Thank you
INTRODUCTION

• Steven van Houttum
• Consultant/Trainer
• MVP Office Server & Services (Skype for Business)
TROUBLESHOOTING
PACKET LOSS, IS IT REALLY THE NETWORK?

- Voice Quality
- Audio Pipeline
- Troubleshooting
- Detect
- Fix
VOICE QUALITY

- Mouth to Ear

- Quality of Experience
  - Mean Opinion Score (MOS)
  - MOS: 1 to 5 (higher=better)
VOICE QUALITY
### Voice Quality – Audio Pipeline

<table>
<thead>
<tr>
<th>Analog audio source</th>
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Bandwidth estimation

RTCP packet pair packet

Host 2 calculate bandwidth for this packet pair. The statistics model needs more packet pairs to converge.

RTCP SR/RR with bandwidth estimation extension:
Bandwidth = -3 (not ready)

RTCP probe packet

RTCP packet pair packet

Host 2 calculate bandwidth for this packet pair. The statistics model now converges to 700000 = 700kbps

RTCP SR/RR with bandwidth estimation extension:
Bandwidth = 700000

Once Host1 receives a positive bandwidth estimation, it sends RTCP SR/RR in regular RTCP interval
Payload Type | Codec | Sampling Rate | Target Bitrate
--- | --- | --- | ---
104 | SILK Wide | 16k | 36kbps
103 | SILK Narrow | 8k | 13kbps

Audio Stream (Caller -> Callee)

- **Codec:** SILK Wide
- **Audio FEC:** False
- **Packet utilization:** 1
- **Max. packet loss rate:** 0.00%
- **Max. jitter:** 0 ms
- **Max. round trip:** 1 ms
## BANDWIDTH & CODECS

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Codec</th>
<th>Max NMOS</th>
<th>Bit-rate (kbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>UC-UC call</td>
<td>RTAudio WB</td>
<td>4.10</td>
<td>62</td>
</tr>
<tr>
<td>UC-UC call</td>
<td>RTAudio NB</td>
<td>2.95</td>
<td>44.8</td>
</tr>
<tr>
<td>UC-UC call</td>
<td>Silk</td>
<td>4.42</td>
<td>69</td>
</tr>
<tr>
<td>Conference call</td>
<td>G-722</td>
<td>4.31</td>
<td>100.6</td>
</tr>
<tr>
<td>Conference call</td>
<td>Siren</td>
<td>3.72</td>
<td>52.6</td>
</tr>
</tbody>
</table>
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MICROSOFT RTP EXTENSIONS (1/2)

- **Bandwidth estimation:**
  - Defines a new mechanism to estimate and communicate the bandwidth on a channel. One host sends two or more "probe packets", and the other host can use the time interval between them to estimate the bandwidth, which is then communicated back through a Real-Time Transport Control Protocol (RTCP) profile extension.

- **Packet loss notification:**
  - Defines an RTCP profile extension that allows a receiver to quickly notify the sender of the loss of a specific packet. The sender can then use this information to hasten recovery, such as by generating a new I-frame or Super P-frame (SP-frame) in the case of a video stream encapsulated through extensions described in [MS-RTVPF].

- **Policy Server Bandwidth:**
  - Defines an RTCP profile extension that allows a host to send its bandwidth provisioned by the policy server, as obtained through the Traversal Using Relay NAT (TURN) protocol to the remote host.

- **TURN Server Bandwidth:**
  - Defines an RTCP profile extension that allows a host to send its bandwidth provisioned by the TURN server as obtained through the TURN protocol to the remote host.
SDES PRIV extension for media quality:
- Defines a private Source Descriptions for Media Streams (SDES) extension for sending the media quality from the CSRC or SSRC to a media receiver. The receiver can use this information to show which source is causing quality issues.

Receiver-side audio healer report:
- Defines an RTCP profile extension that allows a host to send its receiver-side audio healer metrics, local network receive quality, and FEC distance request to the sender to help the sender drive the audio forward error correction (FEC).

Receiver-side bandwidth limit:
- Defines an RTCP profile extension that allows a host to send its receiver-side bandwidth limit request to the remote host to let the remote host know the maximum bandwidth it is capable of receiving.

Peer info exchange:
- Defines an RTCP profile extension that enables a host to send its inbound and outbound network bandwidth throughput limit to the remote host.
PACKET LOSS NOTIFICATION

Host1 (SSRC=1)

Video RTP packet (SeqNum=1, Marker bit=0)

Video RTP packet (SeqNum=3, Marker bit=1)

Host2 detects that packet with SeqNum=2 has been lost

RTCP probe packet

RTCP SR with packet loss notification extension (SeqNum=2)

Host1 forces the video encoder to generate an SP-frame

Video RTP packet (SeqNum=4, Marker bit=0)

Video RTP packet (SeqNum=5, Marker bit=1)

Host2 (SSRC=2)
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# Networking Factors

<table>
<thead>
<tr>
<th>Factor</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Latency</td>
<td>Round-trip time delay</td>
</tr>
<tr>
<td>Jitter</td>
<td>Change in delay</td>
</tr>
<tr>
<td>Packet Loss</td>
<td>% of packets lost</td>
</tr>
</tbody>
</table>
ROUND TRIP

- Distance related
- High average delay might also indicate network congestion
- Can indicate routing issues
JITTER

- High average jitter will add to the delay
- High max jitter might lead to late buffer drops
- Can be improved with QOS
PACKET LOSS

- High average packet loss might mean overloaded network
- Sporadic high max packet loss might indicate failing links
- Can lead to dropped calls
- Can usually be improved QoS
- Asymmetric: Duplex Mismatch
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HEALING

- Compensates for Jitter or Packet Loss
- Extrapolates missing data
- “Time Warping”
- Comfort Noise
- Perception is key

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<tr>
<th>Healer concealed ratio</th>
<th>Healer stretched ratio</th>
<th>Healer compressed ratio</th>
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<tr>
<td>0.00 %</td>
<td>0.10 %</td>
<td>0.40 %</td>
</tr>
<tr>
<td>2.00 %</td>
<td>0.50 %</td>
<td>5.00 %</td>
</tr>
<tr>
<td>6.92 %</td>
<td>2.33 %</td>
<td>1.42 %</td>
</tr>
</tbody>
</table>
FORWARD ERROR CORRECTION (FEC)

- Used to compensate for packet loss
- Costs bandwidth
- Look for incidents vs structural issues
NETWORK RELATED ISSUES

Call Reliability
- Dropped Calls
- Failed Calls

Audio Quality
- Broken Up Audio
- Delay
- Distortion
- Echo
MOS

Network MOS:
- Considers only network factors, such as:
  - Codec used
  - Packet loss
  - Packet reorder
  - Packet errors
  - Jitter
- Useful for identifying network conditions impacting audio quality

Listening MOS:
- Considers codec used, capture device characteristics, transcoding, mixing, defects from packet loss/packet loss concealment, speech level, and background noise
- Useful for identifying payload effects impacting audio quality
MOS

- **Sending MOS:**
  - Considers:
    - Capture device characteristics
    - Speech level
    - Background noise
  - Useful for identifying device issues and contrasting to Listening MOS

- **Conversational MOS:**
  - Considers same factors as Listening MOS plus echo, network delay, delay to jitter buffering, delay due to devices
  - Useful for identifying bi-directional effects
WHAT IMPACTS MOS?

• Network MOS:
  • LAN Congestion
  • Packet Loss
  • Jitter

• Sending MOS:
  • Devices issues
## MAX NETWORK MOS RATING BY CODEC

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<td>RTAudio NB</td>
<td>2.95</td>
</tr>
<tr>
<td>UC-PSTN call</td>
<td>G-711</td>
<td>3.61</td>
</tr>
</tbody>
</table>
MOS DEGRADATION

- Amount of impairment from the maximum
  Lower = better

- Causes:
  - Congestion
  - Bandwidth
  - Wireless interference
  - Overloaded server (anti-virus exclusions)

- DegradationAvg metric
PACKET LOSS TROUBLESHOOTING

- Is it really the network?
  - Monitoring Server
  - KHI
- Incident or structural?
- Network capture
  - Message Analyzer
DETECT ISSUES

Monitoring Server
- Quality of Experience (QoE)
- Call Detail Record (CDR)

Call Quality Methodology (CQM)

Call Quality Dashboard (CQD)

Rate my Call (CQF)
MEDIASTACK BEHAVIOUR

- Jitter -> Jitter buffer
- Packet Loss -> Packet Loss Concealment / FEC
- Audiohealing

- This knowledge can be helpful for understanding QoE
DEMO

Monitoring Server
PRE-CALL DIAGNOSTICS TOOL
## CQM Looks At Quality Three Ways

<table>
<thead>
<tr>
<th>Servers</th>
<th>Network</th>
<th>Endpoint</th>
</tr>
</thead>
<tbody>
<tr>
<td>Lync servers must be healthy and running without resource constraints</td>
<td>Media stream quality between Lync servers—audio/video multipoint conferencing unit (AV MCU), mediation, gateway. Media stream quality between endpoints, and endpoints to servers.</td>
<td>Endpoint factors including system, device, media transport, and media path.</td>
</tr>
</tbody>
</table>
# CQM SCORECARD

**Microsoft Call Quality Methodology Scorecard for Skype for Business Server**

<table>
<thead>
<tr>
<th>Steps</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Select decimal separator used in CSV files</td>
<td>Decimal Separator: Comma</td>
</tr>
<tr>
<td>2. Select date format used in CSV files</td>
<td>Date Format: YYYY-MM-DD</td>
</tr>
<tr>
<td>3. Specify any off-work days</td>
<td>Off-Work Day 1: Saturday, Off-Work Day 2: Sunday</td>
</tr>
<tr>
<td>4. Streams cutoff number</td>
<td>Streams Cutoff: 100</td>
</tr>
<tr>
<td>5. Import CQM Query results as sheets</td>
<td>Import CQM Query Results</td>
</tr>
<tr>
<td>6. Remove results from off work days</td>
<td>Remove Off-Work Days Results</td>
</tr>
<tr>
<td>7. Generate Stream Distribution charts</td>
<td>Generate Stream Distribution Chart</td>
</tr>
<tr>
<td>8. Generate Trending Charts</td>
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</tr>
<tr>
<td>9. Generate Reliability Charts</td>
<td>Generate Reliability Charts</td>
</tr>
<tr>
<td>10. Generate Top issues</td>
<td>Generate Top Issues</td>
</tr>
<tr>
<td>11. Update Scorecard sheet</td>
<td>Update Scorecard</td>
</tr>
</tbody>
</table>

**Poor Stream Distribution**

- **AVMCU Mediation Mediation Gateway**
- **Wireless P2P**
- **Wireless**
- **Wired**
- **External**
- **VPN**
- **Other Wireless**
- **Other Wired**
- **Wired P2P**
KEY HEALTH INDICATORS (KHI)

- 8o KHI
- Exclude non-network issues
- Healthcheck
- KHI Guide

Call Quality Methodology

Monitoring Server
System Center Operations Manager
Synthetic Transactions
Partner Solutions

Key Health Indicators
DEMO

Key Health Indicators
KEY HEALTH INDICATORS (KHI)
STATISTICS MANAGER

Description: The image shows a screenshot of the Skype for Business Statistics Manager. It displays a scenario for Fabric Health with chart data and statistical information. The chart shows various metrics such as Registrar, with data points for dates like 5/12/2016, 6/14/2016, and 7/12/2016. The chart is labeled with various statistical measures such as 'Number of pending Create Service Calls', 'Replicas Pending Role Transition', 'Number of write requests that were denied due to lack of write quorum', 'Number of write requests that were denied due to other errors', and 'Number of routing groups for which the current machine is active secondary replica'. The interface allows selecting a population, with options like FrontEnd and p44.
CALL QUALITY DASHBOARD (CQD)
• Rate my Call
• Aka Call Quality Feedback (CQF)
• Stored in QoE database
When it is the network:
- Adding Bandwidth
- Network Management
- Networking guide recommendations

When it is Skype for Business:
- Use KHI output
- Check hardware
- Anti-virus solutions
- Virtualization done wrong
NETWORK MANAGEMENT

Skype for Business

• Call admission Control (CAC)

Network

• Quality of Service (QoS)
• Software Defined Networking (SDN)
QOS VISUALIZED

Audio Traffic
- Audio
  - DSCP 46 - Queue

Video Traffic
- Video
  - DSCP 34 - Queue

Other Lync Traffic
- Signalling
- App Sharing
- File Transfer
  - Unmarked - Queue

Total Available Bandwidth
<table>
<thead>
<tr>
<th>Scenario</th>
<th>Starting port</th>
<th>Ending port</th>
<th>Port count</th>
<th>Range</th>
<th>DSCP Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Server Audio</td>
<td>40803</td>
<td>49151</td>
<td>8349</td>
<td>40803:49151</td>
<td>46 (EF)</td>
</tr>
<tr>
<td>Server Video</td>
<td>49152</td>
<td>57500</td>
<td>8349</td>
<td>49152:57500</td>
<td>34 (AF41)</td>
</tr>
<tr>
<td>Server Application Sharing</td>
<td>57501</td>
<td>65534</td>
<td>8033</td>
<td>57501:65534</td>
<td>26 (AF31)</td>
</tr>
<tr>
<td>Client Audio</td>
<td>40803</td>
<td>40842</td>
<td>40</td>
<td>40803:40842</td>
<td>46 (EF)</td>
</tr>
<tr>
<td>Client Video</td>
<td>49152</td>
<td>49191</td>
<td>40</td>
<td>49152:49191</td>
<td>34 (AF41)</td>
</tr>
<tr>
<td>Client Application Sharing</td>
<td>57501</td>
<td>57540</td>
<td>40</td>
<td>57501:57540</td>
<td>26 (AF31)</td>
</tr>
<tr>
<td>Client File Transfer</td>
<td>20000</td>
<td>20039</td>
<td>40</td>
<td>20000:20039</td>
<td>14 (AF13)</td>
</tr>
<tr>
<td>Client Media (OCS clients)</td>
<td>20040</td>
<td>20079</td>
<td>40</td>
<td>20040:20079</td>
<td>46 (EF)</td>
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<td>Client SIP</td>
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http://www.ucunleashed.com/2821
DEMO

Networking Guide
PACKET LOSS TROUBLESHOOTING

- Create network capture
- Filter network capture by RTP or RTCP Ports, Conversation ID or SSRC (Synchronization Source ID) to correlate with QoE or S2S Individual streams query.
- Filter the trace for RTCP Loss > 0.
- Identify RTCP Sender/Receiver report blocks where Packet Loss was reported.
- Verify Sequence #'s end to end to determine network loss (sequence not arriving).
- Evaluate RTP Payload Type for packets immediately preceding the reported packet loss to determine causality.
CONCLUSION

• Plenty troubleshoot options
• Context is very important
  • Is it really the network?
  • Incident vs structural issue
• Prevention is better
  • Network Assessment
  • Enough bandwidth
  • Pro-active monitoring
• QoS often recommended
RESOURCES

- Call Quality Methodology: http://download.microsoft.com/download/4/3/2/43252959-AA20-4C3C-819D55A64F49EFB3/Microsoft%20Call%20Quality%20Methodology.zip
Thank you