Voice over IP

Voix Manager is a switchboard for managing VoIP connections and accounts stored on Asterisk PBX server. This is a perfect solution for secretaries in companies using VoIP connection as the main type of communication.

The following features were implemented in Voix Manager:

- **User status.** Software administrator is able monitor status of all current Asterisk PBX user accounts connected to a particular server.
- **Automatic switch to VoIP connection.** When a user makes a call via Voix Manager over land phone line, application identifies whether a receiving party user is calling has VoIP phone and if it does, Voix Manager automatically switches this call to the web making this call free of charge.
- **Custom search.** Customized search is available to administrator enabling to look for Voix Manager user accounts by the following criteria:

  First Name

  Last Name

  Company Name

Picture 1. Interface

- **Call transfer.** Besides monitoring user status, administrator is capable of trasferring calls from different user VoIP accounts using easy drag'n'drop navigation.

What is Asterisk® And What to Do With It?

Asterisk® (www.asterisk.org) is a complete IP PBX (Private Branch Exchange) in software. It runs on a wide variety of operating systems including Linux, Mac OS X, OpenBSD, FreeBSD and Sun Solaris, providing all features you would expect from a PBX including many advanced features that are often associated with high end (and high cost) proprietary PBXs. Asterisk's architecture is designed for maximum flexibility and supports VoIP (Voice over IP) in many protocols, interoperating with almost all standard-based telephony equipment using relatively inexpensive hardware.

Asterisk® needs no additional hardware for VoIP communications. For interconnection with digital and analog telephony equipment, Asterisk® supports a number of hardware devices.

Asterisk® supports a wide range of protocols for handling and transmission of voice over traditional telephony interfaces including H.323, Session Initiation Protocol (SIP), Media Gateway Control Protocol (MGCP), and Skinny Client Control Protocol (SCCP). Using the Inter-Asterisk eXchange (IAX™) VoIP protocol Asterisk® merges voice and data traffic seamlessly across disparate networks. The use of Packet Voice allows Asterisk® to send data such as URL information and images in-line with voice traffic, allowing advanced integration of information.

Asterisk® offers both classical PBX functionality and advanced features, and interoperates with traditional standards-based telephony systems and VoIP systems. It offers the advanced features often associated with large, high end (and high cost) proprietary PBXs. The most common features are: Automated Attendant, Call forwarding, Call recording, Call transfer, Conference Bridging, Caller ID, Flexible Extension Logic, Interactive Voice Response (IVR), Music On Hold, Remote Call Pickup, Talk Detection, Voicemail.

It also supports a number of codecs such as G.711 (A-Law & µ-Law), GSM, iLBC, Speex and some others. The full list of supported features can be found at http://www.asterisk.org/support/features.

Typical Asterisk® installation with minimum configuration comes down to the following:

Original code is downloaded from the web site of its developer. Then it is compiled and installed on the computer which already has Linux or any other OS.

Immediately after its installment just the minimum feature set, announced by the developer, is available. For realization of required function it is necessary to create so-called dialplans, which will realize the announced feature. The dialplan is truly the heart of any Asterisk® system, as it defines how Asterisk® handles inbound and outbound calls. In a nutshell, it consists of a list of instructions or steps that Asterisk will follow.

www.sibers.com

Lavrentyeva av., 6/1-605, Novosibirsk, Russia, 630090

e-mail: info@sibers.com

In a nutshell, it consists of a list of instructions or steps that Asterisk will follow. Unlike traditional phone systems, Asterisk's dialplan is fully customizable. All functionality can be implemented with the help of «applications». Each application performs a specific action on the current channel, such as answering, playing a sound, accepting touch-tone input, or hanging up the call.

It is necessary to create user accounts after dialplans creation. Each user is able to use soft-phone for making calls by installing one of soft-phones found in a big diversity in the Internet. But to use it the user should also have a headset and a microphone.

Now, after diaplans and user accounts are created each user can make calls to other users of the current server. This way internal PBX can be realized for an office. In case the company is spread out through several offices and they are located in different buildings or cities, several Asterisk® servers can be connected with each other via SIP or IAX protocols. In this case we will have distributed PBX network with a single space of numbers. To differentiate the numbers of a local office and numbers of other offices additional prefixes for dialing can be implemented.

But usually calls have to be made not just to colleagues from work but to friends, clients and etc. To do this VoIP provider can be used. Selection of provider is completely up to your preferences. At present there are various providers of IP telephony via SIP protocol, which is de facto international standard. This way by connecting your PBX server to provider you are able to make calls to other cities and countries. A lot of providers can offer you direct numbers of various countries and regions enabling you not only to make calls but to receive them at your number as well.

Up until now we did not used any equipment. The simplest example of VoIP hardware is USB Phones, enabling its user to switch from inconvenient headset and microphone to more common phone tube. Such phone is connected to USB port of your computer and should be supported by soft-phone you've used with other equipment.

Another type of equipment is hardware VoIP telephone. Mainly these phones work with SIP protocol and are represented as regular SIP clients enabling to completely replace soft-phone. Also there are MGCP and H.323 protocol phones. And usually the protocol they support depends on the version of your firmware. Another example of such equipment is Cisco phone, reliable and presentable (although pretty expensive), that works with SCCP protocol, supported in Asterisk®.

There is a possibility of using analog phone by VoIP server. ATA (Analog Telephone Adapter) device is used for connecting analog phone to VoIP server. These devices have installed clients, which work with SIP or IAX protocol, and also cut off point for connecting analog phone and Ethernet network. This way, a regular phone can be used to make calls via VoIP channels after configuring SIP client to work with particular VoIP server. Some ATA class devices enable their users to make calls over an analog phone lines. In such case analog phone can be used with analog phone line and VoIP channels at the same time. The advantage of these devices is that they are pretty cheap (usually, no more than $100 each) and also that they can be used remotely.

Another variant of analog phone connection is the use of PCI cards, plugged in directly into computer, where Asterisk® server is installed. Such cards are manufactured by Digium (www.dugium.com), which develops Asterisk®. These cards make it possible not just to connect analog phones but traditional analog phone lines to PBX server. At present there are PCI cards enabling to connect from 1 to 24 analog phone lines and/or phones. During this type of connection the connected phone should be installed not far from PBX server but in office environment this limitation is not too significant. On another hand, the advantage lies in possibility of connecting traditional analog lines.

Digium company also manufactures PCI cards, enabling to connect digital E1/T1/J1 phone lines. Up to 4 E1/T1/J1 phone lines can be connected to these cards, which in total equals to 128 channels in E1 mode or 96 channels in T1/J1 modes.

Asterisk® server output can vary depending on configuration of computer, on which Asterisk® is installed, and also on the type of channels and codecs used. For clearer orientation only approximate numbers can be demonstrated. Let's imagine some computer with the following configuration: Intel P4 3 GHz with Hyper Threading and 512 MB RAM. Simultaneously this server can process from 100 to 500 concurrent calls via SIP protocol without quality loss of sound and significant delays. It should be noted that a number of calls and a number of users currently registered on a server are two different things. Although registered but not calling at the moment users are also affect server output. The codec used and necessity of various codecs and protocols installment influence on the number of concurrent calls. For example, the most popular g711 codec requires fewer resources for coding/decoding than more compact but more exacting on resources GSM codec. Also it is relevant whether both parties participating in a call have the same or different codecs. It is necessary to take into account the connection technologies used by both parties. For example if calling party is connected via SIP protocol and called party is connected via analog line then communication flow conversion from one format into another will require more resources. Another parameter influencing on the number of concurrent calls is the possibility of direct talk of two subscribers without using PBX server. In this case only connection between two subscribers is happening on the server and voice flow evades the server. Such feature is offered, for example, by SIP protocol but it can be used only when both subscribers are using one network and the same codec.

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e-mail: info@sibers.com