PERFORMANCE ANALYSIS OF SPEECH CODEC (GSM, ILBC, SPEEX) FOR VOIP OVER WIRELESS LOCAL AREA NETWORK (WLAN) WITH RESPECTIVE SIGNAL TO NOISE-RATIO (SNR)

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A report submitted in partial fulfillment of the requirements for the award of the degree of Bachelor of Computer Science (Computer Systems & Networking)

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DECEMBER, 2013

DECLARATION

I hereby declare that the work in this thesis entitled *"Performance Analysis of Speech Codec (GSM, ILBC, and SPEEX) for VOIP over Wireless Local Area Network (WLAN)* with Respective Signal to Noise Ratio (SNR) is my own research except for quotations and summaries which have been duly acknowledged.

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ACKNOWLEDGMENTS

First and foremost praise to Almighty Allah for all his blessings for giving me patience and good health throughout the duration of this research project. I would like to dedicate my highest gratitude for those who have involved directly or indirectly during this research. They have contributed towards my understanding and thoughts.

In particular, I wish to express my sincere appreciation to my thesis supervisor, Mr.Imran Edreeziq, for encouragement guidance, ideas, suggestion, critics and motivation. It is been an interesting and exciting learning period for me. There are lots of things that I learned during this research period.

My sincere appreciation also extend to my entire friend from BCN, who always been supporting towards me. Thank you for all the motivation words, ideas and critics during my research. Their point of view and tips are useful indeed. Unfortunately, it is not possible to list all of them in this limited space.

Last but not least, I am grateful to all my family members for their support. A million thank you to them for their moral support. Thanks for all the encouragement, motivation and prayers for all this time.

ABSTRACT

Voice over Internet Protocol (VoIP) is one of the fastest growing Internet applications. It is a viable alternative to the traditional telephony systems due to its high resource utilization and cost efficiency. Meanwhile, Wireless Local Area Networks (WLANs) have become a ubiquitous networking technology that has been deployed around the world. In this research, 3 types of speech codec (GSM, ILBC, SPEEX) in the same sampling rate of (11-13) kbps are chosen to be test in predefined network environments to measure the performance base on R-Factor, MOS, and packet jitter and packet loss. Thus, a codec is expected to provide good quality of VoIP. And in some circumstances, bandwidth may be a crucial factor between the success and failure of an application. With the likes of Internet applications such as video and audio streaming, video and audio downloading, these has contributed to the increase of Internet users and which directly affect the performance of speech codec when tested with other traffic in the network because it were using the same network bandwidth. All three mention speech codec will be test based on these criteria. The speech quality of three speech codec namely GSM (13kbps) , ILBC (13.33 kbps) , and Speex (11kbps) under various network performance based on pre-determined SNR values will be evaluated and compare against. Several tests are constructed to prove that it meets the interest of investigation. The experimental procedure of this dissertation can be summarized to 2 main experiments which need to be repeated for each speech codec and for each predefined SNR value. Both types of network on two way communication testing; 1) Optimum Network, and 2) Network with others traffic, need to be repeated for all three speech codec GSM, ILBC, Speex with each respective SNR values; 10 dB, 20 dB, and 30 dB. All test criteria will be carry out on real devices simulation. At the end, the performance measurement of VOIP on Quality of Services; such as MOS, R-Factor, packet loss and packet jitter will be observe to determine the best speech codec in each scenario.

ABSTRAK

Suara melalui Protokol Internet (VoIP) adalah salah satu aplikasi internet yang paling pesat berkembang. Ia adalah alternatif yang berdaya saing kepada sistem telefoni tradisional kerana penggunaan sumber yang tinggi dan kekurangan kos. Sementara itu, Rangkaian Kawasan Setempat Tanpa Wayar (WLAN) telah menjadi satu teknologi rangkaian yang sentiasa ada yang telah digunakan di seluruh dunia. Dalam kajian ini, 4 jenis ucapan codec (GSM, ILBC, Speex) dalam kadar pensampelan yang sama (11-13) kbps dipilih untuk diuji dalam persekitaran rangkaian yang telah ditetapkan untuk mengukur asas prestasi ke atas R- Factor, MOS, dan paket tanggoh dan kehilangan paket . Oleh itu, codec yang dijangka untuk memberi kualiti yang baik dalam protocol VoIP. Dalam keadaan tertentu, jalur lebar boleh menjadi faktor penting antara kejayaan dan kegagalan VOIP. Dengan aplikasi Internet seperti video dan streaming audio, video dan muat turun audio, perkara ini telah menyumbang kepada peningkatan penggunaan internet dan secara tidak langsung memberi kesan kepada prestasi ucapan codec apabila diuji dengan rangkaian lain kerana ia telah menggunakan jalur lebar yang sama pada rangkaian tersebut. Ketiga-tiga sebutan ucapan codec akan menjadi ujian berdasarkan kriteria ini. Kualiti tiga ucapan codec iaitu GSM (13kbps), ILBC (13.33 kbps), dan Speex (11kbps) di bawah pelbagai prestasi rangkaian berasaskan nilai SNR yang ditetapkan akan dinilai dan dibandingkan. Beberapa ujian yang dibina untuk membuktikan bahawa ia memenuhi kepentingan penyiasatan. Prosedur eksperimen disertasi ini boleh diringkaskan kepada 2 ujikaji utama yang perlu diulangi untuk setiap codec ucapan dan bagi setiap nilai SNR yang telah ditetapkan. Kedua-dua jenis rangkaian pad ujian dua hala ; 1) Rangkaian optimum , dan 2) Rangkaian dengan lalu lintas luar, yang perlu diulangi untuk ketiga-tiga ucapan codec GSM, ILBC, Speex dengan setiap nilai SNR masing-masing; 10 dB, 20 dB, dan 30 dB. Semua kriteria ujian akan menjalankan simulasi pada keadaan sebenar. Pada kesimpulannya, pengukuran prestasi Kualiti VOIP; seperti MOS, R- Factor, kehilangan paket dan tanggoh paket akan diperhatikan untuk menentukan ucapan codec yang terbaik dalam setiap senario.

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Chapter 1

INTRODUCTION

1.0 Introduction

VoIP is a growing technology that enables the transport of voice over data networks such as the public Internet. Voice over IP (VoIP), also known as Internet telephony, is a form of voice communication that uses data networks to transmit audio signals. When using VoIP the voice is appropriately encoded at one end of the communication channel, and sent as packets through the data network. After the data arrives at the receiving end, it is decoded and transformed back into a voice signal. VoIP became a viable alternative to the public switched telephone networks (PSTNs). It uses a number of protocols which ensure that voice communication is appropriately established between parties, and that voice is transmitted with a quality close to that we are accustomed to in the PSTN.

VoIP uses signaling protocols such as the Session Initiation Protocol (SIP) and H.323. Concurrently, in the access technology used for IP-based networks, a rapid and wide deployment of wireless local area networks (WLAN) is taking place in most corporate buildings, small offices and home offices (SOHO) as well as public spaces such as commercial malls and airport.

WLAN technology is based on the IEEE802.11 network access standards. The use of WLAN enables users to have instant access to the Internet services regardless of their location in the network. In addition, connectivity is continuously offered to the users while roaming from one place to another. As the user moves from one radio coverage to another, the mobile device transfers its control between the Access Points (AP). This transfer process is called handover or handoff. The performance of certain applications can be impacted during a handover.

VoIP is a service that has stringent Quality of Service (QoS) requirements as to the timeliness and the quality of the voice required for users in WLAN-based access networks. Several studies have shown that mobility handover can have an impact on the quality of the voice due to the delays caused by the various operations executed during the handover.

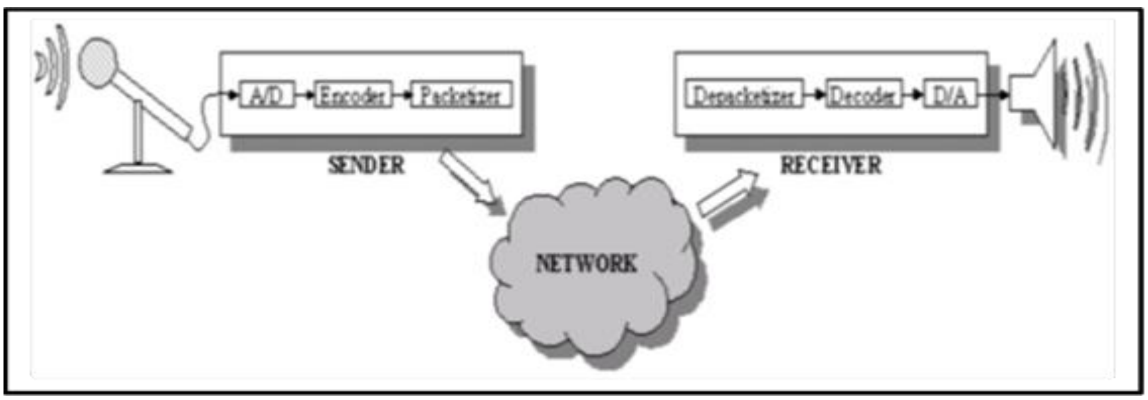

Figure 1: End-to-end data path for VoIP communication [4]

Figure 1 shows the end-to-end path as needed for VoIP communication (a similar path exists in the opposite sense for a bi-directional connection). An audio input device, such as a microphone, is required at the sending end. The audio signal is transformed into digital form by an analog-to-digital converter. Due to the packet-switched nature of computer networks, voice data has to be packetized and encoded prior to being transmitted. Encoding (as well as decoding) is done by codecs that transform sampled voice data into a specific network-level representation and back. Most of the codecs are defined by standards of the International Telecommunication Union, the Telecommunication division (ITU-T). Each of them has different properties regarding the amount of bandwidth it requires but also the perceived quality of the encoded speech signal.

After binary information is encoded and packetized at the sender end, packets encapsulating voice data can be transmitted on the network. Voice packets interact in the network with other application packets and are routed through shared connections to their destination. At the receiver end they are decoded. Decoding may include other steps as well; the most typical being is packet jitter. Other examples are error correction and packet loss concealment. The flow of digital data is then converted to analogue form again and played at an output device, usually a speaker.

Voice over IP (VoIP) involves digitization of voice streams and transmitting the digital voice as packets over conventional IP-based packet networks like the Internet, Local Area Network (LAN) or wireless LAN (WLAN).The goal of VoIP is to provide voice transmission over those networks. Although the quality of VoIP does not yet match the quality of a circuit-switched telephone network, there is an abundance of activity in developing protocols and speech encoders for the implementation of the high quality voice service. In WLAN, as VoIP technology is still in the early stages of commercial deployment, it is necessary to examine if VoIP over WLAN can provide a Quality of Service (QoS) comparable to that of the existing PSTN and cellular networks. So, it is essential to determine the number of simultaneous users a WLAN can support simultaneously without significantly degrading the QoS and also analyze the delay, jitter and packet loss of VoIP over WLAN. The QoS on VoIP network partly depends on the types of voice codec used. These codecs differ in their coding rate (bps), frame rate (frames/s), algorithmic latency that will influence the speech quality or Mean Opinion Score (MOS) in a VoIP network.

1.1 Problem Statement

The voice signal must be encoded (and compressed) in order to be sent over the packet network. Encoding is done via speech codec. Each codec has different characteristics concerning the data rate it uses (and implicitly the compression level) and also the associated user-perceived quality).

Many types of speech codec were available from minimum (2kbps) up to maximum (64kbps) bit rate requirement. Some VoIP clients will offer the specific supported codecs to user, and user can choose the quality of its own VoIP speech sessions in basis of good quality and bad quality.

Thus, a codec is expected to provide good quality of VoIP even after compression. And in some circumstances, bandwidth may be a crucial factor between the success and failure of an application. With the likes of Internet applications such as video and audio streaming, video and audio downloading, these has contributed to the increase of Internet users and which directly affect the performance of speech codec when tested with other traffic in the network because it were using the same network bandwidth. Furthermore, with the given fluctuation of network bandwidth, it will affect the quality of the VoIP session.

Moreover, not many researches were done on specific group of speech codec based on the sampling requirement. Speech codec has its own bandwidth requirement. For this research study, a specific group of speech codecs (GSM, ILBC, Speex) with bit rate, (11-13 kbps) are selected to determine the best and suitable voice codecs for different type of ideal network condition. So, among this three codec, user cannot determine which one of the codec is the best for VoIP session.

1.2 Objective

This research was conducted to meet three objectives. The objectives of the research are:

I. To simulate the group of speech codec (GSM, ILBC, Speex) for VOIP on predefined wireless MESH network.

II. To analyze the group of speech codec (GSM, ILBC, Speex) performances in terms of Packet Jitter (ms), Packet Loss (%), Mean Opinion Score (MOS) and Relative Factor (R-Factor) based on the simulation.

III. To suggest the best speech codec for the predefined wireless mesh network.

1.3. Scope

In this study we will focus on analyzing the performance of speech codec on the predefined wireless mesh network based on several limitations:

I. The simulation is using a group of speech codec within (11-13 kbps) which called (GSM, ILBC, Speex) of VOIP to be analyzed.

II. Tested using SIP server architecture environment only.

III. IEEE 802.11n as the wireless network standard connection will be used as a medium during the simulation. One wireless access points (AP) will be used to establish the connection.

IV. Equipment: Laptop, Access Point (AP), SIP server, VoIP client and real test bit simulation.

V. The end of the simulation, the performance measurement based on QOS Parameter which is Packet Jitter, Packet Lost, Mean Opinion Score (MOS) and Relative Factor (R-Factor) can be generated to see the result of the experiment.

VI. Ten readings of every each QOS Parameter value will be taken for both environment to ensure consistency in reading and data for each experiment analysis.

1.4 Thesis Organizations

The research consists of five chapters:

Chapter 1 provides 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 voice codec (GSM, ILBC, Speex) of VOIP. Besides that, it also explains more on how to measure the performance of the group of speech codec. The literature review is organized in a way that readers can understand.

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 understand 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 and explains most of the configurations of hardware and software involved in the research. The detail of test results will be included in this chapter.

Chapter 2

LITERATURE REVIEW

2.0 SIP (Session Initial Protocol)

SIP is an application-layer protocol that was initially specified by the IETF Multiparty Multimedia Session Control Working Group (MMUSIC WG) in 1999 and updated by the SIP WG in 2002. SIP, which is delineated in RFC 3261 [7], is used for creating, modifying and terminating sessions with one or more participants, and was designed to be independent of the underlying transport protocol. As in H.323, features in SIP are also classified into three similar categories, namely local features, network-based features such as authorization and supplementary services. The main functions of this signaling protocol are: (i) location of resources/parties; (ii) invitation to service sessions; and (iii) negotiation of service parameters. For conveying information about the media content of the session SIP relies on the Session Description Protocol (SDP).SIP is similar to HTTP (Hyper Text Transfer Protocol) and shares some of its design principles. In particular, it adopts client/server (request/response) architecture in which requests are generated by the client and sent to the server. The server then processes the requests and sends a response to the client. Like HTTP, SIP is based on text-based messages which are either requests or responses exchanged between the clients and the servers. The most important types of requests are the INVITE request that is used to invite a user to a call, the ACK that sends the caller to the caller to simply acknowledge the receipt of the latter’s response, and the BYE request that is used to terminate the connection between two users in a session. In addition to these types, three other kinds of requests can be identified, that is, the CANCEL, the OPTIONS and the REGISTER requests. The CANCEL request is used to countermand any pending searching for a user; however, it does not tear down an ongoing call. The OPTIONS request just queries the capabilities of servers. Finally, the REGISTER request is employed to register a user with a SIP server.

2.1 SIP network architecture

A simple paradigm of SIP architecture is illustrated in Figure 2. The main components of SIP are the user agents and networks servers. User agents are the SIP's endpoints that make and receive calls. A user agent can function in two modes: either as a user agent client (UAC) that initiates SIP requests or as a user agent server (UAS) that receives requests and responds on behalf of the user. In practice, a SIP endpoint (for instance, an IP phone) can act as both a UAC and a UAS. However, it functions only as one or the other per transaction depending on whether or not it initiated the request.

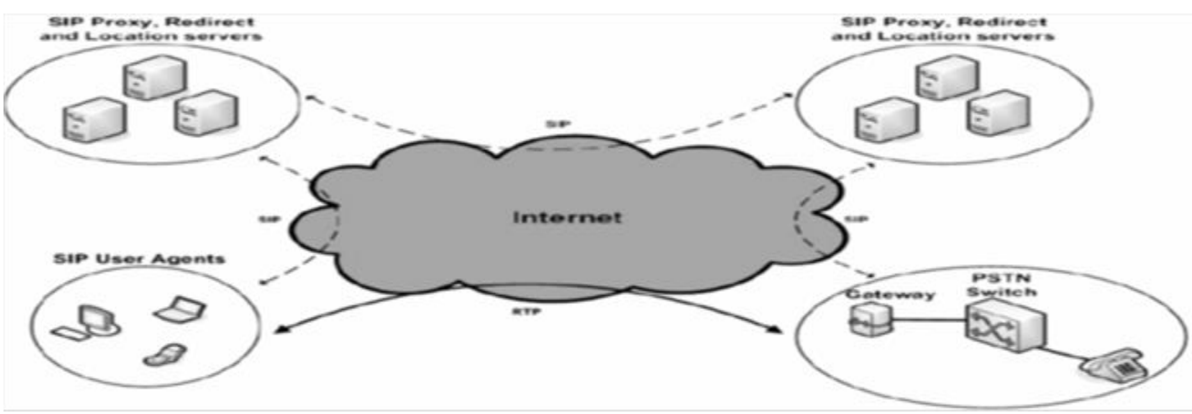

Figure 2: SIP network architecture [7]

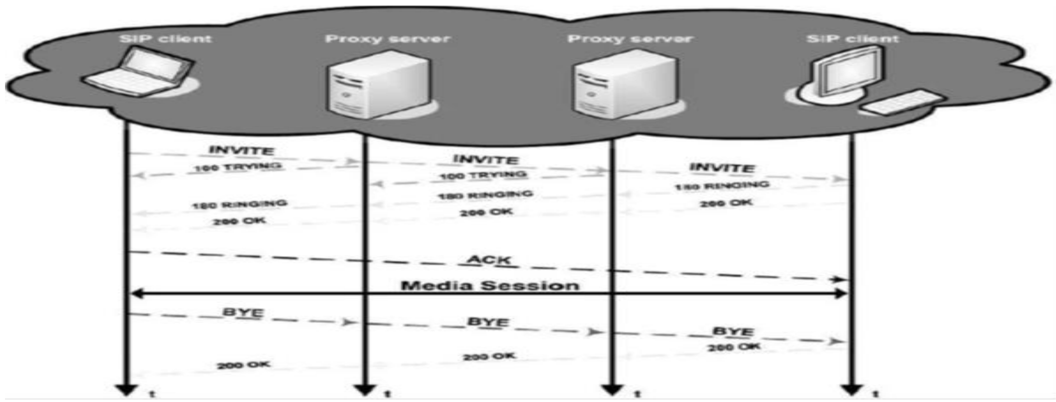

Figure 3: Call setup and tear-down in a SIP architecture. [7]

As far as network servers are concerned, there exist four different types of them in a network: proxy servers redirect servers, location servers and registrar servers. A proxy server receives the requests generated by user agents and decides to which server a request should be forwarded. A request will usually traverse many servers before reaching its destination. The purpose of a redirect server is different from that of a proxy server. A redirect server does not forward requests. Rather it notifies the calling party of the location of the caller. To do so, it contacts a location server that keeps information about the called party’s possible location. As for the purpose of registrar servers, they accept REGISTER requests from user agents and are usually co-located with either proxy servers or redirect servers. Finally, as in the case of the H.323 architecture, a gateway can also be employed to bridge SIP endpoints with other types of terminals.

2.2 VOIP Speech Codec

Since the early days of networking bandwidth has been considered a resource at a premium. Therefore, significant efforts have been drawn towards the minimization of the amount of bandwidth required by specific services in order for the network to be able to serve a greater number of users. In this context, compressing voice signals while keeping the quality perceived by users at acceptable levels represents a daunting challenge. This section is devoted to the methods that either are currently in use or have been proposed for the compression of audio signals, which are referred to as voice codecs. Voice codecs are the algorithms that enable the system to carry analog voice over digital lines. There are several codecs, varying in complexity, bandwidth needed and voice quality. The more bandwidth a codec requires, normally the better voice quality it is. [7].One problem that arises in the delivery of high-quality speech is network efficiency. It is feasible to provide high-quality speech; this comes at the expense of low network efficiency. Nonetheless, a much lower bitrate is desirable for a number of applications on account of the limited capacity or in order to maximize the amount of traffic that can be carried over the network. [7]

2.2.1 GSM (Global System for Mobile Communication)

GSM–Full Rate (GSM-FR) speech codec was specified in ETSI 06.10 and developed in early 1990s and was adopted by the 3GPP for mobile telephony. Full Rate (FR or GSM-FR or GSM 06.10 or sometimes simply GSM) was the first digital speech coding standard used in the GSM digital mobile phone system. The codec operates on each 20 ms frame of speech signals sampled at 8 KHz and generates compressed bit-streams with an average bit-rate of 13 kbps. The codec uses Regular Pulse Excited – Long Term Prediction – Linear Predictive Coder (RPE-LTP) technique to compress speech. The codec provides voice activity detection (VAD) and comfort noise generation (CNG) algorithms and an inherent packet loss concealment (PLC) algorithm for handling frame erasures. The codec was primarily developed for mobile telephony over GSM networks. GSM 06.10 FR codec defines a reference configuration for the speech transmission chain of the digital cellular telecommunications system. The speech encoder takes its input as a 13 bit uniform PCM signal either from the audio part of the mobile station or on the network side, from the PSTN via an 8 bit/A-law to 13 bit uniform PCM conversion.

2.2.2 Internet Low Bitrate Codec (ILBC)

ILBC stands for Internet Low Bitrate Codec and is a royalty-free narrowband speech codec, developed by Global IP Sound (GIPS). The fact of being freeware led to the adoption of iLBC in many commercial and free applications such as Skype, the Gizmo Project, OpenWengo and Google Talk. [7].It has support for two basic frame lengths: 20 ms at15.2 kb/s and 30 ms at 13.33 kb/s. When the codec operates at block lengths of 20 ms, it produces 304 bits per block. Similarly, for block lengths of 30 ms it produces 400 bits per block. Further, this codec uses a block-independent LPC algorithm. The fact of encoding each block of samples independently of the previous ones makes this codec able to withstand a certain degree of frame losses. Not with standing, while this provides better quality when 10% (or more) of the packets are being dropped, this makes the codec suboptimal for clean line conditions.

2.2.3 SPEEX Narrowband

Speex is a patent-free audio compression format designed for speech and also a free software speech codec that may be used on VoIP applications and podcasts. It is based on the CELP speech coding algorithm. Speex is a lossy format, meaning quality is permanently degraded to reduce file size. The Speex Project aims to lower the barrier of entry for voice applications by providing a free alternative to expensive proprietary speech codecs. Moreover, Speex is well-adapted to Internet applications and provides useful features that are not present in most other codecs. Finally, Speex is part of the GNU Project and is available under the revised BSD license. Speex is based on CELP and is designed to compress voice at bitrates ranging from 2 to 44 kbps. Speex is well-suited to handle VoIP, internet audio streaming, data archival (like voice mail), and audio books.

Quality

Speex encoding is controlled most of the time by a quality parameter that ranges from 0 to 10. In constant bit-rate (CBR) operation, the quality parameter is an integer, while for variable bit-rate (VBR), the parameter is a real (floating point) number.

The performance Quality based on bitrate oh the Speex codec.

- Quality 2 => bitrate 5.9 Kbps
- Quality 3 and 4 => bitrate 8 Kbps
- Quality 5 and 6 => bitrate 11 Kbps
- Quality 7 and 8 => bitrate 15 Kbps
- Quality 9 => bitrate 18.2 Kbps

2.3 VoIP Quality Performance Measurements

Important aspect in VoIP communications is the assessment of voice quality. It is imperative that new voice services undergo a significant amount of testing to evaluate their performance. In this section we outline the test methods that are geared towards the evaluation of VoIP systems.

In order to evaluate system performance when using various applications it is necessary to use specific metrics for each application; this makes it possible to measure the User-Perceived Quality (UPQ) for the corresponding application in an objective manner. Modern telecommunication networks provide a large set of voice services using many transmission systems. The rapid deployment of digital technologies in particular has led to an increased need for evaluation of the transmission characteristics of new communication equipment in terms of user-perceived quality. The methods for UPQ assessment can be divided in two main classes: intrusive and non-intrusive. Intrusive methods use special test signals, generally produced artificially by a stimulus generator so as to have similar characteristics with human speech. These test signals are sent through the network between two end points.

Based on the reference input signal and the received degraded signal a quality metric is computed that corresponds to the connection between those two end points.[4]

Non-intrusive UPQ measurement requires the use of traffic monitors. One category of such methods uses general traffic measurements of QoS parameters to predict the quality of a voice communication that would take place over that channel. Another category of methods analyses the content of the real voice traffic transiting the network. By comparing the properties of human speech with such methods can estimate the associated quality metric. Therefore, ITU-T has defined several standards that allow an evaluation of the quality of voice communication. The first of them was a subjective metric, the Mean Opinion Score (MOS) but successive attempts have been made to define objective metrics as well. These ITU-T recommendations are detailed next.

2.3.1 Mean Opinion Score (MOS)

In order to provide quantitative assessment of the quality of VoIP communications, the Mean Opinion Score (MOS) has been introduced. The MOS indicates the perceived voice quality of a VoIP conversation, ranking the call quality as a number in the range 1 to 5. Originally, the MOS was meant to represent the arithmetic mean average of all the individual quality assessments given by people who listened to a test phone call and ranked the quality of that call. Today, human participation is no longer required to determine the quality of the audio stream. Modern VoIP quality assessment tools employ artificial software models to calculate the MOS. MOS score is an indication of what users would think about the call. It was developed using surveys of users of different technologies, but today it is calculated through the use of engineering formulae. In 1996 ITU-T has defined the methodology of determining how satisfactorily given telephone connections may be expected to perform [P.800].[4].The methods described by this recommendation are intended to be generally applicable for any possible form of degradation: loss, circuit noise, transmission errors, environmental noise, talker echo, distortion due to encoding, etc.

Testing methods can be classified into subjective and objective tests [7]. Subjective methods rely on the opinion of a panel of listeners, who are usually asked to rate the quality of the test sentences read aloud by both male and female speakers over the communications medium being tested. Then a Mean Opinion Score (MOS) is computed by averaging all votes out. MOS is expressed as a single number on a scale from 1 to 5, where 1 represents the lowest perceived quality, while 5 is the highest perceived quality. MOS tests for voice are specified in ITU-T Recommendation P.800 [7]. The evaluation procedure is based on subjective tests in which quality is graded by human experimenters. The following values are assigned depending on the quality of the connection:

***Excellent*=5; *Good*=4; *Fair*=3; *Poor*=2; *Bad*=1**

2.3.2 Relative Factor (R-factor)

E-Model provides a powerful method of assessing whether a WLAN data network is capable and ready to carry VoIP calls as well as performing voice-readiness testing. An E-model calculation considers all of the following factors: delay, percentage of packets lost, delay introduced by the jitter buffer, and the behavior of the codec. Once the R value is calculated from these factors, an estimate of the MOS can be directly calculated from it. Furthermore, the maximum number of simultaneous of VoIP calls that can be handled by the WLAN will be determined.

The E-model first appeared in 2000, and was updated several times, the last revision being from 2005 [G.107]. This recommendation proposes a non-intrusive UPQ assessment method. The E-model is a computational model for use in transmission planning, hence a transmission rating model that can be used to help ensure that users will be satisfied with end-to-end transmission performance.

The model integrates in the rating value R, called transmission rating factor (R-value), the impairment factors that affect communication equipment, including delay and low bit-rate codecs. These impairments are computed based on a series of input parameters for which default values and permitted ranges are specified. These should be used if the corresponding impairment situation occurs.

The general formula is:

$$\boxed{\boldsymbol{R = R_0 - I_s - I_d - I_{e\text{-}ef} + A}}$$

Where:

R_0 = *basic signal-to-noise ratio*

I_s = *factor for impairments that are simultaneous with voice transmission*

I_d = *delay impairment factor*

$I_{e\text{-}eff}$ = *packet-loss-dependent effective impairment factor*

A = *advantage factor (system specific)*

Since the computation of the rating factor R involves a large number of parameters, complementary recommendations and appendices have been proposed by ITU-T, such as [G.108] and [G.113] that give the values for these parameters for pre-determined conditions for which the model has been calibrated. The MOS score (equivalent to the mean conversation-opinion score MOS_C from can be obtained from R using the following formulae:

For $R<0$: $MOS=1$

For $0 \leq R \leq 100$: $MOS=1+ 0.035 \cdot R+R+ (R-60)\ (100-R) \cdot 7 \cdot 10$ ^ (-6)

For $R>100$: $MOS=4.5$

The graph of the dependency of MOS on R is shown below. Note that the maximum obtainable MOS is 4.5, the average score that usually results from subjective tests for excellent quality, since experimenters' grades are known to vary between 4 and 5 in such condition grades are known to vary between 4 and 5 in such conditions.

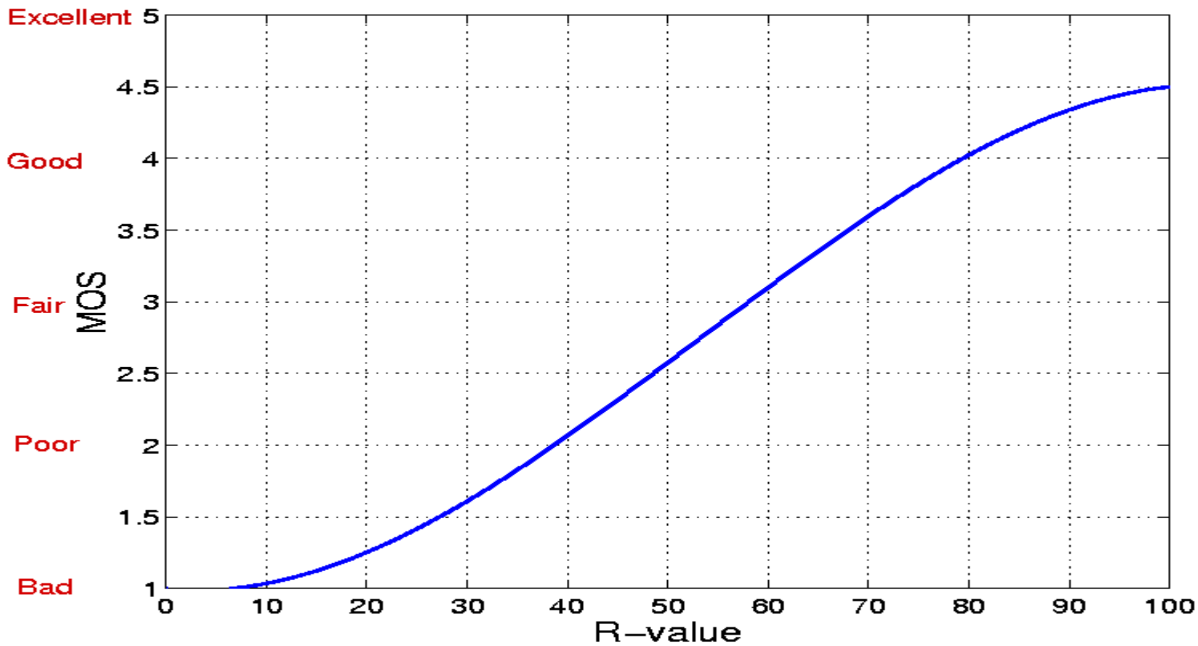

Figure 4: MOS versus rating factor R [4]

A guide of the relationship between the rating factor *R* (R-value), MOS value and user satisfaction is given in the table below:

R-Factor	MOS (lower limit)	Satisfaction Level
90 to 100	4.3-5.0	Very satisfied
80 to 90	4.0-4.3	Satisfied
70 to 80	3.6-4.0	Some users dissatisfied
60 to 70	3.1-3.6	Many users dissatisfied
50 to 60	2.6-3.1	Nearly all users dissatisfied
0 t0 50	1.0-2.6	Not Recommended

Table 1: Relations between the R factor, Satisfaction Level and the MOS rating

2.3.3 Packet Loss

Packet loss occurs in every kind of network. A Packet loss occurs when one or more packets of data fail to reach their destination. A single packet loss is referred to as “packet gap”, and series of packet loss is known as” burst”. Packet loss can occur for a variety of reasons including link failure, high congestion levels, misrouted packets, buffer overflows and a number of other factors. Packet loss causes interrupted playback and degradation in voice quality. Packet loss can be controlled using packet loss concealment techniques within the playback codec. Network protocols are designed to cope with the loss of packets in one way or another. TCP protocol, for example, guarantees packet delivery by sending re-delivery requests for the lost packets. RTP employed by the VoIP protocol does not provide delivery guarantee, and VoIP must implement the handling of lost packets. While a data transfer protocol can simply request re-delivery of a lost packet, VoIP has no time to wait for the packet to arrive. In order to maintain call quality, lost packets are substituted with interpolated data.

A technique called Packet Loss Concealment (PLC) is used in VoIP communications to mask the effect of dropped packets. There are several techniques that may be used by different implementations. Zero substitution is the simplest PLC technique that requires the least computational resources. These simple algorithms generally provide the lowest quality sound when a significant number of packets are discarded. Waveform substitution is used in older protocols, and works by substituting the lost frames with artificially generated, substitute sound. The simplest form of substitution simply repeats the last received packet. Unfortunately, waveform substitution often results in unnatural, “robotic” sound when a long burst of packets is lost. The more advanced algorithms interpolate the gaps, producing the best sound quality at the cost of using extra computational resources. The best implementation can tolerate up to 20% of packets lost without significant degradation of voice quality. While some PLC techniques work better than others, no masking technique can compensate for a significant loss of packets. When bursts of packets are lost due to network congestion, noticeable degradation of call quality occurs. In VoIP, packets can be discarded for a number of reasons, including network congestion, line errors, and late arrival. We need to select right Packet Loss Concealment technique that best matches the characteristics of a particular environment, as well as to implement measures to reduce packet loss on the network.

2.3.4 Packet Jitter

Jitter is a specific VoIP Quality of Service issue that may affect the quality of the conversation if it goes out of control. Unlike network delay, jitter does not occur because of the packet delay, but because of a variation of packet delays. As VoIP endpoints try to compensate for jitter by increasing the size of the packet buffer, jitter causes delays in the conversation. If the variation becomes too high and exceeds 150ms, callers notice the delay and often revert to a walkie-talkie style of conversation.

There are several steps to be taken to reduce jitter both on the network level and in the VoIP end points such as VoIP software, IP phones or dedicated VoIP ATA's (adaptors) or FXS/FXO gateways. By definition, reducing the delays on the network helps keep the buffer under 150ms even if a significant variation is present. While the reduced delay does not necessarily remove the variation, it still effectively reduces the degree to which the effect is pronounced and brings it to the point where it's unnoticeable by the callers. Prioritizing VoIP traffic and implementing bandwidth shaping also helps reduce the variation of packet delay. At the endpoint, it is essential to optimize jitter buffering. While greater buffers reduce and remove the jitter, anything over 150ms noticeably affects the perceived quality of the conversation. Adaptive algorithms to control buffer size depending on the current network conditions are often quite effective. Fiddling with packet size (payload) or using a different codec often helps control jitter. While jitter is caused by network delays more often than by endpoints, certain resource struggling systems that are executed in concurrent environments, such as VoIP soft phones, may introduce significant and unpredictable variations in packet delays. While developing VoIP endpoints or examining call quality problems within existing VoIP infrastructure, it is very important to isolate the cause of jitter. Real-time voice communications over the network are sensitive to delay in packet arrival time or packets arriving out of sequence.

Excess jitter results in calls breaking up. Jitter can be reduced to a certain extent by using jitter buffers. Jitter buffers are small buffers that cache packets and provide them to the receiver in sequence and evenly spaced for proper playback. Buffer lengths can be modified; however, if jitter buffer is increased too much then the call will experience an unacceptable delay. Consequently, a reduction in buffer turns results in less delay but more packet loss. Jitter is measured in milliseconds (ms).

2.4 Wireless LANs

As computer equipment users chose to become mobile, the technology had to adapt and offer wireless connectivity. Wireless will probably replace fixed connections in the same way in which mobile phones became the method of choice for person-to-person communication. However the transition may not be straightforward because of the inherent characteristics of WLAN. On the other hand, the Wireless Local Area Network (WLAN) becomes popular to support high-data-rate Internet access for users in proximity of an access point (AP). The main advantages of WLAN are its simplicity, flexibility and cost effectiveness. In the past several years, the IEEE 802.11WLAN has become a ubiquitous networking technology and has been widely deployed around the world. Although most existing WLAN applications are data centric, such as web browsing, file transfer and electronic mail, there is a growing demand for multimedia services over WLANs s.

2.4.1 Wireless LAN standards

Wireless LAN standards can be grouped into several families. 802.11and 802.11x refers to a family of specifications developed by the IEEE for *wireless LAN* (WLAN) technology. 802.11 specify an over-the-air interface between a wireless client and a base station or between two wireless clients. The IEEE accepted the specification in 1997. The most important will be briefly described next. There are several specifications in the 802.11 family:

The *IEEE 802.11* family is comprised of:

- **802.11** — applies to wireless LANs and provides 1 or 2 Mbps transmission in the 2.4 GHz band using either frequency hopping spread spectrum (FHSS) or direct sequence spread spectrum (DSSS).
- **802.11a** — an extension to 802.11 that applies to wireless LANs and provides up to 54-Mbps in the 5GHz band. 802.11a uses an orthogonal frequency division multiplexing encoding scheme rather than FHSS or DSSS.
- **802.11b** (also referred to as 802.11 High Rate or Wi-Fi) — an extension to 802.11 that applies to wireless LANS and provides 11 Mbps transmission (with a fallback to 5.5, 2 and 1-Mbps) in the 2.4 GHz band. 802.11b uses only DSSS. 802.11b was 1999 ratification to the original 802.11 standard, allowing wireless functionality comparable to Ethernet.
- **802.11e** — a wireless draft standard that defines the ***Q**uality **of** **S**ervice* (QoS) support for LANs, and is an enhancement to the 802.11a and 802.11b wireless LAN (WLAN) specifications. 802.11e adds QoS features and multimedia support to the existing IEEE 802.11b and IEEE 802.11a wireless standards, while maintaining full backward compatibility with these standards.
- **802.11g** — applies to wireless LANs and is used for transmission over short distances at up to 54-Mbps in the 2.4 GHz bands.
- **802.11n** — 802.11n builds upon previous 802.11 standards by adding ***m**ultiple-**i**nput **m**ultiple-**o**utput* (MIMO). The additional transmitter and receiver antennas allow for increased data throughput through spatial multiplexing and increased range by exploiting the spatial diversity through coding schemes like Alamouti coding. The real speed would be 100 Mbit/s (even 250 Mbit/s in PHY level), and so up to 4-5 times faster than 802.11g. [9]

At the moment 802.11b is probably the most widely used WLAN standard, but there are devices that are compatible with all three standards in the same time. As always in the ITC the tendency is to migrate to faster technologies as soon as they become affordable. Each standard from the 802.11 family has its strengths and weaknesses. For example, there is less potential for Radio Frequency (RF) interference for 802.11a, than for 802.11b or 802.11g. Given the larger bandwidth, this solution is better than 802.11b at supporting multimedia voice, video and large-image applications in densely populated environments. However the range is shorter than for 802.11b and they are not interoperable. [4]

In the case of 802.11b fewer access points are required than for 802.11a for the coverage of large areas (with a range of up to 100 m from the base station). A number of 14 channels are available, with three non-overlapping channels. 802.11b is compatible with 802.11g, which may eventually replace 802.11b since it provides higher data rates and security enhancements.

An important element of the 802.11 family of standards is the concept of ad-hoc network. This operation mode is intended to allow wireless communication in locations where an access point is not available, or access to a wired network is not required.

Given the different environments in which WLANs are used, the types of information that need to be transmitted on these networks vary. Convergence, i.e. the use of the same network for multiple purposes, such as communicating data, telephony, video conferencing, is an important trend in the field of ICT. Although convergence has become increasingly prevalent, satisfactory solutions have not yet been found even for the traditional fixed networks. Due to the inherent properties of wireless networks, the situation becomes even more challenging in this case.

First of all the bandwidth available in WLANs is significantly lower than in the case of fixed LANs. For the most widely-spread wireless networks, the maximum theoretical rate is either 11 Mb/s or 54 Mb/s. These rates are considerably lower than the current extensively-used 100 Mb/s and 1 GB/s fixed LANs. [4]

2.4.2 Signal Noise Ratio (SNR)

Signal to noise ratio is a specification that measures the level of the audio signal compared to the level of noise present in the signal. Signal to noise ratio specifications are common in many components, including amplifiers, phonograph players, CD/DVD players, tape decks and others. Noise is described as hiss, as in tape deck, or simply general electronic background noise found in all components. As the name suggests, signal to noise ratio is a comparison or ratio of the amount of signal to the amount of noise and is expressed in decibels. Signal to noise ratio is abbreviated S/N Ratio and higher numbers mean a better specification. A component with a signal to noise ratio of 100dB means that the level of the audio signal is 100dB higher than the level of the noise and is a better specification than a component with an S/N ratio of 90dB.

Why is it Important?

Unfortunately, all components add some level of noise to an audio signal but it should be kept as low as possible. Analog components, such as amplifiers, tape decks and phonograph players generally have a lower signal to noise ratio than digital components, such as CD and DVD players but the goal is still to keep noise as low as possible. As an example, a signal to noise ratio for tape deck or phonograph player is typically about 60dB-70dB, while it is common for a CD player to have a S/N Ratio of 100dB or higher. S/N Ratio is important, but should not be used as the only specification to measure the sound quality of a component.

2.5 Existing Research

In order to require information regarding to the performance analysis of voice codec (GSM ILBC, Speex) of VOIP over Wireless Local Area Network (WLAN) few journals, articles, books, and research studies has been studied and analyzed. The journals reviewed have some comparison toward our research topic. Below are some journals and research studies that are correlated to our research topic.

2.5.1 VoIP over Wireless LAN Survey

In this paper, the researcher has performed a survey of the current state of the art in voice communication over wireless networks. The properties of WLANs and VoIP are presented, and then the issues related to the deployment of VoIP over WLAN are analyzed. The main findings of this survey are the following. WLAN QoS parameters have a high variability in real-world environments, with a significant effect on application performance. Existing WLAN QoS mechanisms are only of limited use for managing contention for applications with different QoS requirements. VoIP is a multimedia application that requires timely servicing of the voice traffic; this is a challenging task in WLANs, even when using QoS enforcement. In this paper, they proposed an experiment testbed which is proposed at the end that allows an objective verification of the properties of existing technologies, as well as the development of new techniques. The testbed can make use of WLAN emulation to allow experimentation in a wide range of controllable network conditions. [4] The speech codec used in this research are G.711, G.726, GSM and G.729. The speech codec performance measurements used are MOS (Mean Opinion Score), E-model and the PESQ score.

2.5.2 Voice over Wireless Mesh Networks: A Case Study in the Brazilian Amazon Region during the Rainy Season

In this paper the researcher present the results of an experimental analysis of the behavior and performance of VoIP calls using different codecs in a wireless mesh network (WMN) based on the IEEE 802.11g standard. Most measurements were done during the rainy season in a Brazilian Amazon region. The call qualities of codecs were accessed using real measurement conditions showing the codecs performance over wireless mesh networks. In this paper they also examined data related to characteristics of latency, packet loss rate, throughput, and MOS values of conversations. The phone calls were done using ITU H.323 signaling and the speech codecs ITU G.711, ITU G.723.1, ITU G.729A and iLBC (RFC 3951). The speech quality of VoIP calls was evaluated by means of the E-model methodology and the result is given in the MOS scale.

2.5.3 Voice Quality Evaluation of Various Codecs

In this paper, the researcher present a test on a large amount of absolute mean opinion scores (MOS) obtained within a single listening. Naive listener's preference on different speech signal properties such as mono/stereo and bandwidth was studied. Various codecs were ranked by their subjective voice quality. The listened speech sequences were recorded and selected to represent several realistic stereo audio capture and background noise configurations, where there are either one or several speakers. The number of conditions was selected to be as large as possible to be listenable in a single two hour session. [11] Due to the test size, the results are divided into smaller graphs where interesting comparisons between different conditions can be easily evaluated. The narrowband and wideband codecs were compared.

2.6 Comparison of the existing research

RESEARCH JOURNAL TITLE / TYPE OF COMPARISON	VoIP over Wireless LAN Survey	Voice over Wireless Mesh Networks:	Voice Quality Evaluation of Various Codecs	Performance Analysis of Voice Codecs (GSM, ILBC, Speex) for VOIP over Wireless Local Area Network (WLAN)
SPEECH CODECS	G.711 G.726 GSM G.729	G.711 G.723.1 G.729A ILBC	Narrowband codec Wideband codecs	GSM ILBC Speex
SPEECH QUALITY MEASUREMENT TOOLS	Mean Opinion Score (MOS)	E-model methodology	Absolute Category Rating MOS (ACRMOS)	Mean Opinion Score (MOS)
	E-model and the PESQ score.	Mean Opinion Score (MOS)		Relative Factor (R-factor)
WLAN ENVIRONMENT	The testbed at JAIST, StarBED has been already used successfully to emulate various environments	They have taken low cost WLAN routers based on the IEEE 802.11b/g standard	They conducted the listening test in Nokia Research Center listening test facilities	IEEE 802.11n as the wireless network standard based on given SNR value with one AP.
SIMULATION MODE	Measuring the quality degradation at the network level by means of QoS parameters (delay & jitter, packet loss, throughput).	By examined data related to characteristics of latency, packet loss rate, throughput, and MOS values of conversations.	The number of conditions was selected to be as large as possible to be listenable in a single two hour session.	To analyze the group of speech codec (GSM, ILBC, Speex) performances in terms of Packet Jitter (ms), Packet Lost (%), MOS, and R-Factor based on the simulation.

Table 2: Comparison between the research journals

CHAPTER 3

RESEARCH METHODOLOGY

3.0 INTRODUCTION

The research project will be done by conducting the performance analysis of voice codec (GSM, ILBC, Speex) for VOIP over Wireless Local Area Network. The analysis will be based on the performance measurement based on Packet Jitter (ms), Packet Loss (%), Mean Opinion Score (MOS) and Relative Factor (R-Factor). For each criteria's, 10 test results will be collected.

To make sure the objective of the research is fulfilled; few steps have been identified as a guideline. The guideline is:

1. Information gathering. Collect all data from all resource based on research topic.
2. Planning and identifying hardware and software tools. All the hardware and software tool must be determined to make sure it is compatible with the test bed.
3. Hardware setup, software configuration and experiment design. It is to ensure that the experiment can be performed.
4. Implementation and experimentation. Performing the testing based on the predefine model is used to collect all the data related to the experiment.
5. Data analysis. All the data is gather to be analyed.

INFORMATION GATHERING

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

PLANNING AND IDENTIFYING HARDWARE AND SOFTWARE TOOLS

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

HARDWARE SETUP, SOFTWARE CONFIGURATION AND EXPERIMENT DESIGN

ARCHITECTURE DESIGN PHASE	
INSTALLATION	Hardware and software tools.
SETTING AND CONFIGURATION	Access point, Laptop for caller (sender) and caller (receiver).
ENVIRONMENT	WLAN Environment
ARCHITECTURE	Two way communication between two callers on Optimum Network Two way communication between two callers with other traffic on the same network.

IMPLEMENTATION AND EXPERIMENTATION

TESTING PHASE
• Generate two ways communication of VoIP session on Optimum Network • Manipulate the network by add another traffic to make it like real network link • Generate report for both networks (Optimum Network and Network with Other Traffic) of the rate of call quality analysis on Packet Jitter (ms), Packet Loss (%), MOS and R-factor using CommView analyzer software.

DATA ANALYSIS

DATA ANALYSIS PHASE
Things to consider are Jitter and Packet loss. Voice/call quality measurement analysis (MOS and R-Factor) stated on reading of CommView.

Table 3: Methodology of this research

3.1 Methodology.

To make sure that this research project can be performed smoothly and completely, five main phases is defined to be used. This methodology helps in collecting more accurate data on performance analysis of the grouping speech codec. The performance measurement based on Mean Opinion Score (MOS) and Relative Factor (R-factor). The research will be performed phase by phase accordingly. The main phases are:

- Preliminary study phase
- Research planning phase
- Architecture design phase
- Testing phase
- Data analysis phase

3.1.1 Preliminary Study Phase

The first important step is by doing preliminary study about the research topic. This is to ensure that the research project will be in the right track by searching the appropriate information which related to the problem statement stated. In addition, the problem statements become a great guideline in planning and extracting the right information. Various sources are used to gather all information from offline-base resources to the online based resources such as books, journal, thesis project, articles and research paper.

Moreover, not many researches were done on specific group of speech codec based on their sampling requirement. Speech codec has its own bandwidth requirement. For this research study, a specific group of speech codecs (GSM, ILBC, Speex) with bit rate, 11-13 kbps are selected to determine the best and suitable voice codecs for different type of ideal network condition. So, which one of the codec is the best for VoIP session. From the preliminary studies, it shows that majority of previous research was done in terms of 2 way communications. Then, all clients were expected to be simulated together.

Besides that, the great recommendations of the speech quality measurement performance tools, the best speech codecs used and various type of WLAN environment related to this research project will be taken into consideration. The limitations of existing research project are used in order to know more about the research topic.

3.1.2 Research Planning Phase

The second step will be the research planning phase. In this phase all the hardware and software equipment will be determine for experimental setup. Then, the testing procedure will be used to make sure all of it work and well-functioning. To achieve the desired objectives, research planning phase is very 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. The hardware uses are two laptops, one act as a sender of packet and another one act as the receiver 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. Besides that, the other hardware that will be used for this research is wireless access point. Wireless access point (AP) also required in this research to act as a bridge to make the two laptops connect and communicate. Then, a dedicated wireless USB card is used specifically to sniff the link between two computers in a VoIP session and analyze of the VoIP packet for performance measurement.

The tools software will be installed on the computer used for this analysis report. The required important software tools such as quality measurement performance of speech codec devices called CommView that includes a network analyzer contain Packet Jitter (ms), Packet Loss (%), Mean Opinion Score (MOS) and Relative-Factor (R-Factor) is very crucial for the research project to be carried out. Mean Opinion Score (MOS) readings will be calculated by the software. It also is used to generate simulation results and reports for VoIP call quality. 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 each of the codec used.

Besides that, the software of SIP Server (Brekeke SIP Server) is installed on one computer to make calls that 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. Then, VOIP Client (EyeBeam and X-Lite) those are compatible and supported to the Speech Codecs are required to perform simulations sessions to generate a simulation report. VoIP client is used as a tool for researcher to establish a VoIP call session. Selected VoIP client should have a voice codec best selection to function during the simulation process. Researchers were able to select the voice codec that want to test in the simulation test bed.

Finally, the testing step and procedure need to be defined so that the desired objectives of this research will be accomplish. Scope limitation of research project also is defined during this phase. Below the table shows the initial software and hardware requirement for this research.

SOFTWARE TOOLS NAME	LICENSE	DESCRIPTION
NETWORK ANALYZER CommView	Proprietary (With 30 day full features trial version)	Tool for network monitoring and analyzer. CommView includes a VoIP analyzer for in-depth analysis, recording, and playback of SIP voice communications (15 seconds only) R-Factor, MOS Score – stream quality estimation and view protocols distribution and payload uses by VOIP session. Generate traffic reports in real time.
SIP SERVER Brekeke SIP Server	Proprietary	Brekeke SIP Server is SIP-compliant that ensures that it has the highest level of interoperability with other SIP devices, services and other client.
VOIP CLIENT EyeBeam	Proprietary	Session Initiated Protocol (SIP) based signaling. Performance Management of the SIP end-point. High Compression codec support Enhanced Quality of Service (QoS) for voice & video calls.

VOIP CLIENT X-Lite		Session Initiated Protocol (SIP) based signaling. A communications freeware product made for those with a VoIP phone system X-Lite supports traditional phone use, and video or conference calls.

Table 4: Software Tools lists and Description

3.1.3 Architecture Design Phase

In this phase the installation and the configuration setting of the required necessary hardware and software is executed to provide the test bed for the wireless local area network (WLAN) environment to perform the testing of the speech codec performance based on Mean Opinion Score (MOS) and R-Factor in order to obtain the data readings for the analysis phase.

In this research study, three speech codecs; GSM (13 kbps), ILBC (13.33 kbps), and Speex (11 kbps) were tested together with several predetermined SNR value ranging from 10dB to 30 dB with a sample rate of 10 second speech. VoIP QoS such as packet jitter, packet lost, MOS, and R-Factor is analyzed in order to make a comparison of speech quality of those three speech codecs in wireless LAN 802.11n environment. Three values of SNR have been chosen for the analysis on wireless performance. The SNR values are 10dB, 20dB, and 30dB.

Two way communications methods will be used in this simulation test bed. The communication is two callers talk simultaneously. Then, other situations will be included in this simulation is the use of network type that have optimum network and network with other traffic in a same network with a VoIP session. VoIP will allow two-way communication process that is very sensitive to delays in a network. Moreover, the simulation result is important to test the performance of the codec used during two-way communication. In addition, optimum network and network with other traffic using the same sample rate and the bitrate of the codec itself. Voice quality performance in a call also may be better and less noise elements found in the calls that are made.

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 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 traffic in a network such as file sharing and others.

For performance result of both types of network proposed VoIP simulation, we use the Packet Jittter (ms), Packet Lost (%), MOS and R-Factor as the value that determines the quality of the call that has been specified by ITU-T recommendation. Usually use MOS and

R-Factor as a value that determines the performance of a voice or the call is used as it has been set by the International Telecommunication Union (ITU). The VoIP session will be capture and analyze by the wireless monitoring tool (CommView) is to display the actual performance result.

The testbed as below:

I. Host A and Host B were set up, one for sender (person that make call) and one for receiver (person that receive a call).

II. Then, both of Host A and Host B will be installed with VOIP Client (EyeBeam and X-Lite) that is compatible and supported to the Speech Codecs (GSM, ILBC, and Speex).

III. Host C will be installed with the SIP Server called Brekeke SIP Server to established VOIP call session.

IV. Next, Host C is dedicated for wireless performance. It will be installed with network analyzer software network monitoring tools to sniff network utilization and bandwidth or bitrate usage, called CommView that includes main performances analysis quality of VoIP calls which are Packet Jitter (ms), Packet Loss (%), MOS and R-Factor.

V. Wireless access point (AP) – Access point will be installed and configured to ensure that both the caller can communicate with each other. Set up the wireless AP with the default setting for the wireless communication to take place between the sender and the receiver.

Diagram shows two way communications of VOIP over Wireless Local Area Network.

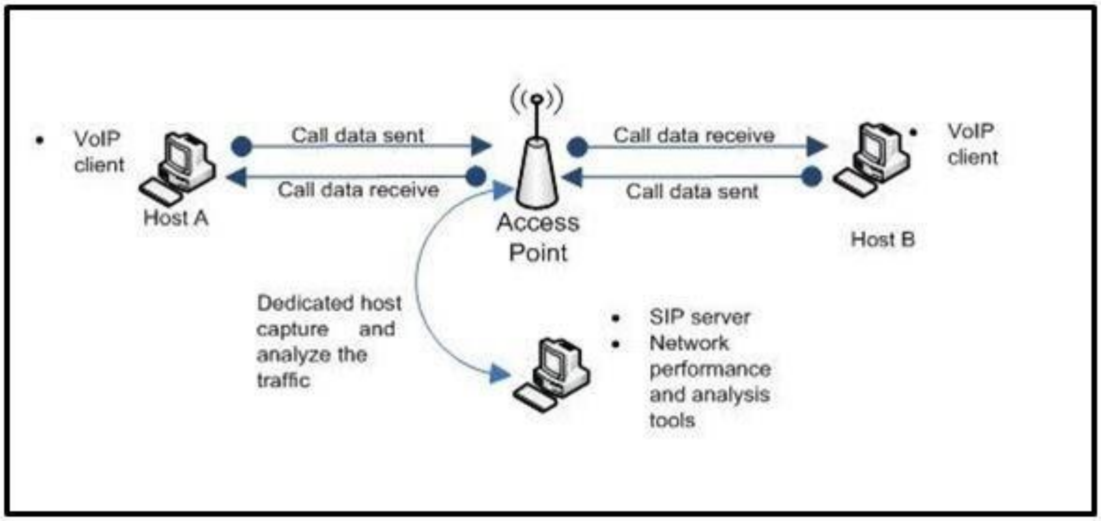

Figure 5: Diagram for the simulation

3.1.4 Testing Phase

After the architecture design phase has been complete, the testing phase need to be implement to test the simulation of the group of speech codec (GSM, ILBC, Speex) for VOIP on predefined wireless MESH network. When simulation environment were set up with hardware and tools, the software should be run and assembled to produce desired performance analysis result. Then, the outcome of analysis report must be accomplished with the objective stated in this project research. Tools for analyzing the simulation results must be functioning well. Moreover, unsatisfactory results will be produce by the device.

CommView software tools are used to generate analysis of VOIP session whether it is bad or good performance. From CommView, the performances of speech codes are determined by the MOS reading and R-Factor values shown. This software tool is able to generate the report of MOS, R-Factor, packet size, jitter, and bandwidth stream to be generated at the end of the VOIP session. This software tool also can provide some correlation graphs that will be used to clarify the results for the simulation experiment.

The simulation result must be accurate to obtain good simulation results. Simulation test bed will be done to ten times simulation. The simulations are done to calculate the average results of the QOS Parameter such as Packet Jitter (ms), Packet Lost (%), MOS, and R-Factor. Therefore, the average value of the simulation results can be made. After the process of collecting the required analysis report completed, the tables will be used to display the results for each situation in the simulation have been generate. The average reading for 10 QOS (Quality of Services) Parameter samples for every each of the tested codec on both types of network will be taken as the final result. Here the will be a table of performance measurement.

<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>											

Table 5: The example of data analysis table

3.1.5 Data Analysis Phase

Data analysis phase is the last phase of the research project. In this phase, the entire data's are obtained and collected. The entire data gather that was collected is vital towards this research project as they will be interpreted and analyzed to answer the research objectives, research questions and problem statements. A comparative analysis will be carried out for the attributes that consist of voice quality performance of speech codecs and the best codec based on their bit rate to be used for the report analysis as the simulation results that accomplished the objective research. Conclusion of this research project is derived from the data analysis phase.

Chapter 4

DESIGN AND IMPLEMENTATION

4.0 Introduction

In this chapter will discuss on the design of the experimental analysis and the implementation of the tools. In this experimental analysis the design will be based on the objectives of the research. This is to ensure that it can fulfill the objectives requirement of this research project that has been defined. It is crucial to state the step-by-step procedure in detail to ensure that this research project is on the right track. Then, the defined procedure of methodology always coincides with the main objectives of this research project. The experimental setup was made to provide the appropriate and necessary simulations for this research. This simulation research will conduct two laptops in the same network interconnection. In this experimental, use Wireless Local Area Network (WLan) where the two nodes will communicate with each other via wireless access point. Then, a wireless access point will act as a bridge for the two laptops to connect and communicate to each other. Based on the research methodology in design phase, it is important to ensure that the hardware and software tools used in the simulation will be able to perform and carry out the task to obtaining the required absolute data.

4.1 Experimental Environment Design

In this phase, the configuration setting of the hardware and installation of the software tools that's use to create a VoIP call session. This is executed to provide the test bed for the wireless environment. This test bed is to conduct the performance analysis of voice codec (GSM, ILBC, Speex) for VOIP over Wireless Local Area Network. Then, network analyzer tool (CommView) used to analyze the performance of VoIP sessions will be performed. The analysis will be based on the performance measurement based on Mean Opinion Score (MOS) and Relative Factor (R-factor). For each criteria's, 10 test results will be collected in order to obtain the data readings for the analysis phase. Some of the activities will be carried out to complete the process of preparing this test bed environment. The activities involved are:

I. **Laptop** – Three laptops called Host A and Host B were set up, one for sender (person that make call) and one for receiver (person that receive a call) as a caller for VoIP to control the VOIP call session. Then, both of Host A and Host B will be installed with VOIP Client (EyeBeam and X-Lite that is compatible and supported to the Speech Codecs (GSM, ILBC, Speex) as a tool to use VoIP service and can determine which codec will be the best to use for each call. Next, Host C will be installed with the SIP Server (BREKEKE SIP Server) and CommView (Network Analyzer) dedicated for wireless performance. Network analyzer software or network monitoring tools function is to sniff network utilization and bandwidth or bitrate usage, called CommView that includes performances analysis which are MOS and R-Factor. In summary, the software tools that have been selected to perform the simulation are SIP Server (Brekeke SIP Server), VOIP Client (Eyebeam and X-Lite), and network analyzer software called CommView.

II. **Wireless access point** - Wireless AP were set up with the default setting for the wireless communication to take place between sender and the receiver. Wireless access points (APs or WAPs) are specially configured nodes on wireless local area networks (WLANs). Access points act as a central transmitter and receiver of WLAN radio signals.

Next, Access points support Wi-Fi wireless communication standards. Access point will provides wireless connection to establish connection between the two hosts in the simulation. Access point is the device hardware used to create a network that will 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.

III. **USB wireless adapter** - USB wireless adapters allows for making internet connections at a growing number of public locations. They also allow for direct ad hoc connections between Wi-Fi enabled devices without the inclusion of infrastructure devices like a router. It can operate in the 5GHz band which is nearly immune to interference and also has the notable advantage of being better able to carry network traffic that requires extensive bandwidth. Examples of such traffic include high definition video streaming, voice applications like VoIP telephony and graphics-intensive games. It’s a far simpler matter to plug in a USB Wi-Fi adapter than it is to run an Ethernet cable to a computer, especially when multiple machines are involved. In this simulation research, wireless adapters that build in the computer has been used to connect the AP to create a VoIP session, experiments need another wireless adapter that will connect to the same access point to allow this USB wireless adapter to monitor and sniff the network that was used by the VoIP call session by the two host.

IV. **Testbed** - There are two types of network environments with respective value of Signal Noise Ratio (SNR) used in this simulations that will be carried out in this research:

TESTBED 1:

- Two ways communication on <u>optimum network</u> with respective value of Signal Noise Ratio (SNR) which are 10, 20 and 30 (dB) on Packet Jitter, Packet Loss, MOS and R-Factor.

TESTBED 2:

- Two ways communication on <u>network with other traffic</u> with respective value of Signal Noise Ratio (SNR) which are 10, 20 and 30 (dB) on Packet Jitter, Packet Loss, MOS and R-Factor.

<u>Environment that will be used is shown in the diagram below:</u>

1. <u>Two ways communication on optimum network (VOIP session only)</u>

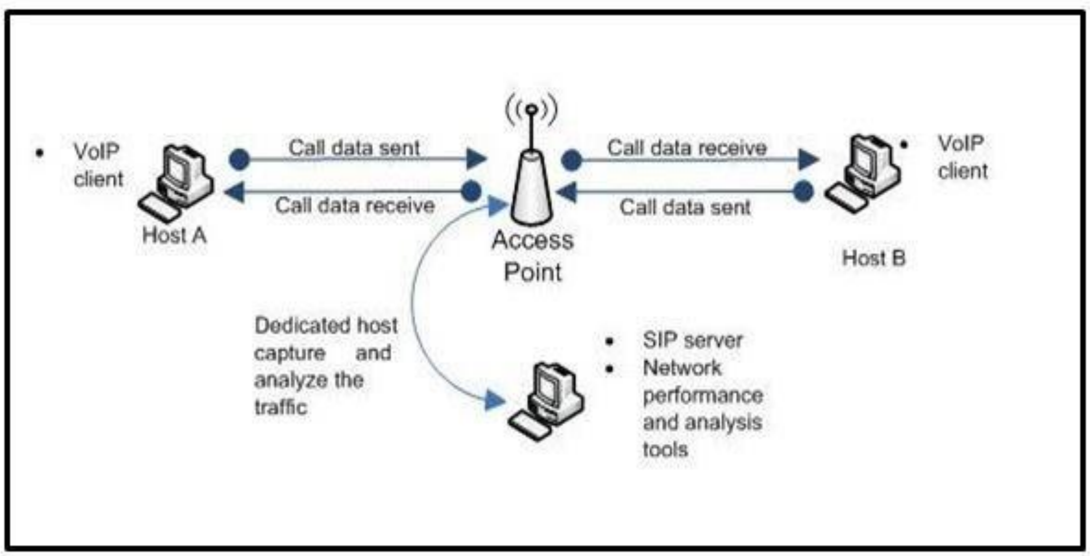

Figure 6: Two ways communication on optimum network

2. Two ways communication on network with other traffic (include others traffic; file sharing)

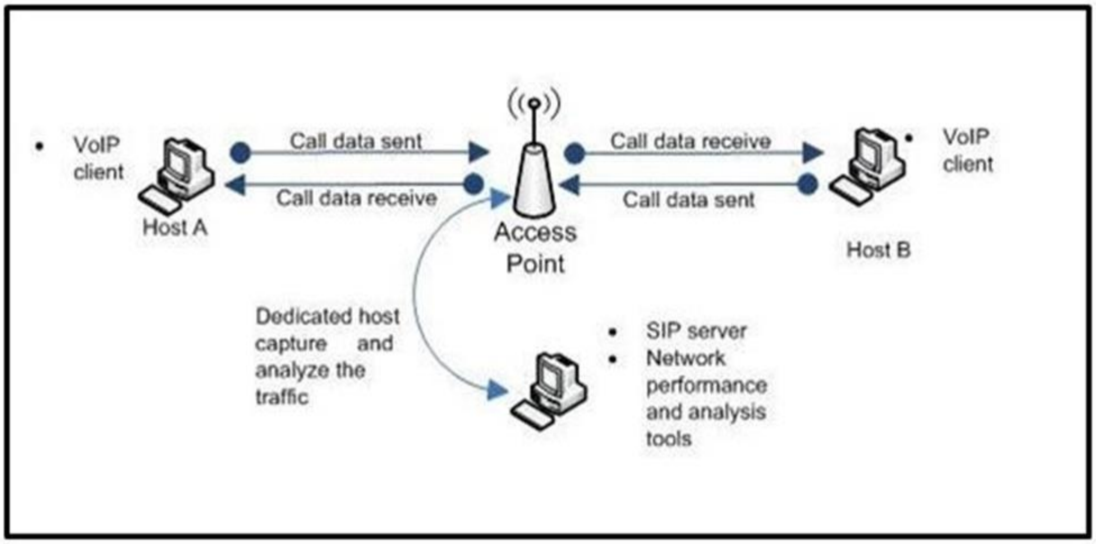

Figure 7: Two ways communication on network with other traffic

The environments that will be conduct on this experiment:

- **The VoIP calls session on optimum network of two way communications with respective value of Signal Noise Ratio (SNR) which is 10, 20 and 30 (dB)**

- For this environment, the VoIP calls session is done with the optimum network condition. In detail, the network will be dedicated totally to VoIP call session without any traffic. This is to test the real actual performance of every each specific speech codec used when VOIP call session is establish.

- **The VoIP call session on network with other traffic of two way communications with respective value of Signal Noise Ratio (SNR) which is 10, 20 and 30 (dB)**

- For the next environment, the network of VOIP call session is included with other traffic. In detail, the caller does not use the network for establish VoIP call session only but the caller using the network to the other traffic too. The example of other traffic such as FTP, and file sharing.

V. Architecture – The architecture as follows:

i. Host A and Host B were set up, one for sender (person that make call) and one for receiver (person that receive a call).
ii. Then, both of Host A and Host B will be installed with VOIP Client (EyeBeam and X-Lite) that is compatible and supported to the Speech Codecs (GSM, ILBC, Speex)
iii. Host C will be installed with the SIP Server called Brekeke SIP Server to established VOIP call session.
iv. Next, Host C is dedicated for wireless performance. It will be installed with network analyzer software network monitoring tools to sniff network utilization and bandwidth or bitrate usage, called CommView that includes main performances analysis quality of VoIP calls which are MOS and R-Factor.
v. Wireless access point (AP) – Access point will be installed and configured to ensure that both the caller can communicate with each other. Set up the wireless AP with the default setting for the wireless communication to take place between the sender and the receiver.

`

4.2 Testing Plan

Once the architecture design were completely installed and configured, the data collection will be done under this testing phase. Testing and analyzing process of the VoIP call session is begin. The selected software tools will be run to obtain the reading of the MOS and R-factor to determine the performance analysis of the speech codec used. Then, the data is collect and analyzed in the next phase. The data will be collect based on the tables below that consist of several environments testing. There are two types of environments used in these simulations that will be carried out in this research that have been described details in environment design phase. For each testing, ten readings will be taken to ensure the accuracy of the analysis that has been generated. Next, the ten readings for every environments condition will be taken. Then, the average of those ten readings will be calculated and used as the final value of the performance of speech codecs used and the quality of a VoIP call.

Furthermore, for each testbed environment condition that will be carried out, nine tables' raw data will be produced. Every table will contain ten results that included the value of Mean Opinion Score (MOS), Relative-factor (R-factor), packet lost (%) and packet jitter (ms). Then, after each test has been completely done, the results are generated. The value of the generated result will be grouped into the tables according to the category of the environment condition analysis. The value that have been added to the table will be used to produce an average charts for the value MOS, R-factor, packet loss and packet jitter. An average chart is creating to show a mean/ average reading. This chart purpose is to determine the performance of the listed speech codec used during VoIP calls. There are eighteen tables in both testbed network environment conditions with SNR reading 10, 20 and 30 respectively that will be used for the data analysis. Here the example of the table as follows:

Two way communications on optimum network based on SNR 10

Two way communication on optimum network SNR 10											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	MOS AVG
	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	
GSM											
ILBC											
SPEEX											

Table 6: Two way communications on optimum network based on MOS score

Two way communication on optimum network SNR 10											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	R-Factor AVG
	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	
GSM											
ILBC											
SPEEX											

Table 7: Two way communications on optimum network based on R-Factor

Two way communication on optimum network SNR 10											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Loss AVG
	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	
GSM											
ILBC											
SPEEX											

Table 8: Two way communications on optimum network based on Packet Loss

Two way communication on optimum network SNR 10											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Jitter AVG
	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	
GSM											
ILBC											
SPEEX											

Table 9: Two way communications on optimum network based on Packet Jitter

Two way communications on optimum network based on SNR 20

Two way communication on optimum network SNR 20											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	MOS AVG
	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	
GSM											
ILBC											
SPEEX											

Table 10: Two way communications on optimum network based on MOS score

Two way communication on optimum network SNR 20											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	R-Factor AVG
	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	
GSM											
ILBC											
SPEEX											

Table 11: Two way communications on optimum network based on R-Factor

Two way communication on optimum network SNR 20											
Readings / Codecs	1st Packet Loss	2nd Packet Loss	3rd Packet Loss	4th Packet Loss	5th Packet Loss	6th Packet Loss	7th Packet Loss	8th Packet Loss	9th Packet Loss	10th Packet Loss	Packet Loss AVG
GSM											
ILBC											
SPEEX											

Table 12: Two way communications on optimum network based on Packet Loss

Two way communication on optimum network SNR 20											
Readings / Codecs	1st Packet Jitter	2nd Packet Jitter	3rd Packet Jitter	4th Packet Jitter	5th Packet Jitter	6th Packet Jitter	7th Packet Jitter	8th Packet Jitter	9th Packet Jitter	10th Packet Jitter	Packet Jitter AVG
GSM											
ILBC											
SPEEX											

Table 13: Two way communications on optimum network based on Packet Jitter

<u>Two way communications on optimum network based on SNR 30</u>

Two way communication on optimum network SNR 30											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	MOS AVG
	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	
GSM											
ILBC											
SPEEX											

Table 14: Two way communications on optimum network based on MOS score

Two way communication on optimum network SNR 30											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	R-Factor AVG
	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	
GSM											
ILBC											
SPEEX											

Table15: Two way communications on optimum network based on R-Factor

Two way communication on optimum network SNR 30											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Loss AVG
	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	
GSM											
ILBC											
SPEEX											

Table 16: Two way communications on optimum network based on Packet Loss

Two way communication on optimum network SNR 30											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Jitter AVG
	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	
GSM											
ILBC											
SPEEX											

Table 17: Two way communications on optimum network based on Packet Jitter

Two way communications on network with other traffic based on SNR 10

Two way communication on network with other traffic SNR 10											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	MOS AVG
	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	
GSM											
ILBC											
SPEEX											

Table 18: Two way communications on network with other traffic based on MOS score

Two way communication on network with other traffic SNR 10											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	R-Factor AVG
	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	
GSM											
ILBC											
SPEEX											

Table 19: Two way communications on network with other traffic based on R-Factor

Two way communication on network with other traffic **SNR 10**											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Loss AVG
	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	
GSM											
ILBC											
SPEEX											

Table 20: Two way communications on network with other traffic based on Packet Loss

Two way communication on network with other traffic **SNR 10**											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Jitter AVG
	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	
GSM											
ILBC											
SPEEX											

Table 21: Two way communications on network with other traffic based on Packet Jitter

Two way communications on network with other traffic based on SNR 20

Two way communication on network with other traffic SNR 20											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	MOS AVG
	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	MOS	
GSM											
ILBC											
SPEEX											

Table 22: Two way communications on network with other traffic based on MOS score

Two way communication on network with other traffic SNR 20											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	R-Factor AVG
	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	R-factor	
GSM											
ILBC											
SPEEX											

Table 23: Two way communications on network with other traffic based on R-Factor

Two way communication on network with other traffic **SNR 20**											
Readings / Codecs	1st Packet Loss	2nd Packet Loss	3rd Packet Loss	4th Packet Loss	5th Packet Loss	6th Packet Loss	7th Packet Loss	8th Packet Loss	9th Packet Loss	10th Packet Loss	Packet Loss AVG
GSM											
ILBC											
SPEEX											

Table 24: Two way communications on network with other traffic based on Packet Loss

Two way communication on network with other traffic **SNR 20**											
Readings / Codecs	1st Packet Jitter	2nd Packet Jitter	3rd Packet Jitter	4th Packet Jitter	5th Packet Jitter	6th Packet Jitter	7th Packet Jitter	8th Packet Jitter	9th Packet Jitter	10th Packet Jitter	Packet Jitter AVG
GSM											
ILBC											
SPEEX											

Table 25: Two way communications on network with other traffic based on Packet Jitter

Two way communications on network with other traffic based on SNR 30

Two way communication on network with other traffic SNR 30											
Readings / Codecs	1st MOS	2nd MOS	3rd MOS	4th MOS	5th MOS	6th MOS	7th MOS	8th MOS	9th MOS	10th MOS	MOS AVG
GSM											
ILBC											
SPEEX											

Table 26: Two way communications on network with other traffic based on MOS score

Two way communication on network with other traffic SNR 30											
Readings / Codecs	1st R-factor	2nd R-factor	3rd R-factor	4th R-factor	5th R-factor	6th R-factor	7th R-factor	8th R-factor	9th R-factor	10th R-factor	R-Factor AVG
GSM											
ILBC											
SPEEX											

Table 27: Two way communications on network with other traffic based on R-Factor

Two way communication on network with other traffic SNR 30											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Loss AVG
	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	Packet Loss	
GSM											
ILBC											
SPEEX											

Table 28: Two way communications on network with other traffic based on Packet Loss

Two way communication on network with other traffic SNR 30											
Readings / Codecs	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th	Packet Jitter AVG
	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	Packet Jitter	
GSM											
ILBC											
SPEEX											

Table 29: Two way communications on network with other traffic based on Packet Jitter

Chapter 5

RESULTS AND DISCUSSION

5.0 Introduction

This chapter will elaborate more on the findings gathered of this project. In this chapter, the results of the experiment will be discussed. The speech quality of three speech codec namely GSM (13kbps) , ILBC (13.33 kbps) , and Speex (11kbps) under various network performance based on pre-determined SNR values will be evaluated and compare against. Several tests are constructed to prove that it meets the interest of investigation. The experimental procedure of this dissertation can be summarized to 2 main experiments which need to be repeated for each speech codec and for each predefined SNR value. Both types of network; 1) Optimum Network, and 2) Network with others traffic, need to be repeated for all three speech codec GSM, ILBC, Speex with each respective SNR values; 10 dB, 20 dB, and 30 dB. In determining the SNR value, we always refer to the SNR reading on the access point (AP) wireless settings for both clients. For comparing all three speech codec, at each scenario the CommView will capture and analyze the VoIP QoS namely packet jitter, packet loss, MOS, and R-Factor. This part will perform all the mechanism involved with the result refers to the group of speech codec (GSM, ILBC, Speex) for VOIP. This chapter also will analyze the group of speech codec (GSM, ILBC, and Speex) performances in terms of Packet Jitter (ms);

Packet Loss (%), Mean Opinion Score (MOS) and Relative Factor (R-Factor) based on the simulation and will suggest the best speech codec on the predefined wireless mesh network.

5.1 Result Analysis

For this experimental analysis, three speech codecs; GSM (13kbps), ILBC (13.33kbps), and Speex (11kbps) were tested together with several predetermined SNR value ranging from 10dB to 30dB with a sample of 10 second speech. VoIP QoS parameter such as packet jitter, packet loss, MOS, and R-Factor are analyzed in order to make a comparison of speech quality of those three speech codecs in wireless LAN (WLAN) 802.11n environment. Ten times of testing have been done on both types of network environment on three values of SNR that have been chosen for the analysis on wireless performance. The SNR values are 10dB, 20dB, and 30dB. Then, evaluate the performance of speech quality for GSM, ILBC, and Speex speech codec by gathering data based on a VoIP QoS Parameter; Packet Jitter (ms), Packet Lost (%), MOS, and R-Factor.

5.2 Optimum Network Environment (TESTBED 1)

For this environment, the VoIP calls session is done with the optimum network condition. In detail, the network will be dedicated totally to VoIP call session without any others traffic. This is to test the real actual performance of every each specific speech codec (GSM, ILBC, Speex) used when VOIP call session is establish. Two ways communication on optimum network with respective value of Signal Noise Ratio (SNR) which are 10, 20 and 30 on Packet Jitter, Packet Loss, MOS and R-Factor is evaluate.

5.2.1 Optimum Network Environment Based on Packet Jitter Comparison

The packet latency is a delay on delivery of a piece of the conversation, which can or not be corrected by the VoIP client, and also interfere on the understanding of the speech. As the latency is not usually constant, its variation, also known as jitter, is also considered in a VoIP call quality evaluation. Jitter is the variations in packet inter arrival time. The difference between when the packet is expected and when it is actually received is jitter.

In this experiment, ILBC (27.7kbps) and Speex have a lower bandwidth (31.2kbps) requirement compare to GSM (36.4 kbps). The result shows on Table 30, GSM have a higher packet jitter compare to ILBC and Speex at SNR 10dB. As SNR decrease so does bit rates, it affects the performance of GSM speech codec due to the needs of higher bandwidth requirement. GSM requires more bandwidth thus it is more sensitive to congestion or resource utilization. Increasing SNR will lead to higher bit rate so its reducing the effect of packet jitter and hence the improvement in performance of VoIP.

As shown in Figure 9, as SNR value increases, GSM records much lower packet jitter compare to ILBC and Speex. We can see that ILBC and Speex performance are almost same at any given SNR. Packet jitter is lower than GSM for both speech codec at SNR 10 dB. At 20 dB, packets jitter for all three codecs still in poor level (> 50ms). To obtain excellent quality calls, the receivable mobile stations needed a SNR of 30 dB or higher for optimum network. Thus, all three tested speech codec shows at SNR 30 dB, the packet jitter value is almost negligible or at acceptable level (20ms-50ms).

OPTIMUM NETWORK	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	PACKET JITTER (ms)	GSM	103.647	55.603	36.287
		ILBC	101.808	56.923	36.711
		SPEEX	95.155	58.606	36.785

Table 30: Optimum Network: GSM, ILBC, Speex on Packet Jitter (ms) Comparison

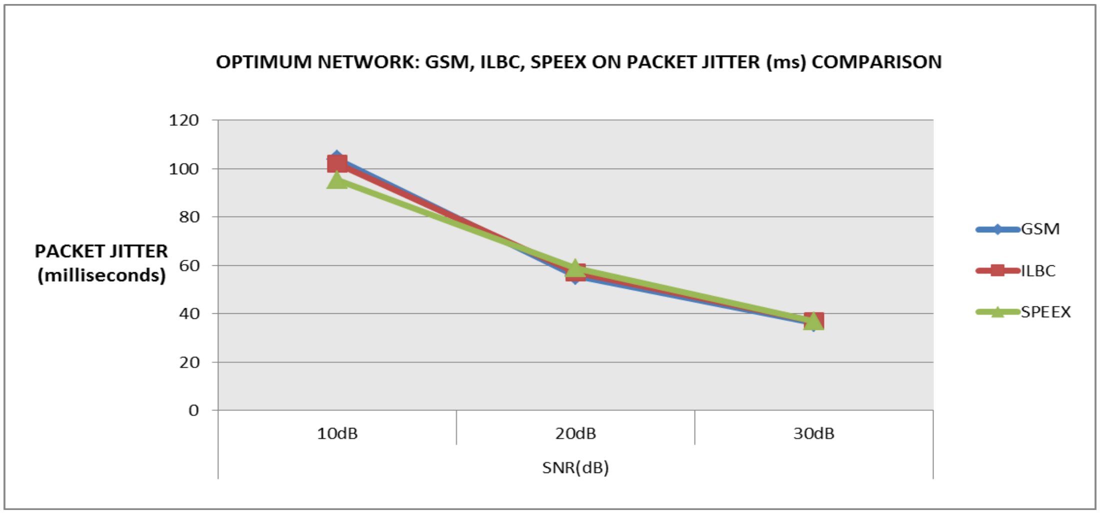

Figure 9: Optimum Network: GSM, ILBC, Speex on Packet Jitter (ms) Comparison

5.2.2 Optimum Network Environment based on Packet Loss Comparison

Packet loss rate can be abstracted as a piece of the conversation lost, which might or not confuse or to obstruct the understanding of the speech. A good behavior of packet loss rate, besides its small value, is a good distribution of packet loss along the entire conversation. Packet loss and delay are caused by both the environment network and the VoIP application itself. The quality obtained is also affected by the nature of losses.

Lower bit-rate codecs shows a larger degradation of speech quality. For high bit-rate codecs, packet loss values up to 10% are acceptable for good quality, whereas for the low bit-rate codecs only 4-5% packet loss rate is acceptable because the number of samples lost within a packet covers more of the speech. [14] Packet loss in the IP network is considered one of the most important factors that cause degradation in the overall voice call quality—packet loss greater than 5% has been shown to have a very detrimental effect on voice quality.[15] However, packet size itself does not have much influence on quality for any of the codecs.
In this experiment, all three low bit rate codecs grouped which is (11-13.33kbps) are tested. We observe that the performance of the codecs is different under packet loss rates. The result shows in Table 31, all three speech codec contain only (4-5%) packet loss on optimum

network condition at 10dB. At 10dB of SNR value, GSM codecs shows the highest packet loss rate at (5.49%). But it drastically decreases the packet loss rate when SNR value increases, means that GSM codec perform well when SNR stable. ILBC codecs also shows the same condition. It records the highest packet loss rate at 10dB, which is (4.2%). Then, the value of packet loss rate drops down to (1.86%) at 20dB. The result proves by GSM codecs recorded almost similar at 30dB SNR value with ILBC and Speex codecs. It proves that all three codecs packet loss rate is acceptable for low bitrate codecs.

As shown in Table 31, at higher SNR reading 20 and 30dB, packet loss rate for all three codecs become decreases. As we can see from the Table 31 at higher SNR reading, (GSM, ILBC, and Speex) codecs reaches to (0-1%) packet loss rate at optimum network condition. We can see clearly from Figure 10, Speex codecs performs (0.44%) packet loss rate at higher SNR given value. It is consider as good level (0-0.5%) of packet loss rate. Speex codecs shows almost same packet loss rate at all tested SNR value, (10-30dB). At 10dB for Speex codecs, the packet loss rate is (1.62%). Then, the value drops a bit to (1.03%) at 20dB compared to GSM and ILBC. It means that Speex codecs voice quality can performs well even at low of SNR value. It is proves by Speex that have less packet loss rate at 10dB. Thus, Speex codecs performs well in packet loss rate. Figure 10 shows clearly of packet loss rate for all codecs; (GSM, ILBC, and Speex) at different SNR value.

OPTIMUM NETWORK	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	PACKET LOSS (%)	GSM	5.49	1.9	0.64
		ILBC	4.2	1.86	0.6
		SPEEX	1.62	1.03	0.44

Table 31: Optimum Network: GSM, ILBC, Speex on Packet Loss (%) Comparison

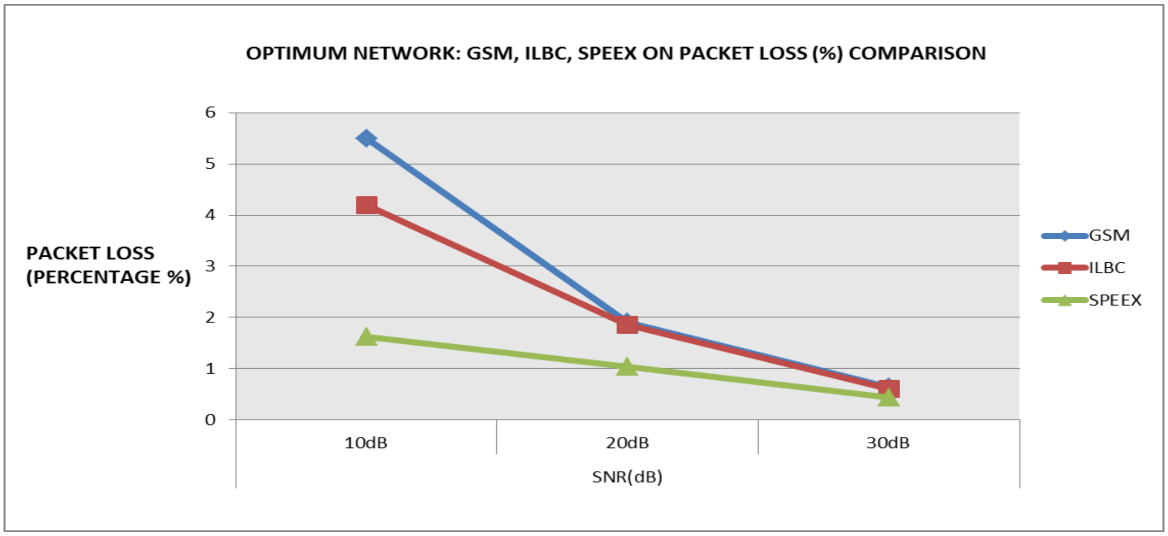

Figure 10: Optimum Network: GSM, ILBC, and Speex on Packet Loss (%) Comparison

5.2.3 Optimum Network Environment based on Mean Opinion Score (MOS) Comparison

In this experiment, we measure our voice quality using MOS. Mean Opinion Score (MOS) technique is the best approach to measure and validate voice quality for three specified tested codecs (GSM, ILBC, and Speex). From the Table 32, it shows that the (Mean Opinion Score) MOS measurement using two way communications of optimum network of VoIP performance over WLAN. MOS gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using codecs. MOS is expressed in one number, from 1 to 5, 1 being the worst and 5 the best.

As shown in Table 32, at 10dB of SNR, MOS value is considered not satisfied by user or poor level on GSM codec. The result shows (2.41) MOS value of GSM codec at 10dB of SNR. ILBC and Speex recorded almost same MOS value for 10dB of SNR which are (3.36) and (3.37) respectively. This is because the value of SNR is very low. When SNR value increases, the value of MOS also increases. It also happened to all codecs. GSM codec also increases in MOS value at 20dB of SNR (3.2). It is considered as fair level and acceptable. At 20dB of SNR value, ILBC and Speex increases in MOS. ILBC record (3.49) while Speex at (3.79) values of MOS. At 30dB of SNR value, all three codecs shows satisfaction to user and acceptable level of MOS on optimum network condition. From three tested codecs on optimum network environment at given SNR value, Speex codecs record the highest MOS value (4.01) at 30dB among two others codecs. It is desirable speech quality satisfied by user and in a good level.

OPTIMUM NETWORK	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	MOS SCORE	GSM	2.41	3.2	3.54
		ILBC	3.36	3.49	3.99
		SPEEX	3.37	3.79	4.01

Table 32: Optimum Network: GSM, ILBC, and Speex on MOS Comparison

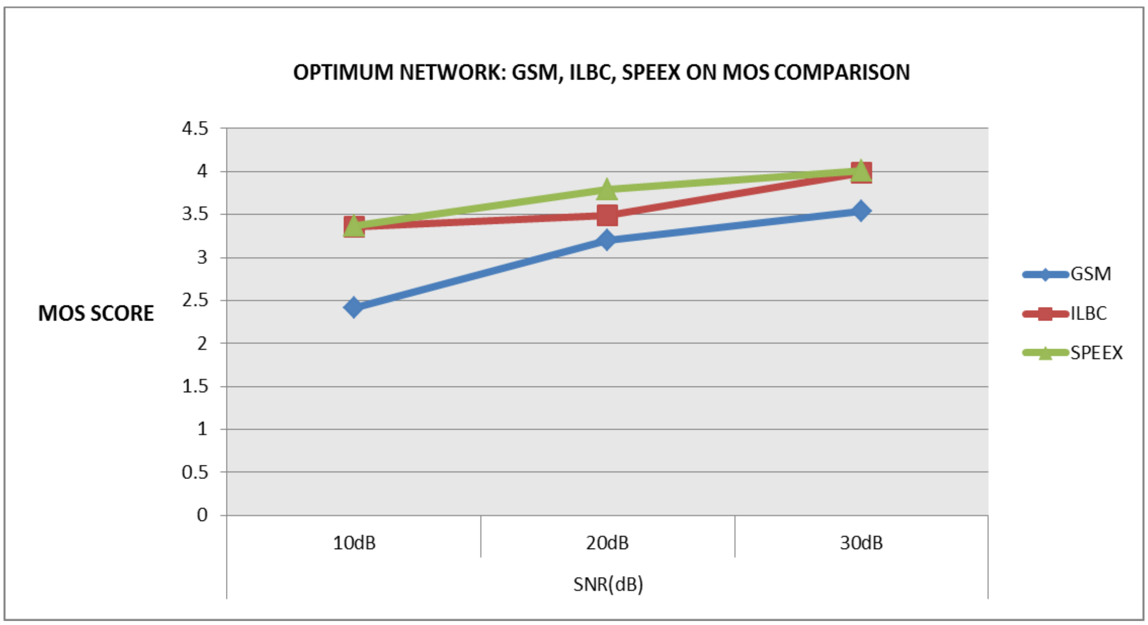

Figure 11: Optimum Network: GSM, ILBC, and Speex on MOS Comparison

5.2.4 Optimum Network Environment based on R-Factor Comparison

For R-factors, alarms trigger when the current value goes below the threshold value. The lower the R-factor, the lower the call quality, so alarms trigger when the R-factor drops below a threshold value. An R-factor between 50 and 60, 60 and 70, 70 and 80, 80 and 90, or 90 and 100 indicates poor, low, medium, high, or best voice quality, respectively. Different approaches have been used to translate these ratings into an overall single measure from which speech quality can be judged. In this experiment, Table 33 shows the (R- Factor) measurement recorded on CommView software using two way communications of optimum network of VoIP performance over WLAN. R-Factor gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using codecs.

As shown in Table 33, at 10dB of SNR, R-Factor value is considered not recommended and poor level by user on GSM codec because it is recorded below 50 which are (46.85). It is because GSM codec cannot perform when SNR is not stable. At 20dB, GSM codec shows improvement on R-Factor reading, which are (62.02). It is described as acceptable level for user to use this codec. The statement proves that GSM codec need stable SNR value to perform better in VOIP. But, for ILBC and Speex codec it is considered acceptable even many users dissatisfied with the codec at 10dB of SNR. It shows (66.1) and (62.75) respectively for ILBC and Speex. This is because the value of SNR is very low. When SNR value increases, the R-Factor also increases. It also happened for another two codecs, ILBC and Speex. At 20 and 30dB of SNR value, all three codecs shows increases value in R-Factor on optimum network condition. From three codecs tested on optimum network environment at given 30dB of SNR value, Speex codecs record the highest R-Factor value (80.17) among two others codecs. It is in good level of R-Factor and user feel satisfied towards the codec. Besides, GSM codec record the lowest quality on R-Factor value at 30dB of SNR value (68.72) but it is still in acceptable level.

OPTIMUM NETWORK	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	R-FACTOR	GSM	46.85	62.02	68.72
		ILBC	66.1	68.35	79.48
		SPEEX	62.75	78.5	80.17

Table 33: Optimum Network: GSM, ILBC, and Speex on R- Factor Comparison

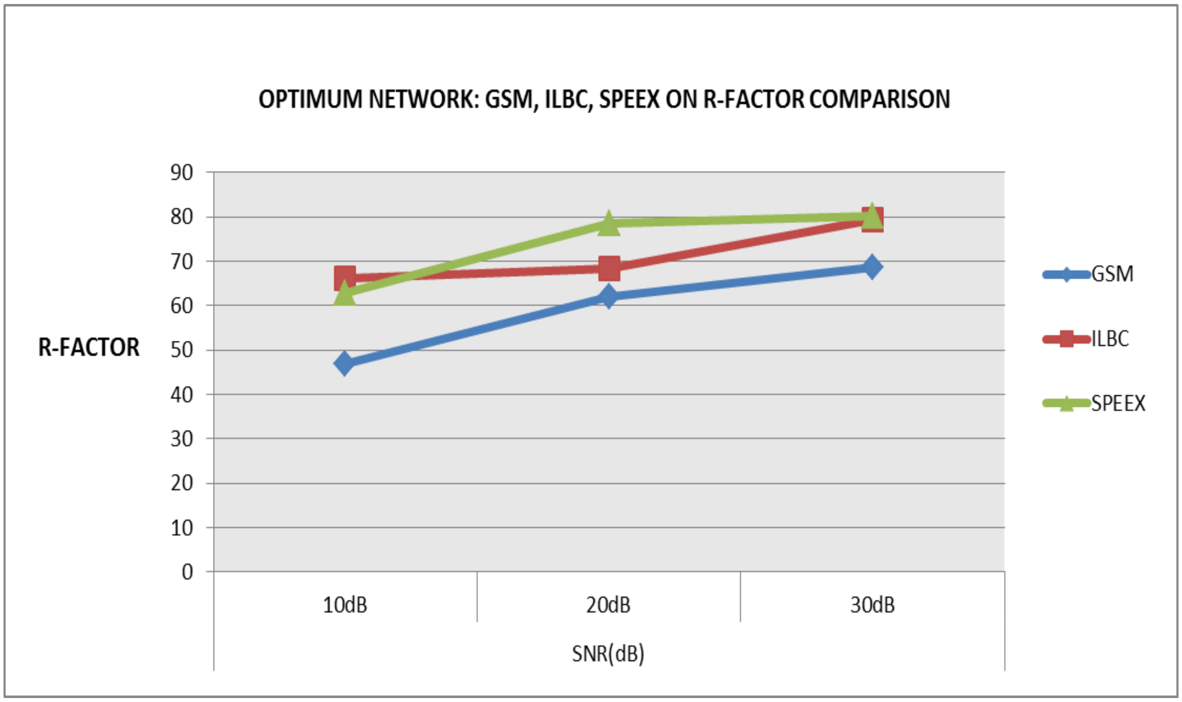

Figure 12: Optimum Network: GSM, ILBC, and Speex on R-Factor Comparison

5.3 Network with Other Traffic (TESTBED 2)

For the next environment, the network of VOIP call session is included with other traffic. In detail, the caller does not use the network for VoIP call session only. This is to test the performance of every each specific speech codec (GSM, ILBC, Speex) used when VOIP call session is establish with others traffic used in the same network. The other traffic is files sharing. In this experiment, the network is no longer dedicated the traffic for VoIP sessions only. 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.

5.3.1 Network with other traffic based on Packet Jitter Comparison

Packet networks exhibit non-ideal behavior that may seriously degrade the performance of real-time communications systems. Particularly, as packets are transmitted from source to destination, they may experience different delays. As a result, packets arrive at the destination with varying delays (between packets) referred to as 'jitter'. Packet delay jitter may result from packets taking different paths to their destination to avoid congested areas or failed links. However, jitter is primarily caused by varying queuing delays encountered by packets at routers (nodes). Network packets compete with other networks traffic at routers. Routers statistically multiplex incoming packets, which results in the varying delay. In a voice over IP (VoIP) application, large inter arrival jitter leads to starvation of the audio play back system. If there is not adequate buffering on the received system these delays will lead to packet loss and a corresponding loss of voice data.

In this experiment, ILBC (27.7kbps) and Speex have a lower bandwidth (31.2kbps) requirement compare to GSM (36.4 kbps). The result shows that GSM have a higher packet jitter compare to ILBC and Speex at SNR 10dB. As SNR decrease so does bit rates, it affects the performance of GSM speech codec due to the needs of higher bandwidth requirement. GSM requires more bandwidth thus it is more sensitive to congestion or resource utilization. Increasing SNR will lead to higher bit rate so its reducing the effect of packet jitter and hence the improvement in performance of VoIP.

As SNR increase, GSM records much lower packet jitter compare to ILBC and Speex. We can see that ILBC and Speex performance are almost balance at any given SNR. Packet jitter is lower than GSM for both speech codec at SNR 10dB. To obtain excellent quality calls, the receivable mobile stations needed a SNR of 30 dB or higher for network with others traffic. At 20 dB, packets jitter for all three codecs still in poor level (> 50ms). In networks that display jitter values that are larger than 50 milliseconds, it is difficult or impossible to play the packets smoothly. Thus, all three tested speech codec shows at SNR 30 dB, the packet jitter value is almost negligible or at acceptable level (20ms-50ms) in this network environment.

NETWORK WITH OTHER TRAFFIC	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	PACKET JITTER (ms)	GSM	115.744	58.855	37.76
		ILBC	109.376	58.968	38.88
		SPEEX	101.372	59.95	39.13

Table 34: Network with other traffic: GSM, ILBC, and Speex on Packet Jitter (ms) Comparison

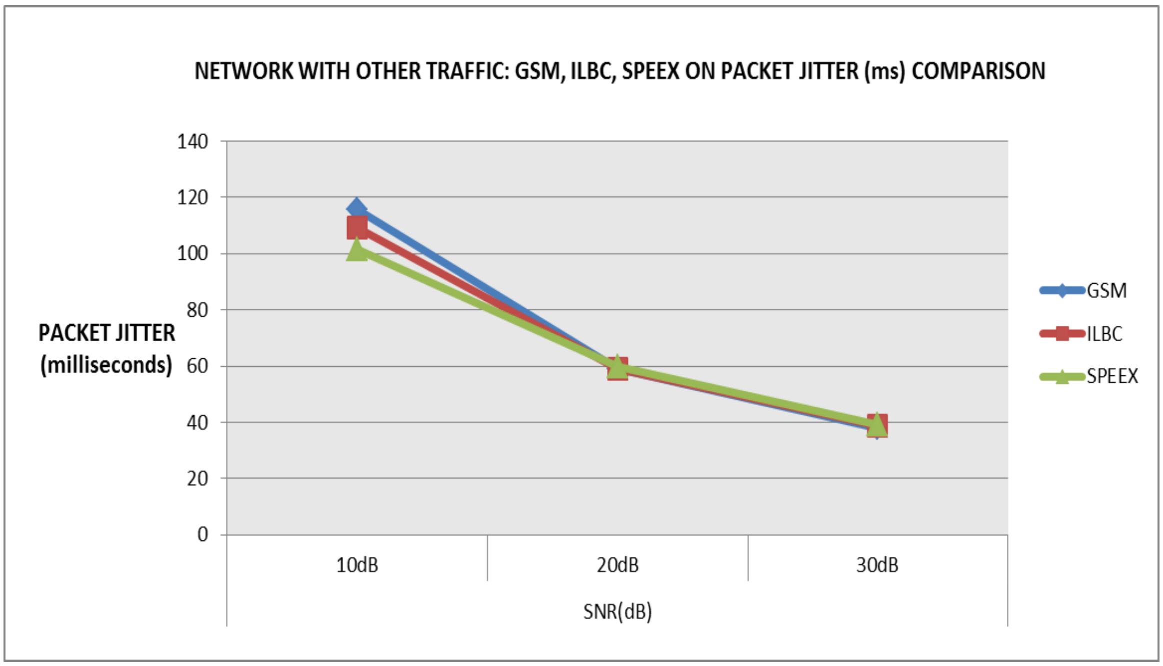

Figure 13: Network with other traffic: GSM, ILBC, and Speex on Packet Jitter (ms) Comparison

5.3.2 Network with other traffic based on Packet Loss Comparison

In VoIP, packets can be discarded for a number of reasons, including network congestion, line errors, and late arrival. Voice over Internet Protocol (VoIP) packet loss occurs when heavy network traffic creates dropped packets, causing portions of conversations to be lost. Packet loss occurs when one or more packets of data travelling across a network fail to reach their intended destination. Call quality deteriorates when this value exceeds 5%. One aspect of network quality is packet loss. This is a quantity that be measured by the CommView software. The quantity is the percentage of packets that are sent from one end of the network connection that do not reach the other end. Networks with a packet loss of more than 5% are not good candidates for VOIP, as there will be dropouts in the audio. Packet loss increases sharply at the point where the network is overloaded with traffic. On a lightly-loaded network, packet loss may be low, but it may become unacceptably high when the number of packets reaches the maximum that the network can accommodate.

In this experiment, packet loss is recorded in percentage for 3 speech codecs, GSM, ILBC, and Speex respectively for a given SNR value on the network with others traffic beside VOIP session. The result shows in Table 34, all three speech codec contain below 5% packet loss on network with other traffic environment at 10dB. It proves that all three codecs packet loss rate is acceptable for low bitrate codecs. GSM codec recorded the highest packet loss (4.3%) at 10dB. It followed by ILBC (3.41%) at 10dB. Then, Speex record the lowest (1.95%) packet loss rate. But, at 30dB of SNR value, GSM codec shows better performance when drops the packet loss rate until it reach (0.95%). It proves that GSM codec can perform well when SNR is stable.

As shown in Table 34 and Figure 14, at higher SNR reading 20 and 30dB, packet loss rate for all three codecs become decreases. As we can see form the table 34, at higher SNR reading, (GSM, ILBC, and Speex) codecs reaches to (0-1%) packet loss rate on network with other traffic condition. From Table 34, Speex is the lowest recorded codecs on packet loss rate until it reach (0.65%) while ILBC recorded the highest packet loss rate at (1.18%) at 30dB. Both of the rate, is considering acceptable packet loss rate.

NETWORK WITH OTHER TRAFFIC	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	PACKET LOSS(%)	GSM	4.3	3.54	0.95
		ILBC	3.41	1.7	1.18
		SPEEX	1.95	1.55	0.65

Table 35: Network with other traffic: GSM, ILBC, and Speex on Packet Loss (%) Comparison

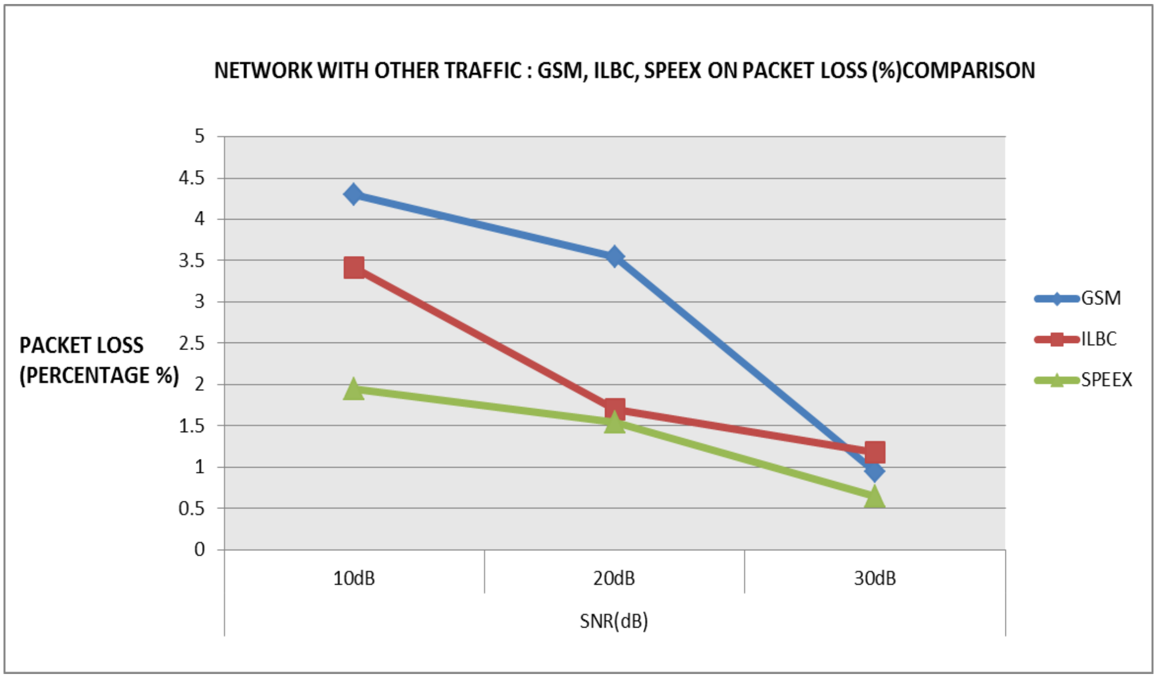

Figure 14: Network with other traffic: GSM, ILBC, and Speex on Packet Loss (%) Comparison

5.3.3 Network with other traffic based on MOS Comparison

The Mean Opinion Score (MOS) is a measurement of overall call quality. Jitter, dropped frames, out of sequence frames, and interval are used to calculate this value. The maximum possible MOS score is 5, which represents two people talking in person to each other. A good

VoIP call has a MOS score of around 4-5. The lower the MOS value, the lower the quality of the VoIP data stream.

In this experiment, we used CommView software to calculate the MOS value. Mean Opinion Score (MOS) technique is the best approach to measure and validate voice quality for three specified tested codecs (GSM, ILBC, and Speex) on network with others traffic. From the Table 35, it shows that the (Mean Opinion Score) MOS performance of 3 codecs recorded using two way communications of network with others traffic on VoIP performance over WLAN. As shown in table 35, at 10dB of SNR, MOS value is considered not recommended by user or poor level on GSM codec. The result shows (2.03) MOS value of GSM codec at 10dB of SNR. Besides, nearly all users dissatisfied for both codecs, ILBC and Speex with 3.13 and 3.15 respectively at 10dB of SNR. This is because the value of SNR is very low. When SNR value increases, the value of MOS also increases. It also happened to all codecs. GSM codec also increases in MOS value at 20dB of SNR (2.36). But, it is still in poor level and not recommended codecs to be used at 20dB of SNR. At 30dB of SNR value, ILBC and Speex codecs shows satisfaction to user and acceptable level of MOS on this network condition. It shows (3.94) and (4.0) values of MOS for both codec respectively. From three codecs tested on network with other traffic environment at given SNR value, Speex codecs record the highest MOS value (4.0) at 30dB among two others codecs. It is desirable speech quality satisfied by user and in considered in a good level.

NETWORK WITH OTHER TRAFFIC	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	MOS SCORE	GSM	2.03	2.36	3.44
		ILBC	3.13	3.47	3.94
		SPEEX	3.15	3.85	4

Table 36: Network with other traffic: GSM, ILBC, and Speex on MOS Comparison

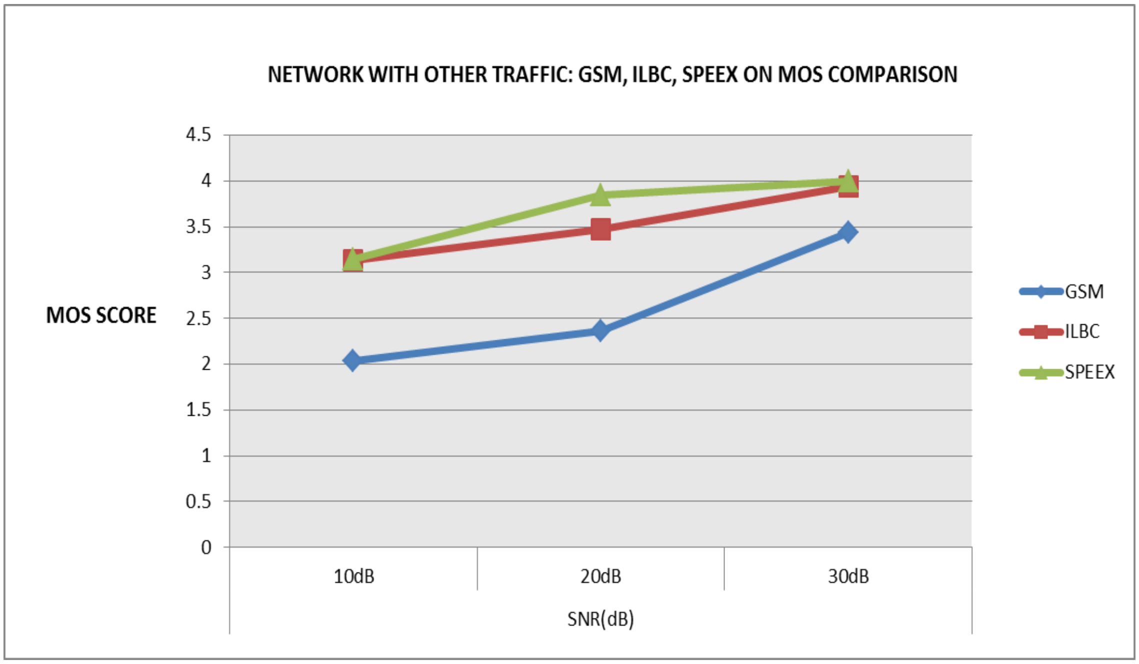

Figure 15: Network with other traffic: GSM, ILBC, and Speex on MOS Comparison

5.3.4 Network with other traffic based on R-Factor Comparison

R-value is a number, or score, that is used to quantitatively express the subjective quality of speech in communications systems, especially digital networks that carry voice over IP (VoIP) traffic, or for which VoIP service is under consideration. The R-value score, which is used in conjunction with voice testing processes, can range from 1 (worst) to 100 (best), with the quality of a test voice signal after it has passed through a network from a source (transmitter) to a destination (receiver).

In this experiment, we use CommView software to give the actual R-Factor performances value of the tested codecs based on given SNR value using two way communications of network with other traffic of VoIP performance over WLAN. As shown in Table 36, at 10dB of SNR, R-Factor value is considered not recommended and poor level by user on GSM codec because it is recorded below 50, which are (39.36). But, for ILBC (66.38) and Speex (60.73) codec is considered acceptable even many users dissatisfied with the codec at 10dB of SNR. This is because the value of SNR is very low. When SNR value increases, the value also increases. It also happened to another two codecs, for ILBC and Speex. At 20 and 30dB of SNR value, all three codecs shows increases values in R-Factor on this network condition. From three codecs tested at given 30dB of SNR value, Speex codecs record the highest R-Factor value (79.57) among two others codecs. It is in good level of R-Factor and user feel satisfied towards the codec. Besides, GSM codec record the lowest quality on R-Factor value at 30dB of SNR value (66.83) but it is still in acceptable level.

NETWORK WITH OTHER TRAFFIC	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	R-FACTOR	GSM	39.36	45.91	66.83
		ILBC	66.38	68.2	77.89
		SPEEX	60.73	75.52	79.57

Table 37: Network with other traffic: GSM, ILBC, and Speex on R-Factor Comparison

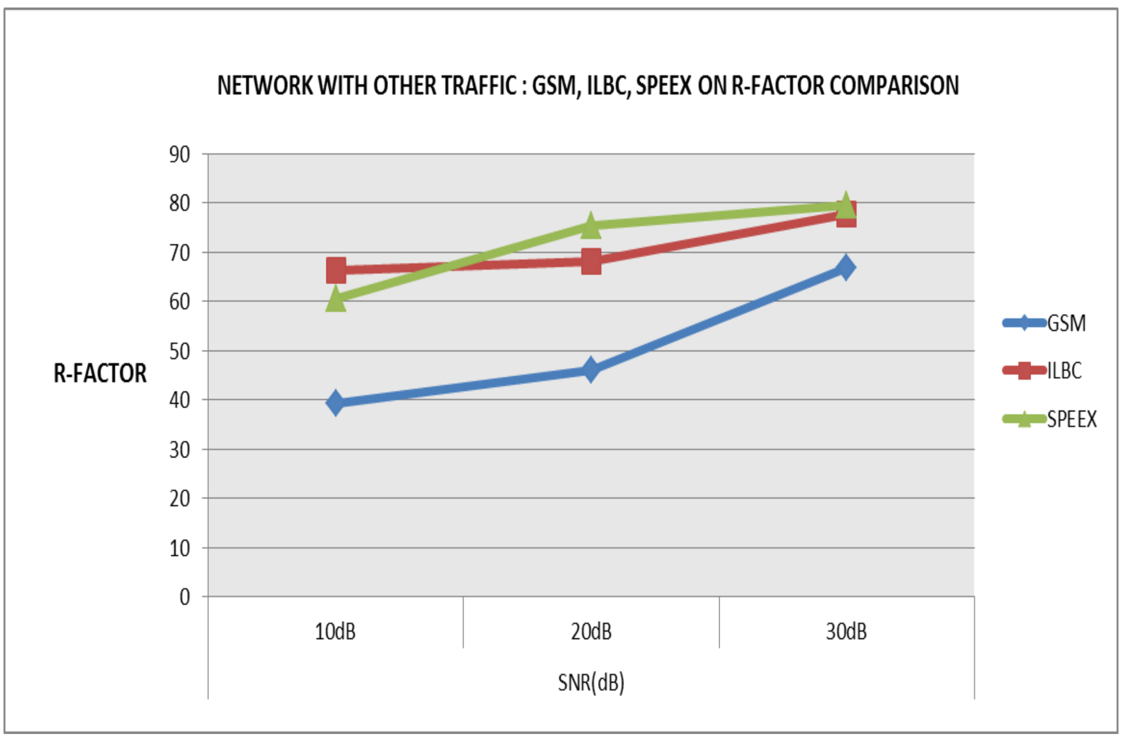

Figure 16: Network with other traffic: GSM, ILBC, and Speex on R-Factor Comparison

5.4 Packet Jitter Comparison of GSM, ILBC, and Speex for Both Types of Network

As we can see from Table 37, packet jitter is higher at 10dB of SNR value for all tested codecs on both types of network; optimum network and network with other traffic. GSM codecs shows higher packet jitter at 10dB than two others codecs for both networks. But, packet jitter slightly decreases when SNR value increases. At higher SNR value, GSM codecs recorded the lowest rate of packet jitter than two others on both network. It is because GSM (36.4kbps) codec have higher bandwidth requirement compared to ILBC (27.7kbps) and Speex (31.2kbps) codec.

Network with other traffic shows highest packet jitter rate at 10dB of SNR value of all codecs. Then, the value drops down when SNR value increases for both types of network. In network with others traffic, packet jitter may be higher than optimum network because of load sharing amongst multiple access links or IP service providers. In order to provide resilience some Enterprise VoIP traffic may be routed over multiple access links to a single IP service provider or diversely routed via several independent IP service providers. This can introduce jitter if the delays across each service or access link differ significantly. Next, it is because load sharing within an IP service. Some IP service providers routinely route traffic over multiple internal routes within their networks in order to improve resilience and provide more even network loading. This introduces jitter resulting from the difference in delay on each route. [17]

From the Table 37, network with other traffic shows highest packet jitter rate at 10dB of SNR value of all codecs. As we can see, GSM codes reach (115.744ms) of packet jitter rate on network with other traffic between (103.647ms) on optimum network. Then, the value drops down when SNR value increases for both types of network. At 20dB, packets jitter for all three codecs still in poor level (> 50ms). All codecs records almost par packet jitter rate at 20dB on both network. Next, all three tested speech codec at SNR 30dB shows the packet jitter value is almost negligible or at acceptable level (20ms-50ms). In summary for packet jitter comparison, in order to obtain excellent quality VOIP calls, the receivable mobile stations needed a SNR of 30dB or higher for both network. Figure 17 shows clearly packet jitter rate for all codecs performances.

PACKET JITTER (ms)	TYPES OF NETWORK	SPEECH CODEC	SNR		
			10dB	20dB	30dB
	OPTIMUM NETWORK	GSM	103.647	55.603	36.287
		ILBC	101.808	56.923	36.711
		SPEEX	95.155	58.606	36.785
	NETWORK WITH OTHER TRAFFIC	GSM	115.744	58.855	37.76
		ILBC	109.376	58.968	38.88
		SPEEX	101.372	59.95	39.13

Table 38: Packet Jitter Comparison of GSM, ILBC, and Speex for Both Types of Network

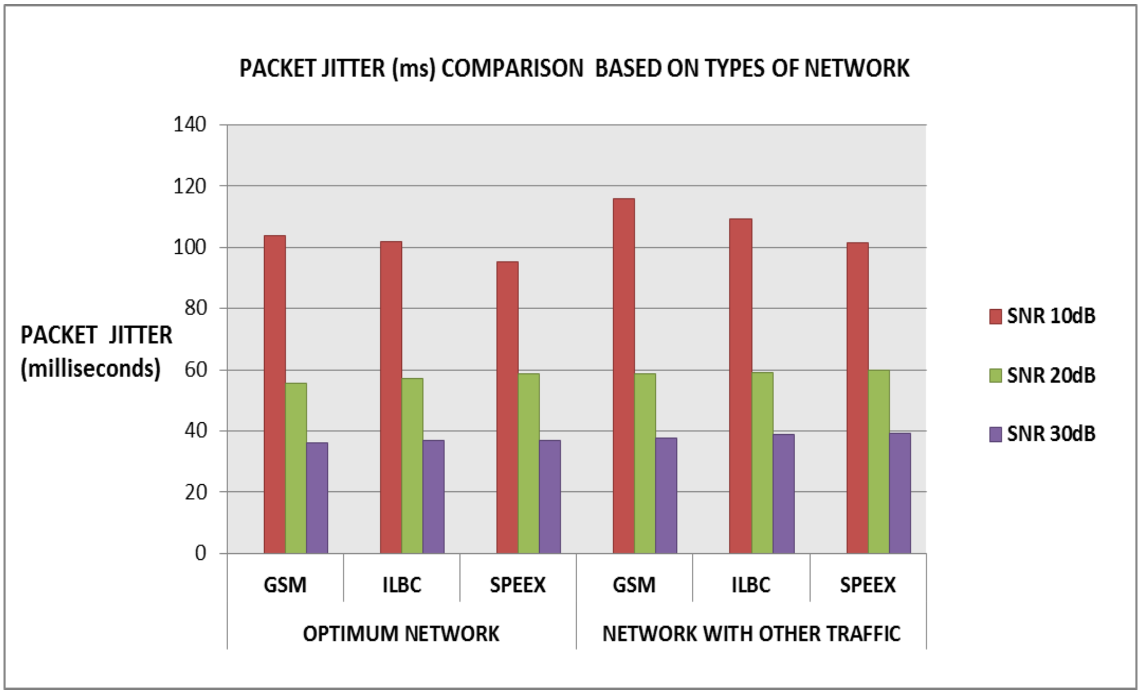

Figure 17: Packet Jitter Comparison of GSM, ILBC, and Speex for Both Types of Network

5.5 Packet Loss (%) Comparison of GSM, ILBC, and Speex for Both Types of Network

Packet loss occurs when one or more packets being transmitted across the network fail to arrive at the destination. This can cause significant problems, especially in Voice over Internet Protocol (VOIP), where information lost cannot be recovered. In some cases, it may be possible to correct for the loss of packets and allow data to be reassembled as it was intended. While overall performance can degrade if excessive retransmissions are required, the data stream stays intact because higher-level sequence numbers are matched to assure that, eventually, every packet is received. The real time communications are based on UDP protocols. This protocol is connectionless and if a packet is lost it is not send again. In addition the packages loss also takes place by discarding packets that do not arrive on time at the receiver.

The result shows in Table 38, at 10dB of SNR value, GSM codecs shows the highest packet loss rate at (5.49%) on optimum network. Then, all three speech codec contain below 5% packet loss on network with other traffic environment at 10dB. GSM codec recorded the highest packet loss (4.3%) at 10dB. The percentage of packet loss rate of network with other traffic was lower than optimum network condition. It proves that others traffic does not influence GSM codec quality, thus effect VOIP call quality at SNR 10dB. At 20dB of SNR value, packet loss rate drastically decrease the value to (1.9%) on optimum network, while (3.54%) on network with others traffic. At 30dB of SNR value, GSM codec shows better performance when drops drastically the packet loss rate (0.64%) on optimum network. But, at 30dB of SNR value of network with other traffic, GSM codec shows better performance when drops the packet loss rate until it reach (0.95%). It proves that GSM codec can perform well when SNR is stable.

ILBC codecs also shows the same condition on 10dB of SNR like GSM codec. It records the second highest packet loss rate at 10dB which is (4.2%) on optimum network. The values decreases to (3.41%) of packet loss rate on network with other traffic at the same given SNR value. It also proves that ILBC codec can perform well when SNR is stable. Then, the result proves by GSM codecs recorded almost par at 30dB SNR value with ILBC and Speex codecs. It proves that all three codecs packet loss rate is acceptable for low bitrate codecs.

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The result shows in Table 38, at 10dB of SNR value, Speex codecs shows the lowest packet loss rate at (1.62%) on optimum network. Speex codec recorded the lowest packet loss (1.95%) at 10dB on network with other traffic. The percentage of packet loss rate of network with other traffic was higher than optimum network condition. At 20dB of SNR value, packet loss rate decrease the value to (1.03%) on optimum network, while (1.55%) on network with others traffic. At 30dB of SNR value, Speex codec shows better performance when drops drastically the packet loss rate at (0.44%) on optimum network. Then, at 30dB of SNR value of network with other traffic, Speex codec recorded (0.65%) packet loss rate. Packet loss rate is considering acceptable packet loss rate for low bitrate codecs. It proves that all three codecs packet loss rate is acceptable for low bitrate codecs. From all three tested codecs, Speex codecs shows stable and balance packet loss rate at all given SNR value.

PACKET LOSS (%)	TYPES OF NETWORK	SPEECH CODEC	SNR		
			10dB	20dB	30dB
	OPTIMUM NETWORK	GSM	5.49	1.9	0.64
		ILBC	4.2	1.86	0.6
		SPEEX	1.62	1.03	0.44
	NETWORK WITH OTHER TRAFFIC	GSM	4.3	3.54	0.95
		ILBC	3.41	1.7	1.18
		SPEEX	1.95	1.55	0.65

Table 39: Packet Loss Comparison of GSM, ILBC, and Speex for Both Types of Network

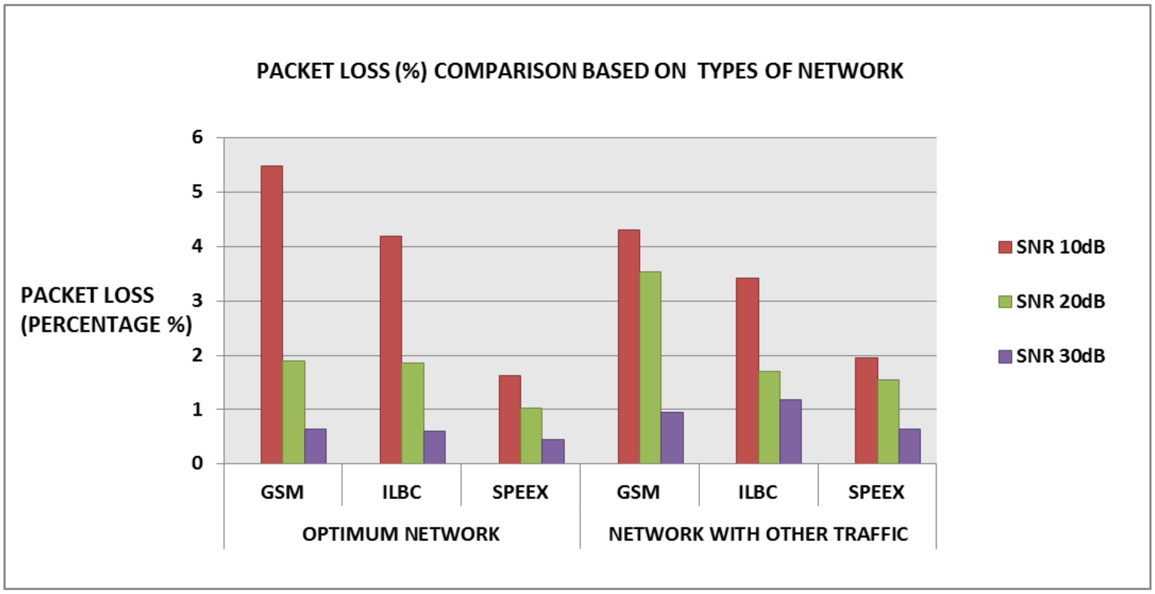


Figure 18: Packet Loss Comparison of GSM, ILBC, and Speex for Both Types of Network

5.6 MOS Comparison of GSM, ILBC, and Speex for Both Types of Network

In this experiment, we measure our voice quality using MOS. Mean Opinion Score (MOS) technique is the best approach to measure and validate voice quality for three specified tested codecs (GSM, ILBC, and Speex). From the Table 39, it shows the (Mean Opinion Score) MOS measurement using two way communications of both types of network of VoIP performance over WLAN. MOS gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using codecs. MOS is expressed in one number, from 1 to 5, 1 being the worst and 5 the best.

As shown in Table 39, at 10dB of SNR, MOS value is considered not satisfied by user or poor level on GSM codec on optimum network and network with other traffic. The result shows (2.41) MOS value of GSM codec at 10dB of SNR on optimum network while (2.03) on network with other traffic. Both MOS value is considered not recommended by user or poor level on GSM codec. This is because the value of SNR is very low. When SNR value increases, the value of MOS also increases. It proves that GSM codec does not perform when SNR is not stable even it is allocated for VOIP session only. GSM codec increases in MOS

(3.2) value at 20dB of SNR on optimum network. It is considered as fair level and acceptable value of MOS. But, GSM codec shows MOS value (2.36) at 20dB of SNR on network with other traffic. It is poor level and not recommended codecs to be used at 20dB of SNR when file sharing is occur or when network is used by others traffic. It proves that GSM codec need stable SNR value when establish VOIP session with others traffic. At 30dB of SNR value, all three codecs shows satisfaction to user and acceptable level of MOS on optimum network condition. GSM codec also produce the same result MOS value (3.54). On network with other traffic, MOS value is (3.44).

ILBC and Speex recorded almost same MOS value for 10dB of SNR which are (3.36) and (3.37) respectively on optimum network. On network with other traffic, values of MOS value drops to (3.13) and (3.15) for ILBC and Speex and nearly all users dissatisfied for both codecs. At 20dB of SNR value, ILBC and Speex increases in MOS. ILBC record (3.49) while Speex at (3.79) values of MOS. On network with other traffic, values of MOS value drops to (3.47) but increase for Speex codecs (3.85). At 30dB of SNR value, all three codecs shows satisfaction to user and acceptable level of MOS on optimum network condition and network with other traffic. At 30dB, all three codecs shows satisfaction to user and acceptable level of MOS on optimum network condition and network with other traffic. From three tested codecs on optimum network environment at given SNR value, Speex codecs record the highest MOS value (4.01) at 30dB among two others codecs on optimum network and (4.0) on network with traffic. It is desirable speech quality satisfied by user and in a good level.

	TYPES OF NETWORK	SPEECH CODEC	SNR		
			10dB	20dB	30dB
MOS SCORE	OPTIMUM NETWORK	GSM	2.41	3.2	3.54
		ILBC	3.36	3.49	3.99
		SPEEX	3.37	3.79	4.01
	NETWORK WITH OTHER TRAFFIC	GSM	2.03	2.36	3.44
		ILBC	3.13	3.47	3.94
		SPEEX	3.15	3.85	4

Table 40: MOS Comparison of GSM, ILBC, and Speex for Both Types of Network

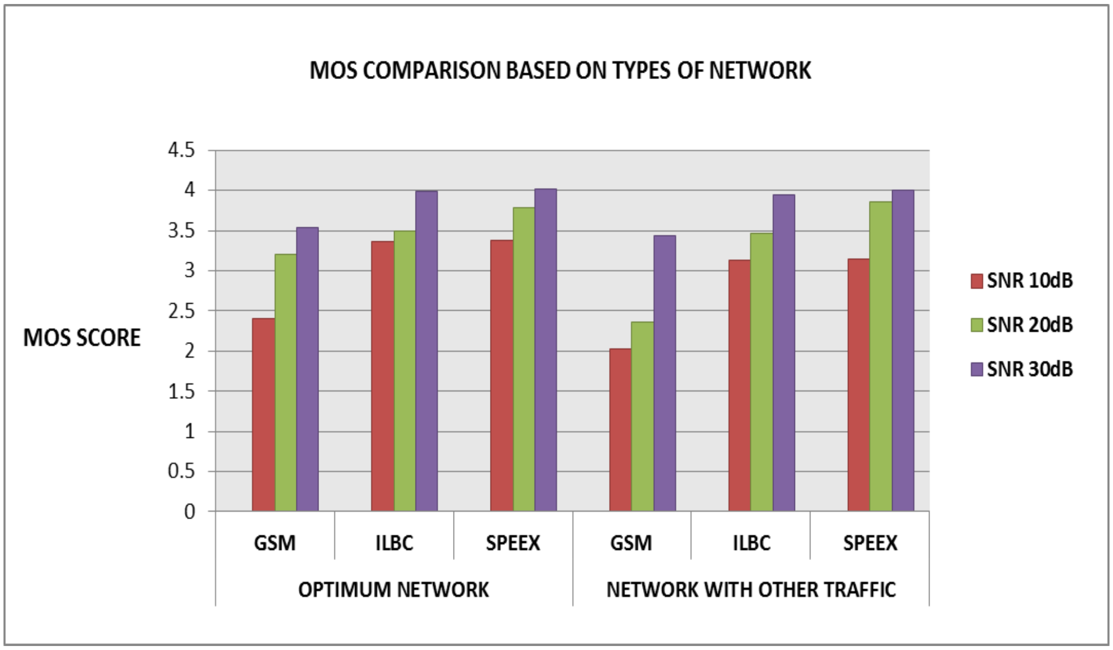


Figure 19: MOS Comparison of GSM, ILBC, and Speex for Both Types of Network

5.7 R-Factor Comparison of GSM, ILBC, and Speex for Both Types of Network

An R-factor between 50 and 60, 60 and 70, 70 and 80, 80 and 90, or 90 and 100 indicates poor, low, medium, high, or best voice quality, respectively. Different approaches have been used to translate these ratings into an overall single measure from which speech quality can be judged.

In this experiment, Table 40 shows the (R- Factor) measurement recorded on CommView software using two way communications of both types of network of VoIP performance over WLAN. As shown in Table 40, at 10dB of SNR, R-Factor value is considered not recommended and poor level by user on GSM codec because it is recorded below 50 which are (46.85) on optimum network condition. Then, R-Factor value for network with traffic also is considered not recommended and poor level by user on GSM codec which are (39.36). Network with other traffic gives effect to R-Factor value. The value shows differences. It means that GSM codec cannot perform well at lower SNR value mostly when the network for VOIP is allocated to others traffic too. GSM codec shows improvement on R-Factor readings, which are (62.02). It is described as acceptable level for user to use this codec. But, the value for network with other traffic condition, still in poor level (below 50) at 20dB of SNR value (45.91). It proves that GSM codec need higher SNR value especially on the network that is not allocated for VOIP session only. Moreover, GSM codec record the lowest quality on R-Factor value at 30dB of SNR value (68.72) than two others codec but it is still in acceptable level on optimum network condition. For network with other traffic, the MOS value almost par with optimum network, (66.83).

But, for ILBC and Speex codec it is considered acceptable even many users dissatisfied with the codec at 10dB of SNR. It shows (66.1) and (62.75) respectively for ILBC and Speex on optimum network condition. As we can see from Table 40, ILBC reach (66.38) and Speex (60.73) R-Factor value. Both codec is considered acceptable even many users dissatisfied with the codec at 10dB of SNR on network with others traffic. At 20 and 30dB of SNR value, all three codecs shows increases value in R-Factor on optimum network condition. From three codecs tested on optimum network environment at given 30dB of SNR value, Speex codecs record the highest R-Factor value (80.17) among two others codecs. Besides, for network with other traffic at given 30dB of SNR value.

In summary, Speex codecs record the highest R-Factor value among two others codecs for both types of network at 30dB. It is in good level of R-Factor and user feel satisfied towards the codec. Moreover, GSM codec record the lowest quality on R-Factor value at 30dB of SNR value on two environments but it is still in acceptable level. It proves that stable SNR value gives greater performances to VOIP quality call on WLAN environment.

	TYPES OF NETWORK	SPEECH CODEC	SNR		
			10dB	20dB	30dB
R-FACTOR	OPTIMUM NETWORK	GSM	46.85	62.02	68.72
		ILBC	66.1	68.35	79.48
		SPEEX	62.75	78.5	80.17
	NETWORK WITH OTHER TRAFFIC	GSM	39.36	45.91	66.83
		ILBC	66.38	68.2	77.89
		SPEEX	60.73	75.52	79.57

Table 41: R-Factor Comparison of GSM, ILBC, and Speex for Both Types of Network

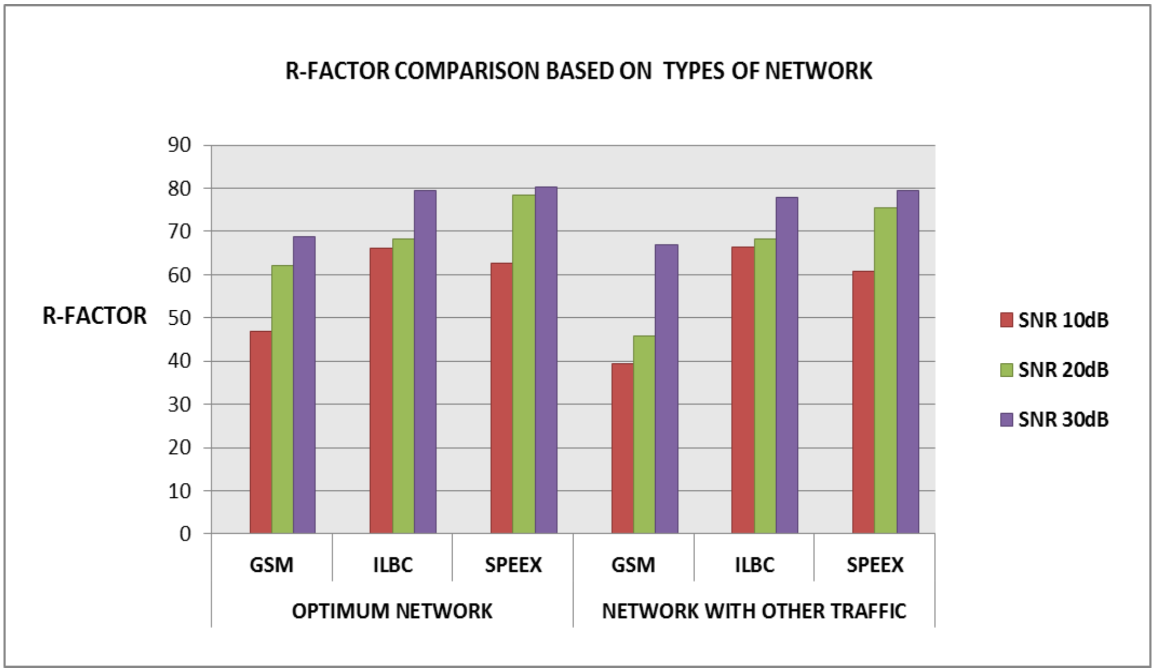


Figure 20: R-Factor Comparison of GSM, ILBC, and Speex for Both Types of Network

CHAPTER 6

CONCLUSION

6.0 CONCLUSION

The growing widespread use of VoIP and its extension to wireless local area network has led to an increased interest in the study of voice over wireless LANs. Users clearly defined the decisive factor in selecting VoIP applications, which is the speech quality. The choice of speech codec has a major influence in the perceived quality. The combination factors of speech codec quality and WLAN signal strength by referring to SNR value will clearly give a substantial impact on voice quality. This paper focuses on three speech codecs; GSM, ILBC and Speex. In the 802.11n wireless environment, with predetermined SNR value (10, 20 and 30dB); the performance of (GSM, ILBC and SPEEX) speech codec was analyzed. VoIP QOS (Quality of Services) ; Packet Jitter, Packet Loss, MOS, and R-Factor became the benchmark in evaluating the effect of speech codec to VoIP performance in wireless 802.11n local area networks (LAN).

When analyzing the effect of speech codec to performance of VOIP WLAN, few factors of causes were identified clearly. Speech codec properties such as bit rate and bandwidth requirements in order to convey voice packet from one client to another client was taken into consideration. The relationship between SNR value, 802.11n wireless performance and capacity and data rate certainly gave an impact to perceived speech quality. Also, an association between all Quality of Services (QOS) and speech quality was analyzed thoroughly. Theoretical knowledge of an acceptable packet jitter, and packet loss rate, then the level of MOS value and R-Factor for VoIP call was needed for us to discuss the impact of every speech codec to VoIP performances. VoIP QOS such as Packet Jitter (ms), Packet Loss (%), MOS and R-Factor are used and analyzed in order to make a comparison of speech quality of those three speech codecs in wireless LAN 802.11n environment.

Result shows that at lower SNR, all tested codec shows lowest performance on QOS parameter and effect quality of the VOIP call. In highest SNR value, all codec shows better performance on QOS parameter, hence give well performance on quality of VOIP. In summary, based on the experiment result, Speex codec shows better performance in VOIP than GSM and ILBC. It follows by ILBC, and then GSM codec based on SNR value and QOS parameter testing result. Speex codecs is lowest on packet jitter and packet loss rate, highest on MOS value and R-Factor on both network environment; optimum network and network with others traffic. Thus, Speex is the best speech codec for the predefined wireless mesh network compared to ILBC and GSM speech codec.

6.1 Recommendation on GSM, ILBC, and Speex Performance affect by SNR value

The SNR of an access point signal, measured at the user device, decreases as range to the user increases because the applicable free space loss between the user and the access point reduces signal level. The same goes for the signals propagating from the user device to the access point. SNR directly impacts the performance of a wireless LAN connection (WLAN). At higher SNR value means that the signal strength is stronger in relation to the noise levels, which allows higher data rates and fewer retransmissions – all of which offers better throughput. Of course the opposite is also true. A lower SNR requires wireless LAN devices to operate at lower data rates, which decreases throughput. A SNR of 30 dB, for example, may allow an 802.11g client and access point to communicate at 24 Mbps; whereas, a SNR of 15 dB may only provide for 6 Mbps.

Based on this performance testing result, each codecs; GSM, ILBC, and Speex shows better result in higher SNR value. It were proves by the value of QOS Parameter; Packet Jitter (ms), Packet Loss (%), MOS and R-Factor on both types of network. We recommended using around 20dB as the minimum SNR for defining the range boundary of each 802.11b/g access point. That ensures a constant association with fairly good performance when performing VOIP session and use the network with others traffic. Keep in mind that 802.11n may require different range boundary definitions. To deploy voice over a wireless LAN with others traffic, then likely need a higher SNR.

For example, Cisco recommends 25 dB for their wireless voice telephony systems. Also, a larger margin (i.e., higher SNR), may be necessary in some venues. Keep in mind that the corresponding level of performance only occurs at the boundary of each access point. Users associating with access points at closer range will have higher SNR and better performance. [16]

7.0 REFERENCES

1. (2012, March 5) Retrieved from http://en.wikipedia.org/wiki/Comparison_of_VoIP_software

2. (2012, March 5) Retrieved from http://www.counterpath.com/eyebeam.html

3. (2012, March 5) Retrieved from http://en.wikipedia.org/wiki/Ekiga

4. Beuran,R. (2006). "*VoIP over Wireless LAN Survey*" Retrieved (January 15, 2013)

5. Jelinek,M. Vaillancourt,T. "*G.718: A New Embedded Speech and Audio Coding Standard with High Resilience to Error-Prone Transmission Channels*" Retrieved [January 15, 2013]

6. K. Jarvinen, J. Vainio, P. Kapanen, T. Honkanen, P. Haavisto "*GSM Enhanced Full Rate Speech Codec*" Retrieved (January 15, 2013)

7. Karapanzatis,S. Pavlidou,F.N. "*VoIP: A comprehensive survey on a promising technology*" Retrieved (January 15, 2013)

8. E. Ekudden, R. Hagen, I. Johansson, and J. Svedberg. "*The Adaptive Multi-Rate Speech Coder*" Retrieved (March 25, 2013)

9. (2012, March 5) Retrieved from http://www.webopedia.com/TERM/8/802_11.html

10. Luis, J.G.Moreira, Mota, Edson, N Silva Jr., Carvalho, L. Quiroz,S and Hoene,C. "*Voice over Wireless Mesh Networks: A Case Study in the Brazilian Amazon Region during the Rainy Season*" Retrieved (March 24, 2013)

11. Ramo, A. "*Voice Quality Evolution of Various Codec*" Retrieved (January 14, 2013)

12. Alias, M. Loon, O.L. "*Performance of Voice Codec Over IP (VoIP) Over a Wireless LAN (WLAN) for Different Audio/Voice Codecs*" Retrieved (January 30, 2013)

13. (2013, November 14) Retrieved from http://www.voiptroubleshooter.com/problems/packetloss.html

14. Mehta, P.Sheveni "Comparative *Study of Technique to minimize packet loss in VOIP*" Retrieved (November 12, 2013)

15. Assem.H., Adely, M, .Jenningsy.B. Malone.D. Dunnez, J., and Sullivanz, O "*A Generic Algorithm for Mid-call Audio Codec Switching*" Retrieved (November 12, 2013)

16. (2013, November 14) Retrieved from http://www.wireless-nets.com/resources/tutorials/define_SNR_values.html

17. (2013, November 14) Retrieved from http://www.voiptroubleshooter.com/indepth/jittersources.html

APPENDIX

8.0 Appendix for Result of Experimental

RAW DATA FOR THREE CODECS ON OPTIMUM NETWORK AND NETWORK WITH OTHER TRAFFIC

	QOS PARAMETER	SNR 10dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	96.68	97.8	97.88	99.68	111.49	111.45	113.47	95.5	115.8	96.72	103.647
	PACKET LOSS	4.4	5.1	5.2	5.4	6	6	6.2	4.2	8	4.4	5.49
	MOS SCORE	2.6	2.5	2.4	2.4	2.3	2.3	2.3	2.6	2.1	2.6	2.41
	R-FACTOR	50.3	47.8	47.5	46.8	45.1	45.1	44.6	51.1	39.8	50.4	46.85
	QOS PARAMETER	SNR 20dB										AVERAGE
OPTIMUM NETWORK (GSM)		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	58.4	59.42	58.14	57.14	50.96	55.67	58.3	57.8	49	51.2	55.603
	PACKET LOSS	3.8	3	1.4	1.4	1.2	2.7	0.7	1.8	1.5	1.5	1.9
	MOS SCORE	2.8	3	3.3	3.3	3.3	3	3.5	3.2	3.3	3.3	3.2
	R-FACTOR	54.9	57.8	63.9	64.2	64.9	57.4	68.3	61.9	63.7	63.2	62.02
	QOS PARAMETER	SNR 30dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	35.67	36	37.1	30.8	37.7	36.1	36.3	37.9	37.1	38.2	36.287
	PACKET LOSS	0.9	0.8	0.7	0.3	0.7	0.2	0.7	0.6	0.6	0.9	0.64
	MOS SCORE	3.5	3.5	3.5	3.6	3.5	3.7	3.5	3.5	3.6	3.5	3.54
	R-FACTOR	67.1	67.9	68.2	70.7	68.2	71.9	68	69	69.2	67	68.72

	QOS PARAMETER	SNR 10dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	96.68	76.99	97.88	99.68	95.5	111.45	112.2	95.5	112.22	119.98	101.808
	PACKET LOSS	3.7	2.9	4.3	4.1	3.9	4.5	4.4	4.6	5	4.6	4.2
	MOS SCORE	3.6	3.6	3.6	3.6	3.6	3.5	3.4	3.1	3	2.6	3.36
	R-FACTOR	71.4	73.3	70.1	70.6	71	68.1	65.9	60.7	58.8	51.1	66.1
	QOS PARAMETER	SNR 20dB										AVERAGE
OPTIMUM NETWORK (ILBC)		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	66.23	68	60.4	56.8	49.4	48	54.9	53.2	69	43.3	56.923
	PACKET LOSS	2.2	2.1	2.2	2	2	2.2	2	1.1	2.4	0.4	1.86
	MOS SCORE	3.2	3.1	3.2	3.3	3.8	3.8	3.7	3.9	2.9	4	3.49
	R-FACTOR	61.3	60.3	61.7	63.1	74.3	75.1	73	77.9	56.7	80.1	68.35
	QOS PARAMETER	SNR 30dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	36	48.77	35.55	33.09	32.99	43.87	34.88	33.88	33.88	34.2	36.711
	PACKET LOST	0.8	2	0.5	0.3	0.1	1.3	0.4	0.2	0.2	0.2	0.6
	MOS SCORE	4	3.9	4	4	4.1	3.9	4	4	4	4	3.99
	R-FACTOR	78.9	75.7	79.7	80.4	80.8	77.6	80.1	80.5	80.6	80.5	79.48

OPTIMUM NETWORK (SPEEX)	QOS PARAMETER	SNR 10dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	95.3	96.86	95.55	97.34	95.57	94.82	98.88	92.33	94	90.9	95.155
	PACKET LOST	1.5	1.5	1.4	2.6	1.6	2	1.6	1.3	1.4	1.3	1.62
	MOS SCORE	3.3	3.4	3.5	3.1	3.2	3.2	3.3	3.7	3.4	3.6	3.37
	R-FACTOR	62.5	62.8	63.2	59.7	62.1	61	62.2	66.6	63.2	64.2	62.75
	QOS PARAMETER	SNR 20dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	67.93	58.8	56.8	56.8	55.8	60.68	54.9	59	57.7	57.65	58.606
	PACKET LOST	1	1.1	1.1	1.1	1	0.9	1.2	1.1	0.8	1	1.03
	MOS SCORE	3.8	3.7	3.7	3.7	3.8	3.9	3.8	3.7	4	3.8	3.79
	R-FACTOR	78	78.4	77.9	77.9	79	79.1	78.8	77.6	79.3	79	78.5
	QOS PARAMETER	SNR 30dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	43.87	36	35.68	34.94	35.68	33.33	36.3	36.75	37.1	38.2	36.785
	PACKET LOST	0.3	0.5	0.4	0.1	0.5	0.3	0.3	0.7	0.5	0.8	0.44
	MOS SCORE	4	4	4	4.1	4	4	4	4	4	4	4.01
	R-FACTOR	80.4	80.3	80	80.8	79.9	80.2	80.3	80.1	79.7	80	80.17

NETWORK WITH OTHER TRAFFIC (GSM)	QOS PARAMETER	SNR 10db										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	110.23	111.23	123.44	99.68	111.49	111.45	113.47	122.34	134.22	119.89	115.744
	PACKET LOST	4.6	4	4.9	4.3	3.9	3.8	4	4.3	4.3	4.9	4.3
	MOS SCORE	2.1	2.1	2	1.9	2.1	2.2	2.1	1.9	1.9	2	2.03
	R-FACTOR	40.8	39.9	37.8	36.2	40.8	42.4	41.7	37.4	37.1	39.5	39.36
	QOS PARAMETER	SNR 20dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	58.9	59.5	58.14	59.3	59.9	59.44	58.3	55.99	58.98	60.1	58.855
	PACKET LOST	3.5	3.6	3.2	3.6	3.4	3.9	3	3.7	3.9	3.6	3.54
	MOS SCORE	2.2	2.2	2.5	2.4	2.4	2.5	2.3	2.2	2.5	2.4	2.36
	R-FACTOR	43.7	43.5	47.6	46.4	46.9	48.4	45	43.2	48.1	46.3	45.91
	QOS PARAMETER	SNR 30dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	37.8	36	39.6	43.3	37.7	36.1	36.3	37.9	35.9	37	37.76
	PACKET LOST	0.5	0.9	2.1	0.6	1	1.1	0.9	1	0.8	0.6	0.95
	MOS SCORE	3.6	3.4	3.1	3.6	3.4	3.4	3.5	3.4	3.5	3.5	3.44
	R-FACTOR	69.7	66.8	60.3	69.3	66.5	66	67.1	66.2	67.6	68.8	66.83

NETWORK WITH OTHER TRAFFIC (ILBC)	QOS PARAMETER	SNR 10dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	120.01	123.3	120.34	90.68	90.56	89.9	122.89	90.65	121.03	124.4	109.376
	PACKET LOST	4	3.7	4	2.9	2.9	2	4.2	2.9	3.2	4.3	3.41
	MOS SCORE	3.1	3.5	3.1	3.1	3.1	3.5	3	3	3	2.9	3.13
	R-FACTOR	70.6	71.1	70.1	70	71	68.1	65.9	60.7	58.8	57.5	66.38
	QOS PARAMETER	SNR 20dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	65.55	67.77	58.14	63.67	55.67	55.34	55.66	56.68	58.98	52.22	58.968
	PACKET LOST	2.9	3	2.9	2.6	1.1	1	1	1.1	1	0.4	1.7
	MOS SCORE	3.2	3.1	3.2	3.3	3.5	3.6	3.7	3.7	3.4	4	3.47
	R-FACTOR	61.3	60.3	61	63.1	69	71	73	73.4	70	79.9	68.2
	QOS PARAMETER	SNR 30dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	39.91	36.77	42.2	43.38	37.7	36.99	36.3	36.89	36.66	42	38.88
	PACKET LOST	1.5	0.9	1.6	1.6	1.4	1	0.3	0.3	1.2	2	1.18
	MOS SCORE	3.9	4	3.9	3.9	3.9	4	4	4	3.9	3.9	3.94
	R-FACTOR	77	78.7	76.7	76.6	77.3	78.3	80.3	80.4	77.9	75.7	77.89

NETWORK WITH OTHER TRAFFIC (SPEEX)	QOS PARAMETER	SNR 10dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	95.3	94	95.55	112.99	95.57	115	98.88	90.23	94	122.2	101.372
	PACKET LOST	1.5	1.5	1.4	2.6	1.6	2.7	1.6	1.3	1.6	3.7	1.95
	MOS SCORE	3.3	3.4	3.5	3.1	3.2	2.6	3.2	3.7	3.2	2.3	3.15
	R-FACTOR	62.5	61	62	58.2	61.9	58	60	63.7	61	59	60.73
	QOS PARAMETER	SNR 20dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	67.93	68.08	56.8	56.8	55.8	60.68	54.9	67.98	56.08	54.45	59.95
	PACKET LOST	1	1.1	1.5	1.4	1.5	1.7	0.6	1.7	2	3	1.55
	MOS SCORE	4	4	3.9	3.9	3.9	3.8	4	3.7	3.8	3.5	3.85
	R-FACTOR	78.4	78.1	76.9	77.3	76.9	73.5	79.3	72	75.7	67.1	75.52
	QOS PARAMETER	SNR 30dB										AVERAGE
		1	2	3	4	5	6	7	8	9	10	
	PACKET JITTER	34.89	38.98	45.55	34.94	39.99	43.89	38.98	38.78	37.1	38.2	39.13
	PACKET LOST	0.8	0.6	0.9	1	0.7	0.8	0.6	0.4	0.3	0.4	0.65
	MOS SCORE	4	4	4	4	4	4	4	4	4	4	4
	R-FACTOR	79	79.5	80.6	78.3	79.3	78.8	79.5	80.1	80.5	80.1	79.57

DATABASE ON OPTIMUM NETWORK AND NETWORK WITH OTHER TRAFFIC

	QOS PARAMETER	SNR		
		10dB	20dB	30dB
AVERAGE OPTIMUM NETWORK (GSM)	PACKET JITTER	103.647	55.603	36.287
	PACKET LOST	5.49	1.9	0.64
	MOS SCORE	2.41	3.2	3.54
	R-FACTOR	46.85	62.02	68.72

	QOS PARAMETER	SNR		
		10dB	20dB	30dB
AVERAGE OPTIMUM NETWORK (ILBC)	PACKET JITTER	101.808	56.923	36.711
	PACKET LOST	4.2	1.86	0.6
	MOS SCORE	3.36	3.49	3.99
	R-FACTOR	66.1	68.35	79.48

	QOS PARAMETER	SNR		
		10dB	20dB	30dB
AVERAGE OPTIMUM NETWORK (SPEEX)	PACKET JITTER	95.155	58.606	36.785
	PACKET LOST	1.62	1.03	0.44
	MOS SCORE	3.37	3.79	4.01
	R-FACTOR	62.75	78.5	80.17

	QOS PARAMETER	SNR		
		10dB	20dB	30dB
AVERAGE NETWORK WITH OTHER TRAFFIC (GSM)	PACKET JITTER	115.744	58.855	37.76
	PACKET LOST	4.3	3.54	0.95
	MOS SCORE	2.03	2.36	3.44
	R-FACTOR	39.36	45.91	66.83

	QOS PARAMETER	SNR		
		10dB	20dB	30dB
AVERAGE NETWORK WITH OTHER TRAFFIC (ILBC)	PACKET JITTER	109.376	58.968	38.88
	PACKET LOST	3.41	1.7	1.18
	MOS SCORE	3.13	3.47	3.94
	R-FACTOR	66.38	68.2	77.89

AVERAGE NETWORK WITH OTHER TRAFFIC (SPEEX)	QOS PARAMETER	SNR		
		10dB	20dB	30dB
	PACKET JITTER	101.372	59.95	39.13
	PACKET LOST	1.95	1.55	0.65
	MOS SCORE	3.15	3.85	4
	R-FACTOR	60.73	75.52	79.57

DATABASE QUALITY OF SERVICE (QoS) PARAMETER BASED ON BOTH TYPES OF NETWORK

PACKET JITTER (ms)	TYPES OF NETWORK	SPEECH CODEC	SNR		
			10dB	20dB	30dB
	OPTIMUM NETWORK	GSM	103.647	55.603	36.287
		ILBC	101.808	56.923	36.711
		SPEEX	95.155	58.606	36.785
	NETWORK WITH OTHER TRAFFIC	GSM	115.744	58.855	37.76
		ILBC	109.376	58.968	38.88
		SPEEX	101.372	59.95	39.13

PACKET LOSS (%)	TYPES OF NETWORK	SPEECH CODEC	SNR		
			10dB	20dB	30dB
	OPTIMUM NETWORK	GSM	5.49	1.9	0.64
		ILBC	4.2	1.86	0.6
		SPEEX	1.62	1.03	0.44
	NETWORK WITH OTHER TRAFFIC	GSM	4.3	3.54	0.95
		ILBC	3.41	1.7	1.18
		SPEEX	1.95	1.55	0.65

MOS SCORE	TYPES OF NETWORK	SPEECH CODEC	SNR		
			10dB	20dB	30dB
	OPTIMUM NETWORK	GSM	2.41	3.2	3.54
		ILBC	3.36	3.49	3.99
		SPEEX	3.37	3.79	4.01
	NETWORK WITH OTHER TRAFFIC	GSM	2.03	2.36	3.44
		ILBC	3.13	3.47	3.94
		SPEEX	3.15	3.85	4

R-FACTOR	TYPES OF NETWORK	SPEECH CODEC	SNR		
			10dB	20dB	30dB
	OPTIMUM NETWORK	GSM	46.85	62.02	68.72
		ILBC	66.1	68.35	79.48
		SPEEX	62.75	78.5	80.17
	NETWORK WITH OTHER TRAFFIC	GSM	39.36	45.91	66.83
		ILBC	66.38	68.2	77.89
		SPEEX	60.73	75.52	79.57

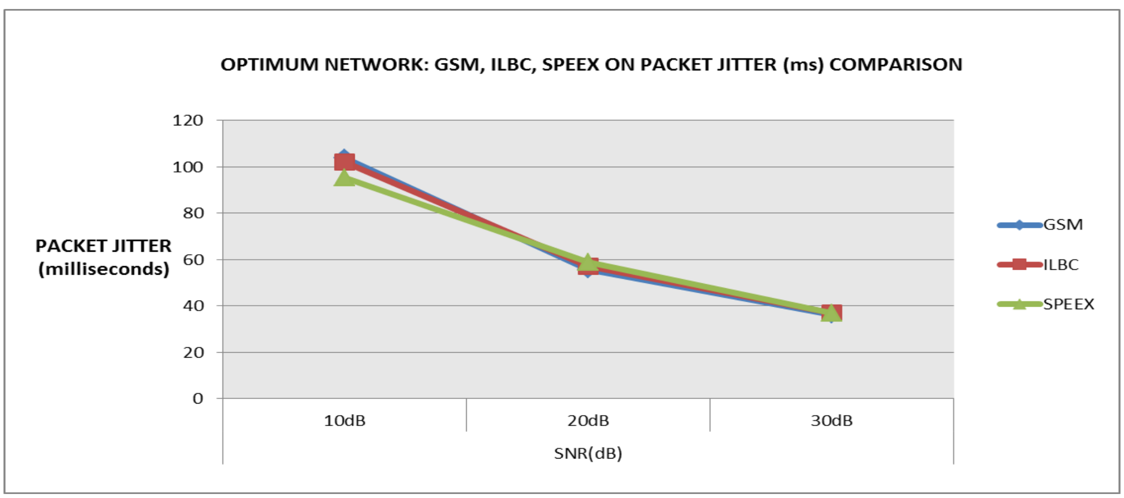
DATA AND CHART BASED ON TYPES ON NETWORK AND QUALITY OF SERVICE (QoS) PARAMETER

DATA AND CHART BASED ON TYPES OF NETWORK AND QOS PARAMETER

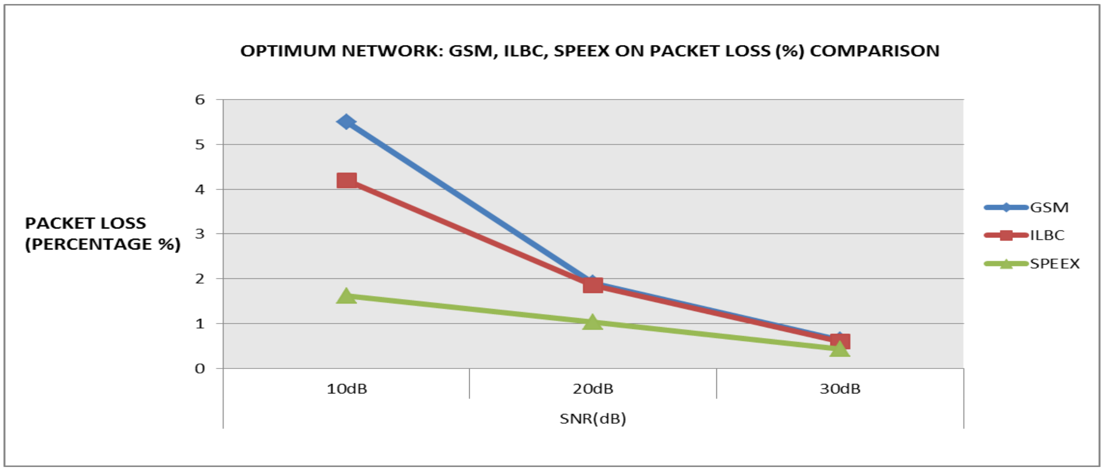
TYPES OF NETWORK: OPTIMUM NETWORK

QOS PARAMETER: PACKET JITTER (ms), PACKET LOST (%), MOS SCORE , R-FACTOR

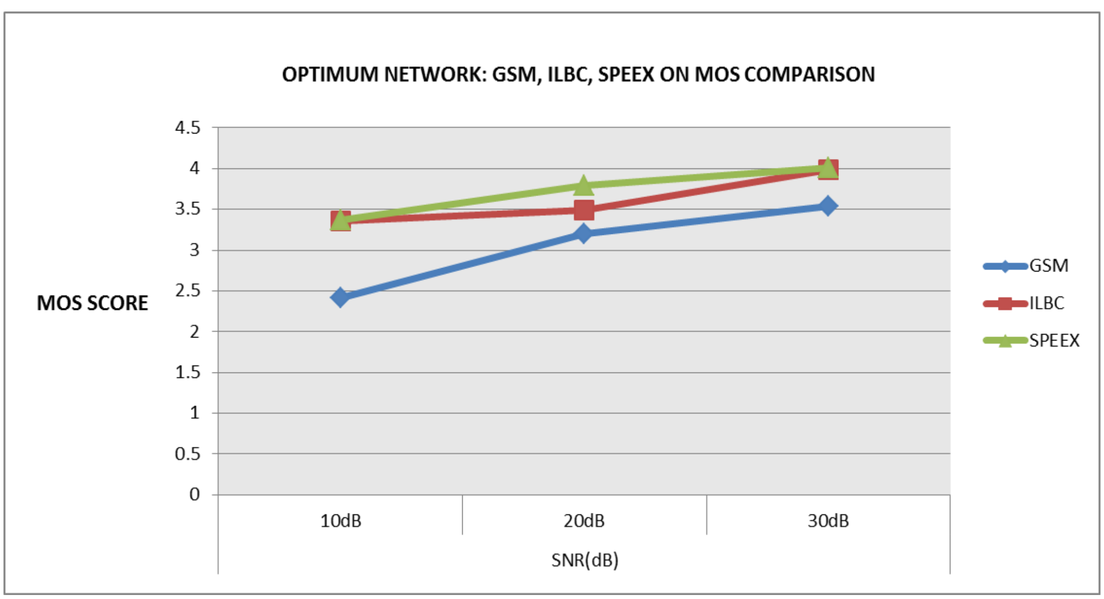
OPTIMUM NETWORK	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	PACKET JITTER (ms)	GSM	103.647	55.603	36.287
		ILBC	101.808	56.923	36.711
		SPEEX	95.155	58.606	36.785



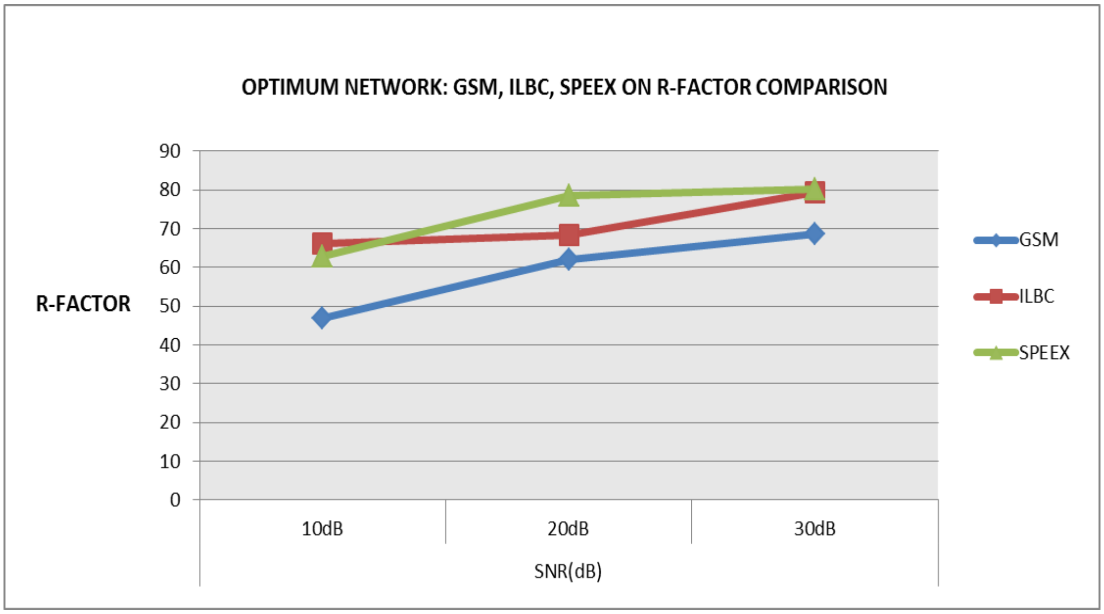
OPTIMUM NETWORK	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	PACKET LOSS (%)	GSM	5.49	1.9	0.64
		ILBC	4.2	1.86	0.6
		SPEEX	1.62	1.03	0.44



OPTIMUM NETWORK	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	MOS SCORE	GSM	2.41	3.2	3.54
		ILBC	3.36	3.49	3.99
		SPEEX	3.37	3.79	4.01



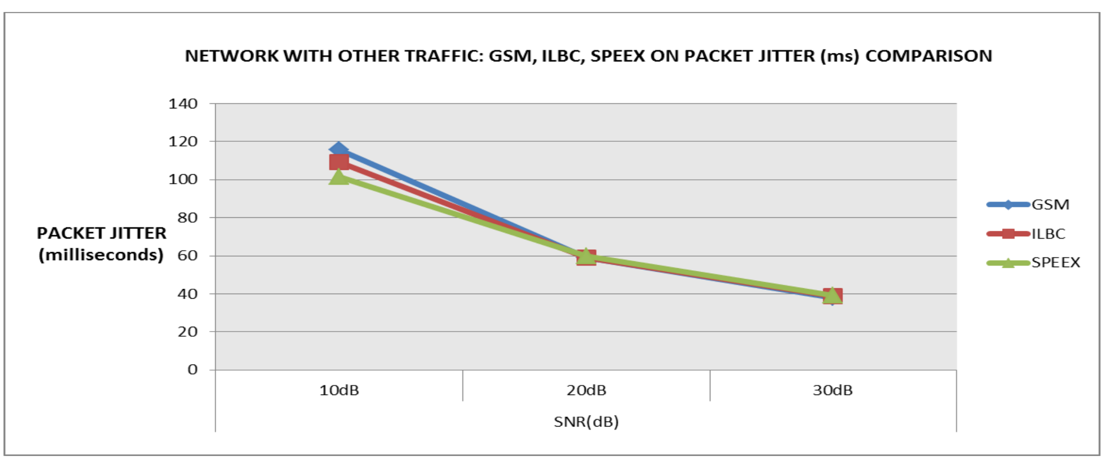
OPTIMUM NETWORK	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	R-FACTOR	GSM	46.85	62.02	68.72
		ILBC	66.1	68.35	79.48
		SPEEX	62.75	78.5	80.17



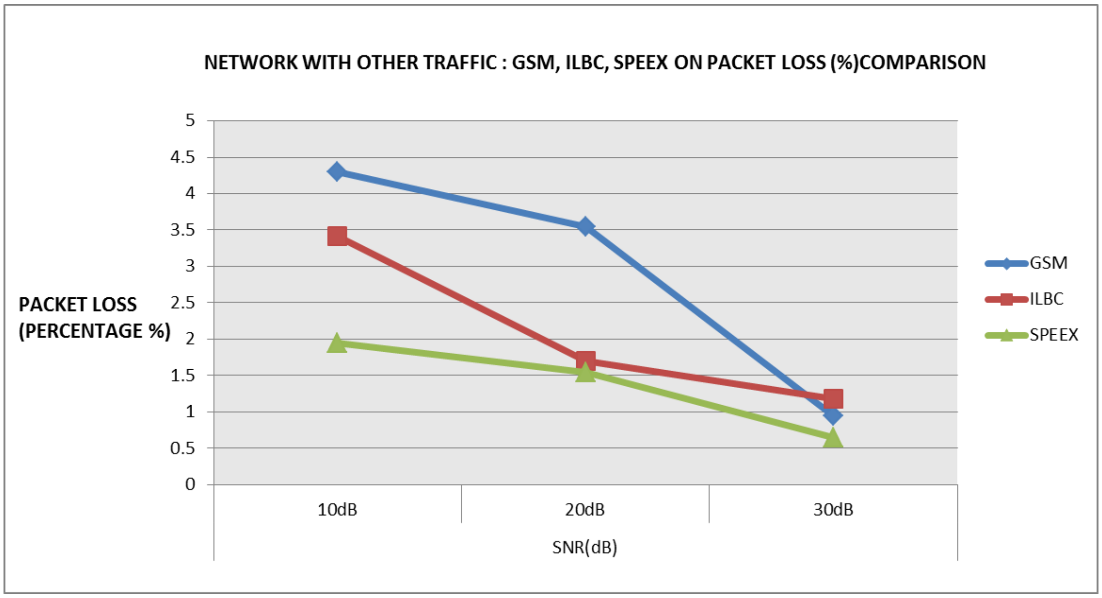
TYPES OF NETWORK: NETWORK WITH OTHER TRAFFIC

QOS PARAMETER : PACKET JITTER (ms), PACKET LOST (%), MOS SCORE, R-FACTOR

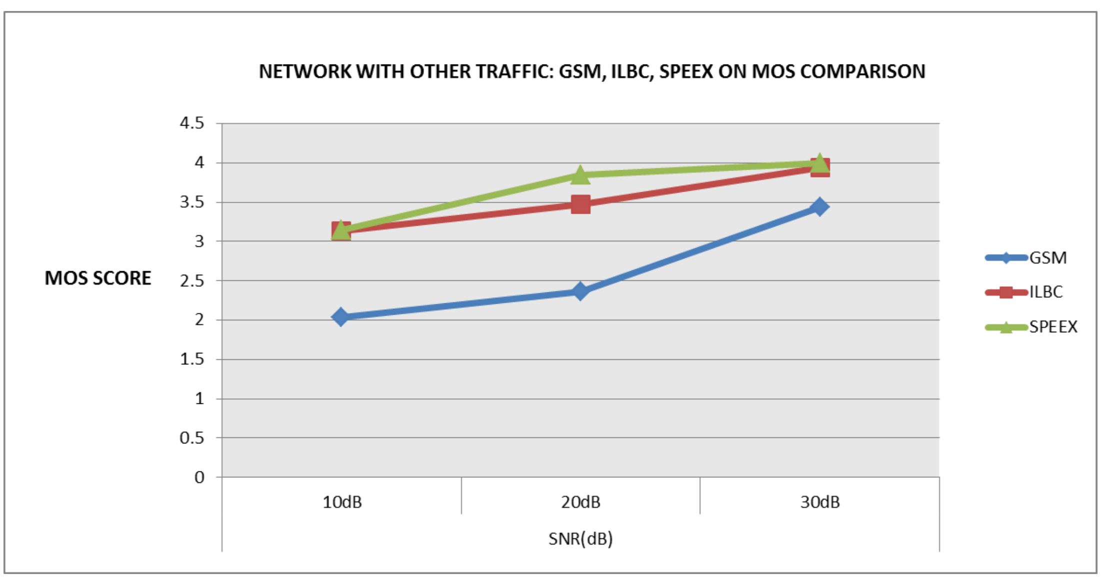
NETWORK WITH OTHER TRAFFIC	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	PACKET JITTER (ms)	GSM	115.744	58.855	37.76
		ILBC	109.376	58.968	38.88
		SPEEX	101.372	59.95	39.13



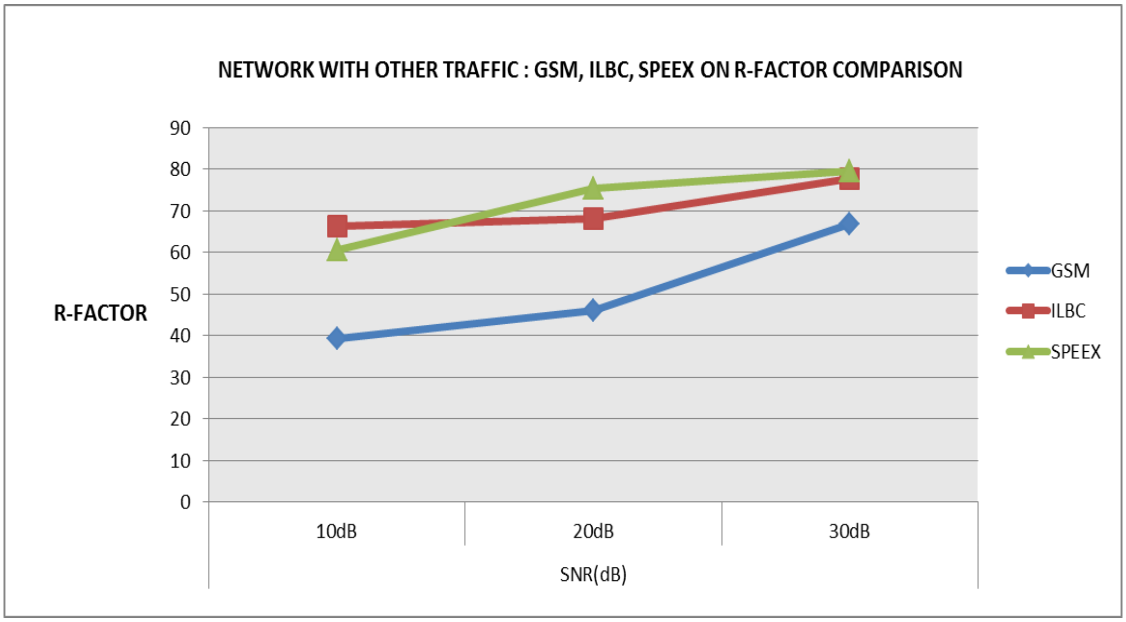
NETWORK WITH OTHER TRAFFIC	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	PACKET LOSS(%)	GSM	4.3	3.54	0.95
		ILBC	3.41	1.7	1.18
		SPEEX	1.95	1.55	0.65



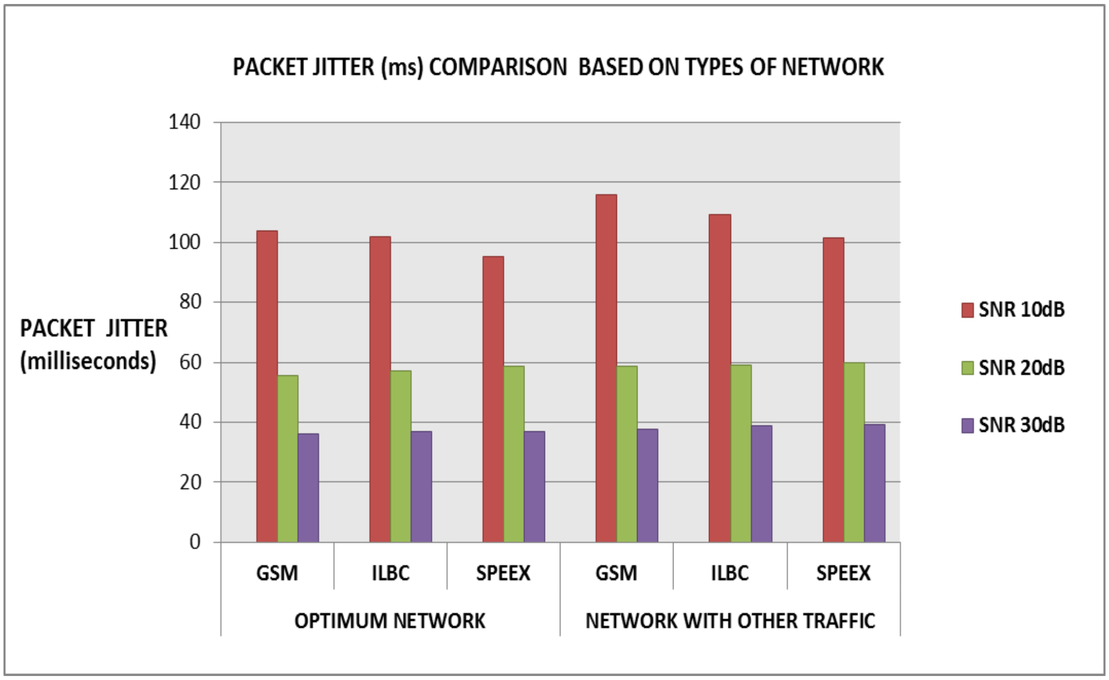
NETWORK WITH OTHER TRAFFIC	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	MOS SCORE	GSM	2.03	2.36	3.44
		ILBC	3.13	3.47	3.94
		SPEEX	3.15	3.85	4

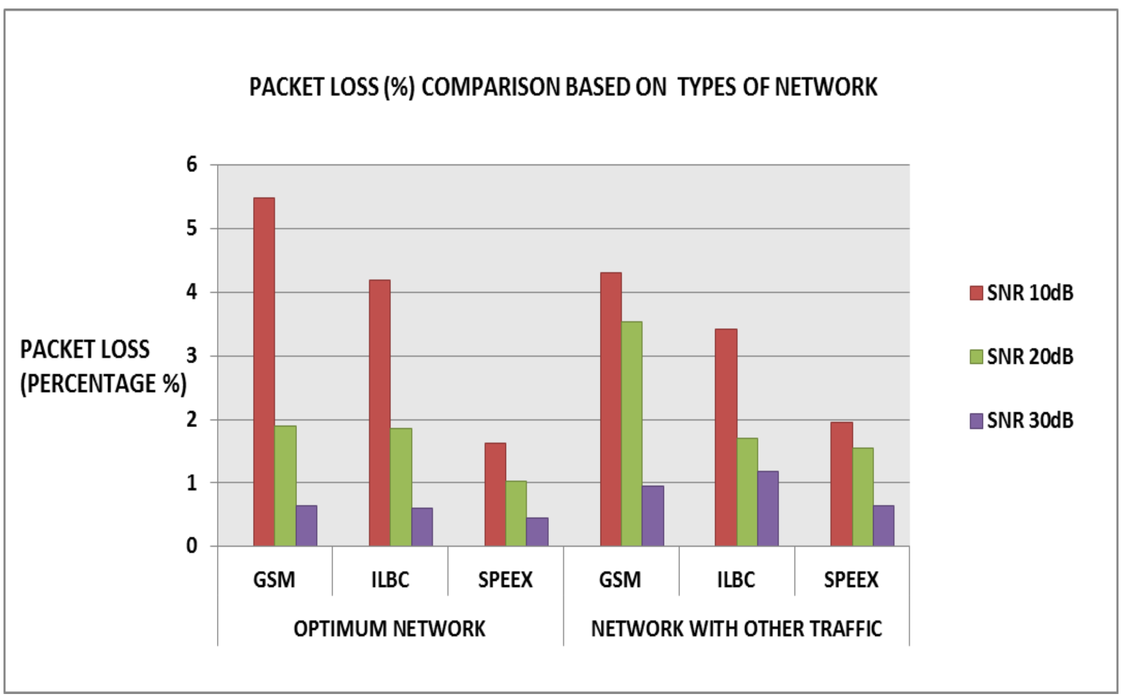


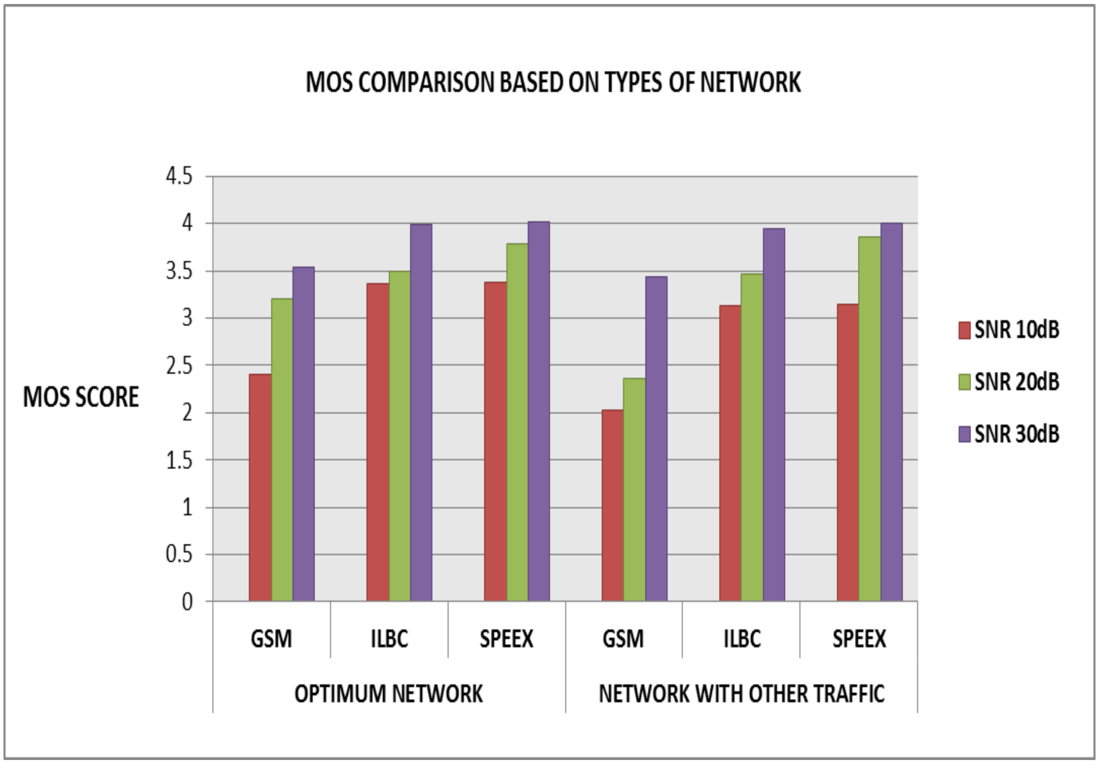
NETWORK WITH OTHER TRAFFIC	QOS PARAMETER	SPEECH CODEC	SNR(dB)		
			10dB	20dB	30dB
	R-FACTOR	GSM	39.36	45.91	66.83
		ILBC	66.38	68.2	77.89
		SPEEX	60.73	75.52	79.57



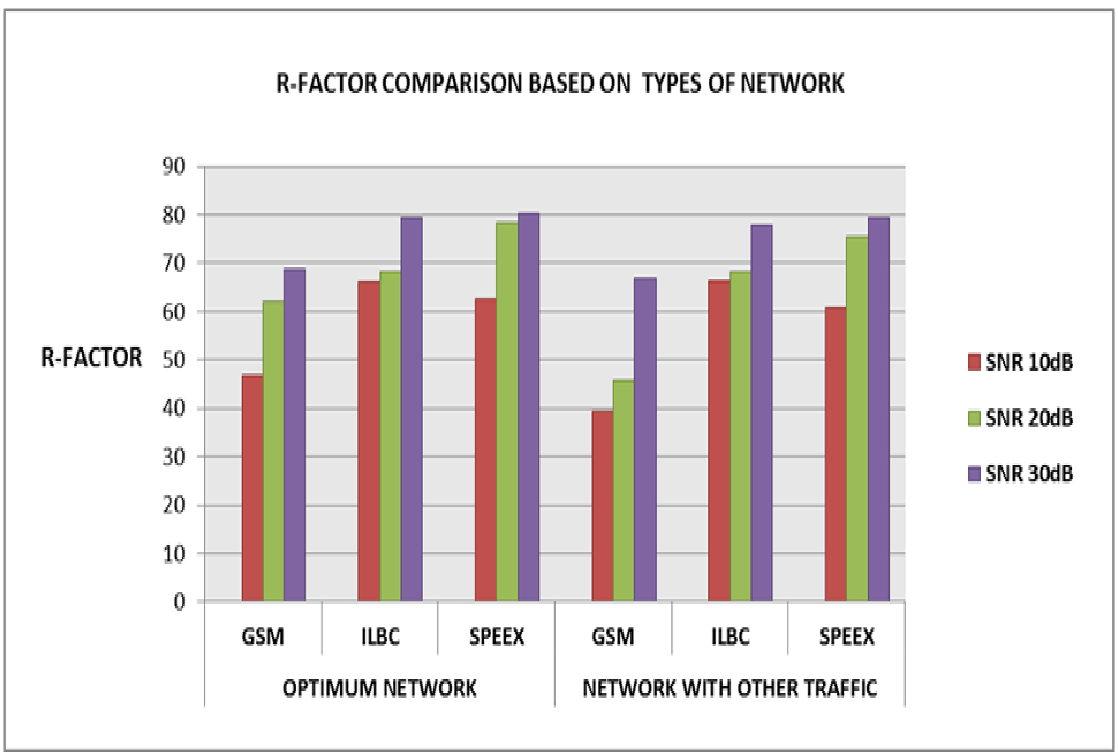
QOS PARAMETER COMPARISON BASED ON TYPES OF NETWORK







MOS COMPARISON BASED ON TYPES OF NETWORK
MOS SCORE
4.5
4
3.5
3
2.5
2
1.5
1
0.5
0
GSM
ILBC
SPEEX
GSM
ILBC
SPEEX
OPTIMUM NETWORK
NETWORK WITH OTHER TRAFFIC
SNR 10dB
SNR 20dB
SNR 30dB



R-FACTOR COMPARISON BASED ON TYPES OF NETWORK
R-FACTOR
90
80
70
60
50
40
30
20
10
0
GSM
ILBC
SPEEX
GSM
ILBC
SPEEX
OPTIMUM NETWORK
NETWORK WITH OTHER TRAFFIC
SNR 10dB
SNR 20dB
SNR 30dB