voxbone

- Asterisk Configuration, past present, and future

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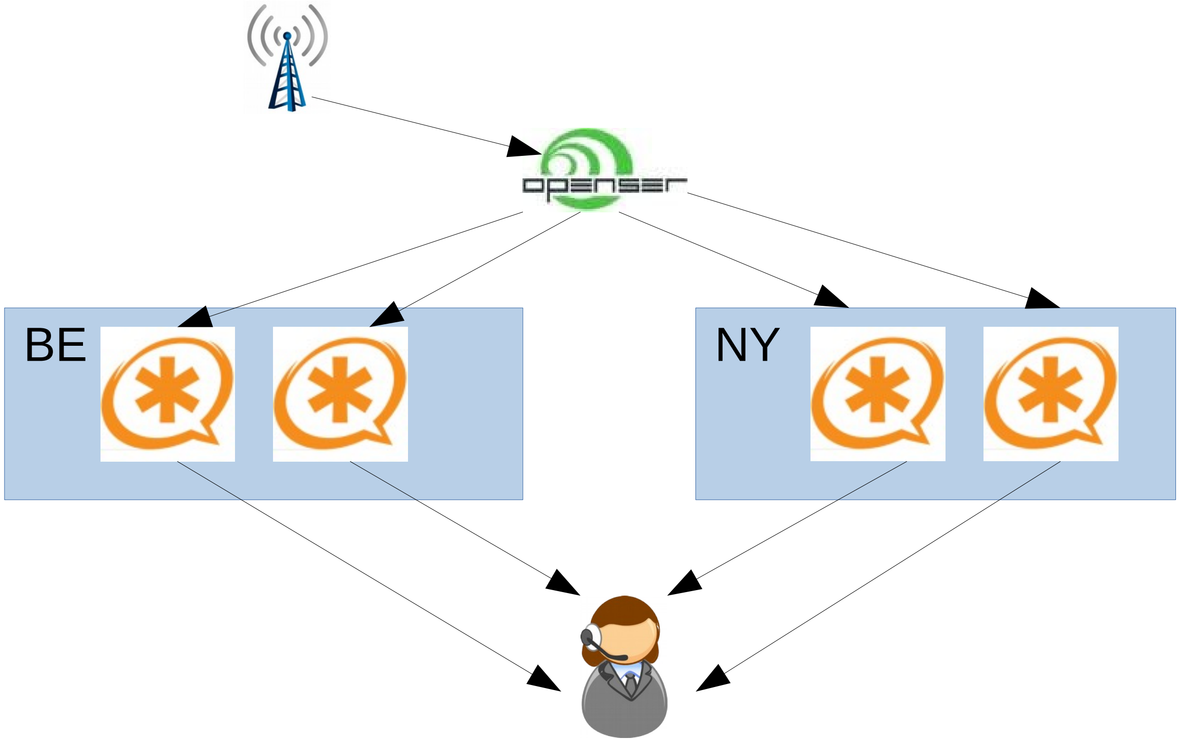
Voxbone - 2005

- 2 POPS – 1 SER
- Asterisk 1.2
- SER the only peer provisioned in sip.conf
- Customers SIP or IAX2 dialed directly by Asterisk using the Default context


BE
NY

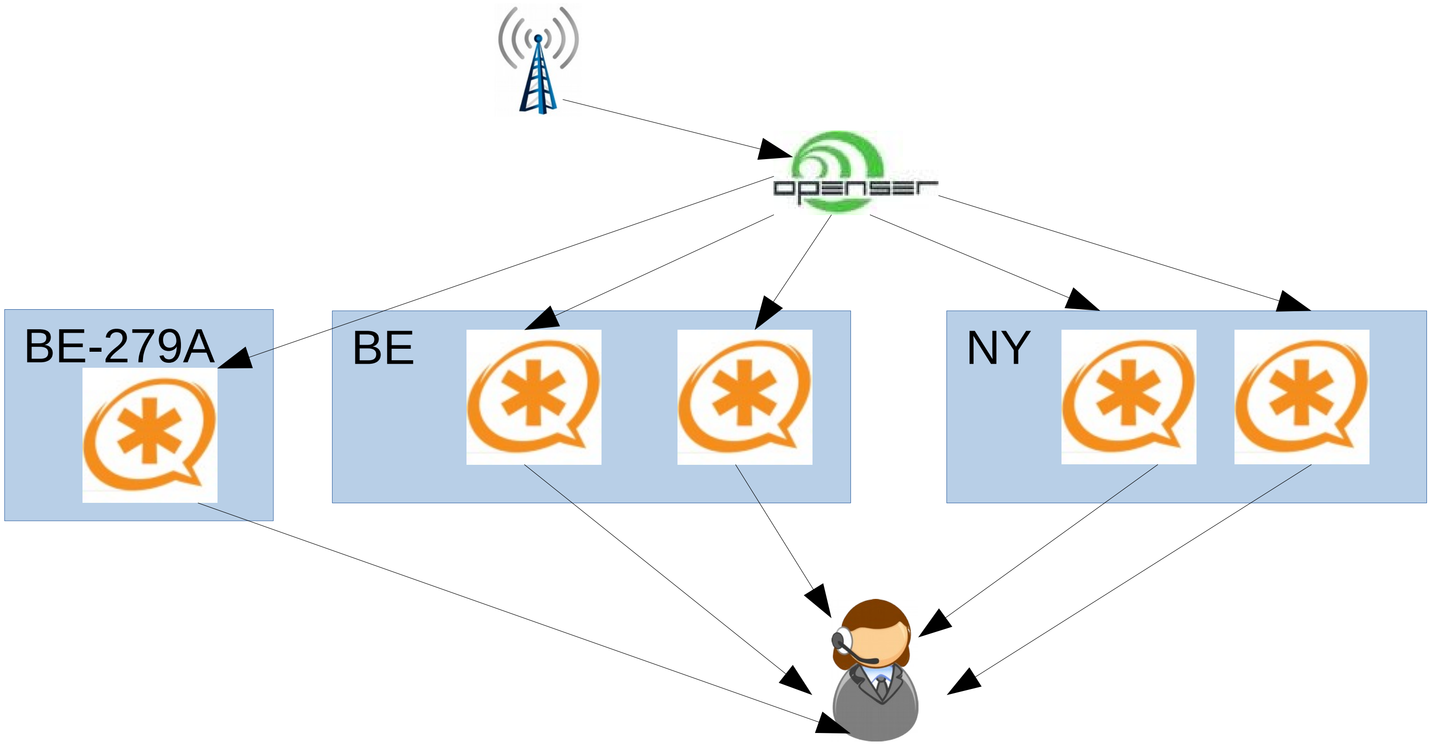
Issues

- All Upstream Carriers appear as 1 peer
- USA likes µLaw
- chan_sip ignores codec preference of invite and always sends alaw prefered in 200 OK
- Asymmetric codec negotiation causes CISCO equipment to have 1 way audio
- Solution: SIP_CODEC variable

Issues

- Dial used Default context – What if customer wants G.729A forced?

- Solution: Special Asterisk Box



Voxbone 2007 - 2008

- I get Hired (Yay!)
- Killed IAX2 (Sorry guys)
- 3 Pops – 1 SER
- Upgraded to Asterisk 1.4
- SIP SBC developed and placed between us and our customers (1 per pop)

Kill 'custom asterisks'

- Service ip address of the SBC was put in the hosts file
- Each configuration permutation is made a different peer in the database pointing to the sbc
- Instead of Dialling a customer, the Dial() command just specifies a peer matching the desired configuration
- Customers can now control their codec and DTMF options on the customer portal

New sip.conf peers

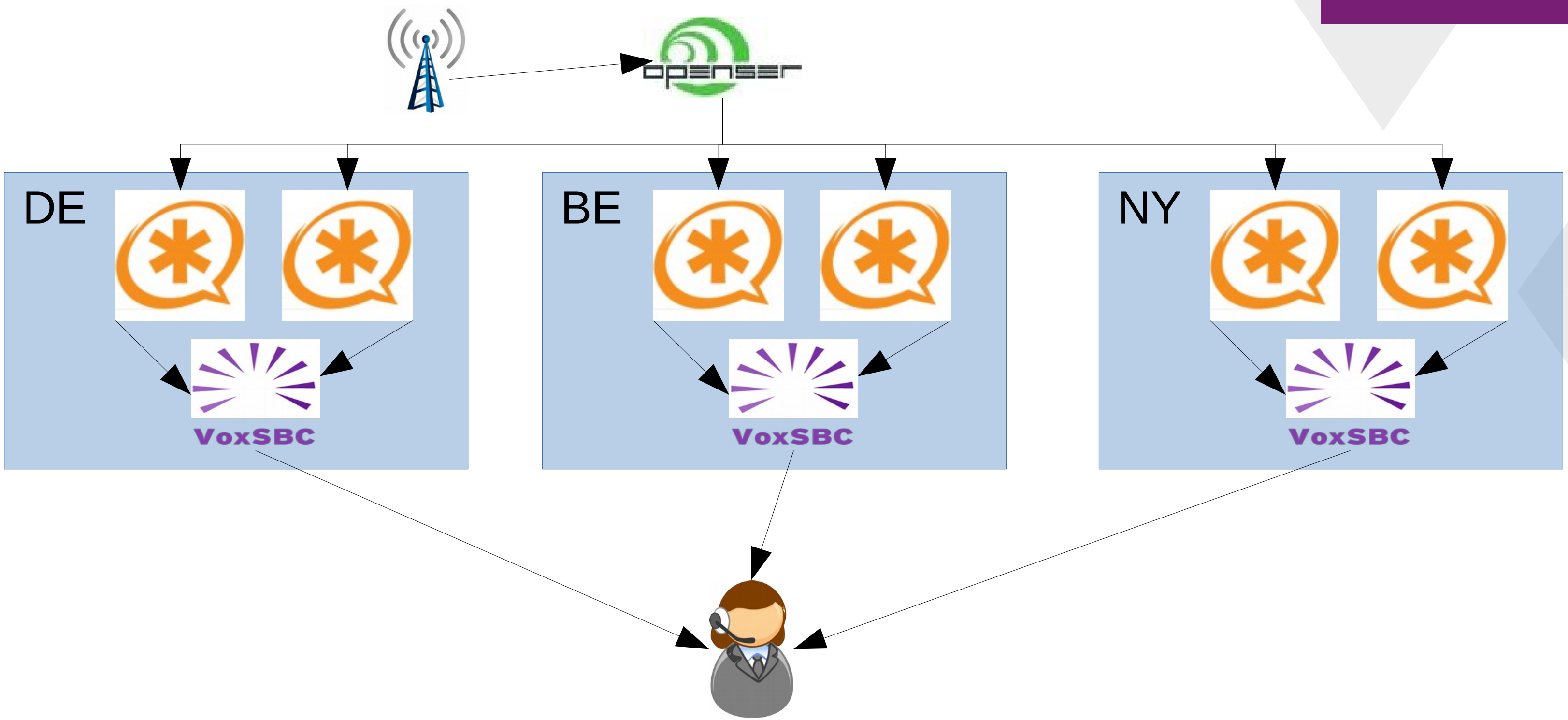
```
[sbc]
type=peer
Host=81.201.82.XX
port=5060
context=default

[sbc_info]
type=peer
host=81.201.82.XX
port=5060
context=default
dtmfmode=info

[sbc_g729]
type=peer
host=81.201.82.XX
port=5060
context=default
disallow=all
allow=g729

[sbc_ulaw]
type=peer
host=81.201.82.XX
port=5060
context=default
disallow=all
allow=ulaw
```

- SBC



Asterisk Realtime

- Would be nice to put the peers in the database, but the ip of the SBC is pop specific
- The primary key is the combination of desired config options
- Service ip address of the SBC was put in the hosts file, peer in the database refers to the hostname 'pop-sbc'
- When customer specifies a configuration, an existing matching peer is looked for, if none exists a new peer is added into the database
- Highly configurable with minimum number of peers in asterisk memory

DTMF Joys

- Some customers may forward the same sip uri to different equipment
- Some equipment might Require INFO others RFC2833
- Solution: DTMF Auto, INFO Failback

Voxbone - 2011

- 5 pops, 5 sers
- Voxout Emergency Services released
- A new Kamailio instance added to authenticate calls from the customer
- All Voxout customers use the same peer in chan_sip
- This peer needs to have a configuration that makes everybody happy

Voxbone - 2013

- Asterisk 1.4 – 11
- Initial investigations into WebRTC
- The Kamailio used for Emergency Services also used for the WebSocket Traffic

Issues

- WebRTC calls require ICE, DTLS-SRTP, and AVPF
- Emergency Service customers don't
- Both are the same SIP Peer from the point of view of Asterisk
- Solution: Add an optional mode for all of the above

Voxbone – 2015 HD Voice & Video

- High Definition Voice
  - Need to Offer HD Codecs in Dial() only if the original call supports HD Voice
  - Need to Accept HD Codecs in the 200 OK
  - Need to get Both Endpoints to agree on a video codec

Voxbone - Future

- New Function added to chan_sip: SIP_SDP_OFFER – returns a csv of codec/bitrate/type
- New Channel Variable SIP_CODEC_OUTBOUND_ORDER – allows a csv of codecs to be specified, controls the order of SIP Codecs on the Outbound Leg, codecs not allowed by the peer are removed
- If a call is bridged copy the codec order from the B leg to the A leg in the answer
- If a call is bridged and there are common codecs between the B leg and the A leg, remove all not supported by B in A's response

THANK YOU

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