



## **Syslog Analysis**

**AudioCodes Academy**

<https://www.audiocodes.com/services-support/audiocodes-academy>

After completing this lesson you will be able to:

- Interpret information collected using the ACsyslog tool

- Syslog is a standard for collecting **log messages** within an IP network.
- A Syslog server can be used to **remotely record** log information.
  - Sent to a central server using UDP port 514
  - Ensure this port is open in your firewall if central server is outside the local network
- Syslog information that is sent by the gateway is a collection of error, warning, and system messages that can record almost every internal operation of the gateway.
  - Syslog protocol defined in the IETF RFC 3164

- When reporting a problem to AudioCodes support, the following information should be provided:
  - **Network setup** (such as network diagram, call direction)
  - **Board.ini file** (renamed appropriately for your site)
  - **Unfiltered Syslog trace** (ACsyslog best tool to use with AudioCodes products)
  - **Unfiltered WireShark files**
  - **Advanced Debug Recordings (per AudioCodes request):**
    - Examples:
      - PSTN signaling problems
      - Problems related to voice quality, modem, fax, DTMF detection

To save the .ini file:

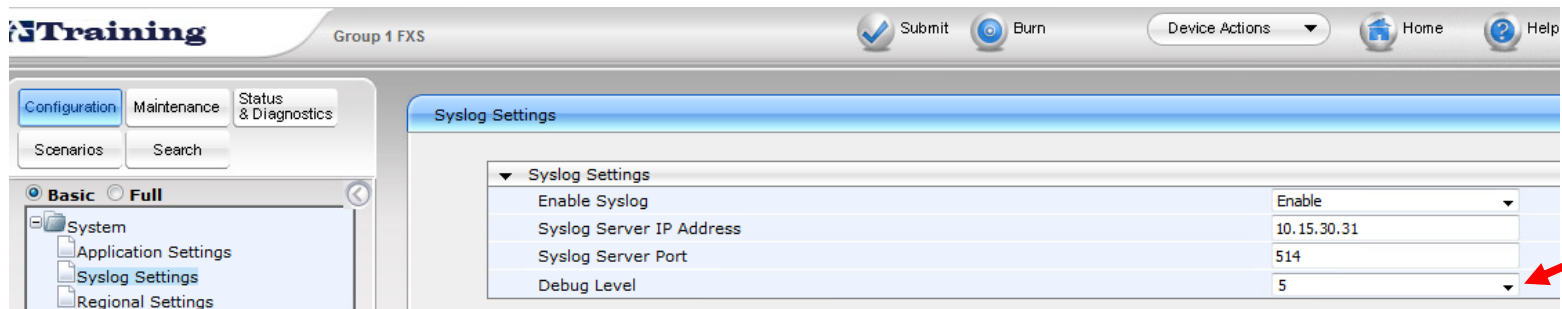
1. Open the 'Configuration File' page (Management tab > Software Update menu > Configuration File).
2. To save the ini file to a folder on your PC:
  - a. Click the **Save INI File** button; the 'File Download' dialog box appears.
  - b. Click the **Save** button, navigate to the folder in which you want to save the ini file on your PC, and then click **Save**; the device copies the ini file to the selected folder.

You can use the supplied proprietary Syslog server ACSyslog or any other third-party Syslog server for receiving Syslog messages.

- A typical Syslog server application enables filtering of messages according to:
  - Priority
  - IP sender address
  - Time
  - Date

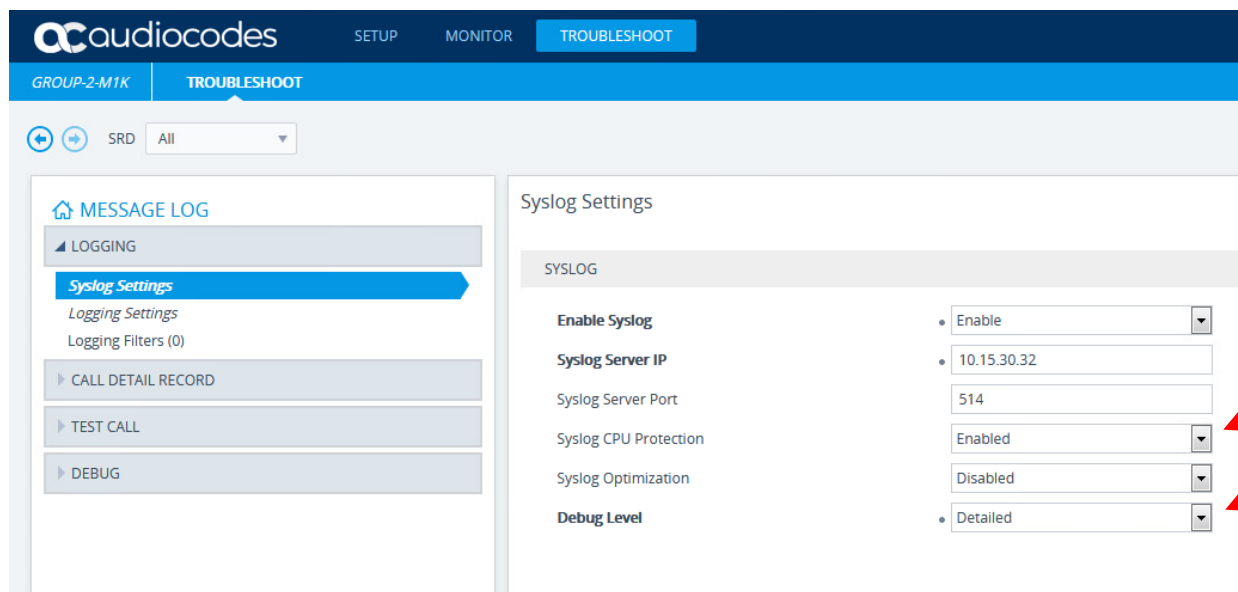
- 0 = Disabled (default)
- 1 = Flow
- 5 = Flow, device interface, stack interface, session manager and expanded device interface
- 7 = automatically changes between level 5, level 1 and level 0, depending on the device CPU consumption

**Note: AudioCodes support REQUIRES level 5 logs be captured for support requests**



- NoDebug = Disabled (default)
- Basic = Flow
- Detailed = Flow, device interface, stack interface, session manager and expanded device interface
- Syslog CPU Protection - Enabled = automatically changes between level 5, level 1 and level 0, depending on the device CPU consumption

**Note: AudioCodes support REQUIRES Detailed logs be captured for support requests**



The screenshot displays the AudioCodes web interface for troubleshooting. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The 'TROUBLESHOOT' tab is active, showing a 'GROUP-2-M1K' and 'TROUBLESHOOT' sub-tab. Below this, there are filters for 'SRD' and 'All'. The left sidebar contains a 'MESSAGE LOG' section with options for 'LOGGING', 'Syslog Settings' (highlighted), 'Logging Settings', 'Logging Filters (0)', 'CALL DETAIL RECORD', 'TEST CALL', and 'DEBUG'. The main content area is titled 'Syslog Settings' and contains a 'SYSLOG' section with the following settings:

Setting	Value
Enable Syslog	Enable
Syslog Server IP	10.15.30.32
Syslog Server Port	514
Syslog CPU Protection	Enabled
Syslog Optimization	Disabled
Debug Level	Detailed

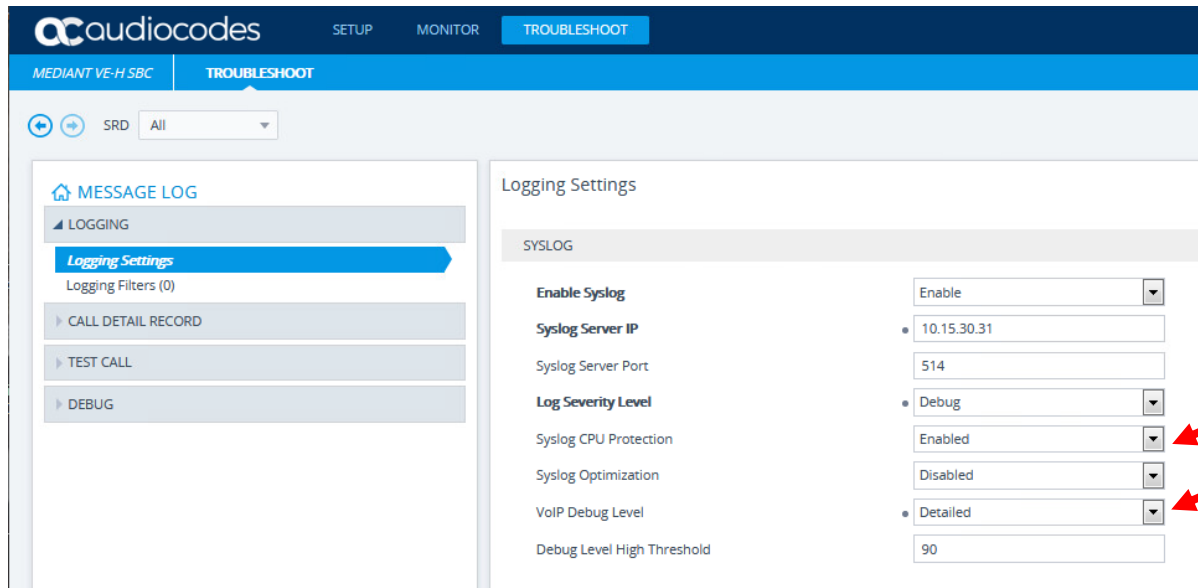
Two red arrows point to the 'Syslog CPU Protection' and 'Debug Level' dropdown menus, indicating they are critical for support requests.



## VoIP Debug Level replaces Debug Level for Syslogs

- NoDebug = Disabled (default)
- Basic = Flow
- Detailed = Flow, device interface, stack interface, session manager and expanded device interface
- Syslog CPU Protection - Enabled = automatically changes between level 5, level 1 and level 0, depending on the device CPU consumption

**Note: AudioCodes support REQUIRES Detailed logs be captured for support requests**



The screenshot shows the AudioCodes Troubleshoot interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The left sidebar shows a 'MESSAGE LOG' section with options for 'LOGGING', 'CALL DETAIL RECORD', 'TEST CALL', and 'DEBUG'. The 'LOGGING' section is expanded, showing 'Logging Settings' and 'Logging Filters (0)'. The 'Logging Settings' section is further expanded, showing 'SYSLOG' settings. The 'SYSLOG' settings include:

- Enable Syslog: Enable
- Syslog Server IP: 10.15.30.31
- Syslog Server Port: 514
- Log Severity Level: Debug
- Syslog CPU Protection: Enabled
- Syslog Optimization: Disabled
- VoIP Debug Level: Detailed
- Debug Level High Threshold: 90

Two red arrows point to the 'Log Severity Level' and 'VoIP Debug Level' dropdown menus, highlighting the 'Debug' and 'Detailed' options respectively.

# Online Syslogs via RS-232 and Telnet via CMD SHELL

- Once syslog is enabled, the logs are visible in both the console and telnet sessions
- Enable SSH or telnet on the gateway and log in using Putty or similar tool
- The commands to start and stop collecting logs are:
  - show log start
  - show log stop
- Log the session to a text file to capture the full trace on your local PC

```
SIP/ SEcurity/ DebugRecording/ Mgmt/ ControlProtocol/ CONFIguration/ IPNetworking/ TPApP/ BSP/
PING SHOW
/>show

SHow - SH
SHow (SH) - Display operational statistics.
Usage:
  SHow INFO           Displays general device information
  SHow DSP            Displays DSP resource information
  SHow IP             Displays information about IP interfaces
  SHow VOICEPROMPT    Displays information about Voice Prompt tones
  SHow LOG            Displays syslog data
  SHow LOG [START|STOP] Displays (or stops displaying) syslog
  SHow TONES          Displays information about special tones

SIP/ SEcurity/ DebugRecording/ Mgmt/ ControlProtocol/ CONFIguration/ IPNetworking/ TPApP/ BSP/
PING SHOW
/>
```


```
SIP/ SEcurity/ DebugRecording/ Mgmt/ ControlProtocol/ CONFIguration/ IPNetworking/ TPApP/ BSP/
PING SHOW
/>show log start

Started syslog display to telnet CLI.

SIP/ SEcurity/ DebugRecording/ Mgmt/ ControlProtocol/ CONFIguration/ IPNetworking/ TPApP/ BSP/
PING SHOW
/>NOTIC:( lgr_psbrdex)(1831) ) recv <-- OFF_HOOK Ch:0
NOTIC:( lgr_flow)(1832) ) #0:OFF_HOOK_EV
NOTIC:( lgr_flow)(1833) ) | #0:OFF_HOOK_EV State:IDLE Substate:sub_None
NOTIC:( lgr_psbrdif)(1834) ) UpdateChannelParams, Channel 0

NOTIC:( lgr_psbrdif)(1835) ) #0:ConfigFaxModemChannelParams NSEMode=0, CNGDetMode=0, FAXTranType=1, VxxTranType=2
, VoiceVol= 0, DTMFVol=-11, InGain=0, RlPRedDepth=0, ECE=1,ECETType=0 SCE=0, ECNlpMode=0, DJBufMinDelay=10, DJBufOptFac=1
0, Result=1)
NOTIC:( lgr_psbrdif)(1836) ) ActivateDigitMap for channel : 0, MaxDialStringLength = 7, MaxEndDialTimer = 4000,
MaxLongInterDigitTimer = 8000, MaxStartTimer = 16000, DigitMap = [0-9*#ABCD][0-9ABCD].T, DPIndex = -1, DPPriority = 0
NOTIC:( lgr_flow)(1837) ) #-100: StartDigitMapDetection with params:
<Pattern=[0-9*#ABCD][0-9ABCD].T>
<MaxStartTimer=16000>
<SendEachDigit=1>
<UseEndDialKey=0>
<MaxLongInterDigitTimer=8000>
```

- Once syslog is enabled, the logs are visible in both the console and telnet sessions
- Enable SSH or telnet on the gateway and log in using Putty or similar tool
- The commands to start collecting logs is:
  - debug log
  - debug log full
- Log the session to a text file to capture the full trace on your local PC

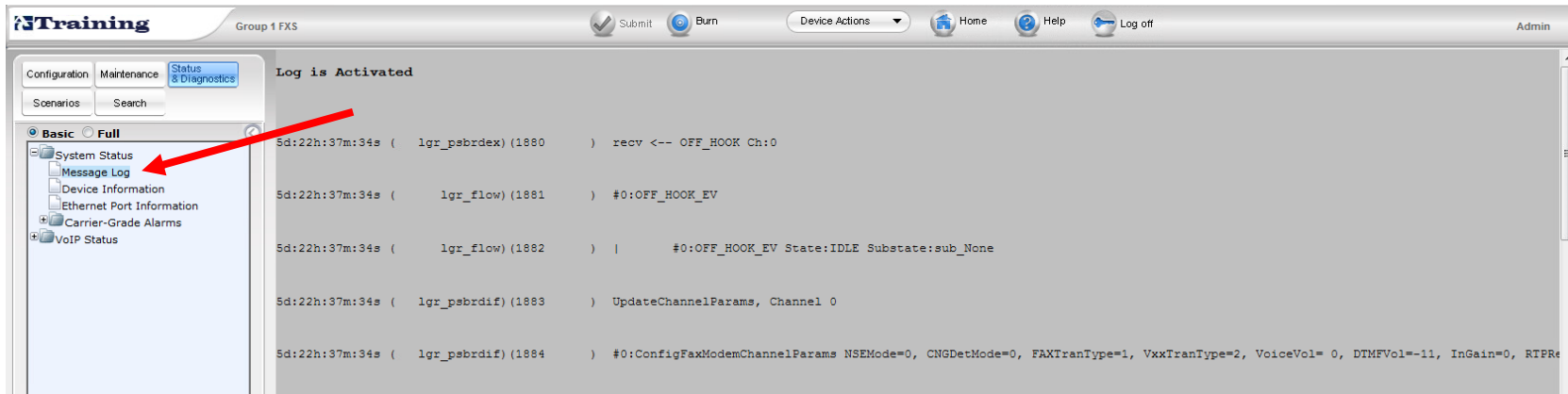


```
Group 1 FXS# debug log full
Logging started

Group 1 FXS# Jul 12 17:30:40 Restart Request Trunk 0 [Time:12-07@17:30:39.968]
Jul 12 17:30:40 Bch:-1 [Time:12-07@17:30:39.968]
Jul 12 17:30:40 LOCAL_RESTART_CONFIRM_EV on trunk:0 BChannel:-1 [Time:12-07@17:30:39.977]
Jul 12 17:30:41 RAISE-ALARM:acBoardEthernetLinkAlarm: Ethernet link alarm. LAN port number 1 is down.; Severity:minor;
Source:Board#1/EthernetLink#1; Unique ID:3; Additional Info1:GigabitEthernet 0/1; [Time:12-07@17:30:40.459]
Jul 12 17:30:41 RAISE-ALARM:acBoardEthernetLinkAlarm: Ethernet link alarm. LAN port number 3 is down.; Severity:minor;
Source:Board#1/EthernetLink#3; Unique ID:4; Additional Info1:GigabitEthernet 0/3; [Time:12-07@17:30:40.459]
Jul 12 17:30:41 RAISE-ALARM:acBoardEthernetLinkAlarm: Ethernet link alarm. LAN port number 4 is down.; Severity:minor;
Source:Board#1/EthernetLink#4; Unique ID:5; [Time:12-07@17:30:40.460]
Jul 12 17:30:41 RAISE-ALARM:acBoardEthernetLinkAlarm: Ethernet link alarm. LAN port number 5 is down.; Severity:minor;
Source:Board#1/EthernetLink#5; Unique ID:6; [Time:12-07@17:30:40.460]
Jul 12 17:30:41 RAISE-ALARM:acBoardEthernetLinkAlarm: Ethernet link alarm. LAN port number 6 is down.; Severity:minor;
Source:Board#1/EthernetLink#6; Unique ID:7; [Time:12-07@17:30:40.461]
Jul 12 17:30:42 TRUNK_RESTART_COMPLETE on trunk:0 BChannel:-1 Operation:3 [Time:12-07@17:30:41.977]
Jul 12 17:30:42 Trunk:0 Notify channels for in service. [Time:12-07@17:30:41.977]
Jul 12 17:48:11 recv <-- OFF_HOOK Ch:0 [Time:12-07@17:48:11.023]
Jul 12 17:48:11 #0:OFF_HOOK_EV [Time:12-07@17:48:11.024]
Jul 12 17:48:11 #0:OFF_HOOK_EV State:IDLE Substate:sub_None [Time:12-07@17:48:11.024]
Jul 12 17:48:11 UpdateChannelParams, Channel 0
[Time:12-07@17:48:11.025]
Jul 12 17:48:11 #0:ConfigFaxModemChannelParams NSEMode=0, CNGDetMode=0, FAXTranType=1, VxxTranType=2, VoiceVol= 0, DTMF
```

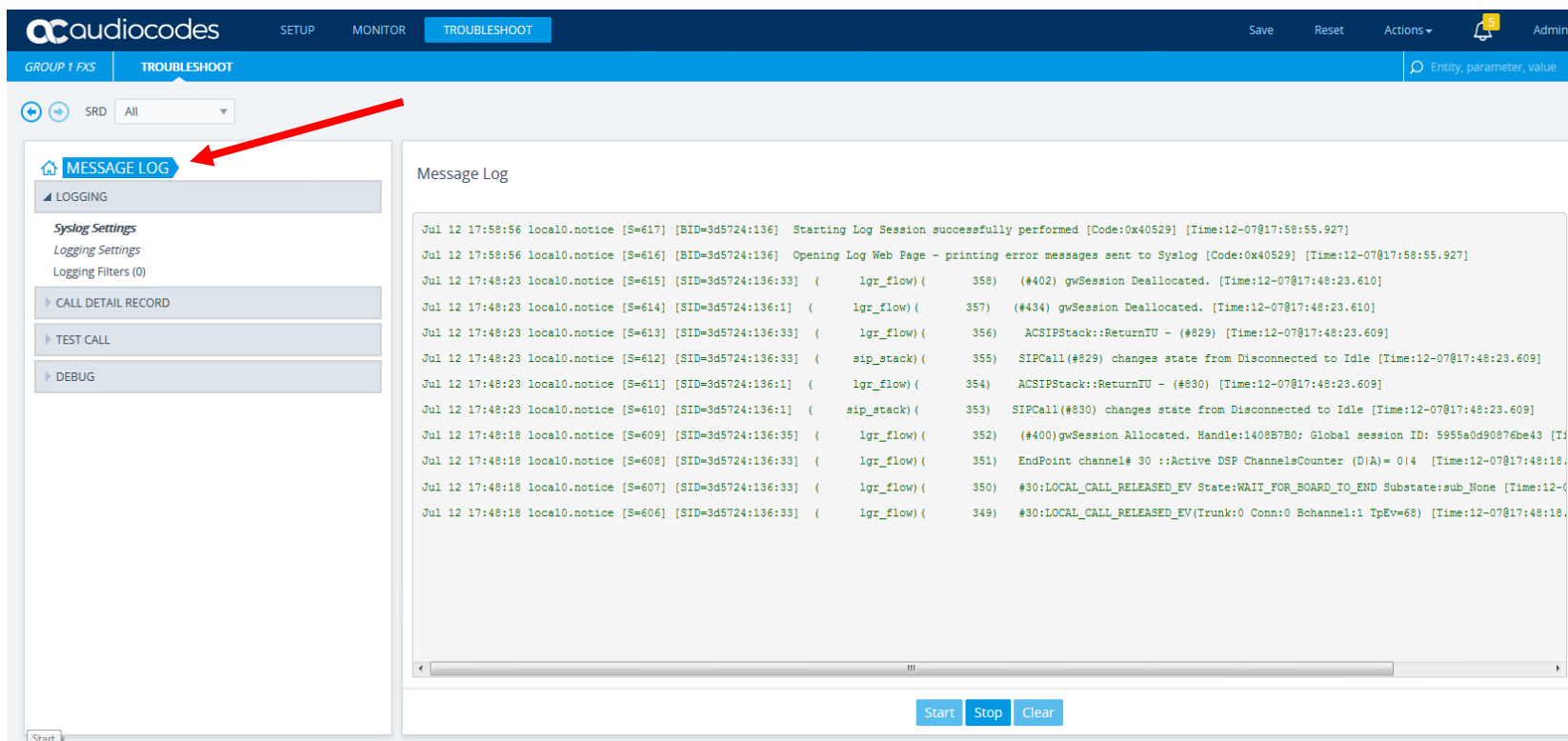
# Online Syslogs via Web Browser Message Log MP 11X MP 124

- Once syslog is enabled, the logs are visible in both the console and telnet sessions
- Access **Status & Diagnostics -> System Status -> Message Log**
  - Logs temporarily stored in PC/Macs RAM



# Online Syslogs via Web Browser Message Log Mediants MP-1288

- Once syslog is enabled, the logs are visible in both the console and telnet sessions
- Access **Troubleshoot -> Troubleshoot -> MESSAGE LOG**
  - Logs temporarily stored in PC/Macs RAM
  - Start/Stop/Clear used to prevent syslogs from eating all of the PC/Macs RAM



The screenshot displays the Audiocodes web interface for troubleshooting. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The 'TROUBLESHOOT' section is active, showing a search bar and a list of options: 'MESSAGE LOG', 'LOGGING', 'CALL DETAIL RECORD', 'TEST CALL', and 'DEBUG'. A red arrow points to the 'MESSAGE LOG' button. The 'MESSAGE LOG' section is expanded, showing a list of log entries. The log entries are displayed in a table format with columns for time, source, severity, and message. The messages include details about log session performance, error messages, and SIP call state changes.

Message Log

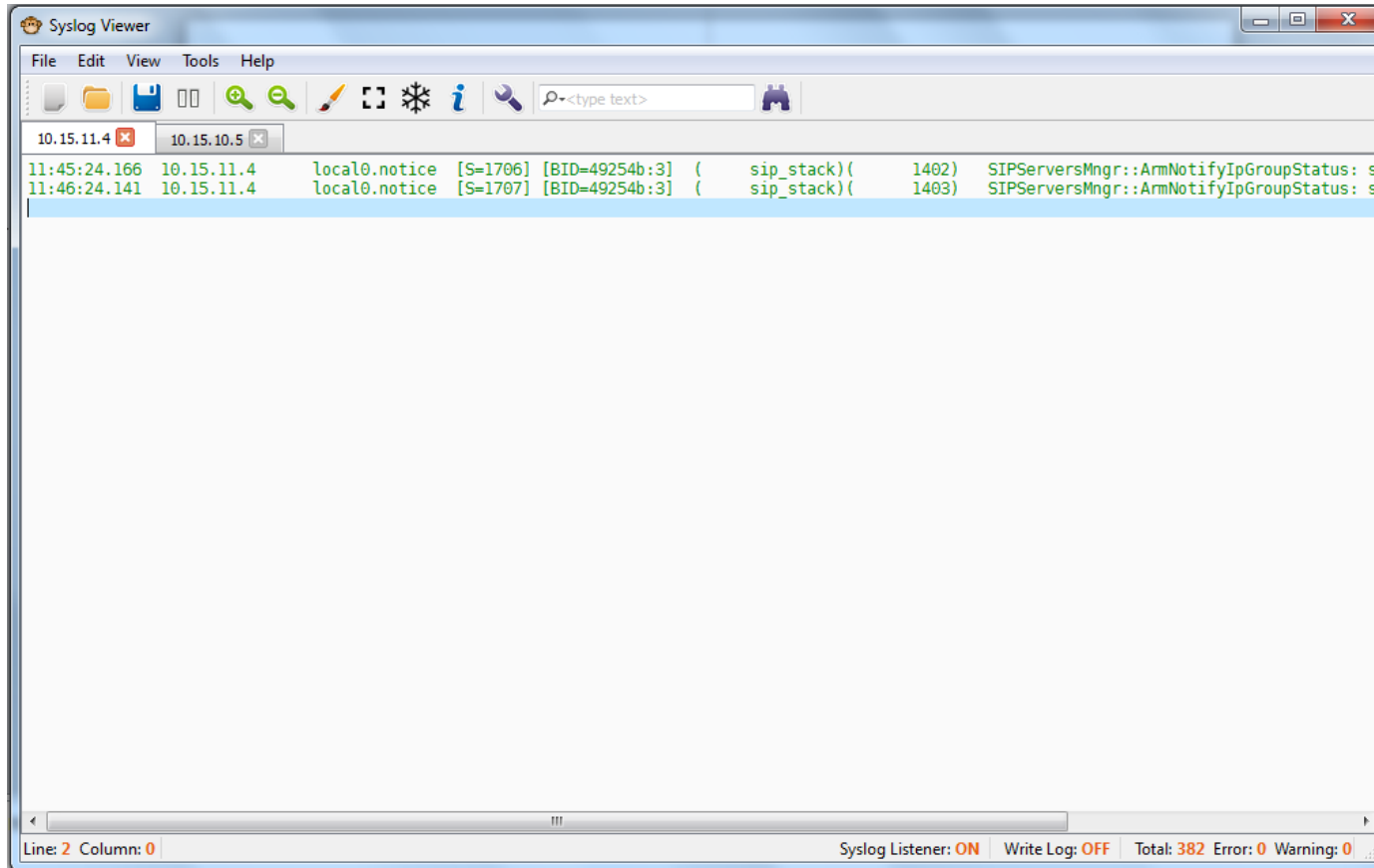
```
Jul 12 17:58:56 local0.notice [S=617] [SID=3d5724:136] Starting Log Session successfully performed [Code:0x40529] [Time:12-07@17:58:55.927]
Jul 12 17:58:56 local0.notice [S=616] [SID=3d5724:136] Opening Log Web Page - printing error messages sent to Syslog [Code:0x40529] [Time:12-07@17:58:55.927]
Jul 12 17:48:23 local0.notice [S=615] [SID=3d5724:136:33] ( lgr_flow) ( 358) (#402) gwSession Deallocated. [Time:12-07@17:48:23.610]
Jul 12 17:48:23 local0.notice [S=614] [SID=3d5724:136:1] ( lgr_flow) ( 357) (#434) gwSession Deallocated. [Time:12-07@17:48:23.610]
Jul 12 17:48:23 local0.notice [S=613] [SID=3d5724:136:33] ( lgr_flow) ( 356) ACSIPStack::ReturnTU - (#829) [Time:12-07@17:48:23.609]
Jul 12 17:48:23 local0.notice [S=612] [SID=3d5724:136:33] ( sip_stack) ( 355) SIPCall(#829) changes state from Disconnected to Idle [Time:12-07@17:48:23.609]
Jul 12 17:48:23 local0.notice [S=611] [SID=3d5724:136:1] ( lgr_flow) ( 354) ACSIPStack::ReturnTU - (#830) [Time:12-07@17:48:23.609]
Jul 12 17:48:23 local0.notice [S=610] [SID=3d5724:136:1] ( sip_stack) ( 353) SIPCall(#830) changes state from Disconnected to Idle [Time:12-07@17:48:23.609]
Jul 12 17:48:18 local0.notice [S=609] [SID=3d5724:136:35] ( lgr_flow) ( 352) (#400)gwSession Allocated. Handle:1408B7B0; Global session ID: 5955a0d90876be43 [Time:12-07@17:48:18.609]
Jul 12 17:48:18 local0.notice [S=608] [SID=3d5724:136:33] ( lgr_flow) ( 351) EndPoint channel# 30 ::Active DSP ChannelsCounter (DIA)= 014 [Time:12-07@17:48:18.609]
Jul 12 17:48:18 local0.notice [S=607] [SID=3d5724:136:33] ( lgr_flow) ( 350) #30:LOCAL_CALL_RELEASED_EV State:WAIT_FOR_BOARD_TO_END Substate:sub_None [Time:12-07@17:48:18.609]
Jul 12 17:48:18 local0.notice [S=606] [SID=3d5724:136:33] ( lgr_flow) ( 349) #30:LOCAL_CALL_RELEASED_EV(Trunk:0 Conn:0 Bchannel:1 TpEv=68) [Time:12-07@17:48:18.609]
```

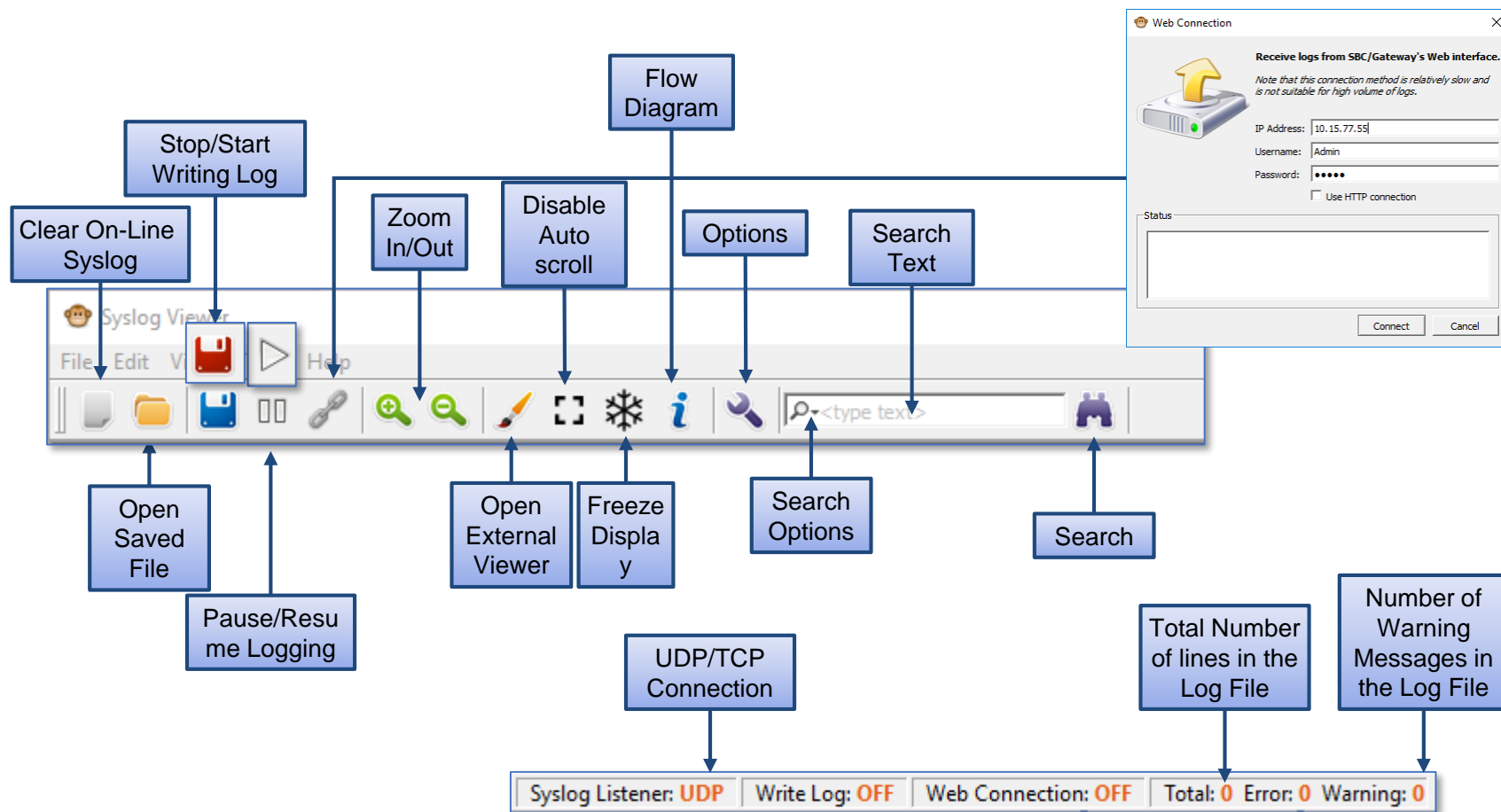
Start Stop Clear

- ACSyslog is a free utility to capture AudioCodes Syslogs
- Installation required
- Requests updates when started each time
- Use on as many workstations as you want
- Contains many options and configuration parameters making it a robust tool for capturing debug logs



- A newer Syslog application provided with the student utilities kit



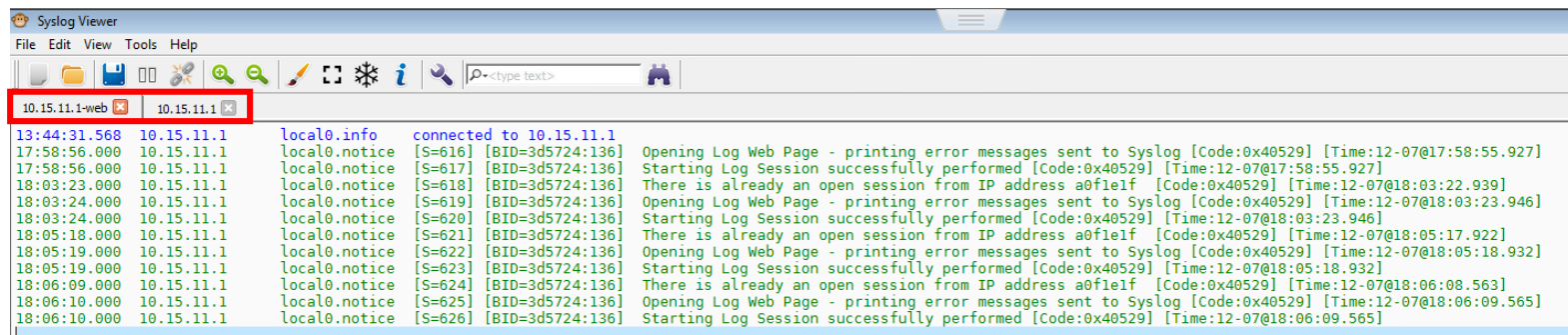
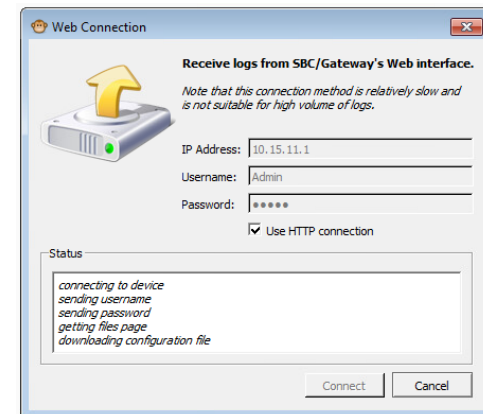
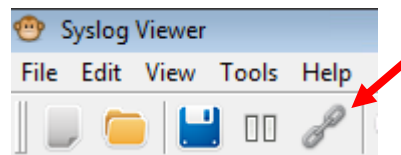




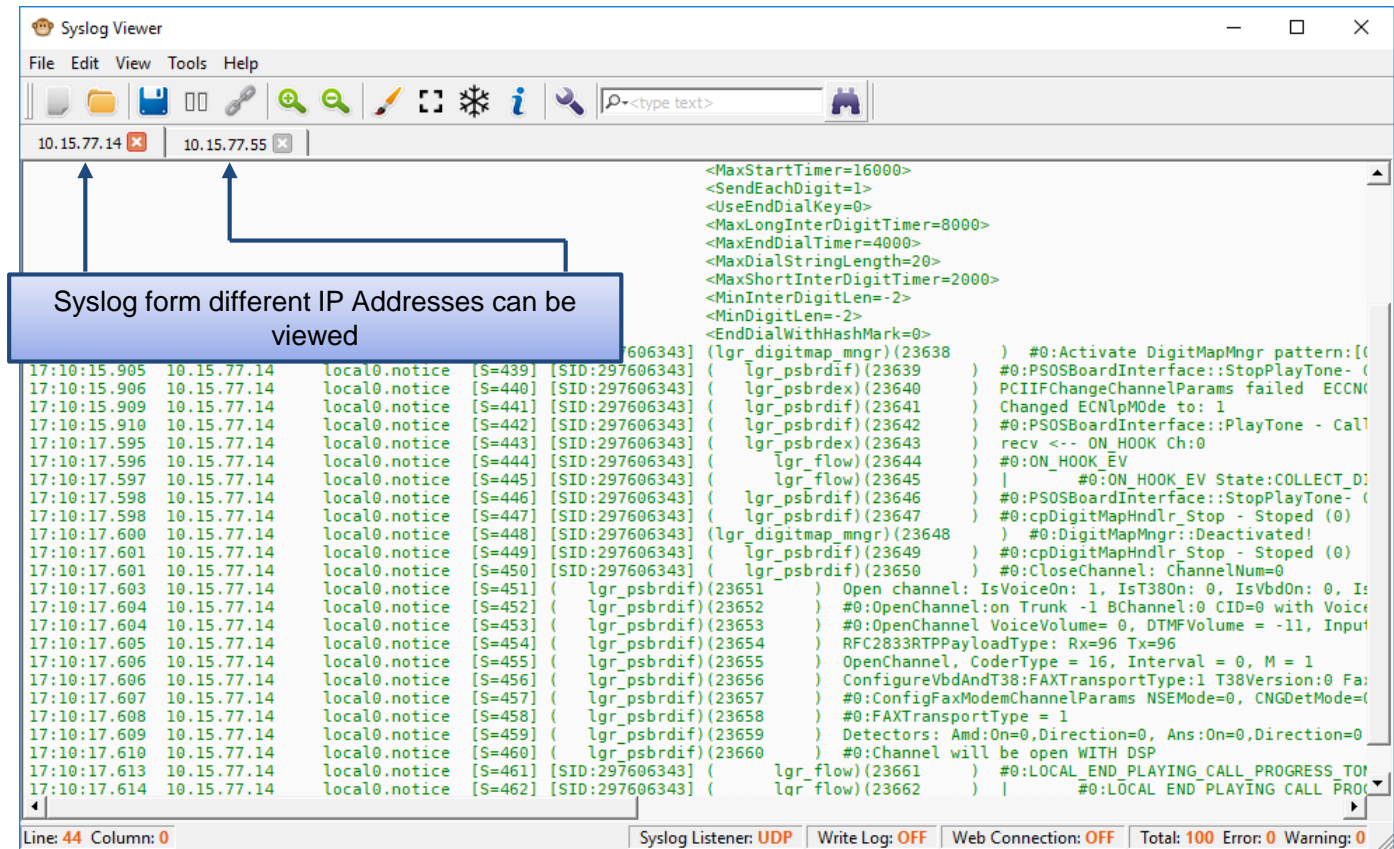
# AudioCodes Syslog Viewer – connect from PC to GW/SBC using Web Connection

- Connecting from ACSyslog to PC

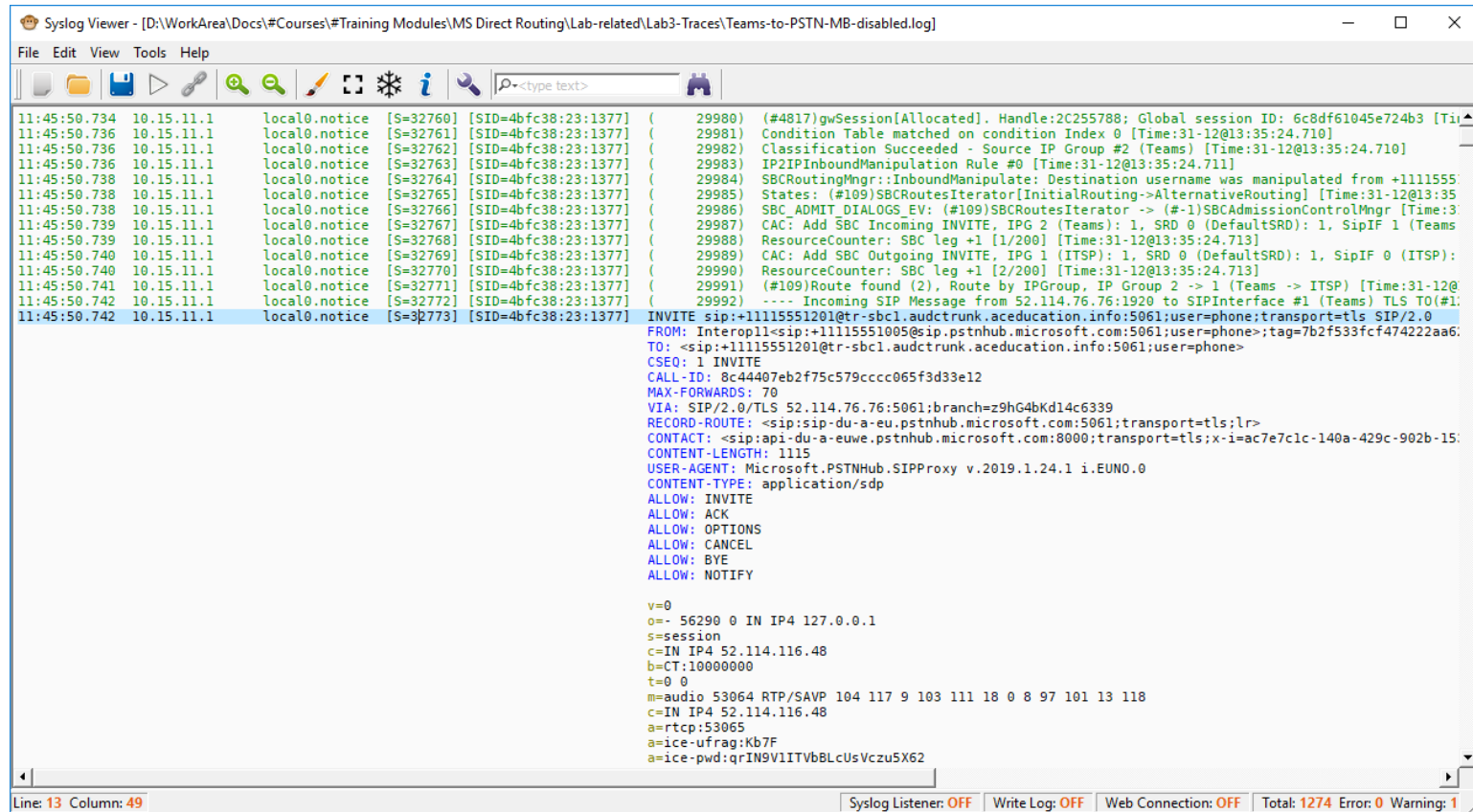
- Click **Connect to** link
- Input IP address, Username and Password of SBC/GW
- Logs recorded by Web Connection appear with **–web** in the display tab (Only 1 web connection can be monitored)
- ACSyslog connected to Specific SBC/GW shown in lower right corner



- Syslog can be enabled simultaneously in several devices, reporting to the same Syslog Server



- SIP/SDP messages are properly arranged to be easily identified for analysis



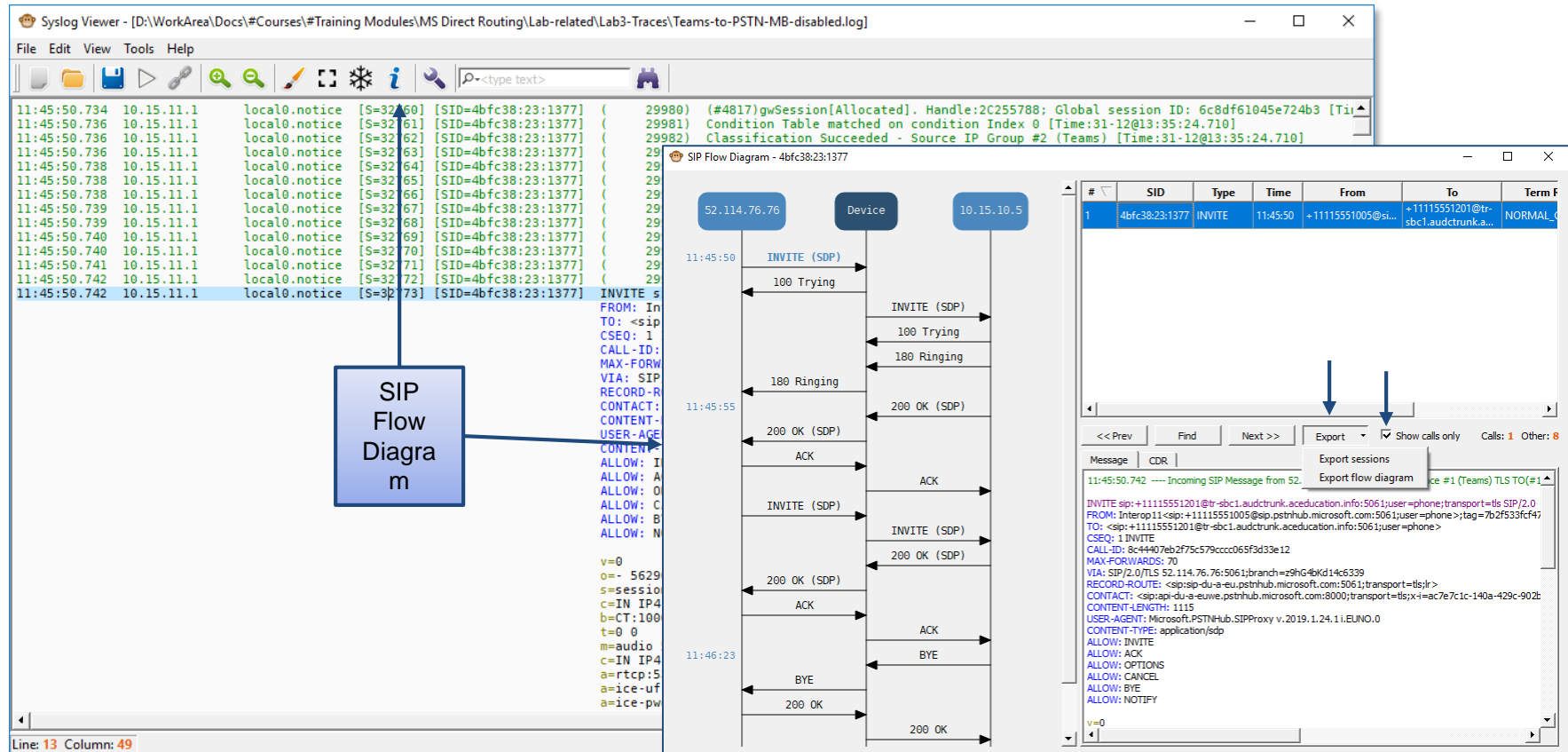
The screenshot shows the AudioCodes Syslog Viewer application. The top window title is "Syslog Viewer - [D:\WorkArea\Docs\#Courses\#Training Modules\MS Direct Routing\Lab-related\Lab3-Traces\Teams-to-PSTN-MB-disabled.log]". The interface includes a menu bar (File, Edit, View, Tools, Help) and a toolbar with various icons. The main display area is divided into two panes. The left pane shows a list of syslog messages with columns for time, source IP, destination IP, message type, and message content. The right pane shows a detailed view of the selected message, which is an INVITE message. The message content is displayed in a monospace font, showing the SIP/SDP details. At the bottom of the window, there is a status bar with the text "Line: 13 Column: 49" and a row of buttons: "Syslog Listener: OFF", "Write Log: OFF", "Web Connection: OFF", and "Total: 1274 Error: 0 Warning: 1".

```

11:45:50.734 10.15.11.1 local0.notice [S=32760] [SID=4bfc38:23:1377] ( 29980) (#4817)gwSession[Allocated]. Handle:2C255788; Global session ID: 6c8df61045e724b3 [Ti
11:45:50.736 10.15.11.1 local0.notice [S=32761] [SID=4bfc38:23:1377] ( 29981) Condition Table matched on condition Index 0 [Time:31-12@13:35:24.710]
11:45:50.736 10.15.11.1 local0.notice [S=32762] [SID=4bfc38:23:1377] ( 29982) Classification Succeeded - Source IP Group #2 (Teams) [Time:31-12@13:35:24.710]
11:45:50.736 10.15.11.1 local0.notice [S=32763] [SID=4bfc38:23:1377] ( 29983) IP2IPInboundManipulation Rule #0 [Time:31-12@13:35:24.711]
11:45:50.738 10.15.11.1 local0.notice [S=32764] [SID=4bfc38:23:1377] ( 29984) SBCRoutingMgr::InboundManipulate: Destination username was manipulated from +1111555:
11:45:50.738 10.15.11.1 local0.notice [S=32765] [SID=4bfc38:23:1377] ( 29985) States: (#109)SBCRoutesIterator[InitialRouting->AlternativeRouting] [Time:31-12@13:35
11:45:50.738 10.15.11.1 local0.notice [S=32766] [SID=4bfc38:23:1377] ( 29986) SBC ADMIT DIALOGS_EV: (#109)SBCRoutesIterator -> (#-1)SBCAdmissionControlMgr [Time:3
11:45:50.739 10.15.11.1 local0.notice [S=32767] [SID=4bfc38:23:1377] ( 29987) CAC: Add SBC Incoming INVITE, IPG 2 (Teams): 1, SRD 0 (DefaultSRD): 1, SipIF 1 (Teams
11:45:50.739 10.15.11.1 local0.notice [S=32768] [SID=4bfc38:23:1377] ( 29988) ResourceCounter: SBC leg +1 [1/200] [Time:31-12@13:35:24.713]
11:45:50.740 10.15.11.1 local0.notice [S=32769] [SID=4bfc38:23:1377] ( 29989) CAC: Add SBC Outgoing INVITE, IPG 1 (ITSP): 1, SRD 0 (DefaultSRD): 1, SipIF 0 (ITSP):
11:45:50.740 10.15.11.1 local0.notice [S=32770] [SID=4bfc38:23:1377] ( 29990) ResourceCounter: SBC leg +1 [2/200] [Time:31-12@13:35:24.713]
11:45:50.741 10.15.11.1 local0.notice [S=32771] [SID=4bfc38:23:1377] ( 29991) (#109)Route found (2), Route by IPGroup, IP Group 2 -> 1 (Teams -> ITSP) [Time:31-12@
11:45:50.742 10.15.11.1 local0.notice [S=32772] [SID=4bfc38:23:1377] ( 29992) ---- Incoming SIP Message from 52.114.76.1920 to SIPInterface #1 (Teams) TLS TO[#1:
11:45:50.742 10.15.11.1 local0.notice [S=32773] [SID=4bfc38:23:1377] ( 29993) INVITE sip:+11115551201@tr-sbcl.audctrunk.aceducation.info:5061;user=phone;transport=tl
FROM: Interop11<sip:+11115551005@sip.pstnhub.microsoft.com:5061;user=phone>;tag=7b2f533fcf474222aa6:
TO: <sip:+11115551201@tr-sbcl.audctrunk.aceducation.info:5061;user=phone>
CSEQ: 1 INVITE
CALL-ID: 8c44407eb2f75c579cccc065f3d33e12
MAX-FORWARDS: 70
VIA: SIP/2.0/TLS 52.114.76.76:5061;branch=z9hG4bKd14c6339
RECORD-ROUTE: <sip:sip-du-a-eu.pstnhub.microsoft.com:5061;transport=tlslr>
CONTACT: <sip:api-du-a-euwe.pstnhub.microsoft.com:8000;transport=tlsl;x-i=ac7e7c1c-140a-429c-902b-15:
CONTENT-LENGTH: 1115
USER-AGENT: Microsoft.PSTNHub.SIPProxy v.2019.1.24.1 i.EUN0.0
CONTENT-TYPE: application/sdp
ALLOW: INVITE
ALLOW: ACK
ALLOW: OPTIONS
ALLOW: CANCEL
ALLOW: BYE
ALLOW: NOTIFY

v=0
o=- 56290 0 IN IP4 127.0.0.1
s=session
c=IN IP4 52.114.116.48
b=CT:10000000
t=0 0
m=audio 53064 RTP/SAVP 104 117 9 103 111 18 0 8 97 101 13 118
c=IN IP4 52.114.116.48
a=rtcp:53065
a=ice-ufrag:Kb7F
a=ice-pwd:qrIN9V1ITvBLcUsVczu5X62
    
```

- The SIP/SDP flow diagram can be viewed and exported



The screenshot displays the AudioCodes Syslog Viewer interface. The main window shows a list of syslog messages. A blue box labeled "SIP Flow Diagram" points to a specific message in the log. To the right, a "SIP Flow Diagram - 4bfc38:23:1377" window shows a sequence diagram of SIP messages between two devices (52.114.76.76 and 10.15.10.5). The diagram includes messages like INVITE (SDP), 100 Trying, 180 Ringing, 200 OK (SDP), ACK, and BYE. Below the diagram, a table shows the details of the selected message (SID: 4bfc38:23:1377, Type: INVITE, Time: 11:45:50, From: +1115551005@si..., To: +1115551201@tr-sbc1.audtrunk.a...). The bottom right pane shows the raw SIP message details, including headers like FROM, TO, CSEQ, and SDP.

**SIP Flow Diagram**

**SIP Flow Diagram - 4bfc38:23:1377**

#	SID	Type	Time	From	To	Term F
1	4bfc38:23:1377	INVITE	11:45:50	+1115551005@si...	+1115551201@tr-sbc1.audtrunk.a...	NORMAL_C

Message: 11:45:50.742 ---- Incoming SIP Message from 52.114.76.76

Export sessions  
Export flow diagram

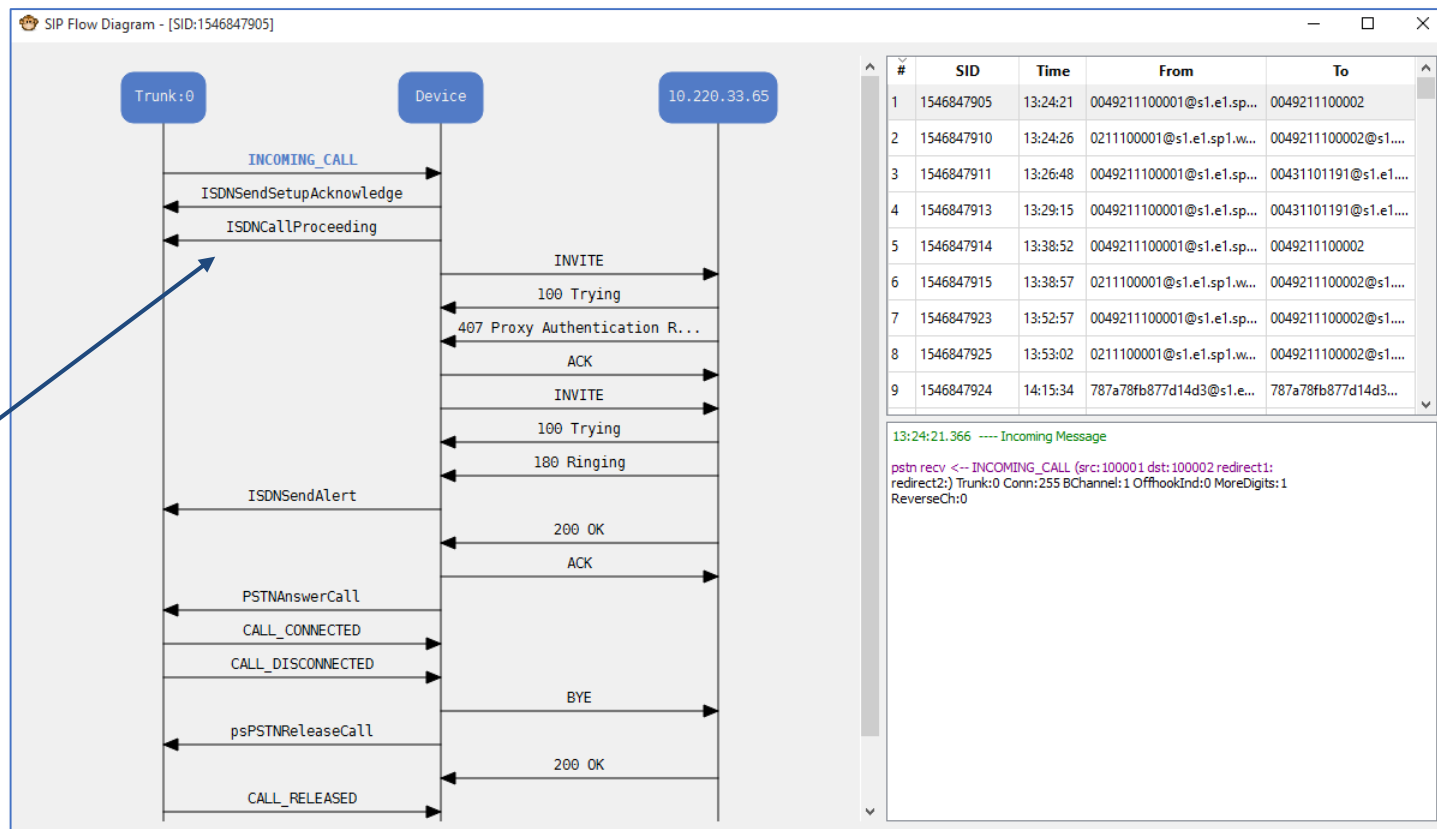
11:45:50.742 ---- Incoming SIP Message from 52.114.76.76

INVITE sip:+1115551201@tr-sbc1.audtrunk.aceduction.info:5061;user=phone;transport=tls SIP/2.0  
 FROM: Interop11<sip:+1115551005@sip.pssthub.microsoft.com:5061;user=phone>;tag=7b2f533cf47  
 TO: <sip:+1115551201@tr-sbc1.audtrunk.aceduction.info:5061;user=phone>  
 CSEQ: 1 INVITE  
 CALL-ID: 8c44407eb2775c579ccc065f3d33e12  
 MAX-FORWARDS: 70  
 VIA: SIP/2.0/TLS 52.114.76.76:5061;branch=z9hGzBkD14c6339  
 RECORD-ROUTE: <sip:sip-du-a-eu.pssthub.microsoft.com:5061;transport=tls>  
 CONTACT: <sip:api-du-a-eu.pssthub.microsoft.com:8000;transport=tls;x-i-ac7e7c1c-140a-429c-902b  
 CONTENT-LENGTH: 1115  
 USER-AGENT: Microsoft.PSTNHub.SIPProxy v.2019.1.24.1 i:EUNO.0  
 CONTENT-TYPE: application/sdp  
 ALLOW: INVITE  
 ALLOW: ACK  
 ALLOW: OPTIONS  
 ALLOW: CANCEL  
 ALLOW: BYE  
 ALLOW: NOTIFY

v=0  
 o=- 5629  
 s=session  
 c=IN IP4  
 b=CT:100  
 t=0 0  
 m=audio  
 c=IN IP4  
 a=rtcp:5  
 a=ice-uf  
 a=ice-pw

Line: 13 Column: 49

- The SIP/SDP <-> ISDN flow diagram can be viewed



- Each arrow on the SIP/SDP flow diagram points to the right place in the trace

The screenshot displays the AudioCodes Syslog Viewer application. The main window shows a syslog trace with columns for Time, IP Address, Local Port, Remote Port, and Message. A specific message is highlighted, and a blue box labeled "SIP Flow Diagram" points to it. To the right, a SIP flow diagram illustrates the sequence of messages between two devices (52.114.76.76 and 10.15.10.5). Arrows in the diagram point to the corresponding entries in the syslog trace, demonstrating how the diagram helps navigate the raw log data.

**Syslog Trace (Highlighted Row):**

Time	IP Address	Local Port	Remote Port	Message
11:45:50.734	10.15.11.1	local0.notice	[S=32] 60	[SID=4bfc38:23:1377] ( 29980) (#4817)gwSession[Allocated]. Handle:2C255788; Global session ID: 6c8df61045e724b3 [Ti...

**SIP Flow Diagram (Highlighted Row):**

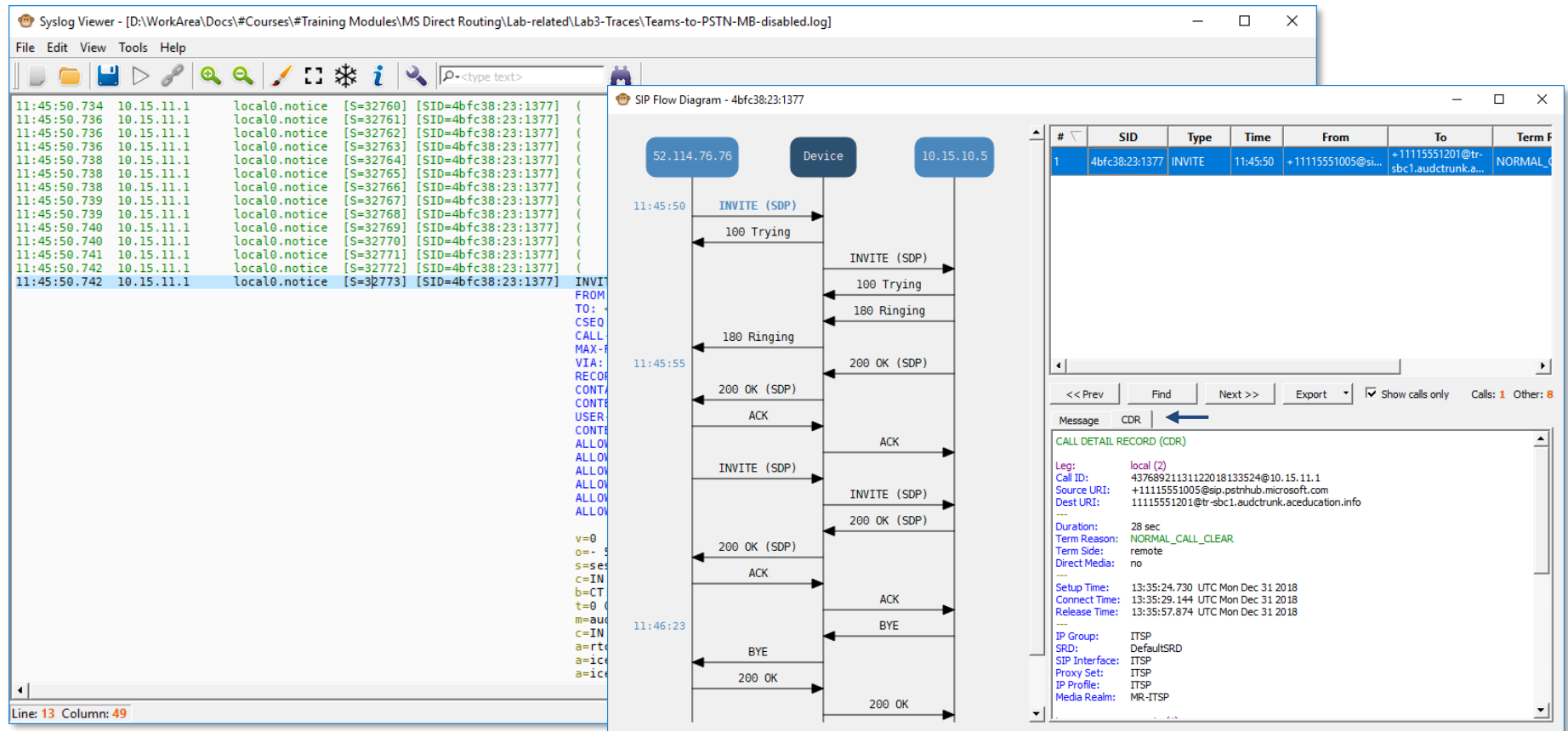
#	SID	Type	Time	From	To	Term F
1	4bfc38:23:1377	INVITE	11:45:50	+11115551005@si...	+11115551201@tr- sbc1.audctrunk.a...	NORMAL...

**SIP Message Details (Highlighted Row):**

```

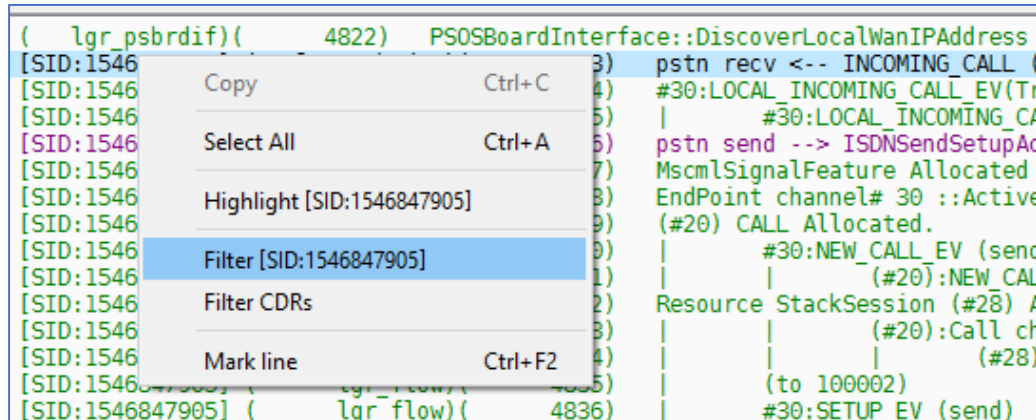
11:45:50.742 ----- Incoming SIP Message from 52.114.76.76:1920 to SIPInterface #1 (Teams) TLS TO(1)
INVITE sip:+11115551201@tr-sbc1.audctrunk.aceducation.info:5061;user=phone;transport=tls SIP/2.0
FROM: Interop11<sip:+11115551005@sip.pstnhub.microsoft.com:5061;user=phone>;tag=7b2f533cf47
TO: <cap:+11115551201@tr-sbc1.audctrunk.aceducation.info:5061;user=phone>
CSEQ: 1 INVITE
CALL-ID: 8c44407eb2f75c579cccc065f3d33e12
MAX-FORWARDS: 70
VIA: SIP/2.0/TLS 52.114.76.76:5061;branch=z9hGzKd14c6339
RECORD-ROUTE: <cap:sip-du-a-eu.pstnhub.microsoft.com:5061;transport=tls>
CONTACT: <cap:sip-du-a-eu.pstnhub.microsoft.com:8000;transport=Bs;x-i=ac7e7c1c-140a-429c-902b
CONTENT-LENGTH: 1115
USER-AGENT: Microsoft.PSTNHub.SIPProxy v.2019.1.24.1 i.EUNO.0
CONTENT-TYPE: application/sdp
ALLOW: INVITE
ALLOW: ACK
ALLOW: OPTIONS
ALLOW: CANCEL
ALLOW: BYE
ALLOW: NOTIFY
    
```

- CDR info





- Extracting Single Call



The screenshot shows a list of syslog messages. A right-click context menu is open over the entry with SID:1546847905. The menu options are:

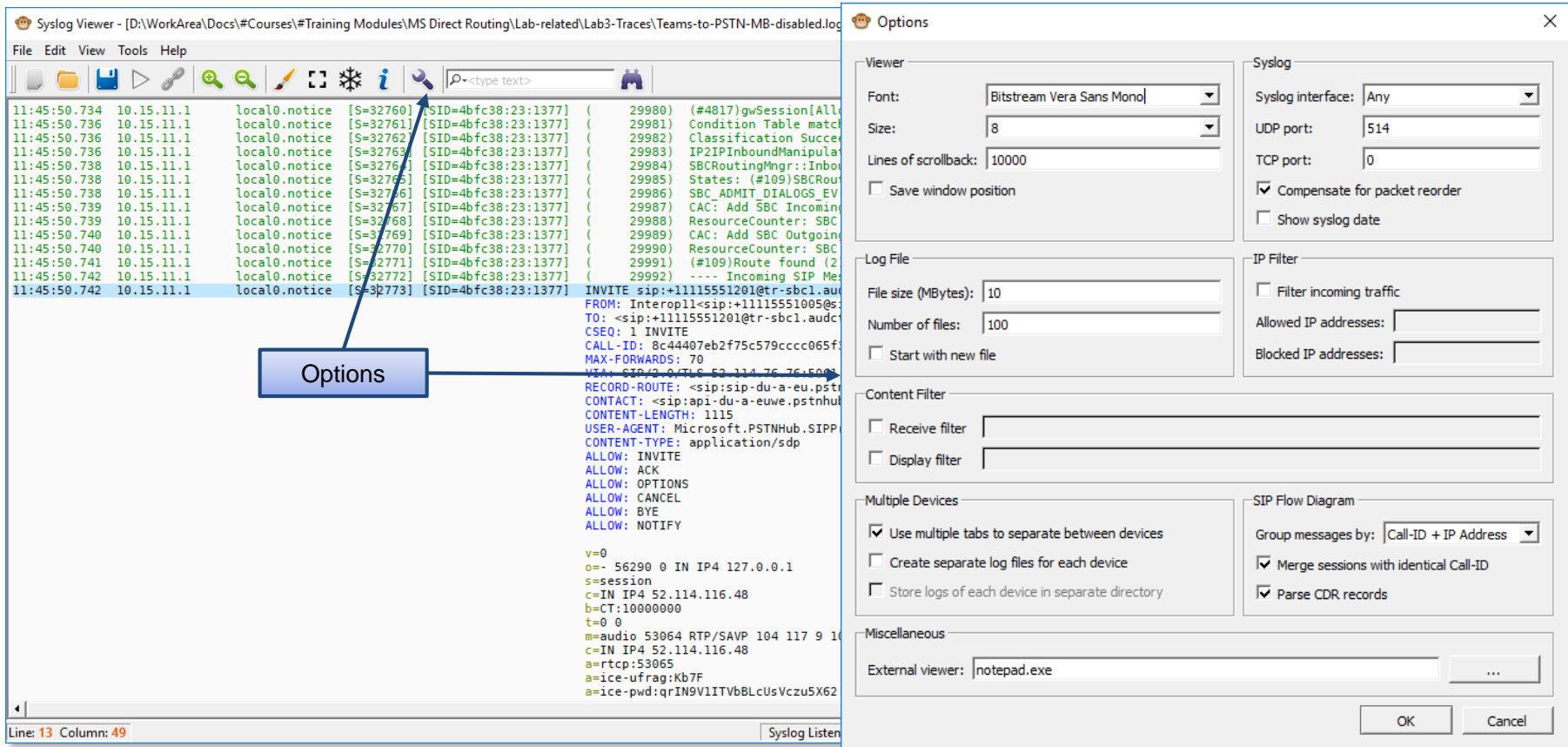
- Copy (Ctrl+C)
- Select All (Ctrl+A)
- Highlight [SID:1546847905]
- Filter [SID:1546847905] (highlighted)
- Filter CDRs
- Mark line (Ctrl+F2)

The background log entries include:

```
( lgr_psbrdif)( 4822) PS0SBoardInterface::DiscoverLocalWanIPAddress
[SID:1546847905] ( lgr_flow)( 4823) pstn rcv <-- INCOMING_CALL (
[SID:1546847905] ( lgr_flow)( 4824) #30:LOCAL_INCOMING_CALL_EV(Tr
[SID:1546847905] ( lgr_flow)( 4825) | #30:LOCAL_INCOMING_CA
[SID:1546847905] ( lgr_flow)( 4826) pstn send --> ISDNSetupAcknowledge()
[SID:1546847905] ( lgr_flow)( 4827) MscmlSignalFeature Allocated
[SID:1546847905] ( lgr_flow)( 4828) EndPoint channel# 30 ::Active
[SID:1546847905] ( lgr_flow)( 4829) (#20) CALL Allocated.
[SID:1546847905] ( lgr_flow)( 4830) | #30:NEW_CALL_EV (send
[SID:1546847905] ( lgr_flow)( 4831) | (#20):NEW CALL
[SID:1546847905] ( lgr_flow)( 4832) Resource StackSession (#28) A
[SID:1546847905] ( lgr_flow)( 4833) | (#20):Call ch
[SID:1546847905] ( lgr_flow)( 4834) | (#28)
[SID:1546847905] ( lgr_flow)( 4835) (to 100002)
[SID:1546847905] ( lgr_flow)( 4836) #30:SETUP EV (send)
```

FILTER #1 <span>✖</span>					
192.168.0.1	local0.notice	[S=5114]	[SID:1546847905]	( lgr_psbrdif)(	4823) pstn rcv <-- INCOMING_CALL (src:100001
192.168.0.1	local0.notice	[S=5115]	[SID:1546847905]	( lgr_flow)(	4824) #30:LOCAL_INCOMING_CALL_EV(Trunk:0 Conn:
192.168.0.1	local0.notice	[S=5116]	[SID:1546847905]	( lgr_flow)(	4825)   #30:LOCAL_INCOMING_CALL_EV State
192.168.0.1	local0.notice	[S=5117]	[SID:1546847905]	( lgr_psbrdif)(	4826) pstn send --> ISDNSetupAcknowledge()
192.168.0.1	local0.notice	[S=5118]	[SID:1546847905]	( media_service)(	4827) MscmlSignalFeature Allocated ResourceID:
192.168.0.1	local0.notice	[S=5119]	[SID:1546847905]	( lgr_flow)(	4828) EndPoint channel# 30 ::Active DSP Channe
192.168.0.1	local0.notice	[S=5120]	[SID:1546847905]	( lgr_call)(	4829) (#20) CALL Allocated.
192.168.0.1	local0.notice	[S=5121]	[SID:1546847905]	( lgr_flow)(	4830)   #30:NEW_CALL_EV (send) : (Unkno
192.168.0.1	local0.notice	[S=5122]	[SID:1546847905]	( lgr_flow)(	4831)   (#20):NEW CALL_EV:(Unkno
192.168.0.1	local0.notice	[S=5123]	[SID:1546847905]	( lgr_stk_mgr)(	4832) Resource StackSession (#28) Allocated
192.168.0.1	local0.notice	[S=5124]	[SID:1546847905]	( lgr_flow)(	4833)   (#20):Call channing stat





The screenshot displays the AudioCodes Syslog Viewer application. The main window shows a list of syslog messages with columns for time, source IP, destination IP, and message content. A blue box labeled "Options" points to the "Options" dialog box, which is open on the right side of the screen.

**Options Dialog Box:**

- Viewer:**
  - Font: Bitstream Vera Sans Mono
  - Size: 8
  - Lines of scrollbar: 10000
  - ☐ Save window position
- Syslog:**
  - Syslog interface: Any
  - UDP port: 514
  - TCP port: 0
  - ☒ Compensate for packet reorder
  - ☐ Show syslog date
- Log File:**
  - File size (MBytes): 10
  - Number of files: 100
  - ☐ Start with new file
- IP Filter:**
  - ☐ Filter incoming traffic
  - Allowed IP addresses: [empty field]
  - Blocked IP addresses: [empty field]
- Content Filter:**
  - ☐ Receive filter [empty field]
  - ☐ Display filter [empty field]
- Multiple Devices:**
  - ☒ Use multiple tabs to separate between devices
  - ☐ Create separate log files for each device
  - ☐ Store logs of each device in separate directory
- SIP Flow Diagram:**
  - Group messages by: Call-ID + IP Address
  - ☒ Merge sessions with identical Call-ID
  - ☒ Parse CDR records
- Miscellaneous:**
  - External viewer: notepad.exe

Buttons: OK, Cancel

## Three Objects:

Endpoint-----Call----- Session

Leftmost – Physical Endpoint

First Tab – Board Endpoint (EP)

Second Tab - Call

Third tab - Session

Rightmost - IP

```
( lgr_call)(7835 ) (#3) CALL Allocated.
( lgr_flow)(7836 ) | #0:NEW_CALL_EV (send) : (Unknown)
( lgr_flow)(7837 ) | | #3:NEW_CALL_EV:(Unknown)
( lgr_stk_mgr)(7838 ) Resource StackSession <#3> Allocated
( lgr_flow)(7839 ) | | #3:Call changing states from:IdleState to:NewCallState_Tel2IP
( lgr_flow)(7840 ) | | | (#3)SIPStackSession <- (#0)ENDPOINT: NEW_CALL_EV (Unknown)
( lgr_call)(7841 ) | | | #3GetNextUI:GlobalUI=67801172, mACAddrLsb=4119367
( lgr_call)(7842 ) | | | #3GetNextUI:GlobalUI=67801173
( lgr_flow)(7843 ) | | (to 1101)
( lgr_flow)(7844 ) | | #0:SETUP_EV (send) : (Unknown)
( lgr_flow)(7845 ) | | | #3:SETUP (TO:1101, FROM:1103):(Unknown)
( lgr_call)(7846 ) new call from EndPoint
( lgr_flow)(7847 ) | | | #3:Call changing states from:NewCallState_Tel2IP to:InitiatedState_Tel2IP
( lgr_flow)(7848 ) | | | (#3)SIPStackSession <- (#0)ENDPOINT: SETUP_EV (Unknown)
( lgr_call)(7849 ) Call::GetStartIndex() return -1
```

- **EndPoint**

- The logical representation of a channel. Channels may come in various types (analog, digital, conference, bct)
- The main functionality of EndPoint is to map global application events into the specific channel logic

- **Call**

- The connecting entity between EndPoint and Session
- The call establishes synchronization between the two legs

- **StackSession**

- The entity which maps application events into the specific Stack (SIP) logic
- The StackSession basically represents the connection point to the IP leg

```
( lgr_call)(7835 ) (#3) CALL Allocated.
( lgr_flow)(7836 ) | #0:NEW_CALL_EV (send) : (Unknown)
( lgr_flow)(7837 ) | | #3:NEW_CALL_EV:(Unknown)
( lgr_stk_mgr)(7838 ) Resource StackSession <#3> Allocated
( lgr_flow)(7839 ) | | #3:Call changing states from:IdleState to:NewCallState_Tel2IP
( lgr_flow)(7840 ) | | | (#3)SIPStackSession <- (#0)ENDPOINT: NEW_CALL_EV (Unknown)
( lgr_call)(7841 ) | | | #3GetNextUI:GlobalUI=67801172, mACAddrLsb=4119367
( lgr_call)(7842 ) | | | #3GetNextUI:GlobalUI=67801173
( lgr_flow)(7843 ) | | (to 1101)
( lgr_flow)(7844 ) | | #0:SETUP_EV (send) : (Unknown)
( lgr_flow)(7845 ) | | | #3:SETUP (TO:1101, FROM:1103):(Unknown)
( lgr_call)(7846 ) new call from EndPoint
( lgr_flow)(7847 ) | | | #3:Call changing states from:NewCallState_Tel2IP to:InitiatedState_Tel2IP
( lgr_flow)(7848 ) | | | (#3)SIPStackSession <- (#0)ENDPOINT: SETUP_EV (Unknown)
( lgr_call)(7849 ) Call::GetStartIndex() return -1
```

- Log tabs are denoted by pipe sign (“|”)
- Each object is prefixed with a pound sign (“#”)
- Board/EP objects are tied to a specific numbering scheme based on the hardware type

- MP-11x
  - #0 represents physical FXO/FXS port 1
- Mediant 2000, 3000, 5000
  - #0 represents physical B-channel 1 on trunk 1
- Mediant 1000
  - Physical Analog ports begin at #0 (range 0-23)

```
( lgr_psbrdex)(          69)  recv <-- OFF_HOOK Ch:0  
( lgr_flow)(          70)  #0:OFF_HOOK_EV [Time:09
```

- Physical Digital ports begin at #30 (range 30-125)

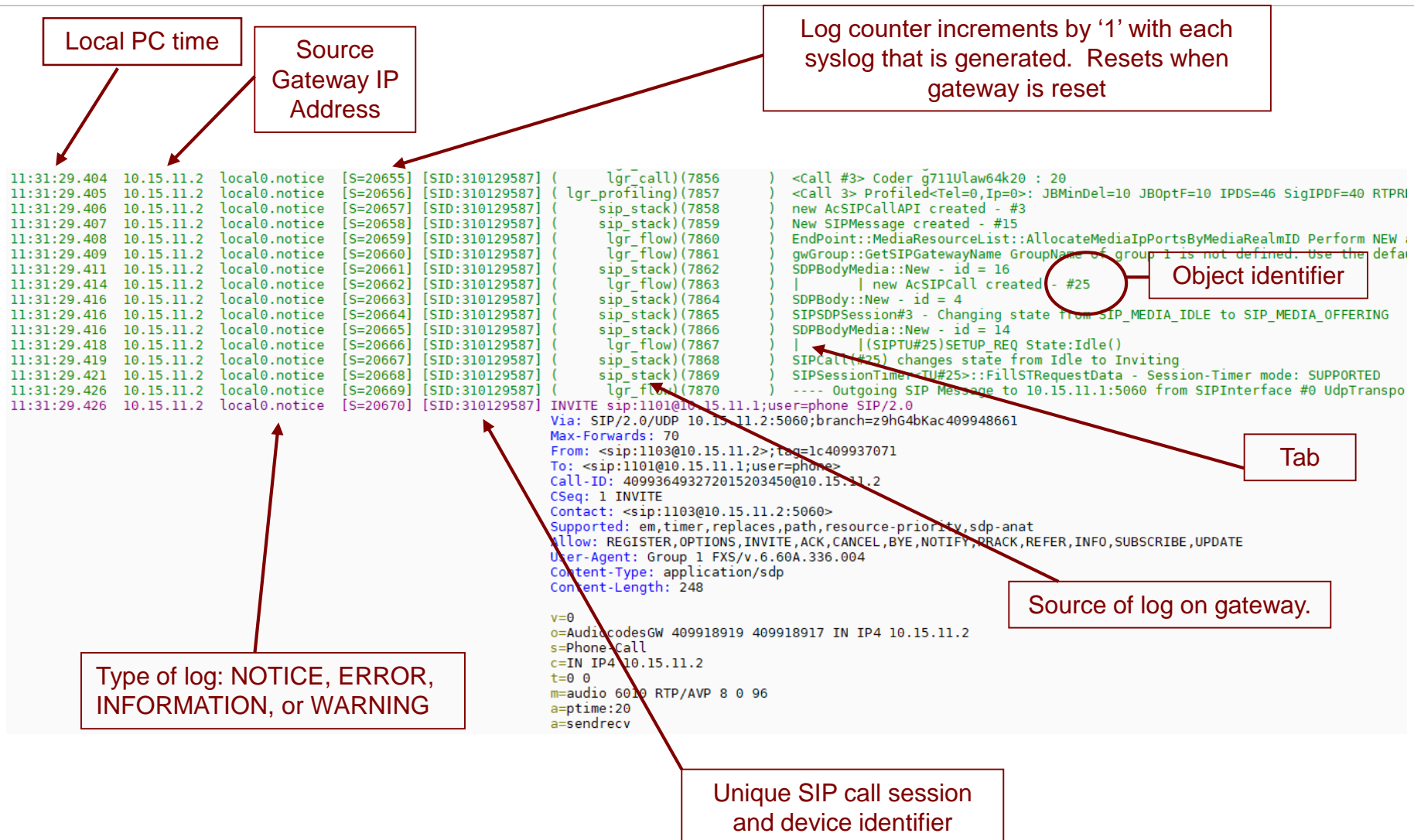
```
( lgr_psbrdex)(          5724)  pstn recv <-- CALL_PROCEEDING Trunk:0 Conn:0 BChannel:6 callhdl:0 Loc:-1 Des:-1 Cmt:67  
( lgr_flow)(          5725)  #35:LOCAL_CALL_PROCEEDING_EV(Trunk:0 Conn:0 Bchannel:6 TpEv=73) [Time:11-07@10:09:48.699]
```

- Call/Session objects are not directly related to any specific port or component in the gateway, but do increment logically

- Call is established by allocating a resource from each object pool on the gateway
  - For IP → Tel calls the rightmost object is created first
  - For Tel → IP calls the leftmost object is created first
- Each object layer invokes an object from the 'next' logical layer
  - Board/EP object → Call Object → Session Object → IP
- Transition from one object to another represents a passing of messages, states, and information from one side of the gateway to the other (PSTN → IP / IP → PSTN)

- Mechanism by which the different parts of the application interact
- Some parts of the application interact internally with events which are mapped into the application main events before passed to other parts of the application
  - For example: SIP\_ACK\_EV is a SIP stack internal event which is mapped to the application CONNECT\_ACK\_EV
- Not all events can be handled by all of the application modules, so if the event will not be processed by a module which can handle it, you will encounter the vague “Unhandled Event...”
  - “Unhandled events” are not necessarily errors, but simply events that are not handled by that particular application module

# Syslog fields at a Glance



- **Information used to quickly identify what is happening in logs**

- SIP messaging
- DTMF / Digit related events
- Endpoint allocation
- Routing and Manipulation Actions
- Sending and Receiving of PSTN messages



- Direction of call flow, calling and called party numbers, Display names and vital network information (coders, payload types, ports)
- Two call flow direction markers:
  - *Incoming SIP Message from 192.168.40.100:5060* ---
  - *Outgoing SIP Message to 192.168.40.100:5060* ---
    - **Incoming** messages are SIP messages received by the gateway from the SIP network
    - **Outgoing** messages are SIP messages SENT by the gateway to the SIP network
- Links below highlight the different components of SIP messages that are useful in troubleshooting:
  - [Incoming INVITE indicates an IP to Tel call](#)
  - [Outgoing INVITE indicates a Tel to IP call](#)



Indicates direction of the message (incoming or outgoing) and the IP address of the entity the gateway is talking to for this dialog.

In this case it is an outgoing INVITE to a SIP network, therefore the call was originated as a Tel→IP call on the gateway

```
11:31:29.426 10.15.11.2 local0.notice [S=20669] [SID:310129587] ( lg_r_flow)(7870) ---- Outgoing SIP Message to 10.15.11.1:5060 from SIPInterface #0 UdpTransportObject[#0] ----
11:31:29.426 10.15.11.2 local0.notice [S=20670] [SID:310129587] INVITE sip:1101@10.15.11.1:user=phone SIP/2.0
Via: SIP/2.0/UDP 10.15.11.2:5060;branch=z9hG4bKac409948661
Max-Forwards: 70
From: <sip:1103@10.15.11.2>;tag=1c409937071
To: <sip:1101@10.15.11.1:user=phone>
Call-ID: 409936493272015203450@10.15.11.2
CSeq: 1 INVITE
Contact: <sip:1103@10.15.11.2:5060>
Supported: em,timer,replaces,path,resource-priority,sdp-anat
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
User-Agent: Group 1 FXS/v.6.60A.336.004
Content-Type: application/sdp
Content-Length: 248

v=0
o=AudiocodesGW 409918919 409918917 IN IP4 10.15.11.2
s=Phone-Call
c=IN IP4 10.15.11.2
t=0 0
m=audio 6010 RTP/AVP 8 0 96
a=ptime:20
a=sendrecv
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15

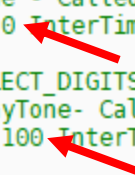
11:31:29.427 10.15.11.2 local0.notice [S=20671] [SID:310129587] ( sip_stack)(7872)
UdpRtxMgr::Transmit 1 INVITE Rtx Left: 6 Dest: 10.15.11.1:5060 CallID: (409936493272015203450@10.15.11.2)
```



The gateway will always prepare to retransmit last message in UDP operations. If there is to be a retransmit, the message will retransmit, decrementing the 'Rtx Left' counter before resending the message.

- 'Digit' in the log indicates an event on the TEL side
- 'recv <-- DIGIT' indicates a DTMF digit has been received on the TEL side
- 'send --> DIGIT' indicates the gateway is sending a DTMF digit to the TEL side

```
( lgr_psbrdif)(7807 ) #0:PSOSBoardInterface::PlayTone - Called Tone=DIAL_TONE Direction=PLAY_TONE_2_TEL
( lgr_psbrdex)(7808 ) recv <-- DIGIT(1) Ch:0 OnTime:0 InterTime:83334034 Direction:0 System:1
( lgr_flow)(7809 ) #0:DIGIT_EV
( lgr_flow)(7810 ) | #0:DIGIT_EV State:COLLECT_DIGITS Substate:sub_None
( lgr_psbrdif)(7811 ) #0:PSOSBoardInterface::StopPlayTone- Called
( lgr_psbrdex)(7812 ) recv <-- DIGIT(1) Ch:0 OnTime:100 InterTime:83334034 Direction:0 System:1
( lgr_flow)(7813 ) #0:DIGIT_EV
( lgr_flow)(7814 ) | #0:DIGIT_EV State:COLLECT_DIGITS Substate:sub_COLLECT_REST_OF_DIGITS_SUB_STATE
( lgr_psbrdex)(7815 ) recv <-- DIGIT(1) Ch:0 OnTime:0 InterTime:100 Direction:0 System:1
( lgr_flow)(7816 ) #0:DIGIT_EV
( lgr_flow)(7817 ) | #0:DIGIT_EV State:COLLECT_DIGITS Substate:sub_COLLECT_REST_OF_DIGITS_SUB_STATE
( lgr_psbrdex)(7818 ) recv <-- DIGIT(1) Ch:0 OnTime:100 InterTime:100 Direction:0 System:1
( lgr_flow)(7819 ) #0:DIGIT_EV
( lgr_flow)(7820 ) | #0:DIGIT_EV State:COLLECT_DIGITS Substate:sub_COLLECT_REST_OF_DIGITS_SUB_STATE
( lgr_psbrdex)(7821 ) recv <-- DIGIT(0) Ch:0 OnTime:0 InterTime:100 Direction:0 System:1
( lgr_flow)(7822 ) #0:DIGIT_EV
( lgr_flow)(7823 ) | #0:DIGIT_EV State:COLLECT_DIGITS Substate:sub_COLLECT_REST_OF_DIGITS_SUB_STATE
( lgr_psbrdex)(7824 ) recv <-- DIGIT(0) Ch:0 OnTime:100 InterTime:100 Direction:0 System:1
( lgr_flow)(7825 ) #0:DIGIT_EV
```



- 'OnTime' indicates how long DTMF was generated/detected
  - OnTime:0 – Start of DTMF digit was detected
  - OnTime:100 – DTMF digit was released – time in milliseconds

- **Tel2IP** is the keyword used to indicate Table Tel to IP Routing was used to find the routing destination
- **FindIPDestination** is the keyword for Tel to IP call routing decisions
  - rmRc:0 (OK) indicates route matching table entry was found
  - rmRc:5 (FAIL) indicates route matching table entry was not found

```
[SID:310129587] ( lgr_call)(7846 ) new call from EndPoint
[SID:310129587] ( lgr_flow)(7847 ) | | #3:Call changing states from:NewCallState_Tel2IP to:InitiatedState Tel2IP
[SID:310129587] ( lgr_flow)(7848 ) | | (#3)SIPStackSession <- (#0)ENDPOINT: SETUP_EV (Unknown)
[SID:310129587] ( lgr_call)(7849 ) Call::GetStartIndex() return -1
[SID:310129587] ( lgr_stack)(7850 ) FindIPDestination: rmRc:0 (OK) SrcIpGroup:-1 IpconnHndl:-1 DstPrefix:1101 DstIp:10.15.11.1
[SID:310129587] ( lgr_stack)(7851 ) RoutingInstance (#3) RtrRouting: trying to find a route according to Routing Table
[SID:310129587] ( lgr_stk_ses)(7852 ) DecideRoutingSetup DestIpGroupId:0
[SID:310129587] ( lgr_stk_ses)(7853 ) <SESSION #3> UpdateAfterDecidingRouting: IpProfileId (0), ChargeCode (255), NewIndex (0), CostGroup (-1)
```

```
[SID:1544990850] ( lgr_stack)(1989 ) FindIPDestination: rmRc:5 (FAIL) SrcIpGroup:-1 IpconnHndl:-1 DstPrefix:1111101 DstIp:
[SID:1544990850] ( lgr_stack)(1990 ) ?? [WARNING] RoutingInstance::DecideRouting: phone:1111101 No relevant entry in routing table
```

- **IP2Tel** is the keyword used to indicate Table IP to Trunk/Hunt Group Routing was used to find the routing destination
- **GetEndPointPhoneNum** is a good keyword to use to search for IP to Trunk/Hunt Group routing decisions
- **GetTrunkGroupId** determines if a route was located in the IP to Trunk/Hunt Group table

```
[SID:310129595] (    lgr_flow)(8352    )    #8:Call changing states from:IdleState to:NewCallState IP2Tel
[SID:310129595] (    lgr_flow)(8353    )    ServicesMgr::GetEndPoint PhoneNum = 1103
[SID:310129595] ( lgr_psbrdif)(8354    )    GetTrunkGroupId- TrunkGroup:1 Trunk:-1 found DstNum:1103 DstPfx:* SrcNum:1101 SrcPfx:* SrcIp:a0f0b01
[SID:310129595] (    lgr_call)(8355    )    Call::SetCoderListForCall #8 Found 2 Common Coders For Call
[SID:310129595] (    lgr_call)(8356    )    <Call #8> Coder g711Alaw64k20 : 20
[SID:310129595] (    lgr_call)(8357    )    <Call #8> Coder g711Ulaw64k20 : 20
```

- For Digital trunks, if the gateway finds a match in the routing and trunk groups tables AND there are resources available (**Current trunks status**) for the trunks associated with the trunk group, the call is passed along

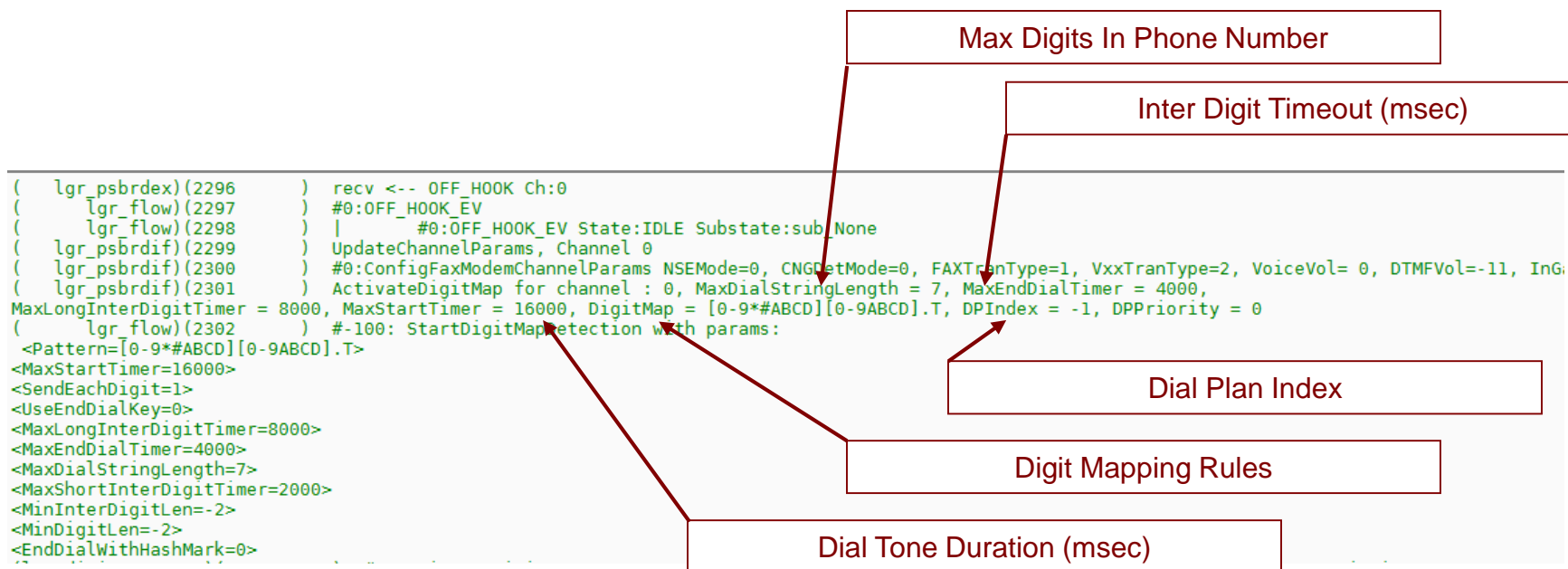
```
[SID:3d5724:135:128] (    lgr_flow)(    8698)    (#120):Call changing states from:IdleState to:NewCallState IP2Tel [Time:11-07@17:17:23.565]
[SID:3d5724:135:128] (    lgr_flow)(    8699)    ServicesMgr::GetEndPoint PhoneNum = 9764000 [Time:11-07@17:17:23.565]
[SID:3d5724:135:128] ( lgr_gw_engine)(    8700)    GetTrunkGroupId- TrunkGroup:2 Trunk:-1 found DstNum:9764000 DstPfx:* SrcNum:1111103 SrcPfx:* SrcIp:168758018 Sr
[SID:3d5724:135:128] (    lgr_psbrdif)(    8701)    Current trunks status: 1 [Time:11-07@17:17:23.566]
[SID:3d5724:135:128] (    lgr_call)(    8702)    Call::SetProhibitedCallParams (#129) Found 1 Common Coders For Call [Time:11-07@17:17:23.566]
[SID:3d5724:135:128] (    lgr_call)(    8703)    Call (#129) Coder g72920 : 20 [Time:11-07@17:17:23.566]
```

- Current trunks status: 1 – Trunk is available
- Current trunks status: 0 – Trunk is unavailable

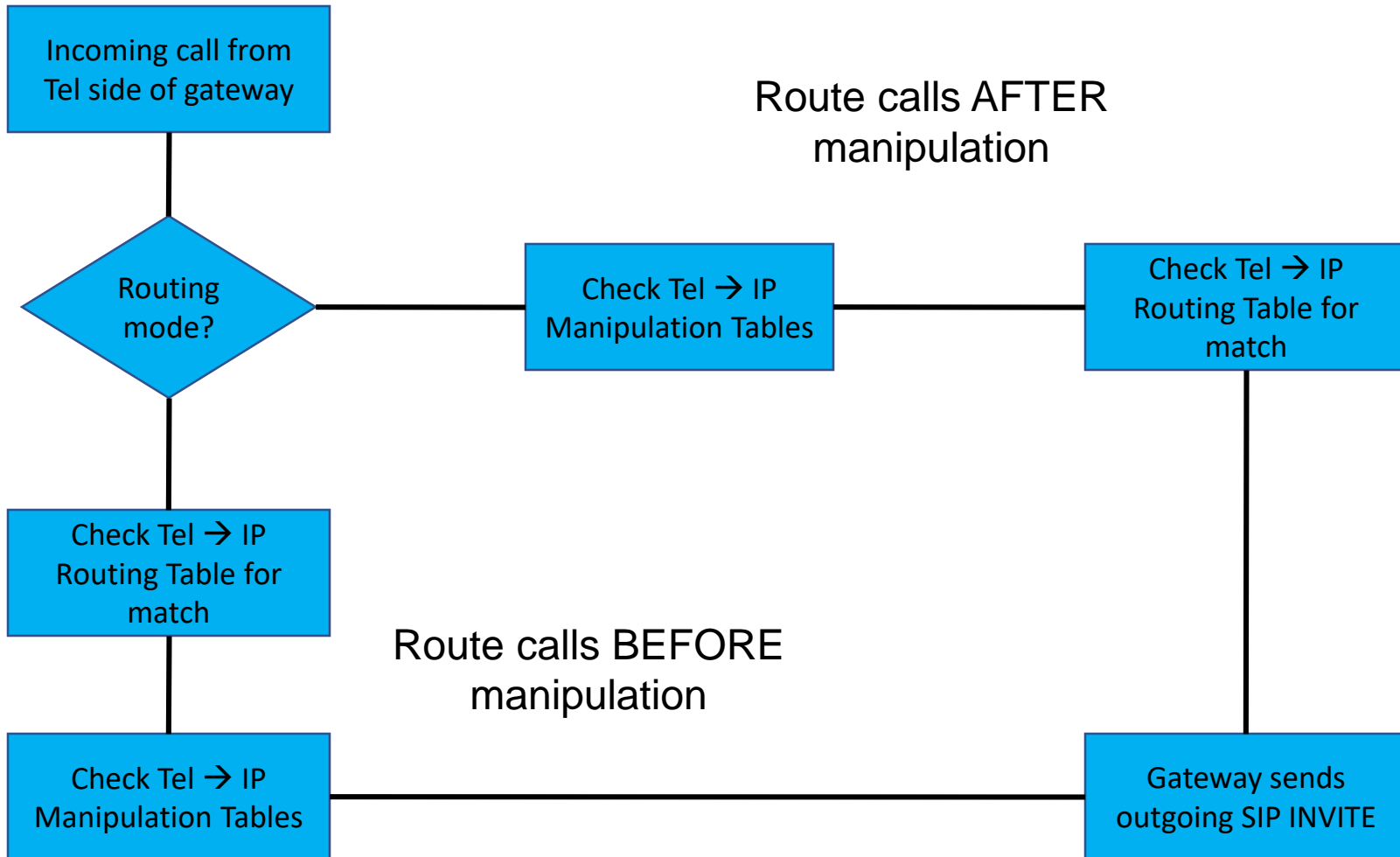
- 'pstn rcv ←' indicates an incoming message from the PSTN network to the gateway.
- 'pstn send →' indicates that the gateway is either sending a new call or related message out to the PSTN network
  - In the example below the message is for a new outgoing call.

```
[SID=3d5724:135:120] ( lgr_flow)( 8389) pstn send --> PlaceCall: Trunk:0 BChannel:14 ConnID:0 SrcPN=1111103 SrcSN= DstPN=9764000 DstSN= SrcNT=0 SrcNP=0 SrcPres=0 SrcScrn=0
[SID=3d5724:135:120] ( lgr_psbrdex)( 8390) pstn rcv <-- CALL_PROCEEDING Trunk:0 Conn:0 BChannel:14 callhdl:0 Loc:-1 Des:-1 Cmt:67 [Time:11-07@16:57:08.190]
[SID=3d5724:135:120] ( lgr_flow)( 8391) #43:LOCAL_CALL_PROCEEDING_EV(Trunk:0 Conn:0 BChannel:14 tpev=73) [Time:11-07@16:57:08.190]
[SID=3d5724:135:120] ( lgr_flow)( 8392) #43:LOCAL_CALL_PROCEEDING_EV State:WAIT FOR BOARD ANSWER Substate:sub None [Time:11-07@16:57:08.190]
[SID=3d5724:135:120] ( lgr_flow)( 8393) #43:PROGRESS_INDICATOR_EV State:WAIT FOR BOARD ANSWER (send) : (1796904715472015213852@10.15.11.2) [Time:11-07@16:57:08.191]
[SID=3d5724:135:120] ( lgr_flow)( 8394) (#133):PROGRESS_INDICATOR_EV(PI=-1)(PC=-1):(1796904715472015213852@10.15.11.2) [Time:11-07@16:57:08.191]
[SID=3d5724:135:120] ( lgr_flow)( 8395) #43:PROCEEDING_EV State:WAIT FOR BOARD ANSWER (send) : (1796904715472015213852@10.15.11.2) [Time:11-07@16:57:08.191]
[SID=3d5724:135:120] ( lgr_flow)( 8396) (#133):PROCEEDING_EV:(1796904715472015213852@10.15.11.2) [Time:11-07@16:57:08.191]
[SID=3d5724:135:120] ( lgr_flow)( 8397) (#17)SIPStackSession <- (#0)ISDNEndPoint: PROCEEDING_EV (1796904715472015213852@10.15.11.2) [Time:11-07@16:57:08.191]
[SID=3d5724:135:124] ( lgr_psbrdex)( 8564) pstn rcv <-- CALL_DISCONNECTED Trunk:0 Conn:0 RetCause:104 NetCause:102 [Time:11-07@17:01:24.052]
```

# Example Gateway Parameter Usage displayed in Syslog



# Tel → IP Routing Table Flowchart





# Example Gateway Table Usage displayed in Syslog

## Destination Number Tel to IP Manipulation Table

## IP to Trunk/Hunt Group Table

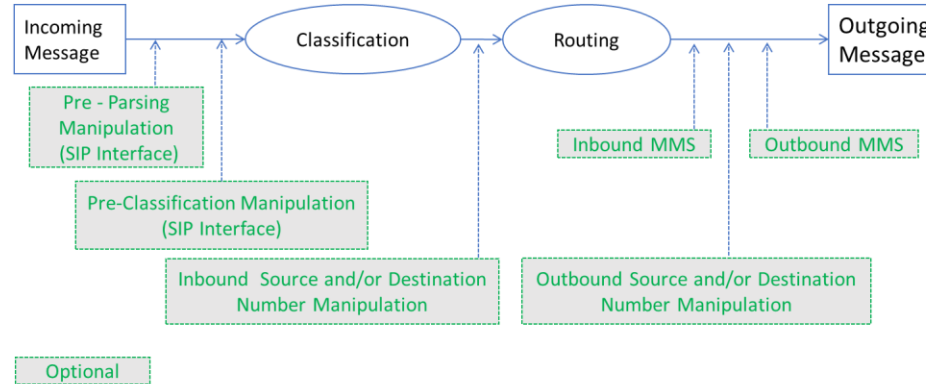
## Coder Group Table

## IP & Tel Profile Tables

## Media Realm Table

```
(lgr_flow)(2335) ) | #0:DIALED_STRING_EV State:COLLECT_DIGITS Substate:sub_COLLECT_REST_OF_DIGITS_SUB
(lgr_digmap_mgr)(2336) ) | DigitMapMgr::HandleDialStringEv Match = 1, MatchNum = 0 STR = 1101
(lgr_call)(2337) ) | (#7) CALL Allocated.
(lgr_flow)(2338) ) | #0:NEW_CALL_EV (send) : (Unknown)
(lgr_flow)(2339) ) | #7:NEW_CALL_EV:(Unknown)
(lgr_stk_mgr)(2340) ) | Resource StackSession <#7> Allocated
(lgr_flow)(2341) ) | #7:Call changing states from:IdleState to:NewCallState_Tel2IP
(lgr_flow)(2342) ) | (#7)SIPStackSession <- (#0)ENDPOINT: NEW_CALL_EV (Unknown)
(lgr_call)(2343) ) | #7GetNextTUT:GlobalUIT=1770634048, mACAddrLsb=4119367
1770634049
n)
(lgr_flow)(2347) ) | #7:SETUP (TO:1101, FROM:111103):(Unknown)
(lgr_call)(2348) ) | new call from EndPoint
(lgr_flow)(2349) ) | NumberMapMgr::HandlePhoneNumberMapping Manipulation on Tel to IP - Destination Number
(lgr_num)(2350) ) | PhoneNumber::AddPrefix - Number change from 1101 to 1111101
(lgr_flow)(2351) ) | #7:Call changing states from:NewCallState_Tel2IP to:InitiatedState_Tel2IP
(lgr_flow)(2352) ) | (#7)SIPStackSession <- (#0)ENDPOINT: SETUP_EV (Unknown)
(lgr_call)(2353) ) | Call::GetStartIndex() return -1
(lgr_stack)(2354) ) | FindIpDestination: rmRc:0 (OK) SrcIpGroup:-1 IpConnHndl:-1 DstPrefix:1101 DstIp:10.15.11.1
(lgr_stack)(2355) ) | RoutingInstance (#7) RTRouting: trying to find a route according to Routing Table
(lgr_stk_ses)(2356) ) | DecideRoutingSetup DestIpGroupId:0
(lgr_stk_ses)(2357) ) | <SESSION #7> UpdateAfterDecidingRouting: IpProfileId (0), ChargeCode (255), NewIndex (0), CostGroup (-1)
(lgr_call)(2358) ) | Call::SetCoderListForCall #7 Found 1 Common Coders For Call
(lgr_call)(2359) ) | <Call #7> Coder g72920 : 20
(lgr_profiling)(2360) ) | <Call 7> Profiled<Tel=0,Ip=0>: <BDIndex=10 JB0ptF=10 IPDS=46 SigIPDF=40 RTPRD=0 ECE=1 Ing=
(lgr_flow)(2361) ) | new AcSIPCallAPI created - #9
(lgr_flow)(2362) ) | New SIPMessage created - #23
(lgr_flow)(2363) ) | EndPoint::MediaResourceList::AllocateMediaIpPortsByMediaRealmID Perform NEW allocation of Media ports for RealmIndex(0) port(6020) current allocations
(lgr_flow)(2364) ) | gwGroup::GetSIPGatewayName groupName of group 1 is not defined. Use the default.
(lgr_flow)(2365) ) | SDPBodyMedia::New - id = 29
(lgr_flow)(2366) ) | | new AcSIPCall created - #15
(lgr_flow)(2367) ) | SDPBody::New - id = 4
(lgr_flow)(2368) ) | SIPSDPSession#7 - Changing state from SIP_MEDIA_IDLE to SIP_MEDIA_OFFERING
()
Inviting
(lgr_flow)(2372) ) | SIPSessionTimer<TU#15>::FillSTRequestData - Session-Timer mode: SUPPORTED
(lgr_flow)(2373) ) | ---- Outgoing SIP Message to 10.15.11.1:5060 from SIPInterface #0 UdpTransportObject[#0] ----
INVITE sip:1111101@10.15.11.1;user=phone SIP/2.0
```

- Session Border Controller uses CMR Process to handle Call flow across legs on the SBC



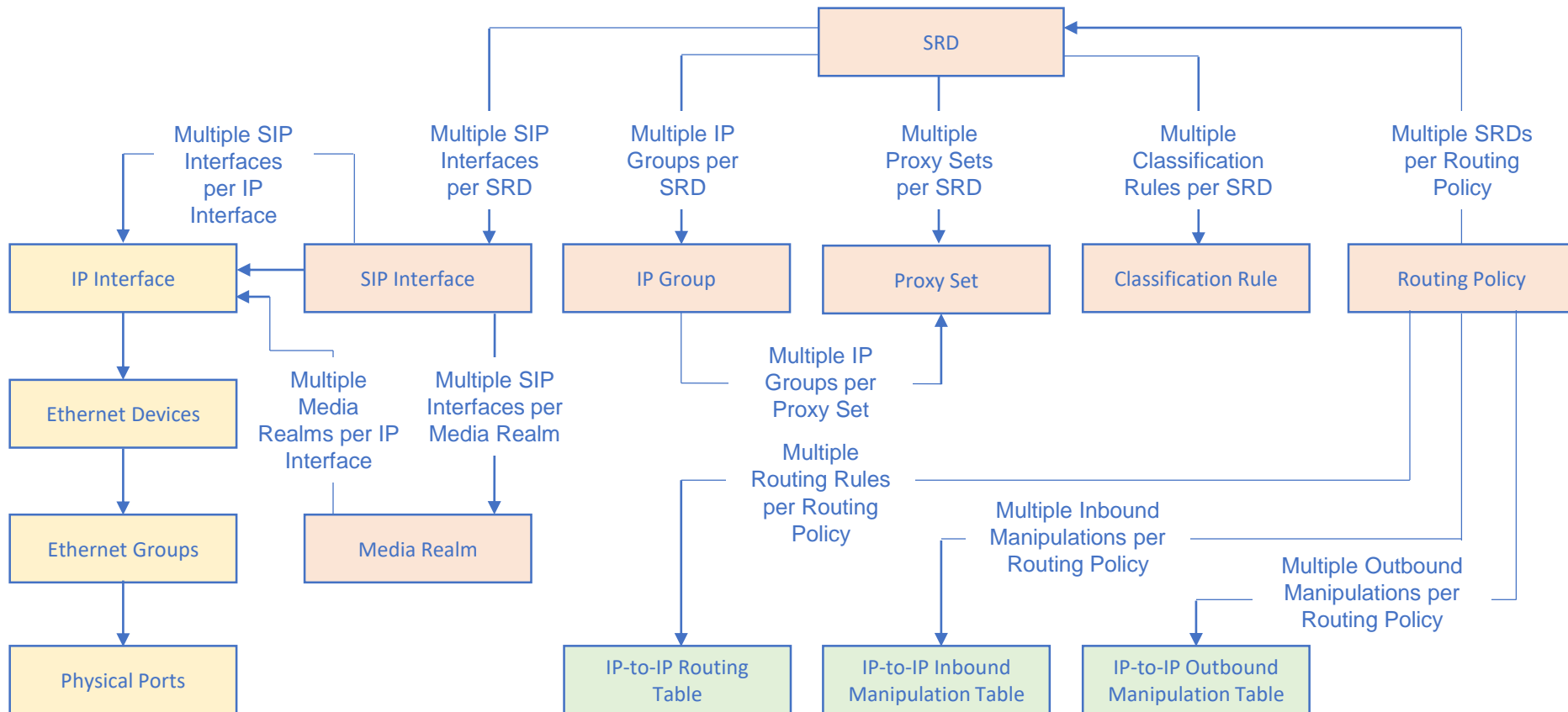
- C = Classify
- M = Manipulate
- R = Route
- This call flow is demonstrated in the SBC Syslogs

Classify

Manipulate

Route

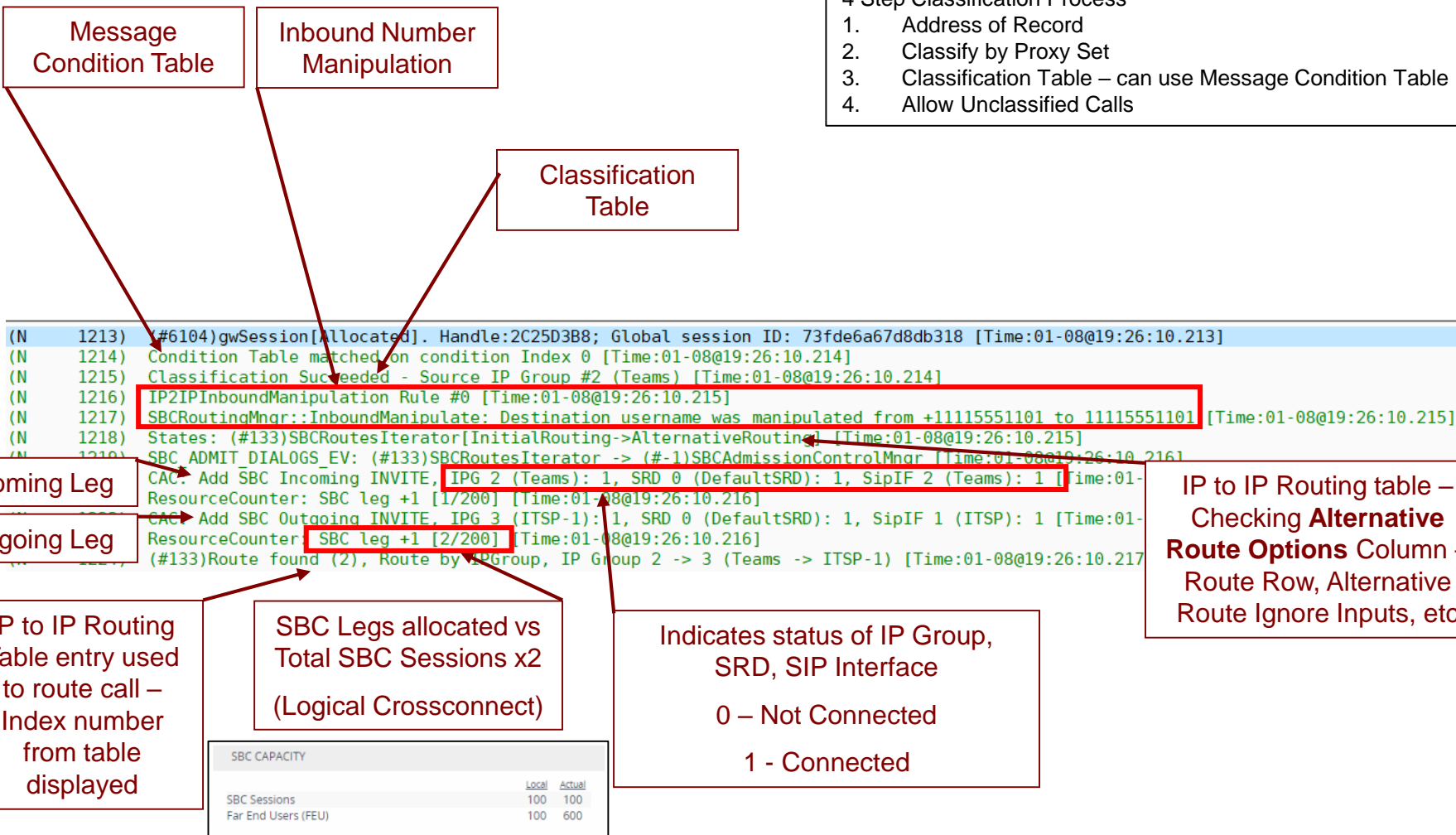
```
(N 1213) (#6104)gwSession[Allocated]. Handle:2C25D3B8; Global session ID: 73fde6a67d8db318 [Time:01-08@19:26:10.213]
(N 1214) Condition Table matched on condition Index 0 [Time:01-08@19:26:10.214]
(N 1215) Classification Succeeded - Source IP Group #2 (Teams) [Time:01-08@19:26:10.214]
(N 1216) IP2IPInboundManipulation Rule #0 [Time:01-08@19:26:10.215]
(N 1217) SBCRoutingMgr::InboundManipulate: Destination username was manipulated from +11115551101 to 11115551101 [Time:01-08@19:26:10.215]
(N 1218) States: (#133)SBCRoutesIterator[InitialRouting->AlternativeRouting] [Time:01-08@19:26:10.215]
(N 1219) SBC_ADMIT_DIALOGS_EV: (#133)SBCRoutesIterator -> (#-1)SBCAdmissionControlMgr [Time:01-08@19:26:10.216]
(N 1220) CAC: Add SBC Incoming INVITE, IP6 2 (Teams): 1, SRD 0 (DefaultSRD): 1, SipIF 2 (Teams): 1 [Time:01-08@19:26:10.216]
(N 1221) ResourceCounter: SBC leg +1 [1/200] [Time:01-08@19:26:10.216]
(N 1222) CAC: Add SBC Outgoing INVITE, IP6 3 (ITSP-1): 1, SRD 0 (DefaultSRD): 1, SipIF 1 (ITSP): 1 [Time:01-08@19:26:10.216]
(N 1223) ResourceCounter: SBC leg +1 [2/200] [Time:01-08@19:26:10.216]
(N 1224) (#133)Route found (2), Route by IPGroup, IP Group 2 -> 3 (Teams -> ITSP-1) [Time:01-08@19:26:10.217]
```



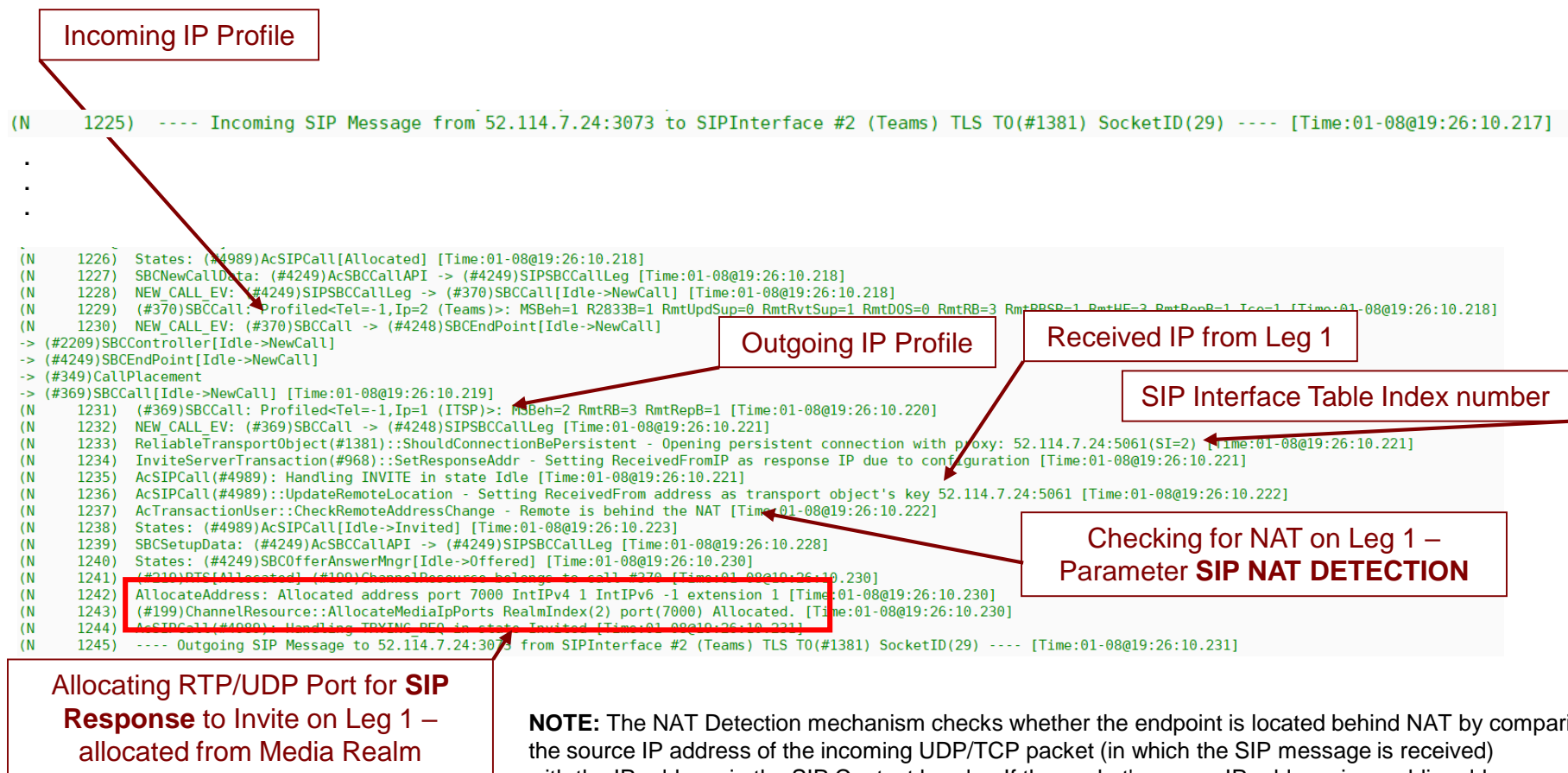
# SBC CMR Process Identified – Deep Dive

## 4 Step Classification Process

1. Address of Record
2. Classify by Proxy Set
3. Classification Table – can use Message Condition Table
4. Allow Unclassified Calls



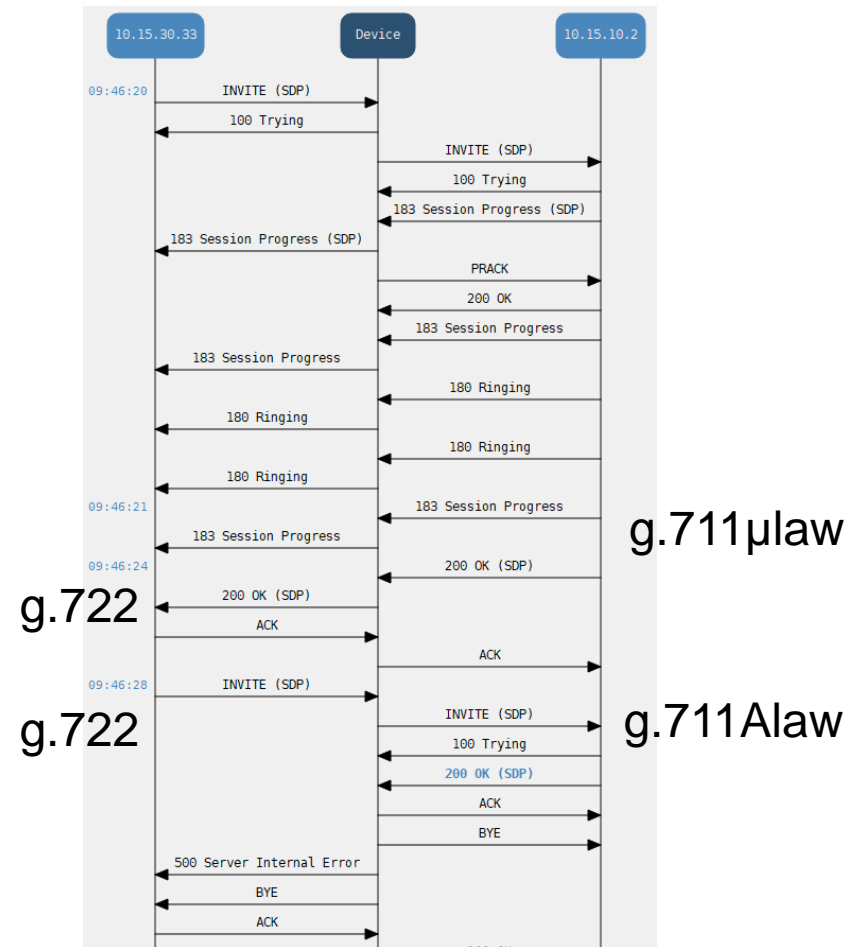
# SBC CMR Process Identified – Deep Dive



**NOTE:** The NAT Detection mechanism checks whether the endpoint is located behind NAT by comparing the source IP address of the incoming UDP/TCP packet (in which the SIP message is received) with the IP address in the SIP Contact header. If the packet's source IP address is a public address and the Contact header's IP address is a local address, the device considers the endpoint as located behind NAT.

# Error Analysis

- Scenario: Music on Hold Call to Skype for Business from IP-PBX fails with Media Mismatch
- Original Invite to SBC from IP-PBX advertises
  - m=audio 6080 RTP/AVP 9 8 0 101
- Invite Traverses SBC to Skype advertises
  - m=audio 6310 RTP/AVP 9 0 101
- 200 OK from Skype advertises
  - m=audio 53656 RTP/AVP 0 101
- 200 OK to IP-PBX advertises
  - m=audio 6570 RTP/AVP 9 101
- Call established
- Re-Invite from IP-PBX to SBC advertises
  - m=audio 6080 RTP/AVP 9 8 0 101
  - a=sendonly
- Re-Invite from SBC to Skype advertises
  - m=audio 6310 RTP/AVP 9 8 0 13 101
  - a=inactive
- 200 OK from Skype advertises
  - m=audio 53656 RTP/AVP 8 13 101
  - a=inactive
- SBC sends Skype a BYE and IP-PBX a 500 Internal Server Error



```
( 454530) SBCSDPFeatureMgr::CheckMediaOfferMatch - match wasn't found for media#0 ImageMatchFound=False GeneralMatchFound=False [Time:07-08@08:45:33.896]
( 454531) SBCSDPFeatureMgr::HandleIncomingAnswer - for media#0CheckMediaOfferMatch failed [Time:07-08@08:45:33.896]
( 454532)?? [WARNING] SBCSDPSession(#1378)::HandleAnswerSDPFromIP - Match was not found. Changing state to last stable state [Time:07-08@08:45:33.896]
```

- Skype tried to renegotiate the media at G.711Alaw instead of using what had been negotiated as G.711μLaw because a message manipulation was added that modified the SDP
- It was found that the Allowed Coder Group table was removing the payload type 8 (g.711Alaw) from the incoming Invite from the IP-PBX, this caused the call to fail
  - (#220)SBCCall: Profiled<Tel=-1,lp=0 (ITSP)>: ExtCGrp=1 **AllCGrp=1** MSBeh=2 R2833B=1 AltDM=2 MltDTMF=1 AssrtID=1 2833PT=101 [Time:07-08@08:45:25.815]

09:46:28.530 ---- Outgoing SIP Message to 10.15.10.2:5068 from SIPInterface #0 (SIPInterface\_1) TCP TO(#3111) SocketID(89) ---- [Time:07-08@08:45:33.897]

BYE sip:FE.TR.local:5068;transport=Tcp;maddr=10.15.10.2 SIP/2.0  
Via: SIP/2.0/TCP 10.15.13.10:5068;alias;branch=z9hG4bKac1475632245  
Max-Forwards: 70  
From: <sip:33335553101@gw3.tr.local>;tag=1c349183516  
To: <sip:+33335553005@tr.local>;tag=4729fd70ed;epid=2F236C8398  
Call-ID: 87797660578201984525@10.15.13.10  
CSeq: 4 BYE  
User-Agent: Mediant VE-H SBC/v.7.20A.250.003  
Reason: SIP ;cause=488 ;text="488 Not Acceptable Here"  
Content-Length: 0

Media Realms (2)

Proxy Sets (2)

IP Groups (2)

▲ CODERS & PROFILES

IP Profiles (2)

Coder Settings

Coder Groups

Allowed Audio Coders Groups (1)

Allowed Video Coders Groups (0)

▲ SBC

Allowed Audio Coders Groups [#1] > Allowed Audio Coders (2)

+ New

Edit

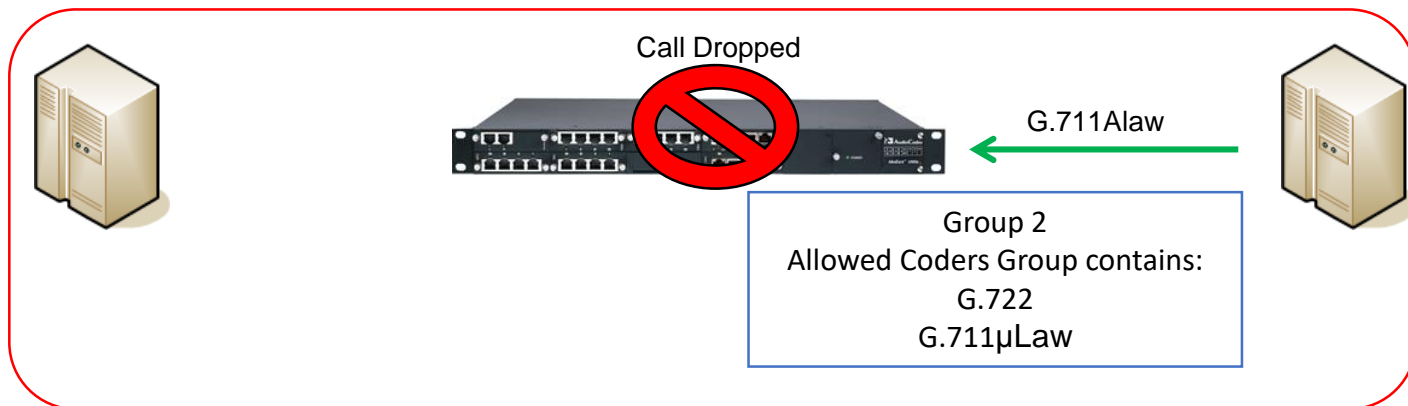
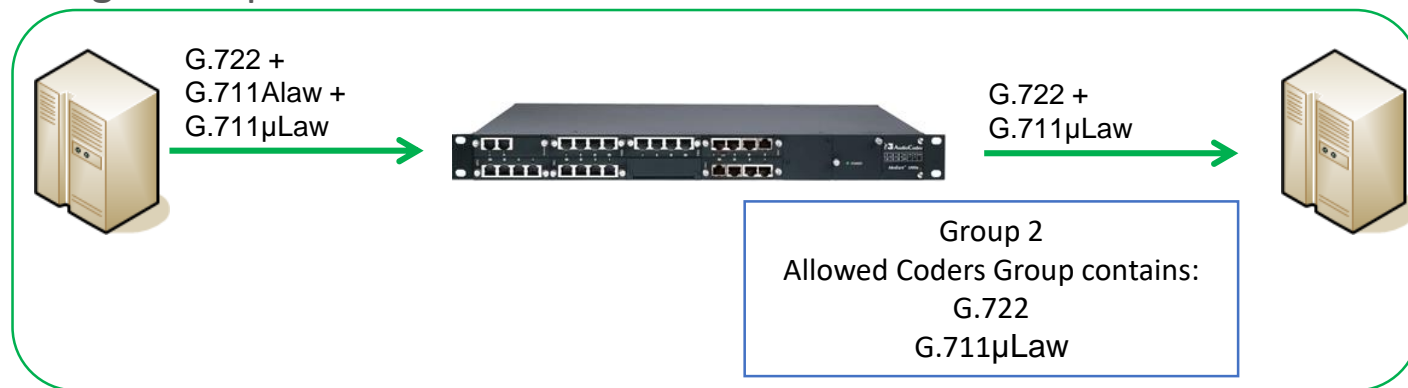
🗑

Page 1 of 1 Show 10 records per page

INDEX ↕	CODER	USER-DEFINED CODER
0	G.722	
1	G.711 U-law	



- At least one incoming coder must be in the Allowed Coders Group
  - Either for a request and/or a response
  - Typically the Allowed Coder Group would remove the Codec not on the list, but a message manipulation was used that modified the SDP



Analyze the log!

Examples of Syslog servers available as shareware:

- Kiwi Enterprises: <http://www.kiwisyslog.com/>
- The US CMS Server: [http://uscms.fnal.gov/hanlon/uscms\\_server/](http://uscms.fnal.gov/hanlon/uscms_server/)
- Netal SL4NT Syslog Daemon: <http://www.netal.com>
- For additional information, refer to [www.Syslog.org](http://www.Syslog.org)
  - a site dedicated to helping to understand and implement logging and analysis systems