SIP Trunking Registration Mode

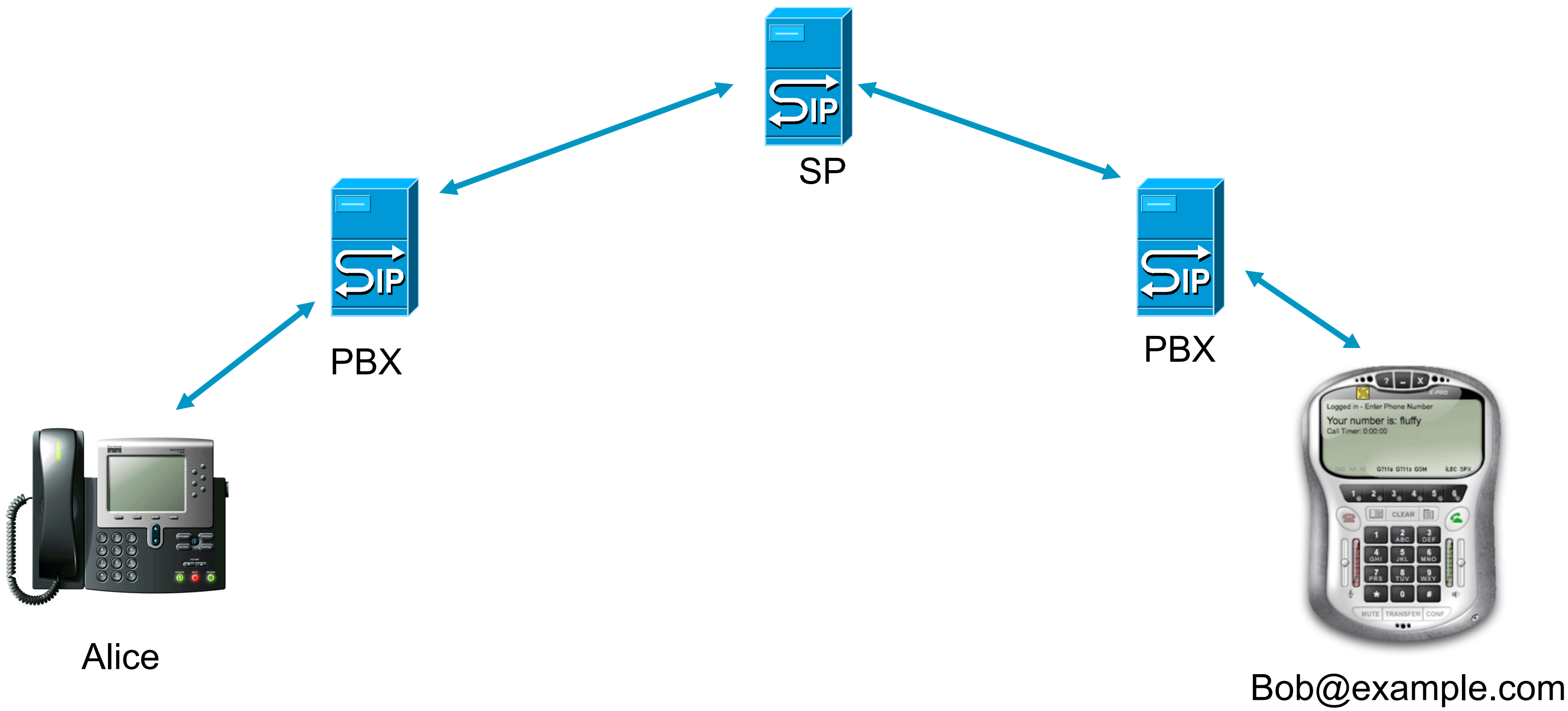
Cullen Jennings, PhD

fluffy@cisco.com

Requirements

- Provide a way for SMB PBX systems to have a DHCP assigned address
- Not require any configuration changes when the PBX's IP address changes

System Architecture



- The key question is about who is responsible for the domain example.com, is it the SP or the PBX?

Solution in SIP Forum draft ver 10

- SP is responsible for AOR such as fluffy@sp.com
- When the call arrives to SP, it loads a loose route to the PBX
- When call arrives at PBX, 3263 forwarding procedures say the call should be forwarded back to the SP but instead, the PBX knows to process the call as if it was responsible for the domain
- Local call from a device on the PBX to fluffy@sp.com, are not routed to the SP

- The problem here is it is unclear who provides the rendezvous service, the SP, or the PBX. At different times they both do. This leads to problems described in later slides

Alternative solution from earlier drafts

- When a call to fluffy@sp.net arrives at the SP, the SP replaces the request URI with the contact from the PBX registration and forward to PBX
- Problem here is many users all get forwarded with same request URI. The contact in the register needed to be unique to the device
- Again causes problem with PBX not understanding which users the call was meant for

Example of something this breaks

- Common telephone feature is having a call simultaneously ringing multiple phones
- Imagine the case where a user had a phone (with same number) in two different offices each with their own PBX. This is not possible with current registration mode as a local call on one PBX would not get routed correctly to ring both phones
- In a similar way, any feature that would normal be inserted by the SP at the rendezvous point, would not work consistently

Proposed Solution

- Sort out if we want SP or PBX to be responsible for AOR
- If the SP is responsible, the propose that each users on PBX registers separately
  - well understood, many deployments do this
  - close to residential voice model
  - Authenticated registration about 1k, if you have enough bandwidth for VoIP for X users, you have enough bandwidth to register X users
- If the PBX is responsible, propose we do some method for PBX to inform SP how to reach the PBX for the PBX domain
- Chose one of :
  1) DynDNS update of domain to IP mapping in DNS
  2) Use TLS cert names (implement connection-reuse draft)
  3) Digest challenge a SIP info message that indicates route availability
  4) Use TRIP