VoIP from A to Z

NAEO 2009 Conference
Cancun, Mexico
VoIP glossary

- What is VoIP?
- Bandwidth
- Signaling
- Codecs
- Quality of Service (QoS)
What is VoIP?

• Voice over Internet Protocol (VoIP) is the method of transmitting voice calls over an IP based network.

• VoIP does not have to operate over the Internet.

• VoIP gives you flexibility with cost, reliability, quality that cannot be obtained in the PSTN.
Bandwidth

• Bandwidth is a measure of data flow rate in digital networks

• You can never have enough bandwidth.

• T1 = 1.5mbps, T3 = 45mbps

• Cable, DSL options

• Carrier Ethernet options (VzON, TLS)
Signaling

- Signaling is used in the telephone network for call creation and supervision.
- Information such as CallerID, DNIS, ANI is sent via signaling
- Many competing standards (SIP, MGCP, H.323, H.248, SCCP)
- Session Initiation Protocol (SIP) has won the battle.
Codec

- Codec stands for enCOding & DECoding.
- Used in the network to digitize the analog voice signal
- Codec choice has a big impact in voice quality and bandwidth requirements
- Major codecs (g.711u, g.729)
- g.711u is the same as a T1 (PCM). uses 80kbps of bandwidth per call. Supports faxing & modem traffic if your network is clean enough
- g.729 is a lower quality, high CPU, low bandwidth codec. 16kbps of bandwidth per call. Cannot support faxing & modem traffic ever
Quality of Service (QoS)

- QoS affects the quality of your voice traffic

- Measured in Latency, Jitter and loss

  - **Latency** = The time it takes for a packet to travel across the network. Latency below 150ms is preferred. Latency = echo which is easily handled by the echo cancelers in the network if the jitter is low enough.

  - **Jitter** = The difference in inter-packet latency. Jitter is bad! Phones & gateways have jitter buffers to compensate but they can only do so much.

  - **Loss** = Packet loss, packets that are dropped by a device along the network path. Commonly caused by queue overruns
QoS part II

- QoS is provided by Marking, Classification and Queuing
  Packet marking is handled by the end points according to class.
  Routers & switches react to the markings by queuing the packets.
  Most routers have 4 queues.

- Nothing helps QoS like MORE BANDWIDTH.
  The faster the interface the easier it is for packets to get through.
  An interface that shows 30% utilization over 5 minutes is actually 100% full 30% of the time! (1.5 minutes of jitter & loss)
  Data packets (web page) are 1500 bytes = 8ms on a T1, 0.12ms on FastE (100mbps)

- Default action for equipment is ‘best effort’/FIFO queuing
  Queuing options vary by vendor and can become extremely complex
  Queues need to be configured on every device.

- 2 types of QoS in the network
  DSCP - IP layer 3 (aka DiffServ) - typically VoIP traffic is marked with DSCP:EF (expedited forwarding)
  802.1p - Ethernet layer 2 (switches)

- DSCP does not survive a trip through the Internet.
  ISPs reset the DSCP marking to DSCP:00 (best effort) on all packets

- 802.1p marking does not survive a trip through a router and must be remapped from the DSCP values at every interface.
VoIP Components

- **Call Agent** - Softswitch, PBX. Handles call control and signaling.

- **Media Gateway** - Converts VoIP traffic into traditional telephony services (ISDN PRI)
  Provides the CPU power through Digital Signal Processors (DSP) to handle the codec work.

- **ATA - Analog Telephone Adapter** - converts VoIP Traffic into traditional analog lines (POTS).

- **User Agent** - Software component of a VoIP end user device (IP Phone, soft phone) registers with a Call Agent.

- **SBC - Session Border Controller**, aka VoIP firewall handles the interaction between devices and the public Internet. Provides NAT Traversal of VoIP packets.
Asterisk?

- Asterisk is a ‘swiss army knife’ VoIP toolbox.
- It can be a Call Agent, User Agent, Media Gateway, IVR, etc.
- Open source, runs on Linux.  [www.asterisk.org](http://www.asterisk.org)
- High degree of complexity. It can do anything but requires Linux knowledge and programming (PHP, Perl, C)
- NOT recommended as a gateway (too many moving parts).
- Many low cost PBX options are based on Asterisk. May be a better choice for companies with small IT staff.
• Operator has a SIP based VoIP phone which registers directly to the SIP gateway.
• Operator logs into WinOP then dials DID access code
• OpAudio enters Infinity via ISDN PRI as a call (Account behavior = OpAudio)
• Operator can make calls through Infinity
• No month recurring charges (no need for VONAGE, etc)
• Can be used for local & remote operators

Equipment options:
• Phones - Linksys SPA-942, CounterPath eyeBeam
• Gateway - Adtran 908e, Quintum, Asterisk
Example 2
Inbound DIDs via VoIP

VoIP Service Provider

Internet

Firewall → LAN → VoIP Gateway → Infinity

- VoIP Service provider sends inbound via SIP to SIP gateway
- SIP Gateway sends call to Infinity via PRI
- Outbound calls leave Infinity via PRI through gateway to SP via SIP
- Avoid using VoIP for Ultracom & Alpha paging unless you have a solid network and use g.711

Equipment options:
- Gateway - Adtran 908e, Quintum, Asterisk
How much bandwidth?

<table>
<thead>
<tr>
<th>Description</th>
<th>Quantity</th>
<th>Bandwidth needed (g.711) 80kbps per call</th>
<th>Bandwidth needed (g.729) 16kbps per call</th>
</tr>
</thead>
<tbody>
<tr>
<td>Operator Audio</td>
<td>10</td>
<td>800kbps</td>
<td>160kbps</td>
</tr>
<tr>
<td>Inbound calls via SIP trunk</td>
<td>25</td>
<td>2000kbps</td>
<td>400kbps</td>
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<tr>
<td>Outbound dispatcher calls</td>
<td>3</td>
<td>240kbps</td>
<td>48kbps</td>
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<tr>
<td>Total</td>
<td>38</td>
<td>3040kbps (3+ mbps)</td>
<td>608kbps</td>
</tr>
<tr>
<td>T1,T3,Ethernet transport (90% efficient)</td>
<td></td>
<td>3370kbps</td>
<td>676kbps</td>
</tr>
<tr>
<td>Cable, DSL transport (60% efficient)</td>
<td></td>
<td>5066kbps</td>
<td>1127kbps</td>
</tr>
</tbody>
</table>
Questions?