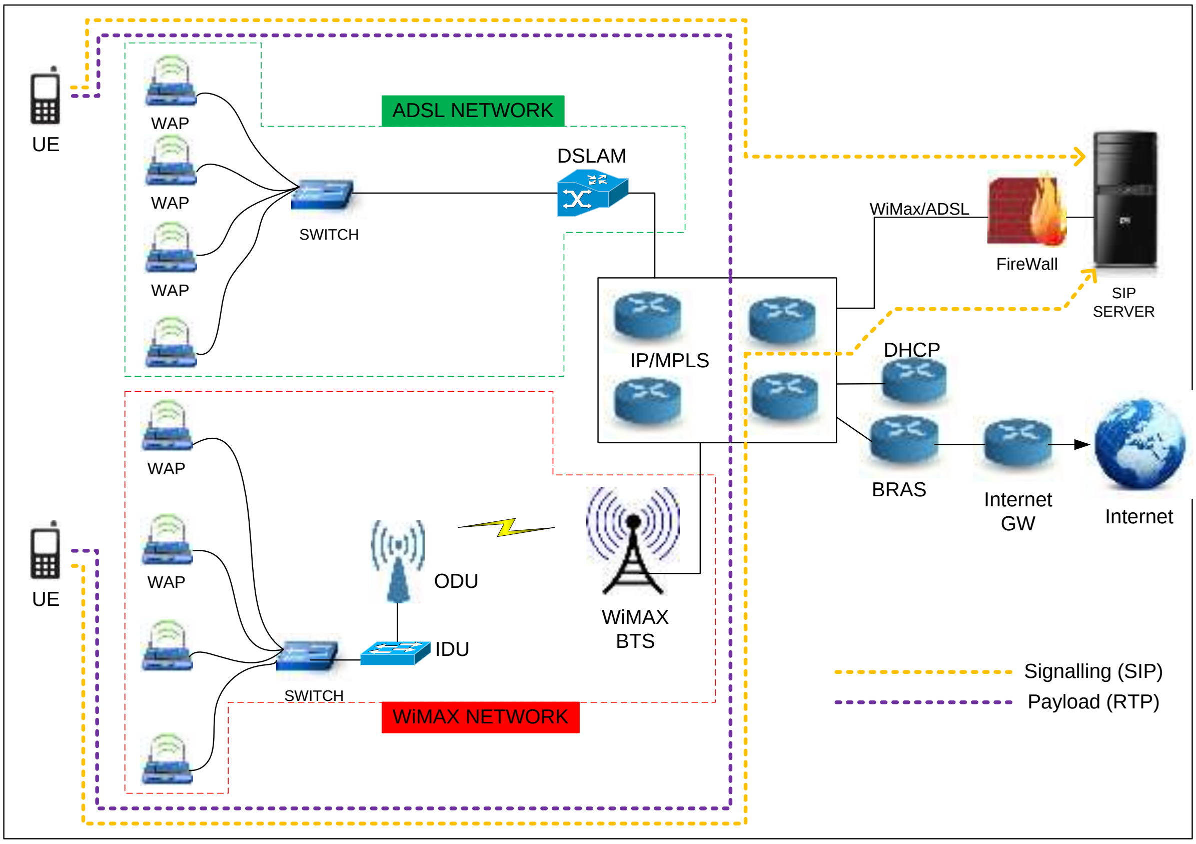


Figure 4.21: mVoIP Deployment on TN BB Network

The SIP Server in this setup can also be connected to the IP/MPLS network via ADSL or WiMax. A Firewall or session border controller is deployed as a point of demarcation between the users and the SIP Server. This is done in VoIP networks to protect the SIP Server from being attacked by users. In this type of setup it is also required that NAT (Network address translation) is executed by the Firewall.

During a voice communication, the Signalling traffic is first established between the user entities involved in the call and the proxy server. After signalling is established and the registration process is finalised, a call can be made. When a call is connected between two users, the actual voice traffic or payload is transported via the MPLS network only and does not reach the SIP Server.

Implementation on a broadband network will improve the service hence the transport network covers a wider area and the network provider already has the means in place to improve the QoS.

4.4.2. *Future (Potential) Deployments*

Figure 4.22 depicts an mVoIP deployment in the mobile data environment. The 3G deployment has been possible with the availability of the 3G network from both Telco's MTC and Telecom Namibia. The LTE (Long-Term Evolution) or 4G has been available from MTC during the current year and will be introduced by Telecom Namibia.

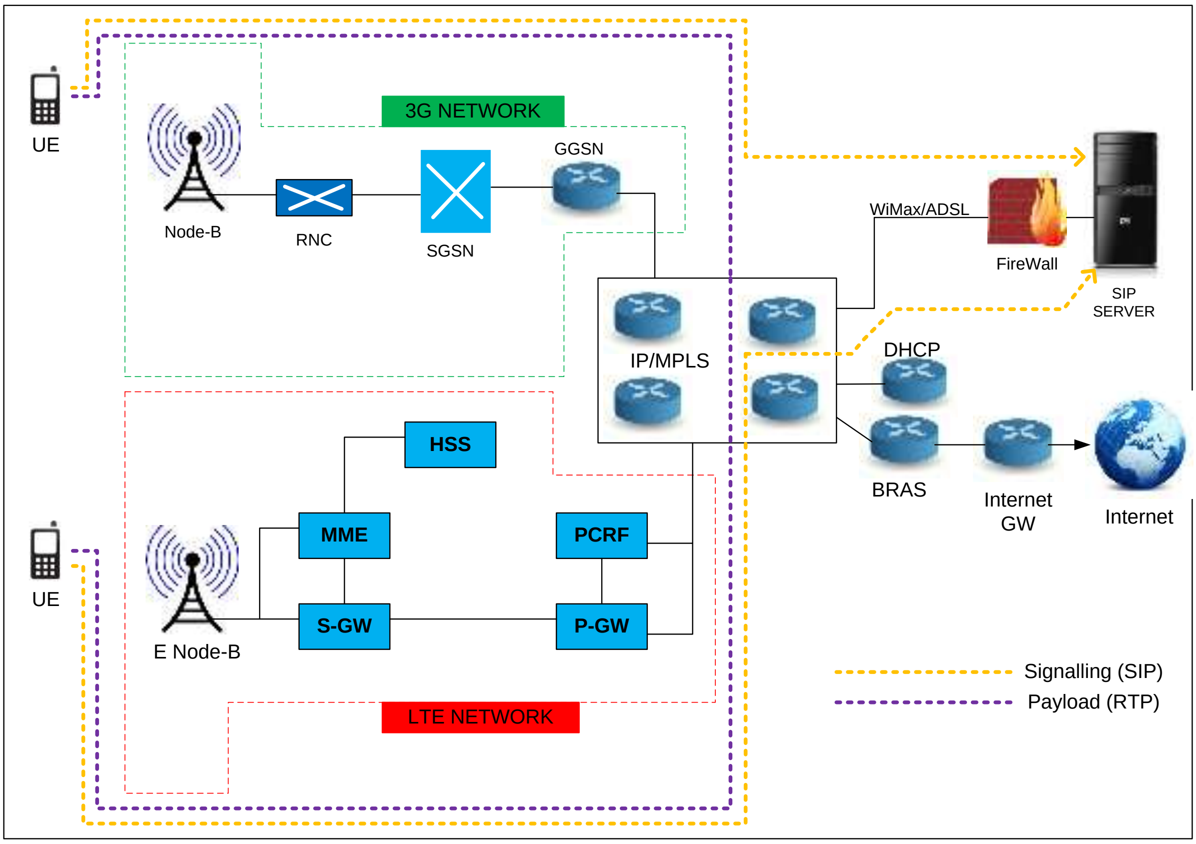


Figure 4.22: mVoIP deployment on 3G/4G Network

With the availability of Broadband Networks, Mobile Broadband in the form of 3G and 4G, one could reason that the Service Providers should have introduced SIP Applications and services long time. But the main question still comes to mind "Are the Service Providers ready to give away their main Service for free?"

Nevertheless a 2G/3G network improves on a lot of the Wi-Fi Capabilities, which are coverage and mobility. Wi-Fi hotspot can cover a cell area of 20 meters, while LTE or 3G can cover cell areas of up to 100 meters (with slow degradation after 30KM). With Access Points that supports 802.11r standards, Wi-Fi can support Voice over Wi-Fi handover or mobility. But still, the 3G and LTE technologies are far more progressive in the mobility service.

5. FINDINGS & DISCUSSIONS

This chapter reviews the research questions that were in the first chapter. It then provides evidence that answers the research questions. The final section of the chapter describes the unanticipated findings of the project.

5.1. ELEMENTS, HYPOTHESIS, RESEARCH QUESTIONS

If the mVoIP solution is implemented then it will be able to provide the users with similar services as those provided by the traditional voice network at less cost and effort. With reference to the assumptions above, the following questions should be addressed:

- To which extent can the mVoIP System provide similar services as the traditional voice network?
  - Does the solution cater for the following essential functionalities of a Voice network:
    - User Security
    - System Security
    - High availability
    - System Management
    - User Management
    - Call Detailed Record
    - Visual and Audible Alarms
    - Network Monitoring
  - What are the CAPEX and OPEX components of the solution?
    - Is implementation cost drastically reduced by the solution?

- How will the user experience be improved?
  - Will there be any improvement in the QoS?
  - How will handover be realised?
  - Can roaming be realised by the mVoIP solution?
  - Are there any new features for the users?
  - Will the mVoIP application be easy to use?
- What criteria's will be used to choose a development platform?
- How about the regulatory framework, how will spectrum licensing be handled?

5.2. EVIDENCE FOUND THAT SUPPORTS OR FAILS TO SUPPORT EACH OF THE HYPOTHESIS, OR RESEARCH QUESTIONS

In support of the assumption made whether the mVoIP solution can provide similar services as a traditional voice network, reference can be made to the call tests made in sub-section **4.3.6.6 Physical Android Device Testing**. It is evident from this tests that the primary service of providing a voice call or any real time communication application can be achieved by the mVoIP solution. Additional services similar to a voice network are also possible as it can be seen in **Appendix C : Asterisk Feature List**.

- On the question of functionalities, evidence is referenced below for each essential function:

    - **User Security –** User Security can be achieved by implementing access user credentials as was discussed in the sub-section **4.3.6.1 Network Testing**.

    - **System Security --** User Security can be achieved by implementing access user credentials as was discussed in the sub-section **4.3.6.1 Network Testing**. Additionally system security can be implemented through NAT setup but this could not yet be proven because the system is currently based on a pilot setup.

    - **High availability –** The implementation of High Availability can be achieved by implementing the VoIP backend systems on Windows Server machines with cluster management. High availability implementation was addressed in sub-section **2.2.1 Requirements of a VoIP Network.**

    - **System Management –** Asterisk PBX management can be realised by using the free PBX manager which comes with the Asterisk package. Some functionalities of the manager can be seen in sub-section **4.3.6.2 Alarm Monitoring.**

    - **User Management –** User Management is also achieved by making use of the PBX manager as it can be seen in Figure 5.1

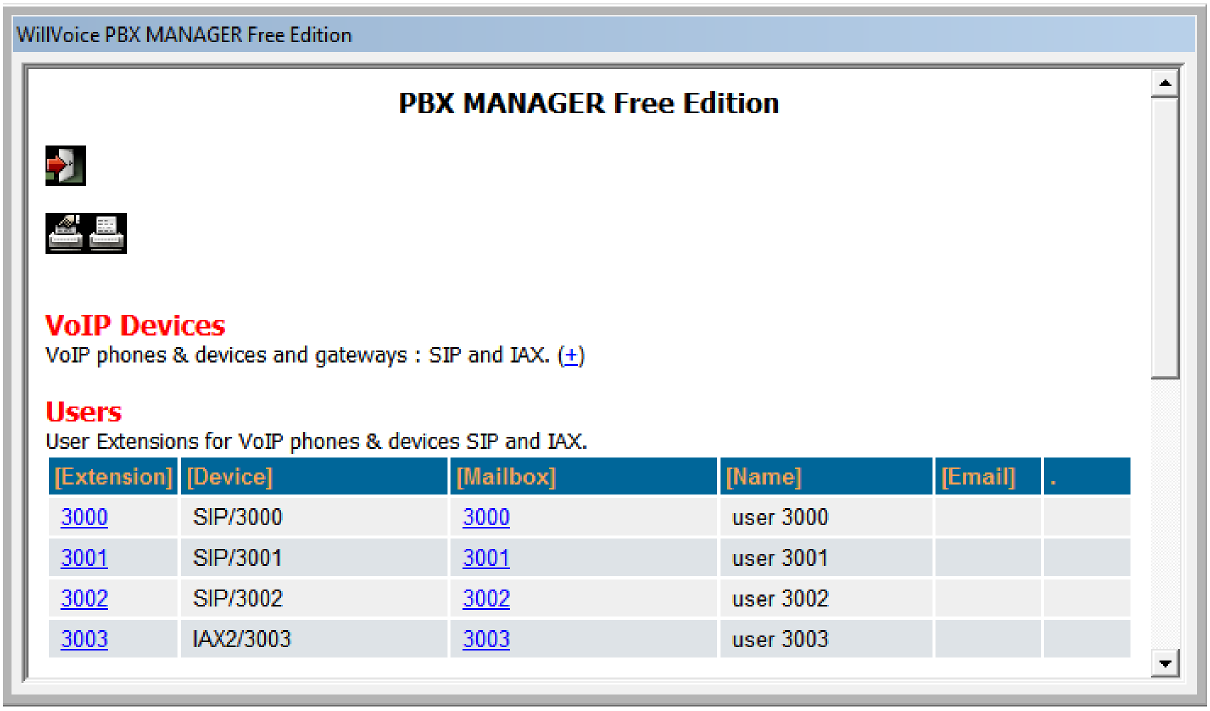


Figure 5.1: PBX Manager -- User Management

- **Call Detailed Record –** Call Detailed Record can be provided by the Asterisk system, the feature is listed in **Appendix C : Asterisk Feature List**

- **Visual and Audible Alarms –** Many systems exist that can be used to implement audible alarms for the mVoIP system but this research does not carry such evidence. One type of visible alarm was recorded in sub-section **4.3.6.2 Alarm Monitoring.**

- **Network Monitoring –** Network Monitoring was achieved by implementing Wire Shark, which is a network analysing tool. More evidence on the functionality of the tool can be seen in sub-section **4.3.6 Testing (laboratory)**.

Operational and Capital expenditures are the two cost components that come with a system implementation. Capital expenditures will be reduced heavily on the backend systems, since normal computers are used which have the following characteristics:

- Lower power consumption then the traditional voice system
- Lower hardware and software cost compared to traditional voice system, with reference to Appendix A
- Free or open-source application software (for service hosting, management and monitoring)

Capital Expenditures savings can be supported by the research. Appendix A describes a voice migration quotation which was received from XON. The offer includes the migration of the EWSD core network elements onto the XON platform. The total cost of this solution is 38 million Namibian dollars. For a fully implemented mVoIP solution back end systems with High-Availability configuration, a total of less than N$100000 can be estimated.

Operational expenditures savings cannot be supported at the moment due the lack of reference information. What is evident on the OPEX reduction at this moment is that the maintenance staff for the backend system can be someone with an A+ certification instead of an Engineering degree and enormous voice system maintenance background.

The **user experience** is crucial in the telecommunication business but since the system testing was based on a laboratory environment, it is too unformed to conclude that the user experience can be improved or not. The following conclusions can be made on the user experience:

- Handover can be realised by implementing a distribution system as highlighted in **2.5 IEEE 802.11 Architecture.**
- Roaming is not a requirement on a VoIP network since the Host is reachable over the network from anywhere in the world.
- The mVoIP application was designed to emulate the normal mobile telephony applications with clear naming conventions, which makes it easy to use.

On the Criteria's used to decide on a **development platform**, the research looked all the possible criteria which need to be explored in order to choose the best platform. Some of the platform characteristics that the research looked at (refer to sub-section **2.7 Factors of Mobile Application Development)** are:

- Targeted Devices
- Learning Curve

On the regulatory front, it should be known that Wi-Fi operates in an unlicensed spectrum and therefore there should not be any regulatory interference in the usage of the spectrum as discussed in sub-section **2.4 Mobile VoIP**. But this benefit also brings up other limitation to the service wide network deployment because of uncontrollable interference.

5.3. UNANTICIPATED RESULTS (FINDINGS)

The Android developer website stated that one of the requirements for developing a SIP application is a mobile device running Android 2.3 or higher (Android D. , 2012). It was discovered during testing of different android devices that not all devices running Android 2.3 or higher includes a full SIP protocol stack and integrated call management services. This was proved by the fact that instantiating a new SIP manager on this platform versions (or mobile devices) always returned a **null** value. The following mobile devices running the respective android versions did not support the SIP protocol stack at the time of writing the paper:

- Samsung Galaxy SII – Android 2.3.4
- Samsung Galaxy SIII – Android 4.0.4
- Samsung Galaxy Note – Android 2.3.6

This was the most challenging part of the project, to find the mobile device with the android version that have the SIP protocol stack. The mVoIP application was finally implemented on the Samsung galaxy Tab 2, running Android 4.0.3. Therefore research into the device capabilities is also very important at the initial stage to prevent unwanted capital expenses.

6. CONCLUSIONS & RECOMMENDATIONS

This chapter starts off with a summary of the work done and continues by revising the deliverables of the project. The chapter then discuss the limitation of the study. Finally the chapter recommends a list of projects to be researched in to help improve the mVoIP system.

6.1. SUMMARY OF THE WORK DONE

A summary of the work done by this project can be highlighted as follow:

- **Literature Review** – The literature review aimed to gather information that could assist to provide direction for the project and also to learn from lessons of the past on how to improve on previous projects. This includes:
    - ***Swotting the requirements of the project*** – This part of the review was carried out to understand the components involved in a fixed and mobile VoIP implementation. It also looked at the features of the components required in a VoIP network in order to understand how to integrate them;
    - ***reviewing the adoption factors of the project*** – This part of the review was carried out to understand the factors that motivates and demotivates the implementation of an mVoIP system, by looking at what similar projects have achieved;
    - ***reviewing the possible architecture of the project*** – This part looks at what are the implementation options of an mVoIP system, by understanding the existing network infrastructures;
    - ***reviewing the development factors of the project*** – This part aimed at understanding the application development requirements such as what OS platform to target and what type of devices to use;
    - ***and comparison of the available open source systems*** – This review looked at the available open source VoIP systems that exist and how they compare with each other in order for the developer to make a choice of which system to use.

- **Evaluation of Voice Systems** – This section compared different criteria's in order to understand the characteristics of the existing systems, how they compare with each other and in which areas should the mVoIP project improve.

- **Design of the mVoIP System** – This section looked at implementing the systems that were selected based on their performance on the comparison charts.

- **Implement the mVoIP System** – This section includes all the tasks that were carried out to integrate an end to end system. This section covers the following tasks:
    - Installing the mVoIP Backend Systems
    - Developing the mVoIP application

- **Testing of the mVoIP System** – The task covers all the tests that was conducted on the project such as
    - Security Testing
    - Network Testing
    - Functionality Testing
    - Virtual Device Testing

6.2. MAIN DELIVERABLES

The main deliverables of this project can be highlighted as follows:

- **An mVoIP network** – The project resulted into an implementation of an ad-hoc mVoIP network, made out of an Asterisk back end server and a Wireless Access Gateway for connecting user devices to the network.

- **An mVoIP application** – The project also provided a mobile VoIP application that can make calls to any other SIP application running on another user device in the ad-hoc network.

- **Documentation** – As part of the project document, a lot of important information that can be reused in industry or education, can be described as follows:
    - ***Mobile Development Process*** – The project has provided information on a mobile development process that can be used by demonstrating each phase.
    - ***Application Program Code*** – The project has also provided the complete program code used to develop the application

6.3. LIMITATIONS OF THE STUDY

The research produced part of its deliverables but could not implement and test a full end-to-end mVoIP network due to the following limitations:

- **Time Limit --** Because of time constraints, the system implementation could only be based on an ad-hoc network implementation making use of few network elements, executing Independent Basic Service Set Testing. Therefore to generalise the result of an end-to-end testing, a wider network is to be constructed so that more functionalities could be tested such as:
    - Calling from two different AP's;
    - Handover functionalities;
    - Testing in a small laboratory setup was also a big problem because the user devices are always next to each other and therefore echo was unavoidable.

- **Lack of Human Resources** – The lack of participants during the testing phase, also limited on how tests could be conducted. One participant has to be the caller and the Callee making it to properly here speech because there is echo at all times. Tests like distance between the parties involved in a session could also not be conducted. Therefore in order to execute better cases more test participants are required.

- **Lack of equipment** – The lack of equipment also prevented a lot of the functionalities from being tested. Some of the functionalities that could have been tested are highlighted below, next to the equipment:
    - Session Border Controller – Security tests, Network address Translations
    - Redundant Server – Failover Tests
    - Gateways – Interconnection testing with PSTN networks
    - Additional Access Points – Handover Tests

6.4. RECOMMENDATIONS FOR FURTHER RESEARCH

6.4.1. ENHANCEMENT ON THE MVOIP PROJECT

After implementing the mVoIP system, the future recommended enhancements are:

- **High Availability** – High Availability is very essential to any voice system, hence users requires services at any time. Therefore High Availability implementations should be explored insightfully and tested. It is only once high availability is implemented that a service provider can guarantee his services
- **Security** – Enhanced security is required on the VoIP network and therefore the mechanisms to implement this without jeopardising the easy of service use (e.g. mobility) should be explored.
- **Handover** – the handover mechanisms should also be explored more and tested since the handover functionality is key to mobility services.
- **Coverage** – a Wi-Fi hot spot can cover a cell area of 20 meters which means that it will be difficult deploy the service in instance like road coverage. Therefore research is necessary in understanding how coverage can be extended.
- **SIP Protocol Stack** – During the development of the mVoIP application, it was realised that not all android mobile devices running android version 2.3.0 or above supports the protocol stack. Therefore it is essential that it is understood how to implement the SIP Stack on all the devices that supports Wi-Fi so that more devices can be targeted.

6.4.2. DUNDi Protocol

Distributed Universal Number Discovery (DUNDi) is a peer to peer system for locating Internet gateways to telephony services. Unlike traditional centralised services (such the simple ENUM standard) DUNDi is fully-distributed with no centralised authority whatsoever. DUNDi is not a VoIP signalling or media protocol but it publishes routes which are in turn accessed via industry standard protocols such as SIP and H.323 (Richardson, 2010).

DUNDi can be used within an organisation to create a fully-federated PBX with no central point of failure, and it also have the ability to allow any additions of resources to a network of communications servers and it will absorb any modifications made to the resources (e.g. gateways) without any additional configuration executed on the DUNDi (Richardson, 2010).

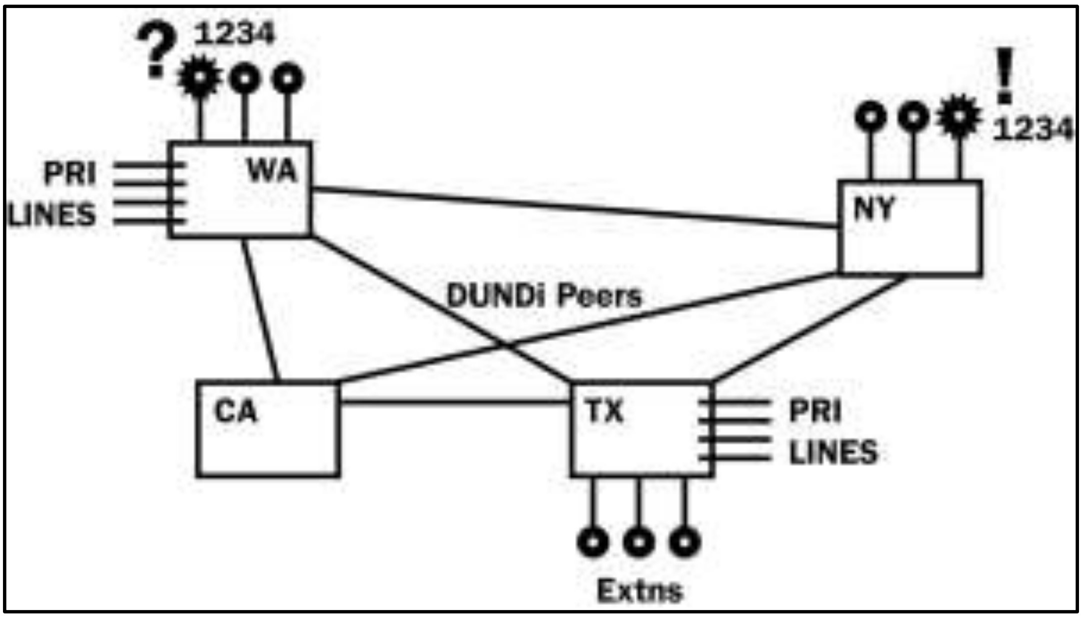


Figure 6.1: DUNDi Publishing Routes (Richardson, 2010)

Richardson (2010) explains DUNDi is a system that can add extra benefits to the mVoIP system as in redundancy of Asterisk or any SIP systems deployed over the Internet. He adds that since it has the ability to locate gateways to telephony services, it can also help identify an available route which is a very important feature in telecommunication, similar to redundancy. The DUNDi system also will help protects the mVoIP service from disruptions; hence it absorbs the modifications done on the resources without any modifications to itself which can be a possible reroute of the VoIP service (Richardson, 2010).

Therefore further research will be able to address the following questions posted by the researcher for functions that are required in the VoIP network:

- Can DUNDi be used to support ENUM services?
- How can the system be used to allow service providers to publish E.164 numbers on the IP network?
- Does the system avail DNS services?
- How will mobility of the user equipment be affected with DUNDi in place?
- How is access security managed with the DUNDi system?

6.4.3. *WI-FI DIRECT*

The Android developer (2012) states that Wi-Fi direct works on Android 4.0 (API Level 14) or later devices that have the appropriate hardware to connect directly to each other via Wi-Fi without an intermediate access point. The website adds that the API allows for the discovery and connection to other devices when each device supports Wi-Fi direct. Communication over Wi-Fi direct is much faster and covers longer distances than a Bluetooth (10 meter) connection. This is useful for applications that share data among users, such as a multiplayer game or a photo sharing application (Android D. , 2012).

As mentioned in the paragraph above the API is normally used by peer to peer applications, but further research can help clarify the following question posted by the researcher:

- In case where a VoIP Server is embedded on the android devices, is it possible then to make use of the Wi-Fi Direct system to do Voice over Wi-Fi communication?
- How will the mobile handset performance be affected if VoIP functionality can be achieved?
- How will mobility be handled in the absence of the 802.11r handover standard?

At short distances of 20 meters it’s not practical to make use of Wi-Fi direct for communication. (Fleishman, 2009)But if the Wi-Fi signal could be re-transmitted (By Wi-Fi range extenders) over longer distances example a couple of kilometres then the implementation of Wi-Fi direct can change the way we do telecommunications in today’s world. The users will not need to register with a SIP/VoIP servers on the network hence they already has a VoIP Server embedded on their device.

6.5. CONCLUSION

The study has satisfied its primary objective of implementing a mobile VoIP which can be hosted by open source software running on inexpensive backend servers. In its current state, the mVoIP system might not be able to replace a traditional voice system, but it does lay the foundation of how inexpensive systems can be implemented. The research has also introduced a step-by-step design approach for mobile applications which can be used in industry for training or implementation purposes.

The findings has also provided sufficient evident that can support the assumption that the mVoIP system can provide telecommunications services. Looking in the future, the mVoIP project should be enhanced in order for it to compete with traditional voice services. This will require additional research and testing in the different aspects of the system for example, Quality of Service, High Availability and Coverage.

Reflecting back on the solution, it is also necessary to do research into the business side of the solution. It should be understood what the total cost of ownership will be at the end of an mVoIP implementation. This will include the enhancement on the solution, the CAPEX and the OPEX of the total solution. Because even though it might seem like the mVoIP system is cheaper on the core network or Back end network, there are still a lot of aspects to consider. For example with a wide network deployment, the backend system will be dimensioned based on the amount of users it must serve. Support for legacy system must also be taken into account which can have a big impact on the monetary value of the solution at the end of the day.

Telecommunication is an essential service of our daily existence which is very expensive for both the user and the provider. Therefore the researcher recommends that telecommunications studies should be incorporated into education especially at tertiary level. By introducing the topic of study to students at an early stage might spark their interest into researching further on how to improve the service and also help reduce its costs for both the user and provider. Because besides the technical factors that the service providers must consider before deploying a VoIP system, there is also the fear of not sufficient and fast return on investment. Therefore more research is necessary in this area.

The development of the mVoIP application has been an eye opener on how to strategically approach the development process. But one important lesson that was learned from the development process is that one should not always go by the information obtained from the Internet, because this can introduce unnecessary delays into the development process. Rather debug the code after completion of every stage of the development process.

The mVoIP implementation was lab based and therefore the service was degraded drastically. Future works requires implementation of the service on 3G or 4G networks, which is assumed will yield better results in the service testing.

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APPENDIXES

Appendix A : VOICE MIGRATION QUOTE FROM XON SYSTEMS

Description	Hardware & Software Pricing (NAD)	Services Pricing (NAD)	Total Total Pricing in NAD
5060 ICS compact IMS for 300K including converged telephony service CTS for 90% residential and 10% corporate users. (Non Geo Redundant)	NAD 6 656 062.41	NAD 13 963.50	NAD 6 670 025.91
5060 MGC-8 for breakout function	NAD 4 243 296.20	NAD 8 872.20	NAD 4 252 168.40
5900 MRF for announcement	NAD 1 517 415.90	NAD 3 183.30	NAD 1 520 599.20
5060 IBC-4	NAD 5 076 986.20	NAD 10 651.20	NAD 5 087 637.40
1360COM for Management	NAD 1 379 678.30	NAD 2 894.40	NAD 1 382 572.70
BTS for CDR collector	NAD 198 555.50	NAD 416.70	NAD 198 972.20
7510 MGW for PSTN breakout gateway function	NAD 7 886 535.80	NAD 16 545.00	NAD 7 903 080.80
VitalQIP	NAD 1 716 000.00	NAD 3 600.00	NAD 1 719 600.00
Spares	NAD 2 654 423.20	NAD 0.00	NAD 2 654 423.20
Integration and IOT test	NAD 0.00	NAD 5 720 000.00	NAD 5 720 000.00

Project Management	NAD 0.00	NAD 1 257 727.90	NAD 1 257 727.90
Total Pricing (Excluding VAT)			**NAD 38 366 807.71**

Appendix B : SIP RESPONSE MESSAGE STATUS-CODE CLASSES (EDWARD OGUEJIOR, 2007)

Code Range	Description
1xx	➢ **Provisional/Informative Response** Provisional response indicate that the associated request was received and being processed. Upon receipt of a provisional response, the request sender should stop transmitting the request. For example, a proxy server can send a response message with a Status-code of 100 upon receipt of an INVITE request. Provisional responses need to be acknowledged with ACK
100	***Trying*** *– Extended search being performed may take a significant time so a forking proxy must send a 100 Trying response.*
180	***Ringing*** *– Destination user agent received INVITE, and is alerting user of a call.*
181	***Call is being forwarded*** *– Servers can optionally send this response to indicate a call is being forwarded.*
182	***Queued*** *– Indicates the destination was temporarily unavailable, so the server has queued the call until the destination is available. A server might send multiple 182 responses to update progress of the queue.*
183	Session in Progress – This response may be used to send extra information for a call which is still being set up.
2xx	➢ **Success** Success responses with Status-codes in the range from 200 to 299 indicate that the request was received, understood and successfully processed. For example a 200 OK response is sent to a User Agent Client when an INVITE request is successfully processed.
200	***OK***
202	***Accepted*** *– indicates that the request has been accepted for processing, but the processing has not been completed*

	204		***No Notification***
3xx		➢	**Redirection** When further action such as a different location is needed to complete a request, redirection responses are used to provide the new location or an alternative service that would satisfy the request. Redirection responses are usually sent by redirect servers. When a redirect server receives a request, it maps the destination address in the request message to one or more addresses retrieved from the Registrar location database and returns the new address list to the originator of the request. The originator is then supposed to re-send the request to the new location.
	300		***Multiple Choices***
	301		***Moved Permanently***
	302		***Moved Temporarily***
	305		***Use Proxy***
	380		***Alternative Service***
4xx		➢	**Client Error** Client error response Status-codes are sent when requests cannot be processed. The request failure could be because of bad syntax in the request message or simply because the request cannot be fulfilled by the responding server.
	400		***Bad Request*** *– The request could not be understood due to malformed syntax.*
	401		***Unauthorized*** *– The request requires user authentication. This response is issued by UASs and registrars.*
	402		***Payment Required*** *– Reserved for future use*
	403		***Forbidden*** *– The server understands the request but is refusing to fulfil it.*
	404		***Not Found (User not found)*** *– The server has definitive information that the user does not exist at the domain specified in the Request-URI. This status is also returned if the domain in the Request-URI does not match any of the domains handled by the recipient of the request.*
	405		***Method not Allowed*** *– The method specified in the request-line is understood but not allowed for the address identified by the Request-URI.*
	406		***Not Acceptable*** *– The resource identified by the request is only capable of generating response entities that have content characteristics not acceptable*

	according to the Accept header field sent in the request.
407	***Proxy Authentication Required** – The request requires user authentication. This response is issued by proxies.*
408	***Request Timeout** – Couldn't find the user in time.*
409	***Conflict** – User Already registered*
410	***Gone** – The user existed once, but is not available anymore*
412	***Conditional Request Failed***
413	***Request Entity too Large** – Request body too large*
414	***Request-URI too long** – The server is refusing to service the request because the request-URI is longer than the server is willing to interpret.*
415	***Unsupported Media Type** – Request body in a format not supported*
416	***Unsupported URI Scheme** – Request-URI is unknown to the server*
417	*Unknown Resource Priority*
420	***Bad Extension** – Bad SIP Protocol Extension used, not understood by the server.*
421	***Extension Required** – The server needs a specific extension not listed in the Supported header.*
422	***Session Interval Too Small** – It is generated by the UAS or proxy when a request contains a Session-Expires header field with a duration below the minimum timer for the server*
423	***Interval Too Brief** – Expiration time of the resource too short*
424	***Bad Location Information***
428	***Use Identity Header***
429	***Provide Referrer Identity***
433	***Anonymity Disallowed***
436	***Bad Identity Info***
437	***Unsupported Certificate***
438	***Invalid Identity Header***
480	***Temporarily Unavailable** – Callee currently unavailable*
481	***Call/Transaction does not exist** – Server received a request that does not match any dialog or transaction*
482	***Loop Detected** – Server has detected a loop*
483	***Too Many Hops** – Max-Forwards header has reached value '0'.*
484	***Address Incomplete** – Request-URI Incomplete*
485	***Ambiguous** – Request-URI is ambiguous*
486	***Busy Here** – Callee is busy*
487	***Request Terminated** – Request Terminated by BYE or cancel*
488	***Not Acceptable Here** – Some aspects of the session description of the Request-URI is not acceptable.*

	489	***Bad Event***
	491	***Request Pending*** *– Server has some pending request from the same dialog*
	493	***Undecipherable*** *– Request contains an encrypted MIME body, which recipient cannot decrypt.*
	494	***Security Agreement Required***
5xx	➢	**Server Error** Server error response Status-codes are sent in cases where the request is valid but the server is unable to fulfil the request. Server internal error (500) and Not implemented (501) are two examples of Server error response Status-codes.
	500	Server Internal Error
	501	Not Implemented – The SIP request method is not implemented here
	502	Bad Gateway
	503	Service Unavailable
	504	Server Time-out
	505	Version Not Supported – The server does not support this version of the SIP protocol
	513	Message Too Large
	580	Precondition Failure
6xx	➢	**Global Failure** When a request cannot be fulfilled by any server, the Global failure response Status-codes are returned. A User Agent Server can return a global failure response with Status-code 603 to decline a request to participate in a session.
	600	***Busy Everywhere***
	603	***Decline***
	604	***Does not Exist Anywhere***
	606	***Not Acceptable***

Appendix C : ASTERISK FEATURE LIST

Asterisk-based telephony solutions offer a rich and flexible feature set. Asterisk offers both classical PBX functionality and advanced features, and interoperates with traditional standards-based telephony systems and Voice over IP systems. Asterisk offers the advanced features that are often associated with large, high end (and high cost) proprietary PBXs. The list below includes a sample of the features available in Asterisk (Digium, 2012).

CALL FEATURES	**CALL FEATURES**	**CODECS**
ADSI On-Screen Menu System	SMS Messaging	Silk
Alarm Receiver	Spell/Say	
Append Message	Streaming Media Access	**VOIP PROTOCOLS**
Authentication	Supervised Transfer	
Automated Attendant	Talk Detection	Google Talk
Blacklists	Text-to-Speech	H.323
Blind Transfer	Three-way calling	IAX (Inter Asterisk Exchange)
Call Detail Records	Time and Date	Jingle/XMPP
Call Forward on Busy	Transcoding	MGCP (Media Gateway Controller
Call Forward on No Answer	Trunking	SCCP (Cisco Skinny)
Call Forward Variable	VoIP Gateways	SIP (Session Initiation Protocol)
Call Monitoring	Voicemail	UNIStim
Call Parking	• Visual Indicator for MW	
Call Queuing	• Stutter Dial Tone for MW	**TRADITIONAL TELEPHONY**
Call Recording	• Voicemail to email	**PROTOCOLS**
Call Retrieval	• Voicemail Groups	
Call Routing (DID & ANI)	• Web Voicemail Interface	E & M
Call Snooping	Zapateller	E & M Wink
Call Transfer	**COMPUTER TELEPHONY**	Feature Group D
Call Waiting	**INTERGRATION**	FXS
Caller ID	AGI (Asterisk Gateway Interface)	FXO
Caller ID Blocking	Graphical Call Manager	GR-303
Caller ID on Call Waiting	Outbound Call Spooling	Loopstart
Calling Cards	Predictive Dialler	Groundstart
Conference Bridging	TCP/IP Management Interface	Kewlstart
Database Store/Retrieve		MF and DTMF support
Database Integration	**SCALABILITY**	Robbed-bit Signalling (RBS)

		Types
Dial by Name		MFC-R2 (Not supported, however a patch is available)
Direct Inward System Access	TDMoE (Time Division Multiplex over Ethernet)	
Distinctive Ringing	Allows direct connection of Asterisk PBX	**ISDN PROTOCOLS**
Distributed Universal Number Discovery (DUNDi)	Zero Latency	
Do not Disturb	Uses commonly Ethernet hardware	AT&T 4ESS
E911	Voice over IP	EuroISDN PRI and BRI
ENUM	Allows for Integration of physically separate installations	Lucent 5ESS
Fax Transmit and Receive	Uses commonly deployed data connections	National ISDN 1
Flexible Extension Logic	Allows a unified dial plan across multiple offices	National ISDN 2
Interactive Directory Listing		NFAS
Interactive Voice Response (IVR)	**CODECS**	Nortel DMS100
Local and Remote Call Agents		Q.SIG
Macros	ADPCM	
Music on Hold	CELT (pass through)	
Music on Transfer	G.711 (A-Law & u-Law)	
• Flexible MP3-based system	G.719 (pass through)	
• Random or linear play	G.722	
• Volume Control	G.722.1 licenced from Polycom	
Predictive Dialler	G.722.1 Annex C licenced from Polycom	
Privacy	G.723.1 (pass through)	
Open Settlement Protocol (OSP)	G.726	
Overhead Paging	G.729a	
Protocol Conversion	GSM	
Remote Call Pickup	iLBC	
Remote Office Support	Linear	
Roaming Extensions	LPC-10	
Route by Caller ID	Speex	

Appendix D : ANDROID APPLICATION CODE