Troubleshooting Avaya SIP

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Vice President Strategy and Technology
SIP Troubleshooting – An Agenda

> SIP Protocol Overview
> Troubleshooting Tools
> Booting an Avaya SIP Phone
> Common Issues
SIP Methods

- INVITE
- ACK
- BYE
- CANCEL
- OPTIONS
- REGISTER
- PRACK
- NOTIFY
- SUBSCRIBE
- PUBLISH
- INFO
- REFER
- MESSAGE
- UPDATE
SIP Response Codes

> Status of server
> Success/failure
> 3-digit status code
  - 1st digit = class of response
  - Class categorized by provisional & final responses

<table>
<thead>
<tr>
<th>Class</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>1xx</td>
<td>Informational</td>
</tr>
<tr>
<td>2xx</td>
<td>Success</td>
</tr>
<tr>
<td>3xx</td>
<td>Redirection</td>
</tr>
<tr>
<td>4xx</td>
<td>Client Error</td>
</tr>
<tr>
<td>5xx</td>
<td>Server Error</td>
</tr>
<tr>
<td>6xx</td>
<td>Global Failure</td>
</tr>
</tbody>
</table>
Common Response Codes

> 100 Trying
  - Extended search being performed may take a significant time so a forking proxy must send a 100 Trying response

> 180 Ringing
  - Destination user agent received INVITE, and is alerting user of call.

> 183 Session in Progress
  - This response may be used to send extra information for a call which is still being set up.

> 200 OK
  - Indicates the request was successful.

> 401 Unauthorized
  - Request requires user authentication (often sent by UAS’s registrars)

> 404 Not Found
  - URI does not match any of the domains handled by the recipient

> 423 Interval Too Brief
  - Expiration of the resource is too short

> 500 Server Internal Error
  - Server could not fulfill the request due to some unexpected condition
SIP Messages / Methods

INVITE

> Creates new session
> Modifies existing session
Simple Call Flow

INVITE
180 Ringing
200 OK
ACK

INVITE
180 Ringing
200 OK
ACK

Registrar

Media
IMS Call Processing

1. IMSORIG
2. ORIGDONE
3. IMSTERM
4. TERMDONE
Example of IMSTERM
Resource Reservation

Unlike H.323 telephones, SIP telephones generate their own dial tone
However, Communication Manager still needs to be aware of off-hook condition
SIP Phone sends INVITE indicating it has gone off-hook
   > To and From are the same
   > There is no SDP
Communication Manager Publishes off-hook event
Off-Hook INVITE is replaced with new INVITE (re-INVITE) when dialing string complete
Off-Hook INVITE ended with 484 Address Incomplete
IMS processing begins
Resource Reservation
Resource Reservation

<table>
<thead>
<tr>
<th>No</th>
<th>Time</th>
<th>Source</th>
<th>Destination</th>
<th>Protocol</th>
<th>Length</th>
<th>Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>502</td>
<td>19:00:0100</td>
<td>135.9.28.76</td>
<td>135.11.180.173</td>
<td>SIP</td>
<td>405</td>
<td>Status: 100 TRYING</td>
</tr>
<tr>
<td>503</td>
<td>19:04:2481</td>
<td>135.11.180.173</td>
<td>135.9.28.76</td>
<td>TCP</td>
<td>0</td>
<td>-cap s sip [ACK] Seq=11371 Ack=9687 win=27248 Len=0</td>
</tr>
<tr>
<td>504</td>
<td>19:13:2470</td>
<td>135.9.28.76</td>
<td>135.11.180.173</td>
<td>SIP</td>
<td>494</td>
<td>Status: 481 Address Incomplete</td>
</tr>
<tr>
<td>505</td>
<td>19:13:2493</td>
<td>135.11.180.173</td>
<td>135.9.28.76</td>
<td>TCP</td>
<td>0</td>
<td>-cap s sip [ACK] Seq=11371 Ack=9127 win=29616 Len=0</td>
</tr>
</tbody>
</table>

- Frame 504: 494 bytes on wire (3952 bits), 494 bytes captured (3952 bits)
- Ethernet II, Src: Juniper_N1c:a9:00 (00:23:9c:1c:a9:00), Dst: Avaya_4e:70:af (00:1b:4f:4e:70:af)

- Session Initiation Protocol
  - Status-Line: SIP/2.0 484 Address Incomplete
  - Status-Code: 484
  - [Request Packet: False]

- Message Header
  - From: <sip:1430131@avaya.com>; tag=2d52534fe46011-603728c0_F1430131135.11.180.173
  - To: <sip:1430131@avaya.com; avaya-cm-fmu-off-hook>; tag=0e85ee3c1e4fbbff4e9400
  - Call-ID: 7_4fe46011-c419bd79-60373d39_1E3135.11.180.173
  - CSeq: 8 INVITE
  - Via: SIP/2.0/TCP 135.11.180.173;branch=9h54b4b8_4fe46011-220f59ce-60373bd6_11430131
  - Server: Avaya CM/R016X.02.0.623.0 Avaya—SM.6.2.2.0.622005
  - Content-Length: 0
Three Bidirectional Flows

SIP Requests + SDP in TCP/UDP

Media (voice, video, etc.) – RTP in UDP

Control Messages – RTCP in UDP

SIP Responses + SDP in TCP/UDP

Media (voice, video, etc.) – RTP in UDP

Control Messages – RTCP in UDP
Making It Secure

- Transport Layer Security (TLS)
  - Encryption of the SIP signaling
  - Think HTTPS
- Secure Real Time Protocol (SRTP)
  - Encryption of the media stream
- Secure Real Time Control Protocol (SRTCP)
  - Encrypted RTP statistics and control information
Plus...

Message Level Security
  > 401 Unauthorized
    – REGISTER
  > 407 Proxy Authentication Required
    – Just about everything else

Nonce (Number Once)
WWW-Authenticate: Digest realm="atlanta.example.com", qop="auth", nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque=""", stale=FALSE, algorithm=MD5
Authentication Challenge

WWW-Authenticate: Digest realm="atlanta.example.com", qop="auth", nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="", stale=FALSE, algorithm=MD5
Authentication Response

Authorization: Digest username="bob", realm="atlanta.example.com" nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="" uri="sips:ss2.biloxi.example.com", response="dfe56131d1958046689d83306477ecc"
Authentication Response

Authorization: Digest username="bob", realm="atlanta.example.com"
nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="",
uri="sips:ss2.biloxi.example.com",
response="dfe56131d1958046689d83306477ecc"
Authentication Response

SMs challenge for authentication

Send second REGISTER Requests, this time with authentication
SIP Subscriptions

SUBSCRIBE
  > Create a relationship between a client and server

PUBLISH
  > Inform subscription server in a change of state / event

NOTIFY
  > Inform all subscribed entities of state change / event
Presence SUBSCRIBE

Presence Server

SUBSCRIBE to James

Louis

Evan

James
Presence PUBLISH
Presence NOTIFY

Presence Server

NOTIFY James is Off-Hook

Louis

Evan

James
AST Feature Subscriptions

An Avaya AST device phone subscribes to the following packages on the primary SM:

- **avaya-cm-feature-status**
  - "Phone features" like SAC, call-fwd, etc.
- **avaya-ccs-profile**
  - Used for reloading configuration, button and contact changes, etc.
- **Dialog**
  - Line appearance state
- **message-summary**
  - Message waiting
- **Reg**
  - Length of subscription, type of registration, reg-id=1 and reg-id=2 with controller addresses, phone's Address of Record (AOR)
AST Feature Subscriptions

You can check phone’s subscriptions in SMGR after registration and subscribes succeed.
AST Feature Subscriptions

You can check phone’s subscriptions in SMGR after registration and subscribes succeed.
AST Feature Subscriptions

You can see the packages being subscribed to with Session Manager’s traceSM
SIP Messages / Methods

REGISTER

> Create a binding between a SIP URI and one or more “Contact”
> Used for initial registration as well as refresh
> Typically challenged with a “401 Unauthorized”
# Multiple Registrations

## Communication Profile

<table>
<thead>
<tr>
<th>Name</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
<td></td>
</tr>
</tbody>
</table>

* Name: Primary
* Default: 

## Communication Address

<table>
<thead>
<tr>
<th>Type</th>
<th>Handle</th>
<th>Domain</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya SFP</td>
<td>4902</td>
<td>training.com</td>
</tr>
</tbody>
</table>

## Session Manager Profile

<table>
<thead>
<tr>
<th>Primary Session Manager</th>
<th>Primary</th>
<th>Secondary</th>
<th>Maximum</th>
</tr>
</thead>
<tbody>
<tr>
<td>CM4</td>
<td>2</td>
<td>0</td>
<td>2</td>
</tr>
</tbody>
</table>

## Secondary Session Manager

<table>
<thead>
<tr>
<th>Origination Application Sequence</th>
<th>Primary</th>
<th>Secondary</th>
<th>Maximum</th>
</tr>
</thead>
<tbody>
<tr>
<td>CM4</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Termination Application Sequence</th>
<th>Primary</th>
<th>Secondary</th>
<th>Maximum</th>
</tr>
</thead>
<tbody>
<tr>
<td>CM4</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Survivability Server</th>
<th>Classroom</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Home Location</th>
<th>Classroom</th>
</tr>
</thead>
</table>
SIP Troubleshooting – An Agenda

> Troubleshooting Tools
  – Avaya Aura System Manager
  – Wireshark
  – traceSM
  – List trace station xxxxxxx/s

> Avaya SIP Phone Boot Sequence

> Common Issues
Avaya Aura System Manager
Avaya Aura System Manager
## Avaya Aura System Manager

### Managed Bandwidth Usage

This page displays system-wide bandwidth usage information for Locations.

<table>
<thead>
<tr>
<th>Details</th>
<th>Location</th>
<th>Audio Call Count</th>
<th>Audio BW Used</th>
<th>Multimedia Call Count</th>
<th>Multimedia BW Used</th>
<th>Multimedia Allow</th>
<th>Multimedia BW Used</th>
<th>Total Usage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Show A</td>
<td>TRM-BSW</td>
<td>5</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>0</td>
</tr>
<tr>
<td>Show B</td>
<td>MN-SBC</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>0</td>
</tr>
<tr>
<td>Show C</td>
<td>MN-SMC</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>0</td>
</tr>
<tr>
<td>Show D</td>
<td>MN-WC</td>
<td>11</td>
<td>287</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>287</td>
</tr>
<tr>
<td>Show E</td>
<td>MNC-MNC</td>
<td>6</td>
<td>492</td>
<td>0</td>
<td>2,430</td>
<td>4%</td>
<td>496</td>
<td>0</td>
</tr>
<tr>
<td>Show F</td>
<td>MN-CCN</td>
<td>23</td>
<td>955</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>955</td>
</tr>
<tr>
<td>Show G</td>
<td>MN-PCC</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>0</td>
</tr>
<tr>
<td>Show H</td>
<td>MN-WD</td>
<td>1</td>
<td>83</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>83</td>
</tr>
<tr>
<td>Show I</td>
<td>MNC-MNC</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>0</td>
</tr>
<tr>
<td>Show J</td>
<td>MN-SMC</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>0</td>
</tr>
<tr>
<td>Show K</td>
<td>MN-SP</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>0</td>
</tr>
<tr>
<td>Show L</td>
<td>MN-PRES</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>0</td>
</tr>
<tr>
<td>Show M</td>
<td>MN-AAC</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>No Limit</td>
<td>N/A</td>
<td>N/A</td>
<td>0</td>
</tr>
</tbody>
</table>

*Note: The bandwidth usage is displayed in GB/sec.*
Avaya Aura System Manager
### Wireshark

#### Debugging SP and PPM on 9841 PCAP

**Wireshark 1.8.3 (SVN-Rev 42556 from trunk-1.8)**

<table>
<thead>
<tr>
<th>Time</th>
<th>Duration</th>
<th>Source</th>
<th>Destination</th>
<th>Protocol</th>
<th>Length</th>
<th>Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>207:19.289</td>
<td>135.9.180.173</td>
<td>135.9.120.118</td>
<td>SIP</td>
<td>1122 request: REGISTER sip:<a href="mailto:1430138@avaya.com">1430138@avaya.com</a></td>
<td></td>
<td></td>
</tr>
<tr>
<td>208:19.298</td>
<td>135.9.180.173</td>
<td>135.9.28.76</td>
<td>SIP</td>
<td>1122 Request: REGISTER sip:<a href="mailto:1430138@avaya.com">1430138@avaya.com</a></td>
<td></td>
<td></td>
</tr>
<tr>
<td>272.19.381</td>
<td>135.9.180.173</td>
<td>135.9.126.76</td>
<td>TCP</td>
<td>6127 &gt; sip [ACK] Seq=1894 Ack=42556 Win=8272 Len=0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>273.19.381</td>
<td>135.9.180.173</td>
<td>135.9.126.76</td>
<td>SIP</td>
<td>668 Request: SUBSCRIBE sip:<a href="mailto:1430138@avaya.com">1430138@avaya.com</a></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### Frame 267: 1121 bytes on wire (9068 bits). 1121 bytes captured (9068 bits)

**Ethernet II**, Src: Avaya_4e:78:2a:0f (00:0b:1f:78:2a:0f), Dst:juniper_1.ca:b3:9c:00 (00:0b:1f:78:2a:0f)


**Session Initiation Protocol**

- Request-URI: REGISTER sip:1430138@avaya.com sip:2.0
- Message-header:
  - Contact: sip:1430138@avaya.com
  - To: sip:1430138@avaya.com
  - Call-ID: 1f2ed45f4abc50c7a-60374000@1430138.135.9.180.173
- Code: 2 REGISTER
- Max-Forwards: 70
- Via: sip/2.0/TCP 135.9.180.173:5663;branch=z9hG4bK55f0169-233a-6037400a-8440131
- Supported: event-list,feature-ref, replaces
- Allow: invite, ack, cancel, subscribe, note, message, refer, info, publish, update
- User-Agent: Avaya one-X deskphone 6.2.0.6 (37452)

**[Truncated]** Contact: sip:1430138@avaya.com;branch=z9hG4bK55f0169-233a-6037400a-8440131
**[Truncated]** Authorization: Digest realm=avaya.com,nonce=18a4f20da97a7c7809a17e757c76728360970cc, url="sip:avaya.com", opaque

**[Truncated]** Content-Length: 0
Wireshark

Request file "Upgrade.txt" file from HTTP server
Wireshark

Right click on entry, select “Follow TCP Stream” to see file contents
Wireshark

“Follow TCP Stream” window displays 96x1Upgrade.txt contents
Configure phone to use TCP instead of TLS

Phone normally must use secure HTTP with PPM (HTTPS) to protect user identity information, system parameters, etc., so trace won’t show that detail.

To see the SOAP message body detail in traces:

- Set the 46xxsettings.txt directive, CONFIG_SERVER_SECURE_MODE, and reboot phone:
  - set CONFIG_SERVER_SECURE_MODE 0 (for HTTP)
  - set CONFIG_SERVER_SECURE_MODE 1 (for HTTPS)
  - set CONFIG_SERVER_SECURE_MODE 2 (HTTP if TCP, HTTPS if TLS)
- Configure the PPM to use HTTP via Session Manager Administration in SMGR:
  - Home/Elements/Session Manager/Session Manager Administration
  - Allow Unsecured PPM Traffic checkbox
Session Manager’s TraceSM
Session Manager’s TraceSM
Session Manager’s TraceSM
### Communication Manager’s “list trace station xxxx/s”

<table>
<thead>
<tr>
<th>time</th>
<th>data</th>
</tr>
</thead>
<tbody>
<tr>
<td>15:20:18</td>
<td>TRACE STARTED 03/27/2014 CM Release String cold-00.0.124.0-21327</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP&lt;INVITE sip:<a href="mailto:4563507@arrows3.com">4563507@arrows3.com</a>;avaya-cm-fnu=off-hook S1</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP&lt;P/2.0</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP From: <a href="">sip:4563507@arrows3.com</a>;tag=-4cc1a1475334808-5</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP To: <a href="">sip:4563507@arrows3.com;avaya-cm-fnu=off-hook</a></td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP From: <a href="">sip:4563507@arrows3.com</a>;tag=-4cc1a1475334808-5</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP To: <a href="">sip:4563507@arrows3.com;avaya-cm-fnu=off-hook</a>;tag</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP</td>
</tr>
<tr>
<td>15:20:24</td>
<td>SIP</td>
</tr>
</tbody>
</table>

press CANCEL to quit -- press NEXT PAGE to continue
SIP Phone Bootup

> Power on and initialize internal hardware, software.
> Set LAN speed and begin talking to DHCP to acquire an HTTP ip address (and more).
> LLDP, if present in network.
  > A 9600 Series IP telephone initiates LLDP after receiving an LLDPDU (Link Layer Discovery Protocol Data Unit) message from an appropriate system. Once initiated, the telephones send an LLDPDU every 30 seconds
> HTTP –check with fileserver and download software, if necessary, based on contents of the 96xxupgrade.txt file – then move onto extracting configuration from 46xxsettings file.
> Begin registration process with a SIP registrar server (SM, BSM).
> May ask for login and password (or use cached values).
> Server downloads data to phone i.e. button config, dial plan, PPM parameters, etc.
Common Issues – SDP and Codec Negotiation

The phone is told, via the 46xxsettingsfile, what codecs it should advertise in its SDP.
Common Issues – SDP and Codec Negotiation
Common Issues – SDP and Codec Negotiation

CM will overwrite the SDP based on the ip-codec-set assigned to the network region.

<table>
<thead>
<tr>
<th>Audio Codec</th>
<th>Silence Suppression</th>
<th>Frames Per Pkt</th>
<th>Packet Size (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711MU</td>
<td>n</td>
<td>2</td>
<td>20</td>
</tr>
<tr>
<td>G.729</td>
<td>n</td>
<td>2</td>
<td>20</td>
</tr>
</tbody>
</table>
Common Issues – SDP and Codec Negotiation
Common Issues – Can’t Register Station

> Verify that user/station is administered in System Manager
> Verify credentials (ie Username/Password)
> Verify that the endpoint has the correct signaling type, registration IP address, and SIP Domain
> If attempting to connect station through SBC-E, verify “Endpoint Flow” is configured correctly.
Common Issues – Registers, But No Feature buttons

> Check for AST/PPM Feature Pack Subscriptions (in SMGR) or watch for the SUBSCRIBE in traceSM

> Restarting Phone often fixes this
Logs in ok, but...

The fact that the client can log in means that the SMGR user profile exists.
Logs in ok, but…

If they can’t place or receive calls, check the Communication Manager and Session Manager Profiles.
Logs in ok, but…

If they can place calls, but can’t receive calls, verify that the Off-Premise Station Mapping has the correct OPS and trunk administered
Thank You!