Advanced OpenSER Course

Voice System SRL

http://www.voice-system.ro

http://www.openser.org

OpenSER & Asterisk Integration

Open Source Media Servers

- Asterisk
- CallWeaver
- Freeswitch
- Yate
- Sems

- An Open Source Modular Multiprotocol PBXa
- Asterisk delivers services on the SIP network
  - Voicemail
  - PSTN gateway
  - Conference
  - Announcement services
- multiple protocols: IAX2, SIP, h323, Skinny, zap, jingle
- multiple codecs: gsm, g711, g729, ilbc, ...
- transcoding
- gateway to PSTN
- protocol translation
- back-to-back user agent

- announcement service
  - play message
    - news
    - clock time
    - date
    - horoscope
    - wake-up call
  - play message within a call
  - play message in early media
  - record message
  - record call

- **voice mail**
- used to leave a message if no one is answering your call
- configuration in Asterisk is in /etc/asterisk
  - voicemail.conf.
- ability to store it on the file system or in database
- pin authentication
- voicemail menu
- configurable welcome message
- send recorded message by email
- listen to recorded message by dialing specific extension
- multi-language support

- voicemail menu – you have to learn it by hart (I will be questioning at the end ... :-))
  - 1 Old Messages
    - 3 Advanced options
      - 1 Send reply
      - 2 Call back
      - 3 Envelope
      - 4 Outgoing call
      - 5 Leave message
      - * Return to main menu
    - 4 Play previous message
    - 5 Repeat current message
    - 6 Play next message
    - 7 Delete current message
    - 8 Forward message to another mailbox
    - 9 Save message in a folder

voicemail menu

- * Help; during msg playback: Rewind
- # Exit; during msg playback: Fastforward
- 2 Change folders
- 3 Advanced options
    - 0 Mailbox options
    - 1 Record your unavailable message
    - 2 Record your busy message
    - 3 Record your name
    - 4 Change your password
    - * Return to the main menu
- * Help
- # Exit

- audio conferencing
  - media stream mixage
  - meetme application
  - admin mode
  - user mode
  - mute/unmute participants
  - record welcome message
  - music on hold
  - storage of conference details in database

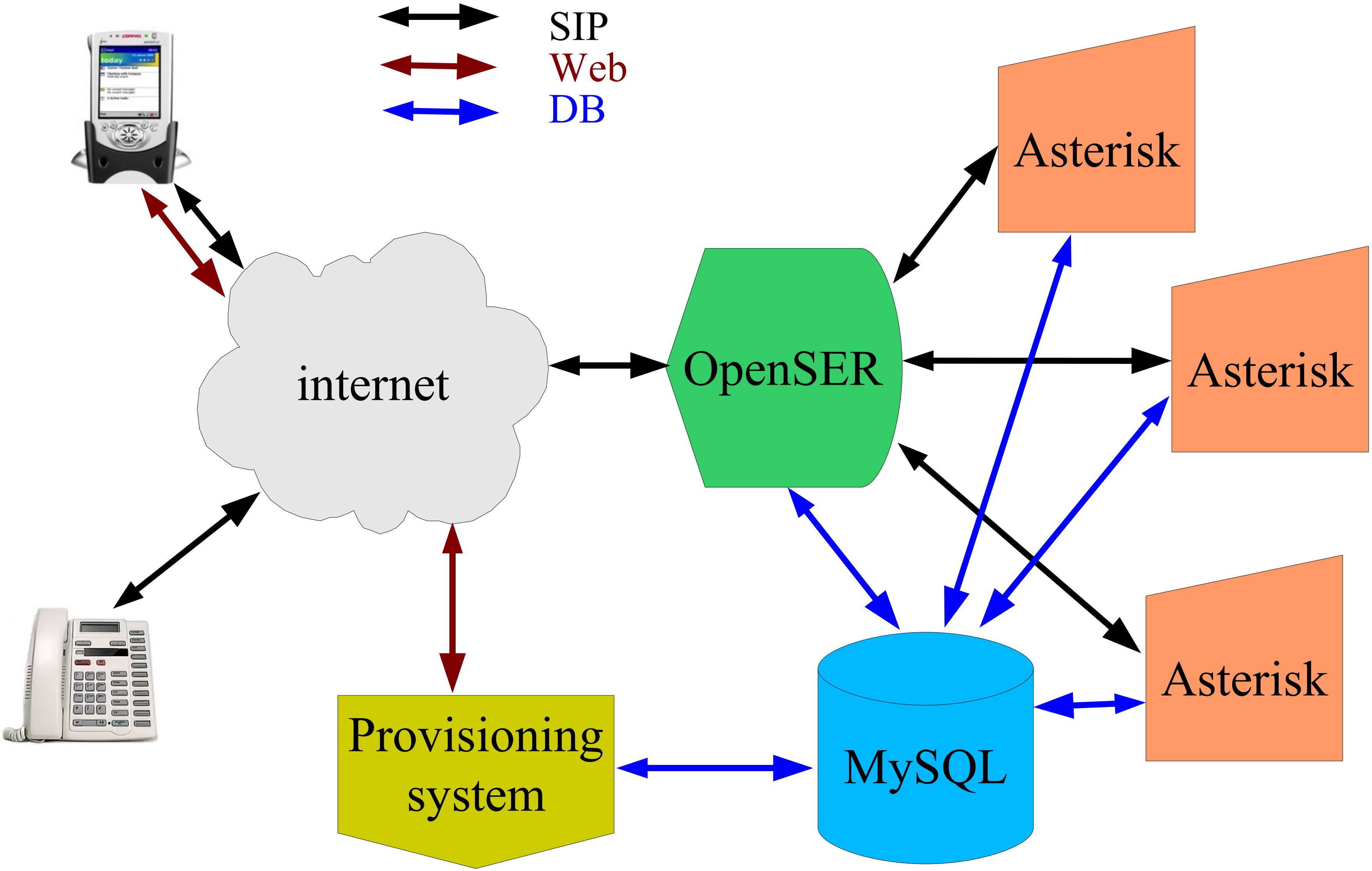
- two stage dialing
  - answer the call and wait for extension
  - ask for authentication via IVR
  - security reasons
  - reducing costs
- auto-attendant
  - selection menu
    - secretary
    - sales
    - distribution
  - call queuing
  - music on hole
  - call recording

- usage of database to share user accounts
- usage of database to store resources (voicemail messages, audio conferences)
- software
  - openser
  - asterisk
  - zaptel drivers
  - unixodbc
  - mysql

- for voicemail
  - create a view over subscriber table of openser to have the voicebox profile
  - create another view over subscriber table of openser to have peers/sip users profile for MWI
- for announcement nothing is required
- for meetme/audio conferencing nothing special is required, eventually, just the provisioning system has to be extended to assign rooms to SIP users for self-management of room access
- for other services, developing an AGI in Perl/Php/etc. should be trivial to get access to openser database

- all these induct a real-time reflexion of changes in openser database to asterisk behavior

- by using database, it si possible to have multiple Asterisk instances connecting to same database to read/write voicemail messages
  - easy load balancing
  - easy replication and failover
  - easy backup and system restore
- keeping user profiles in plain text files for media server in large deployments bring a nightmare of synchronization and reliability
- easy to migrate from SQL views to stand alone tables
- easy to partition the users in views to speed up database access, by just adding a “where” condition in creating the view
- easy to provision from web/gui applications
- ability to use clusters


SIP
Web
DB
internet
OpenSER
Asterisk
Asterisk
Asterisk
MySQL
Provisioning system

Issues

- Asterisk inability to handle SIP spirals
  - cannot return a call to same Asterisk
  - different instances for b2bua and end-point behaviors
- missing “domain” concept in Asterisk
  - it is a multi-protocol PBX, and domain is not common to all
- missing “user alias” concept

- an example of integration
- voicemail
- audio conferencing
- two stage dialing
- announcement services
- echo application

- designing dialing plan
  - do not overlap with user ID
- recommended to use internal mappings, that are not visible outside of openser-asterisk ecosystem
  - some security by “obscurity” in asterisk side
  - in multi-national environments, different public extensions map to same media service
  - for better security, asterisk should communicate with openser only on the signaling channel
- the control is done in openser only
  - asterisk as endpoint or b2bua

- **OpenSER**
- listen voicemail: *188
- leave voice message: *187...
- listen current time: *177
- listen current date: *178
- echo application: *179
- audio conferencing: *122...
- two stage dialing: *144

- **Asterisk**
- voicemailmain: 88
- voicemail: 87
- sayunixtime: 77
- sayunixtime: 77
- echo: 79
- meetme: 22
- read/dial: 44

- Notes:
  - *1 signals a media service
  - digits 2 and 3 map to asterisk extensions

walk through openser-asterisk configuration files

BREAK