VoIP Analysis Fundamentals with Wireshark…

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VoIP / Video Protocol Stack

Call Control & Signalling

- H.323
- H.225
- H.245
- Q.931
- SCCP
- UDP
- TCP
- RTP
- RTCP
- MGCP
- SDP / SIP
- Unistem
- G.711, G.729
- H.261, H.263
- RAS
- Q.931
- SCCP
- Unistem
- MGCP
- RTP
- RTCP
- IPv4 / IPv6

Media
VoIP Protocols Overview (Signaling)

- **MGCP - Media Gateway Control Protocol**
  - Defined by the IETF and ITU
  - Used to control signaling and session management (also known as H.248 or Megaco)

- **SCCP - Skinny Client Control Protocol**
  - CISCO proprietary protocol used to communicate between a H.323 Proxy (performing H.225 & H.245 signaling) and a Skinny Client (VoIP phone)

- **SIP - Session Initiation Protocol**
  - Defined by the IETF / RFC 2543 / RFC 3261

- **H.323 – Defines a Suite of ITU designed protocols**
  - H.225, H.245, Q.931, RAS, etc…
VoIP Protocols Overview (Data)

- **RTP** - Real Time Protocol
  - Defined by the IETF / RFC 1889
  - Provides end-to-end transport functions for applications transmitting real-time data over Multicast or Unicast network services
    - Audio, video or simulation data

- **RTCP** - Real Time Control Protocol
  - Defined by the IETF
  - Supplements RTP’s data transport to allow monitoring of the data delivery in a manner scalable to large Multicast networks
  - Provides minimal control and identification functionality

- **RTSP** - Real Time Streaming Protocol
  - Defined by the IETF / RFC 2326
  - Enables the controlled delivery of real-time data, such as audio and video
  - Designed to work with established protocols, such as RTP and HTTP
VoIP Codecs (Audio Conversion)

- **CODEC** = Compressor / Decompressor or Coder / Decoder or Reader
  - Provides conversion between Audio/Video signals and data streams at various rates and delays

- Designations conform to the relevant ITU standard
  - **Audio Codecs (G.7xx)**
    - G.711a / u - PCM Audio 56 and 64 Kbps (Most common business use)
    - G.722 - 7 Khz Audio at 48, 56 and 64 Kbps
    - G.723.1 / 2 - ACELP Speech at 5.3 Kbps / MPMLQ at 6.3 Kbps
    - G.726 - ADPCM Speech at 16, 24, 32 and 40 Kbps
    - G.727 - E-ADPCM Speech at 16, 24, 32 and 40 Kbps
    - G.728 - LD-CELP Speech at 16 Kbps
    - G.729 - CS-ACELP Speech at 8 and 13 Kbps (Very common for home use)
  - **Video Codecs (H.2xx)**
    - H.261 - Video >= 64 Kbps
    - H.263 - Video <= 64 Kbps

Forensics Analysis of User Traffic
VoIP Codecs

- CODEC = Compressor / Decompressor or Coder / Decoder or Reader
  - Provides conversion between Audio/Video signals and data streams at various rates and delays
Sample VoIP Codec Comparison

<table>
<thead>
<tr>
<th>Codec</th>
<th>Data Rate</th>
<th>Typical Datagram Size</th>
<th>Packetization Delay</th>
<th>Combined Bandwidth for 2 Flows</th>
<th>Typical Jitter Buffer Delay</th>
<th>Theoretical Maximum MOS</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711u</td>
<td>64.0 kbps</td>
<td>20 ms</td>
<td>1.0 ms</td>
<td>174.40 kbps</td>
<td>2 datagrams (40 ms)</td>
<td>4.40</td>
</tr>
<tr>
<td>G.711a</td>
<td>64.0 kbps</td>
<td>20 ms</td>
<td>1.0 ms</td>
<td>174.40 kbps</td>
<td>2 datagrams (40 ms)</td>
<td>4.40</td>
</tr>
<tr>
<td>G.726-32</td>
<td>32.0 kbps</td>
<td>20 ms</td>
<td>1.0 ms</td>
<td>110.40 kbps</td>
<td>2 datagrams (40 ms)</td>
<td>4.22</td>
</tr>
<tr>
<td>G.729</td>
<td>8.0 kbps</td>
<td>20 ms</td>
<td>25.0 ms</td>
<td>62.40 kbps</td>
<td>2 datagrams (40 ms)</td>
<td>4.07</td>
</tr>
<tr>
<td>G.723.1</td>
<td>6.3 kbps</td>
<td>30 ms</td>
<td>67.5 ms</td>
<td>43.73 kbps</td>
<td>2 datagrams (60 ms)</td>
<td>3.87</td>
</tr>
<tr>
<td>MPMLQ</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>G.723.1</td>
<td>5.3 kbps</td>
<td>30 ms</td>
<td>67.5 ms</td>
<td>41.60 kbps</td>
<td>2 datagrams (60 ms)</td>
<td>3.69</td>
</tr>
</tbody>
</table>

- MOS and R value include Packetization delay + Jitter buffer delay
- Common bandwidth – real bandwidth consumption:
  # Payload = 20 bytes/p (40 bytes/s)
  # Overhead includes 40 bytes of RTP header (20 IP + 8 UDP + 12 RTP)
Several different standards are currently competing for dominance in the VoIP field:

- **H.323** - Developed by the International Telecommunications Union (ITU) and the Internet Engineering Task Force (IETF)
- **MGCP / Megaco / H.248** - Developed by CISCO as an alternative to H.323
- **SIP** - Developed by 3Com as an alternative to H.323
- **SCCP** – Cisco Skinny Client Control Protocol – used to communicate between a H.323 Proxy (performing H.225 & H.245 signaling) and a Skinny Client (VoIP phone)
- **UNISTEM** – Proprietary Nortel protocol, developed by as an alternative to H.323
H.323 - Packet-based Multimedia Communications Systems

- An umbrella standard defined by the International Telecommunications Union (ITU) and the Internet Engineering Task Force (IETF)
- Defines a set of call controls, channel set up and Codec’s for multimedia, packet-based communications systems using IP-based networks

<table>
<thead>
<tr>
<th>H.450.1</th>
<th>Supplemental, generic protocol for use under H.323</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.225</td>
<td>Call Signaling / RAS</td>
</tr>
<tr>
<td>H.245</td>
<td>Control messages for the H.323 Terminal (RTP / RTCP)</td>
</tr>
<tr>
<td>H.235</td>
<td>Security Enhancements</td>
</tr>
<tr>
<td>Q.931</td>
<td>Call setup and termination</td>
</tr>
<tr>
<td>G.711, G.723.1 G.728</td>
<td>Audio Codec's</td>
</tr>
<tr>
<td>H.261, H.263, H.264</td>
<td>Video Codec’s</td>
</tr>
</tbody>
</table>
SIP VoIP Standard (SIP)

- Defined in RFC 2543 and RFC 3261 and by the ITU
  - Pioneered by 3Com to address weaknesses in H.323

- Application layer signaling protocol supporting real time calls and conferences (often involving multiple users) over IP networks
  - Can replace or complement MGCP
    - SIP provides Session Control and the ability to discover remote users
    - SDP provides information about the call
    - MGCP/SGCP Provides Device Control
    - ASCII text based
    - Provides a simplified set of response codes

- Integrated into many Internet-based technologies such as web, email, and directory services such as LDAP and DNS
  - Extensively used across WANs
MGCP / Megaco VoIP Standards

• Defined by RFC 2705 / 3015 and the ITU in conjunction with the H.248 standard
  – Pioneered by CISCO to address weaknesses in H.323

• Used between elements of distributed Gateways (defined later) as opposed to the older, single all-inclusive Gateway device
  – Extensively used in the LAN environment

• Utilizes Media Gateway Control Protocol (MGCP) to control these distributed elements
  – Often considered a “Master/Slave” protocol
Quality Of Service (QoS) - Overview

- Provides a guarantee of bandwidth and availability for requesting applications
  - Used to overcome the hostile IP network environment and provide an acceptable Quality of Service
    - Delay, Jitter, Echo, Congestion, Packet loss and Out of Sequence packets
  - Mean Opinion Score (MoS) / R-Factor is sometimes used to determine the requirements for QoS.
  - Utilized in the VoIP environment in one of several methods:
    - Resource Reservation Protocol (RSVP) defined by IETF
    - IP Differentiated Services
    - IEEE 802.1p and IEEE 802.1q
Assessing Voice Quality

- Voice Quality can be measured using several criteria
  
  1. **Delay**: As delay increases, callers begin talking over each other, eventually the call will sound like talking on a “walkie-talkie”. (Over…)

  2. **Jitter**: As jitter increases, the gateway becomes unable to correctly order the packets and the conversation will begin to sound choppy
     - Some devices utilize jitter buffer technology to compensate

  3. **Packet Loss**: If packet loss is greater than the jitter buffer, the caller will hear dead air space and the call will sound choppy
     - Gateways are designed to conceal minor packet loss
Different VoIP Quality Measurement Terms

- MoS – Mean Opinion Score
  - Numerical measure of the quality of human speech at the destination end of the circuit

- PSQM (ITU P.861)/PSQM+ - Perceptual Speech Quality Measure

- PESQ (ITU P.862) – Perceptual Evaluation of Speech Quality

- PAMS (British Telecom) Perceptual Analysis Measurement System

- The E-Model (ITU G.107) – (R-Factor)
  - Send a signal through the network, and measure the other end!
Measures of Voice Quality

- MOS can only be measured by humans
- R-value can be calculated in software
- PMOS values can be determined from R-value

E-Model “R” Factor scores comparison to MOS score
MOS (Mean Opinion Score)

<table>
<thead>
<tr>
<th>MOS</th>
<th>Quality Rating</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>Excellent</td>
</tr>
<tr>
<td>4</td>
<td>Good</td>
</tr>
<tr>
<td>3</td>
<td>Fair</td>
</tr>
<tr>
<td>2</td>
<td>Poor</td>
</tr>
<tr>
<td>1</td>
<td>Bad</td>
</tr>
</tbody>
</table>

1. Quality Goal is the same as PSTN and is widely accepted criterion for call quality

2. Call quality testing has always been subjective (Humans) - International Telecommunications Union (ITU) P.800

**MOS - Mean Opinion Score**
- Numerical measure of the quality of human speech at the destination end of the circuit (affected extensively by Jitter)
- Uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance
- Rating of 5.0 is considered perfect
The E-Model - Recommendation ITU G.107

- The "E-Model" is a parameter based algorithm based on subjective test results of auditory tests done in the past compared with current “system parameters”

- Provides a prediction of the expected quality, as perceived by the user

- The result of the E-Model calculation is “E-Model Rating R” (0 - 100) which can be transformed to “Predicted MOS (PMOS)” (1 – 5; 5 is non-extended, non-compressed)
  - Typical range for R factors is 50-94 for narrowband telephony and 50-100 for wideband telephony

Cascade Pilot Computes the R-Factor and MOS scores
Cascade Pilot computes both “R” Factor and MOS in multiple formats:
1. Average - R Factor / MOS
2. Maximum - R Factor / MOS
Cascade Pilot computes both Jitter and Delta in multiple formats:
1. Average / Maximum Jitter
2. Average / Maximum Delta
So what happens when we engage SIP VoIP?
Expected SIP Operation

• To initiate a session
  – Caller sends a request to a callee's address in the form of an ASCII text command
    • “Invite”
  – Gatekeeper/Gateway attempts phone number -> IP mapping/resolution
    • Trying / Response code = 100
    • Ringing / response code = 180
  – Callee responds with an acceptance or rejection of the invitation
    • “Accept” / response code = 200 “OK”
  – Call process is often mediated by a proxy server or a redirect server for routing purposes

• To terminate a session
  – Either side issues a quit command in ASCII text form
    • “Bye”
SIP Call Setup

End Point 1

Invite

Trying

OK

Trying

End Point 2

Invite

Trying

OK

ACK

ACK
Session Initiation Protocol (SIP - Invite)

- Request-Line: INVITE sip:4697@cisco.sip.ilabs.interop.net;user=phone SIP/2.0
  Method: INVITE
- Request-URI: sip:4697@cisco.sip.ilabs.interop.net;user=phone
  [Resent Packet: False]
- Message Header
  - Via: SIP/2.0/UDP 45.210.3.90:5060;branch=z9hG4bK6137b728
  - From: "Cisco 3290" <sip:3290@cisco.sip.ilabs.interop.net>;tag=003094c3438b00cd52bdf1e8-0d2f4d4b
    SIP Display info: "Cisco 3290"
  - SIP from address: sip:3290@cisco.sip.ilabs.interop.net
    SIP from address User Part: 3290
    SIP from address Host Part: cisco.sip.ilabs.interop.net
    SIP tag: 003094c3438b00cd52bdf1e8-0d2f4d4b
  - To: <sip:4697@cisco.sip.ilabs.interop.net;user=phone>
  - SIP to address: sip:4697@cisco.sip.ilabs.interop.net;user=phone
    SIP to address User Part: 4697
    SIP to address Host Part: cisco.sip.ilabs.interop.net
  Call-ID: 003094c3-438b0083-6f807304-47943c3c@45.210.3.90
  Date: Thu, 13 May 2004 18:11:17 GMT
  - CSeq: 101 INVITE
    User-Agent: CSCO/6
  - Contact: <sip:3290@45.210.3.90:5060>
    Expires: 180
    Content-Type: application/sdp
    Content-Length: 244
    Accept: application/sdp
- Message Body

SIP is data is carried in text format

SIP “Invite”
Session Initiation Protocol (SIP - Bye)

- **Request-Line**: BYE sip:3290@45.210.3.90:5060 SIP/2.0
  - Method: BYE
- **Request-URI**: sip:3290@45.210.3.90:5060
  - [Resent Packet: False]
- **Message Header**
  - Via: SIP/2.0/UDP 45.210.3.36:5060;branch=a84121e1-2d6f00ce-2bb702b0-fd00f62c-1
  - Via: SIP/2.0/UDP 45.210.3.36:5060;received=45.210.3.36;branch=cb89efff-be63b1bc-83f907fe-69cf5fcc-1, SIP/2.0/UDP
  - To: "Cisco 3290" <sip:3290@cisco.sip.ilabs.interop.net>;tag=003094c3438b00cf087acf0f-1340dfed
  - From: <sip:4672@cisco.sip.ilabs.interop.net;user=phone>;tag=614790957
  - Call-ID: 003094c3-438b0085-4ef5a663-56f32b68@45.210.3.90
  - Content-Length: 0
  - Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, INFO, MESSAGE, SUBSCRIBE, NOTIFY, PRACK, UPDATE, REFER
  - User-Agent: PolycomSoundPointIP-UA/1.0.9
  - Max-Forwards: 67
  - k: com.nortelnetworks.firewall,100rel,p-3rdpartycontrol
- **CSeq**: 36515 BYE
  - Sequence Number: 36515
  - Method: BYE

SIP - “Bye”
Challenges of VoIP

• Minimize Delay, Jitter and data loss
  – Excessive Delay variations can lead to unacceptable data lost or distortion

• Implementing QoS
  – RSVP designed to reserve required resources for VoIP traffic

• Interoperability of equipment beyond the Intranet
  – Different vendors Gateways utilize different Codec’s

• Compatibility with the PSTN
  – Seamless integration required to support services such as smart card and 800 service
Factors Affecting Delay and VoIP Quality - 1

- **Latency**
  - Round trip latency is the key factor in a call having an “interactive feel”
  - <100 msec is considered idle

- **Jitter**
  - Occurs when packets do not arrive at a constant rate that exceeds the buffering ability of the receiving device to compensate for
  - If excessive Jitter occurs, larger Jitter buffers will be required which cause longer latency

- **Packet Loss**
  - Loss of > 10% (non-consecutive packets) will be perceived as a bad connection
Factors Affecting Delay and VoIP Quality - 2

• Codec Choice
  – Add delay
    • Processing
    • Encoding / Decoding
  – Greater the compression factors result in lowered quality

• Bandwidth Utilization
  – Less utilization = lower latency, jitter and loss due to collisions

• Priority
  – Voice is extremely sensitive to delay
  – QoS is used to allow network devices to handle VoIP ahead of other traffic
Voice Quality & Delay

Many factors that contribute to the overall delay are fixed:
- Codec delay
- Hardware delay
- Processing delay
- Network physical delay

However, several delay factors are variable:
- Queuing delay
- Network propagation delay

It is the sum of all of these factors that determines overall delay as shown in the chart to the left.
VoIP Delay Example

End-to-End Delay Not to Exceed 250ms

Voice Router

IP or Frame Relay Network

Voice Router

Compression 20ms

Transmission .25 @ T1 7ms @ 56k

Network (FR) 20-40ms

Transmission .25 @ T1 7ms @ 56k

Decompression 10ms

Inter-process 10ms

Inter-process 10ms

Total Fixed Delays (w/o buffer) 71-129ms
The #1 Result of Excessive Delay - Jitter

- Occurs when packets do not arrive at a constant rate that exceeds the buffering ability of the receiving device to compensate for
  - Symptoms
    - Often noticed as garbles or an annoying screech during a conversation
  - Typical Causes
    - Insufficient bandwidth for the conversation
    - Excessive number of hops in the signal path
    - QoS disabled or not supported by one or more devices

Gateway → Internet → Gateway

VoIP Packets leave at constant intervals → VoIP Packets arrive at variable intervals
Customer Symptoms

• Customer Reported Symptoms
  – Cannot place or receive calls
  – Hear foreign voices not supposed to be on call
    • Cross-Talk
  – Volume noticeably low or high
  – Choppy Audio
  – Features do not work properly

• Equipment Alarm Indications
  – Ring Pre-trip Test Fails
  – Internal indications (card, power, etc)
  – Loss of Signal
  – High Error Rate
  – Connectivity failures
**VoIP Analysis Tip:** Wireshark has the ability to reconstruct not only VoIP conversations, but also other media streams for later analysis.
This example contains four (4) calls and is from a VoIP network using Cisco phones and SIP signaling with G.711 audio codec
VoIP Call Detection, Analysis and Playback
Thank You!