Asterisk: A Non-Technical Overview

Nasser K. Manesh
nasser@millenigence.com

Millenigence, Inc.
5000 Birch St., Suite 8100
Newport Beach, CA 92660

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Executive Summary

Asterisk is an open source platform for converged telecommunications. It provides PBX functions and applications, as well as connectivity via TDM and packet voice. Asterisk uses host processing for TDM and DSP, a lightweight protocol (IAX) for packet voice, and a flexible application-centric architecture for PBX services. It also provides interoperability with other VoIP protocols such as SIP, MGCP, and H.323. These features plus a modular, flexible, and expandable design, enable enterprises as well as individuals to deploy VoIP services in a very cost-effective manner. Asterisk is supported by Digium, IAXTEL network, and the open source community.

Introduction

Most of the existing Asterisk literature almost exclusively provides technical details focused on its configuration, setup, and integration. This article explains why Asterisk is important and innovative without going into such technical details. It helps executives and managers in converged telecommunications market have a better understanding of key aspects of Asterisk as a VoIP platform. It also shows how an open source project can be used as a cornerstone in a business model.

Asterisk is an open source converged telecommunications platform, designed to allow different types of IP telephony hardware, middleware, and software to interface with each other consistently. It provides multiple layers, managing both TDM and packet voice at lower layers while offering a highly flexible platform for PBX and telephony applications such as IVR. Asterisk can bridge and translate different types of VoIP protocols like SIP, MGCP, and H.323. At the same time it can provide a full-featured server platform for predictive dialing, custom IVR, remote and central office PBX, and conferencing.

The name Asterisk refers to the "*" symbol which is a "wildcard" in Unix and DOS command line syntax, denoting a symbol of a very versatile component in a voice network. Implementing host based DTM and DSP, allowing multiple packet voice protocols to interact, and offering a modular design with APIs for adding new applications have made Asterisk a real wildcard in converged telecommunications.

History

Open source projects usually start out of a need that somebody with a particular level of pride has. Asterisk is no exception. Mark Spencer, the creator of Asterisk, found it to be too expensive for his new, support-centric *linux-support.net* business to buy a traditional PBX solution to handle phone calls. After all, one could theoretically hook up the voice lines to a computer with some sort of expansion card, and process them in software. He therefore started writing a piece of software to control such cards and perform voice switching services, in order to eliminate the need for a PBX. The result was the first version of Asterisk, which was later rewritten for modularity and flexibility – what we know today as the official version of Asterisk.

With his better understanding of the requirements upon developing the first Asterisk version, Mark teamed up with Jim Dixon of the *Zapata Telephony Project*, to build inexpensive expansion cards for commodity Intel-based hardware to serve as the interface of an Asterisk server platform to the PSTN. The idea was that one could buy a PC from anywhere, run Linux on it, add a card to it for some FXO/T1/E1 connections, add Asterisk to the mix, and end up with a full-featured PBX.

Host Processing for TDM

Making inexpensive interfaces was not easy. Traditional TDM hardware (e.g., Dialogic, later acquired by Intel) was proprietary and expensive. To achieve the goal, Mike and Jim came up with a revolutionary idea: Instead of doing the TDM processing in hardware, offload it to the host processor and let Asterisk deal with it. CPUs were becoming faster and faster, and now it would make sense to let the software use the general CPU of a commodity PC to do the TDM processing. Mark added TDM support to Asterisk, and Zapata Telephony started making pseudo TDM interfaces, which they called Zaptel. The pseudo TDM architecture provides almost the same quality and real-time capabilities of hardware TDM but at a considerably lower price and higher flexibility.

Interestingly Intel has recently taken the same approach and is promoting a new architecture, Host Media Processing (HMP), which is based on moving the TDM processing logic into software, leveraging host CPU.

Packet Voice

While the Zaptel interface worked well for connecting the server directly to the PSTN, there was still a need to communicate over pure IP (or later Frame Relay) in case the termination or origination of the voice session was done somewhere else. Mark did not like the bulky, proprietary H.323 protocol, so he designed his own to do the job in a very lightweight manner. IAX, or Inter Asterisk eXchange, is a protocol that takes care of both signaling and transport of packet voice between two connected nodes. Although the name implies Asterisk at both ends, IAX can in fact connect any two end points supporting the protocol. There

are phones, softphones, and phone adapters available today that speak IAX and can communicate efficiently with an Asterisk node.

Later, to support interoperability with other VoIP systems, support for other packet voice protocols like SIP, H.323, MGCP, and VoFR were added to Asterisk. With such support Asterisk is now an ideal IP PBX for environments with multiple types of phones. Millenigence has successfully used Asterisk to drive different types of IP Phones such as Siemens optiPoint 600 and Cisco 7960 (with both Skinny or SCCP and SIP firmware) as well as low budget SIP Phones such as GrandStream BudgeTone and PCPhoneLine in its DashPhone Application Server™ development lab.

Flexible Architecture

Asterisk follows a modular design. Different functions of the server are implemented as modules that are loaded and initialized by a run-time loader. Asterisk provides four basic sets of APIs for interfacing with different components:

Channel API: The Channel API allows the Asterisk switching core to interface with different TDM or packet voice sources. TDM sources are usually the pseudo TDM devices for which the code performs DSP functions, although some hardware TDM source such as Dialogic boards are also supported as add-ons. VoIP protocols including SIP, H.323, and MGCP are supported.

Codec Translator API: The Codec Translator API provides a flexible way for the core to deal with encoded voice no matter where it is coming from. Formats like GSM, G.723, ADPCM, and MP3 are supported.

File Format API: The File Format API allows Asterisk to be able to read and play sound in different formats including WAV, AU, and MP3. This gives Asterisk based applications more flexibility in dealing with ring tones, DTMF, etc.

Application API: The Application API can be used by third party applications, e.g., calling cards, conferencing, and voicemail to take advantage of Asterisk PBX features.

The Application API allows developers to interface with Asterisk at any stage of call set up and teardown. This makes it possible to write new telephony applications that can directly interact with the core PBX. Asterisk also provides a separate interface, Asterisk Gateway Interface or AGI, which is a mechanism to launch external programs from within Asterisk based on dial plan rules.

The Asterisk dial plan is very flexible and allows very flexible ways of routing calls. Implementing Least Cost Routing for inexpensive call termination and Failover Trunks to reroute calls in case of primary route failure are trivial.

Digium and IAXTEL

After collaboration with the *Zapata Telephony Project* provided a successful proof of concept for the idea that one can really take DSP and TDM out of hardware and implement them in software, Mark saw the opportunity and started Digium, a company dedicated to building inexpensive Zaptel hardware and promoting and supporting Asterisk related open source and commercial software development. Zaptel

libraries, which are equivalent to what Intel will be offering as HMP software, are available from Digium for free.

Based on Digium's web site, "Digium solutions reduce the costs of traditional TDM and VoIP implementations through Open Source, standards-based software and next-generation gateways, media servers, and application servers. Digium hardware supports traditional voice protocols, including PRI, RBS, FXS, FXO, E&M, Feature Group D, Groundstart, and Loopstart. Data protocols include PPP, Cisco HDLC, and Frame Relay. For packet voice, Asterisk supports IAX (Inter-Asterisk eXchange), SIP, MGCP, Skinny, and H.323 VoIP protocols."

At the heart of the offering still resides Asterisk, which provides the host processing as well as the PBX functionalities. Keeping Asterisk free has helped Digium leverage feedback and contribution from developers around the world. People from around the world download Asterisk and play with it, post questions and comments to the Asterisk mailing list, some people happily answer questions, fix bugs, and add new functionality.

Digium's business model is currently based on manufacturing boards that are easily supported by Asterisk, providing Asterisk commercial software add-on modules, and offering Asterisk support and consulting. Most Digium boards like the TE410P, which offers four T1/E1 ports, are FCC, CE, and UL approved. This facilitates Asterisk deployments in the international market.

In July 2003 Digium announced IAXTEL (http://www.iaxtel.com), which is "primarily used by Asterisk developers and hobbyists to test and use their systems and study VoIP." Asterisk users and developers can connect their phones and PBXs for free using the IAX protocol on IAXTEL. The service is co-sponsored by Digium and VoicePulse.

Areas for Improvement

Like any other software project, there are a number of areas where Asterisk has room for improvement. Integrating Fax services, while trivial in most PBX platforms, is still undergoing development in Asterisk. The user interface is a command line, which is very flexible, but at the same time hard for some people to use.

Performance of Asterisk has not been explored much. While the claim is that Asterisk can scale well, there is no test data showing that Asterisk can support a large number of users without a performance penalty. There is no easy way to cluster Asterisk installations to scale up and match the performance of legacy PBX systems.

Flexible APIs make it possible to write any PBX related application, but doing so is out of reach for the average telecommunications expert. It still takes a software person to create a fully functional Asterisk installation for a real enterprise.

While IAX is an extremely efficient protocol, the fact the SIP is now the dominant protocol of IP-based telecommunication and is supported by virtually everybody should give it more visibility in Asterisk. Asterisk supports SIP devices, but does not provide a high level SIP API. This makes it hard to write SIP applications that interact with Asterisk.

Conclusion

Asterisk has become the de facto standard for voice switching and PBX functions in the open source environment. Host processing of TDM and DSP has made it possible to use Asterisk with inexpensive hardware in order to build flexible PBX platforms. Supporting multiple packet voice protocols and providing high-level PBX applications have made Asterisk an efficient platform in virtually any call centric environment. Growing support from the open source community and the IAXTEL network helps Asterisk remain reliable and up to date with new technologies. At the same time, lack of considerable commercial interest in Asterisk has caused less interest in areas like ease of use and canned solutions.

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