UNIVERSITY OF NAIROBI

College of Biological and Physical Sciences

School of Computing and Informatics

A PROTOTYPE FOR MAKING CALLS BETWEEN MOBILE PHONES IN AREAS WITHOUT CELLULAR NETWORK COVERAGE

BY KENNETH M OMBATI

P53/79024/2015

SUPERVISOR: DR. ROBERT OBOKO

Project report submitted to the School of Computing and Informatics in partial fulfillment of the requirements for the award of the degree of Masters of Science in Distributed Computing Technology

Declaration

I hereby declare that this research report is entirely my own work and has not been submitted for assessment at this or any other university.

………………………………………….   ………………………………………….

Signature: Date:

This report has been submitted for examination with my approval as University Supervisor:

………………………………………….   ………………………………………….

Signature: Date:

Acknowledgement

The success of this project work required guidance and a lot of assistance from many people. My accomplishments so far are because of their guidance and assistance that I would not forget to thank them. I am extremely thankful to God for using them to grant me truthful and illuminating guidance on a number of issues during the project work.

I express my profound thanks to my project supervisor, Dr. Robert Oboko, for his immense contribution. His guidance helped in my research and writing of this research project. I acknowledge the invaluably constructive criticism and friendly advice offered to me by all my panelist who gave me a better insight to conduct this academic research and writing the project document.

I am humbly pleased to get endless inspiration, support and guidance from all non-teaching and teaching staffs of the the University of Nairobi who helped me in directly or indirectly in completing my research project work.

My classmates were also supportive in different ways. The way we assisted and motivated each other was encouraging and they enabled me get this far. To mention just but a few, Erik Van Someren, Rotah Ricky, and Kennedy Siika were instrumental in equipping me with a noble attitude in our interactions during the course.

Finally, I wish to thank all my family members for their never ending support and courage that they have always given me, especially my wife Asharine and daughter Shantel. I would like to thank my employer, Kenya National Examinations Council, for the support and understanding to allow me time off work to concentrate on the project and the entire coursework.

Abstract

Mobile phone voice communications using the traditional Global System for Mobile Communications (GSM) signal methods depends on the cellular network operator signal coverage. There is a communication problem when mobile phones users want to communicate to each other when they are located in an area where there is no GSM signal coverage.

This study objective was to develop and evaluate a prototype that can enable mobile phone calls between users who are located within an area where there is no mobile operator signal coverage. The prototype focuses on using Asterisk server which is an open source framework that can be used for building voice-protocol real-time communications applications, Session Initiation protocol which is a call setup and signaling protocol and a wireless local area network (WLAN).

This study results and analysis demonstrated that the prototype can be successfully used to make phone calls between mobile phones without using cellular network GSM or 3G signals with satisfactory voice quality. According to voice quality standards, the prototype provides acceptable sound quality as evidenced by the measured R-Factor of 93.2. The call establishment time was below a second, which is ideal for real time mobile phone calls communication. The prototype also provided the capability of multiple simultaneous calls between mobile phones. However, mobile phone calls conversations were getting poor and eventually being disconnected when either the calling or called wireless network signal becomes weak.

For the wireless network to cover a wide area, an extended distributed wireless local area network using multiple high performance access points, signal repeaters, as well as using a hybrid of wireless and cabled infrastructure is recommended for future studies. Also recommended is a server class hardware to host the IP PBX server as well as integrating the prototype with existing PBXs that are connected to PSTN lines to bridge mobile phones VoIP calls with external PSTN or GSM mobile phone users.

Table of Contents

List of Tables

List of Figures

CHAPTER ONE : INTRODUCTION

1.1 Background

Mobile phone voice communications using the traditional Global System for Mobile Communications (GSM) signal methods depends on the cellular network operator signal coverage. A mobile operator network is used to provide both coverage and capacity for its users (Sarkar, 2012). The cellular technology used includes GSM, third Generation mobile telecommunication (3G), Code division multiple access (CDMA), and Advanced Mobile Phone System (AMPS).

There is a communication problem when mobile phones users want to communicate to each other when they are located in an area where there is no GSM signal coverage.

Signals from a mobile phone are transferred using radio waves through a network of base stations. No matter how strong the mobile operator radio signals can be especially in urban areas, these areas can have some "dead zones" or “blackspots” which are areas where no signal reception can be received.

Mobile phone operators tend to setup their base stations towers in profitable areas such as populated areas and along major road highways. In general, rural areas in Africa are seen as unprofitable by operators and hence not prioritized by mobile operators in providing mobile communications services.

Voice over Internet Protocol is a communication technology that can be utilized with mobile phones to provide making of voice calls using IP networks infrastructure. It can be implemented using either proprietary or open source protocols and standards.

The objective of this project is to develop a VoIP prototype that can enable mobile phone calls between users who are located within an area where there is no mobile operator signal coverage. The prototype focuses on using Asterisk server to build protocols that provide real-time telephony communications applications, using Session Initiation Protocol which is a call setup and signaling protocol together with a wireless local area network.

With this implementation a wireless network enables the mobile phones to connect to an IP PBX server that provides the signaling and data transport of voice packets using the SIP protocol.

Researchers have given various methods for providing mobile to mobile voice calls without using a mobile operator GSM or 3G signals. Researched methods have been based on using peer-to-peer (p2p), VoIP, Wi-Fi, and Bluetooth means. But most of the methods developed do not provide a complete solution that can allow mobile to mobile voice calls providing important features such as centralized user identification and management. Some researchers have proposed a decentralized mobile voice communication system based on P2P-SIP architecture, but their problem is that they use complex algorithms to achieve client identification and the proposed prototypes support only a single mobile call at a time.

1.2 Problem Statement

There is generally a voice communication problem between mobile phones within an area where there is no mobile signal coverage. According to the Global System for Mobile Communications Association report 10 to 15 percent of the world's population is estimated to lack access to mobile coverage as by the "the Final Frontier of Connectivity" (GSMA, 2014).

Radio signal reception from the mobile phone operators' base stations may be affected or attenuated by thick walls in buildings, underground areas like subway stations and tunnels. Such areas will therefore lack or have poor mobile signal reception. Signal gaps can also be created by the mobile operator base stations' contours in areas where there is no complete signal overlap.

According to the Global System for Mobile Communications Association report (GSMA, 2014), an estimated 43% of the world's population is living outside the mobile coverage area representing 332.2 million people living in this areas.

The East, West and Central Africa and South and Southeast Asia are the primary regions with large populations living outside of mobile coverage, representing 60.5 percent of the global uncovered population (Pierre Biscaye et al., 2015).

1.3 Research Objectives

The main objective of this study is to develop and evaluate a prototype that utilizes a wireless distributed network to provide communication between mobile phones within an area where there is no cellular coverage.

Sub-objectives include:

1. To install a wireless local area network (WLAN) that will be used by mobile phones to offer mobile-to-mobile VoIP communication service.
2. To create an IP PBX server based on the Session Initiation Protocol that will be creating and terminating the mobile phone voice communication sessions.
3. To integrate the IP PBX with the wireless local area network (WLAN) to provide mobile phone communication system.
4. Test and evaluate the developed system if it can be used to provide voice communications between mobile phones without using a mobile operator GSM signal.

1.4 Definition of Important Terms

1. Blackspot

This is an area where there is reduced mobile phone signal. This can be caused by a mobile operator cell tower or base station being far away.

2. Callee

The person or device called by a caller (on the telephone). In some situations, the called party may number more than one: such an instance is known as a conference call.

3. Caller

A person or device that originates a call.

4. Codec

An algorithm or computer program used to convert an analog voice signal to digitally encoded signal. They provide varying sound quality, bandwidth and computational requirements. Examples include G.723.1, G.711a, G.711u. Codecs are used achieve the following:

i. Encryption – Decryption

ii. Compression – decompression or

iii. Encoding – decoding

5. Delay

VoIP delay and latency is the time taken for a mobile phone call sound to travel from a caller's mobile phone to the receiver's mobile phone.

6. Propagation delay

This is the time taken to cover the physical distance from a mobile phone speaker and its listener.

7. Transport delay

This is the time taken by network devices in the communication path to forward frames through a network.

8. Packetization delay

This is the time taken by a coder-decoder (CODEC) to digitize and compress a mobile phone speech at one end and to convert it back to analog form at the listener's mobile phone.

9. Queue/Buffer delay

When the mobile phone call packets are held in a queue because of congestion problem of a network device outbound interface, it causes what is referred to as a *queuing delay*. This delay is usually caused when a network device has more packets to send out than its interface can handle at a given time.

10. Jitter

This is the variations of the mobile phone calls packet arrival time. It is one VoIP problem that exists in packet-based communication networks. Packets are sent out to a mobile phone call listener at a constant rate, however the time between this packet to arrive may vary. This is because the voice packet can be delayed along the network path or travel different paths hence causing them not to arrive at the receiving mobile phone at the same regular intervals.

This time difference between when the expected packets arrive is referred to as jitter. It is one of the most common Voice over Internet Protocol call quality problems.

11. IP PBX

This is a telephone switching system that can be used to transfers calls between VoIP mobile phones users or to external users' land lines using shared local lines.

12. Uniform Resource Identifier (URI)

This is a string or sequence of characters used to identify a resource. The resource could be abstract or physical. It can be identified as a locator, a name, or a name and a locator.

13. Uniform Resource Locator (URL)

This is a reference or an address that refers to a resource that can accessed on the Internet. It therefore a subset of URIs that identify a resource, and also provides a means of locating that resource by describing its primary access method.

14. IP Multimedia Subsystem (IMS)

This is a specification used to describe the networking architecture used for implementing IP-based telephony as well as multimedia communications. They define a framework and architecture used to provide the convergence of voice, video, as well as data and mobile network technologies.

15. Softphones

This is a software telephone or an application program that is used to provide telephone calls communication using devices such as computers.

1.5 List of Abbreviations

ACELP	Algebraic Code Excited Linear Prediction
ADC	Analog to Digital Converter
AMPS	Advanced Mobile Phone System
AP	Access Point
ATM	Asynchronous Transfer Mode
CDMA	Code Division Multiple Access
DAC	Digital to Analog Converter
DCF	Distributed Coordination Function
DSS	Direct Sequence Spectrum
DTMF	Dual Tone Multi Frequency
FHSS	Frequency Hopping Spread Spectrum
GSM	Global System for Mobile Communication
IAX	Inter-Asterisk Exchange
ISDN	Integrated Service Digital Network
IMS	IP Multimedia Subsystem
IVR	Interactive voice response
LTE	Long Term Evolution
MGCP	Media Gateway Control Protocol
MOS	Mean Opinion Score
MP-MLQ	Multi-pulse Maximum Likelihood Quantization
OSI	Open System Interconnection

PBX	Private Branch Exchange
PCF	Point Coordination Function
PDA	Personal Digital Assistant
PESQ	Perceptual Evaluation of Speech Quality
PPM	Pulse Position Modulation
PPP	Point-to-Point Protocol
PSTN	Public Switched Telephone Network
RF	Radio Frequency
RFC	Request for Comments
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SDH	Synchronous Digital Hierarchy
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
VoIP	Voice over Internet Protocol
Wi-Fi	Wireless Fidelity
WLAN	Wireless Local Area Network

CHAPTER TWO : LITERATURE REVIEW

Studies have been carried out describing methods that can be used to enable mobile to mobile voice calls without using the conventional mobile operator GSM or 3G methods, that is, provide voice calls between mobile phones in areas without cellular network coverage. The most prominent methods involve using a wireless local area network (WLAN or Wi-Fi), Bluetooth, and Peer to Peer (p2p) infrastructures. Some of the studies in the related area are described next.

Lu et al., (2011) experimented the configuration of a telephone communications system that used an IP PBX together with the SIP protocol to implement voice calls, voice mail, and conferencing using a local network with softphones. Their study addressed a communication problem using computers but never provided a similar solution that can be used for mobile to mobile phone voice calls. Therefore, in a scenario which involves mobile phones this research will not apply since it was done using computers.

Kbar et al., (2010) implemented a telephony program that uses wireless local area network and peer-to-peer as a means of communication between mobile phones at a negligible cost. Their prototype involved using an algorithm that converts mobile phone numbers into IP v6 addresses. However, their proposed system allowed one mobile phone call per at a time. Therefore, their research would not be applicable in scenarios where multiple mobile phone users would want to communicate to each other simultaneously in a distributed environment.

Sundar et al., (2012) designed and implemented a telephony program that uses a wireless network in Peer-to-Peer and Bluetooth to provide free communication between mobile phones. Their prototype allowed mobile phone users to search other individuals within a p2p Wi-Fi range and to make free mobile phone p2p voice calls. They used an algorithm that converts a mobile number into an IP address and as a means for contacting another mobile phone user using the wireless network and Bluetooth. The limitation with their proposed system is that it allowed one call at a time and it was useful within a short distance since the theoretical range of Bluetooth is ten meters. Therefore, their research would also not be applicable in scenarios where multiple mobile phone users would want to communicate to each other simultaneously in a distributed environment.

Matuszewski and Kokkonen,(2008) implemented a multi-service that allowed a mobile phone user make voice mobile phone calls by using the resources of a peer-to-peer overlay network. As much as their system did not need a centralized server of any kind, the p2p overlay architecture required peer nodes based on a Distributed Hash Table (DHT) connected to mobile phones. This required a more complex network architecture consisting of many nodes connected in a p2p and wireless network in order to provide mobile to mobile calls. Their research therefore has a limitation of requiring a p2p overlay network to be formed before mobile phone users can be able to communicate with each other.

After reviewing literature, it is evident that studies have been carried out related to the problem of providing mobile phone to mobile phone calls without using a network operator signal. Researchers have given various methods for providing mobile to mobile voice calls without using a mobile operator GSM or 3G signals. Previous research work has been based on using peer to peer Wi-Fi and Bluetooth means. But most of the methods developed and researched do not provide a complete solution that can be used in areas where there is no cellular signal to allow mobile to mobile voice calls using a distributed network to provide important features such as long range mobility, centralized user identification and management. Some researchers have proposed a decentralized mobile voice communication system that is based on peer-to-peer and Session Initiation Protocol architectures. However, they have used complex algorithms to achieve client identification and such systems support only a single mobile call at a time. In this regard, it is evident that there is a scope in finding out a more efficient solution to fill the available existing gaps in the literature.

2.1 Implementation Concepts Overview

In this research, the two network based technologies, that is, VoIP and WLAN are chosen to develop a prototype that can be used for mobile phone calls as Voice over WLAN (VoWLAN) to provide mobile phone communications solution to areas where there are mobile network coverage challenges. An open source server with an IP PBX is used to provide the required real-time communications system.

2.1.1 Asterisk IP PBX Overview

Asterisk is an open source structure that can be used to build communication protocols, including real-time voice communications applications. According to Asterisk Organization (2016), its components allows it to provide a variety of functions including an enterprise business phone system, a VoIP Gateway, a call center and as an interactive voice response server.

Smartphones together with the SIP protocol, can therefore use Asterisk server as a registrar and gateway for the mobile phone calls and also for communication to external PSTN lines. Asterisk also provides trunking between Asterisk PBXs using the Inter-Asterisk exchange (IAX2) protocol.

2.1.2 VoIP Overview

Voice over Internet Protocol is a set of protocols that transport voice communications over IP packet-switched networks with acceptable voice quality as well as at lower cost (Cai et al., 2006). It refers to the communication protocols and methods use to provide transmission of voice communications over IP based packet data networks.

VoIP consists made of signaling and transportation of data in an IP infrastructure. SIP, MGCP and H.323 open protocol standards are used for the signaling function, whereas the Realtime Transport Protocol performs the data transport function (Mao et al., 2007).

An analog to digital converter digitizes the voice speech and send the data through a data network where it is reassembled to its original analog state using a digital to analog converter (DAC) as shown by Figure 2-1.

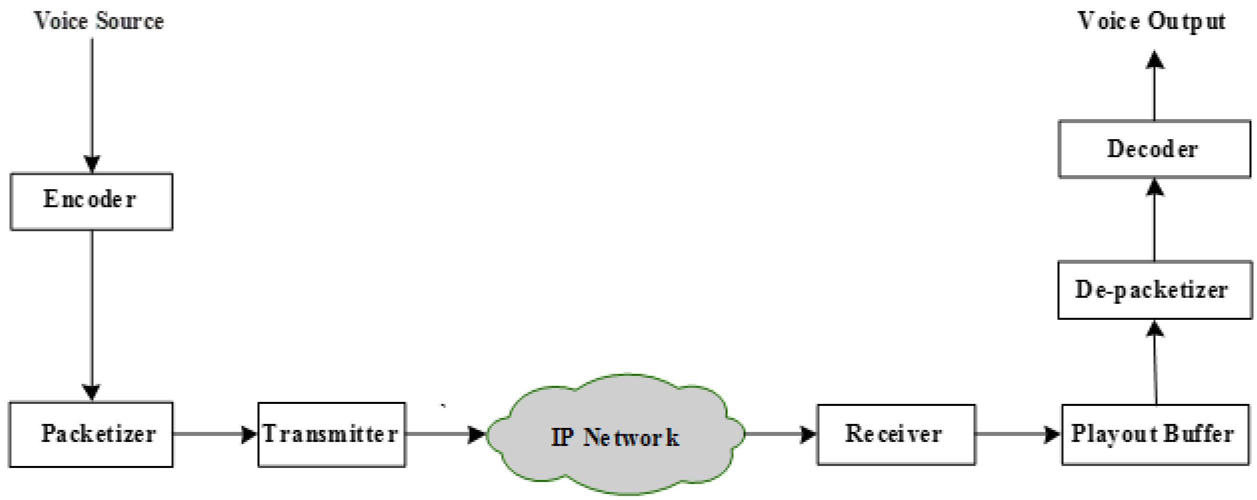


Figure 2-1 *VoIP system*

Source: Int. J. Commun. Syst. 2006; 19:491-508

A. VoIP Protocol Stack

VoIP a Layer 3 protocol, uses Layer 2 protocols including PPP, Frame Relay, ATM, or Ethernet for its transport.

B. VoIP Voice Codecs and Bandwidth Requirements

Codecs are used to digitize and then packetize the voice signals before being transmitted across a network. Some of the codecs also perform compression of the voice signals to preserve network bandwidth requirement. The codecs are implemented in software and/or hardware to achieve various bit rate, complexity and quality requirements (Anderson, 1995).

The bandwidth required during the mobile phone calls conversations depends on the implemented voice codec. Table 2-1 illustrates the various voice codecs and their bandwidth requirements.

In this study the G.711 codec was used because it is natively supported by mobile devices.

Table 2-1 *VoIP voice codecs and bandwidth requirements*

Codec	Sampling Rate	Bandwidth	Payload Size
G.711	8KHz	64Kbps	20ms
G.722	16KHz	48, 48, 64Kbps	30ms
G.723.1	8KHz	5.3, 6.3,Kbps	30ms
G.726	8KHz	24, 42, 40 Kbps	20ms
G.728	Unknown	16Kbps	
G.729	8KHz	8Kbps	20ms
GSM	8KHz	13Kbps	
iLBC iLBC	Unknown Unknown	13.33Kbps 15Kbps	30ms 20ms

Source:(Mao et al., 2007)

Clearly, trade-offs must be considered when deciding a codec to use according to bandwidth and voice quality requirements. This is because low-bit rate codecs provide poorer voice signal quality while higher voice quality is achieved with a higher bandwidth requirement and cost. In this study G.711 codec was used since the 54 Mbps bandwidth between the wireless access point and the mobile phones was the only limiting factor.

C. VoIP Voice Quality

The Mean Opinion Score is a subjective method used to analyze and measure quality of a VoIP call that consists of a scale of 1-5 based on the opinion of human users whereby 5 refers to excellent, whereas 4 is good, 3 is average, 2 weak and 1 is a bad voice quality.

VoIP codecs are given MOS values that are based on their known impairments and rate of conversion. Table 2-2 shows some of the voice codecs and their theoretical maximum MOS values.

Table 2-2 *Voice codecs and MOS*

Codec	Default Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G.711u	64 kbps	20 ms	1.5 ms	2 datagrams (40 ms)	4.4
G.711a	64 kbps	20 ms	1.5 ms	2 datagrams (40 ms)	4.4
G.729	8 kbps	20 ms	15.0 ms	2 datagrams (40 ms)	4.07
G.723.1 MP-MLQ	6.3 kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.87
G.723.1 ACELP	5.3 kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.69

Source: Voice Over IP, 2nd edition

D. Session Initiation Protocol

The Session Initiation Protocol is an application-layer protocol used for control and signaling, as well as to create and terminate mobile phones voice calls over an IP data network (Stallings, 2007). SIP protocol was also used in this study that uses mobile phones since it is capable of handling a user's location.

Since SIP is only a signaling protocol, it is not aware of the actual communication details. For that reason, Session Description Protocol (SDP) is used. According to Penttinen, (2015) the SIP protocol can be used to provide various functions including user's localization, initiation and termination of communication sessions and description of SDP sessions.

A SIP system consists of a client/server as well as network elements that sends SIP requests and also receive SIP responses. This include a user agent client, proxies and a server element that receives requests from clients and then services them by sending back responses. Registrars as well proxies are examples of servers (Stallings, 2007).

A user agent client is responsible for issuing SIP request while a user agent server receives the request and generates responses that accept, reject or sometimes redirect the client request. The registrar server is responsible for accepting registration requests and storing the clients' information in its location service domain.

SIP Uniform Resource Identifier (URI)

SIP uses a text-based syntax similar to the Hypertext Transfer Protocol and Simple Mail Transport Protocol whereby a resource within the SIP network is identified by a Uniform Resource Identifier that is by, user@domain or user@host.

The URI schemes used by SIP are in the form such as sip:ken@uonbi.ac.ke or sip:ken@207.58.138.170 or including a password, a port number and other SIP parameters. For secure communications and transmissions SIP is replaced by “sips”. In such secure cases the SIP messages are transported using Transport Layer Security protocol.

SIP Messages

SIP messages sent from the mobile phones are sent as either responses or requests. Their format indicated by the first line of the request or response message that also indicates a URI where the request or response should be sent.

SIP Requests

SIP communication consist of the methods that are used to enable a user agent and a server locate, invite as well as manage phone calls. A REGISTER request notifies a SIP server it current IP address and also the URL it could receive calls. An INVITE request is used to establish a mobile phone communication session. An ACK response is used to confirm reliable message exchanges. A BYE response is used to terminate and release a call conversation and a CANCEL response terminates a pending request without undoing a completed phone call. Other OPTIONS request messages are used to solicit information regarding the capabilities of the calling mobile phone however it does not perform any call setups.

SIP Responses

SIP response messages are based on the receipt of a corresponding SIP request message. They are used to indicate phone call failure or success as well as the status of the SIP server. A provisional response is indicated with (**1xx**) to show that a request message was received and is being processed. A SIP response that indicates that a SIP request was successfully received and accepted is indicated by a (**2xx**) while a SIP message with response (**3xx**) is a redirection indicating that further action is required to complete the SIP request. A (**4xx**) SIP responses indicates that there is an error from the clients request e.g. a client request containing a bad syntax, while a server error is indicated with a (**5xx**) indicating that the server could not to fulfil a client request. A (**6xx**) SIP response indicates a global failure in the event a client request can longer be fulfilled by any server.

Session Description Protocol

This is a protocol used to describe multimedia sessions. SIP utilizes SDP to describe the requirements of a SIP session. A SIP request contains the SDP parameters and the required values encapsulated within using various fields.

An example of SDP media session related information includes the IP address or DNS hostname, TCP or UDP port numbers, media types, media encoding scheme, session name, bandwidth in kb/s, ending and initiation times and session connection information.

Real-time Transport Protocol

This is a protocol used for sending data that has real-time requirement such as video and phone calls conversations. RTP being an application layer protocol, it uses the User Datagram Protocol for its transport over an IP network.

Useful information contained within an RTP header include timestamp, sequence information and payload type are that are used to reconstruct the data at the receiving mobile phone.

RTP data contains a source identifier that is used in multicast transmissions to identify a member of a group that generates data. A timestamp information is also included to provide proper timing on the receiving mobile phone using a delay buffer and an identifier is also included to identify the payload format for the data transmitted by RTP.

RTP Control Protocol (RTCP)

RTCP is a protocol used to monitor a phone call session quality of service and its participant information. The phone conversation users send each other quality reports, statistics and identity information using the RTCP protocol. A UDP datagram or a lower-level data unit is used to carry the RTCP transmission data.

SIP Signaling

1. SIP Registration Process

A registrar is used to provide a registration service that allows a mobile phone user to make a phone call and be involved in a SIP phone call communication conversation. The SIP registrar server also stores information of the mobile phone clients it has registered and thereby providing their location information. The registrar server therefore has an important role in allowing the mobile phone users change location (Hanzo et al., 2007).

2. SIP Call Setup

Figure 2-2 shows a basic call setup between two SIP mobile phones. This is a successful call beginning with an initial signaling between the mobile phones. A mobile phone user initiates the phone call by sending an invite to the mobile phone being called. An exchange of SIP media information causes an establishment of the phone call conversation and its resulting in both mobile phone users ready to make or receive another phone call.

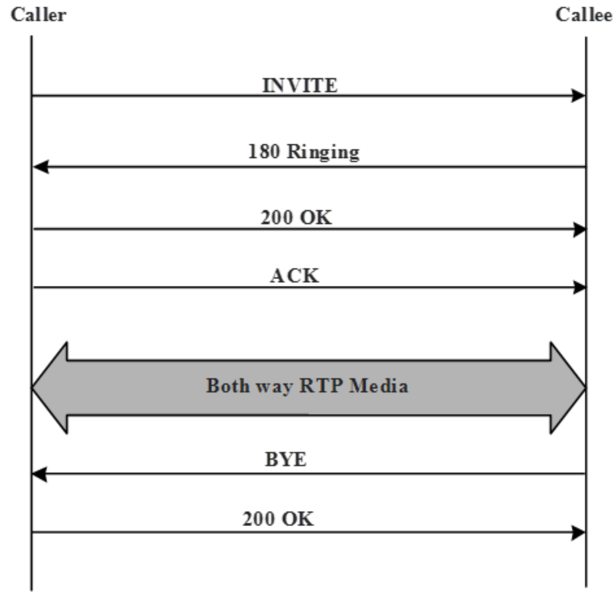


Figure 2-2 *Basic SIP call setup*

3. SIP Call Termination

An active mobile phone call SIP conversation may be terminated by the calling mobile phone user sending a BYE request message. The called mobile phone user terminates the call returning the status SIP response message of 200 with signaling OK code. This is shown in Figure 2-3.

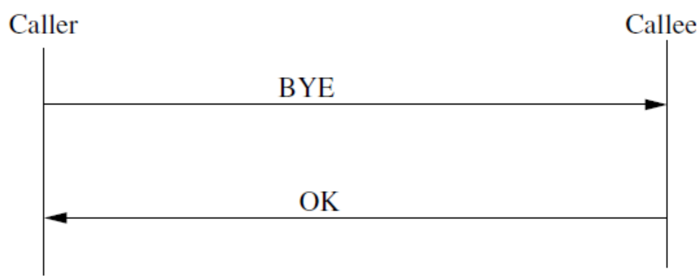


Figure 2-3 *SIP call termination procedure*

4. SIP Call Cancellation

As shown in Figure 2-4, a mobile phone caller can decide to abort a call during its call-setup process. If this is done before the mobile phone caller receives a positive response, the mobile phone caller will issue a CANCEL message to end the call setup process.

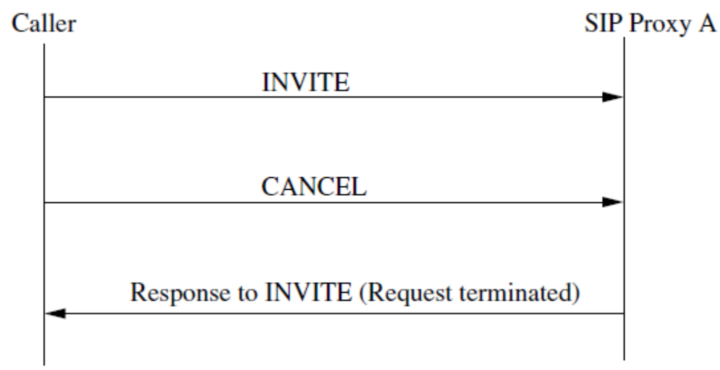


Figure 2-4 *SIP call cancellation procedure*

2.1.3 The Wireless LAN IEEE 802.11 Standard

IEEE 802.11 standard describes the wireless LAN. The IEEE 802.11 Wireless standards are certified by the Wi-Fi Alliance are shown in Table 2-5 and newer versions are still under standardization.

Table 2-3 *WLAN IEEE 802.11 Standards*

Version	**Release Date**	**Name**	**Frequency Band**	**Range**	**Bit Rate (maximum theoretical)**
IEEE 802.11 (legacy)	Jun 1997	WLAN	2.4GHz	~20 meters (~66 feet)	1 Mb/s – 2 Mb/s
IEEE 802.11a	Sep 1999	WLAN (Wi-Fi)	5 GHz	~30 meters (~100 feet)	54 Mbps
IEEE 802.11b	Sep 1999	WLAN (Wi-Fi)	2.4 GHz	~50 meters (~150 feet)	11 Mb/s
IEEE 802.11g	Jun 2003	WLAN (Wi-Fi)	2.4 GHz	~30 meters (~100 feet	54 Mb/s
IEEE 802.11n	Oct 2009	WLAN (Wi-Fi)	2.4 / 5 GHz	~70 meters (~230 feet)	540 Mb/s
IEEE 802.11ac	Dec 2013	WLAN (Wi-Fi)	5 GHz	~50 meters (~150 feet)	1 Gb/s (total for area) and 500 Mb/s (Single station)

CHAPTER THREE : METHODOLOGY

This chapter describes the research design that includes the system implementation architecture associated with the investigation of the research problem and research objectives. It also describes the justification for the application of the specific procedures and methods for data collection and the analysis of the information obtained.

According to (Kothari, 2004), research design is the framework within which research is conducted. In thereby it constitutes the plan for the sampling design, data collection, as well as measurement and analysis of data.

3.1 System Implementation Architecture

Figure 3-1 outlines the implementation architecture of the developed prototype:

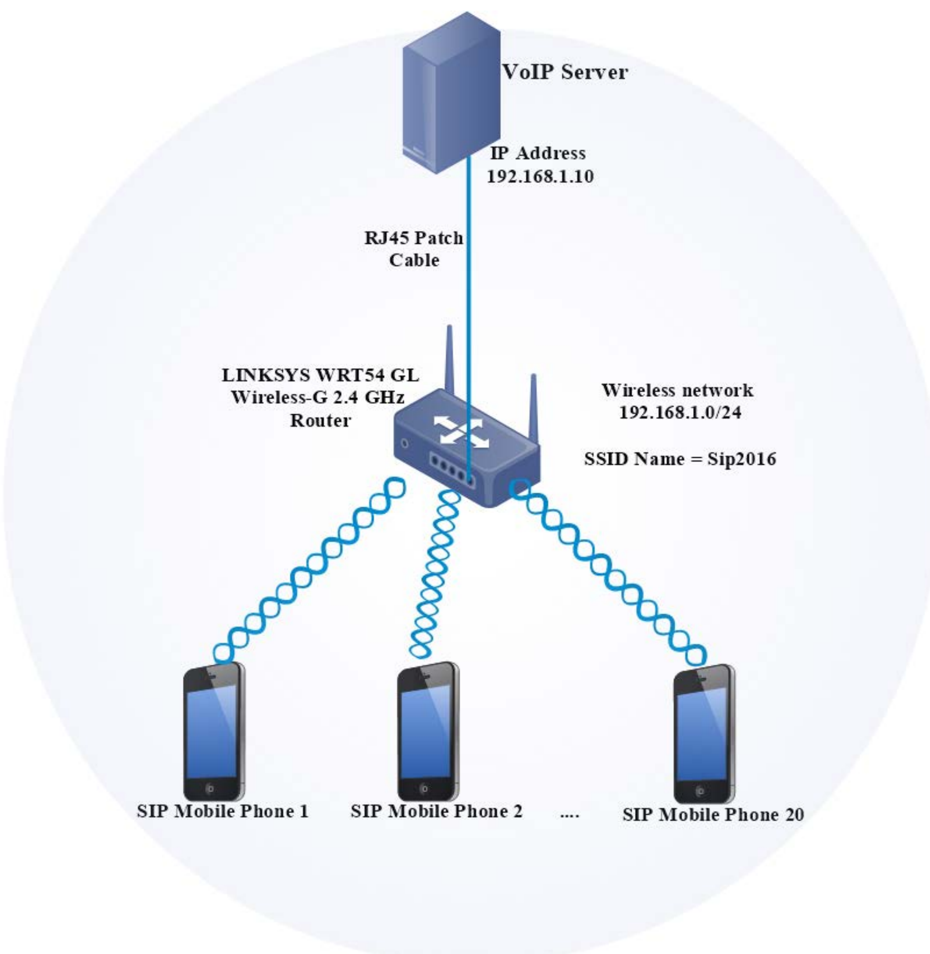


Figure 3-1 *Prototype implementation architecture*

The prototype architecture presented in Figure 3-1 comprises of three parts:

1. Wireless Local Area Network (WLAN)
2. Mobile phones and
3. Asterisk SIP Server

Figure 3-2 shows the prototype architecture of the mobile phones VoIP communications network.

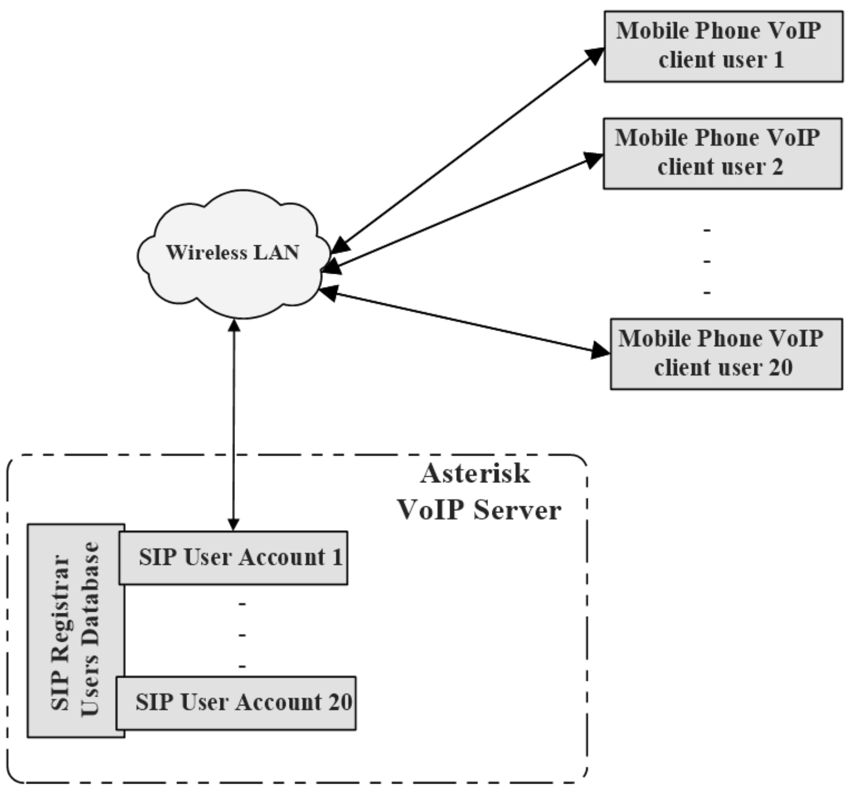


Figure 3-2 *System architecture design*

3.1.1 VoIP Server

An Asterisk VoIP server is used for creating the mobile phone calls communication protocols, including other real-time communications applications such as video. The VoIP signalling function was performed using the Session Initiation Protocol open standard while the Realtime

Transport Protocol used to deliver voice data. Asterisk was installed in a laptop running CentOS with specifications shown in Appendix B.

The server performs the function of receiving client communications requests from the mobile phones and generating responses that accept, reject or redirect this client requests. The server was also used as a registrar to accept client registration requests and then placing the information received including the mobile phones SIP addresses and their IP addresses into its location domain.

In IP multimedia communications codecs digitize and also packetize voice signals prior to their transmission across the wireless IP network. In this study G.711 codec was used because it is natively supported by mobile devices and it gives the best call quality on the basis that it uses no compression at all, and as a result, the call quality sound is acceptable.

Twenty extension accounts database that were to be used by the mobile phones were created in the extensions.conf file and a sample configuration are shown in Appendix C. The user accounts databases were built in the pjsip.auth.conf file and a sample are shown in Appendix D.

3.1.2 Wireless Network

The wireless network was implemented using a LINKSYS WRT54GL Wireless-G 2.4GHz Broadband Router that performed the function of the wireless network access point. The wireless router has an inbuilt 4-Port 100mbps Switch and one 100mbps WAN port. The VoIP server was connected to the wireless network by connecting the server to one of the four switch ports. The access point used supports the IEEE 802.11g and 802.11b wireless standards at a maximum of 54 mbps with a coverage of 40m indoors and 140m outdoors.

The wireless network was configured to broadcast a wireless SSID network named Sip2016 that enabled mobile phones to get connected to the wireless network. The wireless network used a 192.168.1.0/24 subnet. The wireless access point provided an integrated network that enabled communication between the smartphones through the IP PBX server.

3.1.3 Mobile Phones

The research study involved the use of smartphone in the study because of smartphones have embedded wireless network cards supporting IEEE wireless network communications standards and also they are mini-computers that can be enabled to provide multimedia communications using wireless networks.

The mobile phones are the clients that register with the SIP server to enable mobile to mobile communications. Smartphones with Android operating system were used because they provide wireless network access using their embedded Wi-Fi technology and they also have the capability of supporting voice communications.

To simulate using the prototype in an area without a mobile phone operator, airplane mode was enabled on the mobile phones before enabling wireless network access.

3.2 Research Design Introduction

This research study entailed both qualitative and quantitative aspects. This is because to be able to generalize that the system can be used in practice it had to be evaluated based on voice quality provided by the system.

This study has also used a descriptive research design. This is because this study aims at describing the characteristics of a developed prototype by evaluation means. However, a descriptive research to successfully achieve its goals a combination of qualitative and quantitative approaches is adopted.

3.3 Data Collection

To be able to objectively evaluate the developed system, the study required quantitative data to be collected to determine its functionality, completeness, consistence, performance, reliability and usability.

A testbed network as shown in the Figure 3-3 was created to test the system and also enable collection and analysis of the data used between two SIP mobile phone users/clients on the wireless network segment.

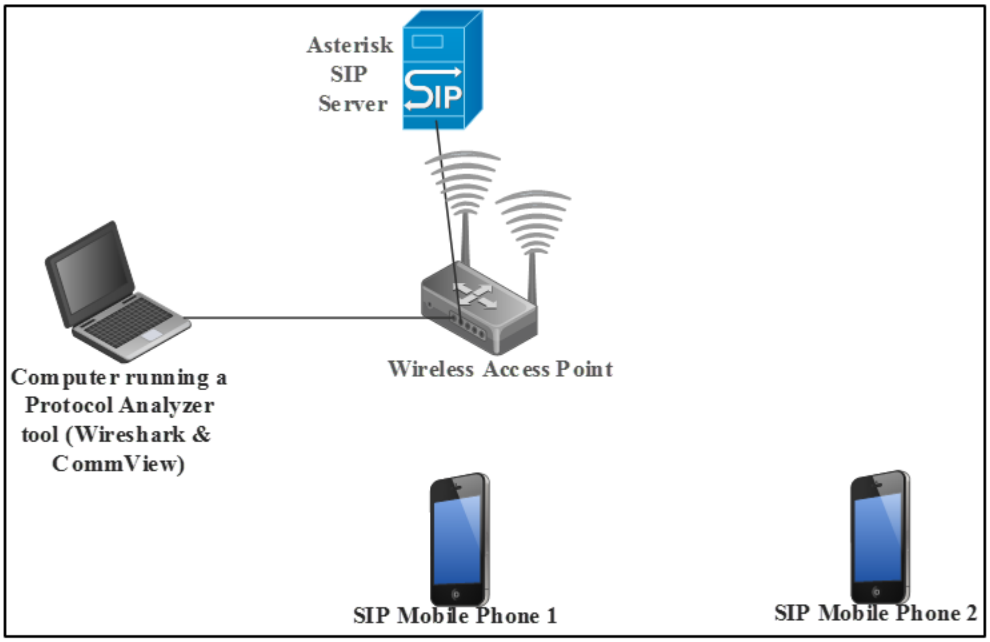


Figure 3-3 *Testbed and data collection network*

3.3.1 Network Data Collection using a Network Analyzer

A network analyzer (sometimes also referred to as a packet analyzer, network monitor, or packet sniffer) is a combination of software and hardware, or in some cases a standalone hardware device, that is installed in a network or computer to provide a detailed statistical data for current and previous network activity and displaying the statistics organized in tabular form, graphic charts or as raw data reports. A network analyzer can also be used to identify packet sources and their destinations as well as monitoring bandwidth utilization at a specific time. It can also be used to search for specific packet data strings, as well as detect unusual packet characteristics and levels of network traffic.

A network analyzer virtually provides a full picture by capturing every packet flowing through a wire or wireless PC or LAN segment. In this study a network analyzer was used for collection of data, in-depth analysis, recording, and playback of SIP and voice communications between the VoIP server and wireless mobile phone devices.

Network analyzers from two vendors namely Wireshark and CommView were used in this study. This is because Wireshark provides more network analysis reports compared to CommView however it Wireshark could not provide R-factor and MOS voice quality reports which can be provided by CommView.

A packet analyzer was used by Bai et al., (2007) to calculate the call establishment time and for SIP packet analysis in a similar study.

In my study Wireshark and CommView were installed on a computer connected to the wireless local area network used by the VoIP SIP server and the wireless mobile phones, and then used in promiscuous mode to monitor, capture and analyze all the RTP stream communication between them as shown in Figure 3-3. These packet analyzers provided interactive browsing and analysis of packet data from the live wireless network and also from a previously captured file. They also provided the lists of all the calls that were found in a trace, and information regarding the start/end time, From/To headers, etc., as well as viewing a call flow in a graphical environment.

The data collected from the packet analyzers enabled measurement and analysis of jitter, latency and packet loss parameters which are important factors that affect a phone call voice quality between the wireless mobile phones.

CHAPTER FOUR : RESULTS AND DISCUSSION

This chapter contains the results, discussion and presentation of the research findings. The main objective of this research work was to develop and create a prototype that utilizes wireless network to provide mobile phone calls communication within an area where there is no cellular coverage. This required installation of wireless local area network that was to be used by mobile phones, building of an IP PBX server based on the SIP protocol and integrating them to provide mobile phone voice communications. Statistical data involving factors that affect voice quality of the prototype such as packet loss, jitter and delay were collected using packet analyzer software (Wireshark and CommView).

4.1 Data Analysis and Presentation of Findings

A protocol analyzer software was used to collect the network communications data, quantitatively analyze it and display the results in tables and graphs. This was achieved by the protocol analyzer following each call from start to finish and collecting data about events that are happening.

In order to evaluate the prototype and determine whether it meets research objectives, it was necessary to establish whether it provides mobile-to-mobile phone call communications, and if it does, measure parameters that affect the voice quality of the communications which include delay, jitter and packet loss.

4.1.1 Sample Call Summary Data

Figure 4-1 shows a call list of four mobile phones conversations captured by the protocol analyzer.

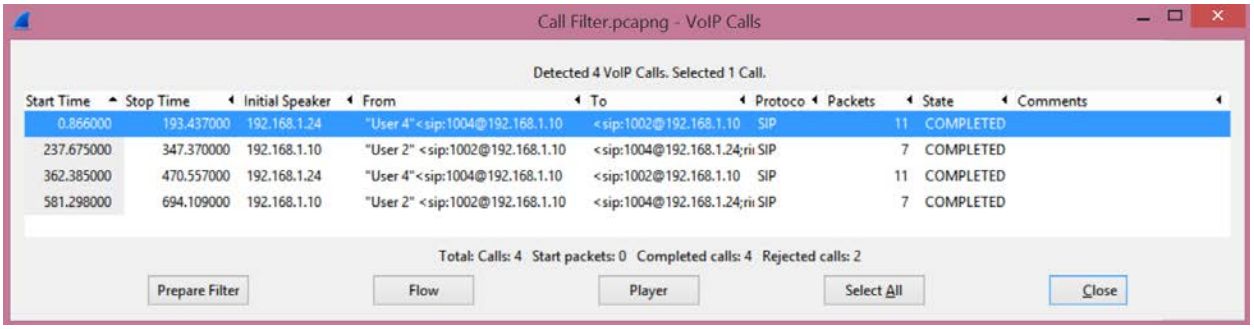


Figure 4-1 *Mobile phones VoIP Calls Captured by a packet analyzer*

The call lists table shows the following information per call; (using the first call in the list for interpretation):

- **Start Time**: This is the relative time of the mobile phone call from the beginning of the capture (after 0.866344).
- **Stop Time**: This is the relative time of the mobile phone call from the beginning of the capture (after 193.4336904 seconds).
- **Initial Speaker**: This is the IP source of the packet that initiated the mobile phone call (User 4 with IP address 192.168.1.24)
- **From**: This is the IP address of the mobile phone that was used to send initiate the mobile phone conversation.
- **To**: This is the IP address of the mobile phone dialed or called.
- **Protocol**: This shows the SIP protocol was used
- **Packets**: This is the total number of packets used in the mobile phone call conversation i.e. 11 packets used in this call.
- **State**: Describes a current mobile phone call state, with the following possible values:
    - IN CALL: state shows that a call is still in progress
    - CANCELLED: state shows that a call was released before it could connect from the originated caller
    - REJECTED: state shows that the call was released before it could connect to the destination side
    - RINGING: state shows that the call ringing, supported only supported for MGCP calls
    - UNKNOWN: state shows the call is in an unknown state
    - CALL SETUP: state shows that the call is in a setup state (Setup, Proceeding, Progress or Alerting)

- o COMPLETED: state shows that a call was connected and then released (used for this example)

- **Comment**: This is a protocol dependent an additional comment.

Figure 4-2 is a Wireshark call flow sequence graph of the eleven packets related to the first mobile phone call in Figure 4-1 of the VoIP call list.

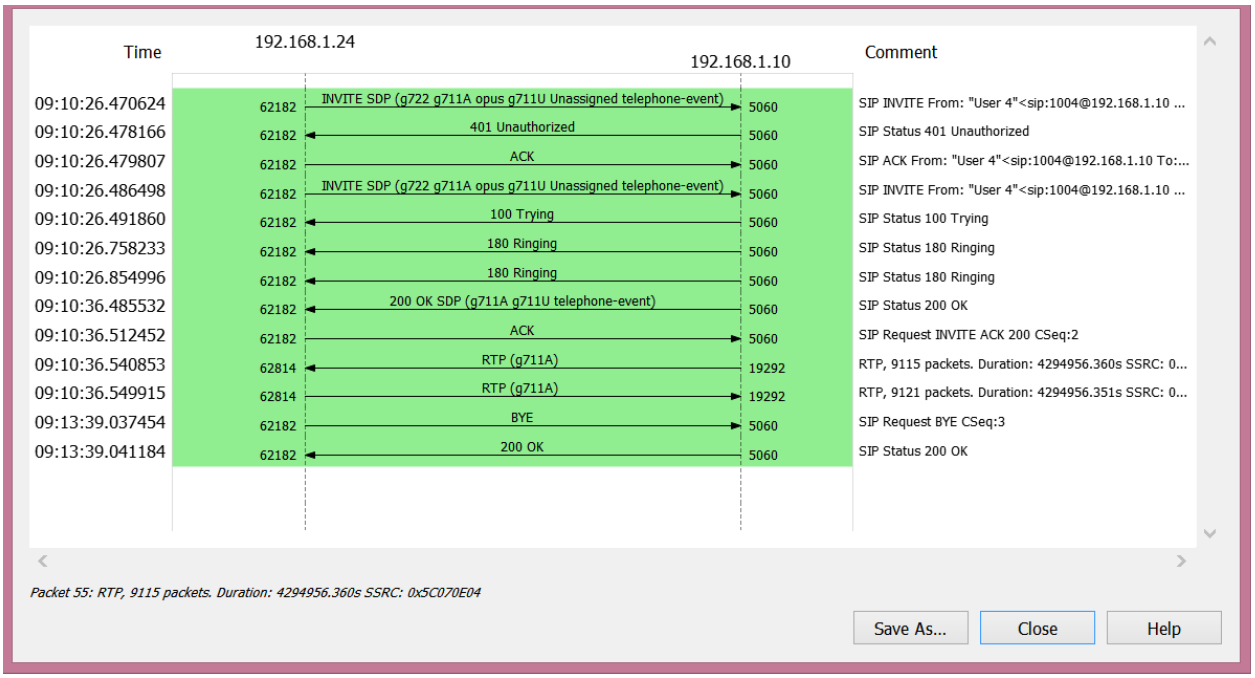


Figure 4-2 *Packet analyzer mobile phone call flow sequence graph*

The Graph shows the following information per call:

- An arrow shows the direction of each mobile phone call packet and the comment column shows protocol dependent information

- The label on top of the arrow shows message type. Also, when available it also shows the used media codec.

- The RTP traffic is summarized by a wider arrow with a corresponding Codec.

- SIP:
  - This shows if phone call the packet is a "Request" or a "Staus".
  - The INVITE packet message shows the "From" and "To" fields
- RTP:
  - This shows the quantity of RTP packets in the phone call stream, with their duration measured in seconds and the SSRC field.

The following figures shows the packet analysis information for the four calls in Figure 4-1 captured using a packet analyzer software.

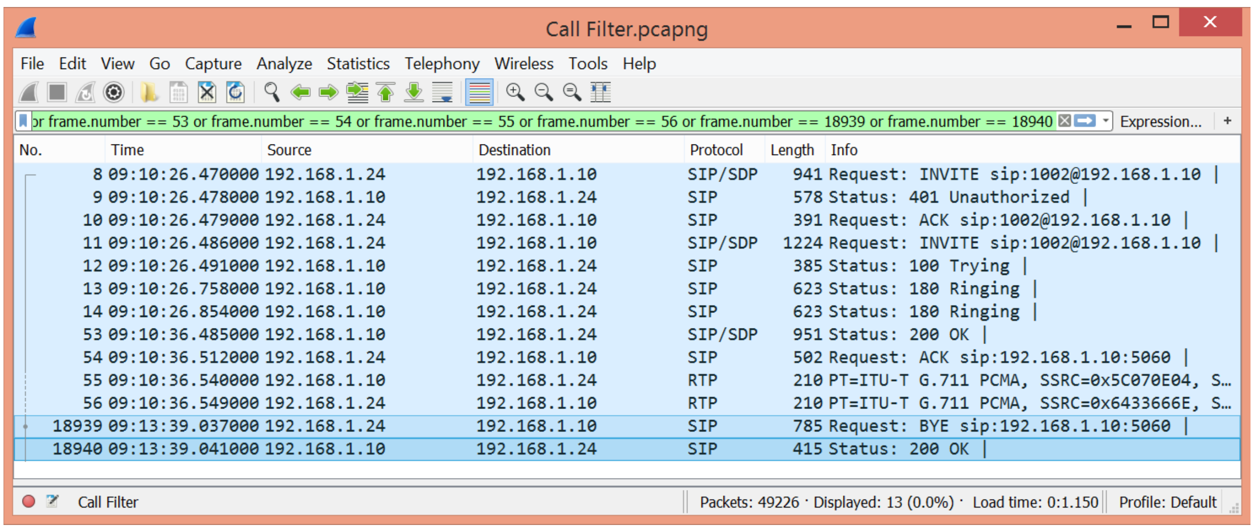


Figure 4-3 *Call number 1 packet analysis*

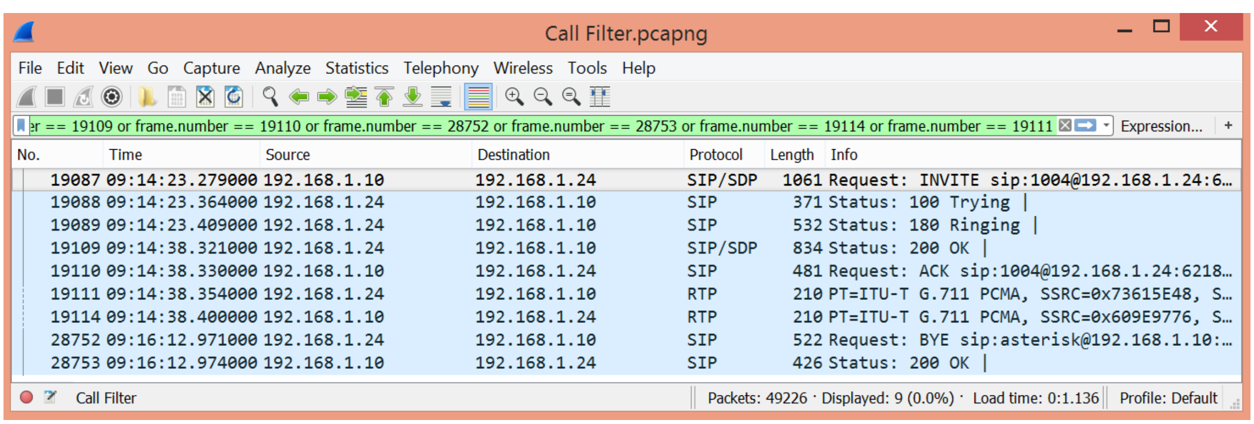
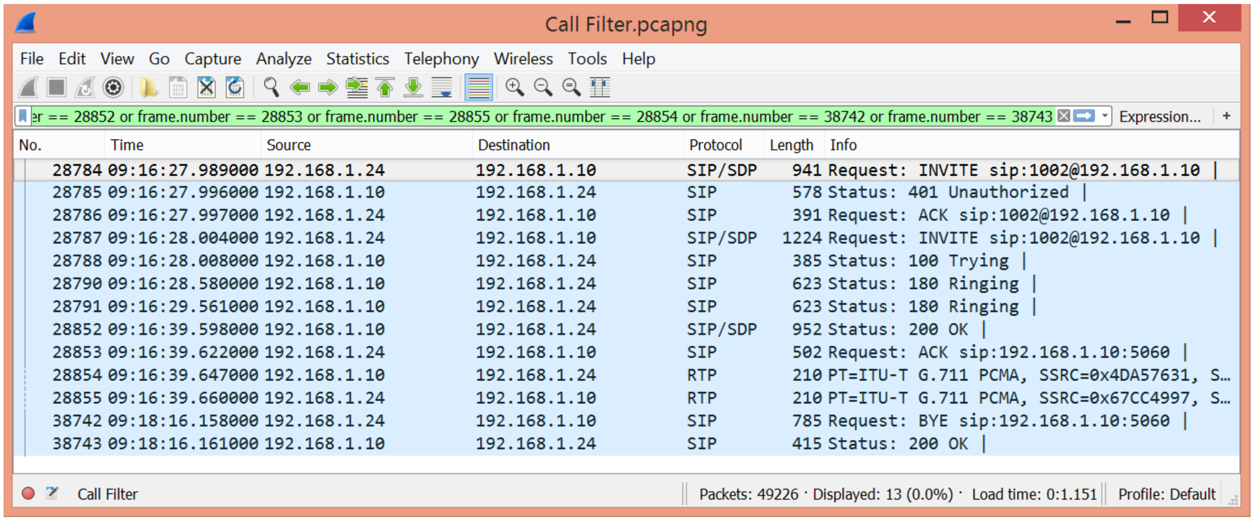


Figure 4-4 *Call number 2 packet analysis*


Call Filter.pcapng

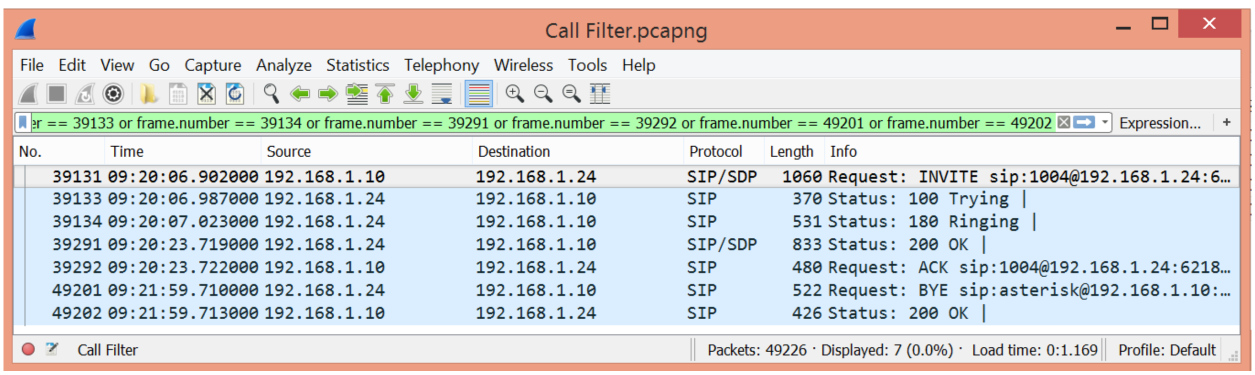
File Edit View Go Capture Analyze Statistics Telephony Wireless Tools Help

er == 28852 or frame.number == 28853 or frame.number == 28855 or frame.number == 28854 or frame.number == 38742 or frame.number == 38743 Expression... +

No.	Time	Source	Destination	Protocol	Length	Info
28784	09:16:27.989000	192.168.1.24	192.168.1.10	SIP/SDP	941	Request: INVITE sip:1002@192.168.1.10 |
28785	09:16:27.996000	192.168.1.10	192.168.1.24	SIP	578	Status: 401 Unauthorized |
28786	09:16:27.997000	192.168.1.24	192.168.1.10	SIP	391	Request: ACK sip:1002@192.168.1.10 |
28787	09:16:28.004000	192.168.1.24	192.168.1.10	SIP/SDP	1224	Request: INVITE sip:1002@192.168.1.10 |
28788	09:16:28.008000	192.168.1.10	192.168.1.24	SIP	385	Status: 100 Trying |
28790	09:16:28.580000	192.168.1.10	192.168.1.24	SIP	623	Status: 180 Ringing |
28791	09:16:29.561000	192.168.1.10	192.168.1.24	SIP	623	Status: 180 Ringing |
28852	09:16:39.598000	192.168.1.10	192.168.1.24	SIP/SDP	952	Status: 200 OK |
28853	09:16:39.622000	192.168.1.24	192.168.1.10	SIP	502	Request: ACK sip:192.168.1.10:5060 |
28854	09:16:39.647000	192.168.1.10	192.168.1.24	RTP	210	PT=ITU-T G.711 PCMA, SSRC=0x4DA57631, S…
28855	09:16:39.660000	192.168.1.24	192.168.1.10	RTP	210	PT=ITU-T G.711 PCMA, SSRC=0x67CC4997, S…
38742	09:18:16.158000	192.168.1.24	192.168.1.10	SIP	785	Request: BYE sip:192.168.1.10:5060 |
38743	09:18:16.161000	192.168.1.10	192.168.1.24	SIP	415	Status: 200 OK |

Call Filter | Packets: 49226 · Displayed: 13 (0.0%) · Load time: 0:1.151 | Profile: Default

Figure 4-5 *Call number 3 packet analysis*


Call Filter.pcapng

File Edit View Go Capture Analyze Statistics Telephony Wireless Tools Help

er == 39133 or frame.number == 39134 or frame.number == 39291 or frame.number == 39292 or frame.number == 49201 or frame.number == 49202 Expression... +

No.	Time	Source	Destination	Protocol	Length	Info
39131	09:20:06.902000	192.168.1.10	192.168.1.24	SIP/SDP	1060	Request: INVITE sip:1004@192.168.1.24:6…
39133	09:20:06.987000	192.168.1.24	192.168.1.10	SIP	370	Status: 100 Trying |
39134	09:20:07.023000	192.168.1.24	192.168.1.10	SIP	531	Status: 180 Ringing |
39291	09:20:23.719000	192.168.1.24	192.168.1.10	SIP/SDP	833	Status: 200 OK |
39292	09:20:23.722000	192.168.1.10	192.168.1.24	SIP	480	Request: ACK sip:1004@192.168.1.24:6218…
49201	09:21:59.710000	192.168.1.24	192.168.1.10	SIP	522	Request: BYE sip:asterisk@192.168.1.10:…
49202	09:21:59.713000	192.168.1.10	192.168.1.24	SIP	426	Status: 200 OK |

Call Filter | Packets: 49226 · Displayed: 7 (0.0%) · Load time: 0:1.169 | Profile: Default

Figure 4-6 *Call number 4 Packets Analysis*

Since the packet analyzer captured all the packets involved in the mobile phone calls conversation, it allows reconstruction of a call from its packets as shown in Figure 4-7 by clicking Play in the RTP Player analysis window. This also enables determination of a sound quality of a given call from the captured packets.

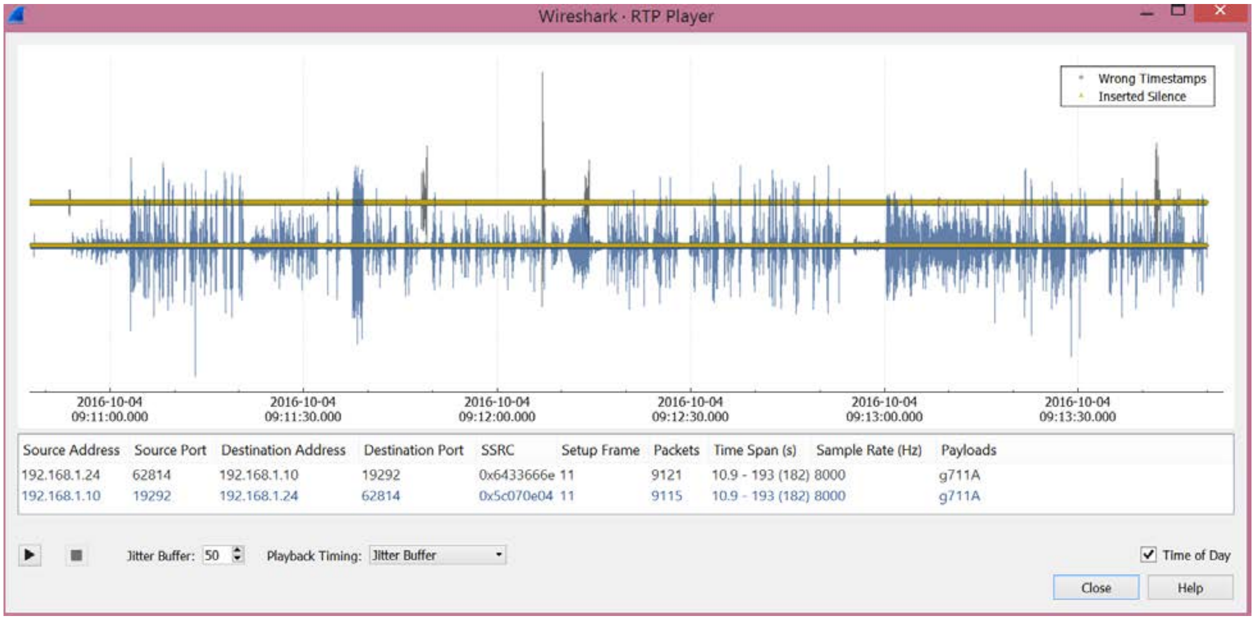


Figure 4-7 *Mobile phone call RTP player analysis*

The data collected by Wireshark packet analyzer from the four mobile phone calls was also used to calculate the Call Establishment Time (CET) based on the difference of time between making a call and getting a ringing tone of the called number using the following formula:

$$\mathbf{T_{CET} = T_{SIP\ INVITE} - T_{SIP\ Ringing}}$$

Table 4-1 *Mobile phone call - Call Establishment Time*

Call No.	**SIP INVITE TIME**	**SIP STATUS RINGING TIME**	**Call Establishment Time (sec)**
1	09:10:26.470000	09:10:26.758000	0.288000
2	09:14:23.279000	09:14:23.40900	0.130000
3	09:16:27.989000	09:16:28:580000	0.591000
4	09:20:06.902000	09:20:07.023000	0.121000
Average Call Establishment Time (for four mobile calls)			**0.282500 (sec)**

It is shows from Table 4-1 that the call establishment time was less than half a second.

For mobile cellular networks, the call setup time is the average interval between the transmission of a Channel Request message from a calling Mobile Station (MS) and the calling MS's reception of the alerting message sent from the Mobile Switching Center (MSC) which is 3 to 4 seconds.

4.1.2 Mobile Calls Analysis: Jitter, Delay and Packet Loss

Figure 4-8 shows the RTP stream analysis function for a selected mobile phone call and its generated statistics using Wireshark protocol analyzer:

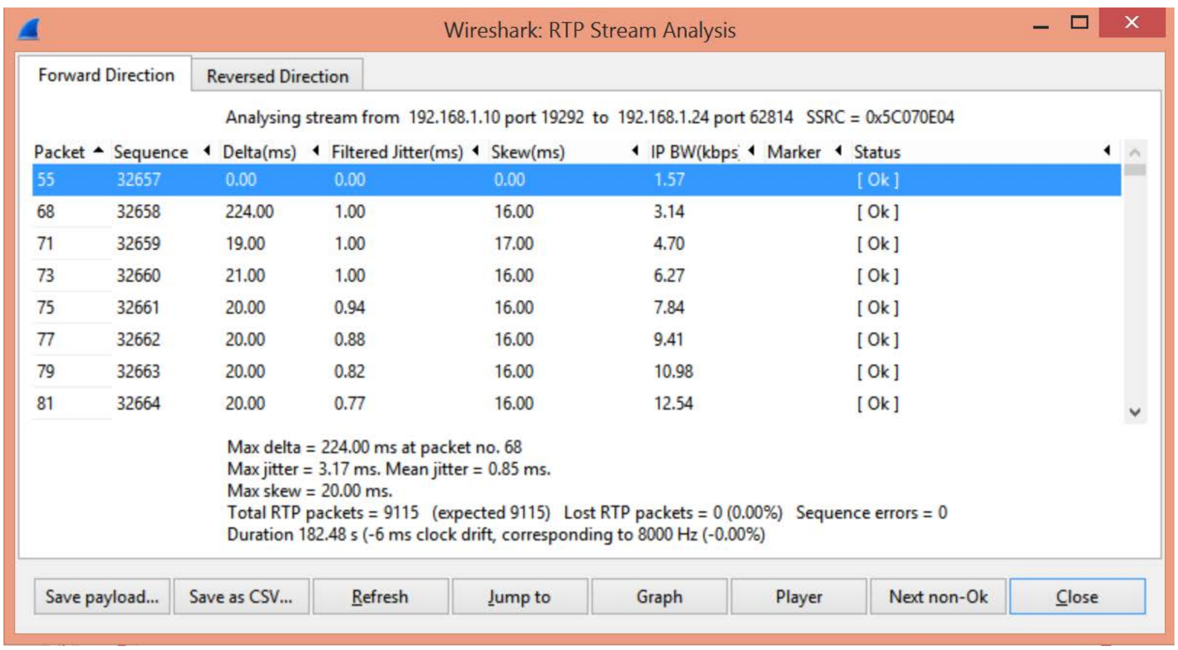


Figure 4-8 *Mobile phone call conversation RTP stream analysis*

The Wireshark sample mobile phone call RTP stream analysis shows data such as, a packet number and its sequence number, and other information based on packet arrival such as time, delay, and jitter. Other statistical data provided include a packet size and consumed bandwidth. As shown in Figure 4-8 in addition to the per packet statistics, also captured are the general packet statistics of the captured mobile phone call, with minimums and maximums values for delta, jitter and as well as clock skew. Also included is a percentage of lost packets.

For the 9115 RTP packets for the sample mobile phone call conversation, the RTP stream analysis gives the following information:

a) Maximum delta of 224.00 milliseconds at packet number 68

b) Maximum jitter of 3.17 milliseconds

c) Mean jitter of 0.85 milliseconds

d) 0.00% packet loss

The above results indicate the prototype provided optimum VoIP sound quality for mobile phone call conversations. This is because the maximum allowable duration of jitter is 40 ms before sound quality deterioration occurs whereas the prototype produced a maximum jitter of 3.17 ms with an average mean jitter of 0.85 ms. In addition, for adequate VoIP quality the required packet loss must be between 1% and 3%. The results from the RTP analysis of the 9115 packets indicated there were no packet loss during the mobile phone conversation. The round trip time measured for a packet to be transmitted from one mobile phone to another phone was established to be a maximum of 8 ms and an average of 2 ms. The maximum duration of latency that a VoIP system can sustain without deterioration is 150 ms in any one direction.

4.1.3 Mobile Calls Analysis: MOS and R-Factor

The Mean Opinion Score (MOS) was introduced in order to provide a subjective quantitative assessment of the quality of mobile phones conversations. In addition, it also indicates the observed voice quality of a mobile phones VoIP communications, classifying the voice call quality as a number in the range 1 to 5 after listening to a call in a “quiet room”.

The MOS was initially meant to provide the arithmetic mean of the quality of a phone conversation by people who tested a phone conversation and ranking its quality. In this study a packet analyzer was used to calculate the MOS value. This is because MOS is highly subjective and an unreliable method to determine the quality of a VoIP conversation prototype. In addition, it does not measure other important network parameters such as network jitter, packet loss and delay.

The packet analyzer was also used to determine the R-Factor, which provides a numerical score derived from other VoIP metrics that affect voice quality, including network jitter, packet loss and delay. R–Factor provides a more precise voice quality measuring tool compared to the MOS since it does a scaling from 0 to 120 compared to the limited MOS factor scale of one to five.

Table 4-2 demonstrates the comparison of the MOS and R-Factor of a perceived phone call quality.

Table 4-2 *MOS comparison value*

User Satisfaction Level	MOS	R-Factor
Maximum using G.711 codec	4.4	93
Very satisfied	4.3-5.0	90-100
Satisfied	4.0-4.3	80-90
Some users satisfied	3.6-4.0	70-80
Many users dissatisfied	3.1-3.6	60-70
Nearly all users dissatisfied	2.6-3.1	50-60
Not recommended	1.0-2.6	Less than 50

Table 4-3 and Table 4-4 display RTP stream analysis (one phone stream in a given direction) between two mobile SIP calls with a measured R-Factor of 93.2 in both directions as measured by CommView packet analyzer software. According to the voice quality standard, the prototype system provides an excellent voice quality for mobile phone SIP calls as indicated with the achieved R-Factor of 93.2. This is also attributed to the G.711 codec that was implemented within the VoIP server, the 0.00% packet loss measured and the excellent wireless signal quality of the mobile phones. G.711 codec gives the best call quality for SIP mobile calls on the basis that it uses no compression at all but at a much higher bandwidth utilization. A mobile phone average bandwidth consumed per call as analyzed by the packet analyzer was 67.09 kbps.

Table 4-3 *MOS and R-Factor values for mobile phone call number 1*

Src IP	Src Port	Dest IP	Dest Port	Duration	RTP Packet Count	Avg BW (kbps)	Total Traffic (bytes)	Max Jitter (ms)	Lost Packets	MOS Score	R-Factor
192.168.1.10	19180	192.168.1.24	59950	02:06.8	6333	66.97	1,089,276	44.51	0	4.4	93.2
192.168.1.24	59950	192.168.1.10	19180	02:06.8	6342	67.19	1,090,824	1.33	0	4.4	93.2

Table 4-4 *MOS and R-Factor values for mobile phone call number 2*

Src IP	Src Port	Dest IP	Dest Port	Duration	RTP Packet Count	Avg BW (kbps)	Total Traffic (bytes)	Max Jitter (ms)	Lost Packets	MOS Score	R-Factor
192.168.1.24	52416	192.168.1.10	12610	02:02.0	6103	67.18	1,049,716	1.23	0	4.4	93.2
192.168.1.10	12610	192.168.1.24	52416	02:02.1	6100	67.01	1,049,200	2.91	0	4.4	93.2

4.1.4 Mobile Calls Voice Quality Analysis in Poor Wireless Signal Conditions

It was observed that the mobile phone call conversation was getting poor and eventually being disconnected when either the calling or called mobile phone moves away from the wireless access point. This is because the wireless network signal power directly affects mobile phones transmission rates and thereby having a direct effect on the selected VoIP codec and quality of a mobile phone call conversation. It was also observed that low wireless signals caused by moving away from the wireless access point causes the mobile phones to select low wireless transmission rates, thereby nullifying the codec chosen initially and causing a high number of packet retransmissions, out of order delivery of packets and consequently high jitter and poor voice quality.

The mobile phone conversations were getting disconnected because the lower wireless signals cause a drop of transmission speeds causing the wireless signal quality to become poor and consequently causing the mobile phone call conversation to get disconnected.

Because wireless packets are sent over a wireless network, a mobile phone call packets are often lost causing the mobile phones and wireless access point hardware to retransmit the same packets repeatedly.

CHAPTER FIVE : CONCLUSIONS AND RECOMMENDATIONS

This chapter contains the conclusions and recommendations of the research work as well as suggestions for future research.

According to literature, previous works have described methods that can be used to enable mobile to mobile phone calls without using the conventional mobile operator GSM or 3G methods. The most prominent literature involves the use of wireless local area network, Bluetooth, and Peer to Peer (p2p) infrastructures. These prototypes allow one call per connection and useful within a short distance since the theoretical range of Bluetooth is ten meters. Therefore, such prototypes would not be applicable in scenarios where multiple mobile phone users would want to make phone calls to each other simultaneously in a distributed network providing important features such as long range mobility, centralized user identification and management. Some researchers have proposed a decentralized mobile voice communication system based on peer-to-peer SIP architecture, but such prototypes require complex algorithms to provide mobile phone client identification and only support a single mobile phone call at a time.

5.1 Conclusion

The objective of this study was to develop a prototype that utilizes a wireless network to provide phone calls between mobile phones within an area where there is no cellular network coverage.

To achieve this objective a wireless local area network was installed to provide the mobile phones a communications network. An IP PBX server built based on the SIP was used to create and also terminate the mobile phone calls communication sessions. Finally, the IP PBX and wireless local area network were integrated to provide the mobile phone communications prototype to be used for voice calls.

This study results and analysis demonstrated that the prototype can be successfully used to make phone calls between mobile phones without using cellular network GSM or 3G signals (e.g. Safaricom, Airtel or Orange) with satisfactory voice quality. According to voice quality standards, the prototype provides acceptable sound quality as evidenced by the R-Factor of 93.2 measured by a protocol analyzer. The call establishment time was below a second, which is an ideal for real time mobile phone calls communication.

The prototype provided the capability of multiple simultaneous calls between mobile phones and this was achieved by use of wireless network that allowed multiple mobile phones access and the IP PBX server implemented with the SIP protocol.

It was observed that the mobile phone call conversation was getting poor and eventually being disconnected when either the calling or called mobile phone user moved away from the wireless access point. This was because the mobile phones' wireless signal strength affected the effective transmission speed and choice of voice codec and consequently the quality of a mobile phone call.

5.2 Limitations of the Research

This study used one wireless access point to provide the wireless network required for the connectivity of the mobile phones. A wireless access point that utilizes the IEEE 802.11b standard provides a wireless network range of 50 meters. This implied that mobile phones could not use the prototype for mobile phone calls beyond 50 meters away from the access point.

A laptop was used to host the IP PBX server even though it met the minimum hardware system specifications. A server class hardware is strongly recommended for such a system to achieve optimum performance including better voice quality and high availability.

5.3 Recommendations

An extended distributed wireless local area network using multiple high performance access points, signal repeaters, as well as using a hybrid of wireless and cabled infrastructure is recommended for future studies. This will expand the wireless network coverage required by the phones in order to cover a wide area.

A server class hardware is recommended for future studies to host the IP PBX server instead of a laptop used in this prototype. A server class hardware will have the capacity to provide optimum performance including better voice quality and high availability requirements.

Future studies may integrate the prototype with existing PBXs that are connected to PSTN lines. This PSTN gateways will bridge between the mobile phones VoIP calls with PSTN lines to support external calls to PSTN or GSM mobile phone users.

REFERENCES

ALCATEL 2003. IP Telephony Design Guide.

ANDERSON, D. V., HARRIS, R.W., AND CHABRIES, D.M., 1995. Evaluation of a hearing compensation algorithm,. *Proc. IEEE ICASSP,*.

ASTERISK ORGANIZATION. 2016. *Asterisk Application* [Online]. [Accessed 31-03-2-16 2016].

BAI, Y., AMINULLAH, S., HAN, Q., WANG, D., ZHANG, T. & QIAN, D. A Novel Distributed Wireless VoIP Server Based on SIP. Multimedia and Ubiquitous Engineering, 2007. MUE '07. International Conference on, 26-28 April 2007 2007. 958-962.

CAI, L., XIAO, Y., SHEN, X., CAI, L. & W. MARK, J. 2006. VoIP over WLAN: voice capacity, admission control, QoS, and MAC. *International Journal of Communication Systems,* 19**,** 491-508.

CHAOUCHI, H. & LAURENT-MAKNAVICIUS, M. 2009. *Wireless and Mobile Network Security : Security Basics, Security in on-the-shelf and Emerging Technologies,* London Hoboken, NJ, ISTE; Wiley.

DAVIDSON, J. 2000. *Voice over IP Fundamentals,* 201 West 103rd Street Indianapolis, IN 46290 USA, Cisco Press.

HANZO, L., SOMERVILLE, F. C. A., WOODARD, J. P., IEEE COMMUNICATIONS SOCIETY. & SAFARI TECHNICAL BOOKS. 2007. Voice and Audio Compression for Wireless Communications. 2nd ed. Chichester, England ; Hoboken, NJ: John Wiley,.

KBAR, G., MANSOOR, W. & NAIM, A. Voice over IP Mobile Telephony Using WIFI P2P. Wireless and Mobile Communications (ICWMC), 2010 6th International Conference on, 20-25 Sept. 2010 2010. 268-273.

KOTHARI, C. R. 2004. *Research Methodology,* New Age International Publishers.

LU, T., NICOLAS, D., QIAO, Q., JIHUA, L., JIANNAN, Z., JING, G. & JI'AO, Z. Study of SIP Protocol Through VoIP Solution of “Asterisk”. Mobile Congress (GMC), 2011 Global, 17-18 Oct. 2011 2011. 1-5.

MAO, G. F., TALEVSKI, A. & CHANG, E. Voice over Internet Protocol on Mobile Devices. Computer and Information Science, 2007. ICIS 2007. 6th IEEE/ACIS International Conference on, 11-13 July 2007 2007. 163-169.

MATUSZEWSKI, M. & KOKKONEN, E. Mobile P2PSIP - Peer-to-Peer SIP Communication in Mobile Communities. 2008 5th IEEE Consumer Communications and Networking Conference, 10-12 Jan. 2008 2008. 1159-1165.

PENTTINEN, J. T. J. 2015. *The Telecommunications Handbook : Engineering Guidelines for Fixed, Mobile and Satellite Systems*, West Sussex, United Kingdom.

PIERRE BISCAYE, JARON GODDARD, MATHEW LANE & ANDERSON, C. L. 2015. Review of Mobile Coverage. *Evans School Policy Analysis and Research (EPAR),* EPAR Brief No. 261 Mathew Lane, & C. Leigh Anderson.

SARKAR, S. K. 2012. *Wireless Sensor and Ad Hoc Networks Under Diversified Network Scenarios*, Artech House, Inc.

STALLINGS, W. 2007. *Data and Computer Communications,* Upper Saddle River, NJ 07458, Pearson Prentice Hall.

SUNDAR, S., KUMAR, M. K., SELVINPREMKUMAR, P. & CHINNADURAI, M. Voice over IP via Bluetooth/Wi-Fi Peer to Peer. Advances in Engineering, Science and Management (ICAESM), 2012 International Conference on, 30-31 March 2012 2012. 828-837.

Appendix A: Asterisk Server Hardware Specifications

The VoIP server used in this system was installed in a laptop with the following specifications

- Laptop Model: HP EliteBook 2760p
- Operating system: CentOS release 6.5 (Final) with SHMZ Final Release
- Memory: 8 GB RAM
- Processor: Intel Core i5 2540M CPU @ 2.60GHz
- Asterisk FreePBX 12.0.76.3
- Network IP Address: 192.168.1.10
- 500 GB hard disk

Appendix B: SIP User Accounts Databases

Table 0-1 *SIP user accounts database*

	Extension Number	Display Name
1.	1001	User 1
2.	1002	User 2
3.	1003	User 3
4.	1004	User 4
5.	1005	User 5
6.	1006	User 6
7.	1007	User 7
8.	1008	User 8
9.	1009	User 9
10.	1010	User 10
11.	1011	User 11
12.	1012	User 12
13.	1013	User 13
14.	1014	User 14
15.	1015	User 15
16.	1016	User 16
17.	1017	User 17
18.	1018	User 18
19.	1019	User 19
20.	1020	User 20

Appendix C: Sample Mobile Phones Accounts Databases

```
[ext-local]

include => ext-local-custom

exten => 1001,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1001/ringtimer)}" > "0"]?${DB(AMPUSER/1001/ringtimer)}:${RINGTIMER_DEFAULT})})

exten => 1001,n,Macro(exten-vm,novm,1001,1,0,0)

exten => 1001,n(dest),Set(__PICKUPMARK=)

exten => 1001,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)

exten => 1001,n,Goto(${IVR_CONTEXT},return,1)

exten => 1001,hint,PJSIP/1001&Custom:DND1001,CustomPresence:1001


exten => 1002,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1002/ringtimer)}" > "0"]?${DB(AMPUSER/1002/ringtimer)}:${RINGTIMER_DEFAULT})})

exten => 1002,n,Macro(exten-vm,novm,1002,1,0,0)

exten => 1002,n(dest),Set(__PICKUPMARK=)

exten => 1002,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)

exten => 1002,n,Goto(${IVR_CONTEXT},return,1)

exten => 1002,hint,PJSIP/1002&Custom:DND1002,CustomPresence:1002


exten => 1003,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1003/ringtimer)}" > "0"]?${DB(AMPUSER/1003/ringtimer)}:${RINGTIMER_DEFAULT})})

exten => 1003,n,Macro(exten-vm,novm,1003,1,0,0)

exten => 1003,n(dest),Set(__PICKUPMARK=)
```

```
exten => 1003,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)
exten => 1003,n,Goto(${IVR_CONTEXT},return,1)
exten => 1003,hint,PJSIP/1003&Custom:DND1003,CustomPresence:1003

exten => 1004,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1004/ringtimer)}" > "0"]?${DB(AMPUSER/1004/ringtimer)}:${RINGTIMER_DEFAULT})})
exten => 1004,n,Macro(exten-vm,novm,1004,1,0,0)
exten => 1004,n(dest),Set(__PICKUPMARK=)
exten => 1004,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)
exten => 1004,n,Goto(${IVR_CONTEXT},return,1)
exten => 1004,hint,PJSIP/1004&Custom:DND1004,CustomPresence:1004

exten => 1005,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1005/ringtimer)}" > "0"]?${DB(AMPUSER/1005/ringtimer)}:${RINGTIMER_DEFAULT})})
exten => 1005,n,Macro(exten-vm,novm,1005,1,0,0)
exten => 1005,n(dest),Set(__PICKUPMARK=)
exten => 1005,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)
exten => 1005,n,Goto(${IVR_CONTEXT},return,1)
exten => 1005,hint,PJSIP/1005&Custom:DND1005,CustomPresence:1005

exten => 1006,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1006/ringtimer)}" > "0"]?${DB(AMPUSER/1006/ringtimer)}:${RINGTIMER_DEFAULT})})
exten => 1006,n,Macro(exten-vm,novm,1006,1,0,0)
exten => 1006,n(dest),Set(__PICKUPMARK=)
```

exten => 1006,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)

exten => 1006,n,Goto(${IVR_CONTEXT},return,1)

exten => 1006,hint,PJSIP/1006&Custom:DND1006,CustomPresence:1006

exten => 1007,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1007/ringtimer)}" > "0"]?${DB(AMPUSER/1007/ringtimer)}:${RINGTIMER_DEFAULT})})

exten => 1007,n,Macro(exten-vm,novm,1007,1,0,0)

exten => 1007,n(dest),Set(__PICKUPMARK=)

exten => 1007,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)

exten => 1007,n,Goto(${IVR_CONTEXT},return,1)

exten => 1007,hint,PJSIP/1007&Custom:DND1007,CustomPresence:1007

exten => 1008,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1008/ringtimer)}" > "0"]?${DB(AMPUSER/1008/ringtimer)}:${RINGTIMER_DEFAULT})})

exten => 1008,n,Macro(exten-vm,novm,1008,1,0,0)

exten => 1008,n(dest),Set(__PICKUPMARK=)

exten => 1008,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)

exten => 1008,n,Goto(${IVR_CONTEXT},return,1)

exten => 1008,hint,PJSIP/1008&Custom:DND1008,CustomPresence:1008

exten => 1009,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1009/ringtimer)}" > "0"]?${DB(AMPUSER/1009/ringtimer)}:${RINGTIMER_DEFAULT})})

exten => 1009,n,Macro(exten-vm,novm,1009,1,0,0)

exten => 1009,n(dest),Set(__PICKUPMARK=)

```
exten => 1009,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)

exten => 1009,n,Goto(${IVR_CONTEXT},return,1)

exten => 1009,hint,PJSIP/1009&Custom:DND1009,CustomPresence:1009


exten => 1010,1,Set(__RINGTIMER=${IF($["${DB(AMPUSER/1010/ringtimer)}" > "0"]?${DB(AMPUSER/1010/ringtimer)}:${RINGTIMER_DEFAULT})})

exten => 1010,n,Macro(exten-vm,novm,1010,1,0,0)

exten => 1010,n(dest),Set(__PICKUPMARK=)

exten => 1010,n,GotoIf($["${DIALSTATUS}"="NOANSWER"]?app-blackhole,busy,1)

exten => 1010,n,Goto(${IVR_CONTEXT},return,1)

exten => 1010,hint,PJSIP/1010&Custom:DND1010,CustomPresence:1010
```

Appendix D: Ten User Account Authentication Database built in Asterisk VoIP Server

The following are ten sample mobile phones user account authentication databases added to the VoIP server.

[1001-auth]

type=auth

auth_type=userpass

password=User 1

username=1001

[1002-auth]

type=auth

auth_type=userpass

password=User 2

username=1002

[1003-auth]

type=auth

auth_type=userpass

password=User 3

username=1003

[1004-auth]

type=auth

auth_type=userpass

password=User 4

username=1004

[1005-auth]

type=auth

auth_type=userpass

password=User 5
username=1005

[1006-auth]
type=auth
auth_type=userpass
password=User 6
username=1006

[1007-auth]
type=auth
auth_type=userpass
password=User 7
username=1007

[1008-auth]
type=auth
auth_type=userpass
password=User 8
username=1008

[1009-auth]
type=auth
auth_type=userpass
password=User 9
username=1009

[1010-auth]
type=auth
auth_type=userpass
password=User 10
username=1010

Appendix E: VoIP Server Mobile Phones Endpoint SIP Setups in the VoIP Server

The following are five sample mobile phones user account endpoint settings added to the VoIP server together with the allowed G 711 ulaw and alaw codecs.

[1001]

type=endpoint

aors=1001

auth=1001-auth

allow=alaw,ulaw

context=from-internal

callerid=device <1001>

dtmf_mode=rfc4733

aggregate_mwi=yes

use_avpf=no

ice_support=no

media_use_received_transport=no

trust_id_inbound=yes

media_encryption=no

timers=yes

media_encryption_optimistic=no

send_pai=yes

rtp_symmetric=yes

rewrite_contact=yes

force_rport=yes

language=en

[1002]

type=endpoint

aors=1002

auth=1002-auth

allow=alaw,ulaw

context=from-internal

callerid=device <1002>

dtmf_mode=rfc4733

aggregate_mwi=yes

use_avpf=no

ice_support=no

media_use_received_transport=no

trust_id_inbound=yes

media_encryption=no

timers=yes

media_encryption_optimistic=no

send_pai=yes

rtp_symmetric=yes

rewrite_contact=yes

force_rport=yes

language=en

[1003]

type=endpoint

aors=1003

auth=1003-auth

allow=alaw,ulaw

context=from-internal

callerid=device <1003>

dtmf_mode=rfc4733

aggregate_mwi=yes

use_avpf=no

ice_support=no

media_use_received_transport=no

trust_id_inbound=yes

media_encryption=no

timers=yes

media_encryption_optimistic=no

send_pai=yes

rtp_symmetric=yes

rewrite_contact=yes

force_rport=yes

language=en

[1004]

type=endpoint

aors=1004

auth=1004-auth

allow=alaw,ulaw

context=from-internal

callerid=device <1004>

dtmf_mode=rfc4733

aggregate_mwi=yes

use_avpf=no

ice_support=no

media_use_received_transport=no

trust_id_inbound=yes

media_encryption=no

timers=yes

media_encryption_optimistic=no

send_pai=yes

rtp_symmetric=yes

rewrite_contact=yes

force_rport=yes

language=en

[1005]

type=endpoint

aors=1005

auth=1005-auth

allow=alaw,ulaw

context=from-internal

callerid=device <1005>

dtmf_mode=rfc4733

aggregate_mwi=yes

use_avpf=no

ice_support=no

media_use_received_transport=no

trust_id_inbound=yes

media_encryption=no

timers=yes

media_encryption_optimistic=no

send_pai=yes

rtp_symmetric=yes

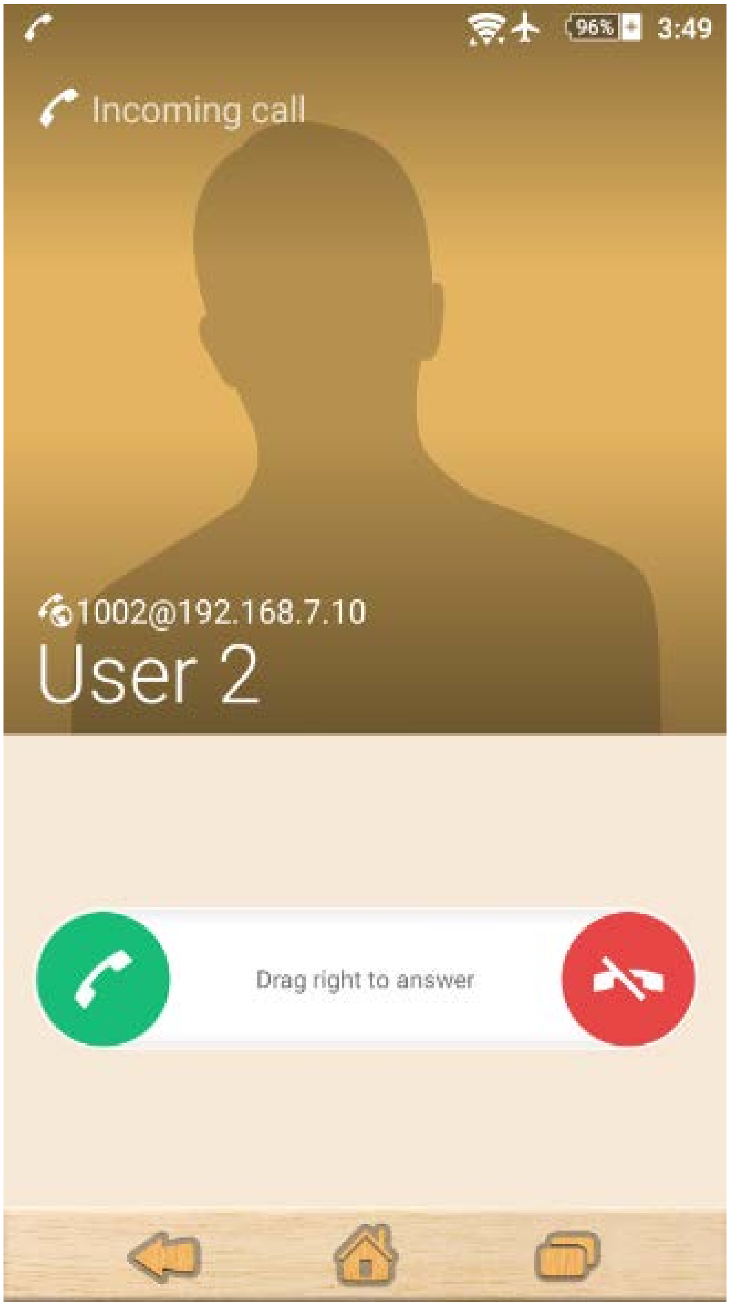
rewrite_contact=yes

force_rport=yes

language=en

Appendix F: Screenshots of a mobile phone user receiving and making a phone call using the prototype

A. A mobile phone user making a phone call with the prototype



B. A mobile phone user getting an incoming call with the prototype