International Journal of Engineering & Technology, 7 (4.29) (2018) 251-254

International Journal of Engineering & Technology

Website: www.sciencepubco.com/index.php/IJET

Research paper

Performance Analysis of Different Audio Video Codecs for Wireless and Wired VoIP

Mohd Nazri Ismail[1]*, Mohd 'Afizi Shukran[2], Mohd Rizal Mohd Isa[3], Mohd Adib[4]

[12345]*Faculty of Defense Science and Technology, National Defence University of Malaysia (UPNM), Sungai Besi, Kuala Lumpur,Malaysia*

**Corresponding author E-mail:m.nazri@upnm.edu.my*

Abstract

This project thesis focuses on quality of voice and video performance in a wired as well as in a wireless network. The analysis was done by using network management system (NMS) to monitor and record the performance of different VoIP codecs in various use cases. Another important challenge is to define the best audio/video codec that can be used in an existing network to satisfy defined QoS parameters. We conclude on our findings that a separation between a wireless and wired environment wasn't reasonable, because the occurring differences of the QoS metrics were too small to measure with the used setup. During the testing of the different codecs the Speex codec reached the best result compared to the others. It can be said Speex is the codec with the least needed bandwidth utilization and therefore the outcome of the testing must be seen divided.

Keywords: *Delay; Jitter; Voice Over IP; codec; VoIP*

1. Introduction

Voice over Internet Protocol (VoIP) has the ability to transmit speech during packet switching in IP networks. The technology consists, a set of facilities and protocols for managing the transmission of voice packets using IP. In IP based computer networks, it is designed to handle data packets, it can arrive out of order or also can be lost and retransmitted. However, in voice conversations are non-tolerant of these kinds of disturbances. Each packet sound also must arrive in correct order, preserve the real time. Nevertheless, the possibilities of having packets lost the conversation sounds, distorted and has a bad audio quality are there. Furthermore, VoIP is extreme sensitive and, issues like staggered and packet loss are common compared to other network applications. Therefore, it is important that network management issues are satisfying defined through QoS parameters.

There are numerous network factors that affect QoS assessment. For example, the network congestion, link failures, routing instabilities or competing traffic. General QoS measurement includes delay, jitter and packet loss. However, several protocols and standards for VoIP do exist. VoIP is currently dominated by two standards. On the one hand the H.323 standard and on the other hand the SIP (Session Initiation Protocol). Both standards describe how a conversation is initiated over the IP network, the required components and the used transport and signaling protocols. Another important challenge is to define the best audio/video codec that can be used in an existing network to satisfy defined QoS parameters.

2. Related work

The reality nowadays, day by day, more people start to use Voice over IP to communicate and this happen to the entire world population. VoIP (voice over IP) is a containerize medium that serve to transmit voice and multimedia over Internet Protocol (IP) networks. Group of technologies and methodologies communicate through channel, level and place. Is either at enterprise level or in local area network or wide area network. VoIP does its job by encapsulate the given audio via codec and transform into several data packets before it transmit over IP network. Once it reaches the destination, the data decapsulates back into audio for caller or user to understand. VoIP endpoints include dedicated desktop VoIP phones, softphone applications running on PCs and mobile devices, and several applications.

A VoIP app is an application or software that end-user can install on computer or mobile device, and it connects to a VoIP services via Internet or a dedicated network. By doing this, it allows user to make VoIP calls. A VoIP is also known as a VoIP client and one of common application to use is softphone app. There are many common program, which make it easy to use VoIP for example Skype, MSN messenger, VoIP-cheap, VoIP-buster, etc. [1]. Today, various of research started to concentrate on wireless technology implementation on VoIP service.

In the digital era, the increase of network bandwidth and the ubiquitous wireless access facilitates become major concern and, -network. Among these services, is Voice over Internet Protocol (VoIP)[2].

Wireless LAN is one of organized wireless technologies that use by all over the world and, it play a major role in next-generation

Copyright © 2018 Authors. This is an open access article distributed under the Creative Commons Attribution License, which permits unrestricted use, distribution, and reproduction in any medium, provided the original work is properly cited.

wireless voice call networks. The architecture is similar as Local Area Network (LAN)'s except the transmission happens via radio frequency (RF) or Infrared (IR) and not through physical wired cables, and at the MAC sub-layer, are use different standard protocol.

The main characteristics of the WLAN technologies are mobility, simplicity, scalability, edibility and cost effectiveness. Today, many organizations are using WLANs as a medium for communication, so it is important to investigate how VoIP over WLAN performs based on previous study [3]. VoIP over Wireless LAN (WLAN) faces many challenges, due to the loose nature of wireless network. Issues like providing QoS at a good level, dedicating capacity for calls and having secure calls is more difficult rather than wired LAN [16, 17, 18].

Therefore, VoIP over WLAN remains as challenging research topic. Wireless VoIP applications makes WLAN looks inefficient due to underutilize resources. Due to large overhead in transmitting small packets (in 802.11 WLAN), the bandwidth only available for VoIP traffic and it is far less than its maximal 11Mbps data rate it currently supports [4]. There has been much activity in the area of WLAN performance analysis in the last few years. For example, Bianchi did their work in this area by developing the analytical model [5], [6], [7] to compute the saturation throughput of the Distributed Coordination Function (DCF) scheme. Other 802.11 related researches have focused on approaches to adapting system parameters [8], [9], [10].

3. Methodology

In this study, we have define to evaluate the performance of VoIP over wireless technology such as i) evaluation /planning phase; ii) Implementation phase; and iii) Analysis Phase (refer to Figure 1)

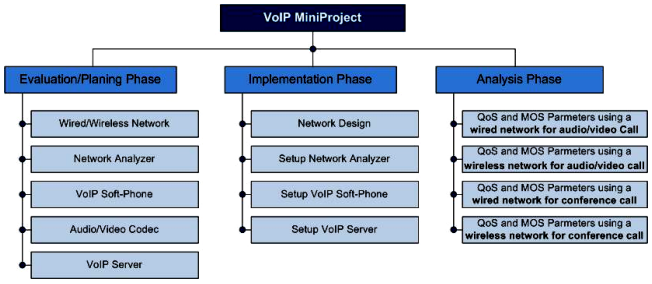


Fig. 1: Work Breakdown Structure Approach

As mentioned in the introduction several protocols and codecs for VoIP do exist. Figure 2 shows the different protocols used on different network layers

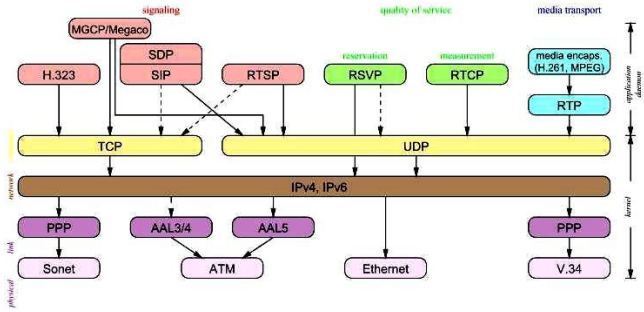


Fig 2: Protocols used on different network layers in VoIP

4. SIP technology

4.1 SIP and signaling protocol

VoIP is currently dominated by H.323 standard and the Session Initiation Protocol (SIP). For this project SIP was used, because SIP is very well supported by many Softphones and VoIP management tools. Hence, application such as SIP is needed. SIP is an application layer that control signaling protocol for creating, modifying, and terminating sessions with one or more participants. The SIP commonly interoperates with i) SDP— describe as the message payload content and characteristics SAP. At same time, use in advertising multimedia session via multicast; ii) RSVP—To reserve network resources for providing QoS ; iii) RTP—For real-time transmission; iv) RTSP—for controlling delivery of streaming media RADIUS —For authentication; v) LDAP—For location discovery.

4.2 SIP server

SIP Servers are essential work under network elements, which enable SIP endpoints to exchange messages and register user locations. At the same time, SIP Servers also enable network operators by installing route and security policies, authenticate users and manage user locations. The SIP Server has three general types of functionality: i) SIP Registrar Server—handles location registration messages and ii) SIP Redirect Server—returns "contact this address" responses. SIP Proxy Server—forwards SIP requests and responses [14, 15].

After evaluating different SIP Servers the decision was made to use the Brekeke SIP Server. The main advantages of the Brekeke SIP Server are that it's free for education usage and runs with Windows XP. Furthermore the administration of the server can easily be done with the help of a web interface.

4.3 Audio/video codec

A codec, which stands for coder-decoder, converts an audio/video signal into compressed digital form for transmission and return back into uncompressed audio/video signal for replay. It also converts each sample into digitized data and compresses it for transmission. Some codecs also support silence suppression, where silence is not encoded or transmitted.

Every VoIP phone contains one or more codecs, and during call establishment, they share their lists of supported codecs such as [11], [12], [13] AMR Codec, BroadVoice Codec 16Kbps narrowband, and 32Kbps wideband, DoD CELP - 4.8 Kbps, GIPS Family - 13.3 Kbps and up, GSM - 13 Kbps (full rate), 20ms frame size, iLBC - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size, G.711 - 64 Kbps, sample-based Comes in two flavors: A-law and mu-law, G.722 - 48/56/64 Kbps ADPCM 7Khz audio bandwidth, G.722.1 - 24/32 Kbps 7Khz audio bandwidth (based on Polycom's SIREN codec), G.723.1 - 5.3/6.3 Kbps, 30ms frame size, G.726 - 16/24/32/40 Kbps, G.728 - 16 Kbps, G.729 - 8 Kbps, 10ms frame size, Speex - 2.15 to 44.2 Kbps

4.4. Softphone

Softphone is an software application to make a telephone calls over the Internet by using a general purpose computer, rather than using dedicated hardware. X-Lite is a proprietary freeware VoIP soft phone that uses SIP. During the project the evaluation was made to work with different softphone applications. Finally the decision was made to use X-Lite, because it's freeware, easy to use and works fine with the Network Monitoring tool.



Fig. 3: X Lite Softphone Interface

4.5. Quality of service (QoS)

Quality of Service (QoS) is one of major issue in VoIP implementations. The issue is related in, how to guarantee that packet traffic for a voice or video connection will not be delayed or dropped due interference from other lower priority traffic. The three most common quality issues affecting VoIP are Latency, Jitter, Packet Loss.

4.6. VQManager

The VQManager is a web-based real-time QoS monitoring tool for VoIP networks. It provides high level call quality scores for each call-both MOS and R factor and detailed analysis of the factors affecting call quality.

It has been chosen because it enables to monitor VoIP network for voice quality, call traffic, bandwidth utilization and keep track of active calls and failed calls. VQManager can[18] also calculate QoS statistics from the RTP streams. For complete QoS reporting by this method, it is necessary that RTP packets from both participants of a call are available at the listening interface (NIC)[18].

5. Experimental, analysis and results

In this project three scenarios of audio/video communication among VoIP were evaluated. These scenarios will be configured and implemented in real network platform. We have setup i) scenario 1 – Wireless Audio/Video call using SIPGate (refer to figure 4); ii) Scenario 2 - Wired Conference Call between 3 parties using local SIP Server (refer to figure 5); iii) Scenario 3 - Wireless conference call using local SIP Server (refer to figure 6)

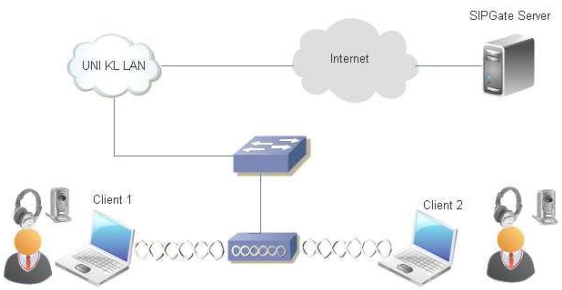


Fig. 4: Wireless Audio/Video call using SIPGate

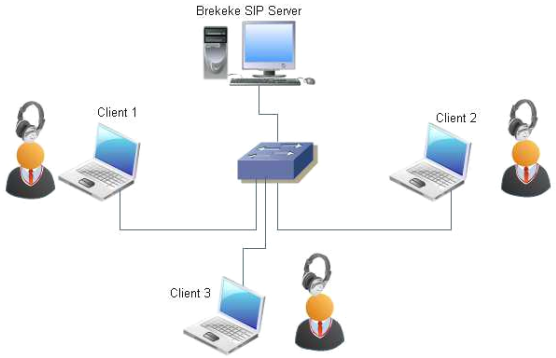


Fig. 5: Wired conference Call between 3 parties using local SIP Server

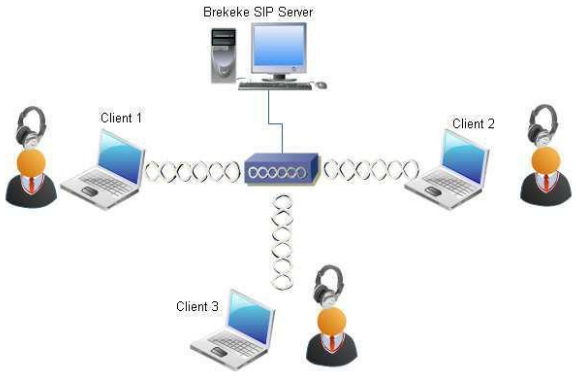


Fig. 6: Wireless conference call using local SIP Server

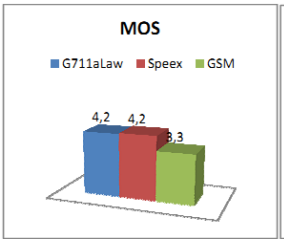
Wired Conference Call - The comparison of three codecs in a wired network brought following results:

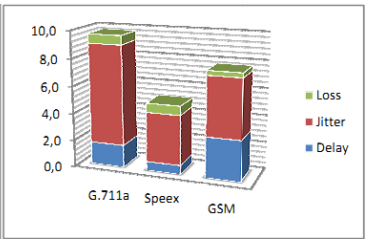
The best MOS factor was achieved with the Speex and G711 Codec (figure 14). The minimum bandwidth utilization was required by the Speex codec. The worst result was achieved by the GSM codec with a MOS of 3.3 and an average delay of 3ms. Moreover, also the bandwidth utilization by GSM compared to Speex was relatively high. To summarize, the Speex codec achieved the best audio quality with the lowest bandwidth utilization (figure 15) for a wireless conference call between three parties.

Table 1: Wired Conference Call Rating

Wired Conference Call												
Codec	G711aLaw				Speex				GSM			
Call Nr.	1	2	3	Ø	1	2	3	Ø	1	2	3	Ø
Delay [ms]	2	1	2	1,7	0	0	2	0,7	3	2	4	3,0
Jitter [ms]	10	2	10	7,3	2	1	8	3,7	2	3	8	4,3
Loss [%]	1	0	1	0,7	0	1	1	0,7	0	1	0	0,3
MOS	4,1	4,2	4,2	4,2	4,3	4,4	4	4,2	3,3	3,4	3,1	3,3
R	94	95	93	94,0	93	94	93	93,3	61	61	60	60,7
B [kbit/s]	340	335	339	338,0	103	106	105	104,7	141	141	138	140,0

Wireless Conference Call												
Codec	G711aLaw				Speex				GSM			
Call Nr.	1	2	3	Ø	1	2	3	Ø	1	2	3	Ø
Delay [ms]	3	1	2	2,0	1	0	1	0,7	6	2	3	3,7
Jitter [ms]	11	7	10	9,3	3	1	4	2,7	2	3	8	4,3
Loss [%]	0	1	0	0,3	0	1	1	0,7	0	1	0	0,3
MOS	4,2	4,4	4,2	4,3	4,4	4,4	4,3	4,4	3,1	3,5	3,3	3,3
R	93	93	92	92,7	93	94	93	93,3	61	60	60	60,3
B [kbit/s]	340	340	339	339,7	104	104	105	104,3	142	146	144	144,0





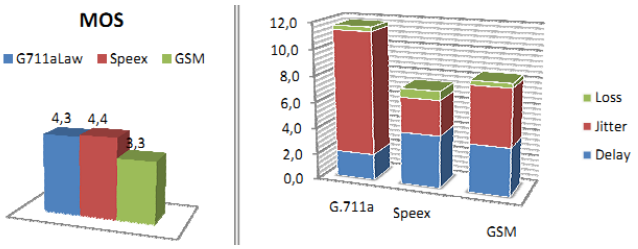


Fig. 7: Codecs - Wired Conference Call Performance

Wireless Conference Call - There were no measurable differences between the wireless and wired infrastructure. Therefore, also in this test case the Speex codec achieved the best audio quality with the lowest bandwidth utilization.

6. Conclusion

We conclude on our findings that a separation between a wireless and wired environment wasn't reasonable, because the occurring differences of the QoS metrics were too small to measure with the used setup.

During the testing of the different codecs the Speex codec reached the best result compared to the others. It can be said Speex is the codec with the least needed bandwidth utilization and therefore the outcome of the testing must be seen divided. As already mentioned earlier the testing environment was an independent network and therefore it is hard to tell if Speex really is the best codec in every environment. For a significant testing of different codecs it is always important in which environment the testing takes place or where the codec should be implemented later on. Hence, it would be necessary to test the codecs in its place of installation over several weeks during peaks and also lows of network traffic.

At the end of the prior mentioned testing a real significant outcome can be given on which codec should be used in which environment and not based on a testing in an independent network which would be never a real place of installation.

References

[1] M. N. Ismail, "A Case Study: Server Performance Measurement for E-Learning System in Campus Environment". *Journal Annals. Computer Science Series, Romania*, Vol. IX, Fascicula I-a, pp. 9–30, 2011.

[2] Andrusyszyn, Mary A, Soeren, Mary V, Laschinger, Heather S, Goldenberg, Dolly, DiCenso A (1999). Evaluation of Distance Education Delivery Methods for a Primary Care Nurse Practitioner Program. J. Dist. Edu [Online], Available: http://cade.athabascau.ca/vol14.1/andrusyszyn_et_al.html

[3] (1999) The Rights Stuff: Ownership in the Digital Academy. [Online]. Available: http://thenode.org/ltreport/ip

[4] Thompson, Herb (1999). The Impact of Technology and Distance Education: A Classical Learning Theory Viewpoint. Education Technology & Society [Online], Available: http://ifets.ieee.org/periodical/vol_3_99/herb_thompson.pdf

[5] (1997) Models of Distance Education. University System of Maryland, Institute for Distance Education Website. [Online]. Available: http://www.umuc.edu/ide/modlmenu.html

[6] Distance Education at a Glance (2003). University of Idaho, Engineering Outreach Website. [Online]. Available: http://www.uidaho.edu/eo/distglan.html

[7] H. Koppelman, H. Vranken, "Experiences with a synchronous virtual classroom in distance education", *ITiCSE '08: Proceedings of the 13th annual conference on Innovation and technology in computer science education*, 4(3): 194-198, 2008

[8] M. Massoth, R. Korn, "The IP multimedia subsystem with an e-learning management system and integrated video conferencing", *Proceeding iiWAS '08 Proceedings of the 10th International Conference on Information Integration and Web-based Applications & Services*, pp: 655-658, 2008

[9] V. Roesler, R. Husemann, C. H. Costa, "A new multimedia synchronous distance learning system: the IVA study case", *SAC '09: Proceedings of the 2009 ACM symposium on Applied Computing*, pp: 1765-1770, 2009

[10] S. Sawai, K. Donyaprueth, J.D. Nicholas, H. Paul, "The study of e-learning technology implementation: A preliminary investigation of universities in Thailand", *Education and Information Technologies*, 11(2):137 – 160, 2006

[11] Mohd Nazri Ismail, 'Implementation of Secure Real Time Transport Protocol on VoIP over Wired in Campus Network Environment', International Journal of Innovation, Management and Technology, Vol. 1, No. 3, pp. 244-247, August 2010.

[12] Mohd Nazri Ismail, 'Study the Best Approach Implementation and Codec Selection for VoIP Over Virtual Private Network', The International Arab *Journal* of Information Technology (IAJIT), Vol. 10, No. 2, pp. 198-203, 2013

[13] Mohd Nazri Ismail, 'Best VoIP CODECS Selection for VoIP Communication Over Internet Service Provider Carriers Network', International Journal of Academic Research, Vol. 4 Issue 4, p206-212, 2012

[14] Liam Murphy. "Adapting WLAN MAC parameters to enhance VoIP call capacity", Proceedings of the 8th ACM international symposium on Modeling analysis and simulation of wireless and mobile systems - MSWiM 05 MSWiM 05, 2005

[15] M. Davis. "Gauging VoIP Call Quality from 802.11 WLAN Resource Usage", 2006 International Symposium on a World of Wireless Mobile and Multimedia Networks(WoWMoM 06), 2006

[16] Mohd Nazri Ismail. "chapter 3 Best Approach for Video Codec Selection Over VoIP Conversation Using Wireless Local Area Network", IGI Global, 2013

[17] Xiuzhong Chen, Chunfeng Wang, Dong Xuan, Zhongcheng Li, Yinghua Min, Wei Zhao. "Survey on QoS management of VoIP", 2003 International Conference on Computer Networks and Mobile Computing, 2003. ICCNMC 2003., 2003

[18] Nurul Syeilla Syazhween Zulkefli, Nurazzah Abd Rahman, Zainab Abu Bakar, Sharifalillah Nordin et al. "Evaluation of Triple Indices in Retrieving Web Documents", 2013 International Conference on Advanced Computer Science Applications and Technologies, 2013

h
-
n
e
t
w
o
r
k
s
-
e
x
t
e
n
d
-
m
i
l
i
t
a
r
y
-
c
o
m
m