ShoreTel
INTELLIGENT PHONE SYSTEMS

ShoreTel SIP Trunks

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THE INNOVATION LEADER IN
ENTERPRISE IP TELEPHONY

Agenda

- ShoreTel SIP
  - Overview
  - Which RFC’s are supported?
  - Common SIP Traffic Flows (Basic SIP Message, SIP Responses & SIP Call)
- SIP Service Providers
  - Overview
  - Information to collect before starting
  - ShoreTel / Ingate
  - ShoreTel Configuration
- What to Test?: 2-way audio / DTMF / Transfer / Conference

Overview of ShoreTel SIP

- ShoreTel supports "SIP Trunks"
- SIP is still evolving, few more years before most devices work together out of box!
  - *Example:* Similar to when Ethernet was introduced.
- ShoreTel Certification Program
  - Provides: "app notes", "product certification" and etc…
  - NOTE: USE certified solutions
    - Certified means ShoreTel and other vendors have open communication.
    - Solution has been gone through extensive testing.
- ShoreTel SIP Trunks best used between two ShoreTel systems, ITSP's or Trunk devices (BRI or Analog Gateway)
  - End points such as WiFi, Analog ATA extensions have limitations
- Simple things such as two way audio normally can work
  - Issues can occur when attempting DTMF, transfer, conference and etc!

Which RFC's are supported?

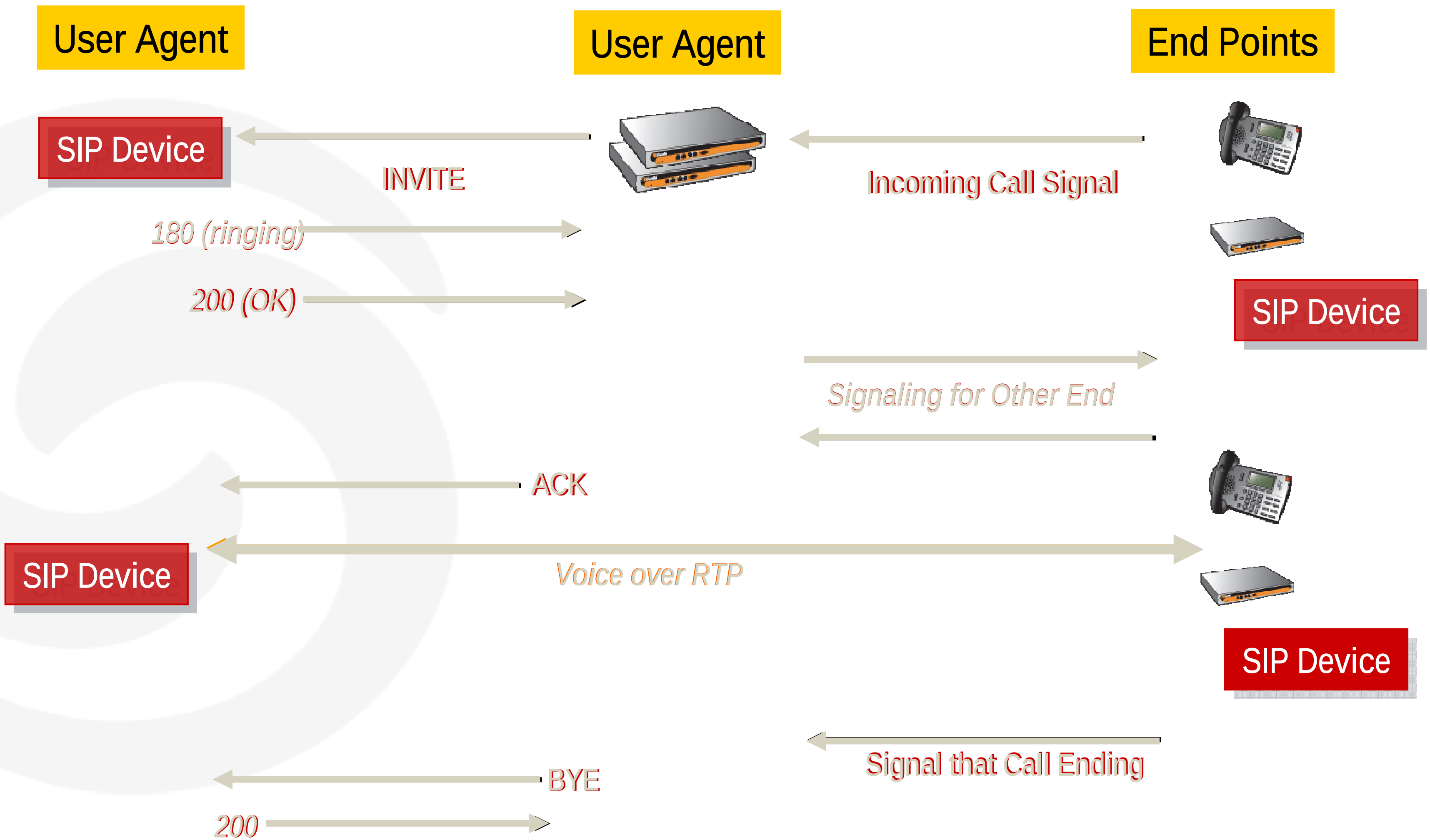
- RFC 3261 - Session Initiation Protocol (SIP)
- RFC 1889 - A Transport Protocol for Real-Time (RTP) Applications
  - Note: Real time transport control protocol (RTCP) is not supported in this ShoreTel release.
- RFC 2327 – Session Description Protocol (SDP)
- RFC 2976 – SIP Info Method
  - For SIP INFO, ShoreTel uses the SIP message *body of type* where:
    Content-Type: application/dtmf-relay
    Signal=1
    Duration=160
- RFC 2396 – Uniform Resource Identifiers (URI)
- RFC 3515 – SIP Refer Method
- RFC 3891 – SIP "Replaces" Header
- RFC 3892 – SIP Referred-By Mechanism
- RFC 2806 – URLs for Telephone Calls
- RFC 3966 - URIs for Telephone Calls
- RFC 2833 - DTMF

Great source for information on RFC's:
http://www.ietf.org/rfc/

Basic SIP Messages

- INVITE—to participate in session
- BYE—release call
- ACK—final response for Invite
- REGISTER—client registers with server
- CANCEL—cancels pending request
- REFER—direct to another party
- INFO—carries information such as DTMF digits

Basic SIP Call Flow



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SIP Service Providers

Overview SIP ITSP

- What are important components for a good ITSP?
  - Does it WORK? Are they ShoreTel certified?
  - Strong SLA
    - Will this connection be voice only or shared with data?
    - What is the QoS story for IP Telephony? How is it prioritized?
    - Do they support rate shaping based on each traffic queue?
    - What type of failover, redundancy and least cost routing do they support?
  - What inbound support do they support? Local DID or 800 #?
  - International support?
  - What codec's do they support (G.711 or G.729)?
  - $, Monthly Fee? Toll Fees?

Information to collect before starting

- How will everything connect? Will it be over the Internet or through a private connection?
- How much bandwidth is being provisioned for IP Telephony?
- Quality of Service or SLA's are offered / supported?
- Codec and DTMF support

VERY IMPORTANT

- Will it connect through the corporate firewall (Private or Public) and if NAT is used (most common), does customer Firewall support SIP NAT Traversal or does it need upgraded / replaced?

"This is where most issues come from when deploying SIP Trunks!"

ShoreTel / Ingate Partnership

- Ingate sits between the LAN / WAN
- Ingate bridges gap between ShoreTel and ITSP
- Provides many key services and compatibility solutions
  - SIP NAT Traversal services (most firewalls don't cut it)
  - Call transfer solution – ShoreTel supports REFER, most ITSP's support reINVITE. Without solution won't work with many ITSP's.
  - Dial plan modification
  - IP Address conversion to Domain Name

    example: 209.118.23.76 = ShoreTel.com
  - Security – Firewall rules

ShoreTel / Ingate Partnership

Ingate Provides

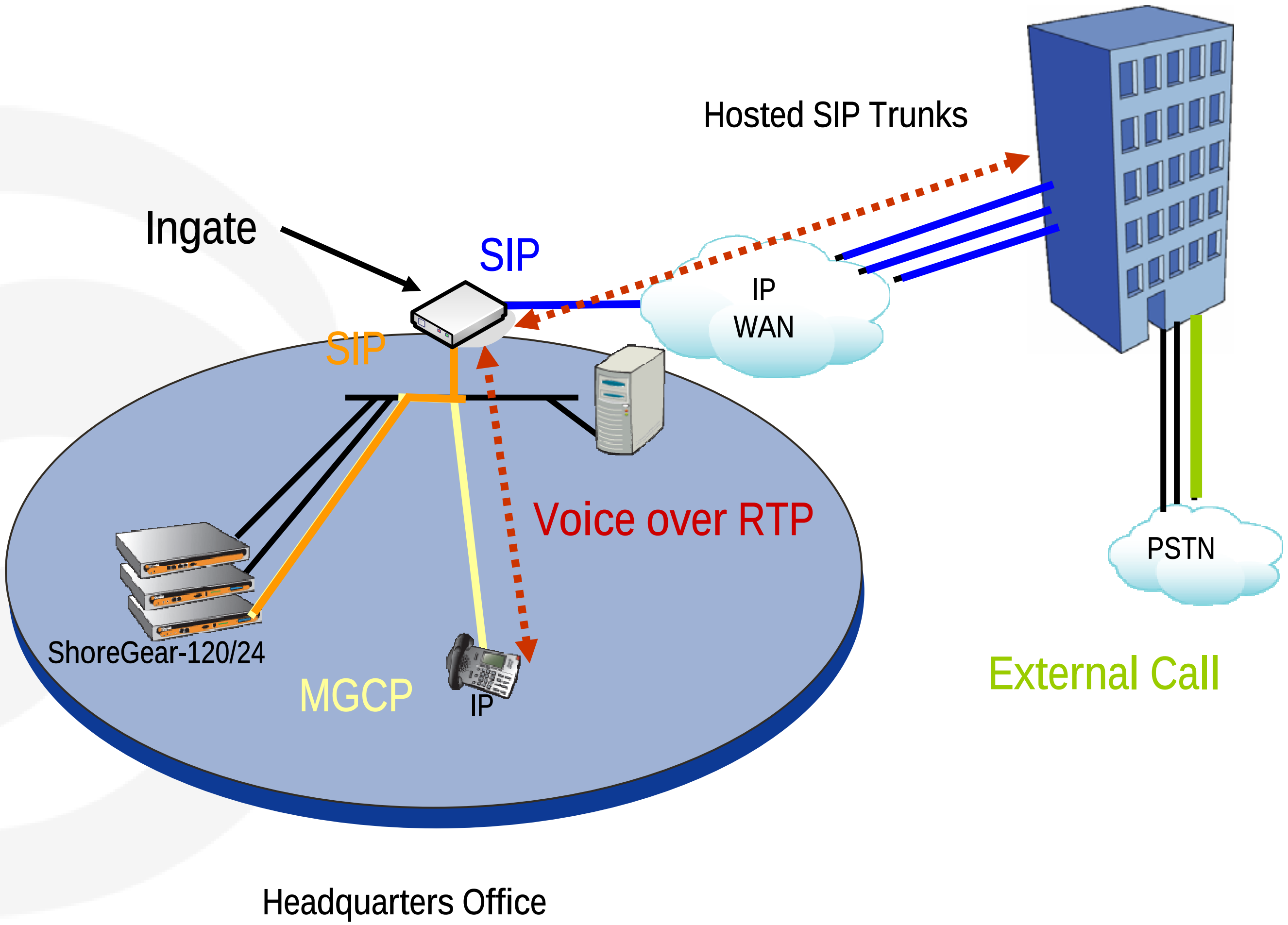
- Support of Registration, Digest Authentication, DNS, Calls from particular #
- Flexibility for ITSP solutions & Far end SIP NAT Traversal

Very Important

- Certification testing ShoreTel / Ingate solution with key ITSP
  - Global Crossing
  - BandTel
  - Others

  Target release of certification and app note documents in 30 to 60 days!

ShoreTel SIP Trunks Example:
Hosted SIP Trunks



What to test prior to deployment???

- Quick validation
  - 2 way call
  - DTMF (verify both directions)
    - Call into AA, VM
    - Call out ShoreTel system to system with AA or VM
  - Transfer
  - 3 party conference

  Note: Always try test in different directions and from different devices!
- More extensive testing should completed based on usage plans
- Consider running pilot before mass deployment

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Thank You