

LOGIN: VOIP

Mark Wallis

Overview

Introduction

Softphones

ATAs

Asterisk

Final Word

Questions

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VOIP in Linux - Overview

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Overview

Introduction

Softphones

ATAs

Asterisk

Final Word

Questions

- Introduction - what is VOIP
- Basic VOIP - Softphone
- Medium VOIP - ATA
- Hardcore VOIP - Asterisk

Introduction - What is VOIP

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Overview

Introduction

Softphones

ATAs

Asterisk

Final Word

Questions

VOIP = Voice Over Internet Protocol

- Allows you to make calls over the Internet
- Cheap
- Flexible
- Standalone or PSTN-interfacing

Providers

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Introduction

Softphones

ATAs

Asterisk

Final Word

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VOIP requires a provider, just like PSTN

- Pennytel
- SIPME
- MyNetPhone
- Skype

Call Routing

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Introduction

Softphones

ATAs

Asterisk

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Questions

You can use multiple providers at once using call routing

- Decide what provider to use based on various factors
- e.g. use ProviderA for mobile calls and ProviderB for long-distance
- e.g. use ProviderA after-hours but ProviderB 9-5

Hardware

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Introduction

Softphones

ATAs

Asterisk

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Whole new range of hardware to enjoy

- VOIP Handsets
- ATA's
- Combo modem/router/ATA's

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Overview

Introduction

Softphones

ATAs

Asterisk

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Questions

ATA = Analogue Telephone Adapter

- Interfaces between your normal handset and VOIP
- Can also provide a PSTN interface for a more complex VOIP configuration
- Quite often provides auto-fallback

Phones

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Overview

Introduction

Softphones

ATAs

Asterisk

Final Word

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VOIP Phones are different to normal phones

- No PSTN plug
- Ethernet plug instead
- Has an IP address, just like your computer
- Must be configured to talk to either an ATA or directly to a VOIP provider

Network Layer

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Introduction

Softphones

ATAs

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VOIP works over a special set of network protocols

- SIP = Session Initiation Protocol
- RTP = Real-time Protocol
- NAT and firewall issues need to be addressed

Basic VOIP - Softphones

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Introduction

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Easy introduction to VOIP - Softphones

- Software on your PC that pretends to be a VOIP phone
- Require's a microphone, speakers and a VOIP account

Common Software

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Introduction

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ATAs

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Your distro probably has a SoftPhone

- Gnome = Ekiga (GnomeMeeting)
- KDE = KIAX
- MacOSX = Gizmo
- Windows = X-Lite

Skype

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Introduction

Softphones

ATAs

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but we can't forget Skype

- Massive user-base
- Free skype-to-skype calls
- Binary-only (not GPL)
- Doesn't use SIP

Mobile Phones

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Introduction

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Many high-end mobile phones also include SIP clients these days

- Windows Mobile has a built-in SIP client
- iPhone (if you have a jailbroken phone only I believe)

ATAs - two functions

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Introduction

Softphones

ATAs

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So, you want to progress past PC-based solutions like Skype and Softphones

- You want to purchase either an ATA or a combo Modem+Router+ATA
- Lots of different features including the number of ports on the device and the number of providers supported

Non-VOIP handsets

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Introduction

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ATAs

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If you want to hookup non-VOIP legacy handsets then you need 'FXS' ports

- Most-entry level ATA's have a single FXS port
- Allows you to hook up one-or-more legacy handsets

PSTN interface

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Introduction

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If you want to be able to route some calls out the PSTN network you need 'FXO' ports

- FXO ports allow you to route some calls out via the PSTN network
- Useful for local calls, free calls and emergency calls

Common Models

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Introduction

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Questions

Two common dedicated ATA's include

- Linksys SPA3102
- Minitar MVA11A

Asterisk - In-house PBX

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Introduction

Softphones

ATAs

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So, you have a couple of days of spare time why not setup your own PBX ?

- Asterisk - The Open Source PBX system
- GPL and non-free versions
- Developed by Digium

What is a PBX

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Introduction

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ATAs

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- Centralised point for all your telephone calls
- Has the concept of 'trunks' and 'extensions'
- Historically a business device

Asterisk - Features

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Overview

Introduction

Softphones

ATAs

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Final Word

Questions

By no means the complete list:

- Multiple extensions
- Multiple providers
- Voicemail
- IVR
- Inbound-call-bouncing
- Extension transfers

Asterisk Distribution

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Introduction

Softphones

ATAs

Asterisk

Final Word

Questions

Installing and configuring Asterisk from scratch isn't fun. Enter
- AsteriskNow

- For the impatient, like me
- Appliance-style distro based on RPath
- Developed by Digium
- Sometimes a version-or-two behind of Asterisk
- GPL
- Seriously easy

Trunks

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Introduction

Softphones

ATAs

Asterisk

Final Word

Questions

Setup your Trunks (aka Service Providers)

- Trunks are VOIP providers and PSTN interfaces

The screenshot shows the Asterisk web interface for configuring Service Providers. The page title is "List of Service Providers". The interface includes a navigation menu on the left, a main content area with a table of providers, and a right-hand sidebar with a "Timeout" warning.

digium | Asterisk

System Configuration
About Digium | Report a Bug | Help

Home Service Providers **Activate Changes** Logout

List of Service Providers

S.No	Service Provider	Type	Options
1	Custom - PSTN	Custom Voip	Options ▼
2	Custom - Pennytel	Custom Voip	Options ▼

Service Providers are outbound lines used to allow the system to make calls to the real world. Trunks can be VoIP lines or traditional telephony lines.

Timeout: This option defines the time in seconds that an Agent's phone rings before the next Agent is rung, i.e. "15"

Add Service Provider

Trunks

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Overview

Introduction

Softphones

ATAs

Asterisk

Final Word

Questions

Trunk to Pennytel

The screenshot shows the Asterisk web interface. The top navigation bar includes 'Home', 'Users', 'Conferencing', 'Voicemail', 'Call Queues', 'Service Providers', 'Setup Hardware', 'mISDN Config', 'Calling Rules', 'Incoming Calls', 'Voice Menus', 'Call Parking', 'Ring Groups', and 'Record a Menu'. The 'Service Providers' section is active, displaying a description: 'Service Providers are outbound lines used to allow the system to make calls to the real world. Trunks can be VoIP lines or traditional telephony lines.' The 'Edit Service Provider' window is open, showing the following configuration:

- Provider Type: Analog VoIP Custom VoIP
- Comment: Pennytel
- Protocol: SIP
- Register:
- Host: sip.pennytel.com
- Username: 61240166163
- Password: *****

Buttons for 'Save' and 'Cancel' are visible at the bottom of the window. On the right side of the interface, there are links for 'System Configuration', 'About Digium', 'Report a Bug', and 'Help'. Below these links are 'Activate Changes' and 'Logout' buttons. A 'Timeout' note is also present: 'Timeout: This option defines the time in seconds that an Agent's phone rings before the next Agent is rung, i.e. '15''.

Extensions

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Overview

Introduction

Softphones

ATAs

Asterisk

Final Word

Questions

Setup your extensions (aka Users)

- Extensions are Hardware VOIP phones, Softphones and legacy hardware phones through FXS ports

The screenshot shows the Asterisk web interface for 'User and Phone Configuration'. The left sidebar contains a navigation menu with items like Home, Users, Conferencing, Voicemail, Call Queues, Service Providers, Setup Hardware, mISDN Config, Calling Rules, Incoming Calls, Voice Menus, Call Parking, Ring Groups, Record a Menu, Active Channels, and System Info. The main content area is titled 'User Extensions:' and lists several extensions: 6000 -- Office, 6001 -- Kitchen, 6050 -- Check Voicemail, 7000 -- Voice Menu, and 8000 -- Call Queue. To the right, there is a form for configuring a specific extension (6000). The form includes fields for Extension, Name (Office), Password, VM Password (6000), E-mail, Caller ID, Analog Phone, Dial Plan (DialPlan1), and Phone Serial. Below these fields is the 'Extension Options:' section with checkboxes for Voicemail, SIP, Call Waiting, Can Reininvite, In Directory, IAX, Is Agent, 3-Way Calling, and NAT. There are also input fields for DTMFMode (rfc2833) and Insecure. At the bottom of the form are 'New', 'Delete', 'Save', and 'Cancel' buttons. The top right of the interface has links for 'System Configuration', 'About Digium', 'Report a Bug', and 'Help', along with 'Activate Changes' and 'Logout' buttons. A tooltip on the right side says 'Move the mouse over to a field to see tooltips' and 'Need a phone, some software, or something else? Click on the 'Where to Buy' button to get directly to products.' with a 'Where to Buy' button below it.

System Configuration
About Digium | Report a Bug | Help

Activate Changes | Logout

Move the mouse over to a field to see tooltips

Need a phone, some software, or something else?
Click on the 'Where to Buy' button to get directly to products.

Where to Buy

Edit Codecs
disallow: all
allow:all

New Delete Save Cancel

Dial Plans

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Overview

Introduction

Softphones

ATAs

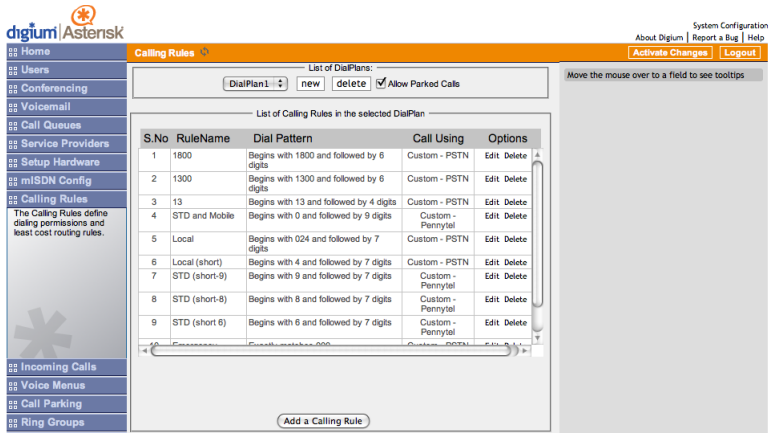
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Setup your dialplans (aka Dialing Rules)

- Based off number length, makeup and prefix
- Prefix 'stripping' is supported



The screenshot shows the Asterisk System Configuration web interface. The left sidebar contains a navigation menu with items like Home, Users, Conferencing, Voicemail, Call Queues, Service Providers, Setup Hardware, miSDN Config, Calling Rules, Incoming Calls, Voice Menus, Call Parking, and Ring Groups. The main content area is titled 'Calling Rules' and shows a 'List of DialPlans:' dropdown set to 'DialPlan1'. Below this is a table titled 'List of Calling Rules in the selected DialPlan' with columns for S.No, RuleName, Dial Pattern, Call Using, and Options. The table lists 9 rules, including 1800, 1300, 13, STD and Mobile, Local, Local (short), STD (short-9), STD (short-8), and STD (short 6). At the bottom of the table is an 'Add a Calling Rule' button.

System Configuration
About Digium | Report a Bug | Help
[Activate Changes](#) [Logout](#)

Home Calling Rules
List of DialPlans:
DialPlan1 new delete Allow Parked Calls

Move the mouse over to a field to see tooltips

List of Calling Rules in the selected DialPlan

S.No	RuleName	Dial Pattern	Call Using	Options
1	1800	Begins with 1800 and followed by 6 digits	Custom - PSTN	Edit Delete
2	1300	Begins with 1300 and followed by 6 digits	Custom - PSTN	Edit Delete
3	13	Begins with 13 and followed by 4 digits	Custom - PSTN	Edit Delete
4	STD and Mobile	Begins with 0 and followed by 9 digits	Custom - Pennytel	Edit Delete
5	Local	Begins with 024 and followed by 7 digits	Custom - PSTN	Edit Delete
6	Local (short)	Begins with 4 and followed by 7 digits	Custom - PSTN	Edit Delete
7	STD (short-9)	Begins with 9 and followed by 7 digits	Custom - Pennytel	Edit Delete
8	STD (short-8)	Begins with 8 and followed by 7 digits	Custom - Pennytel	Edit Delete
9	STD (short 6)	Begins with 6 and followed by 7 digits	Custom - Pennytel	Edit Delete

Add a Calling Rule

Naked DSL

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Overview

Introduction

Softphones

ATAs

Asterisk

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Questions

Naked DSL = DSL without a PSTN service

- Still requires a physical copper line to the house
- No Telstra-PSTN service provided
- All calls VOIP only
- Can you live without a hardline ?

Thanks for listening to me babble, again

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Overview

Introduction

Softphones

ATAs

Asterisk

Final Word

Questions

- Questions ?