Introduction to VOIP and Asterisk

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What are we Talking About?

Voice Over IP (VOIP) allows telephone conversations to travel over a LAN or the Internet instead of traditional telephone wiring

Asterisk is a "phone system" that can connect to many different devices, either traditional phone hardware or VOIP hardware.

Telephone Terminology

Foreign exchange office (FXO) is a port that connects to the phone company

Foreign exchange system (FXS) is a port that connects to a telephone

Analog telephone adapter (ATA) is a device with an ethernet port and a telephone port (FXS)

Basic Phone Example

Phone Line <-> Phone

Telephone <-> ATA <-> Internet <-> VOIP Provider

Phone System Example

Phone Line <-> phone system <-> Telephone wires <-> Phones

Phone Line <-> FXO <-> phone system <-> LAN <-> Phones

VOIP Phone System

VOIP Provider <-> Internet <-> Phone System <-> Lan <-> SIP Phones

Benefits of VOIP

Less expensive calling

More flexible

Location independent

Downsides of VOIP

More complicated

Internet is "best effort"

Requires bandwidth

Not good for 911

Protocol Terminology

Session Initiation Protocol (SIP) manages a telephone connection between two parties

Session Description Protocol (SDP) provides the parameters for communicating between two parties

Real time Transport Protocol (RTP) carries the voice data itself

Protocol Terminology (cont)

COder/DECoder (CODEC) converts voice into various data formats

Simple Tunneling of UDP through NAT (STUN) determines how your firewall will interact with the communication

Inter Asterisk exchange (IAX) is an Asterisk protocol that Asterisk uses to do the same sorts of things.

Anatomy of a Call

You dial your phone and your ATA starts handling the call

It uses STUN to determine how it connects to the internet

It uses SIP to authenticate with the VOIP provider

Anatomy of a Call (cont)

It uses SIP to initiate the call

Once the call is connected to the remote end, SDP is used to determine how you and the other party will talk. SDP is embedded in SIP packets.

RTP transmits the voice back and forth.

SIP is used to shut down the conversation

Linux Soft Phones

SJPhone

Ekiga

Xlite

Gizmo

Voice Hardware

USB Phone

Headset for sound card

Bluetooth

Microphone & Speaker (Not recommended)

VOIP Terminology

Plain Old Telephone System (POTS) is the old "ma bell" phone system

Packet Switched Telephone Network (PSTN) same thing

Direct Inward Dial (DID) is a phone number

Termination is the service for connecting a call to someone else

Free Service Providers

Free World Dialup

SIPPhone

IPKall

Earthlink

Google (eventually)

VOIP Providers

Vonage

Voicepulse

AXVoice

Broadvoice

Sunrocket

Packet8

VOIP Providers (cont)

Voxee

VoipJet

nufone

Many others...

Introduction to Asterisk

Asterisk is a device independent voice platform

Runs on Linux

It works with analog telephone devices

It works with VOIP

It does it's own processing

Features of Asterisk

Multiple extensions aka Portable Branch Exchange (PBX)

Interactive voice response (IVR)

Voicemail

conference calling

Features of Asterisk (cont)

Time dependent processing

Music on hold

Call queues

Scripting

Application integration

Call logging & recording

Real Asterisk Configuration (mine)

Sipura 3000

Linux system (P4)

SPA-841 phone

Analog phone

Real Asterisk Configuration (cont)

Verizon

Several VOIP providers

Broadband internet

Real Asterisk Configuration (cont)

Interactive Voice Response (phone menus)

Group & individual phone numbers

Voice mail emails & pages cell phones

Call queues

Independent call paths

Real Asterisk Configuration (cont)

Caller-ID database

Calling between extensions

Least cost routing

Forces routing

Conference calls

Asterisk Distributions

Asterisk at Home now Trixbox

Asterisk

Fonality

Asterisk as a platform

Read the weather

Dating service

Crack the safe game

Zork

Experiment 1

Get a soft phone

Get a sound card headset ($10)

Join FWD/SIPPhone

Make calls

Experiment 2

Setup Asterisk

Create a simple dial plan that plays a sound

Call Asterisk using your soft phone

Experiment 3

Configure Asterisk to use FWD/SIPPhone

Configure soft phone as an Asterisk Extension

Make calls

Experiment 4

Get an IPKall account

Configure it to point to your FWD/SIPPhone

Call your system using a regular phone

Experiment 5

Buy service from SIPPhone ($25 gets you inward number and 1000 minutes)

Configure Asterisk to use it

Dial in and out

Experiment 6

Build up your Asterisk configuration

Setup an IVR, Voicemail, etc.

Hooked Yet?

If you’re hooked at this point, look at VOIP providers.

Be aware of terms of service

How long are you commited?

How can you use the service?

Make sure they allow "Bring Your Own Device"

Caveats

Make sure you have a reliable 911 service

Realize that you can transfer your main number in to a VOIP provier, but VOIP providers aren’t required to let you transfer it back out.

Be sure you’re happy with the service before you tell other people the number. It’s hard to change that later.

Resources

http://www.voxilla.com

http://www.voipsupply.com

http://www.asterisk.org

http://www.voip-info.org

http://www.trixbox.org