PERFORMANCE ANALYSIS OF 8KBPS VOICE CODEC (G.729, G.711 ALAW, G.711 ULAW) FOR VOIP OVER WIRELESS LOCAL AREA NETWORK WITH RESPECTIVE SIGNAL-TO-NOISE RATIO

MOHAMMAD ZAKI BIN NORANI

THESIS SUBMITTED IN FULFILMENT OF THE DEGREE OF COMPUTER SCIENCE (COMPUTER SYSTEM AND NETWORKING)

FACULTY OF COMPUTER SYSTEM AND SOFTWARE ENGINEERING

2013

UNIVERSITI MALAYSIA PAHANG

BORANG PENGESAHAN STATUS TESIS

JUDUL: PERFORMANCE ANALYSIS OF 8KBPS VOICE CODEC (G.729, G.711 ALAW, G.711 ULAW) FOR VOIP OVER WIRELESS LOCAL AREA NETWORK WITH RESPECTIVE SIGNAL-TO-NOISE RATIO

SESI PENGAJIAN: SEMESTER 1 SESI 2013/2014

SAYA MOHAMMAD ZAKI BIN NORANI

Mengaku membenarkan tesis/laporan PSM ini disimpan di Perpustakaan Universiti Malaysia Pahang dengan syarat-syarat kegunaan seperti berikut:

1. Tesis/Laporan adalah hakmilik Universiti Malaysia Pahang.
2. Perpustakaan Universiti Malaysia Pahang dibenarkan membuat salinan untuk
3. tujuan pengajian sahaja.
4. Perpustakaan dibenarkan membuat salinan tesis ini sebagai bahan pertukaran antara
5. institut pengajian tinggi.
6. **Sila tandakan (√)

☐	SULIT	(Mengandungi maklumat yang berdarjah keselamatan atau kepentingan Malaysia seperti yang termaktub di dalam AKTA RAHSIA RASMI 1972) *
☐	TERHAD	(Mengandungi maklumat TERHAD yang telah ditentukan oleh organisasi/badan di mana penyelidikan dijalankan) *
☐	TIDAK TERHAD	

Disahkan Oleh

………………………………. ………………………………………

Alamat tetap: Penyelia:

Tarikh: ……………………… Tarikh: ……………………………..

*Sila lampirkan surat daripada pihak berkuasa/organisasi berkenaan dengan menyatakan sekali sebab dan tempoh tesis/laporan ini perlu dikelaskan sebagai SULIT atau TERHAD.

DECLARATION

I hereby declare that the work in this thesis is my own except for quotations and summaries which have been duly acknowledged

13/12/2013

Mohammad Zaki Bin Norani
CA 11022

SUPERVISOR DECLARATION

I hereby declare that I have read this thesis and in my opinion this thesis/report is sufficient in terms of scope and quality for the award of the degree of Bachelor of Computer Science (Computer System and Networking)

Signature : ……………………………………….

Supervisor Name:

Date : ……………………………………….

ACKNOWLEDGMENTS

I have put all my efforts to complete this project. However, it will not successfully complete without the help and encouragement of other individuals involved directly and indirectly.

First of all I would like to express gratitude to Allah for allowing this project to be completed properly. I also want to thank my supervisor Mr. Imran Edzereiq Kamarudin who the one that has been very helpful and always give me some advice in the process of making this project. Besides, I would like to express my gratitude towards my parents, my friends, and my academic advisor for their support and encouragement to continue my project to completion.

Contents

ABSTRACT

In this research, 3 types of speech codec (G.729, G.711 aLaw and G.711 uLaw) in the same sampling rate of 8kbps are put to test in predefined network environment and given respective SNR 10dB, 20dB and 30dB to measure the performance base on R-factor, MOS, packet jitter and packet lost. Speech codec is used to convert the analog voice signals into digital signal. Each speech codec have its own speech quality, minimum bandwidth require etc. There are many manufacturers that have been producing various types of speech codecs in the market. The VoIP users are able to choose the desire codec that will be used or enable in the VoIP call based on the service and hardware that can support the speech codec. But, users will face some difficulty in choosing the best codec to use. All 3 mentioned speech codec will be test base on these criteria; VoIP session over optimum wireless network with 10dB, 20dB and 30dB SNR and VoIP session over wireless network that shared with other traffic with 10dB, 20dB and 30dB SNR. Six testbed will be carry out to complete all the criteria and all of the tests criteria will be carry out on real devices simulation. At the end, the performance measurement such as MOS, r-factor , packet lost and packet jitter will be observe to determine the best speech codec in each scenario. The final results of this research should be able to determine the best speech codec among the four codecs that have been selected and match the suitability with the environments.

Chapter 1

INTRODUCTION

1.0 Introduction

Voice over Internet Protocol (VoIP) is a technology that allows user to make voice calls using a broadband Internet connection instead of a regular (or analog) phone line. Some VoIP services may only allow user to call other people using the same service, but others may allow user to call anyone who has a telephone number including local, long distance, mobile, and international numbers. Also, while some VoIP services only work over user computer or a special VoIP phone, other services allow user to use a traditional phone connected to a VoIP adapter.

VoIP services convert user voice into a digital signal that travels over the Internet. If users are calling a regular phone number, the signal is converted to a regular telephone signal before it reaches the destination. VoIP can allow user to make a call directly from a computer, a special VoIP phone, or a traditional phone connected to a special adapter. In addition, wireless "hot spots" in locations such as airports, parks, and cafes allow user to connect to the Internet and may enable user to use VoIP service wirelessly. Usually the device and equipment that we need to use this service is a broadband (high speed Internet) connection is required. This can be over a cable modem, or high speed services such as DSL or a local area network. A computer, adaptor, or specialized phone is required. [1]

This experiment will analyze one of the parts of the VoIP which is speech codec. Speech codec are used to convert an analog voice signal to digitally encoded version. Codecs vary in the sound quality, the bandwidth required, the computational requirements, etc.

In VoIP, voice which is an analog signal is converted into digital signal, it is then encoded by using suitable VoIP codecs and your voice is compressed in the form of stream of binary data and transmitted over the internet. When encoded data arrives at the other end it is then decoded into digital signal and then analog signal. This whole process takes place within less than few milliseconds. [2]

1.2 Problem statement

With many codecs has been produced by various manufacturers, the selection of the codec to use to be a little difficult when user need to consider the appropriateness of a codec suitable with bandwidth of the network. Some VoIP client provides the codec that the user can select is manually. In this type of application, the codec selection is really important because it will give big effect to the VoIP session whether the session will provide a good or bad quality of voice.

Furthermore, certain speech codec have a same minimum bandwidth requirement. For example G.729, G.711 aLaw, and G.711 uLaw have 8kb minimum bandwith requirement. So, among this four codec, user cannot determine which one of the codec is the best for VoIP session.

Many studies have been made to analyze the performance of speech codec based on sampling method and research base on ideal network. However, not many researches have done on specific group of speech codec based on minimum bandwidth requirement and specific type of network.

User that use the VoIP service usually using the VoIP service while using the other internet application such as web browsing, file transfer etc. So, it will affect the performance of the connection of the VoIP session because it's using the same network bandwidth.

1.3 Objective

This study was conducted to meet three objectives:

- To simulate (G.729, G.711 aLaw, and G.711 uLaw) speech codec of VoIP on predefine wireless mesh network.
- To simulate (G.729, G.711 aLaw, G.711 uLaw) speech codec on respective (10dB, 20dB, 30dB) signal-to noise ratio (SNR).
- To analysis (G.729, G.711 aLaw, G.711 uLaw) speech codec performance in term of MOS and R-factor.
- To suggest the best speech codec based on the MOS score and R-factor for the predefined wireless mesh network and respective SNR.

1.4 Scope

Due to the time and resources constraints, this research is limited to the following matter:

- The simulation is using a group of 8kbps speech codec. Four type of speech codec (G. G.729, G.711 aLaw, G.711 uLaw) to be analyzed.
- Using only SIP architecture environment.
- IEEE 802.11n wireless network connection will be used as a medium during the simulation. Two wireless access points (AP) will be used to establish the connection.
- Equipment: computer, WAP, SIP server, VoIP client and real test bit simulation
- The end of the simulation, the performance measurement base on MOS and R-factor can be generated to see the result of the experiment.

Thesis Organization

The research consists of five chapters:

Chapters 1 provide the overall overview of the thesis. Here, the problem statement will be introduced. Then based on the problem statement, the objective of the research is being defined. Lastly, chapter one also will explain about the research scope.

Chapter 2 introduces the hardware and software that will be used in this research project. It is mainly focuses on the performance of the bandwidth estimation tools. The literature review is organized in a way that readers can understand this.

Chapter 3 explains the methodology that will be used to carry out this research. The detail will be elaborated step by step process that is being used to complete the research.

Chapters 4 design the model or know as architecture that will be developed in order to perform the test. It then followed with the continuously design on data analysis.

Chapter 5 concludes all the chapters and the recommendations for future researchers explain most of the configurations of hardware and software involved in the research. Detail test result will be included in this chapter.

Chapter 2

2.0 LITERATURE REVIEW

2.1 Session initiate protocol (SIP)

SIP is a signaling protocol like to HTTP. It is a protocol that can setup and tear down any type of session. SIP call control uses Session Description Protocol (SDP) to describe the details of the call (i.e., audio, video, a shared application, codec type, size of packets, etc.). SIP uses a URI7 to identify a logical destination, not an IP address. The address could be a nickname, an e-mail address, or a telephone number. In addition to setting up a phone call, SIP can notify users of events, such as “I am online,” “a person entered the room,” or “e-mail has arrived.” SIP can also be used to send instant text messages.

PSTN-Like services	Create new services
Caller ID	Web/voice integration
PBX-like features	Programmable services
Call forwarding	Multi-destination routing
Call transfer	Presence
AIN-like features	Instant messaging
Free phone	Multimedia
Find me/follow me	Event notification
Conference calls	Caller and called party preferences

Table 1: defines some of the types of services that can be offered using SIP.

Using a client–server model, SIP defines logical entities that may be implemented separately or together in the same product. Clients send SIP requests, whereas servers accept SIP requests, execute the requested methods, and respond.

The SIP requirement defines six request methods [11]:

- **REGISTER** permits either the user or a third party to register communicates information with a SIP server.
- **INVITE** initiates the call signaling sequence.
- **ACK** and **CANCEL** sustenance session setup.
- **BYE** terminates a session.
- An **OPTION** queries a server about its abilities.

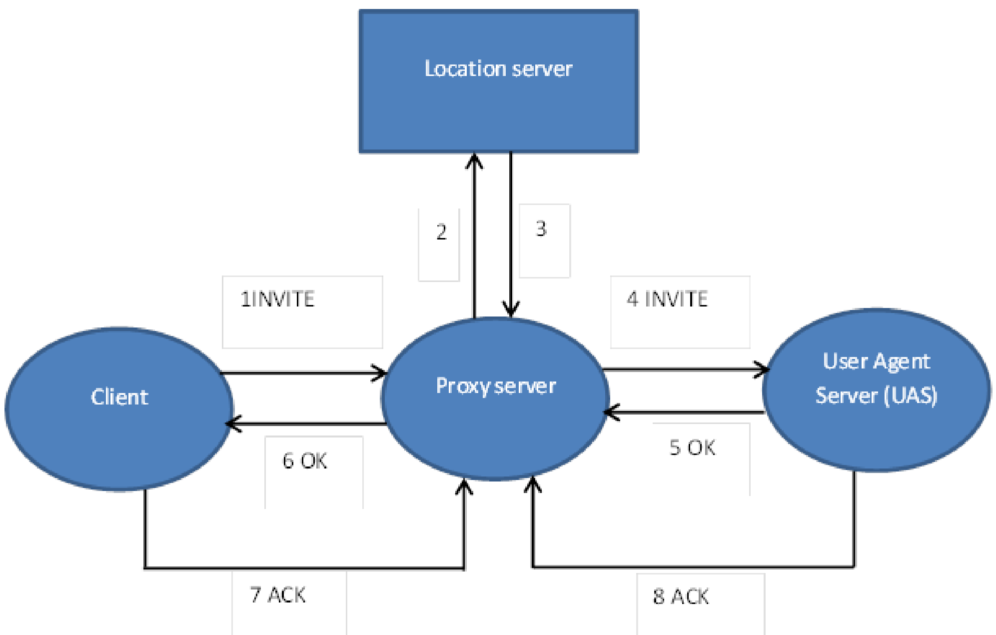


Figure 1: Basic example of a SIP operation

2.3 Speech codec

2.3.1 Overview

When we talk with people which nearby or there is at place which same with where we are, voice sent directly to ear listener through wave. To talk with people who are in other places, especially the different geographical areas such as using the telephone or other equipment, the sound transmitted through various mediums such as cable, microwave, and signal and so on. Therefore, we need communication equipment require human voice codecs to convert the analog form to digital form of the signal to be transmitted through the medium provided. Codecs are used to convert an analog voice signal to digitally encoded version. Codecs vary in the sound quality, the bandwidth required, the computational requirements, etc.

Many factors determine voice quality, including the choice of codec, echo control, packet loss, delay, delay variation (jitter), and the design of the network. Packet loss causes voice clipping and skips. Some codec algorithms can correct for some lost voice packets. Typically, only a single packet can be lost during a short period for the codec correction algorithms to be effective. If the end-to-end delay becomes too long, the conversation begins to sound like two parties talking on a Citizens Band radio. A buffer in the receiving device always compensates for jitter (delay variation). If the delay variation exceeds the size of the jitter buffer, there will be buffer overruns at the receiving end, with the same effect as packet loss anywhere else in the transmission path.

There are many codecs available for digitizing speech. The quality of a voice call through a codec is often measured by subjective testing under controlled conditions using a large number of listeners to determine an MOS. Several characteristics can be measured by varying the test conditions. Important characteristics include the effect of environmental noise, the effect of channel degradation (such as packet loss), and the effect of tandem encoding/decoding when interworking with other wireless and terrestrial transport networks. The latter characteristic is especially important since VoIP networks will have to interwork with switched circuit networks and wireless networks using different codecs for many years [11].

2.3.2 ITU-T G.729

G.729 is ITU-T codec standard that have two version which is A and B. G.729 also offered 8kbps low bit rate with reasonably toll-quality voice. Input frames are 10 milliseconds (10 ms) in duration and generated frames contain 80 bits. The input and output contain 16-bit pulse-code modulation (PCM) samples converted from or to 8-Kbps compressed data. Toll-quality is the service that this codec will provide is can give same quality with public switch network where the call is charged every minute of use [4]. Therefore, it ideally suited for the use of VoIP because VoIP broadband rates should be given serious consideration. G.729 is the proprietary codec; anyone that wants to use this codec on their client should get the license from the company that re-sells the G.729 license. However, there has some non-commercial experimental for this G.729 can be used.

G.729 fits into the general category of CELP (Code Excited Linear Prediction) speech coders [6]. These coders are all based on a model of the human vocal system. In that model, the throat and mouth are modeled as a linear filter, and voice is generated by a periodic vibration of air exciting this filter. In the frequency domain, this implies that speech looks somewhat like a smooth response (called the envelope), modulated by a set of discrete frequency components. CELP coders all vary in the manner in which the excitation is specified, and the way in which the coefficients of the filter are represented. All of them generally break speech up into units called frames, which can be anywhere from 1ms to 100ms in duration. For each frame of speech, a set of parameters for the model are generated and sent to the decoder. This implies that the frame time represents a lower bound on the system delay; the encoder must wait for at least a frames worth of speech before it can even begin the encode process. In G.729, each frame is 10ms, or 80 samples, in duration. This frame is further broken into two 5ms sub frames. The filter parameters are specified just once for each frame, but each sub frame has its own excitation specified. It is also important to note that speech can generally be classified into two types: voiced and unvoiced. Voiced sounds, such as b,d, and g, are generated from the throat, whereas unvoiced sounds, such as th, f, and sh, are generated from the mouth. The model works better for voiced sounds, but the excitation can be tailored for voiced or unvoiced so that it works in both cases.

2.3.3 G.711

An International Telecommunications Union (ITU-T) standard for audio (speech) compression and decompression that is used in digital transmission systems, and in particular, used for the coding of analog signals into digital signals.

G.711 is also known as Pulse Code Modulation (PCM). It is the ITU-T international standard for encoding telephone audio on a 64 kbps channel. PCM samples the signal 8000 times a second; each sample is represented by 8 bits for a total of 64 kbit/s. There are two versions of this standard codec. The u-law (pronounced as mew law) is generally used in North America and Japan digital communications. The A-law is used in European digital communications. The difference between the two standards is the method in which the analog signal is sampled. (See also PCM).

2.3.3.1 G.711 a-Law

A-law is used in Europe and the rest of the world. This type of G.711 has a smaller dynamic range than U-law. Dynamic range is basically the ratio between the quietest and loudest sound that can be represented in the signal. The downside of having a higher dynamic range is greater distortion of small signals. This simply means that a-law would sound better than u-law when the sound input is very soft.

In addition, this codec encoding thus takes a 13-bit signed linear audio sample as input and converts it to an 8 bit value as show in the table below:

Linear input code	Compressed code
s0000000wxyz`a	s000wxyz
s0000001wxyz`a	s001wxyz
s000001wxyz`ab	s010wxyz
s00001wxyz`abc	s011wxyz
s0001wxyz`abcd	s100wxyz
s001wxyz`abcde	s101wxyz
s01wxyz`abcdef	s110wxyz
s1wxyz`abcdefg	s111wxyz

Table 2 audio sample compressed form for a-Law

2.3.3.1 G.711 U-Law

U-Law is a companding algorithm, primarily used in the digital telecommunication systems of North America and Japan. Companding algorithms reduce the dynamic range of an audio signal. In analog systems, this can increase the signal-to-noise ratio (SNR) achieved during transmission, and in the digital domain, it can reduce the quantization error (hence increasing signal to quantization noise ratio). These SNR increases can be traded instead for reduced bandwidth for equivalent SNR. Different from a-Law, u-Law has higher dynamic range than a-Law. U-law encoding takes a 14-bit signed linear audio sample as input. Different from a-Law, u-Law increases the magnitude by 32 (binary 100000), and converts it to an 8 bit value as show in the table below:

Linear input code	Compressed code
s00000001wxyz`a	s000wxyz
s0000001wxyz`ab	s001wxyz
s000001wxyz`abc	s010wxyz
s00001wxyz`abcd	s011wxyz
s0001wxyz`abcde	s100wxyz
s001wxyz`abcdef	s101wxyz
s01wxyz`abcdefg	s110wxyz
s1wxyz`abcdefgh	s111wxyz

Table 3 audio sample compressed form for u-Law

2.2 Speech codec performance measurement

2.2.1 Mean opinion score (MOS) and R-Factor

MOS is the standard to rate the test of audio quality recommended by ITU-T. MOS is determined in one number, from 1 to 5, 1 being the worst and 5 the best. Originally, the test is involve the human that being held with some people sat in the quiet room and listening the audio and the rate it with the MOS. This is why the word "opinion" is used. Nowadays, the test no longer uses human, software to measure and quantify audio MOS was developed. This software is able to calculate the MOS for a test made on audio [1].

R-factor is the alternative way to calculate rate the quality of speech and audio. The function is same as MOS but the rate is different which is from 1 to 120. Because of the limit of rate is more than MOS, R-factor is become the most precise method to do the test of the quality of speech and audio. R-factor evaluates the user perception and all factors that will be effect the quality of the VoIP system. The rate is divided by two results which is network and user R-factor. Most of the user believes that the R-factor is more objective method than MOS [1].

Although r-factor can calculate better evaluation result of the testing, both of the method will be use and generated to get better judgment of the call quality.

Rating	ACR description – MOS	DCR description – DMOS
5	Excellent	Degradation not perceived
4	Good	Degradation perceived but not annoying
3	Fair	Degradation slightly annoying
2	Poor	Degradation annoying
1	Bad	Degradation very annoying

Table 4: MOS and R-factor value

2.3 Typical VoIP Problems

2.3.1 Packet jitter

Jitter is one of the QOS issue in the use of VoIP if the problem can no longer be controlled. Different with network delay, jitter does not happen because of the packet delay, but it is happen when the variation of packet delays occur. Jitter occurs when a packet should be delivered in a steady stream, but due to network problems packet sent not arrive right on time [13].

In VoIP conversation, VoIP endpoints try to control the jitter by increase the size of the packet buffer; jitter will causes delays in the VoIP conversation. The minimum variation of the packet is 150ms and if the variation becomes too high and exceeds the minimum variation, callers will notice the delay and will talking like walkie-talkie conversation. There are several steps you can take to deal with jitter on the network layer and application layer such as VoIP software, IP phones or specific VoIP adaptors. By definition, steps to reduce delays in the network to maintain the buffer is less than 150ms but the variation could not necessary removed. Although variation is not removed by the reduction in network delay, but it is still effective in reducing variation and this is not known by the caller. In addition, the setting for VoIP services as a priority and bandwidth shaping on the network can also reduce the variation in packet delay [1].

At the endpoint, it is crucial to optimize the jitter buffering. While better buffers reduce and eliminate the jitter, anything over 150ms remarkably affects the real quality of the conversation. Adaptive algorithms to manage buffer size depending on the present network circumstances are often quite working. Fiddling with the packet size or using a dissimilar codec (e.g. G.711) will often help control the jitter. While jitter is more affected by network delays than by the endpoints, some resource-struggling systems that are executed in concurrent environments, such as VoIP soft-phones, may present significant and random variations in packet delays. While developing VoIP endpoints or simulate call quality problems within the existing VoIP infrastructure, it is very important to consider about the cause of jitter. A network analyzing and monitoring tool with VoIP analysis can be use in localizing the source of the problem efficiently by produce the value of the jitter and packet loss of the VoIP session.

2.3.2 Packet loss

Packet loss can occur in all types of networks. Therefore, each network protocol designed to handle packet loss occurrences in their own ways. For example, the Transmission control protocol (TCP), which address the problem of packet loss in transmission of a packet with the request for packets that have been lost during the transmission. But in VoIP, VoIP call has no time to wait for the packet to arrive.

VoIP is very concerned about the problem of packet loss, even if only 1% of the packets have been dropped; it will affect the quality of VoIP calls [4]. Speech codecs will play an important role in dealing with the problem of packet loss. Most speech codecs can only assume less than 1% packet loss on a VoIP call this problem should be avoided during VoIP calls from the occurrence of audible errors. Ideally, there must be no packet loss in VoIP call.

There are several techniques to prevent or reduce the packet loss problem. One of the techniques that usually use is called Packet Loss Concealment (PLC) has been used in VoIP session to mask the effect of packet loss. There are some more techniques that may be used in different implementations.

In VoIP, some packets will be discarded for several reasons, including network congestion, line errors, and late arrival. Look at the exact value of packet-loss graphs permit the network administrators to select a PLC technique that best counterparts the characteristics of a certain environment, this method will help them to manage the problem of packet loss effectively.

2.3.3 Signal-to-noise ratio (SNR)

Signal-to-noise ratio (SNR) is one of the factor that VoIP developer should consider when develop VoIP over wireless link. The signal to noise ratio is also referred to as SNR. In other word, it is the ratio between the maximum signal strength that a wireless connection can reach and the noise present in the connection. Here “noise” refers to the stray frequencies that obstruct with the transmission of data in a wireless network.

The signal-to-noise ratio, the bandwidth, and the channel capacity of a communication channel are related by the Shannon–Hartley theorem. Signal-to-noise ratio is sometimes used indirectly to refer to the ratio of useful info to false or unrelated data in a conversation or exchange. The Shannon–Hartley theorem states the maximum rate at which information can be transmitted over a communications channel of a specified bandwidth in the occurrence of noise. Therefore, the SNR of a network have to be as high as possible. The greater the value of SNR, the better the signal strength and the quality of transmission will get.

In VoIP over wireless LAN environment the quality of the call session is depending on how good the voice transmission will be conducted. Based on the Shannon-Hartley theorem, when the SNR interrupt or present on the wireless link that will be used for VoIP session, the call quality and voice data transmission will affected. The value of the SNR can also influence the quality of call and affect the MOS and R-Factor reading in every call because.

2.4 Related research

2.4.1 Capacity of an IEEE 802.11b Wireless LAN supporting VoIP

This research evaluate the capacity of an IEEE 802.11b network carrying voice calls in a wide range of scenarios, including varying delay constraints, channel conditions and voice call quality requirements. It uses G.711 and G.729 voice encoding schemes and a range of voice packet sizes. Firstly, researcher present an analytical upper bound and, using simulation, show it to be tight in scenarios where channel quality is good and delay constraints are weak or absent. Then use the simulation to show that capacity is highly sensitive to the delay budget allocated to packetization and wireless network delays.

This research also shows how channel conditions and voice quality requirements affect the capacity. Selecting the optimum amount of voice data per packet is shown to be a trade-off between throughput and delay constraints: by selecting the packet size appropriately given the delay budget and channel conditions, the capacity can be maximized. Unless a very high voice quality requirement precludes its use, G.729 is shown to allow a capacity greater than or equal to that when G.711 is used, for a given quality requirement [2]. The paper is only evaluated an upper bound on the capacity of IEEE 802.11b network carrying voice calls, and found it to be tight in scenarios where channel quality is good and delay constraints are weak or absent [2]. Furthermore, this paper only uses two types of codec which is G.711 and G.729 that have different bit rate. The test is manipulating the network environment such as delay, packet size and channel condition to test the performance and quality of speech in the conversation [2].

2.4.2 Implementing VoIP: A Voice Transmission Performance Progress Report

This paper is aiming to introduce voice over IP networks and services in ways that satisfy the voice quality expectations of one of the VoIP service provider company customers, they have been conducting laboratory studies of how VoIP transmission affects voice quality while also carefully monitoring and managing several field implementations of VoIP. This article summarizes much of what they have learned in this work, and they hope it provides a useful progress report on the industry's evolution to VoIP.

They review their data on the voice quality effects of packet loss, delay, speech coders, packet loss concealment algorithms, and the compression option of suppressing transmission during silence. Because the familiar problem of echo has emerged repeatedly in the VoIP environment, they review this issue in some detail. Packet loss and delay variation measurements made on private VoIP networks are reviewed, and the data here are encouraging [3]. The test recommended for VoIP session, the codec that practical to use to maintain the performance is G.711, G.726, G.728, or G.729E [3]. This research also use the single VoIP session in the network environment without any other traffic from any network application that use the same network environment. The paper focus on the type of service that has been provides by the service provider to their customer. It suggest a few method to get better service to be provide to customer such as compression can come with significant quality penalties, especially where multiple coding are likely end-to-end and/or where high noise levels (or music on hold) might he a common operating condition.

2.4.3 VoIP Basics: Codec Latency vs. Bandwidth Optimization

This paper is about the codec latency and bandwidth optimization. The researcher has indicated that the codec has low bandwidth is very efficient. This is proved by G.729 will compress 10 milliseconds of audio to 10 bytes and G.723.1 encode 30ms frames to 24 or 20 bytes [9]. However, since we send compressed audio frames as payload in RTP packets which are in turn sent over UDP, The researcher need to consider the overhead for IP, UDP, and RTP headers. The overhead is 40 bytes per packet. This is significant when compared with the size of a compressed audio frame if we are not on a local area network and the bandwidth is limited. The table below shows the overhead for several low-bandwidth codecs. The researcher did the calculation for one frame per packet for G.723.1 and GSM and for 3 frames per packet for G.729 since this codec works with frame size of only 10 milliseconds.

Codec	Nominal bitrate [kbit/s]	Frame length [ms]	Frame size [bytes]	Packet overhead	Actual bitrate [kbit/s]
G.723.1	6.4	30	24	167%	17
G.723.1	5.3	30	20	200%	16
G.729	8	10 *3	10 *3	133%	18.6
GSM 06.10	13	20	33	121%	29.2

Table 5: calculation for one frame per packet for G.723.1 and GSM and for 3 frames per packet for G.729

When calculating the latency, you need to consider the time it takes to send a packet from one end to another (your mileage may vary, try to use "traceroute" to get a clue) and the size of the jitter buffer of the receiving end (which can be 50-60 milliseconds worth of audio). Considering all this, I would say the reasonable maximum is to send 60 milliseconds of audio in one packet. This will result in the following bitrates:

Codec	Nominal bitrate [kbit/s]	Frames in 60 ms of audio	Actual bitrate [kbit/s]
G.723.1	6.4	2	11.7
G.723.1	5.3	2	10.6
G.729	8	6	13.3
GSM 06.10	13	3	18.5

Table 6: result of 60 milliseconds of audio in one packet

In addition to latency, there are two more things we should consider when increasing the number of audio frames per RTP packet:

- If a packet with a larger number of frames gets lost, the loss is more noticeable to the user.
- With greater end-to-end delay, possible echo become more noticeable.

2.4.4 Performance Analysis of Different Codecs in VoIP Using SIP

Converged IP networks look for to incorporate voice, data, and video on the same infrastructure. Nevertheless, the integration of all kinds of traffic onto a single IP network has some benefits as well as weaknesses. While decreasing cost and growing mobility and functionality, VoIP may lead to consistency concerns, degraded voice quality, incompatibility, and end-user complaints due to moving network characteristics. The main purpose of VoIP, various codecs used in VoIP and packet loss, Jitter, delay are analyzed and discussed.

The comparison between three different codecs which is G.711, G.723 and G.729 has been analyzed by implementing peer-to-peer VoIP network using SIP server and caller. Thus we have described the various codecs in VoIP implementation and analyzed three commonly used codecs using peer-to-peer network scenario. These are common narrow band codecs. It can be analyzed from the results that G.711 is an ideal solution for PSTN networks with PCM scheme. G.723 is used for voice and video conferencing however provides lower voice quality. Music or tones such as DTMF cannot be transmitted reliably with G.723 codec. G.729 is mostly used in VoIP applications for its low bandwidth requirement.

Codec	*Data Rate*	*MOS Score*
G 711	64	4.3
G 726	32	4.0
G 726	63	3.8
G 728	16	3.9
G 729	8	4.0
GSM	13	3.7

Table 7: MOS for this analysis

2.4.5 An E-Model Implementation for Speech Quality Evaluation in VoIP Systems

The most common method is very effective for measuring the quality of voice and sound is to use MOS. however, the combination of MOS and R-factor enhance voice quality test results will be made. This article presents a voice quality measurement tool based on the ITU-T E Model. Firstly, the

ITU-T and ETSI specifications of E-Model are briefly reviewed and some errors found in these documents are pointed [12]. After, a measurement tool based on the corrections is described. E-Model can each impairment factor which affects a voice call can be computed separately, even so this does not imply that such factors are uncorrelated, but only that their contributions to the estimated impairments are separable. An expressive amount of delay and lost packets have to be present in the call, at alternating burst and gap conditions, otherwise we always will have excellent MOS scores [12].

In order to validate the measurement tool operation, we have to generate VoIP calls under known QoS environments and evaluate its voice quality using the tool. An expressive amount of delay and lost packets have to be present in the call, at alternating burst and gap conditions, otherwise we always will have excellent MOS scores and the measurement tool would not be completely tested. Thus we used the scenario shown on Figure 2 to generate some calls that could be evaluated by the measurement tool [12].

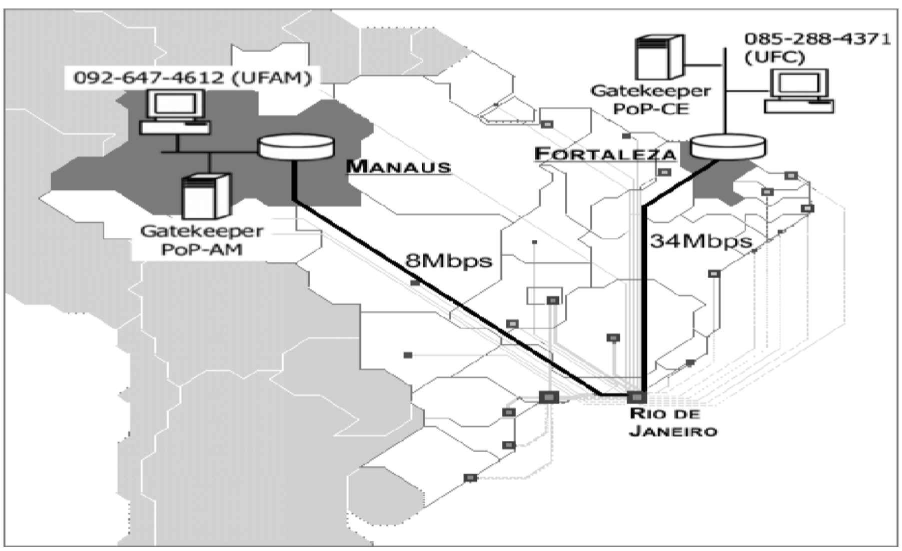


Figure 2 the diagram of the scenario that the researcher created

2.4.6 Best VoIP codecs selection for VoIP conversation over wireless carriers' network

This research is about to determine the best performance of VoIP using different codec. The impact of each element of a VoIP call will be analyzed. The rates will be generated from these VoIP elements will help in determining the best codec between five codec selected (G.711, G.722, G.726, GSM and SPEEX). Below is a two environment that has been use for this research:

1. VoIP over wireless LAN
    - All five codec has been analyze in this environment to be able to determine the best codec if the VoIP session is using the wireless LAN network.

2. VoIP over wireless WAN
    - The codec that has been selected is tested using the wireless WAN as a communication medium. The two different of the WAN ISP has been selected and used for this test.

In the analysis phase, the researcher measures and compares the VoIP performance using different codecs selection. The generating of the packet loss, packet jitter and MOS will help the researcher to determine the best codec that suitable to use in each environment. Five reading has been made to get the most accurate value. The specific tool is used to get the value of the packet loss, packet jitter and MOS.

2.4.7 Comparison between related researches

Research title	Codec used	Method	Result/Analysis
Capacity of an IEEE 802.11b Wireless LAN supporting VoIP	- G.711 - G.729	- Research evaluate the capacity of an IEEE 802.11b network carrying voice calls in a wide range of scenarios, including varying delay constraints, channel conditions and voice call quality requirements.	- Evaluated an upper bound on the capacity of an IEEE 802.11b network carrying voice calls. - the use of G.729 has been shown to allow - Greater capacity than the use of

			G.711 based on the MOS report.
Implementing VoIP: A Voice Transmission Performance Progress Report	- G.711 - G.726 - G.728 - G.729E	- Analyze on how VoIP transmission affects voice quality while also carefully monitoring and managing several field implementations of VoIP. Test the performance and functionality of E-Model tool.	- Get the jitter, packet loss, delay and other VoIP connection problem to compare with the E-Model tool in making measurement of VoIP network performance.
Performance Analysis of Different Codecs in VoIP Using SIP	- G.711 - G.723 - G.729	- Analyze three by implement peer to peer VoIP network using SIP. Simulation using two way communications between two end point and using one SIP server.	- Generate the delay, jitter, and voice traffic value to determine performance of the speech codec that has used. From the analysis, the researcher can determine the which codec suitable for video conference, music or etc.
Best VoIP codecs selection for VoIP conversation over wireless carrier network	- G.711 - G.726 - G.722 - GSM - SPEEX	- Test the VoIP performance over WAN and Using two different ISP wireless service provider. - Using different codec to test the	- Use MOS to measure and validate voice quality between two ISP wireless network providers.

		performance of each codecs.	- Determine the best codec performance based on the MOS value.
An E-Model Implementation for Speech Quality Evaluation in VoIP Systems	- G.711	- Use H.323 environment and test the E-Model result which contain combination of MOS and R-Factor. Using OpenH323 and callgen323 to help measurement tool generate the call quality measurement result.	- A measurement tool referred on the corrections is pronounced. VoIP calls over the backbone were used to prove the tool process.
Performance analysis of voice codec (G.729, SPEEX, iLBC and GSM) for VoIP over wireless LAN.	- G.729 - SPEEX - iLBC - GSM	- Create 4 different testbed to test the performance of 4 selected codec. Selected codec is from the same bit rate (8kbps). Using SIP server and wireless LAN network.	- Generate MOS and r-factor score to determine the best performance between 4 codecs. The score is generate using network monitoring tools base based on the jitter, packet loss and other network interference of the VoIP call.

Table 8: Comparison between related researches

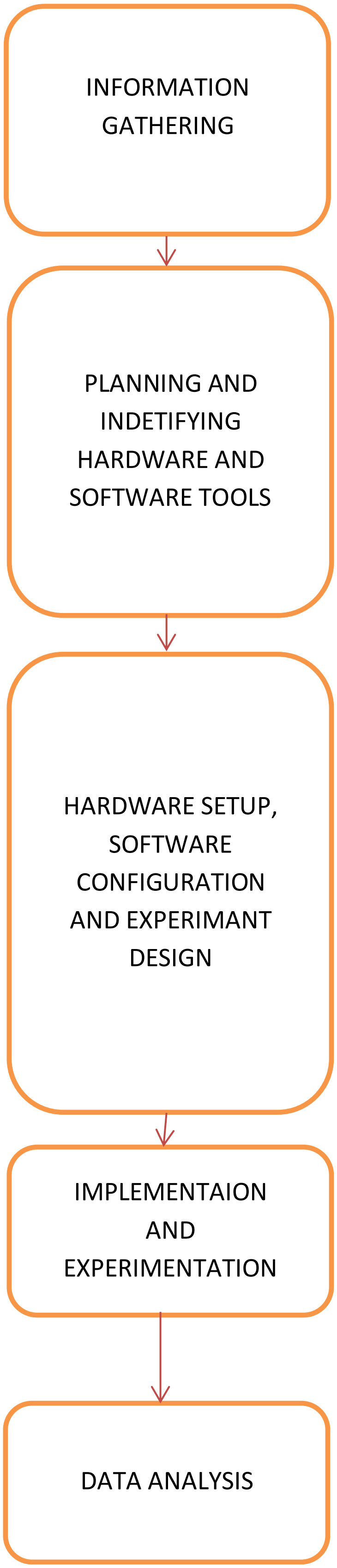
Chapter 3

RESEARCH METHODOLOGY

3.0 Introduction

Research methodology is a method to systematically explain the research problem. It may be understood as a science of reviewing how research is finished systematically. In it we study the numerous steps that are generally accepted by a researcher in reviewing his research problem along with the logic behind them. It is required for the researcher to know not only the research methods but also the methodology. Research methods can be put into the following three groups:

i. In the first group we include those methods which are concerned with the collection of data. These methods will be used where the data already available are not sufficient to arrive at the required solution;
ii. The second group consists of those statistical techniques which are used for establishing relationships between the data and the unknowns;
iii. The third group consists of those methods which are used to evaluate the accuracy of the results obtained.



PRELIMINARY STUDY PHASE	
PRELIMINARY STUDY	Offline: Books, journals, thesis project and research papers
	Online: Articles, journals and proceeding materials

RESEARCH PLANNING PHASE	
HARDWARE	Wireless access point
	Laptop – Caller (Sender)
	Laptop – Caller (Receiver)
SOFTWARE TOOLS	Voice quality and bandwidth measuring tool - CommView
	SIP server – Brekeke SIP server.
	VoIP client – Ekiga and Eyebeam

ARCHITECTURE DESIGN PHASE	
INSTALLATION	Hardware and software tools.
SETTING AND CONFIGURATION	Access point, Laptop for caller (sender) and caller (receiver).
ENVIRONMENT	
ARCHITECTURE	• One wireless access point. • One way communication between two callers. • Two way communication between two callers. • Add traffic on the same link with the call to test the bandwidth usage.

TESTING PHASE
• Generate one and two ways communication to create VoIP session. • Manipulate the link by add another traffic to make it like real network link. • Generate report of the rate of call quality analysis by using MOS and R-factor using CommView.

DATA ANALYSIS PHASE
Things to consider are Latency, Jitter and Packet loss. Voice/call quality measurement analysis (MOS and R-Factor) will be generating based on packet loss and jitter.

Table 9: Methodology of the research

3.1 Overview of Research Method

In order to complete this research, five important phases will be used to ensure that the study will be successful. Phases are defined as in Figure Methodology that was included. Phases involved are:

I. Preliminary Study Phase
II. Research Planning Phase
III. Architecture Design Phase
IV. Testing Phase
V. Data Analysis

3.1.1 Preliminary Study Phase

In preliminary study phase, preliminary studies have been done to gather information related to the study will be done. Study includes finding journals, books, articles on the site and many more materials with information about the study. Preliminary studies made to ensure that the objectives of this research to be achieved successfully. By doing preliminary research, methods and tools used can serve as a reference that can be used in this research.

Preliminary studies show that majority of previous research was done in terms of 2 way communications. All clients were expected to be simulated together.

A few codecs were used in the research that have been made but not in the same sampling rate as it has been defined in this research. Furthermore, environment used for the simulations using the SIP server and also some existing VoIP customers, including the free and the proprietary VoIP client. Methods for produce simulation results are also found in preliminary studies that have been made, the use of MOS and R-factor method has been classified by the tester in-testers and voice sound quality because this method has been specified by ITU-T recommendation.

3.1.2 Research Planning Phase

The next phase is one of the very important phases in this research; research planning phase. In this phase, the stages and activities that need to be carried out will be defined and planned carefully. This phase will ensure that the defined steps and activities will guide the flow of this research project in order to obtain the required data for analysis at a later stage in this research project. Planning for this research must be made carefully so that the time is sufficient for research to be completed successfully. To achieve the desired objectives, careful planning is also important.

The type of tools or software and hardware that need to be used for the simulation and analysis process will be determined in this process. Hardware that will be used for this research is two computers to be used as the hardware to make a VoIP call session. Both computer will be installed VoIP client as the software to use to make a call and one of the computer will be installed the SIP client for control the session of the VoIP and performance analysis tool to generate sniff analysis of the performance and quality of the VoIP session. Other than that, the other hardware that will be used for this research is wireless access point. This hardware is used to create a network for both computers connected to each other in a network. A dedicated wireless USB card is used specifically to sniff and analyze of the VoIP packet for performance measurement. The use of this hardware is to sniff the link between two computers in a VoIP session.

The tools will be installed on the computer used for this analysis such as a VoIP client, SIP server, and network monitoring tool functions will be tested and examined whether the selected tools suitable for use during the simulation. VoIP client is used as a tool for researcher use to establish a VoIP call session. Selected VoIP client should have a voice codec selection to function during the simulation process later, researcher were able to use and select the voice codec that want to test. SIP server is installed on one computer to make calls will work as a controller for VoIP call session to be established. This tool will control the establishment of a call session, the termination of the call session, and a host of other basic elements of the VoIP call session. To generate simulation results and reports for VoIP call quality is, tools such as network monitoring should be used to obtain rate the quality of a VoIP session. The selected tools should be able to generate MOS and R-factor for the call session together with other rates such as packet loss and jitter. Ten simulation results for the rate to be generated by these tools will be calculated as the average of the simulation results for a codec used. Below table shows the initial software and hardware requirement for this research.

Tool name	Type	Description
CommView	Performance analysis tool	Evaluation version. Tool for network monitoring and analyzer. CommView includes a VoIP analyzer for in-depth analysis, recording, and playback of SIP and H.323 voice communications. R-Factor, MOS Score – stream quality estimation based on packet loss and jitter.
Brekeke SIP Server	SIP server	Reliable and scalable SIP system platform for telephony carriers
Ekiga	VoIP client	VoIP or internet telephony client
Eyebeam	VoIP client	• Session Initiated Protocol (SIP) based signaling. • Performance Management of the SIP end-point • High Compression codec support • Multi-party and ad hoc Voice and Video Conferencing

Table 10: Tools and function

3.1.3 Architecture Design Phase

In this phase, the work of installing and configuring the tool and hardware will be conducted to simulate the process. This phase is important to do well and perfect for the accuracy of the simulation results depend on the installation and configuration have been made in hardware and tool. Session Initiation Protocol (SIP) server is used as an operator and control the call session for the call session will be established. Two clients will be set up to make a call session using VoIP client ekiga and eyebeam.

Three SNR value will be set up on the connection of the client. This is to create the environment of the network with some interference that will disturb the transmission of voice data.

Other situations will be included in this simulation is the use of network that have optimum bandwidth and network with other traffic in a same network with a VoIP session. The optimum network link can test and identify the actual performance value apply to a VoIP session that using one of the codecs since the network was dedicated only to the VoIP session only. For a network with other traffic situation, the situation is more of a real environment where not only the use of a special network to VoIP but it is used for other traffic such as ftp, http and others commonly used in traffic in a network.

The most important thing to the results of the simulation analysis is made from the type of codec use and the value of the SNR. Simulation results should have values to illustrate the performance or the effect of using a codec. For performance result of both proposed VoIP simulation, we use the MOS and R-factor as the value that determines the quality of a call that has been set by Telecommunication Union (ITU). The VoIP session will be capture and analyze by the wireless monitoring tool to display the actual performance result. These tools use a USB wireless adapter as a tool for network and session sniffing.

The simulation of this experiment is modeled as the following:

1. “Client 1” talking and the other client is silent
2. “Client 2” talking and the other client is silent

The experiment wills use the recorded voice to maintain the consistent of the voice during 10 calls is made. Sample of the material that will use during the call is:

“*When you’re a carpenter making a beautiful chest of drawers, you’re not going to use a piece of plywood on the back, even though it faces the wall and nobody will ever see it. You’ll know it’s there, so you’re going to use a beautiful piece of wood on the back. For you to sleep well at night, the aesthetic, the quality, has to be carried all the way through*”

The testbed set up as below:

i. Two hosts is set up
ii. One access point as a link between two clients has to be set up and configured with the respective SNR

iii. Tools need to be installed at the required place where the function is needed (user agent, SIP server, network monitoring tool).

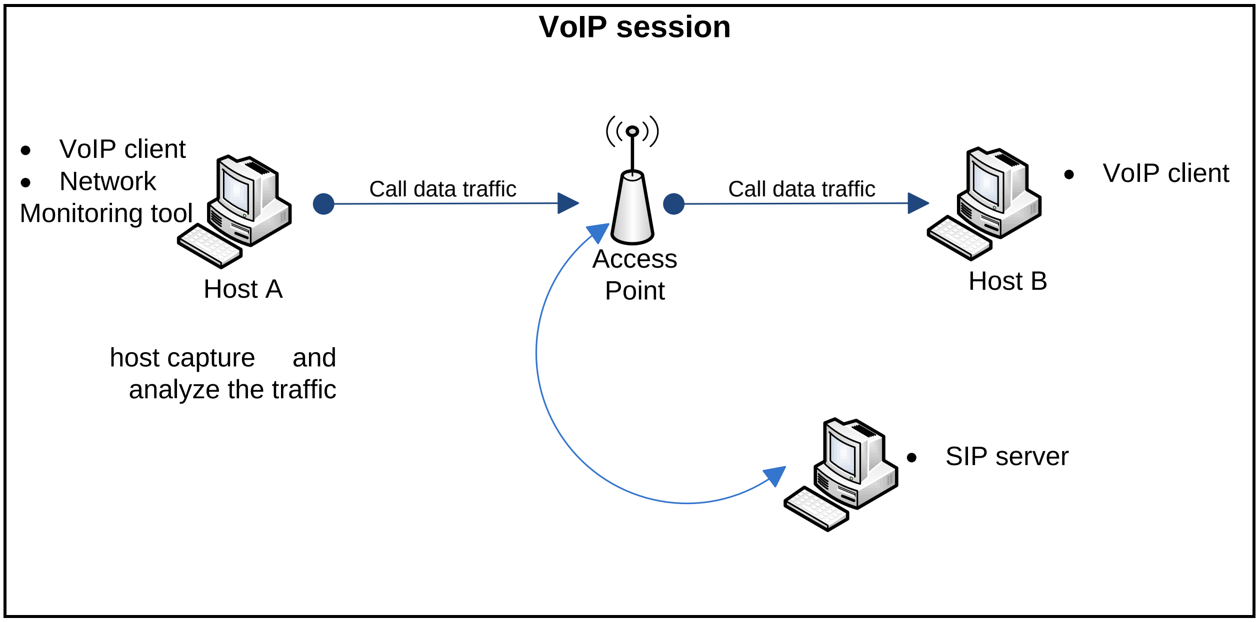


Figure 3: Diagram for the testbed

3.1.4 Testing Phase

After the architecture design phase, the testing phase need to be implement to test the simulation and get the analysis report that state in the objective in this research. When the environment for simulation has been prepared, the tools and hardware should try and run the tools that have been installing to produce the desired analysis report. Tool for analyzing the simulation results must be going well because it can lead to unsatisfactory results will be produced by the tool.

CommView we use as a tool to help generate analysis apply to a session that has been created. This tool is able to generate the report of MOS, R-factor, packet size, jitter, and to be generated at the end of the session. Other than the production values in tabular form, this tool are also able to generate some graphs that will be used to augment and clarify the results generated for this simulation.

Simulation results should be accurate, to obtain accurate simulation results. Simulations will be carried out up to ten times to produce a more than one reading to process of calculating the average of the simulation results can be made. After the process of collecting the required reports completed, the tables will be used to display the results for each situation in the simulation have been performed. The average reading for 10 samples will be taken as the final result.

<Type of environment>											
Readings / Codecs	**1st**	**2nd**	**3rd**	**4th**	**5th**	**6th**	**7th**	**8th**	**9th**	**10th**	**MOS Mean**
	MOS	**MOS**	**MOS**	**MOS**	**MOS**	**MOS**	**MOS**	**MOS**	**MOS**	**MOS**	
<codec type>											
<codec type>											
<codec type>											
<codec type>											
<codec type>											

Table 11: Sample of the MOS analysis result table

3.1.5 Data Analysis Phase

The last phase of the research is data analysis phase. Analysis of data is a process of inspecting, transforming, and modeling data with the goal of highlighting useful information, suggesting conclusions, and supporting decision making. The data were generated from an analysis derived from simulation should achieve the objectives of the study which is the purpose of research is done. For this research, based on analysis of voice quality and bandwidth codec used is a very important analytical report for the report as the simulation results. As an example for the simulation to be made CommView tool will produce MOS and R-factor for a VoIP call is made.

Chapter 4

DESIGN AND IMPLEMENTATION

4.0 Introduction

The design and Implementing phase is very important to plan and state the step by step design and implementation for this simulation, so that the desired outcome is achieved by conducting the simulation process. Design and implementation must be defined according to the objectives that were stated to ensure that the research achieve what actually the main objective of the research. To perform the appropriate and necessary simulations for this research, simulation will be perform using two computers in the same network interconnection and an access point as a bridge for the two computers. Next, hardware used in the simulation topology is supported by tools that carry out their respective functions as intended.

4.1 Experimental Environment Design

In this phase, the hardware installation for experimental architecture and installation of the tools that is use for create a VoIP call session and also a tool to analyze the performance of the codec used in the sessions is setup. Some of the activities will be carried out to complete the process of preparing this testbed environment. The activities involved are:

i. **Computer (Caller)** - Two computers is used as a caller for VoIP call session. One of the two computers is installed with a SIP server that function to control the call session and the other one computer is perform network monitoring tools to sniff network utilization and bandwidth usage, and VoIP client to be used as a tool to use VoIP service and can determine which codec will be used for each call. The tools that have been selected to perform the following tasks are Eyebeam, Ekiga, Brekeke SIP Server and CommView. Table below describes the use of the tools involved in the testbed

Tools	Function
Ekiga, X-Lite and Eyebeam	Used as VoIP client or the softphone.
Brekeke	Function as SIP server that can control the VoIP session.
CommView	Network monitoring tool that capture the packet loss and jitter. Generate MOS score and jitter as well.

Table 12: Function of tools that will be used in tesbed

ii. **Wireless access point** - Access point provides wireless connection to establish connection between the two hosts. This device makes the caller, server and network monitoring tool reside in the same network. Access point is used to create a network that can establish the connection between the two callers. Access point will be installed and configured to ensure that both the caller can communicate with each other. In addition, this access point using DD-WRT system that can measure the SNR between access point and host.

iii. **Testbed** - There are four types of environments used in the simulations that will be done in this research is:

- Two ways voice communication on optimum network with respective SNR value 10
- Two ways voice communication on optimum network with respective SNR value 20
- Two ways voice communication on optimum network with respective SNR value 30
- Two ways voice communication with other traffic with respective SNR value 10
- Two ways voice communication with other traffic with respective SNR value 20
- Two ways voice communication with other traffic with respective SNR value 30
-

Environment that will be used is shown in the diagram below:

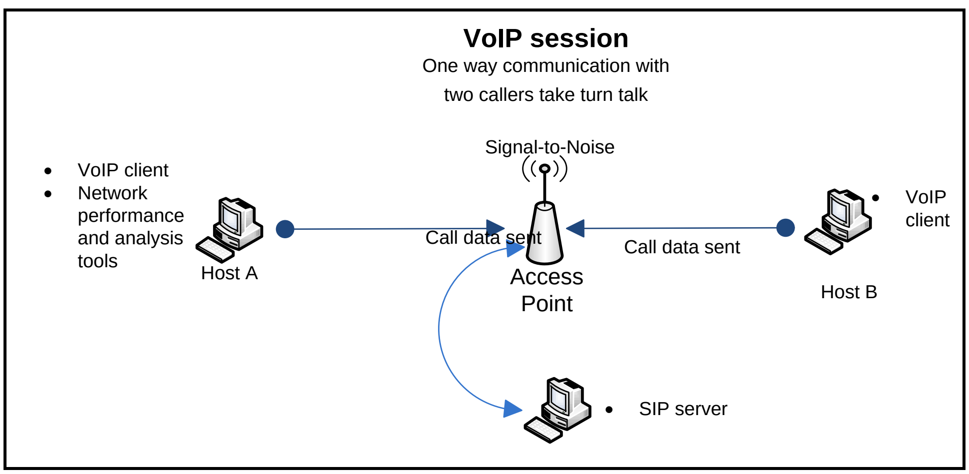


Figure 4: Two ways communication testbed

The environments or situation that will be perform in this research:

- **The VoIP call session using the optimal network connectivity (one way and two way communication)** - For this environment, the network in optimum condition without any traffic. The network will be dedicated totally to VoIP call session only. This is to test the actual performance of a codec call using the codecs listed.

- **VoIP call session with other traffic such as ftp (one way and two way communication) -** For the second environment, the network connection that is used to create a VoIP call session will be included with other traffic such as FTP. Callers not only use the network for VoIP call session, in fact, the caller also using the network to the other traffic. This can result in the performance of such real-world networks.

iv. **Architecture** – There are only one architecture involve in this simulation. Two computers as a tool for callers in touch with the same access point. One of the callers will be put together with a USB wireless adapter that works to monitor the use of the network and the quality of VoIP calls. In one of the caller's computer will also be installed SIP server operators to VoIP call session will be established.

4.2 Testing Plan

When the environment, equipment and tools to do the simulation were complete prepared and installed. Process for testing and analyzing the VoIP call session can be start. Testing will be use the environment and architecture that have been described in environment design phase. Ten readings will be taken to ensure the accuracy of the analysis is generated. The ten readings that have been generated will be taken and the average of the ten readings will be calculated and used for the final value of the performance and quality of a VoIP call.

For each testbed that will be done, four tables will be produced and each table will contain ten results that contain the value MOS, R-factor, packet loss and jitter. After all tests have been completed the results will generated and shown in tabular format, the value of which has been generated will be incorporated into the tables according to the category to make the process of analyze the data easier. The value that have been added to the table will also be used to produce an average charts for the value MOS, R-factor, packet loss and jitter. Produced charts purpose is to show a clearer reading to determine and compare the performance of the codec used during VoIP calls. The for tables that will be used in one way communication testbed environment are as the following:

One way communication on optimum network with respective SNR 10dB											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	MOS Mean
	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	
G.729											
Speex											
iLBC											
GSM											

Table 13: One way communication on optimum network with respective SNR 10dB based on MOS score

One way communication on optimum network with respective SNR 10dB											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	R-factor Mean
	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	
G.729											
Speex											
iLBC											
GSM											

Table 14: One way communication on optimum network with respective SNR 10dB based on R-factor

One way communication on optimum network with respective SNR 10dB											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Loss Mean
	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	
G.729											
Speex											
iLBC											
GSM											

Table 15: One way communication on optimum network with respective SNR 10dB based on packet loss

One way communication on optimum network with respective SNR 10dB											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Jitter Mean
	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	
G.729											
Speex											
iLBC											
GSM											

Table 16: One way communication on optimum network with respective SNR 10dB network based on jitter

The table is same as this four table when the process filling the data in tabular form for one way communication on optimum and network with other traffics with respective SNR 10, 20 and 30 dB.

Chapter 5

Result & Discussion

5.1 Introduction

In this chapter, the result of each testbed will be discuss and analyze. The voice quality that has been processed by the three voice codecs G.729, G.711 uLaw and G.711 aLaw using wireless LAN that has the ratio of SNR 10dB, 20dB and 30dB will be determined. The performance for each codec can be determined by MOS and R-Factor value taken during the test. Many factors can influence the reading of MOS and R-Factor, including what is to be shown in this experiment that the rate of SNR together with optimum network and network with other traffic condition. The SNR, packet loss and jitter are related and can be among of the parameter that has to be taken into consideration to determine the MOS and R-Factor is produced by the network monitoring tool. Theoretically, VoIP packet loss occurs when a large amount of traffic on the network that can cause dropped packets. This results in dropped conversations, a delay in receiving the voice communication, or extraneous noise on a wireless signal (SNR). For jitter to happen, the jitter happens because of at the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Due to the low SNR ratio this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant. Sometimes, the receiver must make a request for a packet that has dropped before; this situation will also can cause a packet delay (jitter). The results for each test performed will be important information to make the final assessment to determine the best performance among the three codecs used. These test results will also be able to provide information to determine the optimal environment to applying voice data transmission using wireless LAN.

5.1.1 Testbed 1: One way communication on optimum network with respective SNR 10dB

The first testbed is using all three codecs to make a call within the network operating with 10dB SNR rate. The results show that the codec can be issued G.711 aLaw is the highest rate of MOS and R-Factor that prove the quality of voice calls made within was the best among the three codecs used. The low packet loss rate has influenced the rate of MOS and R-Factor that has been generated when call using G. 711aLaw is high. For G.729 and G.711 uLaw, the high of the packet loss and jitter rate than the rate when using G.711 aLaw that caused the rate of MOS and R-Factor decreased. Other than the packet loss and jitter factor, another factor that influence the MOS and R-Factor reading is the ratio of the SNR for a client call. On this testbed, the SNR is 10dB; the 10dB SNR rate can be considered as a low ratio and can make interference on voice packet transmission in calls that has been made. With 10dB SNR will keep rates low packet loss and jitter increase as voice data cannot be transmitted properly.

One way communication on optimum network with respective SNR 10dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS
G.729	3.6	3.8	4.1	4.1	4	4.1	4.1	4.1	4.1	4.1	4.01
G.711 aLaw	4.3	4.4	4.4	4.4	4.3	4.4	4.4	4.4	4.3	4.4	4.37
G.711 uLaw	4.4	4.3	4.3	4.3	4.3	4	4.3	4.2	4.4	4.3	4.28

Table 17: MOS readings of one way communication on optimum network with respective SNR 10dB

One way communication on optimum network with respective SNR 10dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor
G.729	70.8	74.7	82.1	82.1	80.5	82.8	81.7	81.3	81.3	82.1	79.94
G.711 aLaw	88.7	91.8	91.8	91.7	89.8	91.8	91.7	93.2	90.4	91.8	91.27
G.711 uLaw	93.2	90.4	90.2	88.9	89.1	79.1	87.2	84.3	93.2	89.6	88.52

Table 18: R-Factor readings of one way communication on optimum network with respective SNR 10dB

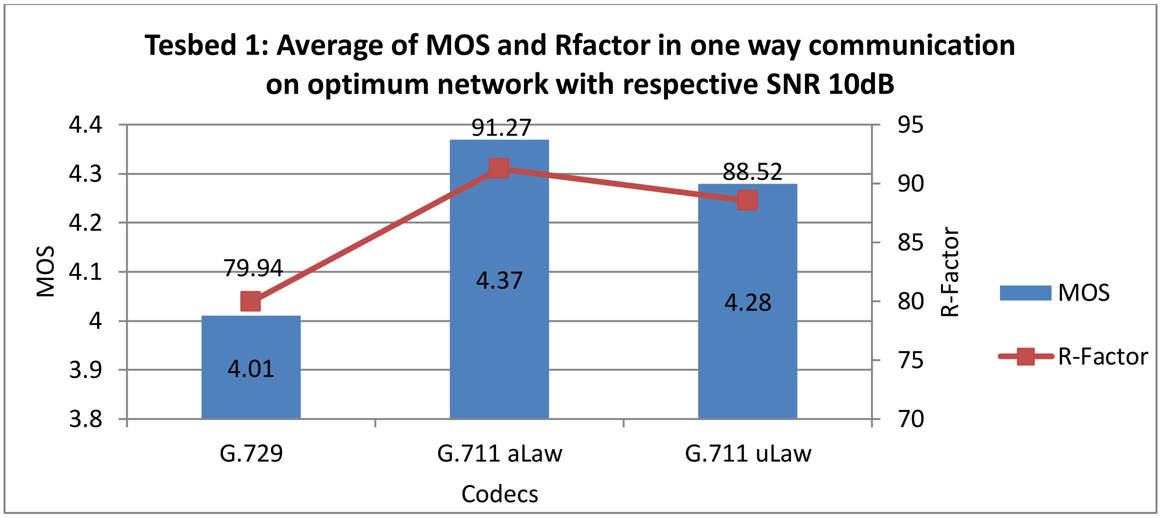


Figure 6: Average of MOS and R-factor in one way communication on optimum network with respective SNR 10dB

One way communication on optimum network with respective SNR 10dB											
Codecs	Readings										Mean Packet Loss
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	
	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	
G.729	2.9	1.9	0.1	0.2	0.6	0.1	0.3	0.4	0.4	0.2	0.71
G.711 aLaw	0.5	0.1	0.2	0.2	0.4	0.2	0.2	0	0.3	0.2	0.23
G.711 uLaw	0	0.3	0.3	0.5	0.5	1.7	0.7	1	0	0.4	0.54

Table 19: packet loss readings of one way communication on optimum network with respective SNR 10dB

One way communication on optimum network with respective SNR 30dB											
Codecs	Readings										Mean Jitter
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	
	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	
G.729	2.6	2.3	2.9	2.3	3.1	1.9	2.1	2	3.4	2.4	2.5
G.711 aLaw	2	3.3	2.2	2.5	2.1	2.7	2	2.3	2	3	2.41
G.711 uLaw	2.3	2	2.5	3.1	2.1	3.2	1.9	2.1	2.3	2.5	2.4

Table 20: Jitter readings of one way communication on optimum network with respective SNR 10dB

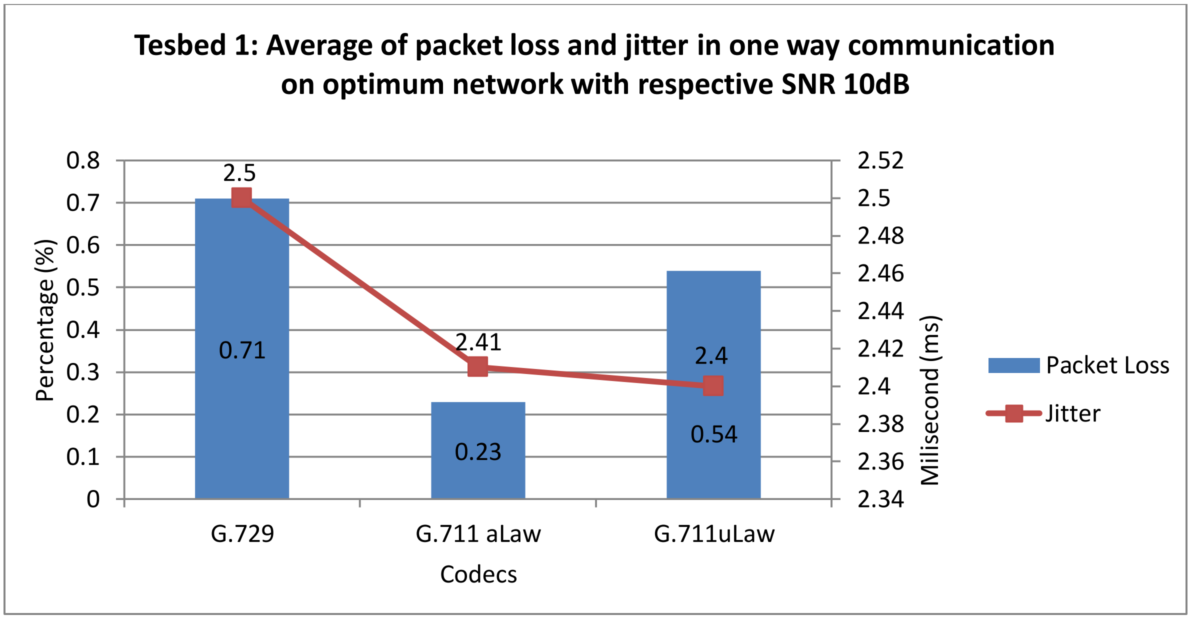


Figure 7: Average of packet loss and jitter in one way communication on optimum network with respective SNR 10dB

5.1.2 Testbed 2: One way communication on optimum network with respective SNR 20dB

Different parameters are used in the second testbed is only the second testbed uses the wireless network has 20dB SNR ratio. As described in the previous chapter, the higher the SNR ratio the better quality voice transmission can be made. With 20dB SNR ratio, the better reading of the MOS and R-Factor will be produced. The average can be produced by the three codecs showed readings in excess of 4.0 MOS and R-Factor readings above 80.0, the value for both of these parameters indicate the quality of voice transmission in the calls made are at a satisfactory level. Different with the readings of MOS and R-Factor, packet loss and jitter readings produce when calls has been made is decrease due to the increased of the SNR ratio for a signal on the network makes the least packet loss rate and jitter on voice data transmission. For the second testbed, call a good quality voice transmission is G.711aLaw call using the gain rate codec 4.37 MOS, R-Factor 91.3, 0.9% packet loss and jitter 2.1 ms.

One way communication on optimum network with respective SNR 20dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS
G.729	4	4.2	4	3.8	3.9	4.1	4	4.1	4.1	4.1	4.03
G.711 aLaw	4.4	4.3	4.4	4.4	4.4	4.4	4.4	4.4	4.2	4.4	4.37
G.711 uLaw	4.4	4.4	4.4	4.2	4.4	4.4	4.4	4.4	4.3	4.4	4.37

Table 21: MOS readings of one way communication on optimum network with respective SNR 20dB

One way communication on optimum network with respective SNR 20dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor
G.729	82.6	81.5	79.1	73.9	81.2	81.7	78.6	82.5	82.8	82.8	80.67
G.711 aLaw	91.8	88.5	91.7	93.2	93.2	91.4	93.2	92.5	85.1	92.4	91.3
G.711 uLaw	92.4	92.5	90.6	86.2	92.5	92.5	92.5	92.3	87.8	93.2	91.25

Table 22: R-Factor readings of one way communication on optimum network with respective SNR 20dB

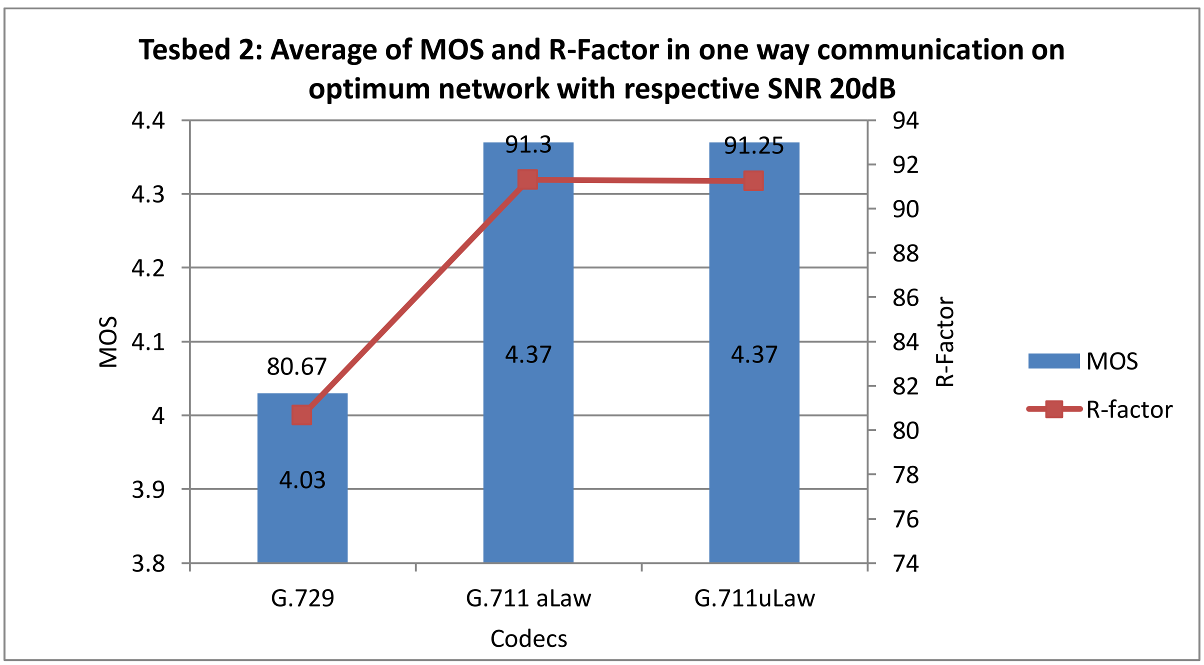


Figure 8: Average of MOS and R-Factor in one way communication on optimum network with respective SNR 20dB

One way communication on optimum network with respective SNR 20dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean Packet
Codecs	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Loss
G.729	0.8	0.2	0.9	2.1	1.3	0	0.1	0.1	0.1	0.1	0.57
G.711 aLaw	0.1	0.1	0	0	0	0.2	0	0.1	0.3	0.1	0.09
G.711 uLaw	0.1	0.1	0	0	0.1	0.1	0.1	0	0	0	0.05

Table 23: packet loss readings of one way communication on optimum network with respective SNR 20dB

One way communication on optimum network with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter
G.729	1.9	2.8	2.5	2.3	2	1.9	2.1	2	2.3	1.9	2.17
G.711 aLaw	2	1.9	2.2	2.5	1.9	1.9	2	2.2	2	2.4	2.1
G.711 uLaw	2	2.2	1.9	2.2	2.1	2.6	1.9	2.4	1.8	2	2.11

Table 24: Jitter readings of one way communication on optimum network with respective SNR 20dB

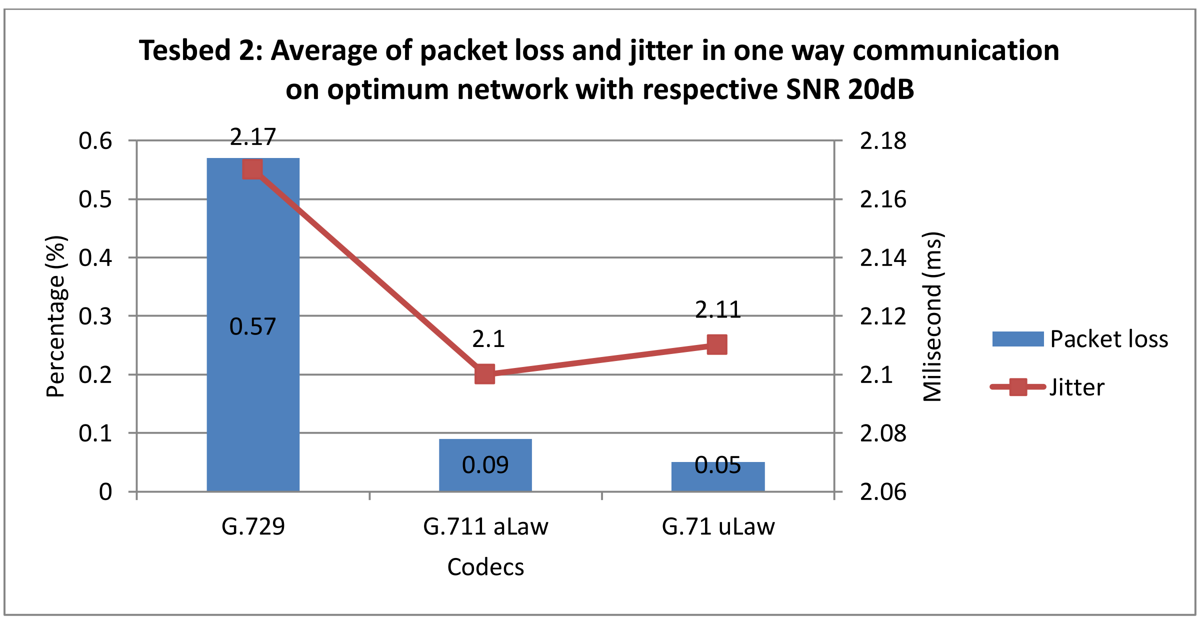


Figure 9: Average of packet loss and jitter in one way communication on optimum network with respective SNR 20dB

5.1.3 Testbed 3: One way communication on optimum network with respective SNR 30dB

For the third testbed, 30dB SNR ratio is set to a network that is used to make calls. The result of ten calls has been made, average MOS, R-Factor, packet loss and jitter can be produced and show calls that produce good sound quality is G.711uLaw codec. The rate of 4.4 MOS, R-Factor 92.79, 0% packet loss and jitter 1.95 ms can be read on the call using codec G.711uLaw. Reading 0% packet loss is very good for data transmission services as cool by VoIP. 30dB SNR ratio is helpful in producing a very good voice quality for VoIP calls.

One way communication on optimum network with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS
G.729	4.1	3.9	4.1	4.1	4.1	4.1	4.1	4.1	4.1	4	4.07
G.711 aLaw	4.4	4.4	4.4	4.4	4.4	4.4	4.4	4.4	4.4	4.4	4.4
G.711 uLaw	4.4	4.4	4.4	4.4	4.4	4.4	4.4	4.4	4.4	4.4	4.4

Table 25: MOS readings of one way communication on optimum network with respective SNR 30dB

One way communication on optimum network with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor
G.729	83.2	77.7	82.9	82.8	80.8	83.2	83.2	81.6	83.2	78.7	81.73
G.711 aLaw	93.2	93.2	93.2	93.2	92.5	91.7	93.2	93.2	92.5	92	92.79
G.711 uLaw	93.2	93.2	93.2	93.2	93.2	93.2	93.2	93.2	93.2	93.2	93.2

Table 26: R-Factor readings of one way communication on optimum network with respective SNR 30dB

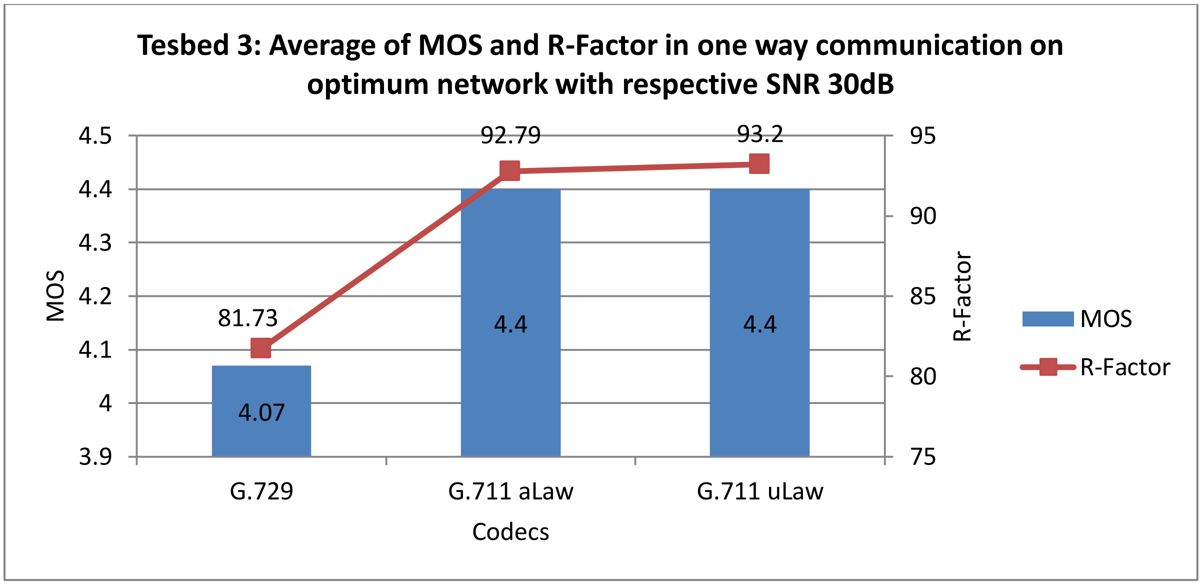


Figure 10: Average of MOS and R-Factor in one way communication on optimum network with respective SNR 30dB

One way communication on optimum network with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean Packet
Codecs	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	Loss
G.729	0	0	0.1	0.1	0.5	0	0	0.1	0	0	0.08
G.711 aLaw	0	0	0	0	0.1	0.2	0	0	0.1	0.1	0.05
G.711 uLaw	0	0	0	0	0	0	0	0	0	0	0

Table 27: packet loss readings of one way communication on optimum network with respective SNR 30dB

One way communication on optimum network with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter
G.729	1.9	1.7	1.7	2.3	1.7	1.9	2.1	2	1.6	1.9	1.88
G.711 aLaw	2	1.9	2.2	1.8	1.9	1.9	2	1.7	2	1.8	1.92
G.711 uLaw	1.8	2	1.9	2.2	2.1	2	1.9	1.8	1.8	2	1.95

Table 28: Jitter readings of one way communication on optimum network with respective SNR 30dB

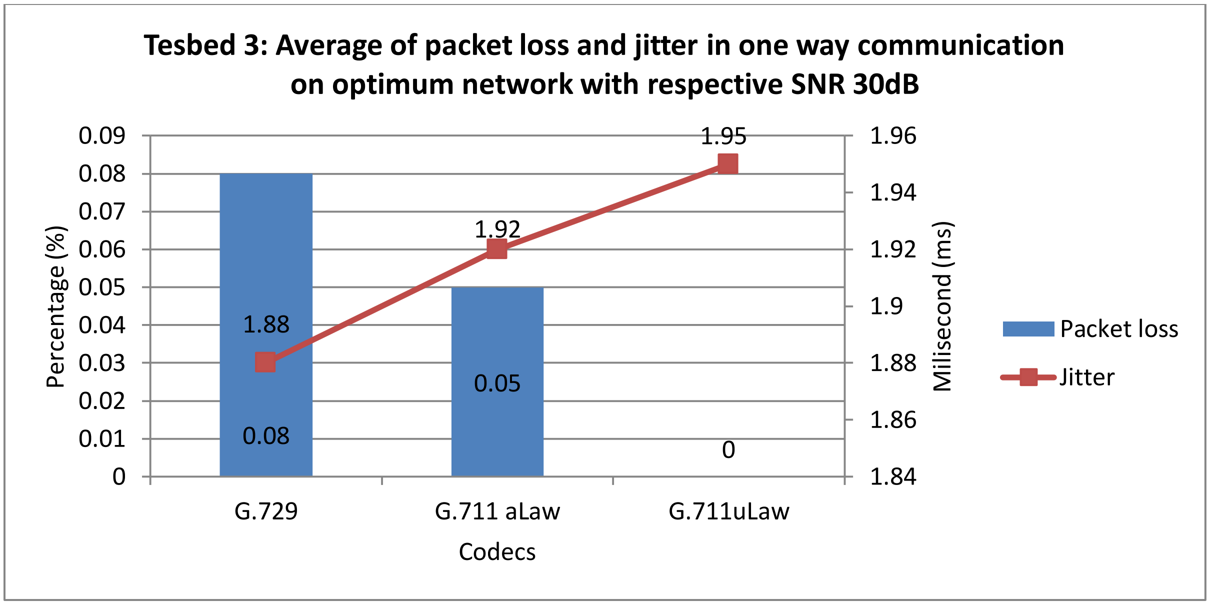


Figure 10: Average of packet loss and jitter in one way communication on optimum network with respective SNR 30dB

5.1.4 Testbed 4: One way communication on network with other traffic with respective SNR 10dB

After the third testbed is using the ratio of 30dB SNR, the 10dB SNR to be returned for the SNR can be implemented in the fourth testbed. But this time the network is no longer dedicated the traffic for VoIP sessions only. For the fourth testbed, the network connection between two callers has been shared by putting the process of transferring files from client A to client B. Such network conditions are very clear impact on VoIP sessions because of the network bandwidth that can be provided has to be shared; it would be the voice quality during a conversation will be dropped. As we can see from the results of tests in the table and graph below, codec G.711aLaw recorded the highest average based on the value of MOS and R-Factor which is 3.97 for the MOS and 80.07 for R-Factor. The packet loss of this codec is less than the G.711uLaw codec but it is more than G.729. However, the value of packet loss was balanced by the value of packet jitter which G.711aLaw have less jitter than the other two codecs. In addition, the QOS for VoIP to be taken into consideration is the ratio of 10dB SNR for this testbed. As we know 10dB SNR is less suitable to be used to process voice data transmission because the signal has 10dB SNR ratio is less stable. The problem of low SNR ratio is added with the traffic of data transfer on a network has produced the results where the value of high packet loss and jitter value.

One way communication on network with other traffic with respective SNR 10dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS
G.729	4	4.1	4.1	4.1	4.1	3.6	3.3	3.5	4.1	3.6	3.85
G.711 aLaw	3.4	3.9	4.3	3.9	4.4	3.6	4.3	4.4	4.2	3.3	3.97
G.711 uLaw	3.2	3.8	3.3	3.6	3.4	2.9	3.4	2.8	3.5	3.6	3.35

Table 29: MOS readings of one way communication on network with other traffic with respective SNR 10dB

One way communication on network with other traffic with respective SNR 10dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor
G.729	79.5	81	83.2	82.5	83.2	69.7	64.7	68.5	82.5	70.5	76.53
G.711 aLaw	65.4	77.9	89.9	77.2	90.9	70.5	88.9	91.9	85.1	63	80.07
G.711 uLaw	61.3	74	63.3	70.9	65.4	56.3	66.3	54.8	68.7	70	65.1

Table 30: R-Factor readings of one way communication on network with other traffic with respective SNR 10dB

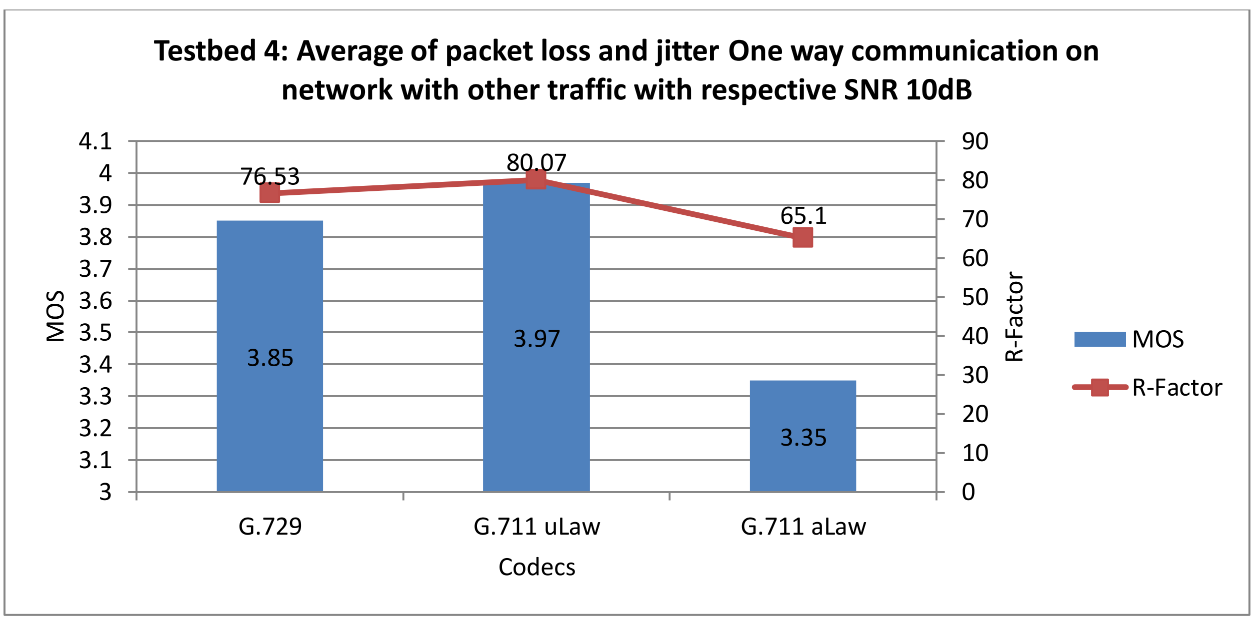


Figure 11: Average of MOS and R-Factor in one way communication on network with other traffic with respective SNR 10dB

One way communication on network with other traffic with respective SNR 10dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean Packet
Codecs	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Loss (%)
G.729	0.8	0.5	0	0.1	0	3.2	4.7	3.5	0.1	3	1.59
G.711 aLaw	4.1	1.9	0.4	2.1	0.3	3.1	0.5	0.1	0.9	4.7	1.81
G.711 uLaw	5	2.5	4.6	3.1	4.1	6.3	4	6.8	3.5	3.2	4.31

Table 31: Packet loss readings of one way communication on network with other traffic with respective SNR 10dB

One way communication on network with other traffic with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter
G.729	10.9	9.1	8.8	9.5	11.1	13.2	9.3	12.2	10.2	11.7	10.6
G.711 aLaw	11.2	10.7	10.2	9.8	9.2	8.2	13.4	8.2	9.7	9.4	10
G.711 uLaw	12.3	10	9.2	10.4	9.3	9.7	9.6	11.9	12.8	13.1	10.83

Table 32: Jitter readings of one way communication on network with other traffic with respective SNR 10dB

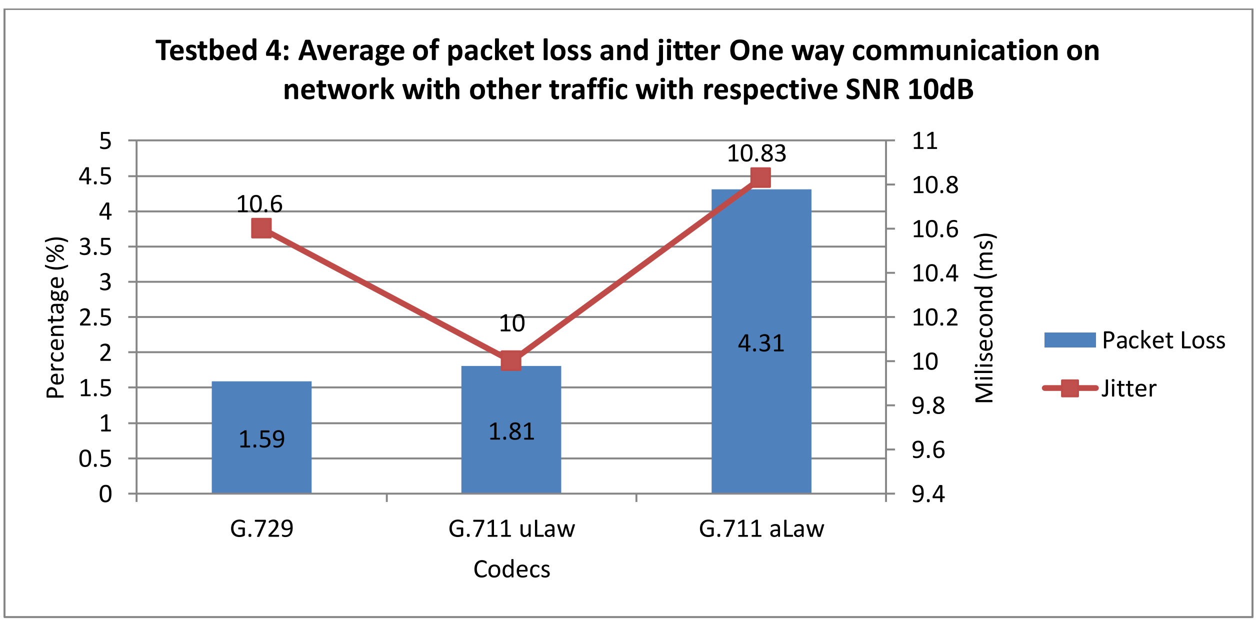


Figure 12: Average of packet loss and jitter in one way communication on network with other traffic with respective SNR 10dB

5.1.5 Testbed 5: One way communication on network with other traffic with respective SNR 20dB

The results for a fifth testbed are the same with the previous testbed but it is used wireless signal that has 20dB SNR ratio. The SNR ratio is increased to 20dB S and tested in a network with other traffic that shared with VoIP session. G.711 aLaw once again showed a good call quality based on the value of MOS and R-Factor is higher than the other three codecs used. MOS recorded reached 4:24, while the R-Factor is 87.14. Value of MOS and R-Factor is proving that the quality of voice data transmission is very good when a VoIP call using a codec G.711aLaw in this network environment. G.711aLaw reduces jitter generated when the call is made. Rate produced by G.711aLaw jitter is less than two other codecs. However, ten readings of packet loss generated by the G.711aLaw are higher than the G.729 codec. However, this does not result in packet loss of high jitter in transmission of voice data.

One way communication on network with other traffic with respective SNR 20dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS
G.729	4.1	4.1	4.1	4.1	4	3.9	4.1	4.1	4.1	4.1	4.07
G.711 aLaw	4.4	4.3	4.4	4.4	4.2	3.8	4.4	3.9	4.4	4.2	4.24
G.711 uLaw	4.4	3.7	4.3	4.3	3.9	4.4	4.3	4.4	4	4.4	4.21

Table 33: MOS readings of one way communication on network with other traffic with respective SNR 20

One way communication on network with other traffic with respective SNR 20dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor
G.729	83.2	83.2	82.8	81.3	79	77.1	82.8	83.2	83.2	82	81.78
G.711 aLaw	91.2	89.1	91.8	93.2	86.1	75	91	76.3	91.1	86.6	87.14
G.711 uLaw	92.4	73.3	88.5	90.4	76.8	92.5	88.2	93.2	78.5	92.5	86.63

Table 34: R-Factor readings of one way communication on network with other traffic with respective SNR 20dB

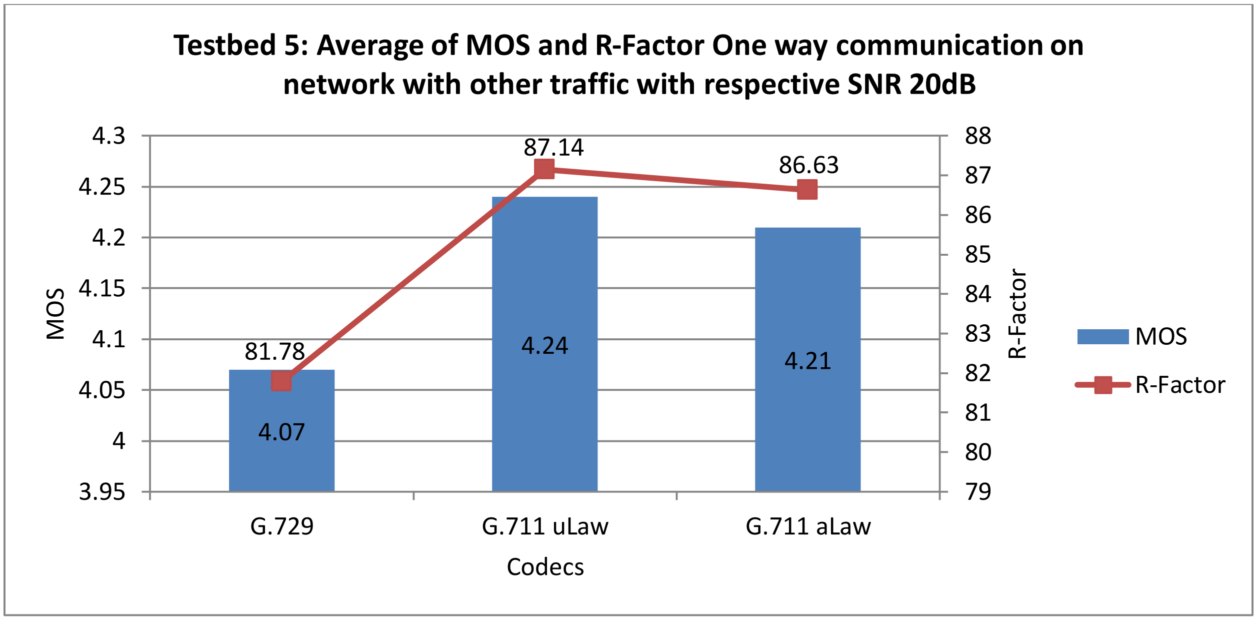


Figure 11: Average of MOS and R-Factor in one way communication on network with other traffic with respective SNR 20dB

One way communication on network with other traffic with respective SNR 20dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean Packet
Codecs	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Loss (%)
G.729	0	0	0.1	0.4	0.9	1.3	0.1	0	0	0.2	0.3
G.711 aLaw	0.2	0.5	0.2	0	0.8	2.4	0.2	2.2	0.2	0.7	0.74
G.711 uLaw	0.1	2.6	0.5	0.3	2.1	0.1	0.6	0	1.8	0.1	0.82

Table 35: Packet loss readings of one way communication on network with other traffic with respective SNR 20dB

One way communication on network with other traffic with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter
G.729	11.4	8.1	8.7	7.8	8.7	11.5	9.3	12.2	8.2	8.5	9.44
G.711 aLaw	10.2	8.3	9.6	11.3	9.2	8.2	7.9	8.2	8.1	9.4	9.04
G.711 uLaw	9.5	11.9	8.9	7.6	9.9	11.7	10.3	9.4	10.1	10.8	10.01

Table 36: Jitter readings of one way communication on network with other traffic with respective SNR 20dB

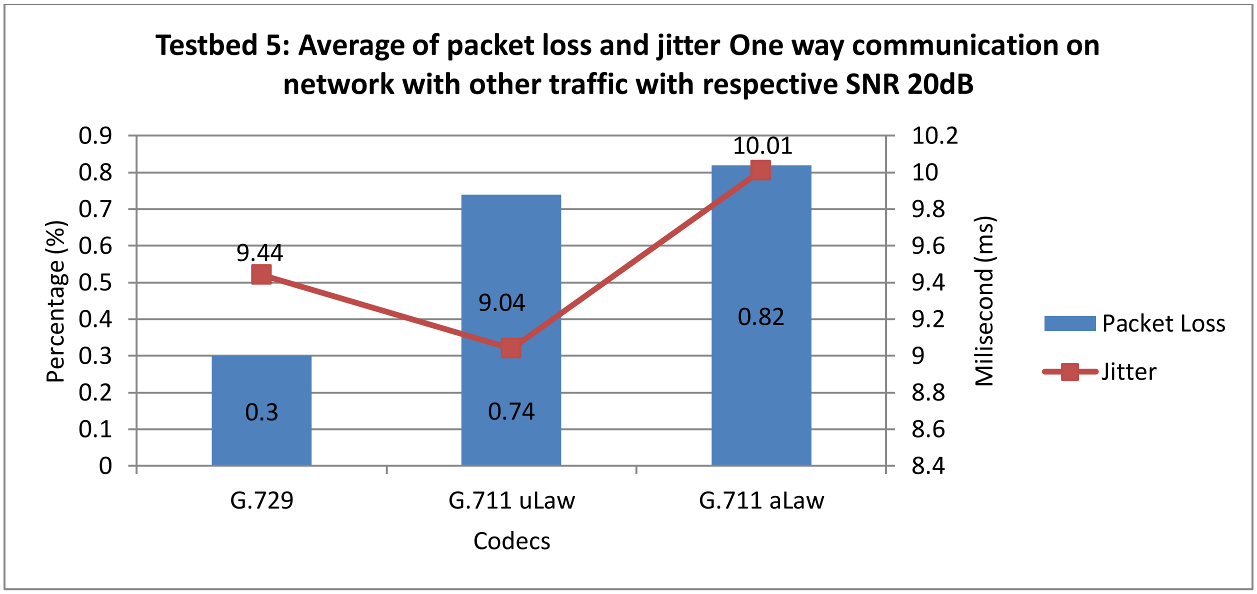


Figure 11: Average of packet loss and jitter in one way communication on network with other traffic with respective SNR 20dB

5.1.4 Testbed 6: One way communication on network with other traffic with respective SNR 30dB

In the final testbed, SNR ratio that can be used for wireless signal is at 30dB. In theory, the ratio SNR is 30dB and above the appropriate rate for doing the voice over data transmission rate below 30dB. SNR ratio is tested with the network used for VoIP sessions shared with other traffic of VoIP session. The results of ten tests performed on three codecs G.729 codec produces the value of show MOS and R-Factor the highest compared to the two other codecs. 3.74 MOS value and R-Factor 73.48 was generated from ten tests that have been made. Value MOS and R-Factor necessarily influenced by the high rate of packet loss and low jitter than other codecs. The rate of packet loss and jitter for G.729 is less than G.711aLaw and G.711uLaw codecs.

One way communication on network with other traffic with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	
G.729	4.1	4.1	3.8	4.1	3.8	3.6	3.8	3.4	3.3	3.4	3.74
G.711 aLaw	3.7	3.9	3.2	4.4	3.8	3.1	3.3	3.3	3.3	3.7	3.57
G.711 uLaw	3.7	3	3	3.4	4.4	4.4	3.8	2.7	4.4	3	3.58

Table 37: MOS readings of one way communication on network with other traffic with respective SNR 30

One way communication on network with other traffic with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor	R-Factor
G.729	82.8	82.8	74.4	82.1	74.7	69.4	73.9	65	64.7	65	73.48
G.711 aLaw	71.6	76.7	62.2	91.1	74.8	60.1	63.8	63.4	64.4	72.3	70.04
G.711 uLaw	73.2	57.3	58.8	65.4	93.2	92.5	73.9	53	91.7	57.8	71.68

Table 38: R-Factor readings of one way communication on network with other traffic with respective SNR 30

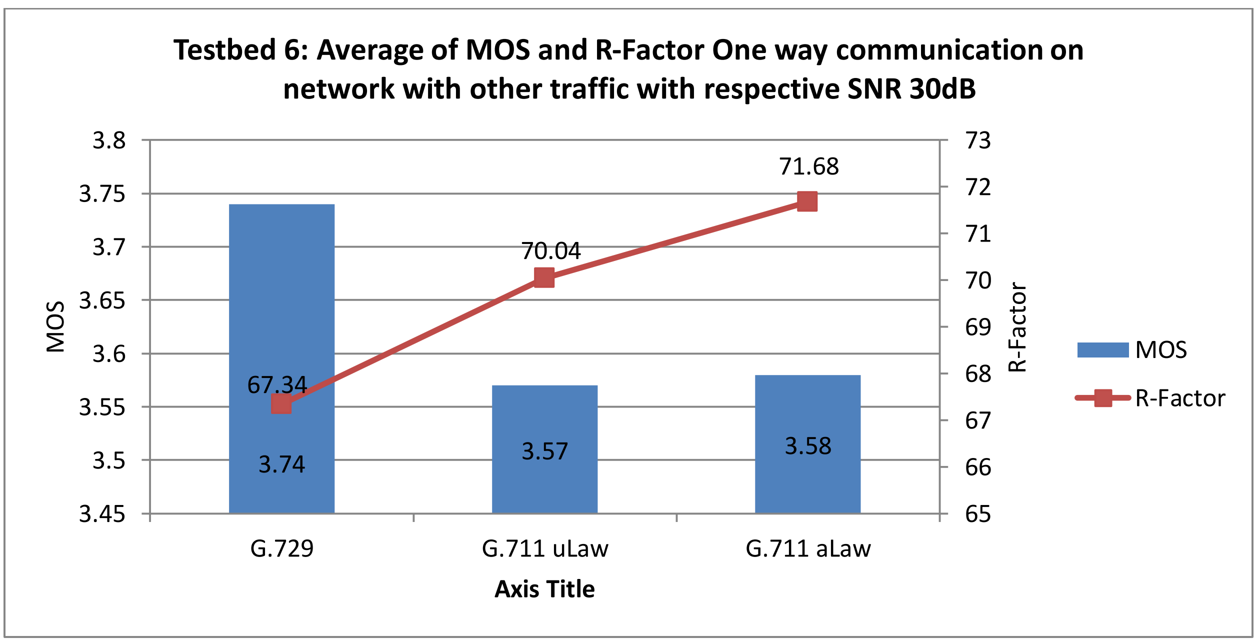


Figure 12: Average of MOS and R-Factor One way communication on network with other traffic with respective SNR 30dB

One way communication on network with other traffic with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean Packet Loss (%)
Codecs	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	Packet Loss (%)	
G.729	0.1	0.1	2	0.2	1.9	3.3	2.1	4.6	4.7	4.6	2.36
G.711 aLaw	2.9	2.1	4.8	0.2	2.4	5.4	4.5	4.6	4.4	2.8	3.41
G.711 uLaw	2.7	6.1	5.7	4.1	0	0.1	2.5	7.3	0.2	6	3.47

Table 39: Packet loss readings of one way communication on network with other traffic with respective SNR 30dB

One way communication on network with other traffic with respective SNR 30dB											
	Readings										
	1st	2nd	3rd	4rd	5th	6th	7th	8th	9th	10th	Mean
Codecs	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter	Jitter
G.729	7.3	8.1	8.7	7.8	7.6	8	7.2	7.7	8.2	8	7.86
G.711 aLaw	7.9	8.3	8	7.8	7.7	8.2	7.9	8.2	8.1	7.7	7.98
G.711 uLaw	9.5	11.9	7.6	7.6	4.9	6.6	10.3	8	6.6	10.8	8.38

Table 40: Packet loss readings of one way communication on network with other traffic with respective SNR 30dB

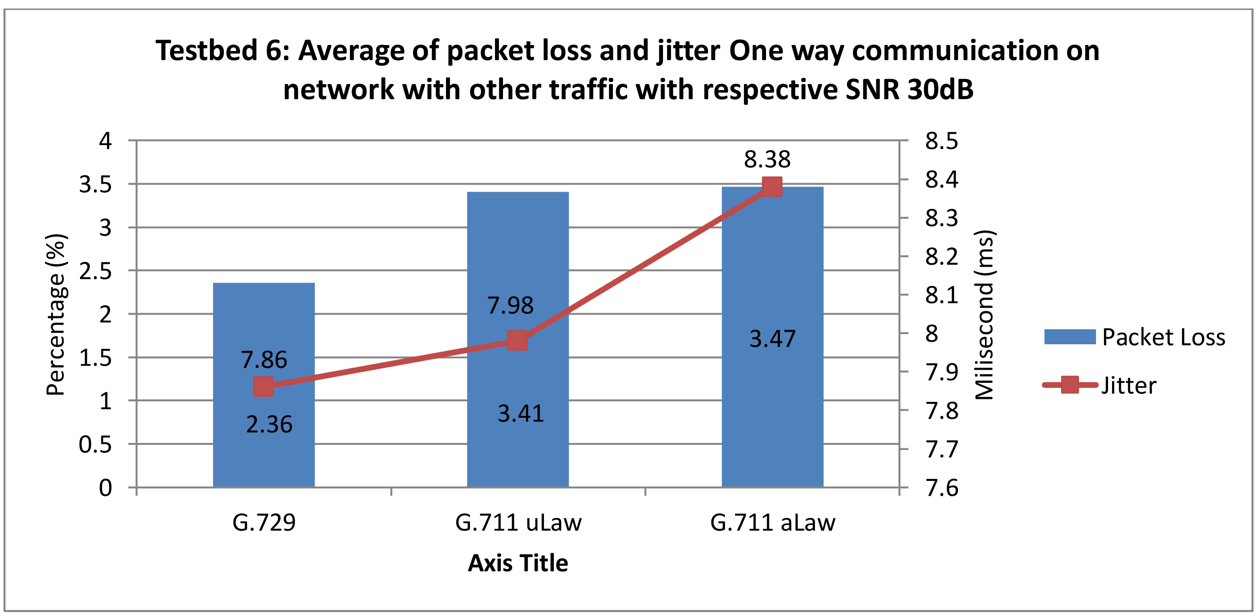


Figure 13: Average of packet loss and jitter one way communication on network with other traffic with respective SNR 30dB

Chapter 6

Discussion and Conclusion

6.0 Introduction

In this chapter, the final decision will be discuss which is to determine the best codec between three 8kbps codec as stated in the objectives of the study in the first chapter of the thesis. The best voice codec that will be chosen will be separate in the tabular and graph form based on the tesbed environment used such as SNR ratio and network conditions to facilitate the determination of the best codec performance.

6.1 Discussion

6.1.1 The Best Codec

Environments with **10dB SNR** ratio and optimal network should use the G.711 codec alaw. This statement is based on the mean value generated from ten VoIP calls that have been made. The good rate of MOS and R-Factor and less packet loss and jitter, making G.711 alaw are the best codec among the three codecs that has been used. The same codec is the best if the same environment is used on networks to\hat shared with other traffic (file transfer) also shows that G.711 aLaw is a codec that has good VoIP call quality.

Moreover, if the wireless network environment has a ratio of **20dB SNR**, the codec has good call quality is G.711 alaw. MOS value equal to G.729, but the R-Factor gives a value greater detail where quality calls using G.711 codec alaw is better than G.729. If the ratio of 20dB SNR is used in a shared network with other traffic, G.711 aLaw codec is also provides the best quality of VoIP calls. The two other codecs which is G.711 and G.729 uLaw also produces a good VoIP call quality, but it still not able to outperform G.711 aLaw.

The last environment is the one that has the highest SNR ratio which is **30dB**. Like the previous environment, using optimal network, G.711 alaw is produced the best quality of VoIP calls. The rates of MOS and R-Factor can be a proved that the high quality of VoIP calls using the G.711 aLaw as speech codec. In a shared network with other traffic, the codec that best demonstrate the quality of the call is G.729. G.729 codec beat two other readings 3.74 MOS, R-Factor is 73.48, 2.36% packet loss, and jitter 7.86.

6.1.3 MOS and R-Factor factor

In determining the quality of VoIP calls, two grading techniques that have been set before by the ITU-T is used such as the MOS and R-Factor. Observations in this experiment showed that the average of MOS value produced in all testbed is above 3.0, which are classified as "degradation slightly annoying". Furthermore, there are MOS value at above 4 which means that the call quality possible "degradation perceive but not annoying". This means that all calls that have been made in the experiment are in good quality and suitable for VoIP session.

However, for more accuracy determines the best call quality; the R-Factor is used to obtain the grade call quality in more detail. R-Factor is very helpful in determining the rate of call quality as MOS has a very limited value of 1 to 5. For example, the second testbed which is one way communication on optimum network with respective SNR 20dB, the mean of MOS between G.711 aLaw and G.711 uLaw is same. In this case, R-Factor can detail the grading of call using longer value range. So, although the value of the MOS is the same value between two codec, maybe the R-Factor value can be slightly different.

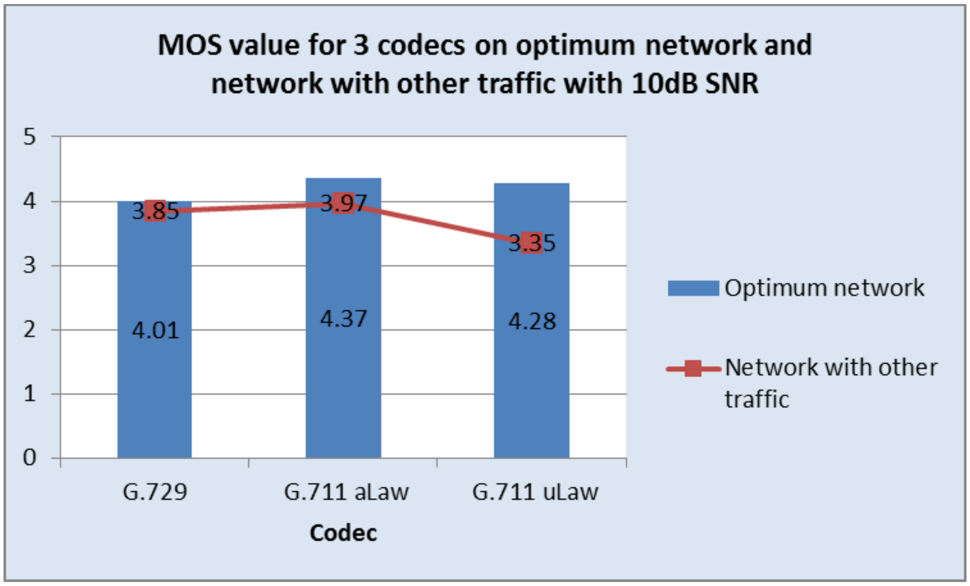


Figure 14: MOS value for 3 codecs on optimum network and network with other traffic with 10dB SNR

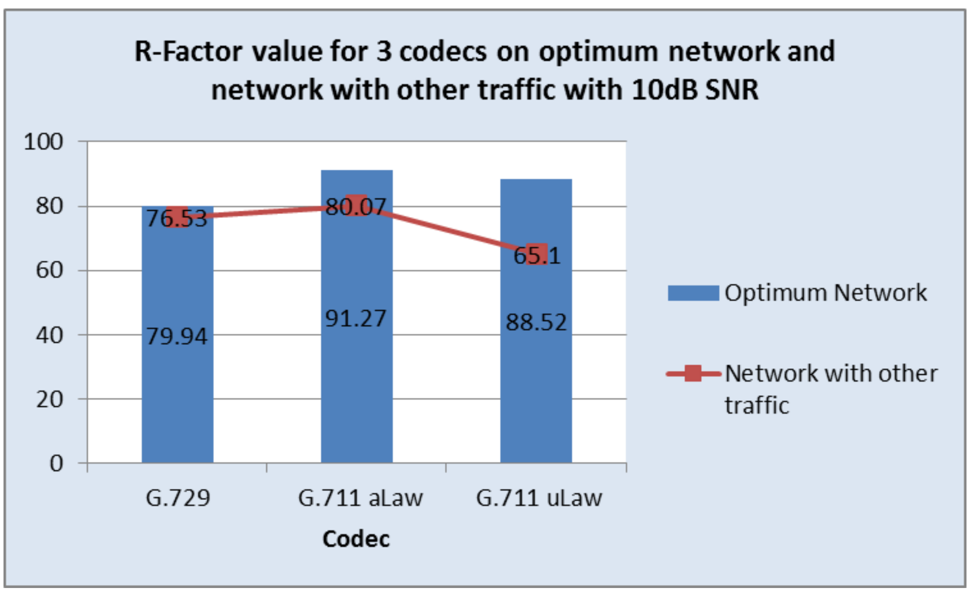


Figure 15: R-Factor value for 3 codecs on optimum network and network with other traffic with 10dB SNR

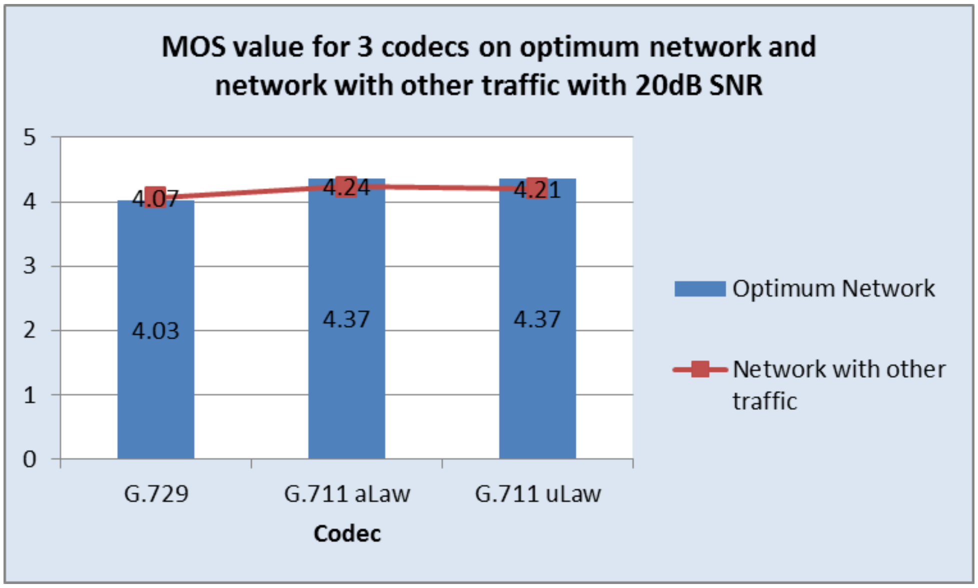


Figure 16: MOS value for 3 codecs on optimum network and network with other traffic with 20dB SNR

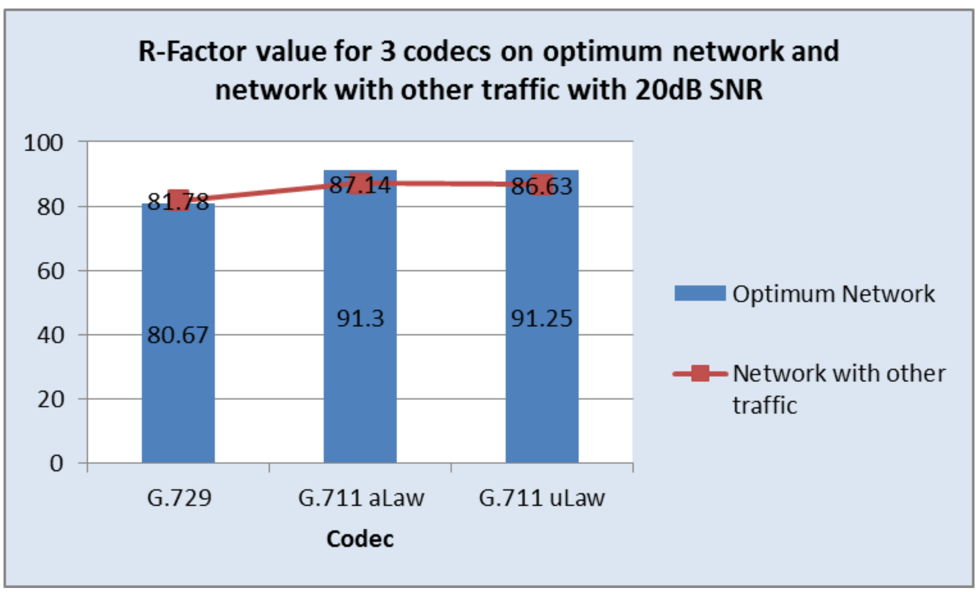


Figure 17: R-Factor value for 3 codecs on optimum network and network with other traffic with 20dB SNR

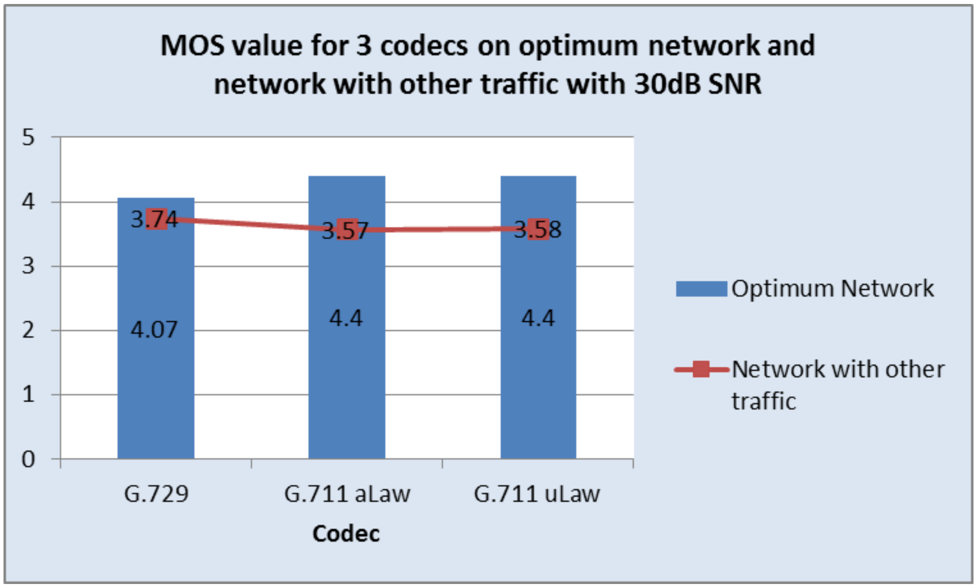


Figure 18: MOS value for 3 codecs on optimum network and network with other traffic with 30dB SNR

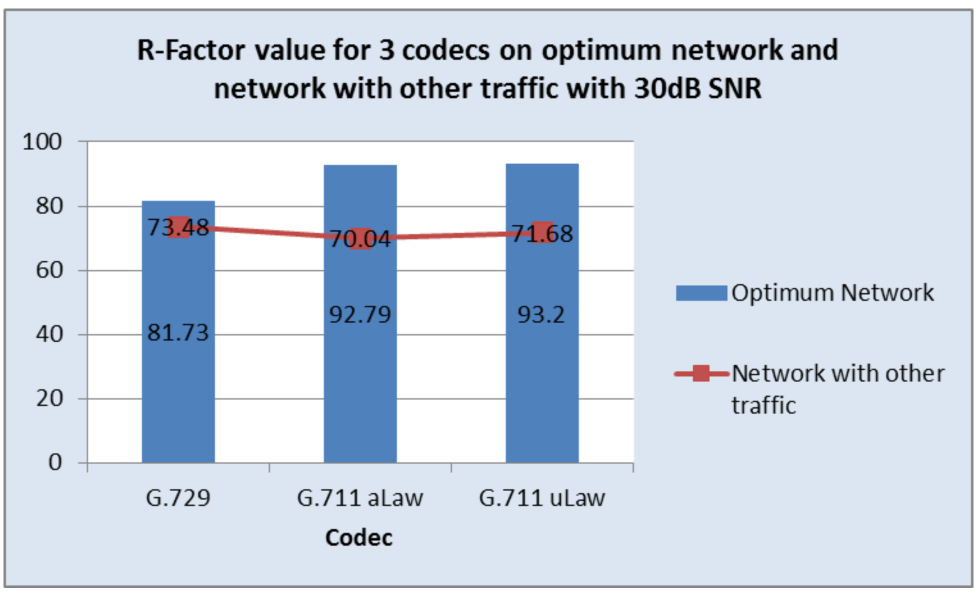


Figure 19: R-Factor value for 3 codecs on optimum network and network with other traffic with 30dB SNR

6.1.4 Packet lass and jitter (QOS) factor

Apart from MOS and R-Factor, some typical problems that occur in a VoIP call can also be analyzed in the result of the experiment. Two QOS issues recorded during the experiment are packet loss and jitter. Both of these problems are really affecting the quality of the calls that has been made in the experiment. The result in the graph and table below can showed that when both problem of the network which is packet loss and jitter reaches a low value, the MOS and R-Factor value is improved or higher compared to the situation of high packet loss and jitter. The high value of packet loss and jitter really affect the voice transmission and reduce the quality of call. Percentage of voice cannot be heard by the receiver is high when the high of packet loss happen during call.

The loss of packets should be taken care in using, manage or choosing a VoIP service. In this experience, on a call using the optimal network, the value of the packet loss is less than the calls made on the network with other traffic. Therefore, optimal network is very suitable for use as a VoIP environment due to the consistency of the transmission of voice. To maintain the good voice transmission, network bandwidth on the network should be able to accommodate with the size of the transmission of voice that is generated by voice codec during a call that allow the transmission to avoiding excessive packet loss. During the experiment, when the packet loss reaches 1%, a voice can be heard at the receiver end is not clear and interrupted voice. It means that the quality of call is decrease and it is not suitable to establish VoIP call.

Theoretically, jitter will obviously be identified when jitter rate exceeds the minimum rate of 150ms jitter that allows to being present in the network. When jitter exceeds 150ms, the caller will speak like using a walkie-talkie. In this experiment, the entire jitter rate that has been capture in the experiment does not exceed the minimum jitter. These results are not surprising because of the VoIP environment that has been used only in LAN environment, mean that the voice data don’t have to go through many gateways to reach the destination. However, there is a slight delay in sound to reach the receiver, but not to reach the 150ms. Callers must listen carefully to hear the voice delay. Nevertheless, there are several ways that you can take to prevent VoIP calls from extreme jitter that will be explained after this in the recommendation sub-topic.

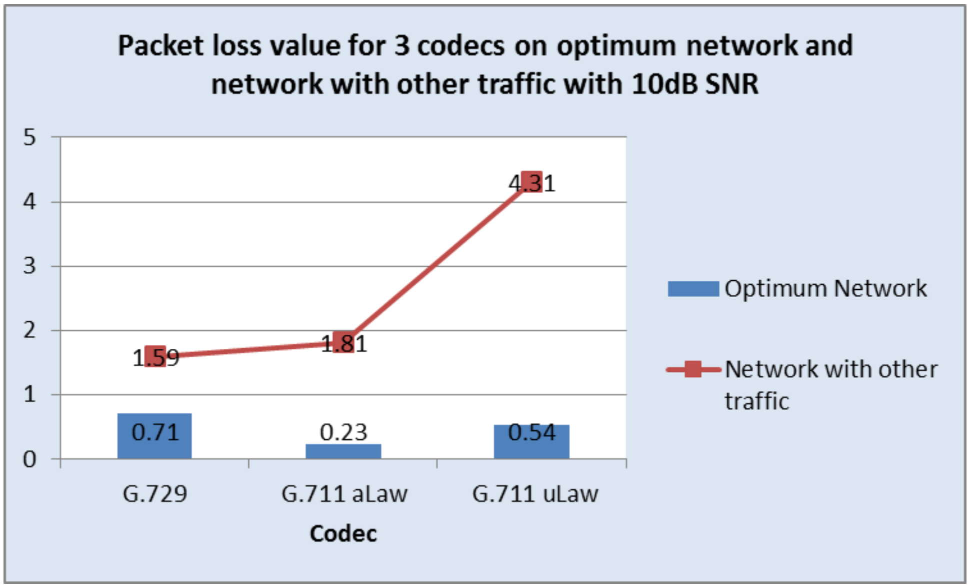


Figure 20: Packet loss value for 3 codecs on optimum network and network with other traffic with 10dB SNR

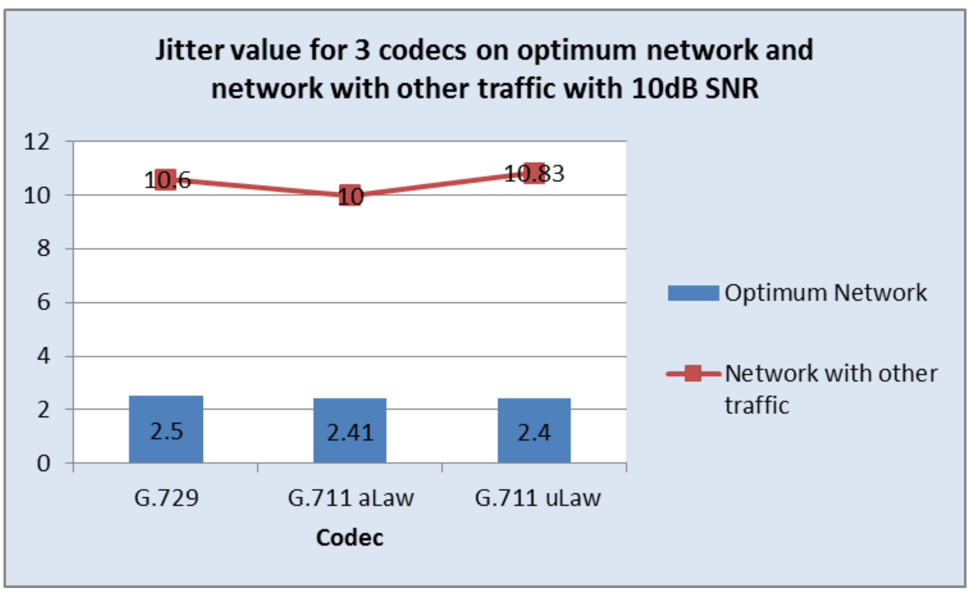


Figure 21: Jitter value for 3 codecs on optimum network and network with other traffic with 10dB SNR

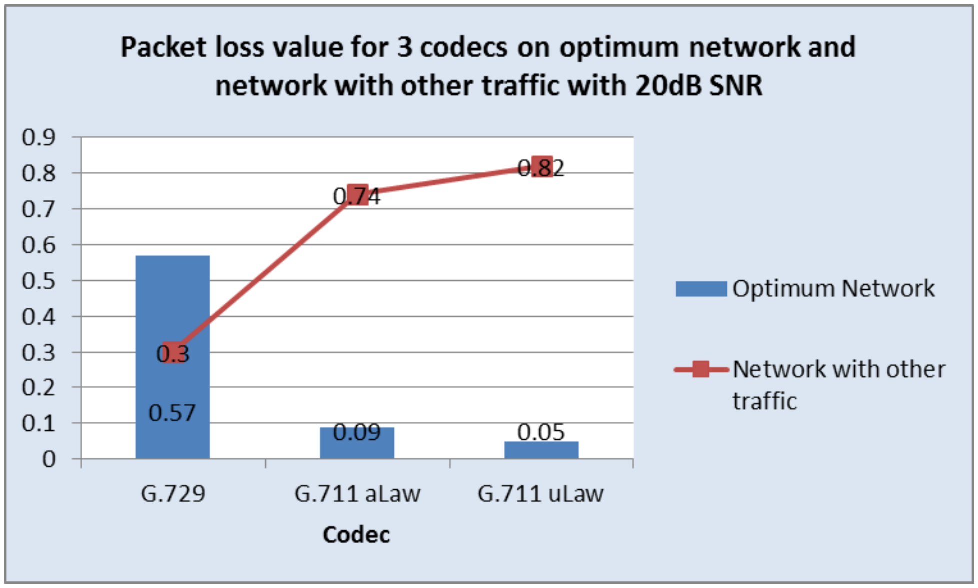


Figure 22: Packet loss value for 3 codecs on optimum network and network with other traffic with 20dB SNR

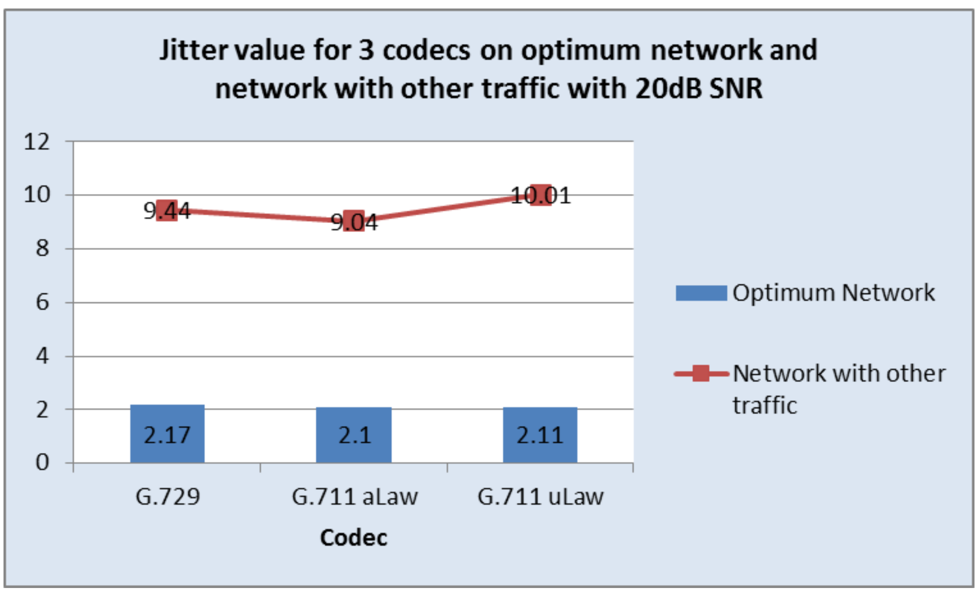


Figure 23: Jitter value for 3 codecs on optimum network and network with other traffic with 20dB SNR

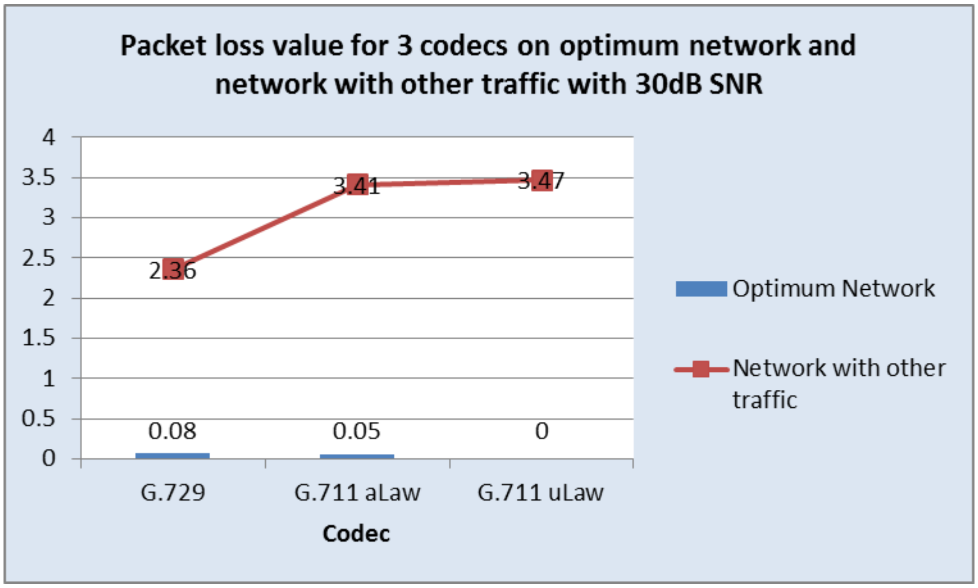


Figure 24: Packet loss value for 3 codecs on optimum network and network with other traffic with 30dB SNR

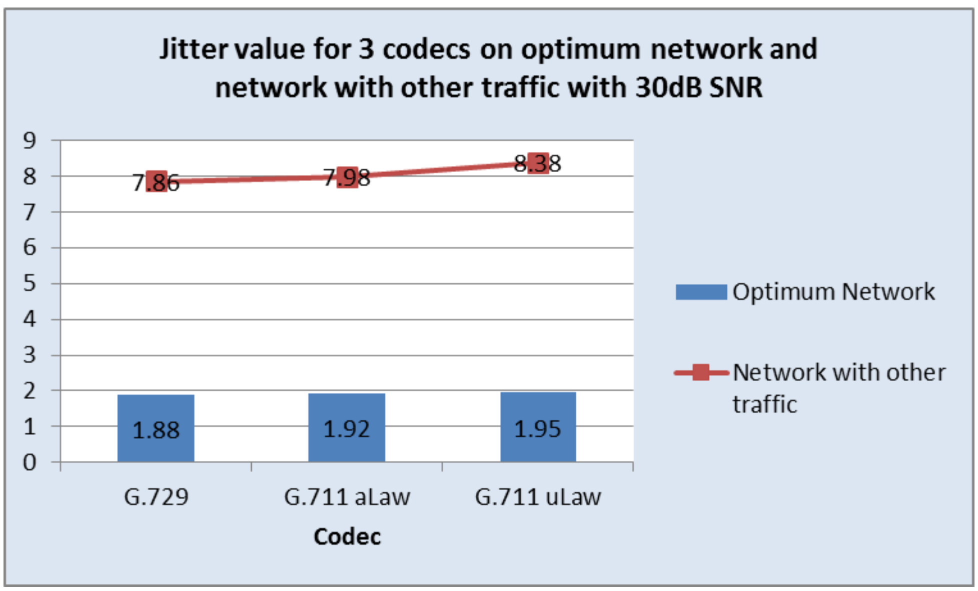


Figure 25: Jitter value for 3 codecs on optimum network and network with other traffic with 30dB SNR

6.1.5 Signal-to-Noise Ratio (SNR) factor

Other than the packet loss, jitter and bandwidth, SNR that present in a network can also impact on the data transmission either the ratio is low or high. According to the Shannon-Hartley Theorem, combination of SNR, bandwidth, and channel are combined to determine the maximum rate at which information can be transmitted over a communications channel of a specified bandwidth in the presence of noise especially on wireless network.

In this experiment, the value of the SNR surely will interrupt the voice data transmission that occurs during VoIP calls are made. If we look at the end result of the experiment, the call quality as measured by MOS and R-Factor is better when the value of the SNR is high. For example, based on the table below, when G.711 aLaw codec is used on the optimum network, the codec can get the best rate of MOS when the SNR of the signal is 30dBwhich is the highest SNR of the experiment. The high of the SNR also can reduce the percentage of packet loss and value of jitter.

G.711 aLaw on network with other traffic			
SNR	**10dB**	**20dB**	**30dB**
MOS	3.97	4.24	4.4
R-Factor	91.27	91.3	92.79
Packet loss	0.23	0.09	0.05
Jitter	2.41	2.1	1.92

Table 41: G.711 aLaw on network with other traffic

Reading of the MOS, R-Factor, packet loss, and jitter are not many visible difference in call quality between the three testbed that using the different SNR. But we can analyze that when the SNR is reached 30dB and higher, the reading of the MOS for a codec can reach the maximum value. Its mean that the transmission of the data is run properly and it’s reduce the packet loss and jitter. For example, in this experiment, G.711 alaw get MOS of 4.4 on a network that has 30dB SNR over the network optimum which is 4.4 is the highest MOS that G.711 aLaw will get (refer to table).

6.2 Conclusion

After running six testbed as planned in design chapter, the decision to meet the objectives of this analysis can be produced in the form of graphs, tables and a brief description. As the final conclusions of this analysis, G.711 aLaw codec is the most stable codec among the three 8kbps codec that has been use in the experiment based on the results of experiments where G.711 aLaw codec have the best call quality (MOS and R-Factor) in four of six conducted testbed. While G.711uLaw and G.729 only recorded the best call quality on one testbed.

Summary result	
Testbed	**Best codec**
Optimum network with respective SNR 10dB	G.711 aLaw
Optimum network with respective SNR 20dB	G.711 aLaw
Optimum network with respective SNR 30dB	G.711 uLaw
Network with other traffic with respective SNR 10dB	G.711 aLaw
Network with other traffic with respective SNR 20dB	G.711 aLaw
Network with other traffic with respective SNR 30dB	G.729

Table 42: Summary result of all experiments

What we can conclude on packet loss and jitter is when the rate of packet loss and jitter increases, the MOS and R-Factor is reduced. SNR that is suitable for VoIP session when using any specific codec is 30dB and above based on the result of the quality of call on each testbed is the best when the SNR is 30dB. On 30dB SNR condition, packet loss and packet jitter almost no occurs because of when the wireless signal in 30dB condition, it does will get 50% signal strength. That is the acceptable signal strength for voice transmission.

In term of the sound quality that we can heard during the call, the sound problem are heard when the jitter is exceed the limit of playout jitter buffer. The situation is like this, Due to network congestion, improper queuing, or configuration errors. The steady stream is interrupted. There is where play out jitter buffer plays the rule. its buffer the continuous voice data at the receiving end and play out with steady stream. When data exceeds the buffer, the problem of the sound can be heard.

Furthermore, we can conclude that when the network has the respective SNR value, we can use the best codec based on the result of this analysis. For example, when on optimum network with respective SNR 10dB, the codec that user can use or enable is G.711 aLaw.

The additional analysis that we will get in this research is the combination of the best codec (G.711 aLaw) and the SNR 30dB and above can come out with the best quality of VoIP call over wireless network as already explain in detail in this analysis. This result of the analysis also can be used as a reference to developer and user of the VoIP service. As we can see, when the new installation of the VoIP environment is about to develop, we can determine and consider the SNR value that will present in the network and what is the best codec that is suitable to use.

6.3 Recommendation

There are many more studies have not been carried out on VoIP quality. Thus, this section describes some of the research proposals that may be made in the future in connection with VoIP call quality. Some suggestions for future research that may be considered are as follows:

1. Research on the codec in other categories of bitrate. For example, other codecs that have sampling rate range 64 kbps, 13kbps and many more.

2. Research on other problem that can affect call quality of VoIP like hardware that being used.

3. Research on other wireless technology such as wireless N.

4. Research to determine the good environment to setup VoIP such as the building floor plan, network design, hardware that is use and other thing that around the VoIP environment.

References

[1] Tamo oft. Monitoring and Troubleshooting VoIP Networks with a Network Analyzer. *http://www.tamos.com/htmlhelp/voip-analysis/mosandr_factor.htm*, Accessed at 31 MAC 2013)

[2] David P. Hole and Fouad A. Tobagi. 2004. Capacity of an IEEE 802.11b Wireless LAN supporting VoIP. *Proc. IEEE Int. Conference on Communication.* 6 pages.

[3] J. H. James, Bing Chen, and Laurie Garrison. Implementing VoIP: A Voice Transmission Performance Progress Report. *IEEE Communications Magazine.* 6 pages.

[4] Voip-Info.org - A reference guide to all things VOIP. *http://www.voip-info.org/wiki/view/ITU+G.729*. Last modification: Thu 07 Apr, 2011. Accessed at 1 April 2013.

[5] What is GNU?. *http://www.gnu.org*. Last modification: 26 Mac 2013. Accessed at 2 April 2013.

[6] Speex: A Free Codec for Free Speech. *www.speex.org.* Accessed at 29 March 2013

[7] What is iLBC. The iLBCfreeware.org project. *http://www.ilbcfreeware.org*. Access at 30 March 2013.

[8] iLBC Freeware. WebRTC.org. *http://www.webrtc.org/ilbc-freeware.* Access at 30 March 2013.

[9] Vladimír Toncar. VoIP Basics: Overview of Audio Codecs. *http://toncar.cz/Tutorials/VoIP/VoIP_Basics_Overview_of_Audio_Codecs.html*. Accessed at 30 March 2013

[10] Jonathan D. Rosenberg. G.729 Error Recovery for Internet Telephony. *http://www.jdrosen.net/e6880/#Introduction*. Accessed at 1 April 2013.

[11] BUR GOODE, SENIOR MEMBER, IEEE. *Voice over Internet Protocol (VoIP).* Proceedings of the IEEE, VOL. 90, NO. 9, September 2002. 23 pages.

[12] Leandro Carvalho, Edjair Mota, Regeane Aguiar, Ana F. Lima, José Neuman de Souza, Anderson Barreto. *An E-Model Implementation for Speech Quality Evaluation in VoIP Systems.* 2005.

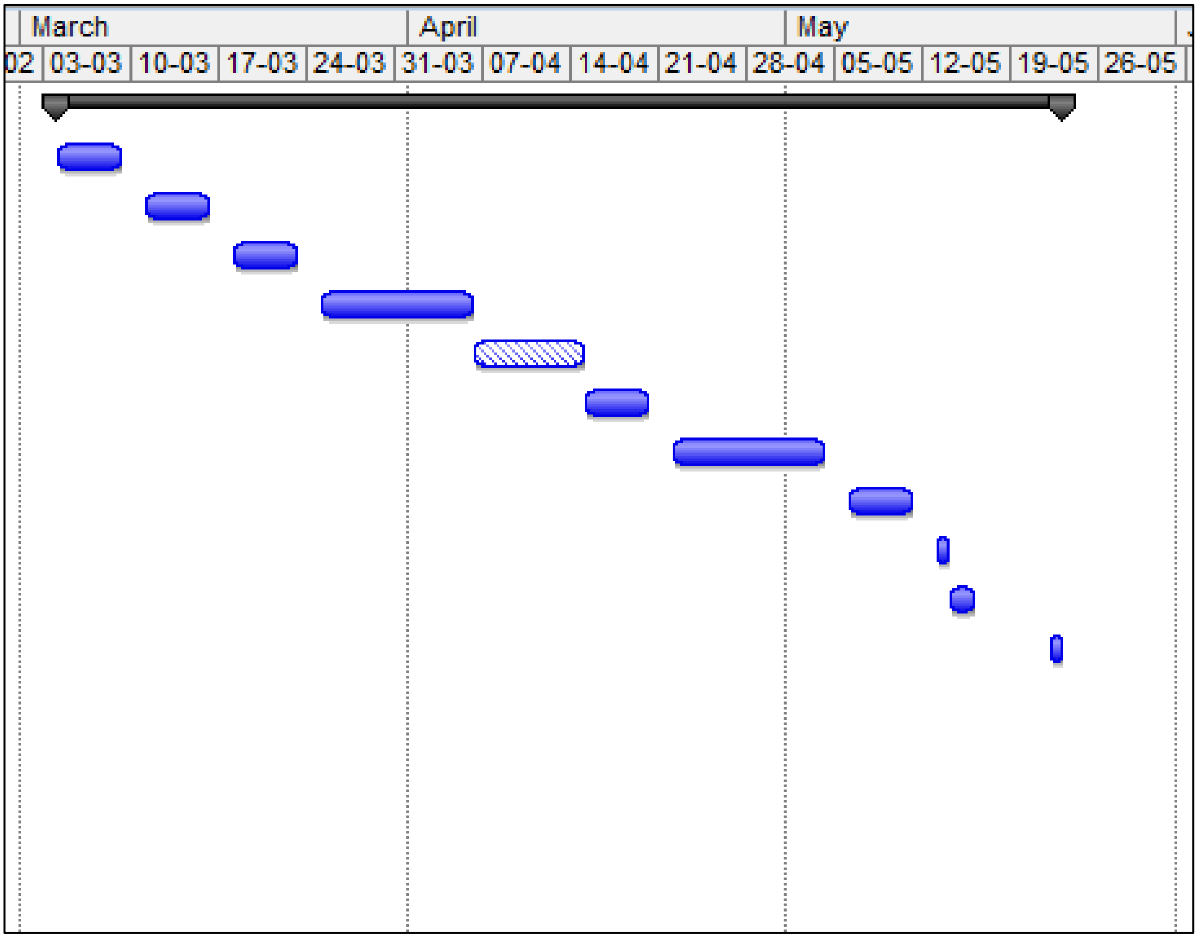
[13] Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms). http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a00800945df.shtml. Last update Feb 02, 2006. Access at 12 May 2013.

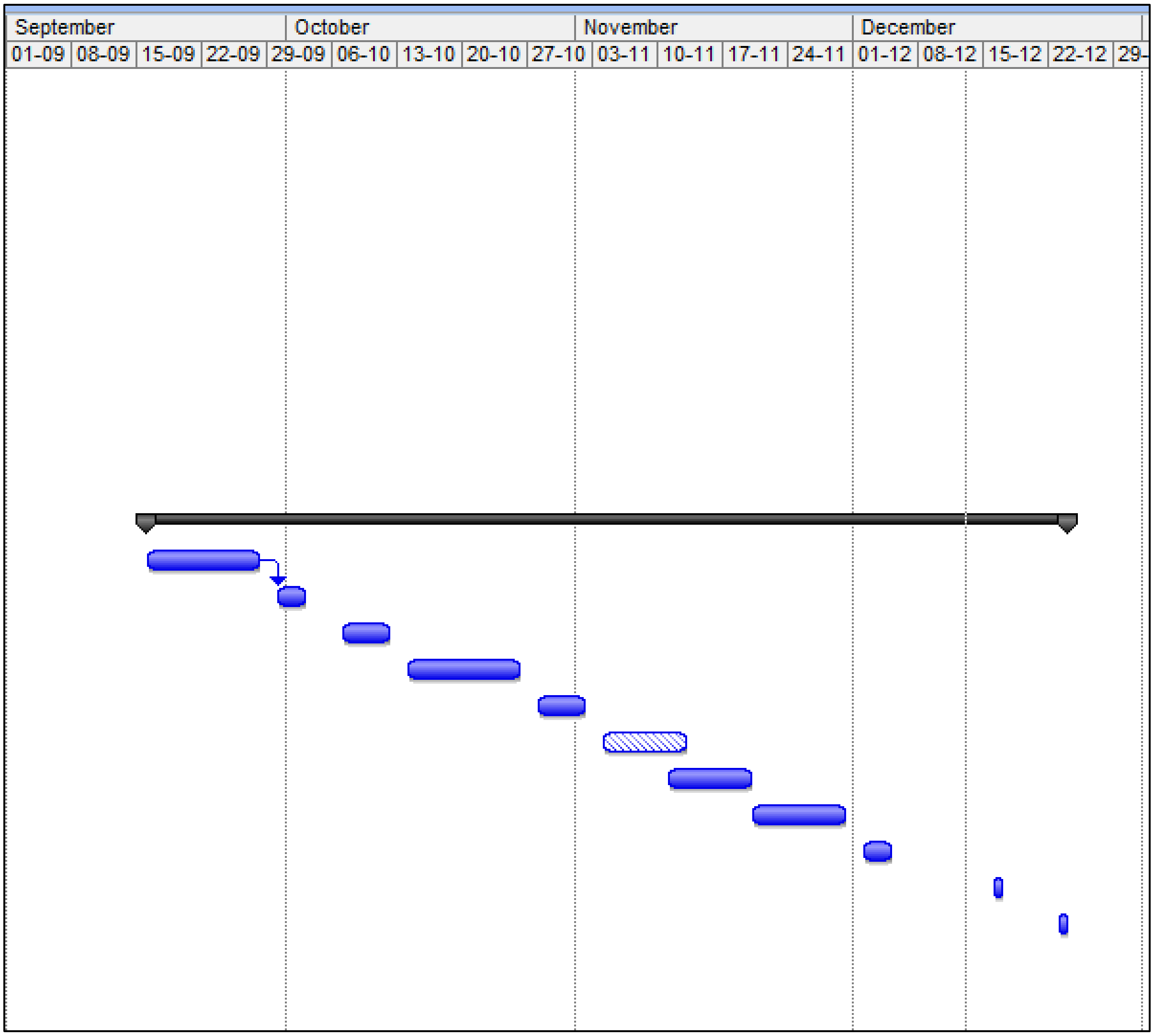
[14] Shannon–Hartley theorem. http://www.princeton.edu/~achaney/tmve/wiki100k/docs/Shannon%E2%80%93Hartley_theorem.html. Last update 15 November 2013. Access at 30 November 2013

APPENDIX

Gantt chart

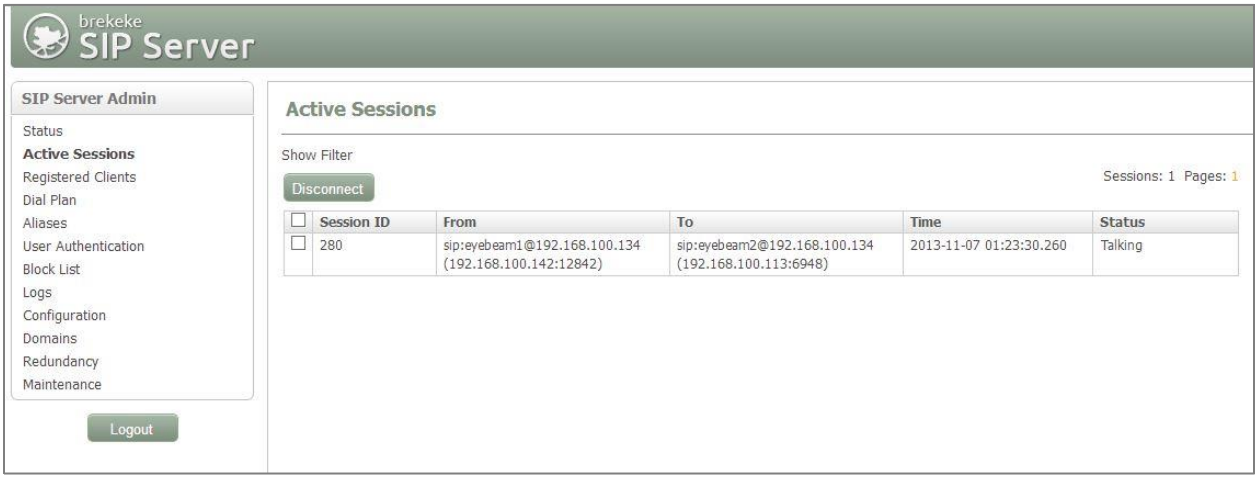
		Task Name	Duration	Start	Finish
1		**Proposal PSM1**	**60 days**	**Mon 04-03-13**	**Wed 22-05-13**
2		Finding title	5 days	Mon 04-03-13	Fri 08-03-13
3		Research the info about title	5 days	Mon 11-03-13	Fri 15-03-13
4		Identify problem statement, objective and scope	5 days	Mon 18-03-13	Fri 22-03-13
5		Chapter 1 and 2 writing	10 days	Mon 25-03-13	Fri 05-04-13
6		***Semester break***	7 days	Sat 06-04-13	Sun 14-04-13
7		Chapter 3 methodology writing	5 days	Mon 15-04-13	Fri 19-04-13
8		Chapter 4 Design and implementation writing	10 days	Mon 22-04-13	Fri 03-05-13
9		Complete the prposal writing	5 days	Mon 06-05-13	Fri 10-05-13
10		Submit full report to supervisor	1 day	Mon 13-05-13	Mon 13-05-13
11		Create the presentation slides	2 days	Tue 14-05-13	Wed 15-05-13
12		Proposal presentation	1 day	Wed 22-05-13	Wed 22-05-13
13		**PSM2**	**72 days?**	**Mon 16-09-13**	**Mon 23-12-13**
14		Identify suitablity of hardware and software	10 days	Mon 16-09-13	Fri 27-09-13
15		Setup the experiment environment	3 days	Mon 30-09-13	Wed 02-10-13
16		Pre testing	5 days	Mon 07-10-13	Fri 11-10-13
17		Start the experiment and raw data collection	10 days	Mon 14-10-13	Fri 25-10-13
18		Tabular and graph the data	5 days	Mon 28-10-13	Fri 01-11-13
19		***Semester break***	7 days	Mon 04-11-13	Tue 12-11-13
20		Analysis	7 days	Mon 11-11-13	Tue 19-11-13
21		Report writing	8 days	Wed 20-11-13	Fri 29-11-13
22		Create and print the poster	3 days	Mon 02-12-13	Wed 04-12-13
23		Final presentation	1 day	Mon 16-12-13	Mon 16-12-13
24		Hard cover printing	1 day?	Mon 23-12-13	Mon 23-12-13

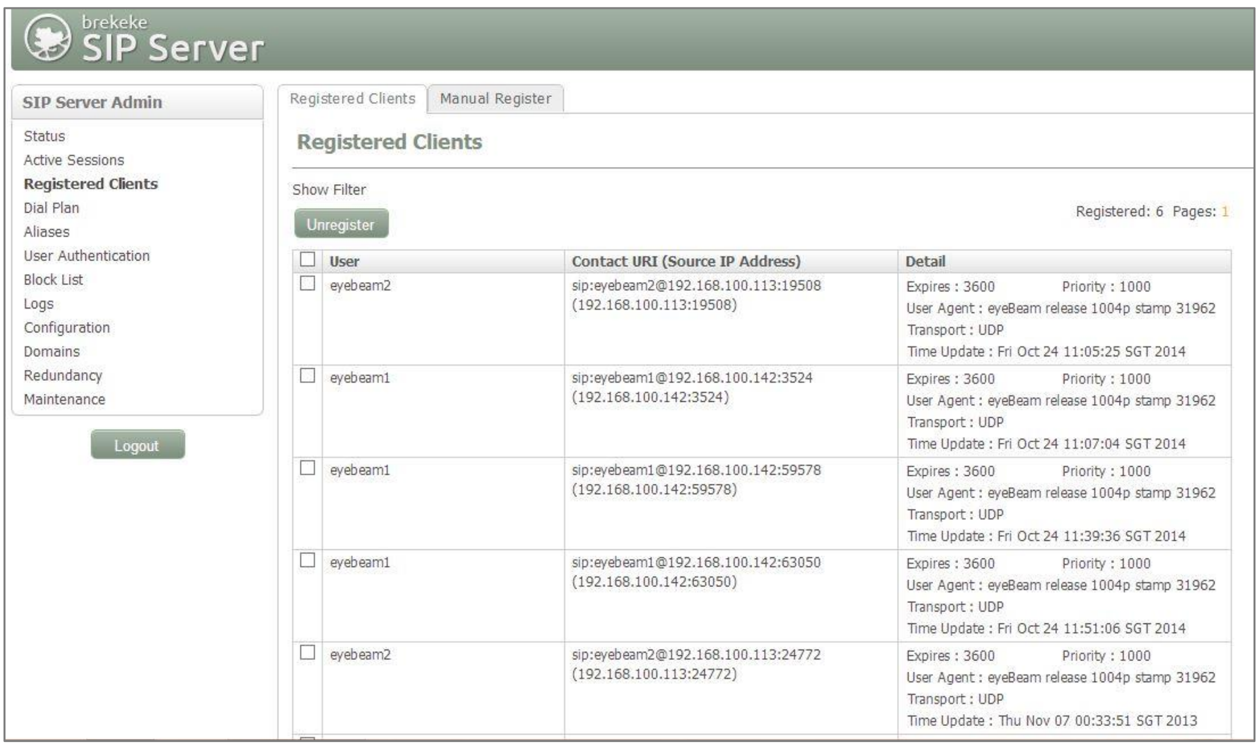

March
April
May
02
03-03
10-03
17-03
24-03
31-03
07-04
14-04
21-04
28-04
05-05
12-05
19-05
26-05



September
October
November
December
01-09
08-09
15-09
22-09
29-09
06-10
13-10
20-10
27-10
03-11
10-11
17-11
24-11
01-12
08-12
15-12
22-12
29-

SIP SERVER - Brekeke SIP server






brekeke
SIP Server
SIP Server Admin
Status
Active Sessions
Registered Clients
Dial Plan
Aliases
User Authentication
Block List
Logs
Configuration
Domains
Redundancy
Maintenance
Logout
Start/Shutdown
Server Status
Start/Shutdown
Restart
Shutdown
Status Summary
Status
Active
Interface
192.168.100.134, 192.168.134.1, 192.168.23.1
Local Port
5060
Active Sessions
0
Multiple Domains
No


brekeke
SIP Server
SIP Server Admin
Status
Active Sessions
Registered Clients
Dial Plan
Aliases
User Authentication
Block List
Logs
Configuration
Domains
Redundancy
Maintenance
Logout
User Authentication
Import/Export
User Authentication
Show Filter
New User
Delete
Results: 6 Pages: 1
User
Name
Email Address
Description
ekiga1
ekiga2
eyebeam1
eyebeam2
xlite1
xlite2

NETWORK MONITORING TOOL – Commview 6.0

Error Description
481 Call Leg/Transaction Does Not Exist
407 Proxy Authentication Required
407 Proxy Authentication Required
407 Proxy Authentication Required
407 Proxy Authentication Required
481 Call Leg/Transaction Does Not Exist
407 Proxy Authentication Required
407 Proxy Authentication Required
481 Call Leg/Transaction Does Not Exist
407 Proxy Authentication Required
407 Proxy Authentication Required
407 Proxy Authentication Required
481 Call Leg/Transaction Does Not Exist

Packet No	Time	Time Interval	Operation	Request/Response	CSeq	Content
1	23:40:45.220571	0.000000	INVITE	INVITE sip:eyebeam2@192.168.100.134	1 INVITE	SDP
2	23:40:45.223133	0.002562		100 Trying	1 INVITE	(none)
3	23:40:45.448341	0.225208		180 Ringing	1 INVITE	(none)
4	23:40:48.468043	3.019702		200 OK	1 INVITE	SDP
5	23:40:49.109449	0.641406		200 OK	1 INVITE	SDP
6	23:40:50.092500	0.983051		200 OK	1 INVITE	SDP
7	23:40:52.110426	2.017926		200 OK	1 INVITE	SDP
8	23:40:52.202074	0.091648		ACK sip:eyebeam2@192.168.100.134:...	1 ACK	(none)
9	23:40:52.669272	0.467198		ACK sip:eyebeam2@192.168.100.134:...	1 ACK	(none)
10	23:40:53.124808	0.455536		ACK sip:eyebeam2@192.168.100.134:...	1 ACK	(none)
11	23:40:53.161703	0.036895		ACK sip:eyebeam2@192.168.100.134:...	1 ACK	(none)

Registrations

Last Activity	Protocol	User IP	User	Domain	Location	Registrar IP	Status	Time To Live	Expires	Last Request/Response
11/10/2013 5:04:43 PM	SIP	192.168.100.134	eyebeam1@192.168.100.137	192.168.100.137	eyebeam1@192.168.100.134:35862;rinstance=c4b60933524ddabe	192.168.100.137	Registered	1:00:00	11/10/2013 6:04:43 PM	200 OK

Pac...	Time	Time Int...	Type	SSRC	Seq...	RTP Times...	Payload Name	Jitter (ms)	Marker	Content
1	23:31:34.877630	0.000000	RTP	640405597	7985	3026600	ITU-T G.711 P...	0.00	Set	
2	23:31:34.887642	0.010012	RTP	640405597	7986	3026760	ITU-T G.711 P...	0.62		
3	23:31:34.907908	0.020266	RTP	640405597	7987	3026920	ITU-T G.711 P...	0.60		
4	23:31:34.928627	0.020719	RTP	640405597	7988	3027080	ITU-T G.711 P...	0.61		
5	23:31:34.945668	0.017041	RTP	640405597	7989	3027240	ITU-T G.711 P...	0.76		
6	23:31:34.966646	0.020978	RTP	640405597	7990	3027400	ITU-T G.711 P...	0.77		
7	23:31:34.991657	0.025011	RTP	640405597	7991	3027560	ITU-T G.711 P...	1.03		
8	23:31:35.006648	0.014991	RTP	640405597	7992	3027720	ITU-T G.711 P...	1.28		
9	23:31:35.030652	0.024004	RTP	640405597	7993	3027880	ITU-T G.711 P...	1.45		
10	23:31:35.046645	0.015993	RTP	640405597	7994	3028040	ITU-T G.711 P...	1.61		
11	23:31:35.066855	0.020210	RTP	640405597	7995	3028200	ITU-T G.711 P...	1.53		
12	23:31:35.085371	0.018516	RTP	640405597	7996	3028360	ITU-T G.711 P...	1.52		
13	23:31:35.105628	0.020257	RTP	640405597	7997	3028520	ITU-T G.711 P...	1.44		
14	23:31:35.125709	0.020081	RTP	640405597	7998	3028680	ITU-T G.711 P...	1.36		
15	23:31:35.145337	0.019628	RTP	640405597	7999	3028840	ITU-T G.711 P...	1.30		
16	23:31:35.167238	0.021901	RTP	640405597	8000	3029000	ITU-T G.711 P...	1.33		
17	23:31:35.194690	0.027452	RTP	640405597	8001	3029160	ITU-T G.711 P...	1.72		
18	23:31:35.206820	0.012130	RTP	640405597	8002	3029320	ITU-T G.711 P...	2.10		
19	23:31:35.226620	0.019800	RTP	640405597	8003	3029480	ITU-T G.711 P...	1.98		
20	23:31:35.247394	0.020774	RTP	640405597	8004	3029640	ITU-T G.711 P...	1.91		
21	23:31:35.265324	0.017930	RTP	640405597	8005	3029800	ITU-T G.711 P...	1.92		
22	23:31:35.286658	0.021334	RTP	640405597	8006	3029960	ITU-T G.711 P...	1.88		

SIP Sessions

Src IP	Dest IP	Start Time	End Time	Duration	Status	Src Display Name	Src SIP Address	Dest Display Name	Dest SIP Address	MOS Score
192.168.100.142	192.168.100.134	11/7/2013 1:11:54 AM	11/7/2013 1:11:59 AM	0:00:04.9	Completed by timeout	eyebeam1	eyebeam1@192.168.100.134	eyebeam2	eyebeam2@192.168.100.134	?
192.168.100.142	192.168.100.134	11/7/2013 1:06:04 AM	11/7/2013 1:06:09 AM	0:00:05.2	Completed by timeout	eyebeam1	eyebeam1@192.168.100.134	eyebeam2	eyebeam2@192.168.100.134	?
192.168.100.142	192.168.100.134	11/7/2013 1:05:38 AM	11/7/2013 1:05:53 AM	0:00:14.2	Completed by timeout	eyebeam1	eyebeam1@192.168.100.134	eyebeam2	eyebeam2@192.168.100.134	?
192.168.100.142	192.168.100.134	11/7/2013 12:57:48 AM	11/7/2013 12:57:52 AM	0:00:04.1	Completed by timeout	eyebeam1	eyebeam1@192.168.100.134	eyebeam2	eyebeam2@192.168.100.134	?
192.168.100.142	192.168.100.134	11/7/2013 12:53:38 AM	11/7/2013 12:54:09 AM	0:00:30.3	Completed	eyebeam1	eyebeam1@192.168.100.134	eyebeam2	eyebeam2@192.168.100.134	?
192.168.100.134	192.168.100.142	11/7/2013 12:49:04 AM	11/7/2013 12:49:10 AM	0:00:05.6	Completed by timeout	eyebeam2	eyebeam2@192.168.100.134	eyebeam1	eyebeam1@192.168.100.134	?
192.168.100.142	192.168.100.134	11/7/2013 12:47:44 AM	11/7/2013 12:48:28 AM	0:00:43.9	Completed	eyebeam1	eyebeam1@192.168.100.134	eyebeam2	eyebeam2@192.168.100.134	?
192.168.100.142	192.168.100.134	11/7/2013 12:40:26 AM	11/7/2013 12:40:31 AM	0:00:04.8	Completed by timeout	eyebeam1	eyebeam1@192.168.100.134	eyebeam2	eyebeam2@192.168.100.134	?
192.168.100.142	192.168.100.134	11/7/2013 12:40:03 AM	11/7/2013 12:40:04 AM	0:00:00.0	Failed	eyebeam1	eyebeam1@192.168.100.134	eybeam2	eybeam2@192.168.100.134	?
192.168.100.142	192.168.100.134	11/7/2013 12:34:50 AM	11/7/2013 12:34:55 AM	0:00:05.0	Completed by timeout	eyebeam1	eyebeam1@192.168.100.134	eyebeam2	eyebeam2@192.168.100.134	?

SNR measurement – DD-WRT

Wireless

Clients

MAC Address	Interface	Uptime	TX Rate	RX Rate	Signal	Noise	SNR	Signal Quality
xx:xx:xx:xx:6B:90	eth1	N/A	N/A	N/A	-71	-81	10	28%
xx:xx:xx:xx:93:C2	eth1	N/A	N/A	N/A	-71	-81	10	28%

Wireless

Clients

MAC Address	Interface	Uptime	TX Rate	RX Rate	Signal	Noise	SNR	Signal Quality
xx:xx:xx:xx:30:48	eth1	N/A	N/A	N/A	-62	-82	20	39%
xx:xx:xx:xx:6B:90	eth1	N/A	N/A	N/A	-62	-82	20	39%

Wireless

Clients

MAC Address	Interface	Uptime	TX Rate	RX Rate	Signal	Noise	SNR	Signal Quality
xx:xx:xx:xx:6B:90	eth1	N/A	N/A	N/A	-53	-83	30	50%
xx:xx:xx:xx:80:AF	eth1	N/A	N/A	N/A	-53	-83	30	50%