An Interoperable Signaling Solution between SIP and H.323

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ABSTACT

The Session Initiation Protocol and H.323 dominate the Voice over IP world. These signaling protocols are responsible for call setup and call tear down by IP telephony solutions. This project analyses and discusses the underlying H.323 and SIP interoperability issues and intends to build a scalable solution for the interoperability of these protocols. The gateway maps the call sequences between the two protocols. A test scenario establishes a barrage of call setup and tear down requests to test the scalability of any H.323/SIP interoperability solution.

1 SIP and H.323 Interoperability

With the rapid acceptance and deployment of Voice over IP (VoIP), two dominant signaling protocols [15] have emerged, Session Initiation Protocol (SIP) [4], [5], [10] backed by the Internet Engineering Task Force (IETF) and the H.323 [7] protocol family backed by the International Telecommunications Union (ITU). Many VoIP vendors (e.g. Cisco, 3Com, etc) that provided pure H.323 are now changing and selling pure SIP or both SIP and H.323 enabled solutions.

These signaling protocols are used by Internet telephony components for call setup and tear down. Both SIP and H.323 (see Table 1) try to solve the same problems, but from different angles. H.323 uses the more traditional circuit-switched approach based on Q.931, while SIP uses a more lightweight Internet approach. H.323 was initially championed by the VoIP industry, but SIP seems to be the favoured protocol for future development. Therefore to achieve universal connectivity, it is desirable to have interoperability between these two protocols. The next several sections present the most significant interoperability issues.

1. 1 Transport

These signaling protocols have many similarities with respect to transport. They both can use the Internet Protocol (IP) as well as the Real-Time Protocol (RTP)

[12], [13] with Real-Time Streaming Protocol (RTSP) [14] to transport voice packets. Both H.323 sand SIP IP-phones may use the same types of CODEC's e.g. G.711, G.723.1, and G.729. These similarities reduce the task of making these protocols communicate and allow only for the translation of signals.

H.323	SIP
Complex Protocol	Comparatively Simpler
Binary Representation for its messages	Textual Representation
Not Very Modular	Very Modular
Not very Scalable	Very Scalable
Complex Signaling	Simple Signaling
Backed by ITU	Backed by IETF
Large Share of market	New upcoming Protocol
MCU required for conferencing	Uses IP multicast
Loop detection is state-full	Loop detection is stateless
ANS.1 Format	Text – Based commands
TCP	UDP/TCP

Table 1: Comparison of H.323 and SIP

1. 2 Gatekeeper Discovery

If we know the IP addresses of the calling and called parties, there is no need for an H.323 gatekeeper or SIP Register. Yet this is often not the case. When we do not know the addresses in advance, or perhaps we only know the URL, we must first contact a gatekeeper/proxy server. Therefore we have to let the gateway register with the SIP Register as well as with the H.323 gatekeeper. There are a few ways to go

about doing this. The gateway could have a SIP register and/or a H.323 Gatekeeper built into it. If not, H.323 requires a Gatekeeper Request (GRQ) message and a wait on a Gatekeeper Confirmation (GCF) message from the gatekeeper [7]. On the SIP side, the User Agent sends out a REGISTER request to the proxy or redirect server, and this message will be acknowledged with a "200 OK" response [4] [see Figure 1]. The reason for this registration is that the gateway will have to know what the addresses of the phones on both sides of the network. [3]
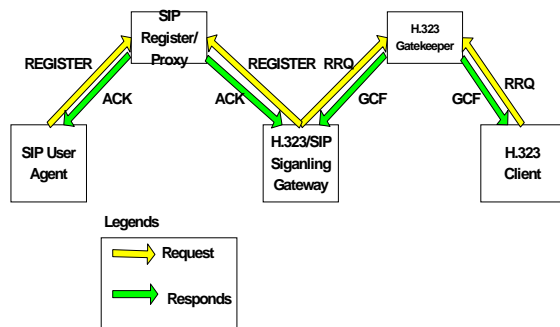

Figure 1:   Gateway Registering

1. 3 Addressing

SIP and H.323 can have various address formats. Examples include usernames, hostnames, E.164 numbers, IP number of the different IP network phones or URL-based names such as SIP:Mike@uwc.ac.za or tel:1234567@ru.ac.za. An interoperable software solution must map H.323 addresses to valid SIP addresses and vice versa [15], [11]. This means that an interoperable gateway has to communicate with both Proxy/Register servers as well as with the H.323 Gatekeeper.

1. 4 Local and Remote Media Capabilities

In the H.323 family, H.245 [8] is used for describing the session. H.245 can negotiate media capabilities, as well as establish and tear down media channels. SIP on the other hand can use any description format, but normally it will use the Session Description Protocol (SDP) [6]. H.245 carries a list of capability descriptors whereas the SDP only carries one at a time. Therefore going from the SIP side to an H.323 environment can be more trivial than going from the H.323 side to SIP. In the case when going from H.245 to SDP, it is proposed that once getting an H.245 descriptor we can send out multiple SDP messages to cater for the H.245's multiple negotiation of media capabilities.

1. 5 Handshakes Between SIP/H.323

For full interoperability between H.323 and SIP we must map the sequences of handshakes that will have to happen when we make calls from an H.323

terminal to a SIP terminal (see Figure 2) and from a SIP to an H.323 terminal (see Figure 3) [2]. From these diagrams we can see that H.323 has more handshakes than SIP. Therefore the gateway has to maintain handshakes with both protocols to be part of the "H.323 network" as well as the "SIP network".
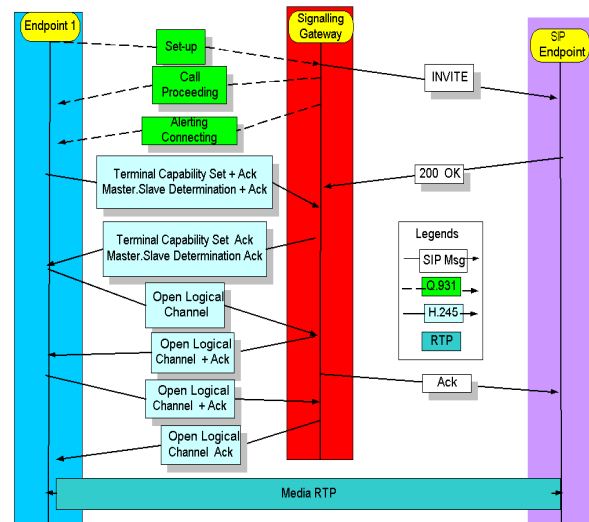
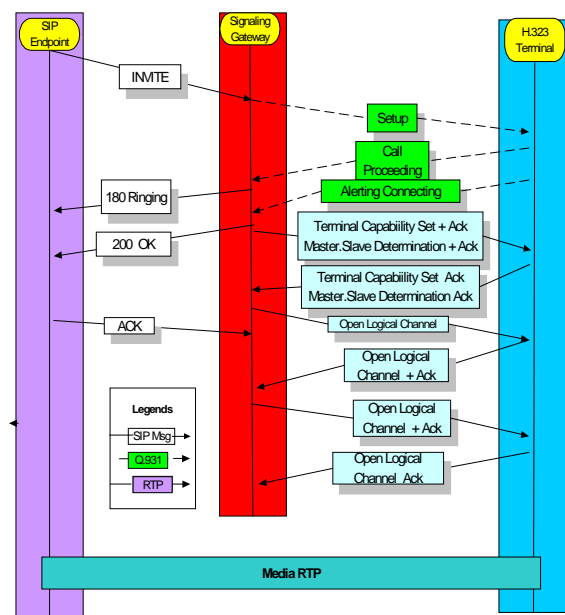

Figure 2:   H.323 to SIP mapping

Figure 3:   SIP to H.323 mapping

Multiple calls are possible with this architecture. Once a call has been set up, and the endpoints have an end-to-end connection, RTP transports the real-time data from one terminal to another. Once the call is in process, the gateway has no control over the connection. The gateway comes into play again once the call connection needs to be broken down. The SIP

terminal will send/receive a "BYE" request and it will wait/send for an "ACK". The H.323 endpoint will send/receive a "Close Logical Channel" and wait/receive a "Close Logical Channel ACK".

2 SIGNALING INTEROPERABILITY GATEWAY (SIG)

Our signaling interoperability gateway (SIG) masquerades as both an H.323 client as well as SIP User Agent (UA). This gateway listens on SIP's well-known UDP (and TCP) port 5060 as well as H.323's well-known port 1720. Once a request comes in on any of these ports, the signaling gateway will do a translation of the signal and act as a proxy by making the request on behalf of the calling terminal.

To allow the gateway to communicate with the H.323 gatekeeper as well as the SIP register, this solution is built with an API to an H.323 open source library [19] as well as a SIP Library that has been obtained from Columbia University [18].

The length of the setup session is dependent on the response from the user. We have to wait for the user to accept the call if auto-accept is not on. This could cause problems when trying to build a scalable solution. These listening ports could become saturated with requests. Then we will not be able to send responses back to the client, as only one application can use a port at one specific time. A way to deal with this problem is to have a daemon that spawns off a process and connects on a new port for each setup. This new port will only be active for the duration of the setup. Once the setup is complete, we free up this port (and process). When one of the terminals wants to end the call, a new port will be allocated to it.

3 SOLUTIONS IN EXISTENCE

The IETF's H.323/SIP interworking group can be found at http://www.softarmor.com/sipwg/teams/sip-h323/. A few implementations of the H.323/SIP gateway exist. Most of the source code for these gateways is available, as the working group is hoping to standardise the interworking of these protocols. We introduce a few of these examples in the next section.

3. 1. SIP323

SIP323 has been developed at the University of Columbia. This solution uses the H.323 open source library [19] as well as the SIP library that has been developed at the University itself. The SIP323 gateway [18] allows communication between an H.323 phone (NetMeeting) and a SIP phone (E-phone). This solution also caters for multiple simultaneous calls [18]. This solution supports H.323v1 and H.323v2 as well as SIPv2. Currently this solution only caters for G.711 Mu Law audio calls. SIP323 has a built-in gatekeeper, as well as a SIP

proxy, but if this is not configured it will use RAS (Registration Admission Status) procedures to find its gatekeeper. An advantage of this gateway is that it attempts to keep the signaling away from the media and allows for a direct connection between the SIP and H.323 endpoints. If this is not possible it has a built-in media switching fabric.

3. 2. THIPHON

The ETSI (the European Telecommunications Standards Institute) project THIPHON (Tele-communications and Internet Protocol Harmonization Over Networks) has issued a news release stating that they have built a signaling gateway that is not signaling protocol specific [1]. This gateway caters for SIP-H.323 interoperability as well. There is no source code to download and no application to test.

3. 3. VOCAL

Vovida has developed VOCAL (Vovida Open Communications Applications Library), which is an open source VoIP library [20]. VOCAL includes a SIP server as well as a Marshal Server but what we are most interested in is the H.323 to SIP translator. This H.323/SIP gateway communicates with the H.323 open source [19] code and allows their SIP system to communicate with NetMeeting. This implementation is still in the development stages.

4 SCALABLITY TESTS

Tests will be run on three of these signaling gateways: VOCAL, SIP323 and SIG, our in-house developed gateway. The tests generate thousands of SIP and H.323 call setups and call teardowns across a translation gateway (see Figure 4). We are most interested in how many calls the gateway can handle at one time, as well as how signaling gets affected when other traffic is present on the network. We are also interested in measuring the delay that gets added when doing H.323 to SIP (and vice versa) translations.
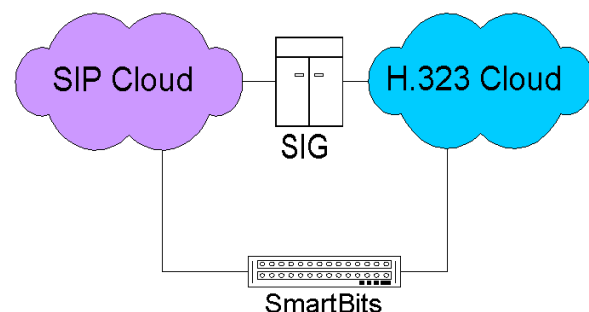

Figure 4:   SmartBits Testing a Signaling gateway

We hope to show that a signaling gateway will scale to a certain threshold. The test results will be used in an iterative process to learn how to design and implement SIG to increase its scalability.

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