International Journal of Trend in Scientific Research and Development (IJTSRD)
Volume 3 Issue 5, August 2019 Available Online: www.ijtsrd.com e-ISSN: 2456 – 6470

Performance Analysis between H.323 and SIP over VoIP

Thet Zaw Aye

Lecturer, Faculty of Information and Communication Technology,
University of Technology (Yatanarpon Cyber City), Pyin Oo Lwin, Myanmar

How to cite this paper: Thet Zaw Aye "Performance Analysis between H.323 and SIP over VoIP" Published in International Journal of Trend in Scientific Research and Development (ijtsrd), ISSN: 2456-6470, Volume-3 | Issue-5, August 2019, pp.1324-1328, **https://doi.org/10.31142/ijtsrd26647**

IJTSRD26647

ABSTRACT

There are a number of protocols that may be employed in order to provide the Voice over IP (VoIP) communication services. In VoIP system, H.323 and Session Initiation Protocol (SIP) are the two major standards. Both of these signaling protocols provide mechanisms for multimedia teleconferencing services. Although the two protocols architecture is quite similar, they have many differences. This system presents Voice/Video over IP communication and summarizes the differences and performance of two major VoIP protocols, H.323 and SIP according to the packet delay variation, jitter, packet loss, and Packet end-to-end delay. It is found that both of them are non-interoperable, approaching each other, their focus and applicability is still different. In this paper, the system is designed and configured by Graphical Network Simulator (GNS3) and analyzed performance by Opnet Modeler Simulation.

KEYWORDS: *VoIP; H.232; SIP; performance*

I. INTRODUCTION

A signaling protocol is a type of protocol used to identify signaling encapsulation. Signaling is used to identify the state of connection between telephones or VoIP terminals (IP phones or PCs). The most popular signaling protocol that used in VoIP are H.323 and Session initiation protocol (SIP).

H.323 is a recommendation from the International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) that defines the protocols to provide audio-visual communication sessions on any packet network. The H.323 standard addresses call signaling and control, multimedia transport and control, and bandwidth control for point-to-point and multi-point conferences. It is widely implemented by voice and videoconferencing equipment manufacturers, is used within various Internet real-time applications such as GnuGK and NetMeeting and is widely deployed worldwide by service providers and enterprises for both voice and video services over IP networks.

The system is demonstrated VoIP network with specified protocol components. This system is configured H.323 based VoIP network and SIP based VoIP network and then test the VoIP call and collect the performance parameters. The same network topology is used for both H.323 and SIP. The main goal of this system is to compare the proposed signaling protocols and to evaluate them based on some performance metrics such as jitter, delay variation, end to end delay, and packet loss. This evaluation is performed theoretically by simulation.

II. H.232 PROTOCOL

H.323 was the first VoIP standard to adopt the Internet Engineering Task Force (IETF) standard Real-time Transport Protocol (RTP) to transport audio and video over IP networks. One strength of H.323 was the relatively early availability of a set of standards, not only defining the basic call model, but also the supplementary services needed to address business communication expectations.

H.323 is a system specification that describes the use of several ITU-T and IETF protocols. The protocols that comprise the core of almost any H.323 system is: H.225.0 Registration, Admission and Status (RAS), which is used between an H.323 endpoint and a Gatekeeper to provide address resolution and admission control services.

The H.323 system defines several network elements that work together in order to deliver rich multimedia communication capabilities. Those elements are Terminals, Multipoint Control Units (MCUs), Gateways, Gatekeepers, and Border Elements. Collectively, terminals, multipoint control units and gateways are often referred to as endpoints. H.323 uses port number 1720.

While not all elements are required, at least two terminals are required in order to enable communication between two people. In most H.323 deployments, a gatekeeper is employed in order to, among other things, facilitate address resolution.

III. SESSION INITIATION PROTOCOL

SIP was designed to provide a signaling and call setup protocol for IP-based communications supporting the call processing functions and features present in the public switched telephone network (PSTN) with a vision of supporting new multimedia applications. It has been extended for video conferencing, streaming media distribution, instant messaging, presence information, file transfer, Internet fax and online games.

SIP is distinguished by its proponents for having roots in the Internet community rather than in the telecommunications industry. SIP has been standardized primarily by the IETF, while other protocols, such as H.323, have traditionally been associated with the ITU.

The protocol defines the specific format of messages exchanged and the sequence of communications for cooperation of the participants. SIP is a text-based protocol, incorporating many elements of the Hypertext Transfer Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP). A call established with SIP may consist of multiple media streams, but no separate streams are required for applications, such as text messaging, that exchange data as payload in the SIP message.

SIP works in conjunction with several other protocols that specify and carry the session media. Media type and parameter negotiation and media setup is performed with the Session Description Protocol (SDP), which is carried as payload in SIP messages. SIP is designed to be independent of the underlying transport layer protocol, and can be used with the User Datagram Protocol (UDP), the Transmission Control Protocol (TCP), and the Stream Control Transmission Protocol (SCTP). For the transmission of media streams (voice, video) SIP typically employs the RTP Protocol or the Secure Real-time Transport Protocol (SRTP).

IV. H.323 SYSTEM DESIGN AND IMPLEMENTAION

This system has two H.323 zones and three main components as H.323 gatekeeper, H.323 gateway and H.323 clients. H.323 gatekeeper provides zone mapping as local and remote H.323 zones and bandwidth control, proxy support and account registration for each client. H.323 gateway allow connections from H.323 to H.323 network and provide dial numbers and H.323 trunk. H.323 clients are generally softphones that provide H.323 protocol for VoIP calls.

Figure 1 shows H.323 system design with two zones. Zone one is controlled and mapped by GK1 and zone two is controlled and mapped by GK2. Clients call through the service provider.

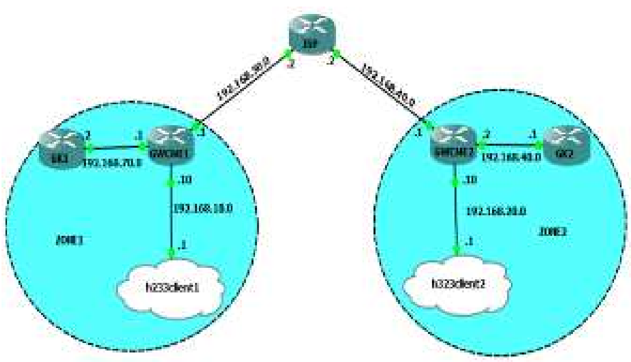

Figure1. Multi-zone H.323 System Design

A. H.323 Gateway Implementation

Figure 2 shows that how to configure H.323 gateway for zone 1 (GW1) in console. Gateway assigns VoIP interface that receives media and control data from clients. Gateway registers to local gatekeeper for endpoints registration and bandwidth control in general. Gateway of zone 2 (GW2) is also configure as like this with appropriate IP addresses.

```
interface FastEthernet0/0
 ip address 192.168.10.10 255.255.255.0
 load-interval 30
 duplex auto
 speed auto
 h323-gateway voip interface
 h323-gateway voip id gk1 ipaddr 192.168.70.2 1719
 h323-gateway voip h323-id CME1
 h323-gateway voip tech-prefix 1#
 h323-gateway voip bind srcaddr 192.168.10.10
```

Figure2. H.323 Gateway (GW1) Assigns Gatekeeper and VoIP Interface

Figure 3 shows how to configure H.323 phone registration in GW1. In this system, gateway serves as call manager express to access phone registration and it allows maximum four phones for register. Gateway uses username and password for authentication and g711 voice codec for client's voice data encoding. In the following figure, the dial number 5001 is registered with "Nay" as user name and "1234" as password. In GW2 of zone 2, another dial number is configured like this process.

```
voice register global
 mode cme
 max-dn 4
 max-pool 4
!
voice register dn  1
 number 5001
 name Nay
 label Nay
!
voice register pool  1
 id mac 0200.4C4F.4F50
 number 1 dn 1
 username Nay password 123456
 codec g711ulaw
 no vad
```

Figure3. Dial Number 5001 Registers to Gateway (GW1)

Figure4. Show how gateway route VoIP data to another gateway over the H.323 trunk link and defines source address of telephony service. The gateways of both zone 1 and zone 2 are configured with the address which are described in system design.

```
dial-peer voice 6000 voip
 destination-pattern 60..
 session target ras
 no vad
!
!
gateway
 timer receive-rtp 1200
 security password 135445415F5952 level all
!
!
!
telephony-service
 ip source-address 192.168.10.10 port 2000
 max-conferences 8 gain -6
 transfer-system full-consult
```

Figure4. H.323 Gateway (GW1) Route VoIP Traffic to Remote Gateway

B. H.323 Gatekeeper Implementation

Figure 5 shows how to configure gatekeeper of zone 1 and zone 2 to map local and remote zones, bandwidth control and proxy support.

```
gatekeeper
 zone local GK1 test.com 192.168.70.2
 zone remote GK2 test.com 192.168.80.1 1719
 gw-type-prefix 1#*
 use-proxy GK1 remote-zone GK2 inbound-to gateway
 use-proxy GK1 default inbound-to gateway
 bandwidth check-destination
 load-balance
 no shutdown
!
```

```
gatekeeper
 zone local GK2 test.com 192.168.80.1
 zone remote GK1 test.com 192.168.70.2 1719
 gw-type-prefix 1#*
 use-proxy GK2 remote-zone GK1 inbound-to gateway
 use-proxy GK2 default inbound-to gateway
 bandwidth check-destination
 load-balance
 no shutdown
!
```

Figure5. H.323 Gatekeeper Zone Mapping and Control

Figure 6 shows active endpoints that registered to gatekeepers using H.323 ports, RAS signal addresses and H.323 IDs.

```
GK1(config)#do sh gatekeeper endpoints
                    GATEKEEPER ENDPOINT REGISTRATION
                    ================================
CallSignalAddr  Port  RASSignalAddr   Port  Zone Name         Type    Flags
--------------- ----- --------------- ----- ---------         ----    -----
192.168.10.1    1720  192.168.10.1    51909 GK1               TERM
    H323-ID: 5001
192.168.10.10   1720  192.168.10.10   63237 GK1               H323-GW
    H323-ID: CME1
    Voice Capacity Max.=  Avail.=  Current.= 0
Total number of active registrations = 2

GK2(config)#do sh gatekeeper endpoints
                    GATEKEEPER ENDPOINT REGISTRATION
                    ================================
CallSignalAddr  Port  RASSignalAddr   Port  Zone Name         Type    Flags
--------------- ----- --------------- ----- ---------         ----    -----
192.168.20.10   1720  192.168.20.10   51898 GK2               H323-GW
    H323-ID: GWCME2
    Voice Capacity Max.=  Avail.=  Current.= 0
Total number of active registrations = 1
```

Figure6. Registered Endpoints at Gatekeepers

V. SIP SYSTEM DESIGN

In this system, SIP was configured as two sides SIP covered networks. It has two gateways and two SIP clients. Unlike H.323, gatekeeper isn't required in SIP. The network design for SIP is shown in Figure 7.

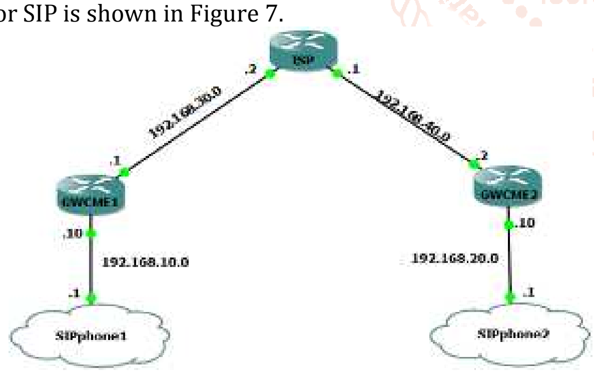

Figure7. Multi-Side SIP Covered Network

a. SIP Gateway Implementation

Figure 8 shows SIP gateway allows SIP to SIP calls and binds media and control source interface to receive VoIP data from clients using UDP port.

```
GWCME2(config)#voice service voip
GWCME2(conf-voi-serv)#allow-connections sip to sip
GWCME2(conf-voi-serv)#redirect ip2ip
GWCME2(conf-voi-serv)#sip
GWCME2(conf-serv-sip)#bind control source-interface f0/0
GWCME2(conf-serv-sip)#bind media source-interface f0/0
GWCME2(conf-serv-sip)#session transport udp
GWCME2(conf-serv-sip)#registrar server
GWCME2(conf-serv-sip)#privacy user session
GWCME2(conf-serv-sip)#ex
GWCME2(conf-voi-serv)#ex
```

Figure8. Allowing Service and Binding Source Interface

Figure 9 shows SIP user agent configuration. It supports user authentication with username and password to clients. Then, it provides registration address and registration timer for clients.

```
sip-ua
 authentication username Nighfroz password 7 08701E1D5D4C53
 retry invite 2
 retry response 2
 retry bye 2
 retry cancel 2
 retry register 2
 registrar ipv4:2.2.2.2 expires 3600
 permit hostname dns:test.com
!
```

Figure9. SIP User Agent Configuration

Figure 10 and Figure 11 show configuration of defining phone numbers, number registration pools, account authentication and call manager express's VoIP source addresses.

```
voice register global
 mode cme
 source-address 192.168.20.10 port 5060
 max-dn 4
 max-pool 4
 authenticate realm local
 tftp-path flash:
!
voice register dn  1
 number 2001
 name Nig
 label Nig
!
voice register pool  1
 id mac 0200.4C4F.4F50
 number 1 dn 1
 username Nig password 123456
 codec g711ulaw
 no vad
```

Figure10. Number Defining and Account Authentication in GW1

```
voice register global
 mode cme
 source-address 192.168.10.10 port 5060
 max-dn 4
 max-pool 4
 authenticate realm local
 tftp-path flash:
!
voice register dn  1
 number 1001
 name Nay
 label Nay
!
voice register pool  1
 id mac 0200.4C4F.4F50
 number 1 dn 1
 username Nay password 123456
 codec g711ulaw
 no vad
```

Figure11. Number Defining and Account Authentication GW 2

Figure 12 show SIP trunk implementation to route VoIP data from one gateway to another. It provides target address to route data to remote gateway through the trunk.

```
dial-peer voice 2000 voip
 destination-pattern 20..
 session protocol sipv2
 session target ipv4:192.168.40.2
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
```

```
dial-peer voice 1000 voip
 destination-pattern 10..
 session protocol sipv2
 session target ipv4:192.168.30.1
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

Figure12. SIP Trunk Configuration

VI. PERFORMANCE COMPARISON

This session compares the performance parameters like jitter, end-to-end delay, and delay variation, packet loss of H.323 and SIP in line graph. All tests are measure from zero to 100 seconds.

A. Jitter Comparison

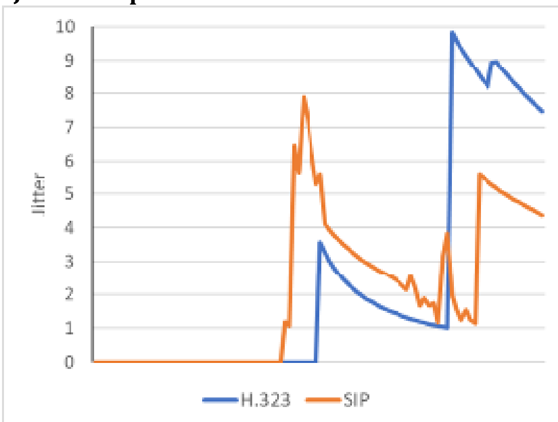

Figure13. Jitter Comparison (Phone1 Vs Phone3)

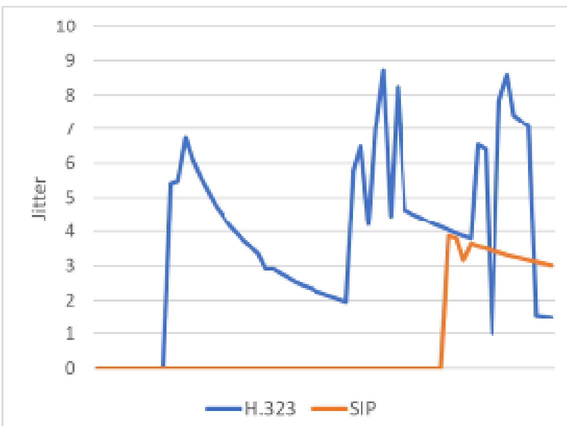

Figure14. Jitter (Phone2 Vs Phone4)

As Figure 13 and Figure 14 comparison graphs, H.323 has more jitter value in average than SIP. Jitter can generally be affected from packet loss and network delay and no steady packet generating. H.323 has more packet loss rate than SIP. Average peak jitter values occur when packet loss rate and packet delay rate are high.

B. Delay Variance Comparison

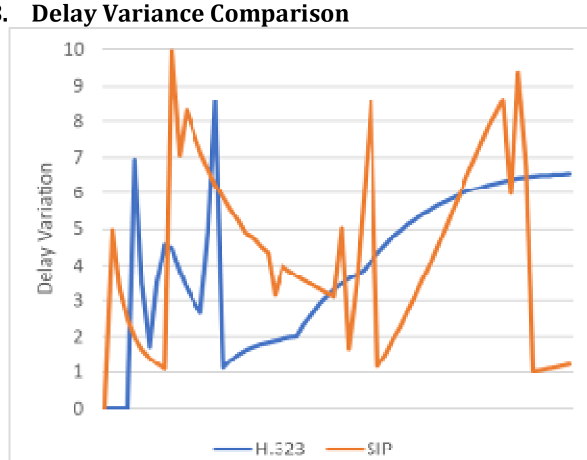

Figure15. Delay Variation Comparison (Phone1 Vs Phone3)

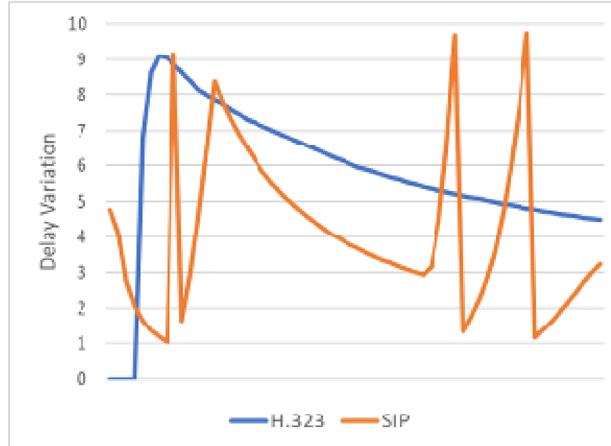

Figure16. Delay Variation (Phone2 Vs Phone4)

Delay variation is like jitter in meaning but delay variation ignores packet loss. Delay variation occurs when same size packets are not transmitted and received in same speed. As Figure 15 and Figure 16 comparison graph, H.323's delay variation is little more than SIP because H.323 gatekeeper's priority queuing mechanism is not enough to handle on packet transmission when many clients call simultaneously.

C. Delay Variance Comparison

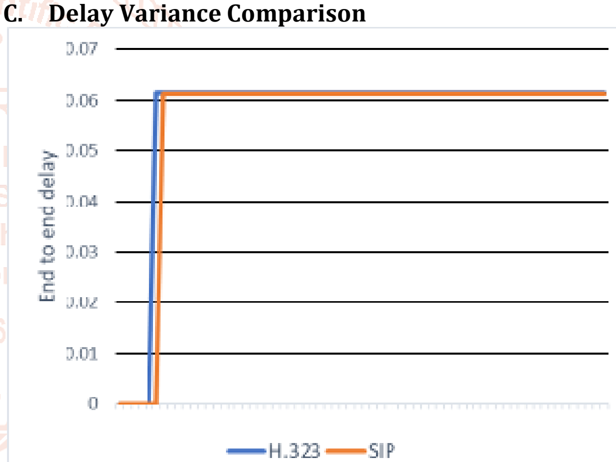

Figure17. End to End Delay Variation (Phone1 Vs Phone3)

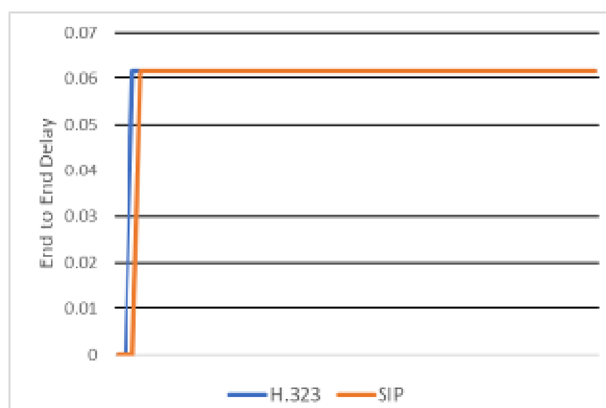

Figure18. End to End Delay (Phone2 Vs Phone4)

In end-to-end delay comparison graph, Figure 17 and Figure 18, the graphs indicate that there is no distinctly difference because end-to-end delay highly depends on network infrastructure and this system has same network design for two networks.

D. Packet Loss Comparison

Figure19. Packet Loss (Phone1 Vs Phone3)

Figure20. Packet Loss (Phone2 Vs Phone4)

According to Figure 19 and Figure 20, H.323 has highly packet loss rate. Frame handling is really important to reduce the packet loss. H.323 frame handling is not good in handling of simultaneous interactive media stream and so, packets are sent twice that are already arrived.

VII. CONCLUSION

This paper present about the fundamental configuration of H.323 and Session Initiation Protocol (SIP) protocols. And the performance statuses of these protocols with some parameter like jitter, delay variation, end to end delay and packet loss are also described. As the performance comparison graphs described above, the results show that SIP protocol is better performance than H.323 in VoIP signaling for reliability. H.323 is not flexible with internet but SIP is quite flexible and so SIP is better to choose for business and personal uses of today.

REFERENCES

[1] Guoyou He (Helsinki University of Technology) "H.323 Protocol Suite "Hasan, Md" A Study on VoIP"

[2] https://forums.oneplus.com/threads/about-sip-settings.862792/

[3] https://youthcarnival.org/bn/session-initiation-protocol-sip-overview-5-use- cases/

[4] https://www.cisco.com/c/en/us/support/docs/voice/h323/5244-understand-gatekeepers.html

[5] http://www.h323.org/

[6] https://www.hackthis.co.uk/articles/sip

[7] https://www.revolvy.com/page/Round%252 Dtrip-delay-time

[8] https://www.techopedia.com/definition/3041/jit ter-voip-voice-over-internet- protocol-voi