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# VoIP watch

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**WE COVERED THE VARIOUS POTENTIALS** and benefits of VoIP in the last issue of *;login:*. This month we will spend some time investigating how we can take advantage of VoIP for some toll-bypass fun. Whether you are engulfed in technologies or are involved in managing the tight budget that allows your staff to play with these technologies we will hopefully keep you thinking about or occupied by VoIP.

Building and deploying your own VoIP platform to bypass expensive telecommunication costs among partners and employees can be fun. With the amount of involvement in planning, designing of features, and managing and maintaining stability of the system, coupled with user expectations, VoIP managers and operators can be overwhelmed by the workload very quickly. Offloading some of these responsibilities to existing VoIP communities can leverage the technology and allow creation of an innovative way of communicating within an organization without much subsequent overhead.

## Private VoIP Service Providers and Communities

Some VoIPs do not work well with NAT because they use layer 3 addresses in a layer 4 protocol. For example, an SDP message (the payload within a SIP packet) may contain the private RFC1918 address for which the SIP INVITE (beginning of a SIP conversation) may have originated. If the SIP recipient beyond a NAT firewall tries to respond and contact this private address, the conversation can never be established. There are different techniques for getting around this issue by mangling layer 3 addresses in layer 4 protocols so that replies can be routed properly.

Of the various protocols widely used today, SIP seems to have dominated over its competitors. Even though SIP shares some of the limitations and restrictions around NAT gateways and firewalls much like its predecessor (i.e., H.323), many still prefer the protocol because of its wide acceptance.

There are a handful of private VoIP communities supporting various protocols and applications, including, among others:

- Free World Dial-Up (SIP)
- Skype (proprietary protocol)
- Gizmo project (SIP)
- Vonage (SIP)

Whether they are bridging between a VoIP user and the PSTN world or providing peer-to-peer (P2P) communication over the Internet, these private communities allow your VoIP packets to travel within their framework. Although pure VoIP communication is certainly a free service, it is important to understand that VoIP service isn't always a giveaway—especially when one of the legs of the conversation originates from or terminates on the PSTN. The costs are usually associated with the capacity (or the lack thereof, in this case) that is and can be provisioned on the PSTN service subscribed by the VoIP service provider (or whoever is providing the bridge between VoIP and PSTN).

Free World Dial-Up (FWD, http://www.freeworlddialup.com), one of the first such private services, is based on SIP. Members can connect to other members by dialing an account ID or via a SIP URI. FWD has supernode(s) with which a client application (also known as a SIP User Agent) registers.

The SUA can be a soft SIP phone running on a PC, or it can be a dedicated computer with an embedded OS and a real handset and keypad (i.e., IP phones). Besides P2P communication, FWD also supports delivery to regular telephone numbers in cities around the world via SIP, as well as offering a global calling plan that allows calls to be made to worldwide destinations at competitive rates.

With Cisco's migration from the H.323 protocol to SIP-based IP phones and Microsoft's introduction of MSN Messenger to replace its H.323-based NetMeeting conferencing software, there is no doubt that SIP is gaining in popularity. However, this does not preclude others from inventing their own VoIP protocol.

Skype is no longer just hype; it's proven to be of interest to many home users and business travelers, as its ease of installation and call quality surpasses some of the early IP telephony software running on Windows and Mac platforms. To expand its footprint, Skype has partnered with hardware manufacturers to maximize usage of their VoIP products.

The biggest obstacle between the open source community and Skype is the lack of openness in the Skype protocol. There are hacks for, say, an Asterisk server wanting to send VoIP traffic to the Skype network—through the use of a middleware machine running the Skype software. However, this is not an elegant solution; thus the Gizmo project was born. Gizmo is not yet an open source project, but it employs open standards—SIP, allowing someone to call to and from other SIP networks on hardware/software clients around the world. Gizmo is a relative newcomer compared to Skype, but both offer similar paid services—the ability to accept inbound PSTN calls from cities around the world and to terminate outbound calls to the PSTN for just a few pennies per minute.

Most people may not consider Vonage to dwell in the same realm as some of the VoIP communities discussed. In principle, Vonage allows P2P calling without any additional service charge with an ATA (analogue telephone adapter). But Vonage takes it one step further by replacing the traditional POTS line and providing all subscribers with call-in and call-out type services for a monthly charge.

## Build Your Own VoIP Platform, and More . . .

With the availability of open source PBX and other telephony platforms, one can easily go about replicating the infrastructure of some of these private communities. All that is required is a machine that can route and bridge calls between the VoIP and the PSTN, using an appropriate protocol:

- SIP, with a way of mangling the layer 3 address described above
- IAX (Inter Asterisk Exchange) protocol, which is designed to work well with NAT

For simplicity, let us assume a 64-kbps CODEC (i.e., the ITU G.711), which is used since it is the most widely supported CODEC for sampling voice stream; an ATA device that supports DHCP for those who work remotely at their home office; and a software-based IP phone and a Bluetooth headset for traveling users to use the VoIP service where Internet is available, via Ethernet at airport lounges or a WiFi connection at hotel hotspots. Last but not least, provide a Web tool so that employees may forward their office extensions to wherever they wish—to a soft VoIP phone or a landline.

If there isn't already a corporate standard in Instant Messaging and VoIP, choosing one that does not rely on a proprietary protocol is highly recommended. Some may argue that software that supports open standards tends to have fewer compatibility issues and a quicker turnaround time for patch releases. The Gizmo project definitely places high on the list in this respect, as it mimics the capability of Skype with the use of SIP.

A Gizmo user can populate the contact list with friends who already have a Gizmo account; he or she can also use the same software to access company voicemail and connect to any IP-enabled extensions in the office while traveling. This is made possible by creating SRV records in the company's DNS:

```
_sip._udp        IN       SRV     20 0 5060 pbx.mycompany.com
```

SIP requests made to the URI sip:1508@mycompany.com (along with any SIP URI requests) will be directed to pbx.mycompany.com. In this case, using Asterisk as an example, extension 1508 can handled by directing the caller to a SIP-based IP phone, followed by an IAX-based IP phone, and finally to a voicemail account after 40 seconds of ringing:

```
exten => 1508,1,Dial(SIP/cisco_phone,20) ; using SIP
exten => 1508,2,Dial(IAX2/iax_phone,20)  ; using IAX no one answered after 20 seconds
exten => 1508,3,Voicemail(u1508)   ; finally, place the caller into voicemail
```

It is also possible to allow Gizmo users using Asterisk to bridge into another private VoIP community. Special extensions of Asterisk can be created to handle bridging of specific Vonage numbers, FWD numbers, or even another Skype user (using the aforementioned hack). If users want to make calls to the PSTN, they can choose either to use the native Gizmo CallOut features or to use a special extension on Asterisk to make outgoing calls and let the Asterisk dial plan decide how the call should be made (either via IP or the PSTN).

This process is good enough to support outgoing calls made from a Gizmo client to anyone:

- within the corporation
- with a Gizmo account
- with any other private VoIP community account
- on the PSTN

How about incoming calls? How can we provide a convenient way for partners and customers to contact employees without incurring steep long distance or international charges? Of course, the most favorable move is to advertise SIP URI contacts. Instead of calling the main office number followed by keying in extension numbers (or dialing the DIDs directly), the callee can simply become sip:ext@mycompany.com:

Heison Chak +1-416-977-1414 (ext. 1508) or +1-416-348-1508 becomes
sip:1508@mycompany.com

Gizmo and CounterPath (formerly Xten) X-Lite (another SUA) are capable of connecting to such a URI from Windows, Linux, or Mac OS.

To support callers on the PSTN who prefer to dial directly from their cell phones and landlines, simply giving out a SIP URI is not helpful, unless their communication devices can handle such context—although such capability shouldn't be too far off, given that there are already cell phones that are Skype-capable. However, until this capability is widely available, the easiest workaround is to find a provider who can supply a SIP-based call-in number in or near the city from which most overseas calls are made (i.e., where the callers are based).

Currently, Gizmo provides call-in numbers in the United Kingdom and in the United States, and Skype with is bigger landscape can provide numbers in 13 different countries around the world (covering Europe, Asia, North America, and Latin America). Until the Gizmo project increases its coverage significantly, one may need to do more research to find local SIP providers that allow soft-phone options. For example, a VoIP account with Hong Kong Broadband costs around U.S. $6.15/month, and Asterisk can be set up to register with the VoIP server so that incoming calls from the PSTN will be delivered via the Internet to a machine that is physically hosted in North America:

```
sip.conf:
register => 1234567:password@hkbn.net/1234 ; Register 1234567 at SIP provider as 1234 here
extensions.conf:
exten => 1234,1,Dial(SIP/cisco1)        ; calls dest. for 1234567 from HK will ring IP phone cisco1
```

This can provide great savings to a North American–based corporation for communicating with overseas customers or partners. However, since VoIP accounts are not managed under the umbrella provider (e.g., Gizmo or Skype), be prepared for considerable management headaches in judging whether the savings are worth it. For a corporation with high call volume, it may be. Otherwise, it is still a fun IT project for you fellow admins out there.