Concepts Meet Reality…

Session Initiation Protocol (SIP) in the Contact Center

Introduction

Executives with Contact Center oversight will soon be inundated with new Session Initiation Protocol (SIP) product offerings from the vendor community. SIP is being heralded as the next great enabler of quick application development, vendor interoperability and call context multimedia support. Many vendors will state that SIP is past the Early Adopter phase, has crossed the chasm and is ready for the pragmatist majority. Even so, any changes to existing Contact Center operations must be in support of business drivers (e.g. reducing costs, increasing revenue, reducing risk, achieving and/or maintaining competitive advantage, increasing customer satisfaction, and increasing worker productivity), rather than due to new technology perceived as "cool." Without the direct linkage to business drivers, any introduction of new SIP enabled technology into the Contact Center is nothing more than an expensive science fair project. This document provides an introduction to the protocol, SIP initiatives focused on the Contact Center and a strategic approach to introduce SIP technology into the Contact Center.

Background and Current Environment

In 1992, the Multicast Backbone (MBONE) network was established between various academic and research institutions. The MBONE's charter was to experiment and develop technology to handle live multimedia communications via multicast. Multimedia communications were concentrated on audio and video conferencing. These one-to-many, many-to-one or many-to-many streams needed some mechanism to invite users, determine their media capabilities (i.e. video and/or CODEC set, audio only, etc…) and incorporate them into the session. The earliest development and ratification of SIP by the Internet Engineering Task Force (IETF) provided this mechanism. SIP was first addressed in Request for Coments (RFC) 2543 and has been enhanced to become RFC 3261. Over time additional features, such as IP Telephony and Presence, have been added under the SIP umbrella. Currently, there are over 40 RFC's published by the IETF's SIP Working Group.

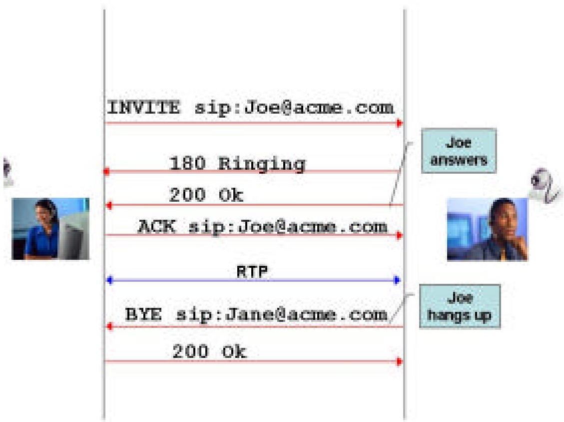

Figure 1: Basic SIP Call

SIP efficiently leverages existing IETF standards, such as Hyper Text Transfer Protocol (HTTP) for messaging formats, Real Time Protocol (RTP) for media, Domain Name System (DNS) and Uniform Resource Locator (URL) for addressing, Session Description Protocol (SDP) for media-negotiation and Multipurpose Internet Mail Extensions (MIME) for messaging and attachments. The vendor community is very excited about the fact that SIP is not a "start from scratch" protocol. So in theory, any new applications built around SIP will also have the embedded usage of mature standards. However, SIP is not just a recipe where the IETF has thrown in some MIME, DNS, HTTP and out comes SIP. There are many new features and aspects of the protocol.

It is important to note that within SIP, the term "multimedia" includes many contexts, such as, voice, video, and Internet chat. SIP allows the user to elevate the conversation between the various contexts. For example, with appropriate applications a user chatting with a peer via a chat client on a PC, may elevate the session to a phone call by pressing a desktop icon; the user can further elevate the conversation to a video call, if the endpoints have video capability. SIP is an enabling technology that allows communications between Web users and contact centers.

So expanding our previous example, assume a customer on your corporate Internet site has a question about a product or his account, using on-line chat as the media exchange. From the call center perspective, this is just another contact via chat media versus a traditional voice call. The contact is "routed" to an agent. Granted, this may be an agent logged into the chat queue or split. At this point the agent can elevate the contact to a traditional voice call, if desired.

SIP is a peer-to-peer (P2P) protocol. P2P implies that there's some intelligence in the session originating endpoints (aka user agent clients) that allows the endpoints to know the location of other endpoint clients and negotiate directly with them when activating multimedia sessions. In its' simplest form SIP endpoints can establish sessions with distant peers directly. Peers in a session are called User Agents (UA's). Indirectly, an SIP UA can send an invite to an SIP Proxy/Redirect Server where the originating SIP UA will be redirected to the distant endpoint. A user agent can function in the following roles.

User agent client (UAC) - A client application that initiates the SIP request.

User agent server (UAS) - A server application that contacts a distant endpoint when a SIP request is received from an originating endpoint and returns a response on behalf of the distant endpoint.

A UA can function as UAC and a UAS, and change between roles dependent on a specific transaction.

The UAS provides functions such as registration, location, proxy, and/or redirection:

Registrar - The Registrar is responsible for accepting dynamic registrations from SIP endpoints. The registrar is often co-located with a Proxy, Location or a Redirect Server and updates these applications when SIP endpoints register or unregister.

Location Server - The Location Server is a service used by an SIP Proxy or Redirect server to obtain information about a distant endpoint's location.

SIP Proxy - The SIP Proxy is a function that can make requests on behalf of endpoints. A proxy begins the process of centralizing the intelligence of the P2P network. Rather than every endpoint knowing about every other endpoint on the network (n entries in some database), the endpoint can off-load the "routing" function to another device.

Redirect Server - The Redirect Server responds to SIP requests (INVITE's) by sending redirects back to the originating endpoint so that it may contact the distant destination. In other words, the originator asks, "How do I get to Joe@acme.com?" The redirect server resolves/ finds Joe @acme.com and sends the resulting information to the originator so that the session may be established directly with Joe.

SIP is an ASCII-based protocol that uses requests and responses to establish communication among the various endpoints in a network and ultimately establishes a multimedia conference between two or more endpoint UAs. UA' request access to services via defined SIP Request Messages.

The SIP Request Messages are:

- **REGISTER -** Register contact information
- **INVITE, ACK, CANCEL -**Setting up sessions
- **BYE -** Terminating sessions
- **OPTIONS -** Querying servers about their capabilities
- **SUBSCRIBE, NOTIFY** (RFC 3265) -Event notification framework
- **MESSAGE** (RFC 3428) - Instant messages

SIP Responses between UACs and UASs are generally defined as:

- **1xx: Provisional -** Request received, continuing to process the request
- **2xx: Success -** The action was successfully received, understood, and accepted
- **3xx: Redirection -** Further action needs to be taken in order to complete the request
- **4xx: Client Error -** The request contains bad syntax or cannot be fulfilled by a specific server
- **5xx: Server Error -** The server failed to fulfill an apparently valid request
- **6xx: Global Failure -** The request cannot be fulfilled at any server

Another important feature of SIP is the concept of "Presence," or simply the ability for others on the network to sense if you're available, away from your endpoint (the endpoint could be a PC or telephone), or do not want to be disturbed. Session Initiation Protocol for Instant Messaging and Presence Leveraging Extensions (SIMPLE), RFC 3428, is an instant messaging (IM) and presence protocol suite. The IM communication tool within the Enterprise Desktop space is featured in products such as

Microsoft's Office Communicator or IBM's Sametime Connect. At VoiceCon 2006 many vendors announced their alliance with these desktop giants to develop SIP based products to integrate telephony presence with the desktop giants.

Within SIP, several aspects of telephony are addressed. The IETF's Session Initiation Protocol Project INvestiGation (SIPPING) Working Group is chartered to "document the use of SIP for several applications related to telephony and multimedia, and to develop requirements for any extensions to SIP needed for those applications." SIP Services Examples draft (draft-ietf-sipping-service-examples-06), defines example telephony features implemented in SIP. Another important SIP RFC is built around Message Waiting Indicator (MWI) for SIP stations, RFC 3842.

The SIPPING Working Group document, draft-ietf-sipping-service-examples-06, provides example call flows detailing an SIP implementation of the following traditional telephony services:

Call Hold
Music on Hold
Unattended Transfer
Consultation Hold
Unconditional Call Forwarding
Attended Transfer
No Answer Call Forwarding
Busy Call Forwarding
Single-Line Extension
3-way Call
Incoming Call Screening
Find-Me
Call Pickup
Call Park
Outgoing Call Screening
Automatic Redial
Click to Dial

Recall that a UA acts as a UAS or UAC depending on the transaction. If a SIP Proxy is introduced into the scenario is it a UAS or UAC? The answer is both.

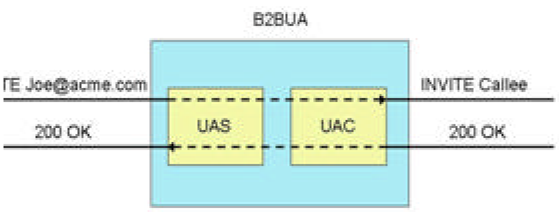

Figure 2: B2BUA Call Setup

SIP has defined a Back-to-Back User Agent (B2BUA) that acts as a user agent to both ends of a call. To SIP clients, the B2BUA acts as a User Agent *server* on one side and as a User Agent *client* on the other (back-to-back) side (Figure 2). The B2BUA is also responsible for handling all SIP signaling between both ends of the call. Important note: this is how vendors extend the value-added features of their traditional telephony products to SIP endpoints. The features distinguish the point in which products leave the standards trail and begin to become more proprietary.

SIP Technology in the Contact Center

SIP coupled with IP and the previously mentioned IT standards will further evolve the virtualization of the Contact Center. Throughout this document the term Contact Center has been used instead of Call Center. This subtle difference reflects the evolution of different multimedia contexts (voice, chat, video) with presence that are infiltrating the legacy call center environment. The vendor community will continue to offer or evolve their SIP solution sets to leverage converged communications in the following areas around the Contact Center:

- Virtual Contact Center
- Hosted or Outsourced Contact Center
- Endpoints and Agent Desktops
- Internet Customer Care

Another area of vendor development, outside the scope of this paper, is the concept of Service Oriented Architectures (SOA). Avaya Inc. calls their concept “Service Oriented Architecture,” Cisco Systems Inc. uses the term “Service Oriented Network Architecture (SONA).” SOA will be another area of development whereby vendors will attempt to treat communications resources such as conferencing, messaging and presence, as service reuse engines and integrate business application events to utilize telecommunications services, as needed.

SIP will formulate the underlying communications glue to integrate telecommunications services with the various business applications (CRM, ERP.)

Virtual Contact Center

The concept of a Virtual Contact Center is not new; however, vendors will continue to offer products to "virtualize" the Contact Center. For traditional "brick and mortar" contact centers housing hundreds of agents, SIP allows a location-independent contact center and presents a unified customer facing presence where ever the agent may reside. Customers have a consistent experience across all locations and modes of access (web based chat, voice). Administrative changes can be made in a single place, and these changes are immediately effective for all calls and all agents registered to the virtual contact center. SIP reduces the cost and complexity of implementing home-based agents, who now only need a multimedia PC. Finally, SIP has infiltrated the Service Provider community, such that a Service Provider could host the entire contact center in their network, and use only IP connections to deliver multimedia (voice, web chat, email) contact types to their customers' contact center agents.

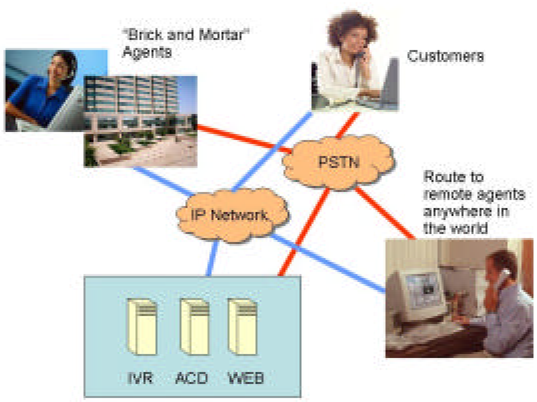

Figure 3 – Virtual Contact Center

Hosted or Outsourced Call Centers

Many contact center operators may consider moving operations to a hosted or out-sourced solution, in an effort to minimize the labor costs of running a contact center in-house. Calls originating in one country can be transmitted to agents in another country over an IP network. (Some research is required around in-country hop off regulations to ensure you company is not violating the telecommunications laws of the host country.) An SIP-based ACD for the contact center can be maintained in a data center, anywhere in the world, via IP connectivity.

Endpoints and Agent Desktops

The flexibility of SIP-based architectures enables many options. Some vendors are able to separate the voice and data streams associated with a contact. For example, agents receive calls and customer information via screen pops to their desktop; the call can be routed concurrently to the agent via a "hard" handset, rather than streaming the voice to the PC. The handset could reside on a totally different system (PBX, home agent's PSTN line). The call remains under control of the originating virtual contact center for metric analysis. This is a very important feature when considering ways to reduce the risks associated with maintaining the quality of the voice stream on a converged network. The handsets themselves are migrating to SIP signaling. Now contact centers will have to consider what type of handset is best for them: digital, H.323, SCCP, MCGP or SIP handsets.

Internet Customer Care

Vendors will continue to offer products that exploit the multimedia capabilities of SIP. It is not enough to only handle phone calls in the contact center; consumers now have become more comfortable with their PC as their primary communication device. To maintain customer satisfaction contact centers must enable customers to contact the contact center via any mechanism; web-based chat, click to talk voice, video or email. In the customer care environment, SIP replaces CTI and merges voice, video and data together. Using instant messaging and presence, agents can "see" who is able to support them, allowing your business to assign an expert to resolve the customer issue as quickly as possible. The possibility that the expert could be located halfway around the world remains totally transparent to the customer.

Business Decision Criteria

It is important to note that technology implementation projects do not happen independently. Successful technology projects are directly correlated to other business imperatives, such as increasing revenue or increasing customer satisfaction. Traditionally, the project manager is clearly focused on determining the requirements to satisfy the new system. However, it is important to determine which aspect of the project ensures that the project actually made the business better. Too often the vendor has provided "assistance" creating the Return on Investment (ROI) study that justified the project in the first place. Therefore an established mechanism to validate the ROI is necessary.

For example, if a business objective is to reduce calls to the Contact Center through Self-Help, generally through a Web browser, how will you validate that the changes introduced are providing the customer with self-help, without reducing customer satisfaction? A self-help solution would result in decreased agent call volumes, calls queued and an increase in web hits. But is the customer satisfied with this new access method or has he just moved to a competitor? How will you know?

Obviously someone or some organization, serving as an honest broker, needs to be focused on the end-to-end business analysis aspects of the project, along with process change management, technical implementation, monitoring and overall program/project management. The project's honest broker should be capable of demonstrating knowledge and experience around a wide array of best practices and technological/business challenges relevant to the Contact Center, SIP, IP telephony, and network performance. Ultimately, the project requires leadership and the application of proven, innovative and leading edge methodologies to meet the challenge of deploying SIP technology in the Contact Center.

Conclusion

When comparing SIP to the English language, rules exist for both in composing sentences and the use of grammatical syntax. It takes a writer to compose a novel; likewise vendors currently are feverishly composing novels for the Contact Center. Each vendor's offering will have different strengths and weaknesses. Ironically, the same vendors who state SIP is not ready for the Contact Center today will be back within the year stating their products are ready for primetime. Be aware of the influence of your favorite vendor to propose inaction; SIP is here now and your competitors are examining for application within their business. The most challenging transition for Contact Center leadership will not be in implementing SIP technology; it will be the governance of business processes, changing the business culture, and establishing measurable criteria to validate that the introduction of the new SIP technology into the Contact Center is directly contributing to improved customer satisfaction, increased revenue, reduced costs, improved efficiencies and competitive advantage.

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