Interoperation with the Circuit-Switched Telephone System

- interoperation scenarios
- mapping E.164 phone numbers to IP addresses
- finding hop-off gateways: TRIP
Interworking scenarios

In increasing complexity:

- phone - analog line (FXS) - Ethernet
- connect to PBX
- corporate gateway – “Internet PBX”
- carrier with multiple gateways – “IP Centrex”

Often, same equipment, different location
Internet PBX

- PSTN/IP Interworking
- Internet PBX
- SIP proxy server
- Ethernet
- ISP
- T1, ISDN (BRI/PRI) or analog lines

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IP Centrex
IP Carrier

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Interoperation

- from PSTN to IP:
  - can’t dial IP addresses
  - map E.164 number to SIP address ➔ ENUM
  - gateway identified by phone number (or mobile gateway…)

- from IP to PSTN:
  - use `tel:+1-212-555-1234` or `sip:+1-212-555-1234@i2p.com;user=phone` to address user
  - need to find gateway

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Mapping from IP to PSTN

Dialing 7042

- Ip address: 240.7.9.3.9.2.1.2.1
- Sip://alice@somewhere.com
- Tel: 7042
- Outbound proxy or pre-configured table or enum
- Outbound proxy: sip:7042@cs.columbia.edu
- Outbound proxy: tel:7042
- Outbound proxy or pre-configured table or enum: sip.cs.columbia.edu
- Outbound proxy or pre-configured table or enum: tel:7042
- Outbound proxy or pre-configured table or enum: sip:alice@somewhere.com
- Outbound proxy or pre-configured table or enum: somewhere.com
- Outbound proxy or pre-configured table or enum: TRIP

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DNS for number mapping

- PSTN may treat DNS as yet another number routing database (800, 900, mobility, number portability, ...)
- use DNS NAPTR rewriting for local number portability (LNP)
- find out which signaling services are supported
- can also be used for E.164 numbering, e2e IP transport

Issues for any scheme:

- rapid, secure, globally visible updates (e.g., for LNP)
- ownership of addresses or neutral third party?
**e164.arpa**

1. convert to digits: +46-8-56264000 → 46856264000
2. insert dots: 4.6.8.5.6.4.0.0.0
3. reverse: 0.0.0.4.6.5.8.6.4
4. append TLD: 0.0.0.4.6.5.8.6.4.e164.arpa
Phone number rewriting

- NAPTR (RFC 2168, being updated): originally for URN rewriting
- flag: “a” for A RR, “s” for SRV RR

2.8.0.4.6.2.6.5.8.6.4.e164.arpa.
IN NAPTR 10 10 "a" "sip+N2R"
    "sip:paf@swip.net".
IN NAPTR 102 10 "s" "potscall+N2R"
    _potscall._tcp.paf.swip.net.
IN NAPTR 102 10 "a" "smtp+N2R"
    mailto:paf@swip.net".
Local number portability

- old number: +46-346-23232, but has moved to a ’telcoy’.
  2.3.2.3.2.6.4.3.6.4.e164.int. IN NS ns.telcoy.net.
- Telco Y allocates +46-8-919191:
  $ORIGIN 6.4.e164.arpa.
  _potscall._tcp.2.3.2.3.2.6.4.3 IN SRV 10 10 1.9.1.9.1.9.8
PSTN-to-VoIP gateway location

- currently, not specified
- use hop-off closest to IP destination or where doing number translation?
- maybe get IP address via SIP OPTIONS and then determine “best” hop-off point?
Gateway location protocol architecture
Gateway location protocol

- inter-domain distribution of gateway properties
- multiple signaling hops - aggregation → additional server
- aggregation of entries (e.g., Cologne/Bonn area +49221, +49228, +492222, ... → +4922)

- but difficult in NANP: NJ has 201, 609, 732, 908, 973.
- local calling areas may have a hundred exchange codes!
- generally, use most specific match
- policy filtering to restrict advertising prefixes
## Attributes

<table>
<thead>
<tr>
<th>Attribute</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>DestinationPhoneNumbers</strong></td>
<td>phone number prefix (+1201)</td>
</tr>
<tr>
<td><strong>NextHopSignalingServer</strong></td>
<td>direct H.323, SIP, ... here</td>
</tr>
<tr>
<td><strong>AdvertisementPath</strong></td>
<td>( \approx ) BGP AS_PATH, object path</td>
</tr>
<tr>
<td><strong>GatewayCapacity</strong></td>
<td>load balancing?</td>
</tr>
<tr>
<td><strong>SignalingProtocols</strong></td>
<td>H.323, SIP, ...</td>
</tr>
<tr>
<td><strong>Pricing</strong></td>
<td>$/minute, setup charges, increment, ...</td>
</tr>
<tr>
<td><strong>LastModifiedBy</strong></td>
<td>verifiable source of attribute</td>
</tr>
<tr>
<td><strong>NumSignalingHops</strong></td>
<td>min/max <em>signaling</em> (not media) hops</td>
</tr>
<tr>
<td><strong>AtomicAggregate</strong></td>
<td>aggregation indicator</td>
</tr>
<tr>
<td><strong>MultiExitDisc</strong></td>
<td>preference between multiple LS–LS links</td>
</tr>
</tbody>
</table>
Telephony Routing Information Protocol (TRIP)

- from BGP-4: transport (TCP), message format (TLV)
- from OSPF and SCSP: flooding
- initially, exchange routing database when establishing connection
- incremental updates (no refreshes)

**UPDATE**

- advertised route
- signaling server
- price
- withdrawn route
- withdrawn route
TRIP

open handshake

KeepAlive (30s)

error condition

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Signaling: SIP to ISDN
Signaling: SIP to ISUP

INVITE sip:+1−212−555−0101

100 Trying

183 Session Progress
duplex RTP media

200 OK

ACK

BYE

REL

200 OK

IAM

ACM

ANM

one-way voice

+1 212 555 0202

+1 212 55 0101

 caller

SIP

NGW

ISUP

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