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Performance Evaluation of Voice over Internet Protocol (VoIP) on Wired and Wireless Networks

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ABSTRACT

Voice over Internet Protocol (VoIP) facilitates voice communication over an IP network such as internet, intranet etc. Basically, VoIP works by converting analogue voice signal to digital signal, which is then converted to IP packets and sent over the IP network. A sine qua non for suitability of a network for using VoIP is that it must be an IP network. VoIP offers cost-effective telephony service in that it involves sharing of existing data network facilities. VoIP uses signaling protocols to achieve high-quality voice communications and the protocols are responsible for establishing and tearing down calls and enables network protocols to communicate with each other. The paper evaluates the performance of VoIP over wired and wireless networks using a laboratory experimental approach. A wireshark software is utilised to monitor communication while the Oberserve-17 is used to evaluate and measure quality of service (QoS) of VoIP call. It was discovered that the main issue to be addressed is that of security and quality of voice communication. This is because VoIP architecture differs from that of traditional circuit-based, analogue telephony service. Although, there are other issues associated with VoIP, but the major concern of this paper are security and voice communication quality.

Keywords: Digital signal, protocol, wired, wiredless, VoIP

1. INTRODUCTION

The voice codecs and quality of service (QoS) are the determinant factor in the delivering of effective and efficient communication in a voice over internet protocol (VoIP). The voice codecs are the algorithms that allows the system to carry analog voice over digital lines. Several codecs available with varying degree in complexity, bandwidth required and voice quality. The more bandwidth a codec requires, normally the better voice quality is (Karapantazis and Pavlidou, 2009). Codec, which stands for compression-decompression encodes the voice data to be embedded in the network packet to use minimal amount of bandwidth and the data will be decompressed at the receiving end for maximum voice quality. Voice codecs are classified into three, namely: narrowband codecs, broadband codecs, and multimode codecs (Karapantazis and Pavlidou, 2009).

1. Narrowband Codecs: They operate on audio signals that range from 300 to 3400GHZ sampled at 8KHZ. Codecs under this category include G.711, G.723.1, G.726 among others.
2. Broadband Codecs: They operate on audio signals filtered to a frequency range from 50 to 7000 HZ sampled at 16 KHZ. Popular codecs in this category include G.722, G.722.1, AMR-Wb+, GSM-HR, AMR etc.
3. Multimode Codecs: They operate on either narrowband or broadband signals and they include Speex, BroadVoice etc.

For the quality of service (QoS), the VoIP applications require a real-time data streaming and the quality of a call can be measured using one of several call quality metric computations. The most used system is the mean opinion score (MOS). The MOS score of a call is between 1 (for unusable) and 5 (for excellent) call quality. VOIP calls that are working properly fall between 3.5 and 4.2 while the toll quality is pegged at 4.0. Other systems for quality measurement are R-factor, PSQM, PESQ, and PAMS. The MOS rating from 5, 4, 3, 2 and 1 represents excellent listening quality and complete relaxation listening effort, good listening quality and attention needed listening effort, fair listening quality and moderate effort listening effort, poor listening quality and considerable effort listening effort and bad listening quality and no meaning listening efforts respectively. Packet loss is a major setback to VoIP as packets may not be delivered as a result of loss either due to security or bandwidth issues.

Table1: Showing Quality of Service Scale based on MOS

MOS SCORE	CALL QUALITY
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

2. LITERATURE REVIEW

2.1 Important Issues affecting VoIP Technology

The internet architecture is associated with many issues such as security, congestion, quality and so on, since VoIP technology makes use of the internet architecture it is therefore right to say that VoIP will inherit all the issues associated with the internet. Moreover, before users can accept this technology to the traditional PSTN, this technology must be able to provide more advantages than the traditional PSTN. Hence there is need for quality of service which acts as a network resource reservation and prioritization (Tim Szigeti 2014). This is so because QoS helps to measure packet loss, jitter, bandwidth and delay in network as well as ensure they are improved to some certain extent in advanced before the actual data is transmitted. QoS operation is based on viewing and treating all network packets as not equal. QoS gives some sessions such as delay-sensitive sessions priority over the other sessions which are less sensitive to delay. These high priority sessions bypass other sessions. The numbers of simultaneous calls a particular network bandwidth size can support is referred to the concurrent call capacity of such network. This has a direct relationship with the bandwidth available on the network as well as the type of CODEC employed. Consider a lossless CODEC type which takes about 64kbps to process a voice signal, the number of concurrent calls supported can be calculated as

$$\text{Number of Concurrent Call} = \frac{\text{Available Bandwidth}}{\text{CODEC bit rate+Overhead}} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots \text{ eqn 1}$$

It can be concluded from the equation 1, that the higher the bit rate of a CODEC, the lower the number of simultaneous calls allowed. SIP trunking is a technology that supports concurrent VoIP calls and this is illustrated fully by (Ayokunle 2012). (SmartBits 2001) points out that there is a general correlation between the voice quality and the data rate. This means that the higher the data rate, the higher the voice quality; hence, this strongly leads to the choice of CODEC. The choice of CODEC depends on some factors such as the communication distance, the bit rate required, the bandwidth available, drop sensitivity among other factors. (Daniel Minoli 2002), describes that CODEC speech quality is a function of bit rate, complexity, and processing delay. This means that the choice of CODEC is greatly affected by the aforementioned attributes. A low-bit-rate CODEC tends to have more delay than higher-bit-rate CODECs. This shows that for applications such as voice which requires no or low delay, a higher-bit-rate CODEC will be preferred. Another issue associated with low-bit-rate CODEC is the complexity involved in their implementation. This complexity results in higher costs and greater power usage (Minoli 2006).

Finally, a low-bit-rate CODEC have lower speech quality as compared to higher-bit-rate CODECs. Therefore, this shows that the quality to be expected from a voice call will have much correlation with the type of CODEC used. For a low effective bandwidth network such as WAN, a low bit rate CODEC is preferable if not the quality of calls will suffer due to bandwidth limitations which will lead to loss of packets. LAN is known to provide high bandwidth (greater than 100Mbps) therefore a high bit rate CODEC can be employed. This leads to another important discussion of private LAN. The size of private LAN infrastructure makes it relatively easy to control the quality of transmission of voice over either LAN or WLAN by controlling network parameters such as bandwidth, packet loss, delays and so on. The requirements for VoIP on a LAN will be further illustrated in the paper. Bandwidth/Concurrent Call Capability is very high bandwidth is necessary for VoIP communication for better voice quality. Low bandwidth can cause packet loss or poor voice quality.

Thus, proper bandwidth reservation and allocation is essential to VOIP quality. Low bandwidth can also lead to delays during packet routing. It also determines if a VoIP system will have the capacity to sustain concurrent calls. The choice of codec significantly determines the performance of VoIP (Karapantazis and Pavlidou, 2009). This is so because the codecs have different features which determines which is suitable in a particular scenario. These features include frame/packet speed, number of bits per frame/packet, algorithmic delay, codec delay, compression type, complexity, and the average mean opinion score (MOS). Choice of codec needs to be determined during the requirement analysis phase of the system setup and hybrid usage of codecs contribute to codec delay due to different coding/decoding schemes used by individual codecs. Ismail (2009) studied the analyses the effect of codec selection on the performance of VoIP technology in a campus environment using MOS as measurement parameter. He carried out experiments using a soft phone and IP phone. He concluded that WAN contributes higher delay, higher packet loss and higher CPU usage than LAN in a campus environment.

In another experiment, Ismail (2011) studies the effect of five codecs mainly: G.711, G.722, G.726, GSM and Speex on both wireless LAN and WAN. For LAN, the codecs have MOS score of 4, 2, 3, 4 and 1 respectively. For WAN, the MOS score are 1, 1, 2, 3 and 1 respectively. From the analysis of results obtained, he concluded that wireless LAN offers better voice quality than wireless WAN and that the best codec for wireless WAN is GSM. Siradeghyan and Kirakossian (2012) also did a performance evaluation of VoIP over wired and wireless networks and in their analysis, wired network outperforms that of wireless network. VoIP security deals with ensuring that only authorised persons can make calls and the eavesdropping on the communication channel is prevented or even total hijack of the entire communication through attack on the communication servers. Threat to VoIP systems are classified into six (Stanton,) namely; denial of service (DoS), theft of service, telephone fraud, nuisance calls, eavesdropping and misinterpretation. Therefore, the first step towards security is identification of clients and authentication through privacy mechanisms such as encryption.

2.2 Session Initiation Protocol (SIP) AND Real Time Protocol (RTP) Used for VoIP Calls

SIP is an application-layer protocol is used for creating, updating, and terminating sessions among users, and was designed to be independent of the underlying transport protocol i.e the real time transport protocol (RTP). A SIP system is made up of end nodes, a proxy, location server and also the registrar. Considering a SIP model, a user is not attached to a particular host. The user at the start their location to the registrar which may then be integrated into the proxy server or redirect server. Consequently, the information will be stored in the external location server. The messages from the end nodes can only be transmitted through either using a proxy or redirect sever. Messages coming from end nodes and other services are usually intercepted by the proxy server and check for the destination username and subsequently inform the location server to resolve username into appropriate address and the despatch the message to the designated end node or any other sever. The same function can also be performed by Redirect Sever but end nodes is responsible for the actual routing. That is, Redirect servers obtain the actual destination address of the destination from the location server and return this information to the original sender, which then must send its message directly to this resolved address (Kuhn et al., 2005).

SIP devices can be categorised into end-to-end devices and workhorses (Sisalem and Kuthan)

SIP end-to-end devices include:

- User Agent Client (UA Client): It originates call
- User Agent Server(UA Server): It listens to incoming call.

SIP Workhorses include:

- SIP Proxy Server: It relays call signals, i.e. acts as both client and server.
- SIP Redirect Server redirects callers to other servers when it cannot handle a request.
- SIP Registrar accept registration requests from users and maintains users' whereabouts at a Location Server.

3. METHODOLOGY

The methodology adopted was the use of laboratory experiment with test results to ascertain the behaviour of VoIP on a wired and wireless networks using Wireshark and Observer-17 software packages.

3.1 SIP Method/Requests (RFC 2543)

SIP uses the following methods/requests for communications among users.

- **INVITE:** These initiates sessions and the session description is embedded in message body. It can also re-INVITE when session state needs to be changed.
- **ACK:** Confirms establishment of sessions. It can only be used with INVITE
- **BYE:** It terminates sessions
- **CANCEL:** It cancels a pending INVITE
- **OPTIONS:** It communicates users about the capability of SIP phones, both calling and receiving.
- **REGISTER:** It communicates user's location through the IP.

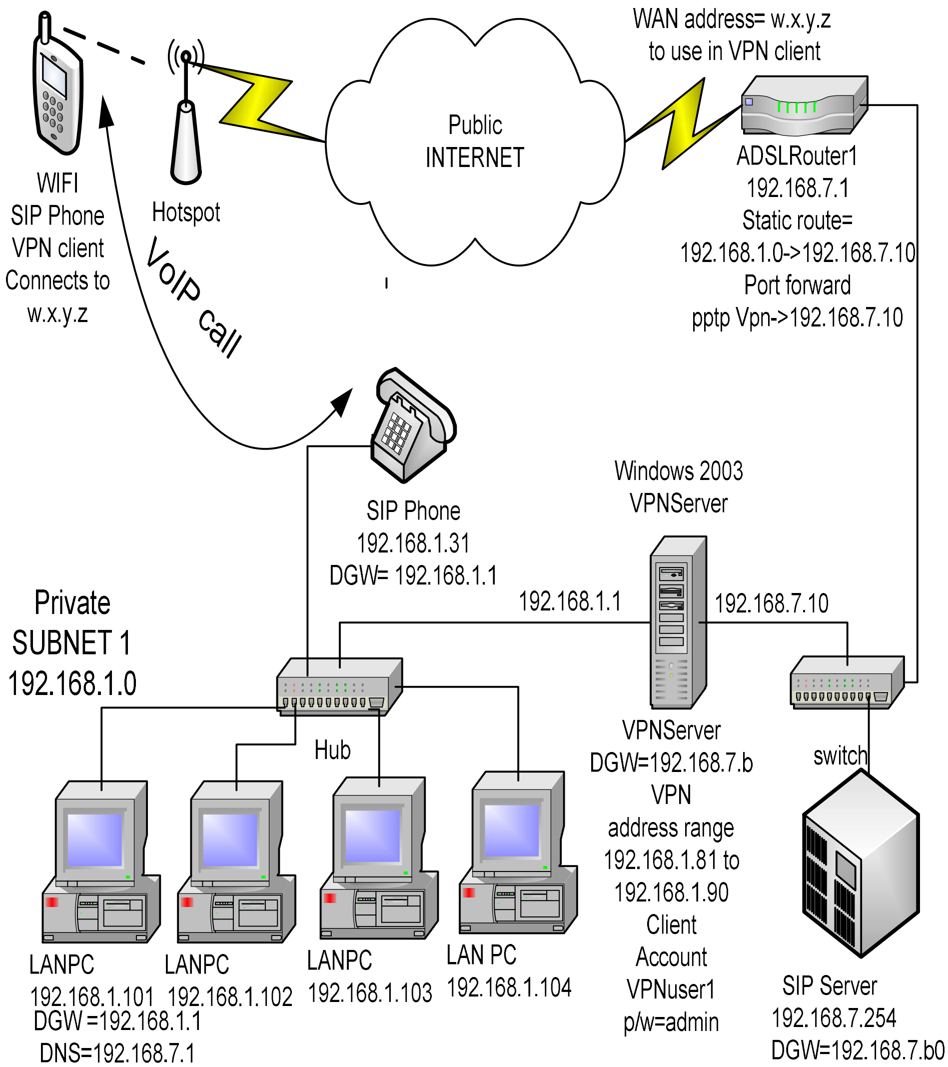

Figure 1: The Physical Topology of the Network

Figure 1 shows the physical topology of the network for the laboratory investigation and figure 2, displays the screen shot of the laboratory investigation of the communication between a user agent (UA) server with IP address (192.168.2.101) and SIP sever. Figure 3, illustrates the graph flow of SIP user registration on the server while figure 4, shows the call set-up and call take-down periods on the VoIP communication.

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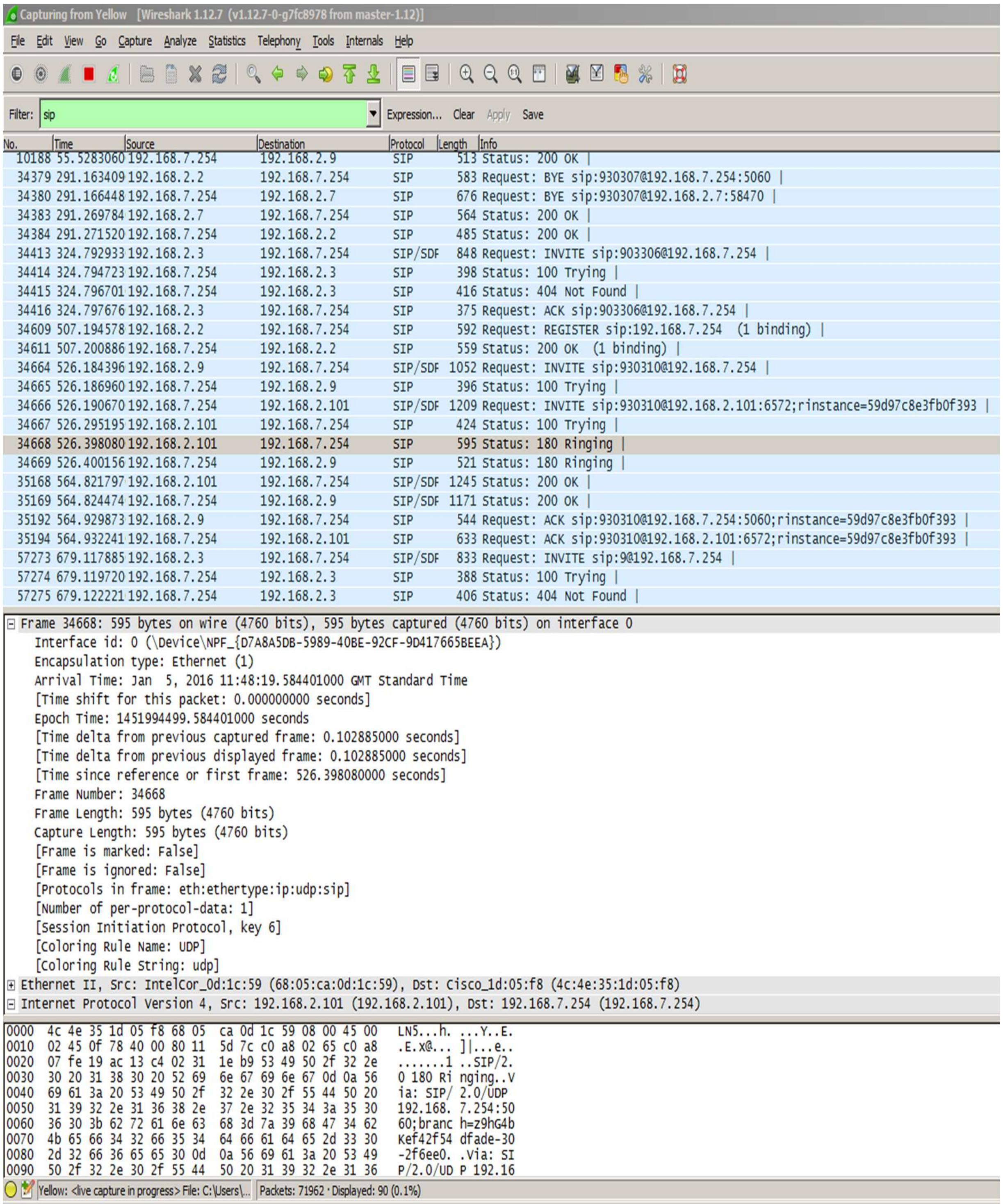

Figure 2: SIP User Registration

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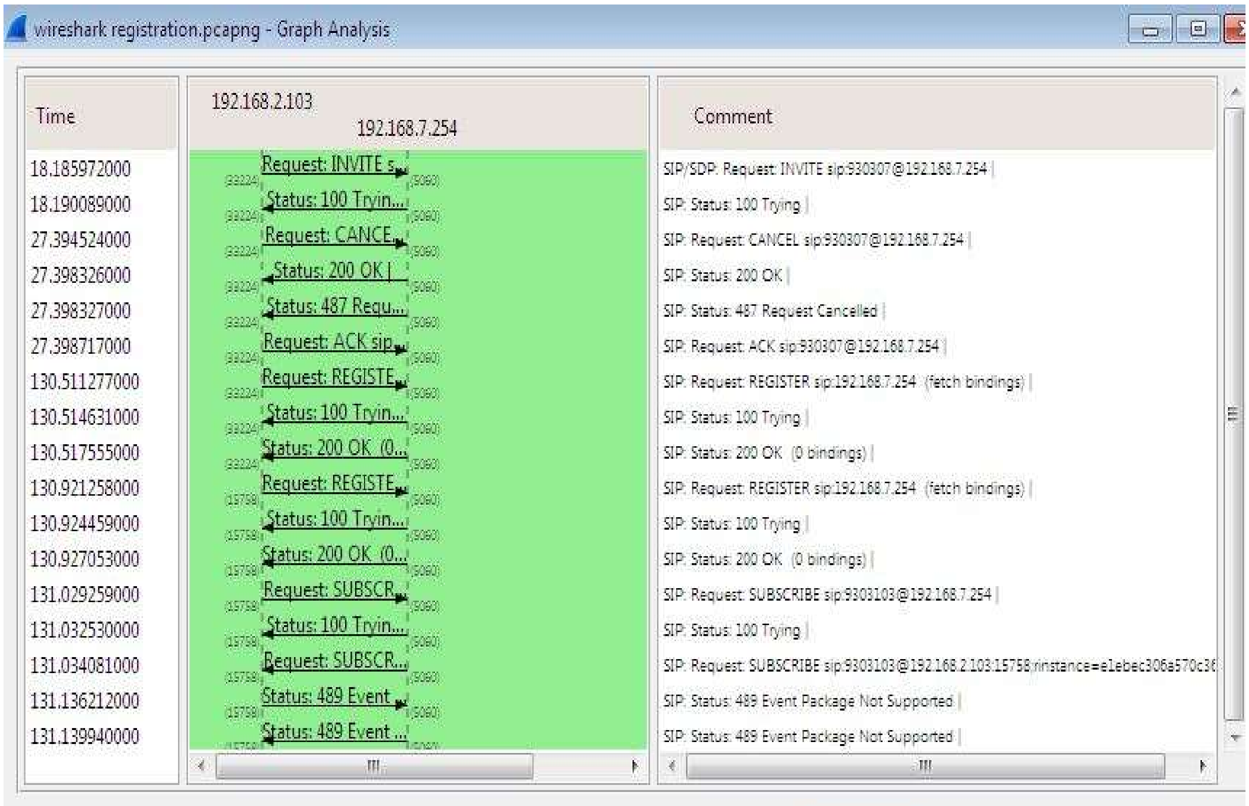


Figure 3: Graph flow for a SIP user Registration to the Server

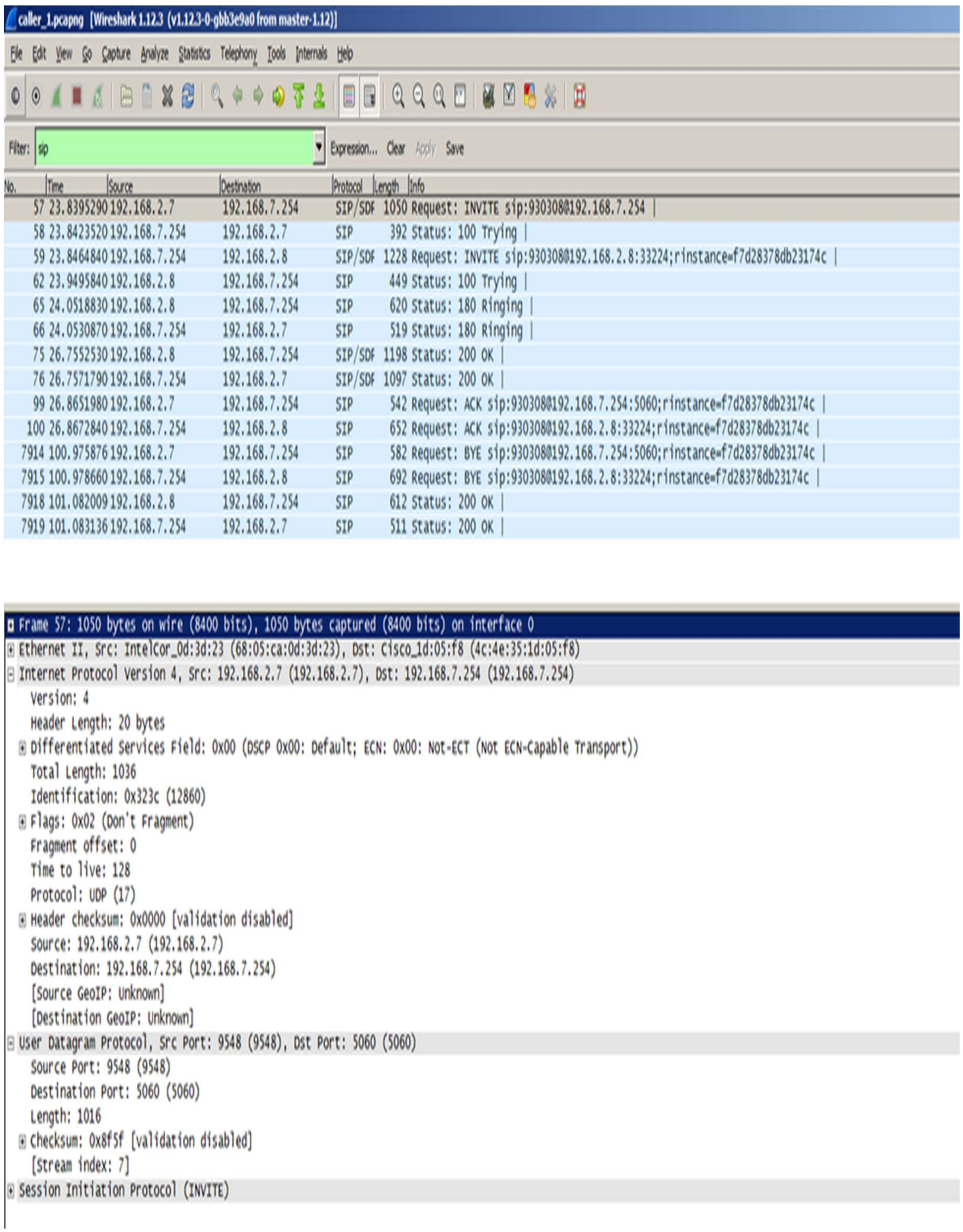


Figure 4: Call set up and call take down

3.2 Response Codes

- **1xx:** This refers to provisional response with no clear definition which basically means the server is still performing some actions and does not have definitive response yet.
- **2xx:** This means the invitation request was successful
- **3xx:** This implies the server has redirected the request to some other available servers
- **4xx:** This means the request has failed as a result of the SIP client error.
- **5xx:** This implies the request has failed due to server error.

3.3 Real Time Transport Protocol (RTP)

Real Time Protocol (RTP) (RFC 1889) performs the following functions: detects media content type, sender identification, data synchronization, data loss detection, segmentation, and security (encryption). In an SIP based VoIP, once the session has been successfully established, the RTP takes over the activities and handles all the packet transmission related issues. Figure 5 illustrates the takeover of activity from SIP by RTP during call initiation.

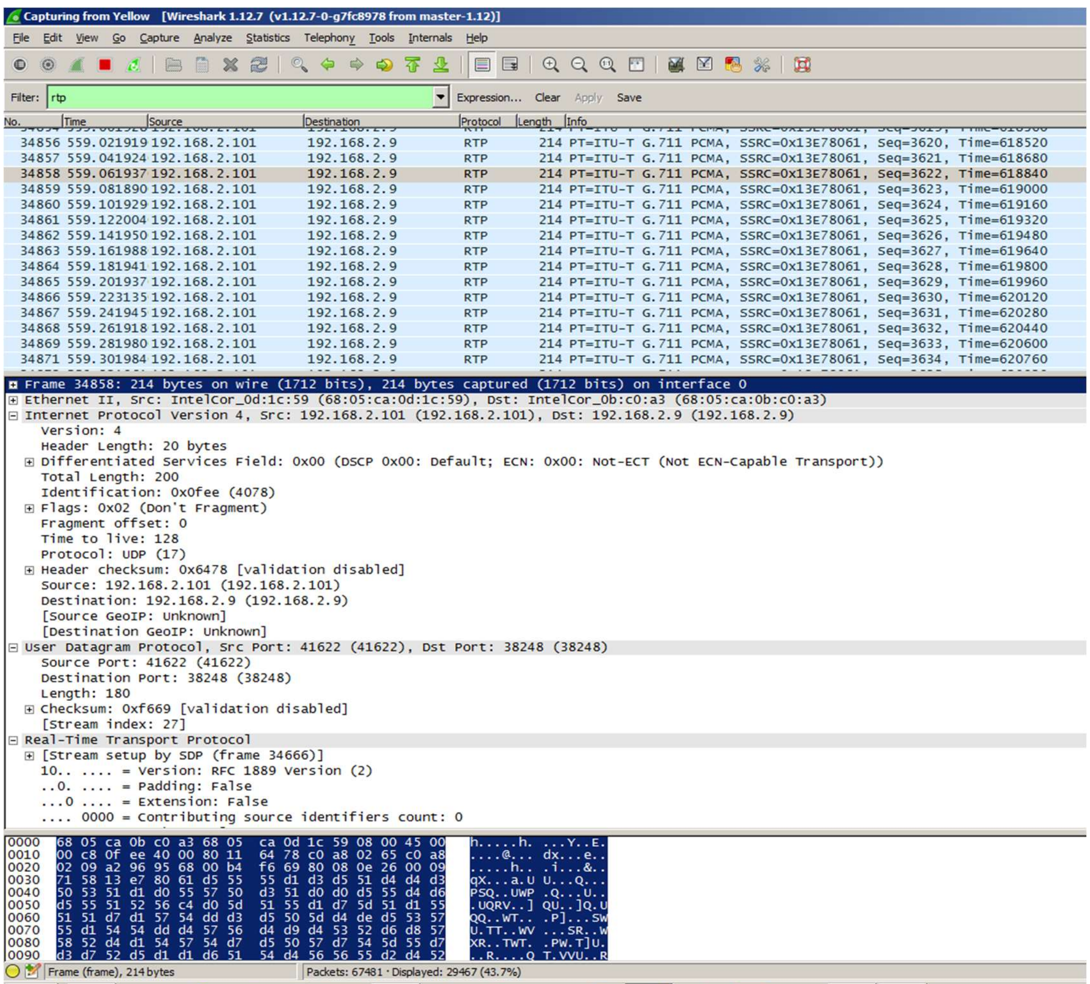
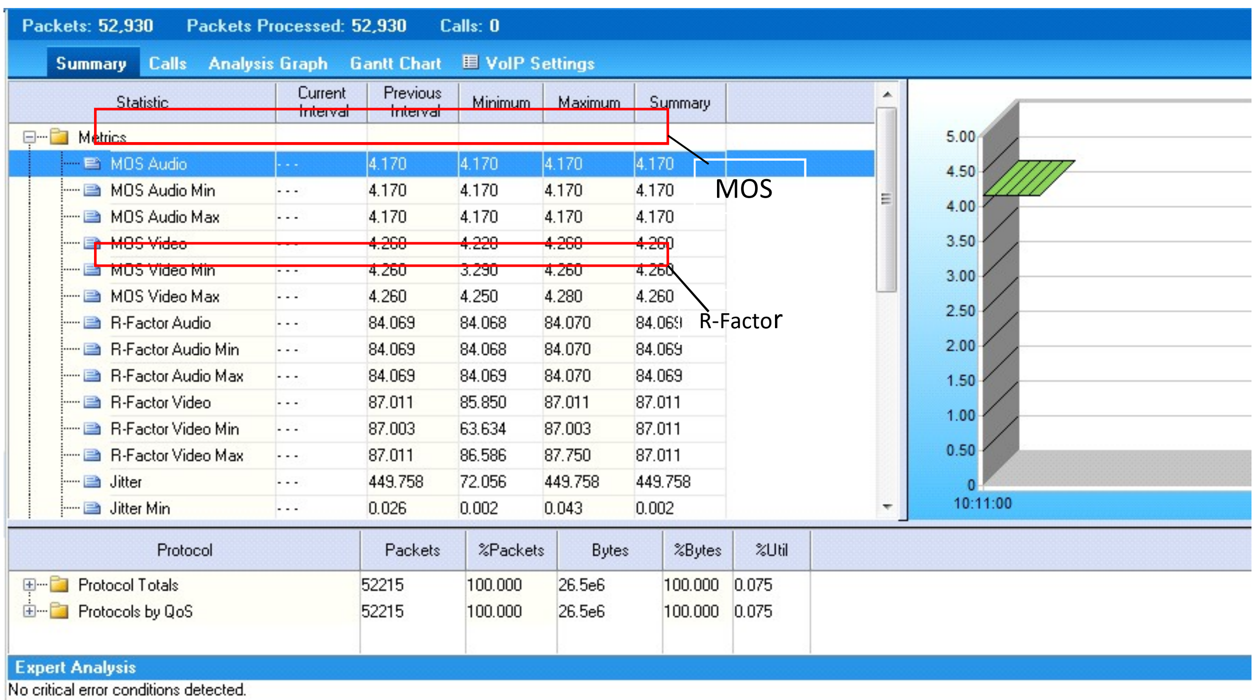


Figure 5: Operation of Real Time Protocol (RTP)

The header of the packets being routed by RTP typically consists of Synchronization Source Identifier (SSRC), a time stamp, payload, and a sequence number.

4. LABORATORY RESULT FINDINGS

Implementing VoIP over Wi-Fi network is plagued by two problems: quality of voice and security of the medium. As earlier mentioned, the MOS is the main evaluation metric for voice quality. Packet loss leads to complete lack of communication. From the laboratory experiment juxtaposing the results of the VoIP over Wi-Fi and wired LAN, we present the MOS scores and R-factor for the two scenarios among other quality evaluation metrics. Figure 6 shows the MOS and R-factor for VoIP on Wi-Fi network using observer 17 software while figure 7 displays the MOS and R-factor for VoIP on wired LAN captured with Observe-17 software.



Statistic	Current Interval	Previous Interval	Minimum	Maximum	Summary
Metrics					
MOS Audio	- - -	4.170	4.170	4.170	4.170
MOS Audio Min	- - -	4.170	4.170	4.170	4.170
MOS Audio Max	- - -	4.170	4.170	4.170	4.170
MOS Video	- - -	4.260	4.220	4.260	4.260
MOS Video Min	- - -	4.260	3.290	4.260	4.260
MOS Video Max	- - -	4.260	4.250	4.280	4.260
R-Factor Audio	- - -	84.069	84.068	84.070	84.069
R-Factor Audio Min	- - -	84.069	84.068	84.070	84.069
R-Factor Audio Max	- - -	84.069	84.069	84.070	84.069
R-Factor Video	- - -	87.011	85.850	87.011	87.011
R-Factor Video Min	- - -	87.003	63.634	87.003	87.011
R-Factor Video Max	- - -	87.011	86.586	87.750	87.011
Jitter	- - -	449.758	72.056	449.758	449.758
Jitter Min	- - -	0.026	0.002	0.043	0.002

Protocol	Packets	%Packets	Bytes	%Bytes	%Util
Protocol Totals	52215	100.000	26.5e6	100.000	0.075
Protocols by QoS	52215	100.000	26.5e6	100.000	0.075

Figure 6: MOS and R-Factor for VoIP on Wi-Fi Network

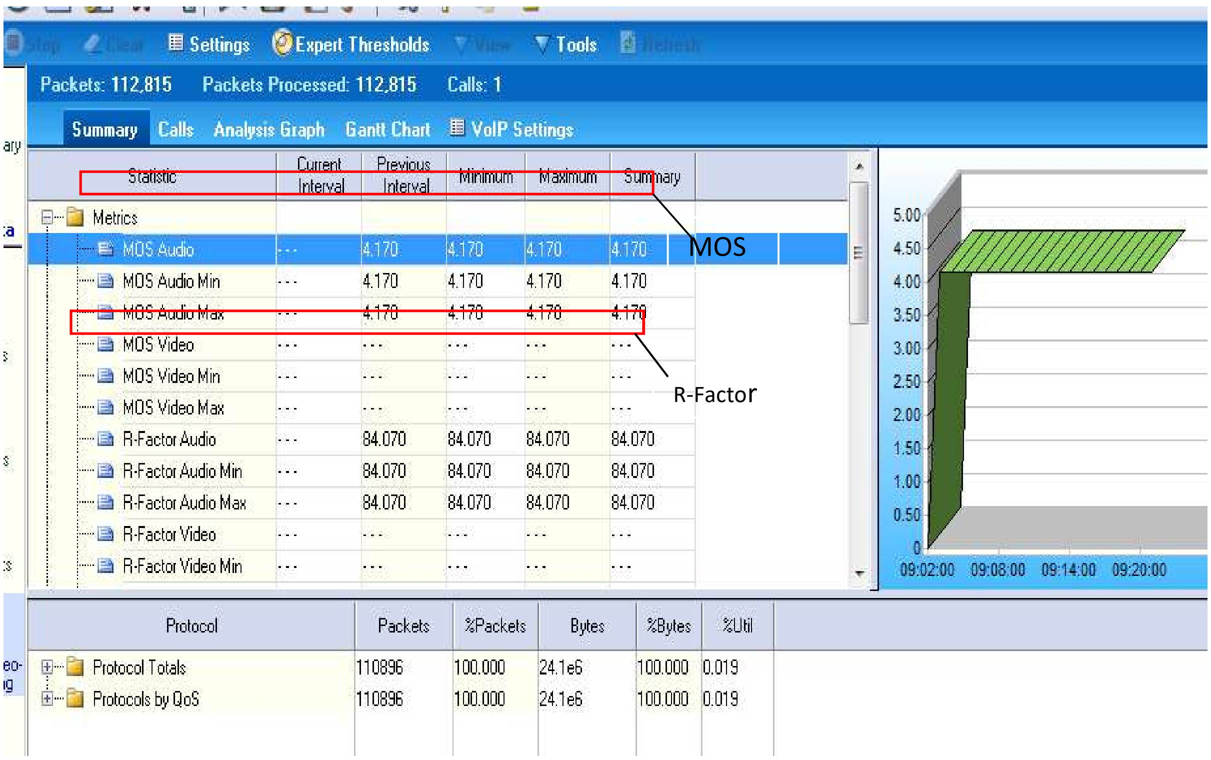


Figure 7: MOS and R-Factor for VoIP on Wired LAN

In contrast to MOS, R-factor score ranges from 0 to 120. Also, the MOS originally represent the arithmetic mean average of all the individual voice quality evaluation given by people who listened to a test phone call, but it is now computed using intelligent software. On the other hand, R-Factor, an alternative metric for voice quality assessment, is calculated by evaluating user perceptions as well as the objective factors that affect the overall quality of a VoIP system. These factors consist of the network R-factor and User R-factor. Packet loss is another problem affecting secured Quality of Service (QoS). It leads to voice communication outage. We also present the packet loss rate and jilter for both scenarios in Figure 8 and Figure 9, show the packet loss and Jilter for VoIP over Wi-Fi Network and the packet loss and Jilter for VoIP over Wired LAN.

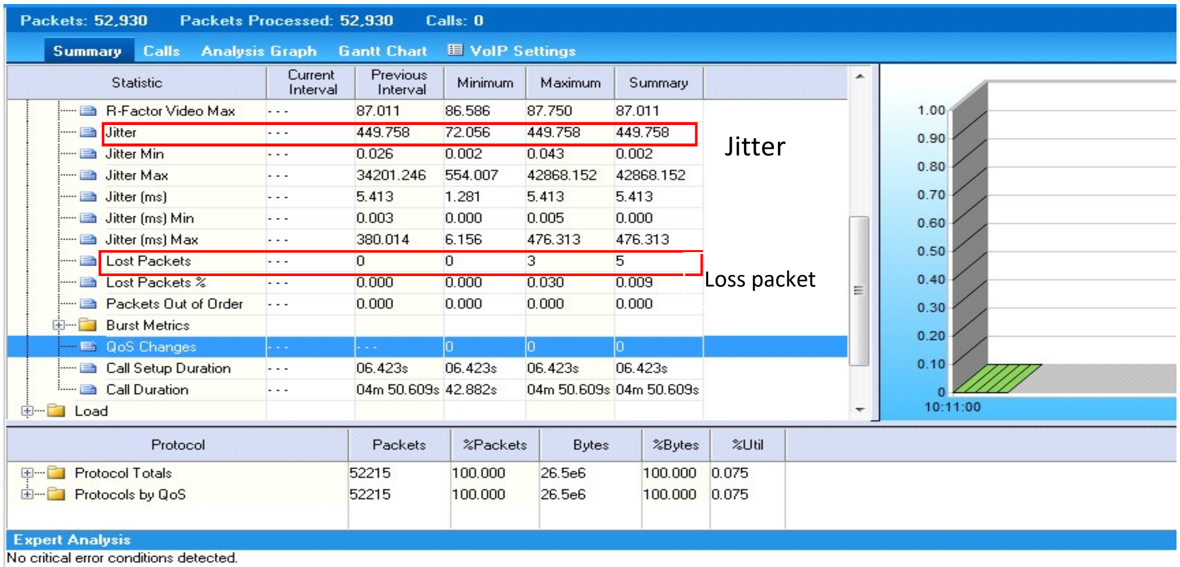


Figure 8: Packet Loss and Jilter for VoIP over Wi-Fi Network

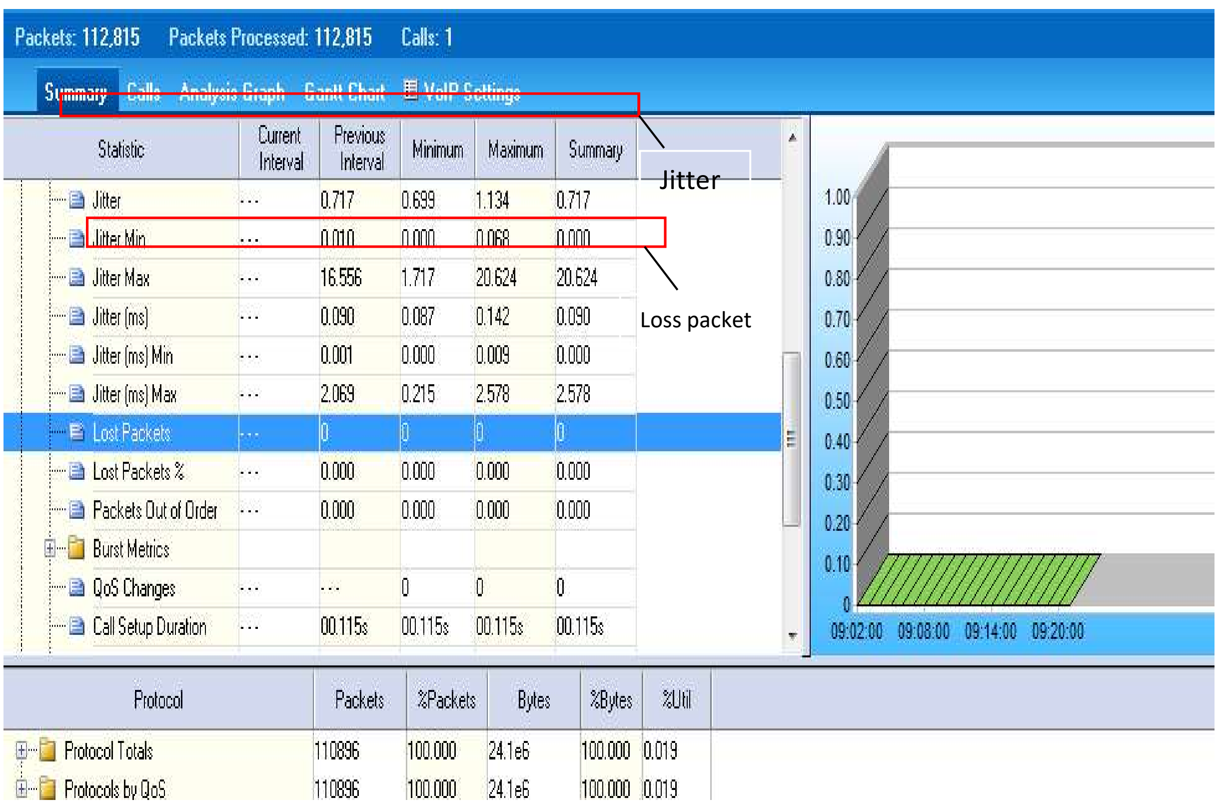


Figure 9: Packet Loss and Jilter for VoIP over Wired LAN

4.2 Jitter can be described as delay in network which also affects call quality however, the occurrence of jitter is because of packet delay, but due to the of a variation of packet delays. As end nodes increase the size of the packet buffer in order to compensate for the jitter, jitter produces delays in the conversation. If the variation becomes very high and exceeds 150ms, callers can notice the delay.

4.3 Out of Sequence Error, also referred to as packets out of order is a problem which adversely affects call quality. It occurs when packets do not follow the other in which they were sent thereby causing wrong or mismatch voice communication. Unlike traditional data communication networks, correction out of sequence packet are expected in real time in order for the communication to make sense (Executive Summary, Monitoring and Troubleshooting VoIP Networks with a Network Analyzer). The experimental results further present a juxtaposition of the metrics for both VoIP over Wi-Fi and wired LAN in Table 2:

Table 2: Comparison of QoS Evaluation Metric between VoIP over Wi-Fi and VoIP over wired LAN

QoS Evaluation Metric	VoIP over Wi-Fi	VoIP over Wired LAN
MOS (5)	4.170	4.170
R- Factor (120)	73	84.070
Loss Packets Rate (%)	3	0
Out of Sequence Packets	0	0
Jitter	449.758	0.707
Codec Category	Broadband	Broadband

6. CONCLUSION

VoIP is still an emerging technology making it open for active research and generating lots of research interest. Implementing VoIP comes with lots of problems unlike traditional data networks ranging from quality to security and bandwidth management issues. Implementation of VoIP can be in many forms such as wired or wireless connection to network. In this paper, operation of VoIP and Codecs was reviewed, issues relating to VoIP protocols was discussed and narrowed to specific scenarios of wireless and wired network experiments. From the analysis of our laboratory results, it can be seen that voice over Wi-Fi experienced a higher jitter when compared to VoIP over wired LAN. This can be attributed to congestion issues on the internet. We can therefore say VoIP setup over wired LAN produces better performance than VoIP over wireless (Wi-Fi) network using the evaluation metrics discussed earlier.

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