Comparison of Call Signalling Protocols for Ad-hoc Networks

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ABSTRACT
The two major standards in the multimedia services over IP area are the protocol suites H.323 (ITU-T) and SIP (IETF).Both have emerged as competing protocol standards for the signaling and call control of IP telephony. SIP is designed with a broader scope, offering functions specifically designed to enable easy extensions; it should be the advantage for new potential services. H.323 is still the more mature standard; H.323 provides better interoperability and interworking (PSTN, ISDN). We can assume a coexistence of both protocols. We compare voice performance parameters like talking time, total sessions, delay, jitter packet delivery ratio, call set up times and throughput, establish the differences in performance of these two VoIP protocols through an Qualnet simulation. SIP and H.323 are used for establishment and release of Voice over IP calls as well as Video and Media calls. Both these protocols play a very important role in terms of optimizing the call set up time and call reliability and flexibility over IP networks for real time applications like voice and video. The obtained results are discussed to highlight the impact of both H.323 and SIP in ad-hoc networks.

Keywords
Voice over IP (VOIP), H.323, SIP, Call Set up Time, End to End delay, internet telephony, signaling Protocol.

1. INTRODUCTION
Since the late 90's IP telephony, commonly referred to as Voice over IP (VoIP), has been presented as a revolution on communications enabling the possibility to converge historically separated voice and data networks, reducing costs, and integrating voice, data and video on applications. The ability to communicate properly over long distances has become an integral part of society today. Businesses are expanding to different regions in the world, but need to keep the same deadlines [2].This means it is necessary for employees in two different regions to communicate with each other over long distances, cheaply and trouble free. The public switched telephone network (PSTN) has developed itself to accommodate these requirements. But internet has become a very popular means of communication in a very short period of time. It was set up as a network where people could share files and access other people's work [7]. It has since established itself as a massive communication infrastructure that provides many services such as electronic mail .In the recent years it has further developed itself into providing Internet Telephony or Voice over internet protocol (VOIP). One of the most important parts of a telephone call is the establishment of the call itself. [1] In a packet switched network this is accomplished by a protocol. That performs signalling. This paper addresses such protocols, namely the well spread H.323 and Session initiation Protocol (SIP).

2. VOICE OVER INTERNET PROTOCOL (VOIP)
Voice over Internet Protocol (VoIP) is a general term for a family of transmission technologies for delivery of voice communications over IP networks such as the Internet or other packet-switched networks. Other terms frequently encountered and synonymous with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, and broadband phone.

3. H.323 BASIC ARCHITECTURE
H.323 is an umbrella Recommendation from the ITU 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, multimedia transport, and bandwidth control for point-to-point and multi-point conferences [4]. It is widely implemented by voice and videoconferencing equipment manufacturers; it is used within various Internet real-time applications like NetMeeting and is widely deployed worldwide by service providers and enterprises for both voice and video services over Internet Protocol (IP) networks [6].H.323's strength lies in multimedia communication functionality designed specifically for IP networks.

Distributed VOIP H.323 was originally developed for multimedia conferencing on LANs but was later extended to cover Voice over IP. The standard encompasses both point-to-point communications and multipoint conferences[11].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.

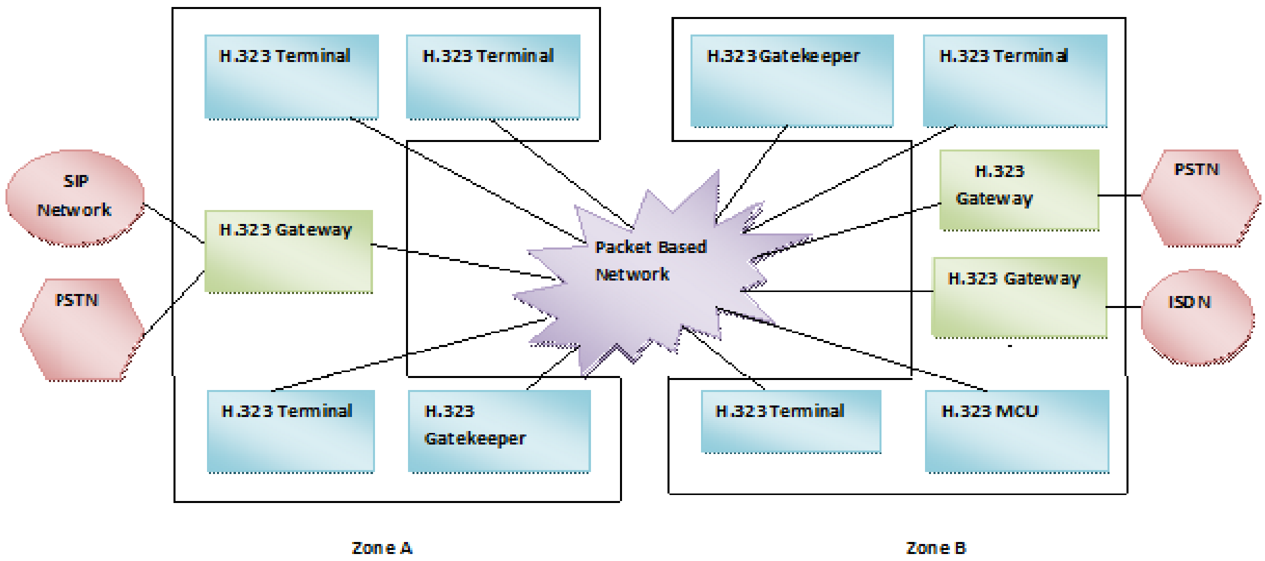

Fig 1: H.323 Architecture (Inspired from [4])

3.1 H.323 is "Multimedia over IP"

H.323 makes it possible to create and deploy new services quickly and to take advantage of multimedia capabilities. These services can embrace audio, video, and data conferencing. H.323 provides a strong foundation for new multimedia products and services [5]. Multimedia conferencing devices show the real potential of H.323 and multimedia communication. H.323 has very strong support for video.H.323 allow users to work side by side on a document using voice, video, text, and application sharing technologies.

Table 1. H.323 Signaling Statistics

No. of Calls Initiated	No. of Calls Initiated by a terminal.
No. of Calls Received	No. of Calls received by a terminal.
No. of Calls Established	No. of Calls established by a terminal.
No. of TCP Connection Rejected	No. of Calls rejected by a terminal.

3.2 Session Initiation Protocol (SIP)

Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IETF) standard designed for Initiating, maintaining and terminating interactive Communication sessions between users. These sessions may include voice, video, instant messaging, and interactive games.[9]. SIP makes minimal assumptions about the underlying transport and network layer Protocol, which can provide either a packet or byte stream service with either reliable or unreliable service [3]. A SIP system is based on a client/server model and is comprised of the following logical entities:

User Agent (UA) is an application that acts on behalf of the user, both as a client (User Agent Client) and as a server (User Agent Server). As a client it initiates SIP requests and as a server it accepts calls and responds to SIP requests made by other entities[11]. The user agent is usually part of a multimedia terminal whose media capabilities it controls without having any media capabilities of its own.

Registrar Server is a SIP server that accepts only registration requests issued by user agents [4]. A registrar server never forwards requests.

Location Server is a server which provides information to a proxy/redirect server about the possible current locations of a user. Usually, this entity is part of the proxy/redirect servers.

Redirect Server is a SIP server that provides address mapping services [12]. It responds to a SIP request destined to an address with a list of new addresses. A redirect server doesn't accept calls, doesn't forward requests nor does it initiate any of its own.

Proxy Server is a SIP server that acts both as a server to user agents by forwarding SIP requests and as a client to other SIPS servers by submitting the forwarded requests to them on behalf of user agents or proxy servers [8]. SIP providing equivalent Services through a simpler and more lightweight web based approach [8]. A SIP all is defined as the **multimedia** conference consisting of all participants invited by a common source.

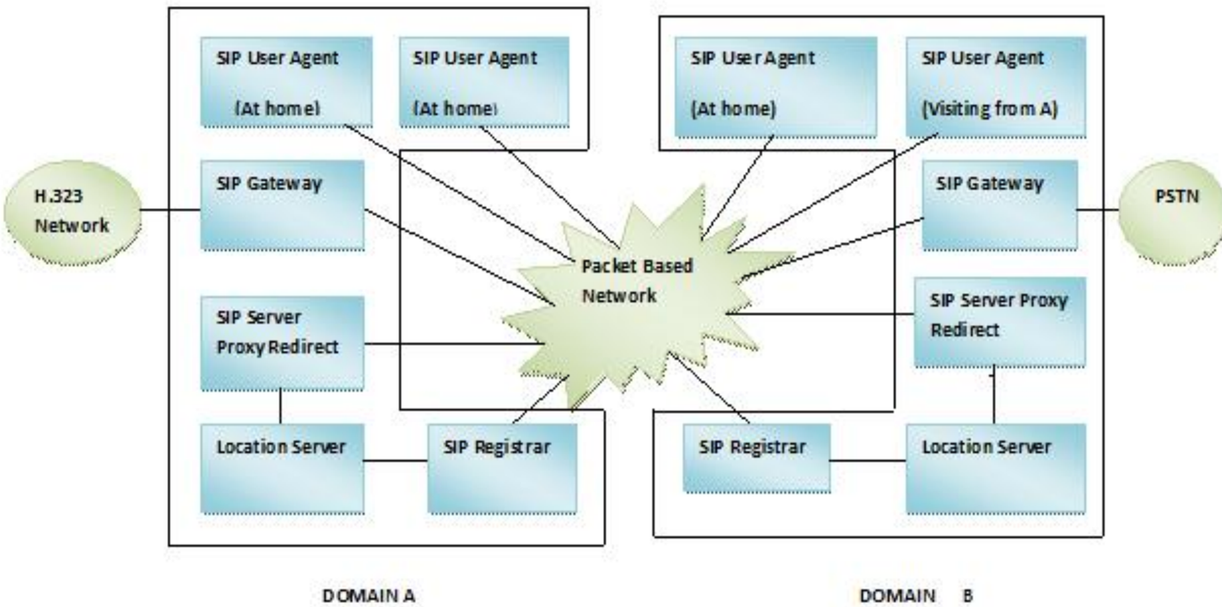

Fig 2: SIP Architecture (Inspired from [4])

Table 2. SIP Requests

Message Name	*Function*
INVITE	Invite user(s) to a session. The session description is contained in the body of the message, e.g. using the session description protocol (SDP)[8].The Session description contains the address where the host wants to receive media streams.
ACK	Acknowledgment of an INVITE request
BYE	Sent when a call is to be released
OPTIONS	Query user agents about capabilities
CANCEL	Cancel a pending request
REGISTER	Register with a SIP server

Table 3. Characteristics Comparison

Characteristics	*H.323*	*SIP*
Client Type	Intelligent	Intelligent
Network Intelligence and Services	Provided by Gatekeepers	Provided by Servers (Proxy, Redirect, Registrar)
Model Used	Telephony/Q.SIG	Telephony/Q.SIG
Addressing	E.164, URI, E-mail address	URI
Message Definition and Encoding	ASN.1 - Binary	ABNF - ASCII
Media Transport	RTP/RTCP, SRTP	RTP/RTCP, SRTP
Transport Protocol	TCP/UDP	TCP/UDP
Authentication and Encryption	H.235	HTTP (Digest and Basic) - SSL, PGP, S/MIME
Capability Negotiation	Good	Limited
PSTN Integration	Well suited	Non-Native

4. PERFORMANCE EVALUATION

4.1 Performance metrics:

Performance metric is a postulate that transforms the results of the task into measures of performance for drawing conclusions about the task objective .By these metrics the success or failure of tasks is evaluated. The metrics used to measure the performance of VOIP traffic is the Time between 1^{st} and last packet, talking time, and total sessions, no of packets, Avg packet size, jitter and throughput.

4.2 Simulation environment:

We use Qualnet simulator as our performance analysis platform. Various evaluation parameters include the time between 1st and last packet, no of packets, Avg packet size, delay, jitter and throughput of the simulated scenario. The simulation parameters are summarized in table 4. We designed the infrastructure mode scenario consisting of 11nodes. We are comparing the results of H.323 and SIP using 1 simulation and the application between the randomly chosen source and destination is VOIP traffic.

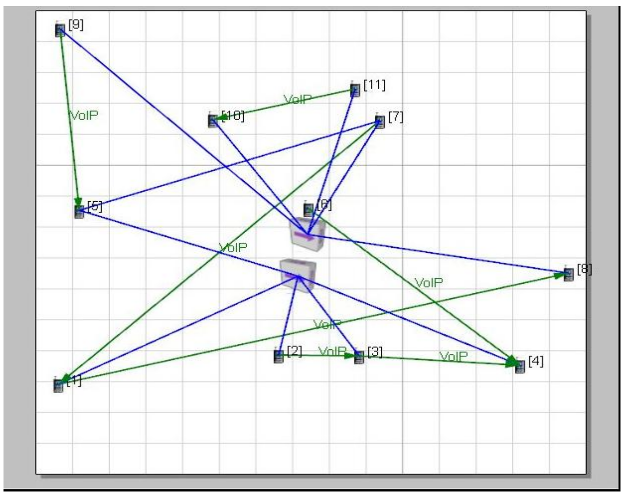

Fig 3: Snapshot of network in Qualnet Simulator

4.3 Wireless subnet properties

1. Two channels are used for simulating separately the receiving and transmitting frequencies.
2. Listenable and listening mask are set accordingly.

4.4 Application layer parameters

VOIP (H.323, SIP) applications are chosen to verify the end to end simulation.
The statistical results obtained through Qualnet simulation shows successful implementation of traffic with respect to the call established between VOIP users.

5. RESULT ANALYSIS

In our simulation, we consider a network of 11nodes that are placed randomly within a 1500m X 1500m area. Here H.323 and SIP application is used to simulate the same scenario in which we consider time between First packet and second packet .and no of packets sent and received. The parameters we used to configure comparative analysis of both protocols for simulation scenario are shown in Table 4. Session is initiated at 60s and ended at 240s. Thus, the total transmission time or session time is 180s.

Table 4 . Parameters for Simulation Evaluation

No. of nodes	11
File Name	Terminal-Alias-Address-File/DNS Address File
Source & destination position	1 & 11 respectively
Subnets	2
Application	VOIP(H.323/SIP)
Simulation time	15 Mins
Data traffic	VOIP
SEED	1
VOIP-CALL-TIMEOUT	60s
VOIP- Connection Delay	8s
Starting time	1Min
End time	4Min
No. of simulations	1

5.1 End to End Delay

The delay attribute indicates the acceptable transfer time of a packet from source to its destination. Mean delay is the average end-to-end delay of packets transmitted and 95- percentile delay is the time within 95 percent of packets has reached the destination. As shown in fig. 4, when we are increasing number of VoIP applications the delay at H.323 is 0.3124 and at SIP is 0.511,hence we can say that in case of delay H.323 is better that SIP.

5.2 Average Packet Size (PDR)

It refers to No of packets received at destination [1]. Following are the values of PDR at H.323 and SIP, according to figure 5.

(a).H.323 – 0.001272

(b). SIP - 0.0012128

5.3 Jitter

It is defined as the time variation of a periodic signal in electronics and telecommunications. Jitter can be quantified in the same terms as all time-varying signals. For better call signaling jitter should be minimum as shown in figure 6 the average jitter by using H.323 application is 0.01001 and in case of SIP average jitter is 1.155.Again it shows H.323 is better than SIP.

5.4 Throughput

Throughput is normally defined as time average of the number of bits per second that can be transmitted by every node to its destination. The throughput is usually measured in bits per second. The system throughput or aggregate throughput is the sum of the data rates that are delivered to all terminals in a network. According to our simulation results in figure 7, H.323 shows maximum throughput i.e. 336.647 bits per second and SIP has throughput of 293.64 bits per second

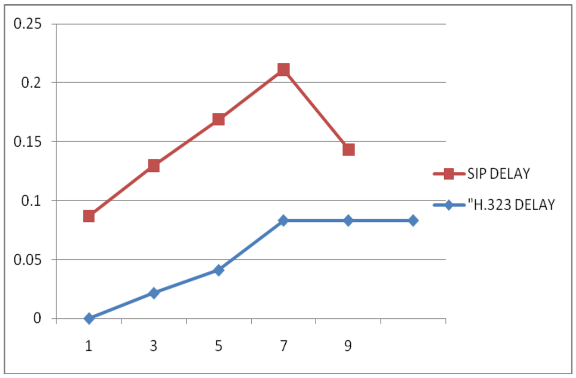

Fig 4 :Delays - H.323 & SIP vs. No of VOIP Applications

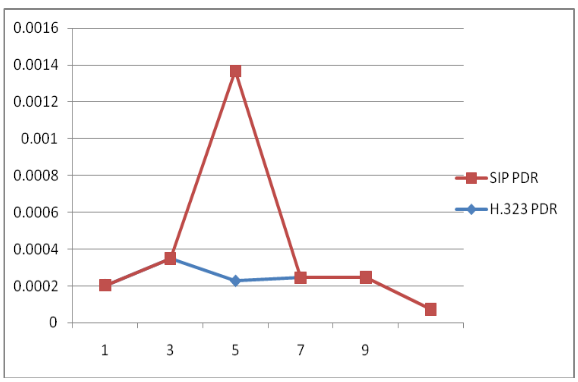

Fig 5: PDR - H.323 & SIP vs. No of VOIP Applications

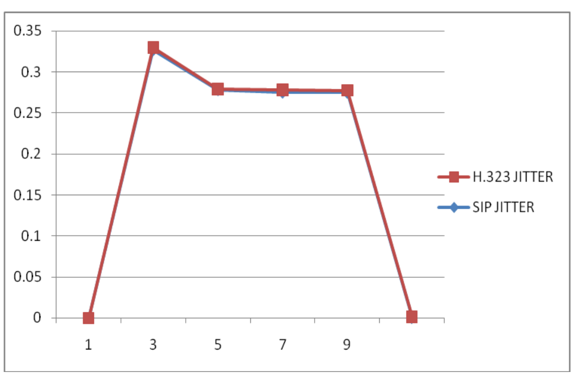

Fig 6: Jitter - H.323 & SIP vs. No of VOIP Applications

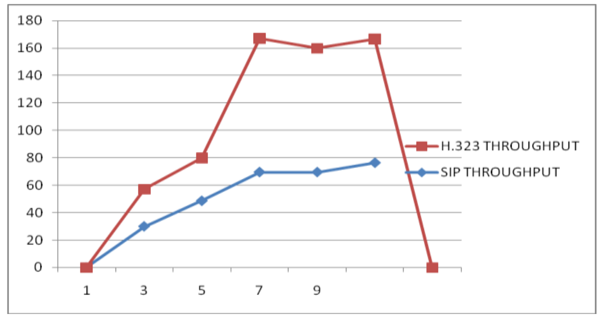

Fig 7: Throughputs - H.323 & SIP vs. No of VOIP Applications

6. CONCLUSION

H.323 has solid foundation and Technical capabilities are reasons for its success over SIP. However, SIP is less complex than H.323.

Therefore, nowadays, SIP is increasing its utilization whereas H.323, given its complexity, is being less used.

In this paper, we have compared these two signaling protocols based on four performance matrices as end to end delay, PDR, throughput and jitter.

Our observations clearly depicts that as we increase the number of VOIP applications, H323 performs better in regards to delay and SIP is better than H323 if compared on the basis of talking time.

The reality is that most H.323 products on the market today also support SIP, including SIP/H.323 interworking. It can be assumed, that neither of the two protocols will succeed over the other. They will probably coexist in different environments, bringing a strong required We can use H.323 and SIP for Global Communication There will be more efficient call handling and processing and also fast call set up time. This work may be extended for analyzing the behavior of these protocols in case of mobility with many more metrics for evaluation. And further we can add more complexities to this like port Reservation (in case of emergency services), QOS monitoring, third party call control etc. Further, a combined protocol scenario may also be presented suitable in different situations with varying network type and complexity.

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