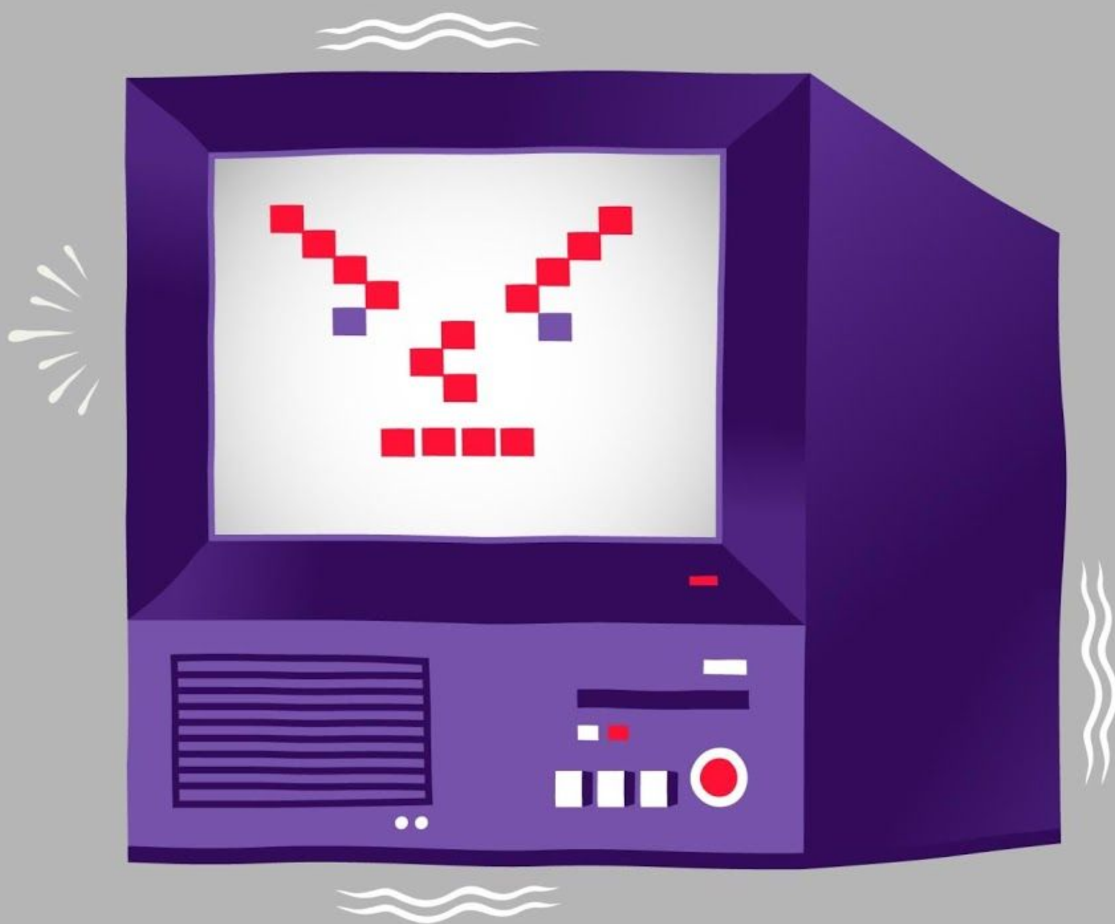
WHAT IS ASTERISK



sas

Specialty Answering Service

sas

Specialty Answering Service

Contents

1 Introduction

Asterisk is a hugely popular open source framework that can be used for building communications applications such as an IP PBX, VoIP gateways and other solutions. Asterisk was created by Mark Spencer of Digium in 1999. It derives its name from '*' the asterisk symbol. Asterisk is one of the earliest pioneers of open source PBX software packages, helping to convert a normal computer into a communications server, thus allowing the attached telephone lines to call each other as well as connect to other services such as VoIP and PSTN networks. Asterisk is released under a dual license model – the free version is released using the GNU GPL (General Public License) and there is also a proprietary software license to allow licensees to distribute the system components that are previously unpublished and are proprietary in nature and not free.

Today, Asterisk based communication systems are used by large and small businesses alike including Fortune 1000 companies. Its popularity is widespread as it is used by call centers and government agencies across 170 countries. Asterisk is truly a multi-platform product which runs on a wide variety of operating systems such as NetBSD, FreeBSD, OpenBSD, Solaris and Mac OS X, not to mention Linux for which it was originally designed. There is also a Microsoft Windows port known as AsteriskWin32. It is not bulky and can even run in an embedded environment. Asterisk is a favorite among system integrators and developers, and is flexible enough to develop an entire telephone communications system or add features to an existing system.

Asterisk can be useful in building creative and innovative applications. It is also a great learning tool for students who would like to know more about telephony and telecommunications.

2 Features

Asterisk is a feature rich software suite and has a whole host of features that are commonly found in commercial PBX systems such as automatic call distribution, voice mail, conferencing, and interactive voice systems. Users also have the flexibility to add more features by programming in C or creating AGI (Asterisk Gateway Interface) programs in any programming language that supports stdin and stdout. One can also write dial plan scripts in any of the extension languages of Asterisk or create additional functionality by using network TCP sockets. Thus, Asterisk is more a construction kit for creating a PBX rather than a PBX by itself.

2.1 Hardware Requirement

In order to convert a computer into an Asterisk server, additional hardware known as PCI cards need to be attached to the computer so that analog telephones, PSTN lines and even digital phone services can be connected to the server. There are several vendors including Digium itself who sell these PCI cards.

2.2 Protocol Support

Asterisk supports both traditional and VoIP telephony. As a result, it allows the flexibility either to build an entirely new telephone system or gradually move from the existing system to cutting-edge technology. It can be used to replace traditional PBXs or to provide add-on features such as voicemail or voice based menu systems. It can even result in cost savings for long distance calls by routing them through the Internet.

A wide range of VoIP and video protocols can be supported by Asterisk including H.323, the MGCP (Media Gateway Control Protocol) and SIP (Session Initiation Protocol). In addition to these, Asterisk also supports traditional circuit-switching protocols such as SS7 and ISDN. However, this requires the right kind of hardware interface as well as the respective software modules such as Libss7, wanpipe, Zaptel and so on. Asterisk also has a native protocol, the Inter-Asterisk Exchange (IAX2), which provides trunking among Asterisk PBXs as well as distributed configuration logic and call forwarding to VoIP service providers that support the protocol. Asterisk is compatible with most SIP telephone and it can play the dual role of registrar as well as the gateway between the PSTN and the IP phones.

In short, its wide spectrum of features make Asterisk a truly world class product.

2.3 Configuration

There are several steps required to configure Asterisk as a fully operational system. These include:

1. Creating VoIP or TDM channels or analog telephony devices that allows Asterisk to use these channels and devices as a communication voice path

2. Creating a dial plan to configure the workflow which can be used to respond to incoming calls to the Asterisk server through any of the devices or channels connected to it. The dial plan is written in the Asterisk control language. Depending on the nature of the functionality required, customized dial plans have to be created for each functionality such as using it as a PBX or as a VoIP gateway.

3. Customizing the configuration text files is the next step. One of the key configuration files is the extensions.conf file which contains the operational flow logic and acts as a starting point for the processing of calls. Elements of call processing such as variables, macros, contexts, extensions and actions are defined using a native scripting language. A context is a group of valid destination codes that can be applied to a set of channels on which the calls can be placed. Each channel declares a context, and a dial plan defines which extensions and facilities can be accessed or restricted by each device. Each extension can be thought of as an algorithm with multiple steps each of which will have a logical operation, a program flow or a call to execute an in-built application in Asterisk.

2.4 Applications

There are several applications that come packaged with Asterisk. They have a modular architecture and can be used independently to perform specific tasks such as dialing a telephone number (either external or an internal extension), performing a conference call, or handling voicemail. These applications act as the basic tool set that can be used to easily create complex algorithms that can be used to perform a totally customized telephony scenario.

The Asterisk Gateway Interface (AGI) can be used to control an Asterisk system with the help of standalone external applications. The AGI is a software interface and communications protocol that can be used for inter-process communication between external, user-written programs and the Asterisk platform. These external programs are launched from the Asterisk dial plan via pipes in order to control the workflow on the voice channels. This can be thought of as similar to the CGI feature in a web server where programs written in any language can communicate with the central server using standard streams.

2.5 Graphical User Interfaces

Several GUIs have been developed for Asterisk including the Asterisk-GUI developed by Digium itself. Others include FreePBX, Distro, Elastix, and Trixbox. These GUIs makes Asterisk installation and configuration easier by allowing administrators to view and edit various aspects of Asterisk through a user-friendly interface.

Yet another user-friendly variant aimed at novice administrators who want to have an Asterisk based PBX is AsteriskNow, which is an off-the-shelf PBX from the house of Digium. The user needs to only create the dial plans and connect the hardware. This is a boon to those who do not have prior server configuration experience as it comes packaged with FreePBX and all the required software.

3 History and Evolution of Asterisk

Asterisk began as an open source project in 1999 when Mark Spencer released the original source code and started accepting submissions from a growing community of communication systems users and developers. The end result has been a product that acts as an engine which handles all the low level details of calls between phones such as initiating, maintaining and manipulating real time media streams between end points. Asterisk has grown to be a robust system due to its enormous popularity. As per the Asterisk Website, the product has been tested and refined by a community of more than 65,000 developers spread across 170 countries to date.

Using the Asterisk system as a core engine, several complex applications have been built above it. Taking an analogy of web applications and web servers, Asterisk is to communication systems what Apache server is to web applications. A web application built on an Apache server can be as simple as a static web page or as complex as a search engine or social networking application. Similarly Asterisk can be used to build an entire telephony system or be used only for one of its features by application developers who can write programs to make it behave as a PBX, a VoIP gateway, a dialer or a combination of all these. While some Asterisk applications are simple with just the core engine, some configuration files, and Dialplan scripts, others can be more complex, connecting Asterisk to databases, web services and other external interfaces or applications.

Asterisk is now in its Version 11, which was released in the second half of 2012. There are multiple supported feature-frozen releases of Asterisk. Typically, every release series is supported for some initial period of time when bug fixing changes are included. At some point, the release series will be deprecated and only maintained with security fixes for security issues. Once a release reaches its End of Life, it will no longer be supported and no changes of any kind are made.

Asterisk versions are released as a Long Term Support (LTS) release or as a standard release. Long Term Support (LTS) releases are made from Asterisk branches where the focus has been on stability and user experience, whereas standard releases are made from branches of Asterisk that have major new features.

The duration of support depends on the nature of the release. LTS releases are fully supported for 4 years followed by another year of maintenance before reaching its End of Life. Standard releases are typically supported for a shorter period which is usually one year of full support and one year of maintenance (security fixes only).

New releases of Asterisk are made every year, alternating between standard and LTS releases. During the full support period, bug fix updates are released every month, whereas during the maintenance phase, updates are made on an ad-hoc basis based on need.

As an end user, it is advisable for call centers to choose the latest LTS release as it will have a longer support period, although the latest release (which may be a standard release) will be more feature rich. From a developer's perspective, it is important to know when the feature freeze for a particular branch occurs, which is typically 3 months prior to the release of a new version, occurring on the 3rd Wednesday of October.

The first version of Asterisk was released in 2004. A major release upgrade in 2005 included several new features as well as performance improvements and more efficiency in memory usage. Today there are a number of supported versions of Asterisk available, including the current Long Term Support release 11, the previous LTS release 1.8 and the Standard Support release 10. Additional branches include the Certified Asterisk branch of 1.8 and the "Digium Phones" branch of 10. These branches include additional support, not available in the mainline releases, for Digium's IP Telephones. Some of the key features in the various releases are detailed below:

3.1 Asterisk 1.4 in 2006

The Asterisk 1.4 version had several enhancements to improve call quality and simplify programming. It also offered compatibility with other networks such as Jabber, GoogleTalk and Jingle, and had support for Spanish and French in addition to English. Language capabilities were enhanced in terms of new sounds as well as improved support for sentence structures.

This version introduced the Generic Jitter Buffer to improve call quality during network congestion. The AEL V2 (Asterisk Extension Language Version 2) has simplified the programming and dial plan configuration. The Unified Messaging feature allows voicemail, fax and email to be collated into a central mailbox, making it easier for users to manage all forms of communication using any device. The Whisper Paging feature allows call interruption to be pre-programmed with the ability to control the volume and even mute the line.

It also offers performance enhancements such as improved interoperability of SIP call transfers, IAX2 scalability improvements, enhanced IAX2 media stream capabilities, Cisco® SCCP support, SNMP monitoring, and RTP native bridging capabilities. It also includes variable length DTMF support, the option for programming shared line appearance, centralized RADIUS storage for call detail records, a built-in web manager interface and a simplified, single user configuration for SOHO/SMB users.

3.2 Asterisk 1.6.0 in 2008

This version had several changes in multiple areas such as the Dialplan functions, IAX2, MGCP, Console Channel Driver, Phone Channel and other channels, DUNDi, ENUM, Voicemail, and MeetME. It also had an improved queue handling and had additional call features.

3.3 Asterisk 1.8 in 2010

This version had a new call logging system – the Channel Event Logging. Other changes include support for IPv6 in the SIP Channel driver, Call Completion Supplementary Services support, Connected Party Identification support and Advice of Charge support. It also had Secure RTP and Calendaring Integration.

3.4 Asterisk 10 in 2011

Asterisk 10 introduced several new features including:

- Advanced, high-performance wide and ultra-wideband conferencing application for 8-192kHz clients
- Re-architected media negotiation framework featuring support for an array of common sampling rates
- Support for SKYPE's SILK codec, offering narrow, wide and ultra-wideband audio
- Pass-Through Support for the CELT low-latency audio codec at 32 and 48kHz
- Support for the SPEEX codec at 32kHz
- New receive-side jitter buffer capabilities
- CCSS Device State Information

3.5 Asterisk 11 in 2012

Asterisk 11 being an LTS release is focused on stability, security and performance and not much on new features. Still, it does have some significant new features such as:

- WebSockets SIP Transport for real time communications with web browsers by allowing browser-based SIP clients to establish media sessions with Asterisk.
- DTLS-SRTP Support for secure transport for RTP media streams
- ICE, STUN and TURN Support for establishing live media streams between software agents running behind network address translators (NATs) and firewalls.
- A new channel driver, Motif, which combines the functions of multiple channels present in earlier versions and makes use of a more standardized XMPP implementation that can support Jingle and Google Talk.

3.6 Internationalization

Although Asterisk was developed initially in the United States, it has become universally popular due to its extensible modular architecture and its free availability under the open source licensing model. As a result developers from different geographies have contributed to its growth by introducing American-English, French, and Mexican Spanish female voices as well as Australian English prompts for the IVR system. In addition, commercially available voice sets in both genders can be easily integrated to make Asterisk truly international across languages and dialects.

3.7 Derived Products

The popularity of Asterisk has also led to several derived products – both commercial as well as freely available under open-source license. These also include commercial hardware and software bundles such as TrixBox and Elastix, where the manufacturer would release only the software as an open source.

4 Applications

Although Asterisk is most widely used as an IP PBX, it also has several components that can be used to serve a wide range of functions, thus making it the core engine of several communication applications such as a VoIP gateway, an IVR server, a voicemail server, and a conference bridge.

The modular architecture of Asterisk offers complete flexibility, by allowing its components to be combined in different ways to build various applications. For example, by combining the SIP channel with the PSTN interface channel, and using some Dialplan script, it can function as a gateway. By modifying the Dialplan to move the calls into a ConfBridge session, a conference call server can be created. A voicemail server can be created by introducing the logic to route the calls to voice mailboxes.

Since Asterisk can be easily integrated with existing applications, it can be used to build multimodal communication capabilities into existing applications such as Workforce management tools or CRM tools. This helps to offer features such as remote working for your staff or web-based call back for customers, enhancing the capabilities of the existing systems.

In the following sections, we will look at how Asterisk can be used for performing various functions.

4.1 Asterisk – PBX

Asterisk was originally created as the engine for a PBX system and it includes all the components needed for a robust and scalable business phone system, including advanced features such as voicemail, call queueing, conferencing, intercom calling, automated attendant, and call parking.

Asterisk is compatible with most technologies and protocols. It can communicate through VoIP as well as analog telephone systems. Asterisk includes drivers for SIP and other protocols thus making it easy to use with any IP phone. Being an open source product, it is constantly evolving, making it one of the most modern and feature rich PBX systems.

Depending on the skill level of the user, a PBX system can be developed using Asterisk in several ways. If you need complete control over the functions offered by the phone system then a PBX system can be built from scratch using the raw Asterisk engine. The flip side is that it is time consuming and difficult to maintain. Unless the project has extremely unique requirements, this may not be the right option for a call center.

Building a PBX system using a software appliance such as AsteriskNow, is easier as the initial installation is fully automated and a computer can be customized into a business phone system in less than half an hour. The operating system, Asterisk server, drivers for telephony cards and phones and even an administrative user interface (FreePBX) are all auto-installed.
The easiest option for a call center or any other organization requiring a PBX system is to buy a turnkey solution based on Asterisk. Digium itself offers one such product known as Switchvox, which uses the core Asterisk engine along with an easy to use user interface and additional features such as instant messaging, universal inbox, and fixed/mobile convergence.

4.2 Asterisk – Conference Bridge

A conference bridge is a common number to which participants dial in so as to participate in a virtual meeting. In contrast to three way calling where the number of participants is restricted to three and is a standard feature is most phones, a conference bridge is typically an expensive add-on feature that allows several participants (up to hundreds) to join the call simultaneously. A conferencing system typically would have several virtual conference rooms, each of which can accommodate multiple participants. The limits on the number of conferences and the participants in each one varies depending on the hardware capabilities of the system as well as the terms of the license.

A conference bridge system either allows an administrator to assign different DID numbers to conference rooms or uses a single DID number along with an IVR application to route the calls to the specific room number. Some systems also offer increased security with the help of PIN numbers which can either be common to all participants, or have a different PIN number for the ChairPerson and the rest of the participants, or can even be custom generated for each participant. Conference systems with graphical user interfaces allow participants to see the speaker and the other participants as they join the call.

Asterisk offers the functionality of a conference bridge through a standard application known as ConfBridge which can be used as a stand-alone conference service or be integrated with other solutions such as an IP PBX. It is very easy to configure a conference room using Asterisk and can be done with a few lines of Dialplan scripting. ConfBridge is feature full and offers facilities such as muting/unmuting participants, adding and removing participants and so on. It also supports basic video conferencing facilities and offers a rich event structure for developers to create robust user interfaces.

The biggest benefit of a conferencing system is the significant savings on travel time and costs associated with on-site meetings. It is a key enabler for a collaborative work environment and facilitates virtual teams. A conference bridge combined with VoIP connectivity enables a call center to have agents work from diverse geographies as a single team.

4.3 Asterisk – VoIP Gateway

A VoIP gateway creates a bridge between legacy telephone systems and VoIP systems allowing them to communicate between each other. Introducing VoIP to a legacy PBX phone system helps to reduce costs and increase the feature set. A VoIP gateway connects to the legacy PBX through either digital or analog trunk ports. It converts outgoing calls from the PBX into VoIP calls and sends them over the Web to a VoIP service provider or another VoIP peer. Incoming VoIP calls are converted into the legacy protocol supported by the PBX.

A VoIP gateway offers the reliability of the PSTN to a VoIP phone system. Another advantage of using a VoIP gateway is the redundancy it offers by making it easy to communicate with a backup system in the event of a failure with the primary IP PBX with which it normally communicates. It can also act as a great tool for a phased migration from a legacy PBX system to a new IP PBX system. The PSTN trunks are connected to one interface on the gateway and the other interface is connected to the legacy PBX. The IP PBX is integrated to the gateway using a VoIP protocol such as SIP. The gateway then acts as a bridge by routing some calls to the legacy PBX (for departments which are not migrated yet) and others to the IP PBX. It also allows calls to be transferred from one PBX to another.

With a standard computer and telephony interface cards, a customer can build a VoIP gateway from scratch using Asterisk. Turnkey solutions created by Digium can also be used with Asterisk as the base.

One of the key benefits of using a VoIP gateway is that it extends the useful life of legacy equipment by allowing it to take the advantages of VoIP by replacing the traditional trunks with SIP trunks or by routing some traffic over VoIP using toll bypass. It also offers flexibility by allowing organizations to phase out the migration to a new PBX system by using the gateway as a bridge between the legacy and the new system.

4.4 Asterisk in the Call Center

With its host of features and flexible design, Asterisk is a powerful tool that can be used for building call center systems for centers of all sizes. Some of the key features of Asterisk that are useful for a call center system include its support for call queues, outbound dialing, IVRs, call recording, real time monitoring and reporting. In fact a single Asterisk server is sufficient for creating a small call center. Large call centers typically use a cluster of Asterisk servers, which can be easily scaled as the business volume goes up. Call centers with legacy ACD systems use Asterisk to provide add on features such as the IVR menu for enabling skills-based routing.

4.5 Asterisk – Voice Messaging System

Asterisk can be used to create a robust voice messaging system using its standard voicemail components. Asterisk offers multiple message store options and integration techniques, making it ideal for building a world class messaging solution. Voice messaging systems offer the twin benefits of increasing the productivity of the agents in a call center as well as enhancing customer service levels.

4.6 Asterisk – IVR

Asterisk can be used for creating IVR applications by scripting using Dialplan or by using the Asterisk Gateway Interface. The Asterisk based IVR applications can integrate with almost all external systems, as Asterisk is platform and protocol agnostic in nature.

One of the biggest advantages of using Asterisk as a tool to create an IVR system is the cost factor. It offers a free, feature rich solution. Compare this to proprietary IVR systems which would often have a per port license fee. Thus, call centers wishing to experiment with IVR systems can easily use an Asterisk based system without having to worry about the cost of the solution. Some of the functions offered by Asterisk include audio playback and recording, digit collection, calendar integration, speech recognition and synthesis, and database and web service access. These functions allow users to create a powerful IVR system that can communicate with the other systems in the call center environment to offer superior customer service. It is easy to build an IVR system from scratch using Asterisk, and all you need to know is Linux, script development and telephony. What is more, it offers the flexibility to customize the system for the specific needs of your call center as the source code is available and can be modified as required. In a proprietary system, it can be an additional cost to request a feature change to the vendor.

5 Building an Asterisk Based Call Center

Today, the world of telecommunications is no longer restricted to voice alone. The combination of multiple media such as voice and data has led to the emergence of a new protocol – VoIP, which uses an IP infrastructure as the backbone for telecommunications. Almost all contact centers make use of tools and technology based on VoIP in order to achieve superior results while keeping operational costs low, without having to compromise on the features available. Some of the key factors that call centers must analyze while migrating to a new technology include the capital investments required, the equipment lock-ins if any, change management during migration and getting the maximum operational efficiency using the platform.

Asterisk has all the necessary features required for building world class call center software. With Asterisk as its base, call center software can have all the features necessary to achieve world class operational efficiency such as an ACD with Skills-Based Routing, CTI integration with APIs, IVR, Predictive Dialer, Call Scripting, Campaign and List Management, VoIP support, quality monitoring, call recording, reporting capability, and the ability to integrate with other software such as CRM. There are several benefits of using Asterisk as the underlying telephony platform for a call center solution, the key among them being the cost savings. Proprietary software which use Asterisk as a base help contact centers to achieve world-class capabilities at a fraction of the cost of legacy systems.

Using Asterisk as a telephony switch allows a contact center to drastically cut down on its technology costs. This is because, for most proprietary contact center solutions, a large portion of the initial cost is due to the underlying telephony switch. If that cost can be removed, then the contact center can focus on getting a feature rich ACD that would utilize all the features of the Asterisk switch. Since Asterisk supports both TDM (PRI E1/T1) and VoIP protocols (SIP/IAX) and allows seamless integration between the two, it can easily work with existing TDM connectivity and allow companies to migrate at their own pace to a VoIP platform. In a real life contact center, modern data applications may often coexist with legacy switching equipment. In such cases, Asterisk can act as the glue that bridges VoIP to TDM and digital to analog.

There are several other benefits in choosing Asterisk. Take for instance, voice recording – which is an inherent requirement in all contact centers for quality monitoring purposes. In the past, this meant additional expenditure and time. However, Asterisk has an built-in voice recording feature which means savings in terms of time and effort, not to mention the avoidance of change management.

The Asterisk server can be configured on any normal computer and this helps to avoid proprietary equipment lock-in. It is also easy to scale as additional Asterisk servers can be added to the existing ones, as they reach their capacities. In this way, the initial capital expenditure can be kept minimal. Additionally, this also helps to make it easy to build redundancies into your contact center architecture.

5.1 Inbound Call Centers

Asterisk in its basic form is sufficient to build an inbound-only call center. It offers several advantages such as convenient set up, its platform and protocol modular architecture, and its ability to work with QueueMetrics reporting software, which is one of the best metrics tracking software for call centers. On the other hand, a base Asterisk server can only function as an inbound call center and has limited features. The interfaces are also limited to the configuration options available and any changes to the standard options require the Asterisk source code to be modified. In addition, it can be cumbersome to use multiple servers with the same queue. There is also no built-in interface for agents.

5.2 Outbound and Blended Call Centers

There are several Asterisk based options available for use as outbound and blended call centers. Most of these are proprietary software which come with a licensing fee, but there are open source free versions available as well. Popular outbound dialers based on the Asterisk platform include:

a. SineDialer with a rich user interface that can be customized per customer, and has multi-server load balancing capabilities.

b. Skyy Consulting Dialer with a web-based client interface is capable of handling heavy call volumes. It also comes in a hosted version which is priced based on usage.

c. OmegaDial which has an IVR builder along with the dialer software.

d. GnuDialer which is an open-source GPL licensed product which also has an agent interface and has multi-server capability.

e. VICIDIAL which is yet another open-source GPL licensed product with an agent interface, multi-language support and multi-server load balancing capabilities.

From a server hardware perspective, an outbound call center is more resource intensive than an inbound call center. For an enterprise level outbound call center, it is ideal to have separate server boxes to handle the database, web server and telecommunications server requirements. If call recording is also necessary, then the server requirements could be even more from a storage requirements perspective. For the agent station hardware, softphones which run on the agent's machines can be used, or VoIP hardphones may be considered.

5.3 Asterisk Based Solutions for the Contact Center

Some of the popular software options based on Asterisk are listed below:

5.3.1 Aheeva CCS

Aheeva has an entire suite of contact center products, which are commercial closed systems, leveraging on open source products such as Asterisk. Aheeva is one of the earliest commercial predictive dialers created on the Asterisk platform. Some of the key features offered by the Aheeva CCS include call recording and screen recording, skill-based routing, and web-based agent interface.

5.3.2 Q-Suite 5.5

This ACD software is feature rich and utilizes the Asterisk switching capability to offer a great product at a reasonable price. It can be used for inbound and outbound, as well as blended call centers. It has a Web based agent interface and GUI based IVR setup, dialplan builder, and script builder. It offers an ACD with skills based routing and queue prioritization. It also has the feature of hot-desking which allows agents to be dynamically added and removed from the call center queue. The software also supports outbound call center requirements with predictive dialing, campaign and list management and do-not-call compliance. It also has detailed reporting capabilities with both real time and historical reporting.

5.3.3 QueueMetrics

QueueMetrics is a powerful call center tool that can help track nearly 150 quantitative metrics on a real time basis. It helps to identify problems related to achieving budgets and SLAs, and also helps to track agent productivity at a granular level. As the saying goes, "If you want to improve it, you must measure it," and QueueMetrics helps in achieving that.

5.3.4 OrderlyStats from Orderly Software

This is yet another metrics-tracking tool that can be effective in the operational management of the call center. It helps key decision makers in the contact center in analyzing performance measures to find opportunities for improvement in areas such as the effectiveness of the call center, agent availability, staff utilization and so on.

5.3.5 Pbxtra

This solution allows you to manage an inbound call center and offers features for efficient call routing, operations management, and performance monitoring.

5.3.6 Foehn

Foehn has several VoIP and data convergence solutions based on the Asterisk platform and is an established player in the field of IP communication solutions for call centers.

5.3.7 Switchvox

Switchvox is the commercial IP PBX based on the Asterisk platform from the house of Digium. According to Digium itself, 'Asterisk is an engine, whereas Switchvox is a fully fitted vehicle'. It supports unified communications with support for voice, fax, chat and video. It also has a much stronger support for queue management and IVR. Thus it offers an affordable solution for contact centers with upto 400 users.

Switchvox is primarily meant for small and medium enterprises who want a feature rich, cost effective and easy to operate solution, whereas Asterisk is meant for telecom developers who want to create completely customized solutions. Switchvox has an easy to use GUI for ease of configuration, whereas Asterisk will need to be configured using configuration file changes and program scripts. In Asterisk, each phone would need to be configured manually, whereas Switchvox has the capability to automatically detect and configure phones and Digium interface cards connected to it.

The major negatives about running Asterisk as a PBX is the time and effort involved in deployment and maintenance. Building an IP PBX from scratch with Asterisk requires advanced technical skills in the areas of IP networking, Linux system administration, telecommunication protocols, and script based programming. In addition, even after deployment, a contact center would need a technical expert to handle change management and day-to-day administration of the Asterisk server.

6 Conclusion

For a qualified technical expert, the Asterisk learning curve is much faster, when it comes to adding additional features. Customizing a proprietary solution would often require programming using a raw C language API. In short, if you have a technically expert team and you want a completely custom made solution or product, then Asterisk is the right choice. If not, it is better to opt for a proprietary solution based on Asterisk. This will help you get the best of both worlds, as you can keep the costs down (because of the free Asterisk engine) and still get the bells and whistles of a commercial solution.

7 References

1. http://en.wikipedia.org/wiki/Asterisk_(PBX)

2. www.asterisk.org