Digium Technical Paper | Custom Telephony Solutions with Asterisk

Asterisk As Conference Server

Description

Teleconferencing is an important part of modern business communications, connecting offices, remote workers and customers. Conferencing can save thousands of dollars and hours otherwise spent on travel, lodging and logistics for face-to-face meetings. Unfortunately, traditional conferencing systems tend to be proprietary, difficult to integrate and prohibitively expensive. Asterisk changes that.

With Asterisk's integrated conferencing features any company can operate its own conference server for a fraction of the price of traditional commercial systems. Asterisk-based conference servers can operate as stand-alone devices connected directly to the PSTN, or as private resources connected behind existing PBX systems. The Asterisk dialplan scripting language makes it easy to integrate the conference server with ODBC-compatible databases or LDAP directories.

Best of all, the price of an Asterisk conferencing solution can't be beat. VoIP-only systems require no additional hardware and no per server or per seat license fee. Systems that connect with analog or digital telephony technologies require only a Digium interface card. Support subscriptions for individual (non-clustered) systems start at only $595 per year. Conferencing has never been this flexible, this simple or this inexpensive.

Supported Scenarios

- Private conference bridges
- Ad hoc/open meet-me (dial-in) conferencing
- Scheduled conferencing

Features

- Multi-party full duplex conferencing.
- Multi-protocol access enables connections using both VoIP and PSTN technologies.
- Integrates with calendaring systems for scheduled conferences.
- File or database PIN numbers for security.
- Per-user or per-conference room PIN access.
  - Conference recording
  - No per user or per seat license restrictions
  - Connects directly to PSTN resources or behind a PBX

Benefits

- Improve customer interactions, internal meetings, remote employee training and virtually any other kind of meeting.
- Save significantly over the cost of either traditional premise-based conference servers or hosted conferencing services.
- Reduce or eliminate travel costs by switching to virtual meetings with audio conferences.

Components

The components required to create a conference server with Asterisk range from the basic (simply a computer running Asterisk for VoIP-only systems) to slightly more complex for systems that integrate with either the PSTN or a legacy PBX. Most implementations require a combination of the following:

- Generic x86 computer platform (server or desktop)
- Linux operating system
- Asterisk telephony engine
- Digium digital or analog interface card(s)
- Interface cable(s) to legacy system

The computer can be any standard x86 (Intel or AMD) computer. The system will need to include either PCI or PCI-Express expansion slots. If the system will be connected to any legacy telephony interfaces (public or private) the chassis must be large enough to accommodate the interface cards. The system should be at least a Pentium IV or equivalent for a small bridge (up to 8 parties connected over SIP or analog ports). Larger bridges and bridges that require transcoding (translation of the audio media from one format to another) will require more powerful hardware.

The operating system can be virtually any modern 2.6-series distribution of Linux. Digium recommends the AsteriskNOW distribution, which comes with Asterisk and the interface card drivers pre-installed. Digium offers support subscriptions for systems running RedHat Enterprise Linux 4/5, CentOS Linux 4/5, Ubuntu Server Long Term Support (LTS), Debian stable (currently "Lenny"), SUSE Enterprise Linux 10/11 and OpenSUSE 10/11.

The current Asterisk release is available as a binary installation for the RedHat and CentOS family of Linux distributions, and comes pre-installed on the AsteriskNOW distribution. It can be installed from source code with a few simple commands on any other supported Linux distribution. Details on downloading and installing Asterisk or the AsteriskNOW distribution are available at www.asterisk.org.

Interface cards are required to tie the conference bridge in with legacy telephony technologies. To connect a PBX with analog station ports or to connect directly to analog PSTN lines, the system will need one or more Digium analog card. Connections to larger scale legacy systems or the

Company	Product	Price	T1 Port Price	E1 Port Price
Avaya	IP Office Conf. Bridge w/ 1 T1	$10,569.95	$440.00	$74.00
Aastra	CNX-30 Conf. Bridge w/ 4 T1/E1	$5,800.00	$62.00	$48.00
Digium	Asterisk Conf. Bridge w/ 4 T1/E1	$3,495.00	$36.00	$29.00

Table 1. Cost comparison of commercial conference bridge solutions

PSTN over digital T1 or E1 connections require a Digium single, dual or quad-spand digital card. Connection to ISDN-BRI PBX station ports or ISDN-BRI lines is accomplished using a Digium BRI or analog/BRI hybrid card.

Comparison With Commercial Conference Bridge Solutions

Assembling a conference bridge from Asterisk and a standard Linux computer can save significantly over the cost of purchasing a commercial system. For example, the table above comparea the cost of building an Asterisk-based bridge with 24 connections to a PBX over a T1 span plus multi-protocol VoIP support with equivalent solutions from Avaya and Aastra.

Asterisk Gateway Component Prices

$1,200 – Server Computer with Quad Core Intel Xeon (estimated)

$1,700 – Digium Quad Span (4 T1/E1) Interface Card w/ Hardware Echo Cancellation (estimated)

$0.00 – Linux Operating System

$0.00 – Asterisk Telephony Engine

$595.00 – Digium L1 Support Subscription

digium
The Asterisk Company

More Information

To download Asterisk or AsteriskNOW or for more information on how to create a VoIP gateway or other solution with Asterisk, see: www.asterisk.org

To buy Digium interface cards, support subscriptions, add-on software, check out the Digium web store at: **http://store.digium.com** or contact Digium to find a reseller in your area.

For information on Digium's complete line of Asterisk training courses, visit **www.digium.com/en/training**

For information on support subscriptions for Asterisk, visit **www.digium.com/en/support**

The Asterisk Company

For more information, go to **www.digium.com**

Digium, Inc. • 445 Jan Davis Drive NW, Huntsville, AL 35806, USA • Phone: +1 256-428-6000 • Fax: +1 256-864-0464 • www.digium.com