Asterisk Overview

Berkeley In Munich Tech Talks
17.01.2007

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Contents

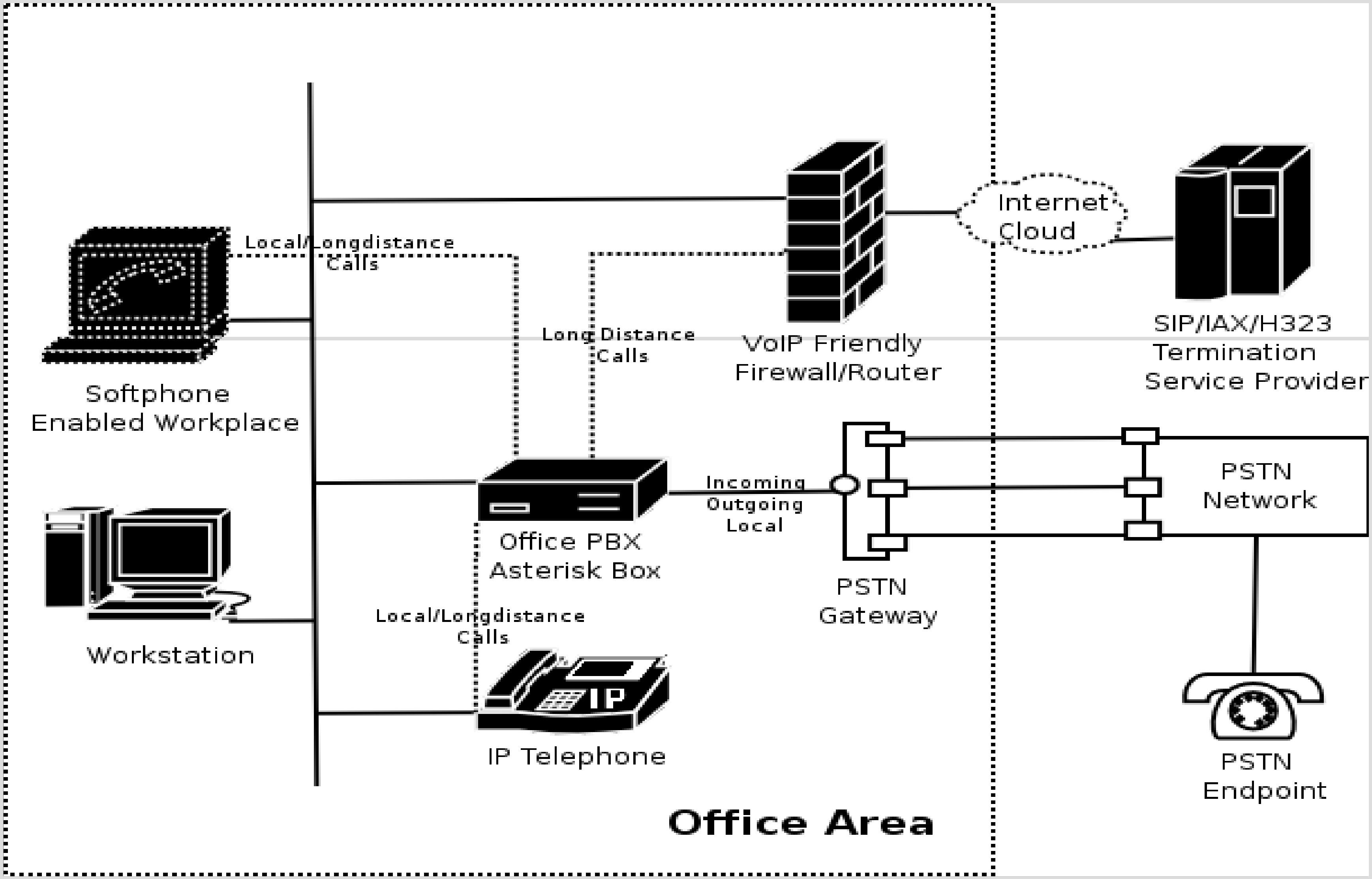
What is Asterisk?

- Private Branch eXchange (PBX)
  - Telephone Exchange for private offices
  - Focus on Voice Over IP (VoIP) Support
- An Interactive Voice Response Platform
  - Extensible with modules
  - Shipped with a set of applications enough to build modest but efficient middle-sized office systems
- Open Source Solution
  - GPL Licensed
  - Commercial License (embedded proprietary code)
  - Commercially supported by Digium Inc.

Usage Scenarios - 1

- Home VoIP PBX
  - Saves costs on long distance calls
  - Flexible answering machine
- Office VoIP PBX
  - Free voice communication between offices
  - Flexible on-line customer support (+ call queues)
  - Video & audio conferences with customers/partners
- VoIP Termination Services
  - Inet providers have advantage
  - Offer VoIP service/traffic to business customers
- VoIP Gateway
  - Asterisk is portable and scalable
  - Usable with appliances and gateways

Usage Scenarios - 2



Asterisk appliance in small offices

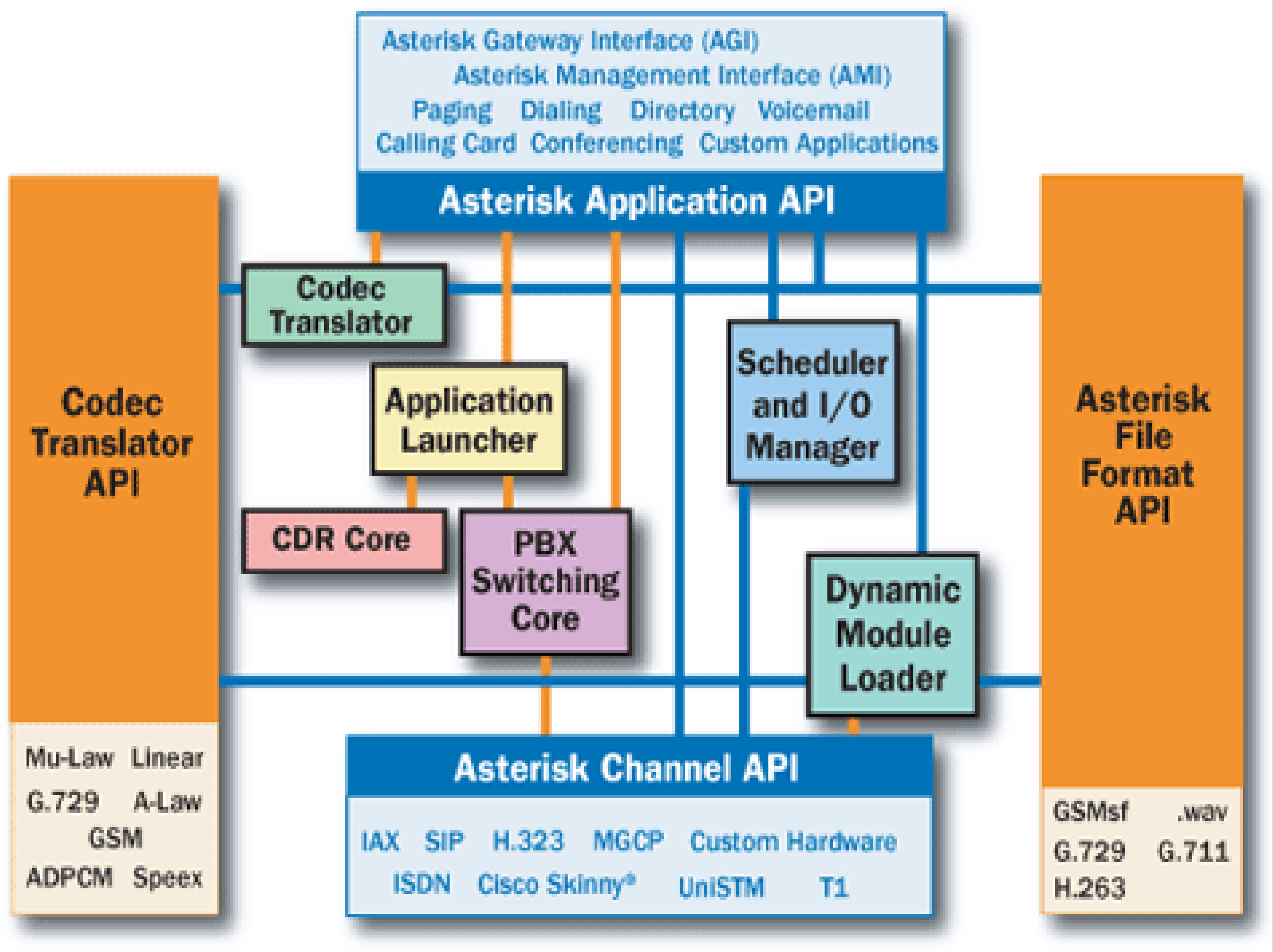
Characteristics

- Portable
  - Crossplatform programmed (pure C)
  - Initially written for Linux
  - Ported on other Unix-like OS
  - Version 1.4 compiles on FreeBSD without patches
- Performance
  - Written not with performance in mind as a priority
  - Behind some SIP-only solutions (SER, OpenSER) or dedicated commercial systems
- Considering Asterisk
  - Integrates different channels and hardware
  - Growing variety of supported hardware

Capabilities

- Protocols
  - H.323, SIP, MGCP, IAX2, XMPP, ISDN, etc..
- Codecs
  - Transcodable and passthrough-only
  - Audio G.711, GSM, iLBC, Speex, G.723, G.729, etc
  - Video H.263, H.263+, H.264 (MPEG4)
- Dialplans
  - Pattern-based matching/routing
  - Macros
  - Asterisk Extension Language (AEL)
  - EMUM (number mapping), DUNDI (dialplan sharing)
- Call Detail Record (CDR)
  - Support for CSV
  - ODBC, postgresql, mysql, FreeTDS

Architecture



Source: http://www.digium.com

Architecture - Core

- Codec Translator
  - Handles translation of media between peers
- Application Launcher
  - Responsible for executing internal * applications
- Scheduler & IO Manager
  - Handles IO and Threading operations
- Dynamic Module Loader
  - registers system and application modules
- CDR Core
  - Manages CDR resources and call statistic operations

Architecture - Modules

- Application Modules
  - Pluggable commands
  - Native way of application integration
- PBX Modules
  - Inner parts of Asterisk are defined as modules
- Codec Modules
  - Coding/decoding, payload processing
- Format Modules
  - Media format Handling
- Channel Modules
  - Protocols for Communication
  - Call state handling and representation
- CDR Modules
  - Resource drivers for CDR storage (odbc, mysql,etc)

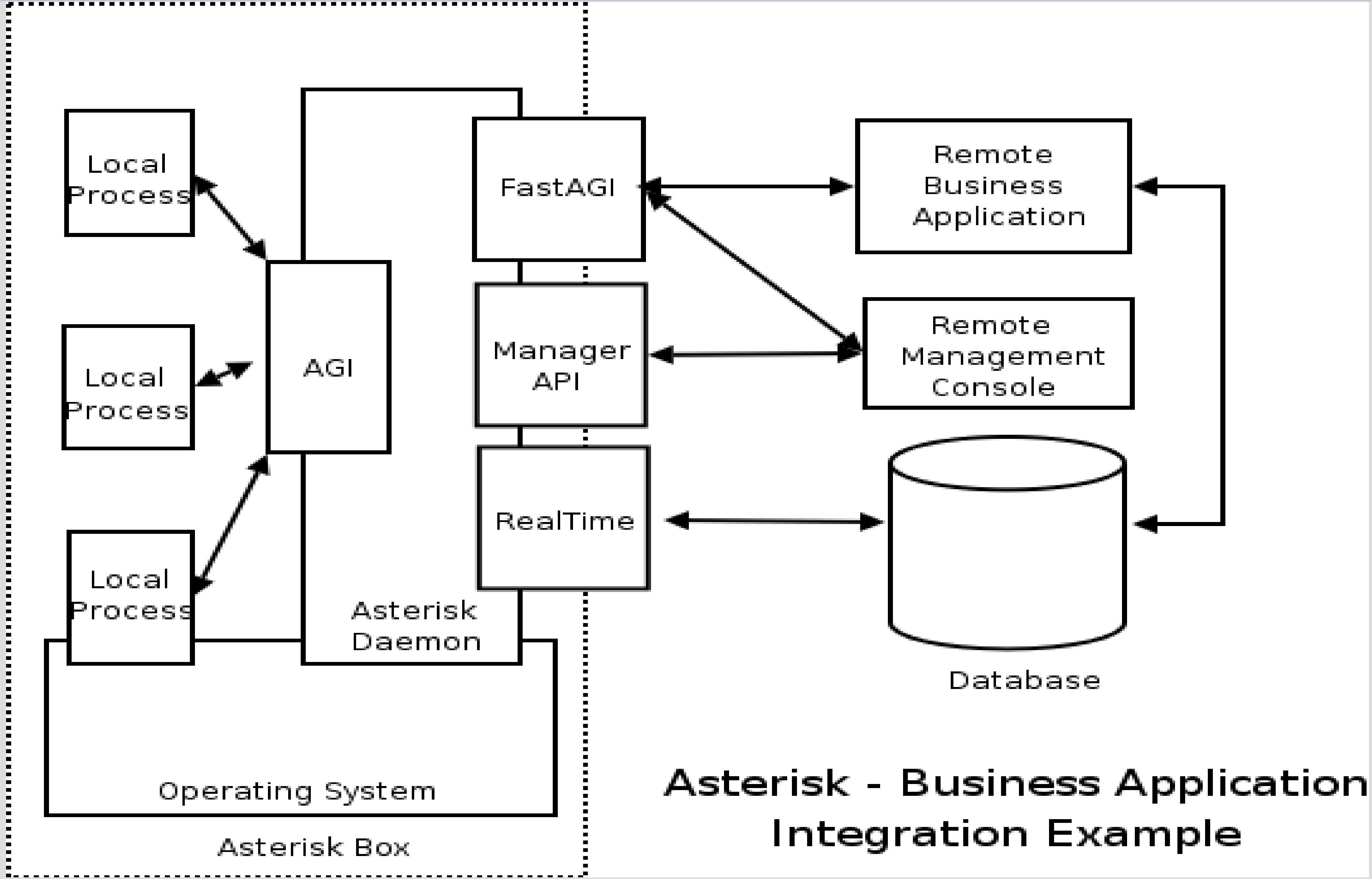
Architecture - Integration

- Asterisk RealTime
  - Configuration mapped onto abstract db schema
  - Database support for ODBC, MySQL, Postgresql
  - Makes billing and user management lot easier
  - Good way of integration with existing business apps (with custom driver)
  - Misses NAT Support
- Asterisk Gateway Interface (AGI)
  - Implemented as app_agi, allows execution of external programs that understand agi environment
  - Common practice – pass dialplan logic to scripts via AGI (supports Perl, Python, Ruby, Java)

Architecture - Remoting

- Asterisk Manager API
  - TCP based control interface
  - Call and channel state monitoring
  - Call origination, transfer, answer, etc.
  - Bindings for Perl, Python, Ruby, Java
- FastAGI
  - Inherits properties and functionality of AGI
  - Enables remote communication to business applications
  - Perl, Python,Java

Architecture - Example



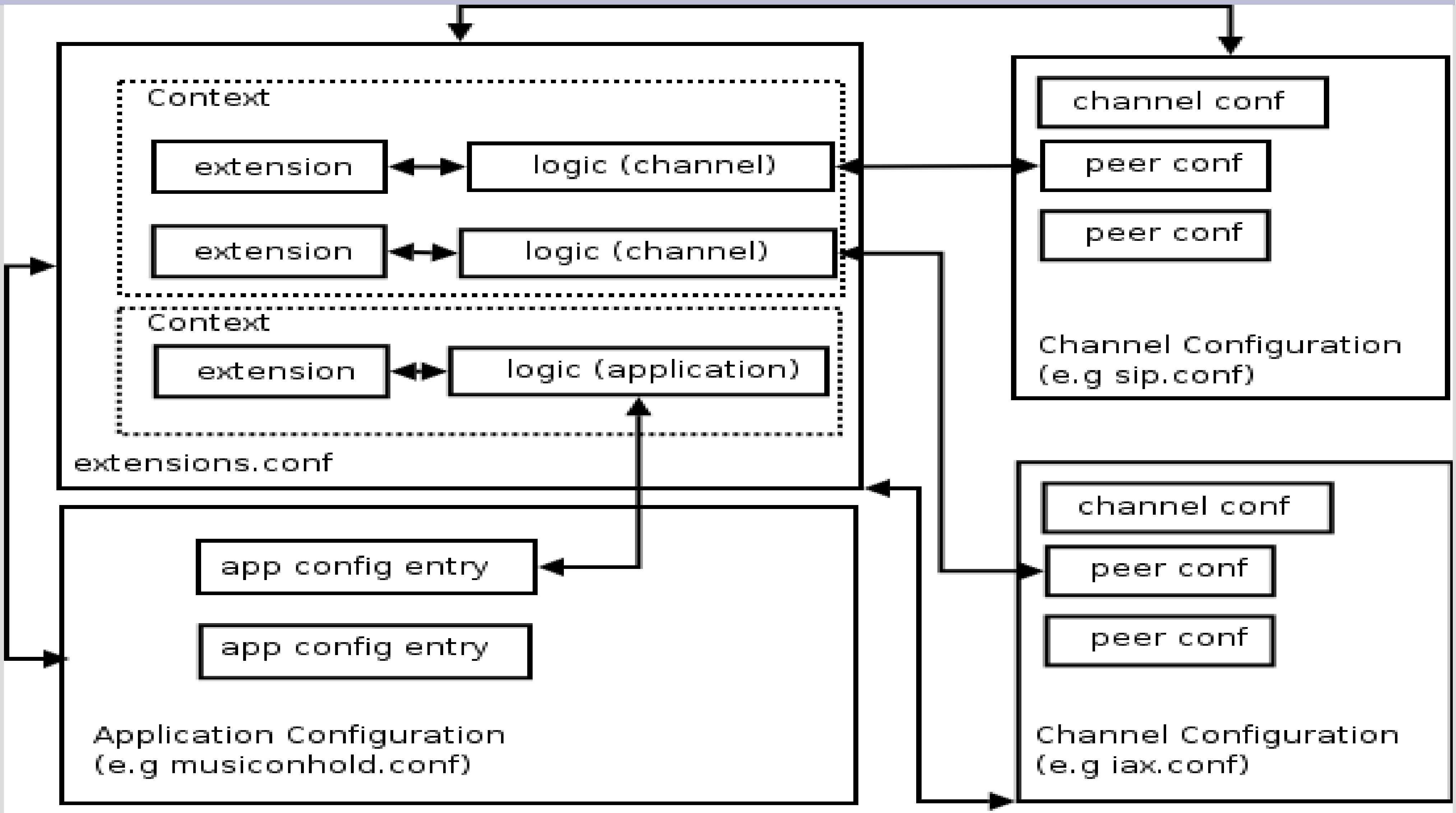
Configuration Basics

- Asterisk Configuration
  - modules.conf – enables/disables autoloading, prohibits loading
  - rtp.conf – specify port range for rtp media
  - musiconhold.conf – define music classes
- Peer Configuration
  - sip.conf – specify SIP peers (users, accounts, etc.)
  - iax.conf – specify IAX2 peers ..
- Extensions Configuration
  - extensions.conf – define dialplan (contexts, extensions)
  - AEL – flexible dialplan language
    - Loops, conditions, subroutines

Configuration Files

- extensions.conf
  - Extensions are grouped in contexts
  - Every peer is associated to a set of contexts
  - Extensions are mapped to a sequence of execution of applications or macros with a specified priority
  - Jumps across contexts and macros with Goto
- sip.conf, iax.conf, etc (any channel setup)
  - Channel setup
    - networking, ports, codecs, buffers, access control
  - Peer definition
    - Peer type
    - Authentication, credentials, per-peer codecs, channel behavior for peers,
- Concrete examples during the demo

Configuration Dependencies



Configuration Dependencies

Service Providers

- Proxying/Brokering
  - **FreeWorldDialup** (FWD Net - freeworlddialup.com)
    - One of the first FREE VoIP networks with SIP service
    - Toll-free numbers access, optional IAX access
  - **SIPBroker**
    - Free peering and extension mapping
    - Free ENUM service (e164.org, enum.org, etc)
  - **e164.org**
    - ENUM mapping of PSTN numbers to VoIP addresses
    - Works over DNS, highly scalable, free and open
  - Supports Email, ICQ, XMPP address records

Service Providers

- Termiantion – a rich variety of service providers, over 500 available according to SIPBroker:
  http://www.sipbroker.com/sipbroker/action/providerWhitePages
  - **MutualPhone.com**
    - US-Based
    - Cheap A-Z Termination, reported low delay
    - Only SIP with GSM, ilbc, ulaw, alaw, speex, g729
  - **VoipJET.com**
    - US-Based
    - Cheap A-Z Termination, Europe high delay, US low
    - IAX2 with GSM, ilbc, ulaw, alaw, speex, g729
  - German Terminators traditionally expensive (but in Germany low delay) – pepphone.de, sipgate.de, bluesip.de

Service Providers

- Dedicated In-Dial (DID) – become reachable wherever over the internet on a local call rate
  - **IPKall.com**
    - FREE US-Washington State area codes (360,206,253)
    - Free SIP Forwarding (for example to FWD)
    - Supports most codecs (GSM, A/ULAW, iLBC, G729)
  - **VoIPUser.co.uk**
    - UK-based VoIP community
    - FREE UK local, mobile and international rate DIDs
    - For every incoming call on the DID you get free minutes for international PSTN calls.
    - Free forwarding to any SIP/IAX address
  - Most mass-market terminators provide also DIDs for variety of international areas on monthly rate (good prices under 4 USD / month)

End-User Applications

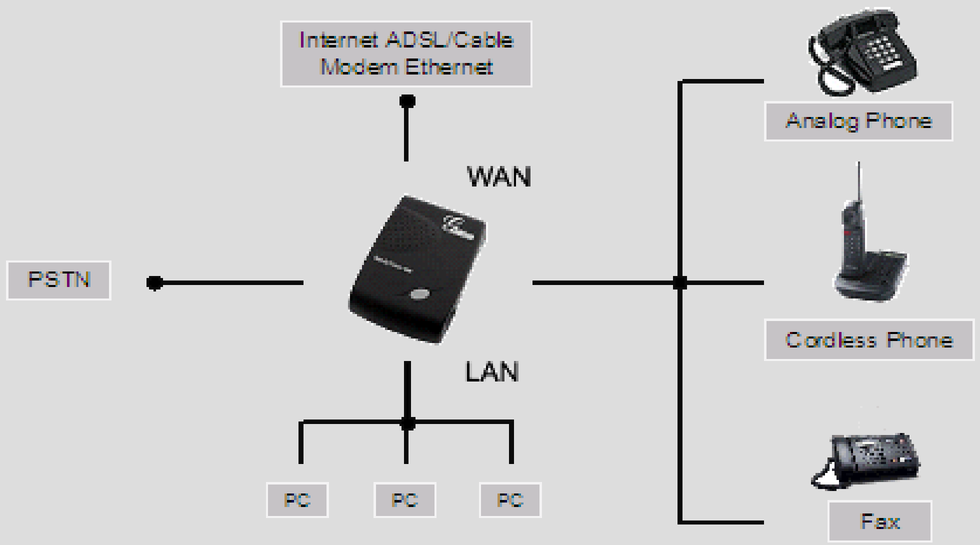
- SIP Softphones
  - **Linphone** – Unix/linux, GTK2 + libosip, no STUN support, no Video
  - **Kphone** – Unix/linux, KDE + libosip, no STUN, but support Video
  - **Ekiga** – Unix/linux, GTK2 + openh323 + custom SIP lib, video support, popular application on linux distros
  - **Eyebeam** (X-Lite) – Excellent Win32 support, bad linux/unix, Full STUN, ICE, X-tunnels. H263,H263+ video support, not freely rebrandable, scaled down free version
  - **SJPhone** – excellent Win32/WinCE(mobile), worse linux/unix support, STUN, rebrandable, adware

End-User Applications

- IAX2 Softphones
  - **IaxComm** – Basic UI, runs on win32,linux,mac. WxWindows + libiaxclient, Free Opensource, GPL
  - **Kiax** – Simple UI, runs on win32,linux,freebsd, libqt3 + cusomized libiaxclient, opensource GPL
  - **QtIAX** – Minimalistic UI, libqt3 + scaled down libiaxclient (misses some codecs)
  - **IDEFisk** – Fancy Interface, only Windows, libiaxclient, freeware
  - **MozIax** – XUL based, libiaxclient, windows/linux
  - Many custom apps – voipbuster client, Freshtel Firefly (extended IAX2), etc.
  - Asterisk's **dial app** + asterisk itself ;-) works with all channels of course!

End-User Devices

- Analog Telefone Adapters (ATA)
  - Keep your old phone while placing and getting calls selectively over PSTN or VoIP
  - Relatively cheap (from 30-60 EUR) depending on number of ports
  - Large variety, check out voip-info.org



Grandstream ATA 486

End-User Devices

- VoIP Hardware Phones
  - Looks like phone (with cord), acts like phone
  - Usually High quality and reliability
  - Recently Video phones under 250 EUR!

Grandstream GXV3000

H264 Support!

References

- **asterisk.org** – download and test tha Asterisk PBX
- **voip-info.org** – the ultimate VoIP guide (redundant sections/categories but very useful)
- **asteriskdocs.org** – an asterisk documentation project
- **digium.com** – hardware, asterisk support
- **siproker.com** – peer your asterisk with other VoIP networks
- **ipkall.com** – get US DIDs
- **voipuser.co.uk** (.org) – get UK DIDs
- e164.org – register your PSTN numbers and get called for free over the net

Summary

- Flexible PBX with strong community and support
- Saves money in small and middle-sized offices
- Large variety of service providers, end user applications and devices
- Modular approach for better customization and integration
- Language bindings for AGI and Manager Interfaces - not difficult to write your own business logic

The End

Thank you for the attention!

Asterisk Demo

- IAX Client (Kiax) interaction with Asterisk
  - EchoTest
  - DTMF
  - Call Answering
  - etc.