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A Comparative Study of Signalling Protocols Used In VoIP

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ABSTRACT: Voice over Internet Protocol (VoIP) involves the transmission of telephone conversations using data connections that transmit packets of data on an Internet Protocol (IP). Unlike in the past, it's no longer necessary to have separate networks for voice and data. For establishing a VoIP Communication we need to establish the connections between the users, for this signaling protocol is used. In this paper, we compare some of the protocols available for VoIP such as Session Initiation Protocol (SIP), InterAsterisk eXchange Protocol (IAX) and H.323 protocol.

KEYWORDS: VoIP; signalling protocol; SIP; H.323; IAX

I. INTRODUCTION

VoIP allows users to make calls over the internet. It allows the user to make call to others who are also using VoIP services. It is another way of making phone calls, it is much cheaper compared to traditional PSTN (public switched telephone network) as it consumes data. VoIP has recently become very famous as it is offered by many mobile apps. The most famous provider of VoIP is Skype.

To call via VoIP first we have to establish the connection. Signaling protocols are used to connect a call and end call, locate users and negotiate capabilities. Some of these protocols are Session Initiation Protocol (SIP)[1], H.323[3][4], InterAsterisk eXchange Protocol (IAX)[2]. The actual conversation occurs over Real-Time Protocol (RTP). The RTP is used to transmit voice, but it is useful for the transmission of any data, such as video streams etc. It is worth noting that even if there are number of signaling protocols available, most of them use RTP protocol for data transmission, though not all. Combining Signaling protocols and RTP we can establish VoIP calls.

II. VOIP PROTOCOLS

Session Initiation Protocol (SIP)

The SIP[8] is Signaling protocol by Internet Engineering Task Force (IETF) used in VoIP. SIP is a signaling protocol, which means that its purpose is to establish connection, coordinate and terminate communication session between two endpoints peers. Comparing SIP to a traditional telephone we can say that the ringing of a phone, the busy tone and the ending of a call are the function that the SIP protocol is responsible for.

A SIP network consists of the following entities: end points, a proxy and/or redirect server, location server, and a registrar[6]. End points or User Agents (UA) represent phone devices or software modems. SIP users are not bound to specific devices; they register themselves with the registrar and use a special form of address resolution to identify other users. SIP user identification is based on a special type of Uniform Resource Identifier (URI) called SIP URI, similar to email addresses[7]. A location server stores the address bindings of users when they register themselves with the registrar.

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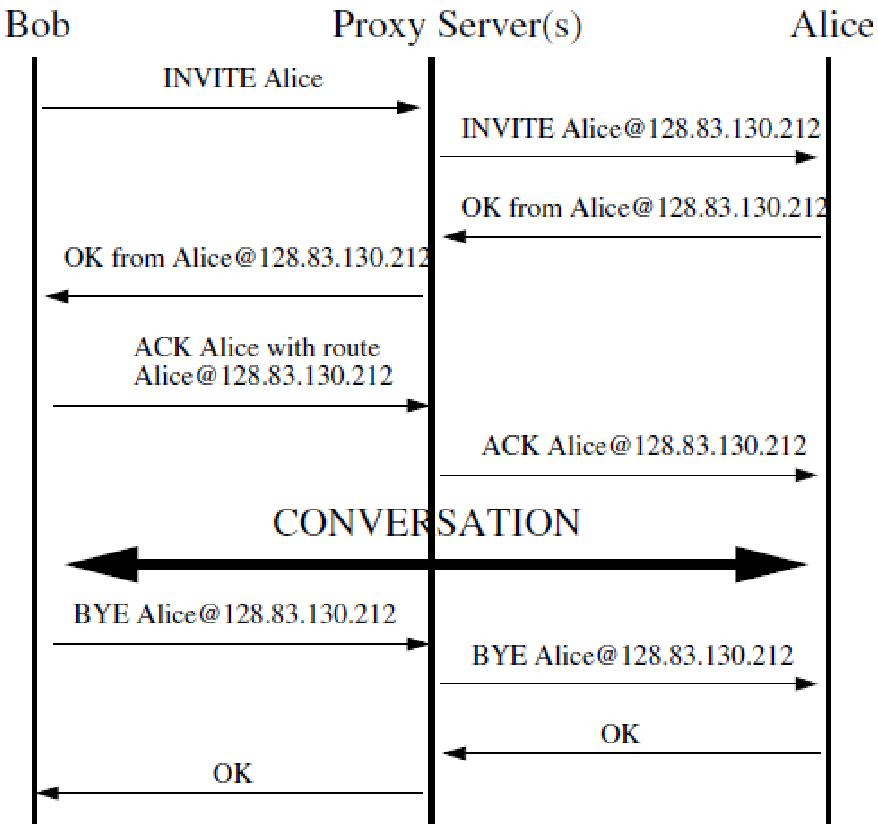

Fig 1: an example of SIP dialog

SIP is a text-based application layer protocol that addresses the signalling and session management. It uses a "request-response" model similar to that of the HTTP protocol. The protocol itself is modeled on the three-way TCP handshake. Fig 1 shows a SIP connection setup between the end points with an intermediate proxy server in between them. To setup a connection between Alice's and Bob's UAs, Alice's SIP URI is first converted into the IP address of the end point or User Agent (UA) under which Alice is currently registered. UA itself does not perform SIP address resolution and routing, but delegated it to the proxy server for the UA's domain. In our example, Bob's proxy will search the DNS to lookup the address of Alice's proxy server. During this setup process, communication details are negotiated between UAs using the Session Description Protocol (SDP) [10]. To place a call to Alice, Bob's UA sends an INVITE request to the proxy server containing SDP info, which is then forwarded to Alice's UA, possibly via her proxy server (after address resolution by Bob's proxy). If Alice wants to accept the call from Bob, she sends an OK message back to Bob containing her SDP preferences. Bob then sends an ACK response. Media exchange takes place directly between Alice's and Bob's respective UAs.

H.323

The International Telecommunication Union (ITU) recommends h.323 and it consists of family of protocols that are used for set-up call, authentication, registration, call termination and other functions. These protocols are transported over TCP or UDP protocols. fig 2 shows the H.323 suit of protocols with their transport mechanisms. H.323 family of protocol consists of firstly H.225 which is used for registration, admission, and call signaling. H.245 is used for establishing and controlling the media sessions. T.120 is used for conferencing applications. The audio codec is defined by G.7xx series by H.323, while video codec is defined by H.26x series of specifications. H.323 uses RTP for media transport and RTCP is used for purpose of controlling RTP sessions. The following fig 2 & fig 3 shows the H.323 architecture and call set-up process.

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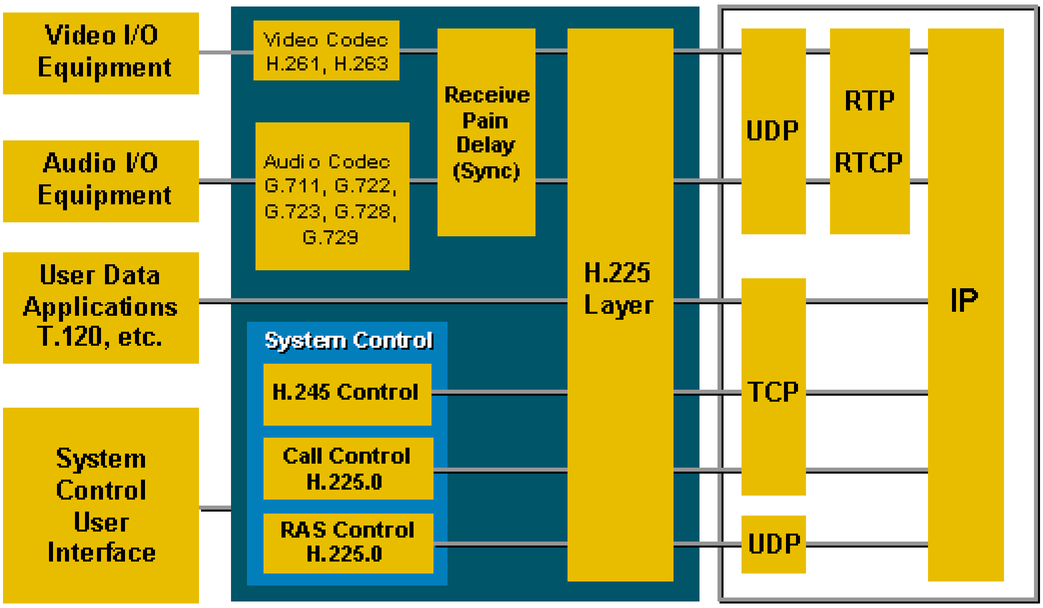

Fig 2: H.323 Protocol family

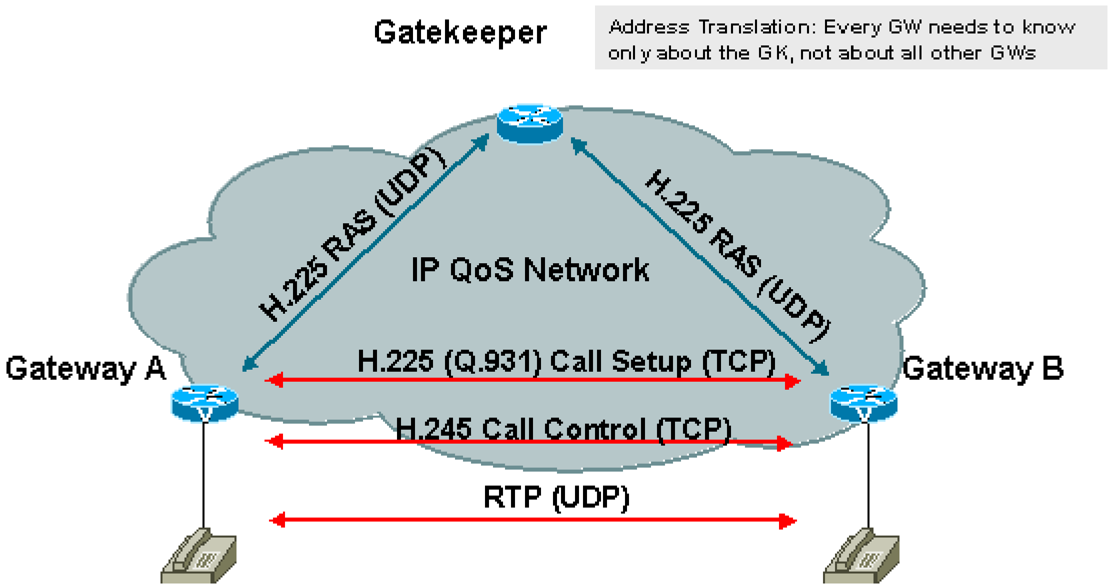

Fig 3: An example of a h.323 call flow

Inter-Asterisk eXchange (IAX)

IAX[9] protocol is able to carry a larger number of concurrent phone calls compared to other protocols. IAX achieves this by being a binary protocol and not a plain-text protocol like SIP. Using binary lets IAX compress the commands and codes into the smallest size possible. It also means that IAX is a bit more robust than SIP, which needs a parse in order to split and recognize the different commands that are being given.

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IAX's stability held it in good stead. This method rarely suffers from a shaky voice. Being less bandwidth hungry than its counterpart, its efficiency is also noteworthy. Simplicity came out tops. IAX only uses one port for signalling and audio. It is thus also easier to configure the firewall.

IAX also has its disadvantages. The first setback is encountered in extending the IAX protocol. It is necessary for any new additions in IAX to be added to the specification, as there aren't much of generic extension mechanisms available in IAX. The second setback is the use of a single port. On one hand, this is good as it simplifies network translation. But the drawback of doing this is it makes it easier to execute resource exhaustion or denial of service attacks. By flooding the port used by IAX, the system can be overcome and this can prevent legitimate users from accessing the service.

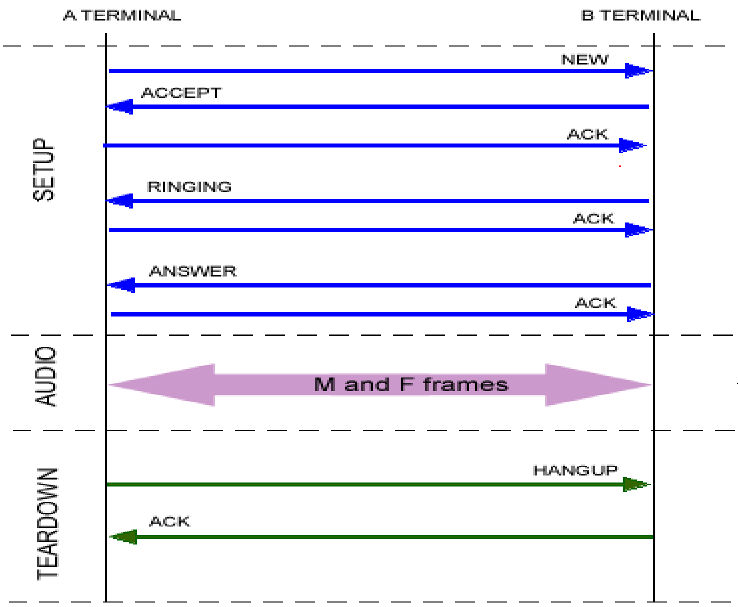

Fig 4: An example of IAX call flow

Result of Comparison between the protocols in shown in table 1.

Table 1: Comparison between SIP, H.323, IAX

	SIP	H.323	IAX
Transport protocol	**TCP, UDP**	**TCP, UDP**	**UDP**
Media Transport	**RTP/RTCP, SRTP**	**RTP/RTCP, SRTP**	**Full/mini Frames**
Server Needed	**Proxy server**	**Gatekeeper**	**Peer to peer**
IP Port for TCP/UDP	**5060**	**3230-3253 5001 5004-6004**	**4569**
Call Setup	**Invite=> <=200OK Ack=>**	**Setup=> <=Connect Ack=>**	**New=> <=Accept Ack=>**
Header Used	**RTP Header**	**RTP Header**	**Full/Mini Headers**

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III. CONCLUSION

In Table 1, we present a summary of the comparison presented in this paper. SIP, H.323 and IAX2 are all different protocols and are not directly interoperable. and each has their own flow. The SIP protocol is the most popular of the three. SIP has now become the industrial standard in the world, that's why it is not possible to totally replace it with others. But IAX is more bandwidth efficient and less overhead used by it. However, supplementary services in H.323 are more rigorously defined. Therefore, fewer interoperability issues are expected and better interoperability with PSTN (public switched telephone network). So different protocols will be good in different situations.

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