

The Internet2 SIP.edu Initiative

Dennis Baron < *dbaron@mit.edu* >,
Jeremy George < *jeremy.george@yale.edu* >,
Ben Teitelbaum < *ben@internet2.edu* >

June 2, 2003

Introduction

The Internet2 SIP.edu initiative seeks to promote the convergence of voice and email identities, grow SIP-reachability in Internet2, and encourage experimentation with new enterprise SIP services. SIP.edu is one of several new Internet2 initiatives seeking to promote advanced peer-to-peer communications applications that integrate voice, video, and instant messaging through the “glue” of presence services and open standards.

To engage what is estimated to be nearly five million Internet2-connected users in an exploration of new communications paradigms, these users require connectivity. SIP [1] and SIMPLE [2] are regarded as essential to providing this connectivity. Users require SIP addresses and a call routing infrastructure to initiate peer-to-peer communications sessions with each other. Presence routing infrastructure is also required to support the publication and notification of user-level state changes and to support many advanced integration functions.

Merely providing connectivity to enable innovation at the edge is not enough. The richest new communications applications will combine peer-to-peer design principles with new enterprise middleware services. Call control policy, mobility, location presence, and integration with calendaring all require support from the enterprise. Through the Internet2 VoIP working group and related activities, campus IT directors and telecommunications staff are working to understand the evolving requirements of next-generation enterprise communications services.

SIP.edu

The primary goals of the Internet2 SIP.edu initiative are: the convergence of email and voice identities, the rapid growth of SIP reachability, and the

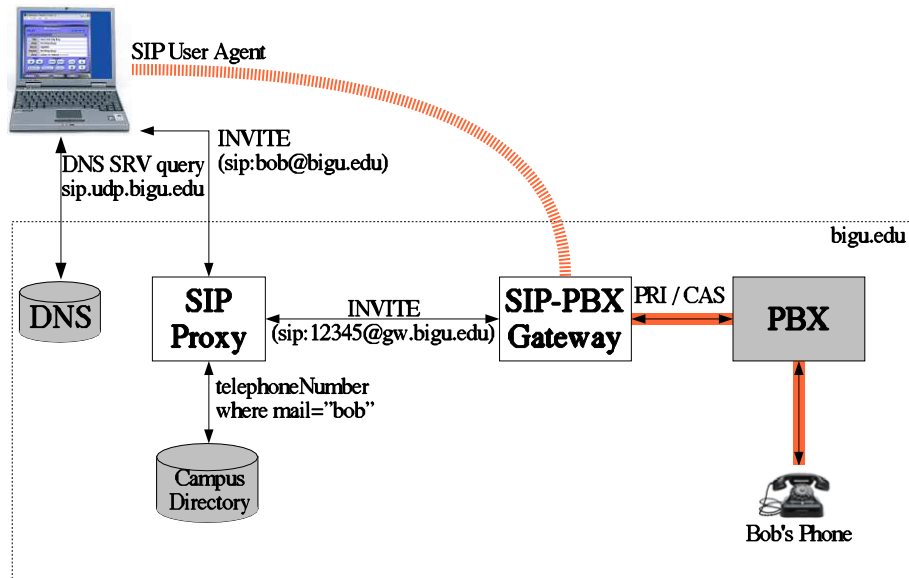


Figure 1: Initial SIP.edu architecture

widespread deployment of enterprise SIP servers to serve as learning platforms for schools looking to experiment with new enterprise SIP services. The means towards this end will be a low-cost “cookbook” architecture that an Internet2 school can deploy to enable email address style SIP dialing to all its users.

Generically, the “ingredients” in a SIP.edu implementation include:

- a SIP DNS SRV record pointing to a SIP proxy server (or multiple servers, if fault tolerance is desired)
- a SIP proxy server integrated with the campus directory to alias internal extension numbers to usernames
- a SIP-PRI gateway to terminate inbound SIP calls and gateway to the correct “black phone” through the campus PBX (or CENTREX trunk)

When a school becomes “SIP.edu enabled”, its users are reachable by voice at their email addresses (*e.g.* sip:username@bigu.edu). A simplified SIP.edu call-flow from Alice to user Bob at BigU would proceed as follows (see figure 1).

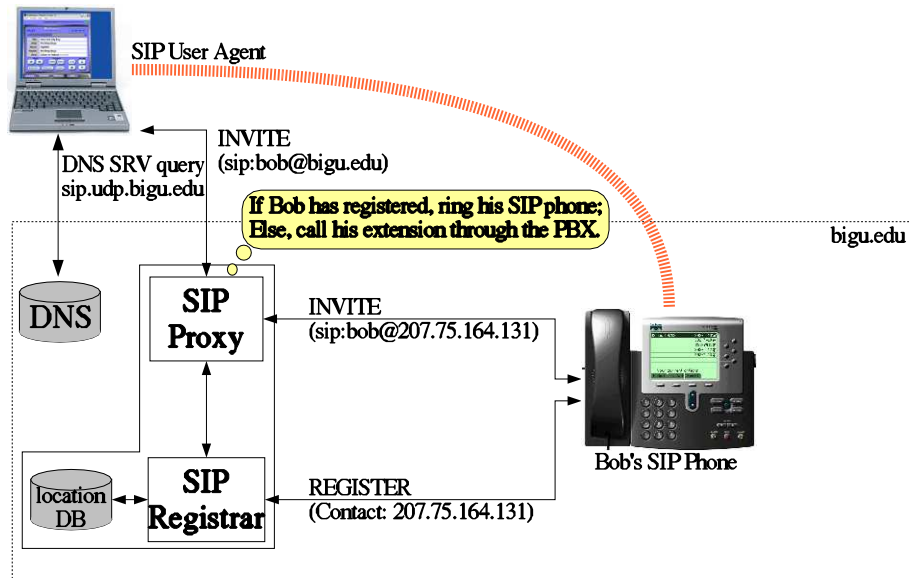


Figure 2: Extension of the initial SIP.edu architecture to support the registration of SIP user agents

1. Alice types “bob@bigu.edu” into her SIP client to initiate a call to Bob (Alice’s client could be either a hard phone or a soft phone)
2. Alice’s client performs a DNS SRV resolution on `_sip._udp.bigu.edu` discovering that BigU’s SIP proxy is at `proxy.bigu.edu`¹
3. Alice’s client sends `INVITE sip:bob@bigu.edu` to `proxy.bigu.edu`
4. The proxy re-writes the `INVITE` with the extension published for Bob in the campus directory, sending `INVITE sip:12345@gw.bigu.edu` to the gateway `gw.bigu.edu`
5. The gateway rings Bob’s desk phone (ext. 12345) through the PBX

Once campuses gain confidence in this approach, they will begin to experiment with supporting the registration of their users’ SIP user agents (UAs). At this stage of deployment, the proxy in step 4 above would check whether Bob has registered a SIP UA; if so, the proxy would forward the `INVITE`

¹As support for NAPTR[3] matures, the SIP.edu architecture will evolve to recommend its use. In the example above, Alice’s client would first perform a NAPTR resolution to learn which services are available and then perform the appropriate SRV resolution.

to Bob's registered SIP UA; if not, Bob's extension would be rung through the PBX as described above (see figure 2).

MIT and Yale have demonstrated the feasibility of this approach. Approximately 12,000 MIT users and 15,000 Yale users are now SIP reachable to anyone with a SIP user agent.

Identity Convergence

Through the SIP.edu initiative, we hope to converge email and voice user identities throughout Internet2, explore new enterprise SIP services, and spark a "SIP chic" fad similar to the "email chic" that existed in the late 1980's. Metrics of success would include: the appearance of SIP addresses on business cards, people who have not exchanged phone numbers saying to each other "just SIP me", and greater user pull for enterprise SIP services, including instant messaging and presence.

SIP.edu Cookbook

The SIP.edu cookbook is a work in progress. It will include a description of the generic architecture (as above), as well as a canonical "recipe" for deployment. To scale SIP reachability rapidly, it is important to design the canonical recipe to make deployment as simple as possible in the typical case. It is also essential to keep the cost of entry low, while affording a growth path for future exploration and deployment of SIP services.

The canonical recipe will include:

- a recommended SIP proxy server product
- a recommended SIP-PBX gateway product and example configurations for common PBXs
- patches or scripts to integrate an LDAP eduPerson campus directory with the SIP proxy server

Although participating SIP.edu institutions may choose to implement the generic architecture with a variety of combinations of proxy server, gateway, person directory schema, and PBX, the canonical recipe will be heavily promoted.

Glossary

Black Phone Non-IP telephone (analog or digital)

Hard Phone Physical IP telephone

SIMPLE SIP for Instant Messaging and Presence Leveraging Extensions; IETF working group developing extensions to SIP to support instant messaging and presence

SIP Session Initiation Protocol; IETF signaling standard for initiating, terminating, redirecting, proxying, and performing other operations on real-time multimedia sessions

Soft Phone Software only implementation of an IP telephone

UA SIP User Agent; could be a phone (hard or soft) or a logical component of a middlebox

URI Uniform Resource Identifier; a compact string of characters for identifying an abstract or physical resource; *e.g.* `sip:bob@bigu.edu`

VoIP Voice over IP (Internet Protocol)

References

- [1] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, "SIP: Session Initiation Protocol," RFC 3261, Internet Engineering Task Force (IETF), June 2002.
- [2] SIP for Instant Messaging and Presence Leveraging Extensions (simple), IETF working group, <http://www.ietf.org/html.charters/simple-charter.html>.
- [3] Mealling, M. and R. Daniel, "The Naming Authority Pointer (NAPTR) DNS Resource Record," RFC 2915, Internet Engineering Task Force (IETF), September 2000.
- [4] Gulbrandsen, A., Vixie, P. and L. Esibov, "A DNS RR for Specifying the Location of Services (DNS SRV)," RFC 2782, Internet Engineering Task Force (IETF), February 2000.
- [5] J. Rosenberg, and H. Schulzrinne, "Session Initiation Protocol (SIP): Locating SIP Servers," RFC 3263, Internet Engineering Task Force (IETF), June 2002.