Asterisk & ENUM

Extending the Open Source PBX

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Why a ENUM-enable a PBX?

- your PBX „doubles“ as an IP/PSTN gateway – for your existing numbers
    - becomes a „dual contact point“
    - NAPTR record „attracts“ traffic on IP just like MX
- <u>no operator buy-in required</u>
- unilateral decision to deploy
    - default is standard PBX behaviour
    - if another IP PBX is visible, call drops to IP

What is Asterisk?

- A PBX software for the Linux platform developed by Digium.
- Does PBX call switching, Codec translations, and various Applications.
- Available for free in source code under the GNU Public Licence.
- nic.at funded Digium to implement ENUM in call processesing.
- See www.asterisk.org

Voice Interfaces (1)

- PRI (E1/T1)
  - With cards sold by Digium
  - Can be used to drive channel-banks
- ISDN BRI
  - ISDN4Linux or CAPI
- Analog lines
  - FXO and FXS
  - PCI and USB versions available from Digium
- Linux Soundcard

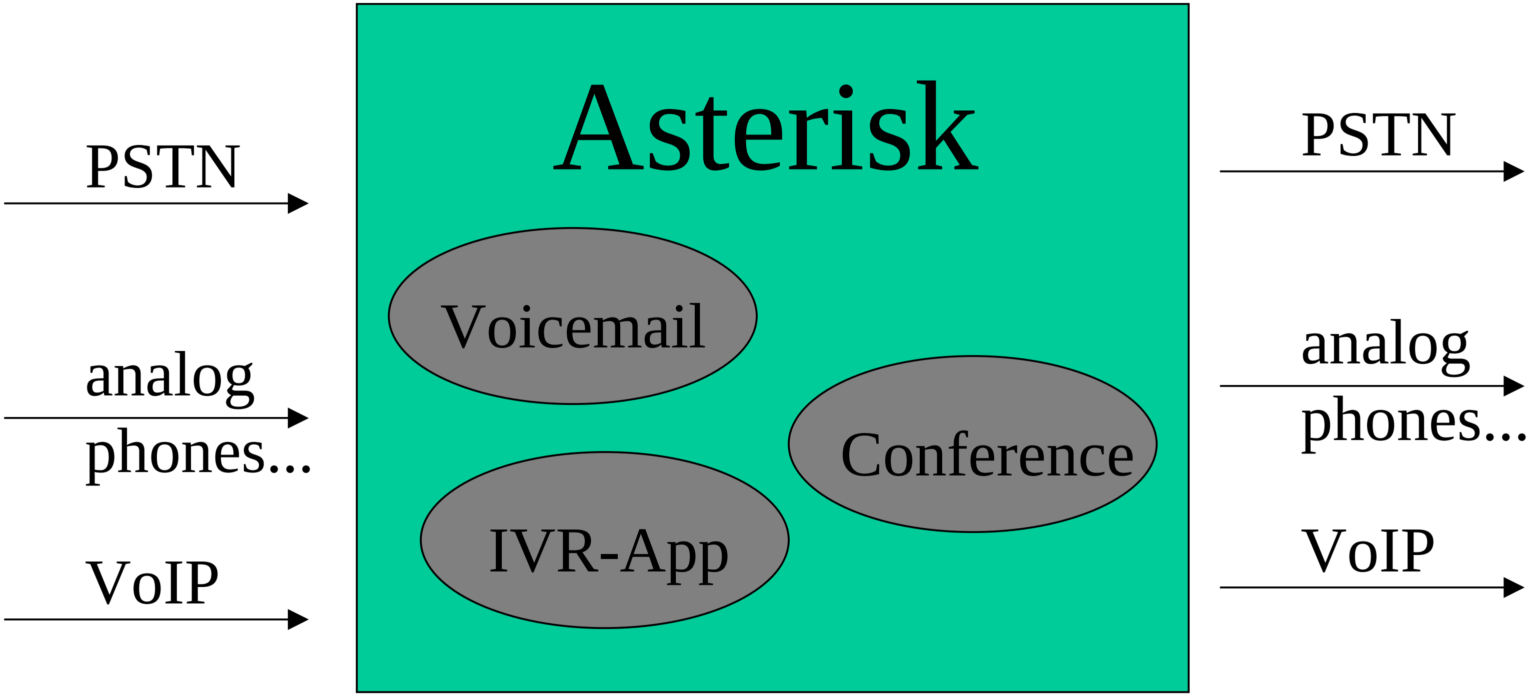
Voice Interfaces (2)

- SIP
  - Includes codecs for G.711(a, µ), ILBC, GSM
- H.323
  - Utilizes OpenH323 code
- IAX
  - Inter-Asterisk-eXchange
    - proprietary; TLS & X.509 certficates for signaling
- MGCP

Applications

- Voicemail
- Conference Bridge
- ACD Queues (Automatic Call Distribution)
- IVR Applications ("press x for Sales")
- File Playback
- Scripting using "extension.conf" for simple Applications
  - Can do Database operations
  - Can do ENUM lookups
- CGI-like interfaces for advanced programming

Overview



PSTN
analog
phones...
VoIP
Asterisk
Voicemail
Conference
IVR-App
PSTN
analog
phones...
VoIP

Call Routing

- Asterisk implements a **State Machine** which is defined in terms of
  - The origin of the call (Which SIP user? PSTN? Anonymous SIP? Local POTS?)
    - **= CONTEXT**
  - The number dialed by the user (or Direct Dial In, or SIP URI)
    - = **EXTENSION**
  - A "Program Counter" which orders sequences of commands (like line numbers in BASIC)
    - **= PRIORITY**

State Machine Example (1)

- Make "80" in *context* call the Echo Application.

```
[context]
; Let them know what's going on
exten => 80,1,Playback(demo-echotest)
exten => 80,2,Echo                          ; Do the echo test
exten => 80,3,Playback(demo-echodone) ; Let them know it's over
exten => 80,4,Hangup                        ; End the call
```

State Machine Example (2)

- Map extension "200" to a analog extension port with fallback to Voicemail:
- zapata.conf

```
[channels]
context=localuser
signalling=fxs_ls
channel => 1
```

- extensions.conf

```
exten => 200,1,Dial(Zap/1,30)          ; ring for 30 secs
exten => 200,2,Voicemail(u200)         ; if not answered
exten => 200,3,Hangup
exten => 200,102,Voicemail(b200)       ; if busy
exten => 200,103,Hangup
```

Using a SIP phone

- sip.conf

```
[mylogin]
type=friend
context=authorized      ; in which context start calls from that phone?
username=mylogin        ; Authentication info
secret=no1knows
callerid=300            ; Set the callerID for this phone
host=dynamic            ; Dynamic Address: wait for it to REGISTER
```

- extensions.conf

```
exten => 300,1,Dial(SIP/mylogin,30)
...plus voicemail & co ...
```

Using an external SIP service like FWD

- extensions.conf

  exten => 301,1,Dial(SIP/19343@fwd.pulver.com,30)

  ...plus voicemail & co ...

extension.conf Syntax

- Extension rule for a specific context follow after a [*contextname*] line. (cf. .ini files) and have the form

  exten => *pattern,priority,command*

- *pattern:*
  - 12345 ; a fixed string
  - _[1-4]XX. ; a regular expression
  - s ; "start": match the empty extension
  - i ; "invalid": a default entry
  - t ; "timeout"

nic.at Asterisk Demo

- Connected via a PRI to the Vienna PSTN.
- Configured to act as SIP server for local soft- and hardphones.
- Accepts anonymous SIP calls to configured extensions.
- Authorized users can call out via SIP and the PSTN.

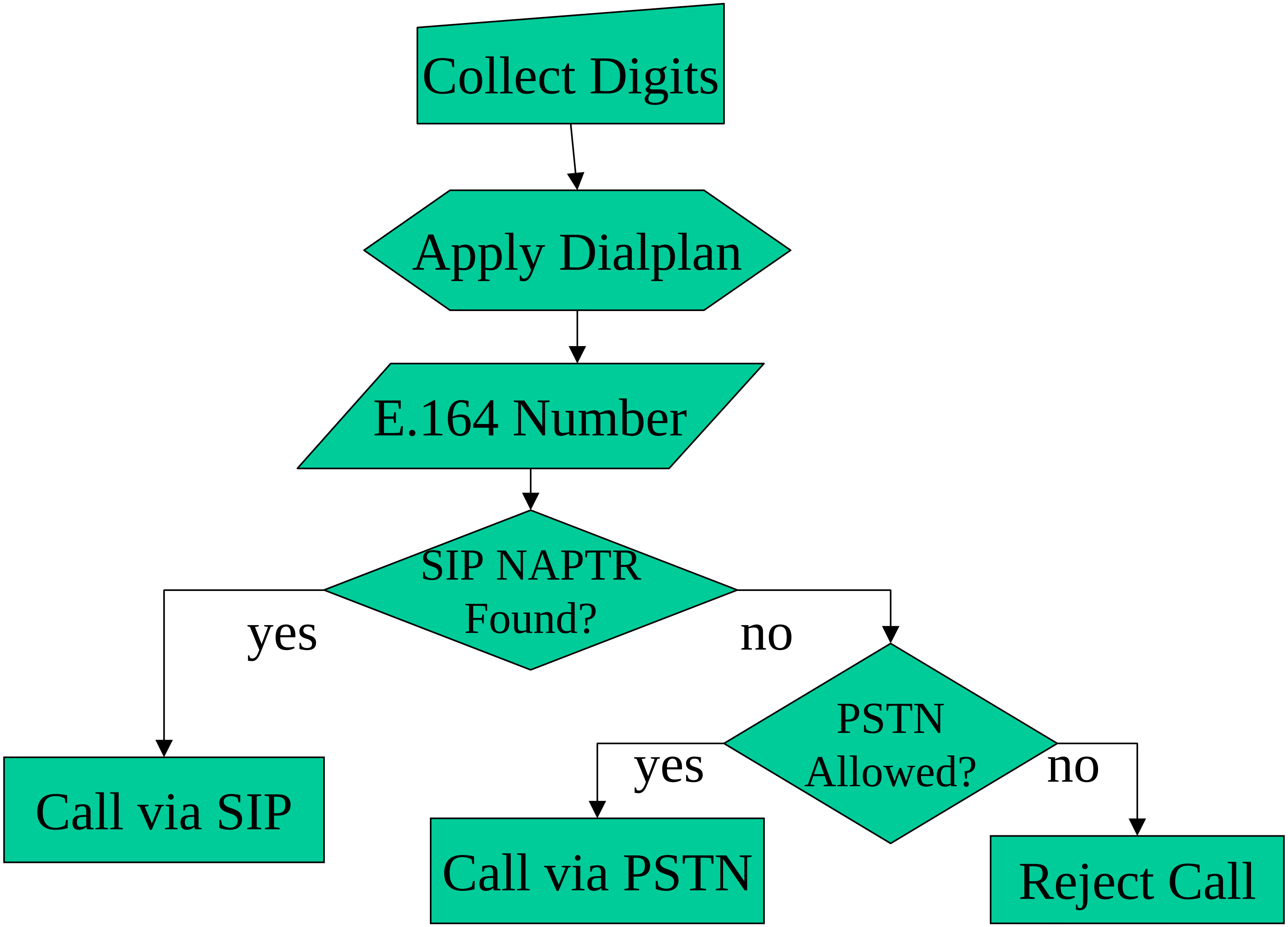
Dialing Plan

- Asterisk is configured according to the standard Austrian PBX dialing plan.
- Numbers <u>**not**</u> starting with '0' are considered local extensions.
- <u>One leading '0' signifies a local call within the Vienna calling area.</u>
- 00xxxyyyy is a call to area code xxx.
- <u>000zz... corresponds to +zz...</u>

ENUM lookups

1. The dialed number is converted to an E.164 number (if it's not a local extension):
   - 0xyz... --> +43 1 xyz...
   - 00abc... --> +43 abc...
   - 000def... --> +def...
2. The e164.arpa tree is searched for a NAPTR record with a SIP service entry
3. If found: Send a SIP INVITE to this address
4. If not found and the user is authorized: Call using the PSTN

Asterisk Call Logic



ENUM for local Calls

```
[globals]
TRUNK=Zap/g2        ; This will be our link to the PSTN
[fullaccess]
exten => _0[1-9]XXX.,1,BackGround(nic.at/enum-doing)
exten => _0[1-9]XXX.,2,EnumLookup(431${EXTEN:1})
; ${EXTEN:1} is the number dialed by user with the leading 0 stripped.
; Thus "431${EXTEN:1}" is the E.164 number.
; EnumLookup sets ${ENUM} on success. On failure jumps to priority+101.
exten => _0[1-9]XXX.,3,BackGround(nic.at/enum-successful)
exten => _0[1-9]XXX.,4,Dial(${ENUM},30)
exten => _0[1-9]XXX.,5,Goto(104)        ; No answer on SIP, fallback to PSTN
exten => _0[1-9]XXX.,103,BackGround(nic.at/enum-failed)
exten => _0[1-9]XXX.,104,Dial,${TRUNK}/${EXTEN:1}
; our trunk in inside the Vienna dialing plan: thus just strip the 0.
```

No PSTN permission?

- Calls from the PSTN or anonymous SIP calls should be in a context like this:

```
[nopstn]
exten => _0[1-9]XXX.,1,BackGround(nic.at/enum-doing)
exten => _0[1-9]XXX.,2,EnumLookup(431${EXTEN:1})
exten => _0[1-9]XXX.,3,BackGround(nic.at/enum-successful)
exten => _0[1-9]XXX.,4,Dial(${ENUM},30)
exten => _0[1-9]XXX.,5,Goto(104)
exten => _0[1-9]XXX.,103,BackGround(nic.at/not-allowed)
exten => _0[1-9]XXX.,104,Hangup
```

Handling tel: Records

- EnumLookup jumps to
  - *extension+1* on encountering SIP URIs (${ENUM} will be set to "SIP/user@domain")
  - *extension+51* for tel: URIs (${ENUM} is set to the E.164 number without the leading '+'.)
  - *Extension+101* on no matching NAPTR
- EnumLookup does currently **not** handle multiple NAPTR records.
- tel: URIs are dangerous as they can point to expensive 0900xxx numbers

International calls + tel:

```
[fullaccess]
exten => _000[1-9]XXXXX.,1,BackGround(nic.at/enum-doing)
exten => _000[1-9]XXXXX.,2,EnumLookup(${EXTEN:3})
exten => _000[1-9]XXXXX.,3,BackGround(nic.at/enum-successful)
exten => _000[1-9]XXXXX.,4,Dial(${ENUM},30)
exten => _000[1-9]XXXXX.,5,Hangup
exten => _000[1-9]XXXXX.,53,BackGround(nic.at/enum-successful)
exten => _000[1-9]XXXXX.,54,Dial,${TRUNK}/00${ENUM}
exten => _000[1-9]XXXXX.,55, Hangup
exten => _000[1-9]XXXXX.,103,BackGround(nic.at/enum-failed)
exten => _000[1-9]XXXXX.,105,Dial,${TRUNK}/${EXTEN:1}
exten => _000[1-9]XXXXX.,106,Hangup
```

Asterisk Usage Scenario

- For a small company:
  - PBX with local phones attached either as IP-Phones or via POTS cards / channel-banks
  - Voicemail system
  - IVR and ACD
  - Teleworker integration with SIP phones
  - Outgoing calls routed via PSTN
  - Outgoing least-cost routing with ENUM
  - VoIP & ENUM educational vehicle
  - ENUM trial vehicle