An Interoperable Signaling Solution for IP-based Next Generation Networks

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Abstract

Current VoIP signaling protocols are interoperable with the PSTN SS7 via H.323-to-SS7 or SIP-to-SS7 gateways. As we move toward the Next Generation Network, the PSTN falls away and we work toward carrier-grade VoIP interoperability between H.323, SIP, and other future VoIP signaling protocols. This paper present a Work in Progress to design and implement a solution based on the Internet Protocol itself.

1. VoIP/PSTN Signaling

The Public Switch Telephone Network (PSTN) makes use of two distinct layers, a circuit layer and a switched transport layer. The separation of these layers frees up the transport entities from signaling responsibilities as all signaling occurs on the Signaling System 7 (SS7) control network. [20]

IP is connectionless and packet-switched. At the transport layer, protocols such as TCP and UDP provide a form of signaling, but the overall picture remains "best-effort"[8], and packets that make up virtual connection may follow multiple paths from endpoint to endpoint. Despite this, the IP protocol can deliver data, voice, and video with astonishing clarity.[1,20] Voice over the Internet Protocol (VoIP) is of particular interest to us.

The PSTN traditionally offered the highest Quality of Service (QoS) for voice communications. The systems are reliable, highly distributed, well managed, and deliver speech with exceptional clarity. Currently, digital switching provides even more reliability and quality. There are several significant challenges to apply Internet technologies toward a 99.999% reliable VoIP system.[20]

The first challenge is that the packet switching paradigm requires reliable call-signaling capabilities. The second is that QoS must be provisioned and controlled. The third is to build converged VoIP/PSTN solutions. The final challenge is to evolve this migration into the Next Generation Network (NGN).

The two prominent VoIP signaling protocols are H.323[5,23,24] and the Session Initiation Protocol (SIP)[2,6,10,15,17]. Currently there exist H.323/SS7 gateways as well as SIP/SS7 gateways. These gateways allow VoIP solutions to communicate with the PSTN, but the problem is that H.323 and SIP are not necessary interoperable with each other *(See Figure 1)*. Companies and telcos need to handle SIP and H.323 as well as any other signaling protocol that comes down the line.

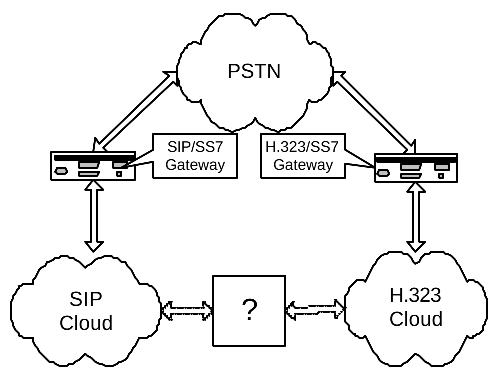

Figure 1 : VoIP Signaling Interoperability Problem

Unlike Asynchronous Transfer Mode (ATM) [7], Quality of Service (QoS) remains a concern for VoIP. However, due to mechanism such as RSVP, QoS can be supported over IP networks. Yet on dedicated IP networks, we believe carrier grade, QoS is attainable.

The NGN of the future will assume all communication needs. Engineers from both Circuit Switch (Intelligence Networks) [9] and packet switch paradigms are working towards the NGN. A Media Gateway (MGW) glues an IP network to the PSTN and allows for seamless voice transfer. These MGWs convert between SS7/SIP or SS7/H.323, etc. The PSTN might disappear and most carrier-grade services, of which voice is just one of them, are likely to migrate to dedicated IP, Frame Relay, ATM networks, and/or any combination that is capable of providing carrier grade QoS. Of course, such dedicated networks will require connectivity to the uncontrollable Internet. This paper addresses VoIP signaling since we feel IP will be the dominant protocol in the NGN.

With new VoIP hardware and software flooding the market, issues of interoperability emerge a force to be reckoned with. Today's service providers and enterprises must consider interoperability as the most urgent technical issue to deal with. Thankfully, there exist standards such as H.323 and Session Initiation Protocol (SIP) to provide tools that we can work with to develop solutions.

1.1 H.323 Signaling

The International Telecommunications Union (ITU-T) defined the H.323 protocol suite [5,24]. This recommendation provides the technical requirements for voice communication over LANs while assuming no QoS is being provided by the LAN. H.323 is currently dominant in the VoIP world. The H.323 protocol is actually a family of protocols. H.225 RAS (Registration, Admission and Status) uses UDP (an unreliable channel) to transmit registration, admission, bandwidth changes, and status messages between the endpoints and the gatekeeper. Since it is sent over UDP, it recommends timeouts and retry counts for messages[23]. H.225 Call Signaling (Q.931) uses TCP for setup and termination. H.245 is used for negotiating the usage of channels [23,24]. RTP (real-time transport protocol) uses UDP to transmit the digitized voice streams once the call and logical channels has been established[11]. In general H.323 can be defined with four logical components:

Terminal – All H.323 terminals must support H.245, Q.931, RAS and RTP.

Gateway (GW) –When terminals on different networks want to communicate, they do so via gateways using H.245 and/or Q.931. Incorporating gateway technology into the H.323 specification, the ITU has positioned H.323 as the glue of standards-based multi-media conferencing.

Gatekeeper – Each H.323 device in a "zone" has to register with a Gatekeeper. The gatekeeper acts as the central point in the network. Functions of the Gateway are, address translation, admission control, call control signaling, call authorization, bandwidth management, and call management. The gatekeeper holds a routing table of where all the nodes are situated.

Multi-Point Control Unit (MCU) - The MCU is an endpoint that provides the capability for three or more H.323 entities to participate in a multi-point conference.

1.2 SIP Signaling

On the other hand, the International Engineering Task Force (IETF) backs SIP as an alternate solution [17]. SIP is an application layer control protocol for creating, modifying and terminating sessions with one or more participants. The architecture of SIP is similar to that of the HTTP (client-server protocol) : SIP a text-based protocol harnesses the power of the Internet Paradigm by borrowing common elements such as the format of Domain Name Server (DNS) and email addressing formats. SIP signaling commonly runs over TCP because of inexpensive widespread connectivity, directory services, and naming services.

2. VoIP Gateways

A gateway (GW) solution consists of a signaling GW, a media controller, and a MGW. The Media Gateway Control Protocol (MGCP) [16] enables control and management of data communications equipment at the edge of the emerging multi-service packet network[6]. The problem is the interoperability between these GWs themselves. In order to accomplish interoperability between all VoIP terminals one must consider the following : [4]

- **GW to GW interoperability** requires each gate-keeper to share call routing tables. Each service provider should have a (full or partial) view of partner networks so that calls can be made from anywhere on any connected network.

- **GW to Gatekeeper interoperability** allows a call control engine to administer GWs. Carriers need this because the gatekeeper controllers are the heart of application development and delivery, network control and administration. Also, service providers need to manage multiple control elements centrally.

- **Gatekeeper to Gatekeeper interoperability** allows a gatekeeper to decide where to let the GW route a call. A gatekeeper must "know" the addresses of terminals on other networks.

- **True Service interoperability** allows a service provider to manage voice services across a multi-vendor network. That is, if each vendor complies with interoperability standards, services providers can charge and control billing.

3. An Interoperability Solution

Inspecting both protocol stacks *(see figure 2)* reveals that both H.323 as well as SIP runs over IP and uses the Real Time Transport Protocol (RTP) for transferring real-time audio/video data, which in turn runs over UDP. This commonality reduces the task of inter-working between H.323 and SIP, thus allowing us to concentrate on the interoperability of the signaling protocols, such as H.323's H.225 and SIP's sessions description protocol (SDP)[18].

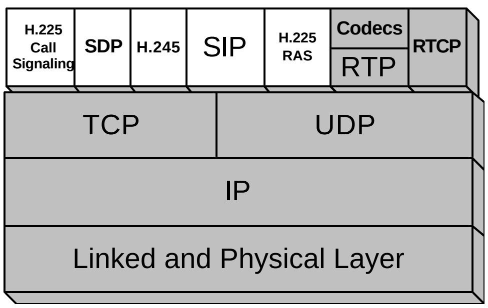

Figure 2 : SIP and H.323 protocol stacks

We will translate the calling and called addresses (alias or URL) to IP addresses, we will be able to provide a framework for the user to dial an address without him knowing if it is on the H.323 network or the SIP network. With a generic gateway (GGW) connected to both the H.323's Gatekeeper as well as the SIP register server, we allow softphones from any point on the network to communicate with each other.

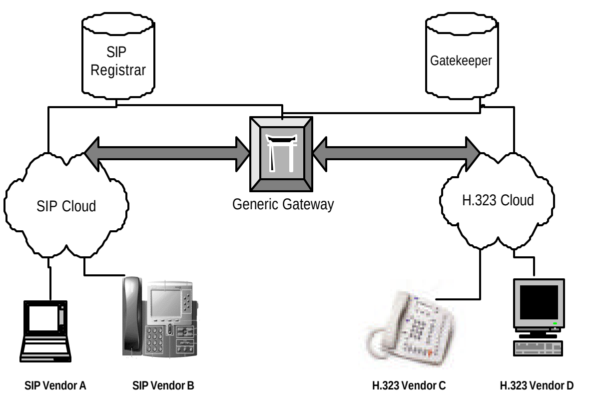

Figure 3: My vision on how the GGW will work

The SIP registrar or the H.323 gatekeeper should be able to decide if it can handle the connection or not (i.e. to check if the calling party uses the same protocol as the called party). If not, it will send a request to the GGW who will convert the request and act as a bridging router to the correct destination. Once the connection is made the GGW and all the call parameters are agreed on the GGW will release control of the call (in order to reduce latency) and data transfer will happen across RTP. This solution will also cater for yet-to-be-defined protocols for future interoperability.

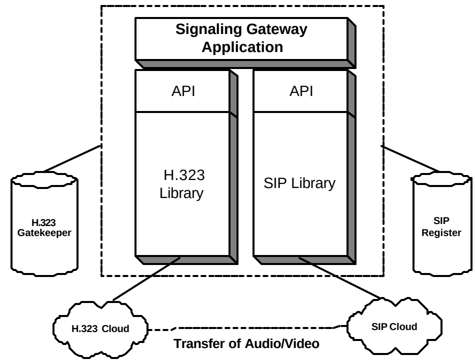

Figure 4 : Address Translator

4. Test Scenario

We will be testing this system using NetComs Systems SmartBits 200 Starter kit to generate traffic on the network. FTP and Telnet sessions as well as SmartBits will be adding traffic on the network. This is to test the capabilities of the gateway and how it copes with on a live network. *(See figure 5)*

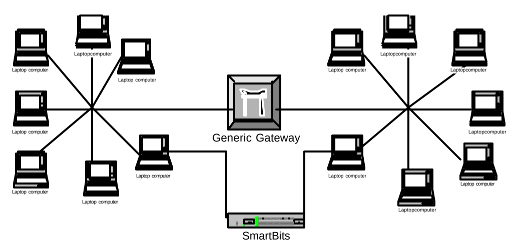

Figure 5: A test scenario

A few calls will be setup, and gradually adding more, so that we can observe how many calls the GGW can handle at one time. We will also be using a combination of different call scenarios from SIP phones to H.323 phones and visa-versa.

5. Work In Progress

This work is based on work done at the University of Columbia by Kundan Singh. [13,14]. The Voice over IP Forum, in conjunction with the International Multimedia Teleconferencing Consortium (IMTC) has published initial interoperable recommendation.

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