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deviating alternatives to Asterisk

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SINCE THE RELEASE OF 1.2 IN 2005 and 1.4 to follow in 2006, the Asterisk open source project has gained momentum and critical mass. With the increasing number of deployments and user base, stability and scalability are becoming critical concerns. Although Digium is standing behind Asterisk, maintaining stability in such a large-scale project is easier said than done.

Major cleanup and code audit procedures have been put in place. As a result, more efficient codes and channel drivers enhance system compatibility and stability. All latest Digium (e.g., PRI and analog FXO) interfaces also feature hardware echo cancellers to help reduce host CPU utilization. These are examples of commitments from Digium to ensure stability of the Asterisk PBX software.

Despite community effort and dedication from developers to make Asterisk a better system, many are still looking for controversial changes. Although they do not provide the same functionality and some are not even designed to be a PBX, here are a few deviating alternatives to Asterisk:

- SIP Express Router (SER)
- Broadsoft
- FreeSWITCH
- sipX
- OpenPBX

By and large, these alternatives were developed by those who started doing research on Asterisk and quickly realized that it wasn't what they were looking for.

SIP Express Router

Asterisk can be configured as a SIP registrar, acting as an endpoint user agent to the originating call leg and then creating a new call to the receiving phone—thus staying in the middle of the call. A real proxy, such as SER, is never the endpoint of a call, handles call control on behalf of user agents, and does not maintain state during a call. SER supports SIP connections with more features (it can be a registrar, proxy, or redirect server) and has better scalability.

It is quite common to use SER in conjunction with Asterisk, especially when someone is looking for PBX functionality and has a multitude of networks to traverse. Remember, SIP doesn't play nicely with NAT. Using a proxy can lift some of the burden.

Broadsoft

With Asterisk Realtime, some of the configuration management can be off-loaded to the backend database, and alteration to config files will not require a reload of Asterisk. This may be a step ahead, but it is nowhere near the big leap Broadsoft has put together: Its carrier-grade soft-switch platform provides a homogeneous view of configuration steps. From provisioning SIP accounts to assigning numbers to call features (e.g., call waiting, voicemail) on accounts, whether they are administrative tasks or user-driven options, the configuration interface is quite similar. This is a big benefit to carriers, as they can use the same interface day in and day out.

The soft-switch runs under Solaris and is usually installed and supported by the vendor. The software can be provisioned to support administrative view (for provisioning SIP accounts or assigning numbers) and customer view (for end users to manage their subscribed features).

FreeSWITCH

FreeSWITCH is an open-source soft-switch started by a bunch of Asterisk developers who respect many Asterisk architecture decisions but disagreed with the following:

- Monolithic architecture—processing, user interface, and data all residing within the same entity
- Limited support of UNIX flavors and GCC
- Limited support of 8-kHz audio sampling

FreeSWITCH has been available since January 2006 and supports a wide range of protocols, including SIP, IAX, H.323, and Jingle (GoogleTalk). Although it has IVR capability, it is most beneficial to carrier-grade applications and is not really a PBX system.

Asterisk IP PBX

Despite the richness of features in Asterisk, many applications have undergone redesign to cope with user demands and community feedback. Let us find out where some of these deficiencies may be.

MeetMe is well known for its ability to take over traditional conference servers based on TDM (time-division multiplexing). Because of the nature of VoIP, it can easily support a larger number of participants; the number of legs on a teleconference is limited by CPU, memory, and bandwidth rather than the number of licensed ports, as on legacy systems.

However, MeetMe relies on Zaptel devices as a timing source. Upon a closer look at this dependency, one sees that it is actually the Zaptel driver that is responsible for mixing audio in its buffers, and it does so for PRI or analog legs. When an IP-based audio stream (e.g., RTP or IAX) is added to the mix, MeetMe creates a proxy pseudo device and handles the conversion of reading RTP or IAX frames and writing out to the pseudo device as well as reading from the pseudo device followed by writing out RTP or IAX frames. In the absence of a Zaptel device, a software-based dummy driver (ztdummy) generates timing from the usb-uhci kernel module or uses the internal high-resolution kernel timer in Linux 2.4 and 2.6, respectively. Such dependency adds complexity to normal operations and an additional burden to upgrades.

Voicemail

Users generally praise Asterisk's ability to deliver voicemail as email attachments. However, many may soon realize that deleted voicemail attachments reappear in their voicemail box when they dial into the system—there was no easy way to synchronize email deletion with the actual voicemail storage. Some have implemented periodic removal of voicemail messages: After an email attachment has been sent, voicemail storage is swept for messages older than a predefined age. Others have implemented hooks in their IMAP storage to allow voicemail messages to be sent and to allow IMAP clients to delete messages as well as for Asterisk to remove voicemail if so requested by users calling from their phones.

The latest development provides support for IMAP voicemail storage in Asterisk, so that users can manage voicemail messages in a synchronized fashion with either their phone or their favorite mail user agent.

Fax and Asterisk

We love HylaFAX for what it does, and even if there are less complicated ways to manipulate faxes in Asterisk, many are still sticking with HylaFAX. HylaFAX is a software fax machine that communicates with fax modems and has been around for quite some time. SpanDSP is a library for digital signal processing. Two Asterisk applications, app_rxfax and app_txfax, use SpanDSP to send and receive faxes. When it works, it is very efficient: It receives the fax as a tiff image, which can be converted into PDF and emailed to a user with just a few lines of code in the dialplan. However, because it is not distributed with Asterisk, it requires some patching, and whenever there is a major upgrade (Asterisk or Zaptel), the application is prone to failure.

Alternatives

SIPfoundry's sipX is a SIP-based IP PBX. Because of its SIP proxy capability, it is believed that sipX can scale better. As long as the PBX system only has SIP user agents and all of the IP voice trunks are delivered via SIP, sipX may be a viable option. Also note that sipX does not support fax communication.

Another alternative is OpenPBX, which is a fork of Asterisk 1.2. It maintained some of the features and applications that were deprecated in Asterisk 1.4 and has significantly improved the build process and environment as compared to 1.2. However, community support of OpenPBX is nowhere near that of Asterisk. On a recent SIP exploit that alerted Asterisk and other SIP vendors to patch their systems, OpenPBX revealed its weakness in responding to the incident. Only time will tell whether this fork will survive.

A number of service providers running Asterisk 1.2 are holding off on upgrading to 1.4 and awaiting Asterisk 1.6's arrival, due by the end of this year. Other have moved away from Asterisk totally and onto FreeSWITCH or Broadsoft.

REFERENCE

http://www.asteriskguru.com/downloads/asterisk_stability_and_security.ppt.