Answering Machine Detection and Call Progress Analysis for Asterisk-Based Call Centers

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What is Call Progress Analysis?

- From Asterisk Dial, to Answer or Hangup/Cancel
- Detects the following pre-Answer events that Asterisk cannot:
  - Disconnect
  - Carrier Congestion
  - Other SIT(Special Information Tones)
- Sometimes referred to as CPD(Call Progress Detection)

What is Answering Machine Detection?

- Detects Human Answer or Answering Machine based upon patterns in audio stream AFTER receiving an Answer signal from the carrier
- Other post-Answer detection includes Fax machine detection

Why use CPA / AMD in a Call Center?

- More efficient use of agent time, less waiting for a live customer
- Leaving full recorded messages on customer Answering Machines
- Cleaning calling lists of disconnected and changed phone numbers
- Avoiding fax machines, or redirect them to fax responder
- Lower dropped call rates, important for regulatory compliance

Asterisk Built-In and Free Add-On Options

- app_amd – Contributed by Aheeva in 2005 at Astricon Europe in Madrid, Spain
  - Accuracy: 70-80% at best (at 3+ seconds of analysis time)
  - Built-in to Asterisk since 1.4.X
- “nv”detect – Contributed by Newman Telecom
  - Not supported beyond Asterisk 1.4.X
- Built-in Fax detection added to Asterisk 1.6.2
- Detection cannot start until after Answer signal

Commercial Add-Ons for CPA / AMD

- Sangoma Netborder Call Analyzer
  - Asterisk support added in 2008
  - Works as a SIP Proxy
- Dialogic Diastar Perfect Call
  - Asterisk support added in 2010
  - Works as SIP Proxy and with chan_woomera
- Lumenvox Call Progress Analysis
  - New offering in 2012
  - Works as add-on to existing speech platform

Sangoma Netborder Call Analyzer

- Operates as a SIP Proxy server
- Optimal limit at 200 channels per proxy for low-end single quad-core CPU server
- Configurable parameters set at start-up time
- No easy licenses-in-use interface
- Licenses only in use while analysis is happening at the beginning of the call
- Based upon a speech recognition model
- Also called “AMD for Asterisk”

SANGOMA

http://sangoma.com/products/call_center_software/netborder_call_analyzer.html

Dialogic Diastar Perfect Call

- Operates through chan_woomera in Asterisk
- Peak capacity at 400 channels per server for a high-end server
- Configurable parameters set at start-up time
- Licenses-in-use can be seen through CLI interface
- Licenses are in use for the duration of each call
- Requires Dialogic SIP licenses as well as CPA licenses to use

http://www.dialogic.com/Solutions/Open-Source/DiaStar-Server.aspx

Lumenvox CPA

- Operates through the Asterisk res_speech module using the SpeechBackground app
- Peak capacity above 500 channels on high-end server
- Configurable parameters can be set per call
- Licenses in use can be seen through Windows GUI app or C++ API
- Licenses are in use only during analysis
- Only post-Answer tested with Asterisk

http://lumenvox.com/products/speech_engine/cpa.aspx

CPA Pricing

- CPA software is all priced per channel. Sangoma and Lumenvox licenses are only in use while the analysis is active at the beginning of the call
- Sangoma licenses are more expensive, but you need less of them because of how their licensing works, Lumenvox is the lowest list price option, but it is also very new
- Lumenvox and Sangoma offer per-month licensing purchase options.

CPA Integration With Asterisk

- CPA software is often best installed on separate proxy server, although it is possible to install directly on the Asterisk server
- Your software will parse Asterisk Manager Interface API events or channel variables to gather results from analysis
- Sangoma and Dialogic integration requires patching Asterisk, to convert SIP messages to AMI output

Tips for CPA Use

- The longer the analysis time, the better chance of a result, instead of a hangup
  - For messaging campaigns: 100 seconds
- Carrier post-dial-delay can cause problems
- Monitor your channel usage, running over your concurrent limit will mean overage calls aren't going out
- Monitor server CPU load, overloaded servers can lead to more delay in calls getting to agents

Tips for AMD Use

- The longer the analysis time, the better the accuracy
- The longer the analysis time, the more likely the customer will hang up
- Loud background noise on the customer end will cause problems for all AMD engines
- To leave answering machine messages at the right time, use either Asterisk WaitForSilence() app or build-in options from commercial products

Asterisk Built-In AMD Example

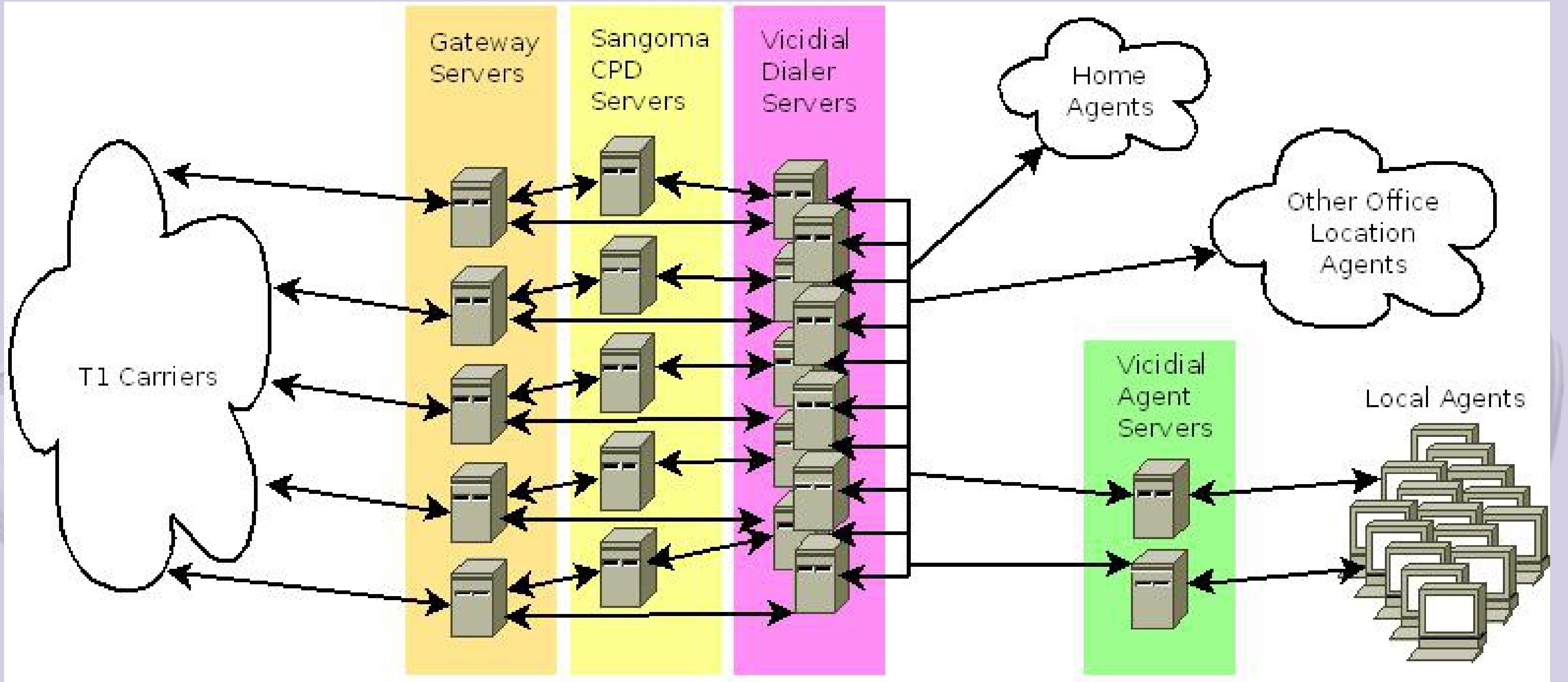
- Dialplan example(Asterisk 1.4) for AMD use with WaitForSilence:

```
[outbound]
exten => s,1,Background(sip-silence)
exten => s,n,AMD(2000|2000|1000|5000|120|50|4|256)
exten => s,n,GotoIf($[${AMDSTATUS}=HUMAN]?humn:mach)
exten => s,n(mach),WaitForSilence(2000|2|30)
exten => s,n,Playback(message-when-machine)
exten => s,n,Hangup
exten => s,n(humn),WaitForSilence(500)
exten => s,n,Playback(message-when-human)
exten => s,n,Hangup
```

- For better results, use a Background(short-silence-audio-file) before running AMD

Call Flow Example

Calls going from Vicidial dialers, through Sangoma CPA to gateway servers with T1s



Thank you!

For more information, go to:

www.vicidial.org