Using Asterisk with Odin's OTX Boards

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Abstract

Odin TeleSystems supports corporate open-source telephone systems that take advantage of VoIP. Open-source products are still at the leading edge of the enterprise telephony market, they are less expensive than the home-grown systems from companies such as Avaya and Nortel.

Odin's open-source phone systems can save companies 50% or more compared with proprietary voice systems. This is largely because open-source software is available at little or no cost, and it runs on off-the-shelf computers or servers. With open-source telecom systems, companies also avoid getting locked into a relationship with one vendor and do not depend on that vendor to create new features. Instead, any programmer versed in Linux can customize a telephone system to the needs of a particular business.

This paper highlights the advantages of Asterisk support and its prospects for the future.

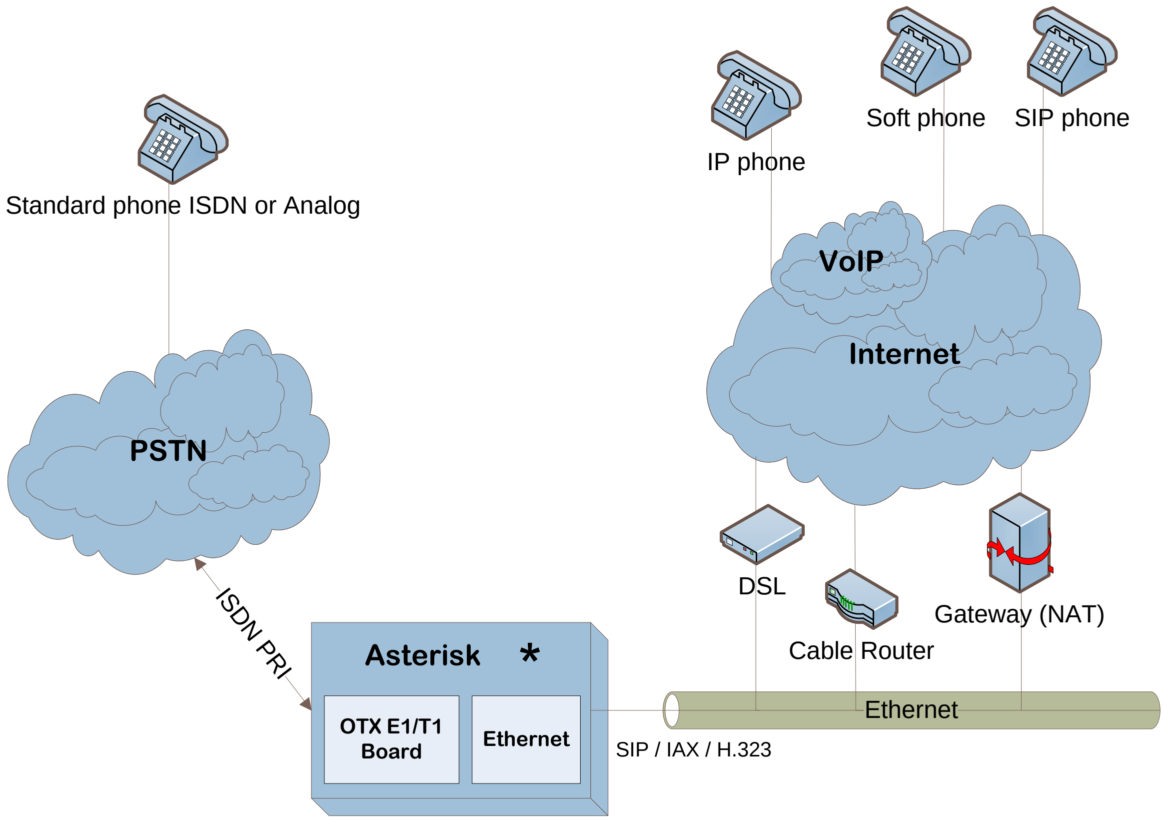
Overview

Asterisk is a complete PBX software. It runs on Linux, BSD, Windows (emulated) and OS X and provides all of the features you would expect from a PBX and more. Asterisk does VoIP in four protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk provides voicemail services with directory, call conferencing, interactive voice response, and call queuing. It has support for three-way calling, caller ID services, ADSI, IAX, SIP, H.323 (as both client and gateway), MGCP (call manager only) and SCCP/Skinny.

Asterisk needs no additional hardware for VoIP, although it does expect a non-standard driver that implements dummy hardware as a non-portable timing mechanism (for certain applications such as conferencing). A single (or multiple) VoIP provider(s) can be used for outgoing and/or incoming calls. Outgoing and incoming calls can be handled through entirely different VoIP and/or Telco providers.

For interconnection with digital and analog telephony equipment, Asterisk supports a number of hardware devices like Odin's OTX compatible devices. Odin has single, quad and octi span T1 and E1 interfaces for interconnection to PRI lines and channel banks also a complete system integration all-in-one Asterisk enable devices like Alvis-4-CSI and Alvis-8-CSI.



Asterisk support on OTX Hardware

For interconnection with the cellular network (GSM or CDMA), Asterisk can use the Celliax channel driver or chan mobile that is in the trunk now and there is also an unofficial back ported version.

The standalone devices are available to do a wide range of tasks including providing FXO and FXS ports that simply plug into the LAN and register to Asterisk as an available device.

Features

1. Supports traditional PBX functions

- **Call Transfer -** enables to use the external lines and numbers more efficiently by their division between subscribers, and easily switch calls from one subscriber to another or on additional services (conferences, menu).
- **Call Hold / Music Hold -** allows at the time of call to interrupt it, and dial another number, then come back.
- **Call Waiting -** when the caller is already connected, it notifies of a new incoming call with the choice of method of response (accept, reject, ignore).
- **Call Forwarding -** allows “busy”, “unavailable” subscriber to set the other numbers on which he will be available, including mobile and international ones.

- **Call Parking -** allows you to “park” the call, to put the tube and come to another phone and continue the interrupted conversation.
- **Call Pickup** - when rings the phone, located in the same group of interception, it is possible to intercept the incoming call on your phone.
- **Call Completion -** allows by clicking a special button in the tone mode (usually *) to interrupt the ongoing call and to dial the new number.
- **Caller ID and number displaying on a subscriber’s device -** allows you to inform the caller about incoming calls by displaying this information on the phone screen, and also allows to save the record in the history of calls, to display caller’s name and number in the voice mail, to set up a special type of call, etc.
- **Call Detail Records (CDR)** – allows you to save records about made calls in a text file for further analysis and pricing.

2. Supports advance PBX functions

- **Teleconferencing -** few people are simultaneous participating in a phone conversation. It is possible to dynamically create conference rooms, invite the other participants in rooms, and to protect the conference by password.
- **Voicemail and Message Waiting Indication (MWI) -** PBX function that allows you to record a voice message if the caller is not available or is busy, with sending (if needed) the recorded message by e-mail. Asterisk offers a well-designed voice mail system that can compete with many business decisions, and is already used by operators of voice mail service for their clients. The MWI feature allows you to display the number of new messages as they arrive.
- **Interactive voice response (IVR)** – enables to create a «menu» of organization with saying voice phrases and detection of pressings in the tone mode.
- **Call Center** – where queues of calls provide the strategy of calls distribution in customer service. Asterisk supports various strategies (algorithms) of the distribution of incoming calls among agents of the queue.
- **Presence -** allows you to display on your system phone and on software the status of the subscriber (available, busy, calling, do not disturb, etc.).
- **Call Recording -** allows you selectively or in the transverse mode to record conversations, as initiated by the administrator (automatically), so by the request of the user, i.e. selectively (one touch recording).
- **Call Spy -** administrator has the authority to connect in to an existing conversation.
- **Call Intrusion** – authorize connection to the conversation with the opportunity to speak with any of the parties (with one or both).
- **Speed Dial -** the ability to designate a number. This feature allows you to significantly facilitate the dialing of international numbers and to increase the speed of it.
- **Redial -** allows you to quickly dial the last dialed number, not entering it entirely (without knowing it). In many phones this function exists, but the realization of this possibility at PBX will allow you to use this feature from the old phones.
- **Recall** - in determining the number of the caller allows you to recall him back without dialing of number.
- **Hotline -** connects the caller immediately after the lifting of tube, without dialing any numbers.
- **Blacklist** - allows you to put some numbers in the “black list” and does not switch calls from them and for them.
- **Whitelist -** allows you to disable all incoming and outgoing calls with the exception of the list of unauthorized numbers switching.

- **Callback -** allows you to call on the dedicated callback number of the company, where instead of the connection it happens the connection break. After that PBX itself initiates a call to the defined number, requests a PIN code (if there is no number in the base), and then simulates the usual algorithm for taking the call.
- **Different algorithms for processing of calls, depending on the time and date, and time restrictions of access -** allows you to create different voice menus for weekday and weekend, and technically allows the PBX to put any function in depending on the time / date.

3. Traditional channels support

Asterisk supports analog lines (FXS, FXO). For integration with other PBXs, there are support for digital communication protocols ISDN PRI and SS7.

4. Modern channels support (VoIP)

Modern telecommunications are based on the Internet Protocol (IP) and on the technologies of voice transferring over the Internet - VoIP. According to the protocols VoIP, it interact the communication with other IP PBX (peering), as well as to reduce the cost of communications and to conduct secure negotiations and teleconferencing.

- **Office phones with computer -** users and administrators can control access to the telephone system via the built-in WEB interface.
- **Remote connection of users and offices** - allows you to organize the mobility of users.
- **Voice traffic termination -** reduces the cost of international calls by their directing to providers of IP telephony.
- **Geographical numbers connection -** with the help of VoIP technology it is possible to connect multiple numbers from different countries. This will greatly simplify and facilitate communication with the company and its staff on all parameters.
- **Quality of service (QoS) support** – allows you to increase the priority of voice traffic.
- **Unlimited growth of the subscriber base -** using VoIP channels allows you to solve the problem of growth of the subscriber base, as the maximum number of simultaneous calls and subscribers depends only on server’s and communication channels’ capacity.

5. Billing

Telecom system must have a full system of AAA (Authentication, Authorization, and Accounting). It should be very flexible abilities to manage subscriber connections (security), permitted lines (cards), counting of the cost (termination). Each user must have an individual account, which records all of his calls, balance, rate plans, addresses, etc. Subscription such opportunities are available with the integration with external billing system.

6**. Number plans**

In order to maximize the flexibility of IP PBX Asterisk supports an unlimited number of isolated number plans, allowing employees of different offices have the same internal numbers.

7. Monitoring

- **SNMP support to collect statistics -** allows you to monitor the number of connections, the load of the system and network interfaces, the availability of VoIP peers, and to notify and respond to the problem.
- **Operator Panel** – uses for the visualization of connections and ongoing calls, interception, management conferences, conversation listening, the number of waiting in the queue, connected operators, parked calls.

8. Security

Using of passwords to restrict access and authentication: every function of PBX may be "under the password" as a PIN code.

9. Networking opportunities

A firewall, router and VPN server. Asterisk runs on Linux and FreeBSD, known as reliable and high-performance network operating system, what allows you to combine the functions of PBX and network router with functions of NAT, Internet access server or remote access with PAP / CHAP authentication support, to establish on the base of Asterisk PBX a virtual private network using IPSec, LT2P, OpenVPN.

Conclusion

Asterisk support on Odin TeleSystems cards presents significant cost savings without sacrificing high performance, which becomes more dramatic when you compare it to feature-rich telephone systems like those used for call centers or for conferencing. Features like conferencing are usually very expensive, but when you have commodity PCs—and we make very inexpensive hardware that allows you to connect those PCs to the conventional phone network—and you combine those together. There is also the flexibility to be able to customize the product.

About Odin TeleSystems Inc

Odin TeleSystems Inc. is a privately held Texas corporation specializing in manufacturing, design, and sale of OEM-subsystems for the Telecommunications industry. Odin's award-winning products represent outstanding cost/performance value for today's service providers and telecom equipment manufacturers. Innovative and flexible systems enable service providers and equipment manufacturers to provide reliable and leading-edge communications services and products for T1/E1/J1, Integrated Services Digital Networks (ISDN), Frame Relay, Voice over IP (VoIP), Signaling System Number 7 (SS#7), and Digital Wireless (e.g. GSM).
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