Verizon T.38 FAX Configuration Guide for AudioCodes MP-11x

Version 1.0

Revision History

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Glossary & Acronyms

Term	Definition
CPE	Customer Premise Equipment
MP	Media Pack
EMS	Element Management System

Table of Contents

1. Preliminary

The primary purpose of this document is a configuration guide to be used by Verizon Employees and other authorized personnel as an aid in implementing T.38 FAX on the AudioCodes MP Media Pack Analog Gateway with the Avaya Communication Server 1000 (CS-1K). It is intended to provide a high level view of sample configurations as well as show as many critical configuration details as practical without duplicating the existing installation, operation, and configuration documentation. This document may also be useful to customers or potential customers who are interested in standard installations involving these components.

This configuration guide provides you with supplementary information on AudioCodes SIP-based, Voice-over-IP (VoIP) devices specific to Avaya CS-1K deployments. Refer to the User Manual and other references at the end of this section for a full description of the capabilities of these products.

1.1 Applicable AudioCodes Systems

Media Pack Series:

AudioCodes Media Pack Series is designated by MP-1xx (for example the 2-line MP-112, the four line MP-114, or the 24 line MP-124). The MP-1xx is a multi-line analog Media Gateway which provides excellent voice quality and optimized packet voice streaming over IP networks, enabling voice, fax, and data traffic to be sent over the same IP network.

Based on AudioCodes' award-winning, field-proven TrunkPack design, the MP-1xx uses AudioCodes' well-established DSP voice compression technology. The MP-1xx incorporates up to 24 analog ports for connection, either directly to an enterprise PBX (MP-10x/FXO), and to phones or fax (MP-1xx/FXS), supporting up to 24 simultaneous VoIP calls. Additionally, the MP-1xx is equipped with a 10/100 Base-T Ethernet port for connection to the LAN. MP-1xx Gateways are best suited for small to medium-sized enterprises, branch offices or for residential Media Gateway solutions.

Figure 1 – AudioCodes MP-1xx (MP-118 shown)

AudioCodes EMS

The AudioCodes Element Management System (EMS) is an advanced solution for standards-based management of Media Gateways within VoP networks, covering all areas vital for the efficient operation, administration, management and provisioning (OAM&P) of AudioCodes' families of Media Gateways. The EMS features a

Client/Server architecture, enabling customers to access the EMS from multiple, remotely located work centers and workstations.

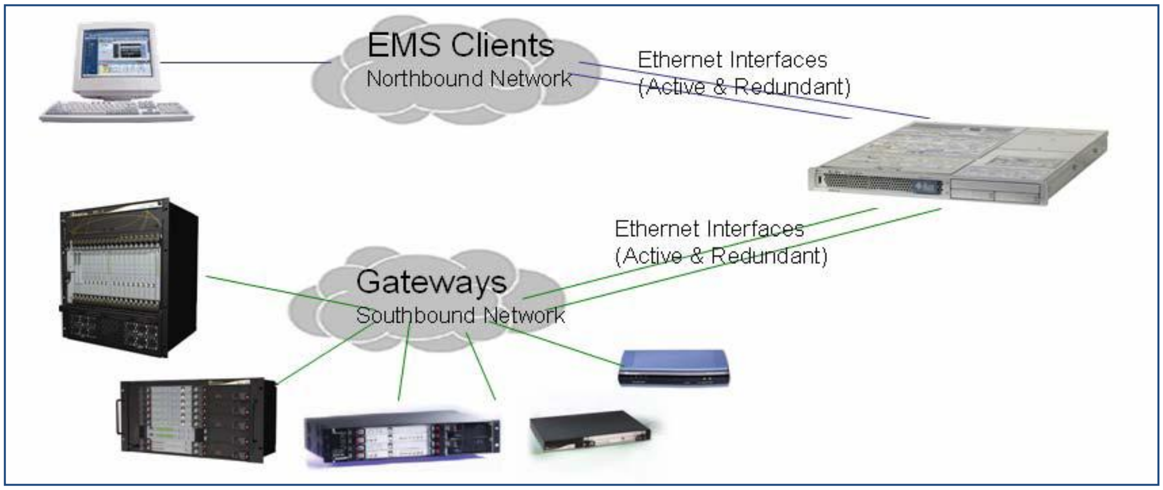

Figure 2 - AudioCodes EMS

1.2 Management by Web

The AudioCodes EMS is commonly used in solutions which involve multiple gateways or in high availability or high security applications. Each gateway is also equipped with a configuration and management interface accessible via a standard web browser via HTTP or HTTPS. The web design of the various digital gateway products in general share a common web design and the references within this configuration guide are applicable to any of the deployed products. The web interface provides a simple search capability which makes it easy to find the various parameters when some or all of the name is known (for example entering "Alt Routing").

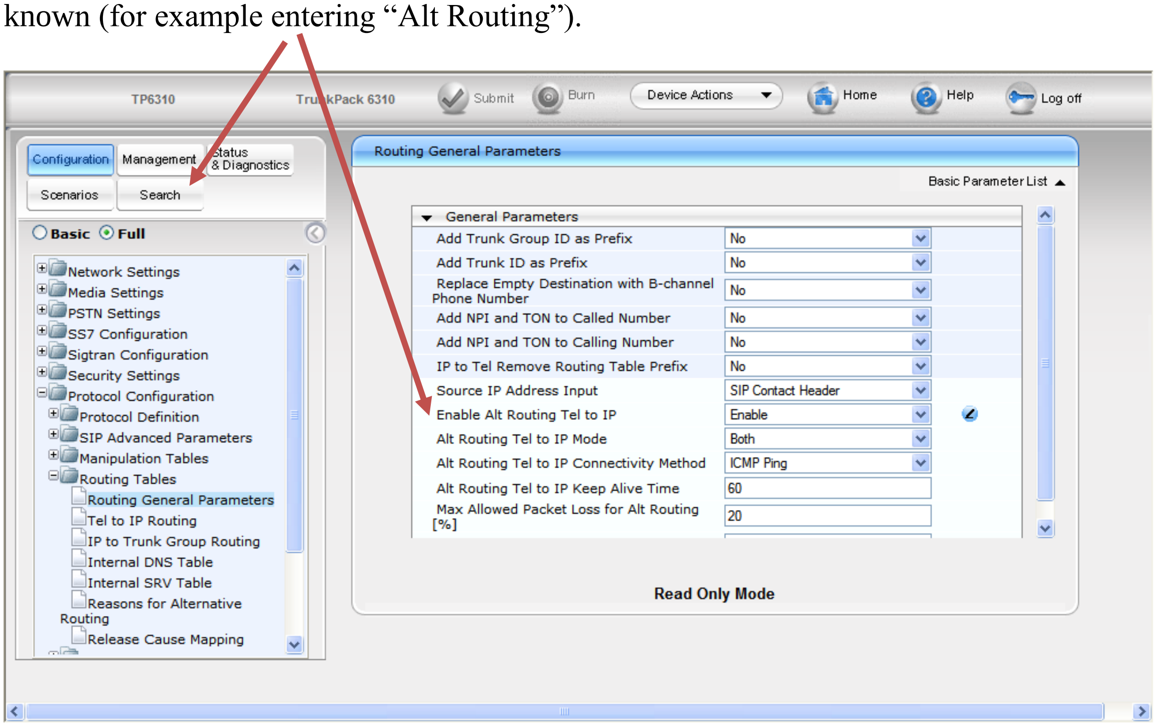

Figure 3 - Sample Web Configuration screen

1.3 Management by EMS

The EMS supports a Help menu option which provides a search similar to the Web GUI. There is a convention used in this document to indicate navigation for menus, sub-menus, and tab headings (both top of frame or side of frame). The example shown below would be denoted in bold as follows:

SIP->Routing->Route Modes

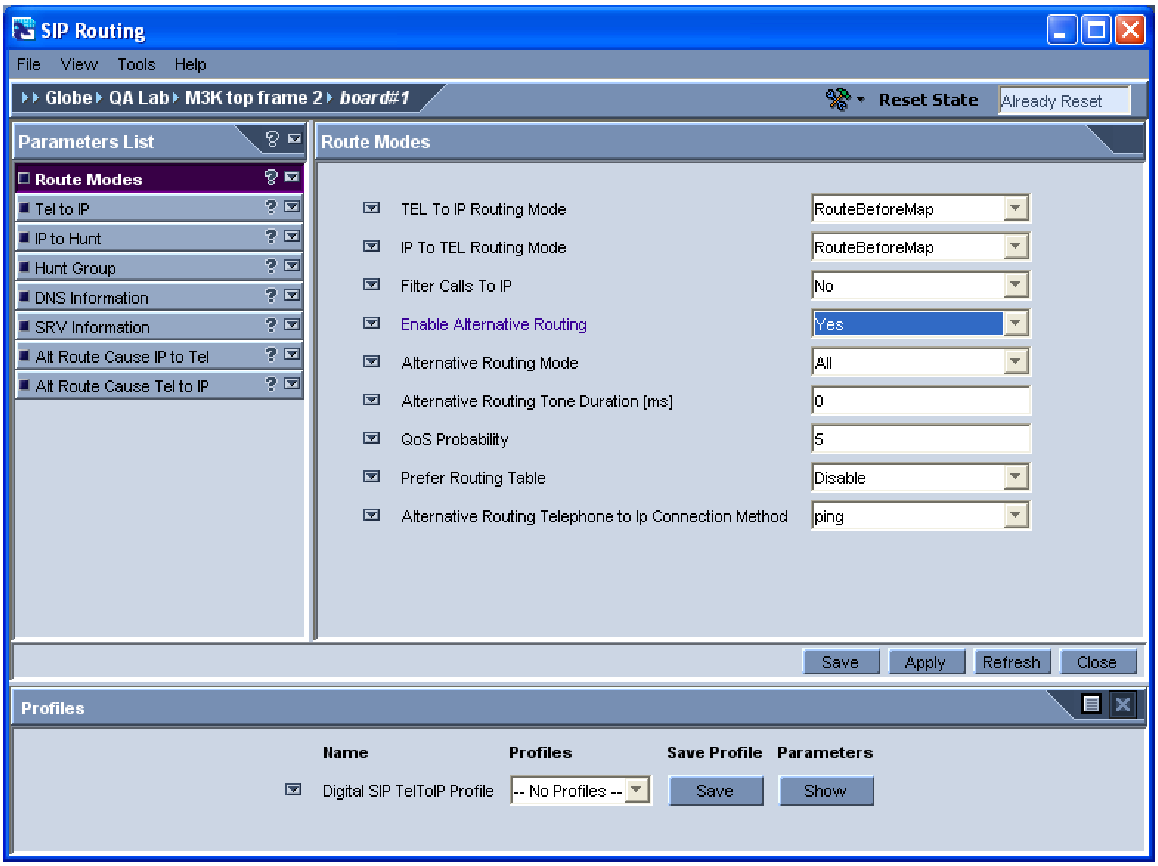

Figure 4 - Sample EMS Screen

1.4 AudioCodes Configuration File Basics

In addition to the Web and EMS view, each configuration parameter is part of an ASCII file known as the initialization file or INI file (a.k.a. the conf file). Most parameters follow in a simple NAME = VALUE, line by line sequence as shown below:

```
ALTROUTINGTEL2IPENABLE = 1
```

In general, the value 0 indicates FALSE or NO and 1 indicates TRUE or YES.

For more complex configuration parameters, the INI file supports tables. Table entries in the document are indicated by using an underscore notation as shown below:

```
SNMPManagerTableIP_2 = 0.0.0.0
SNMPManagerIsUsed_2 = 0
```

Because the INI file numbering is zero-based, this example shows that the third SNMP manager in the table has not been enabled.

The INI file can be easily retrieved from the media gateway by either the WEB or EMS.

IMPORTANT NOTE: The INI file only shows a parameter value when it differs from the default. When a parameter is not found in the INI file, it can be assumed to be at the default value.

1.5 Document Conventions

In most cases, a screen capture of the referenced parameter will be provided for both the web GUI and the EMS view. The user should reference the screen which matches the management mode for the deployment (either Web or EMS). For clarity, each referenced parameter will first be discussed by referencing the INI file name (for example, ALTROUTINGTEL2IPENABLE). The reference will be followed by a short definition which will include a description of valid ranges and default values.

1.6 AudioCodes Software Releases

Each AudioCodes software release will have four numerical indicators, for example 6.2.34.4. The first two numbers indicate the major release. This indicator progresses as follows: 5.0, 5.2, 5.4, 5.6, 5.8, 6.0, 6.2, etc. The second two indicators represent the minor release. In general, the minor release indicators are not usually significant to end users.

The software version can be easily determined from either the web or EMS views. The software version can be referred to with leading or trailing zeroes which can be ignored. For example, the web GUI may display a version 06.20.034.004 for the 6.2 version referenced above. In some cases, a letter designation will be included, as in 6.20A.043.002.

This manual in general is applicable to the 6.0 and 6.2 versions. The screens are captured from the 6.2 version and may vary slightly from version to version.

1.7 References

The last 3 digits of the document reference number (beginning with LTRT) vary by release. These digits are replaced by xxx in the table below.

LTRT-59xxx MP-11x and MP-124 SIP Installation Manual Ver. 6.x.pdf
LTRT-65xxx_MP-11x_and_MP-124_SIP_User's_Manual_Ver_6.x.pdf
LTRT-66xxx CPE Configuration Guide for IP Voice Mail Ver 6..x.pdf
LTRT-26xxx SIP CPE Release Notes Ver. 6.x.pdf

LTRT-29xxx Errata-Addendum for SIP CPE Documentation 6.x.pdf
LTRT-52xxx SIP CPE Product Reference Manual Ver 6.x.pdf
LTRT-94xxx EMS Server Installation, Operation, and Maintenance (IOM) Manual v6.x.pdf
LTRT-91xxx EMS User's Manual v6.x.pdf
LTRT-90xxx EMS Release Notes v6.x.pdf

A complete set of applicable AudioCodes documentation is shipped by CD/DVD with the product. Updated versions (as new major releases come out) can be obtained by registering at www.audiocodes.com in the Support section.

2. FAX Configuration Settings

Per the requirements of the tested solution, the values shown in this guide are in general the defaults for the basic T.38 FAX. Because the AudioCodes MGW supports several alternate ways of transporting FAX and Modem data (Transparent, Transparent with events, Bypass, NSE, V.152, etc.), there are many settings and combinations of settings that are best left to solving specific situations. Complete information about these non-standard values is included in the User’s manual referenced above.

If you are using the default settings and you are still experiencing problems with T.38 FAX, check first the end to end call flow. In previous releases, most of these FAX issues result from issues revolve around routing, and or the proper server settings for Avaya TON/NPI headers.

DO NOT CHANGE FAX SETTINGS THAT ARE NOT WELL UNDERSTOOD in an effort to fix a specific FAX failure since that will cause an unsupported configuration to be placed in service with possible support issues down the road.

Additionally, make sure that the software release running on the Audiocodes MP-1xx matches exactly the approved load line up for the release. Occasionally, the devices shipped from stock have been manufactured prior to the final release date or have been in a warehouse for so long that there is a good chance that the version is not correct. IT IS THE RESPONSIBILITY OF THE INSTALLER to check and verify that the software placed in service is in accordance with the solution as it was tested for interoperability.

2.1 SIP General Parameters

The values shown are basic T.38 examples. Other than some that are highlighted below, most of the default values are usually used in the SIP General Parameters.

```
IsFaxUsed = 1
```

Fax Signaling Method
In the T.38 Relay Mode, upon detection of a fax signal the terminating device negotiates T.38 capabilities using a SIP Re-INVITE message. If the far end device doesn't support T.38, the fax fails. In this mode, the parameter FaxTransportMode is ignored. To configure T.38 mode using SIP Re-INVITE messages, make the following setting:

> [0] No Fax = No fax negotiation using SIP signaling. Fax transport method is according to the parameter FaxTransportMode (default).
> **[1] T.38 Relay = Initiates T.38 fax relay.**
> [2] G.711 Transport = Initiates fax / modem using the coder G.711 Alaw/ µ-law with adaptations (refer to User’s Manual).
> [3] Fax Fallback = Initiates T.38 fax relay. If the T.38 negotiation fails, the device re-initiates a fax session using the coder G.711 A-law/µ-law with adaptations (refer to the Note in the User Manual).

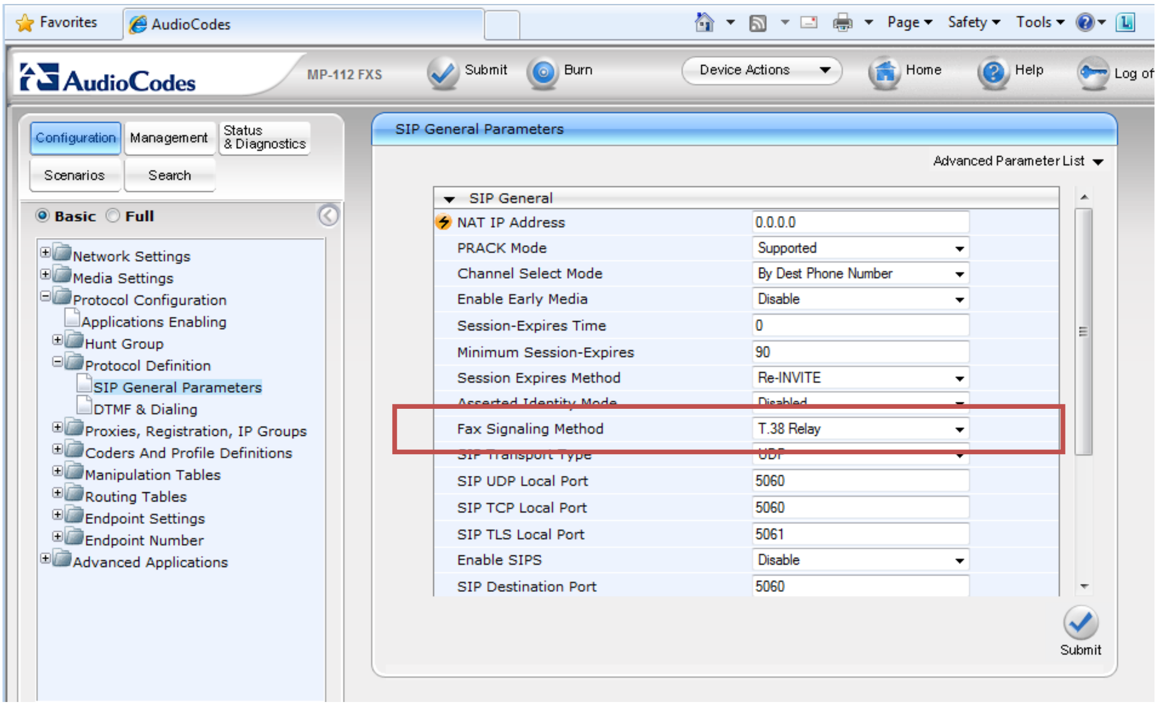

Figure 5 – Web view of SIP General Settings (T.38 Relay Selected)

For T.38 signaling, log into the web GUI (http:<your_ip_address>, using the user id "Admin" (case sensitive with no quotes). The password will be the default, "Admin" (case sensitive with no quotes), or whatever the installer may have chosen. After logging in, navigate to **Protocol Definition** and select the **SIP General Parameters** tab and the make sure that the **Fax Signaling Method** is set to "T.38 Relay".

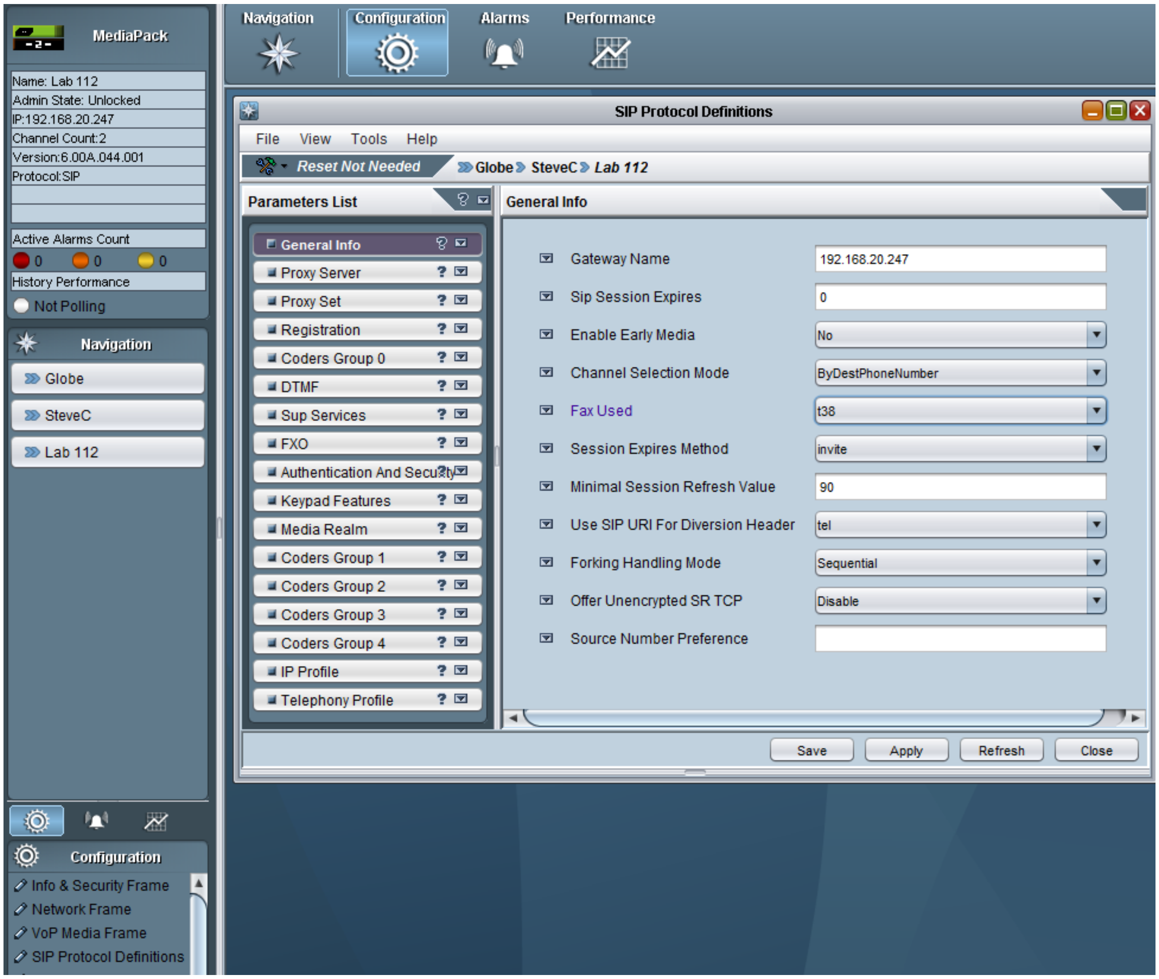

Figure 6 – EMS View SIP General Settings (T.38 Relay Selected)

For T.38 signaling in the EMS, navigate to your MP-11x (“Lab 112” shown) and under **Configuration**, at the bottom, choose **SIP Protocol Definition**. In the **General Info** frame, make sure that **Fax Used** is set to “T.38”.

2.2 FAX and Modem Settings

There are several other inter-related parameters involved in changing the T.38 behavior from the most common settings. In most situations, these are best left at the default values also. With the default settings, as shown in the examples below (and described previously), the terminating endpoint detects the FAX tone and is responsible for sending the notifications which contain a modified SDP with a T.38 codec.

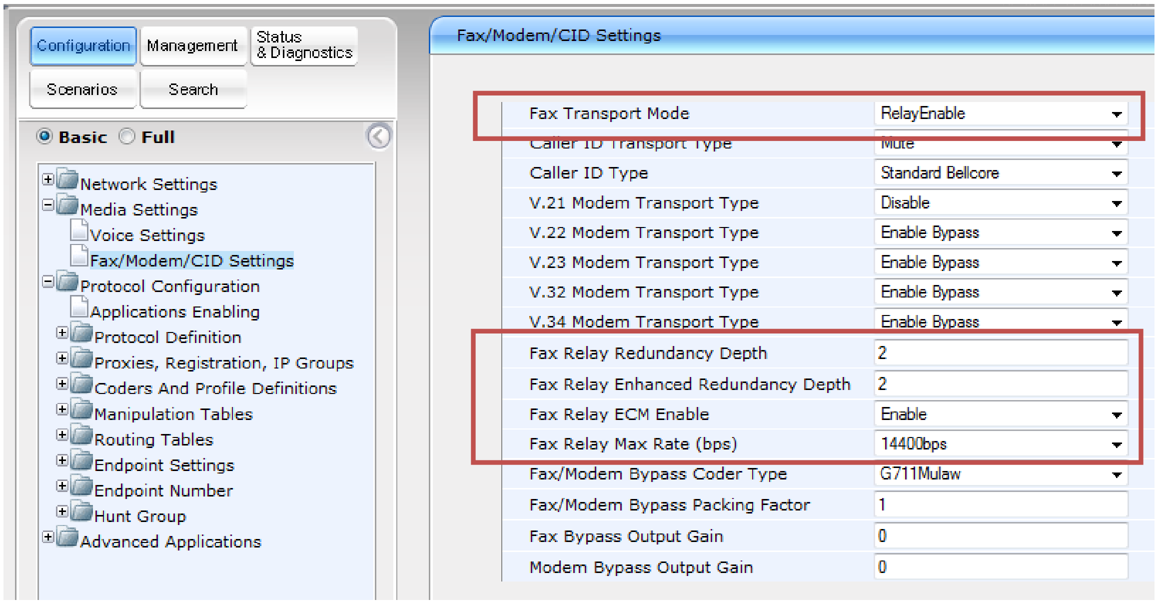

Figure 7 – Web View FAX/Modem/Channel ID settings (T.38 Relay Selected)

For T.38 signaling, in the web choose **Fax/Modem/CID Settings** under **Media Settings**.

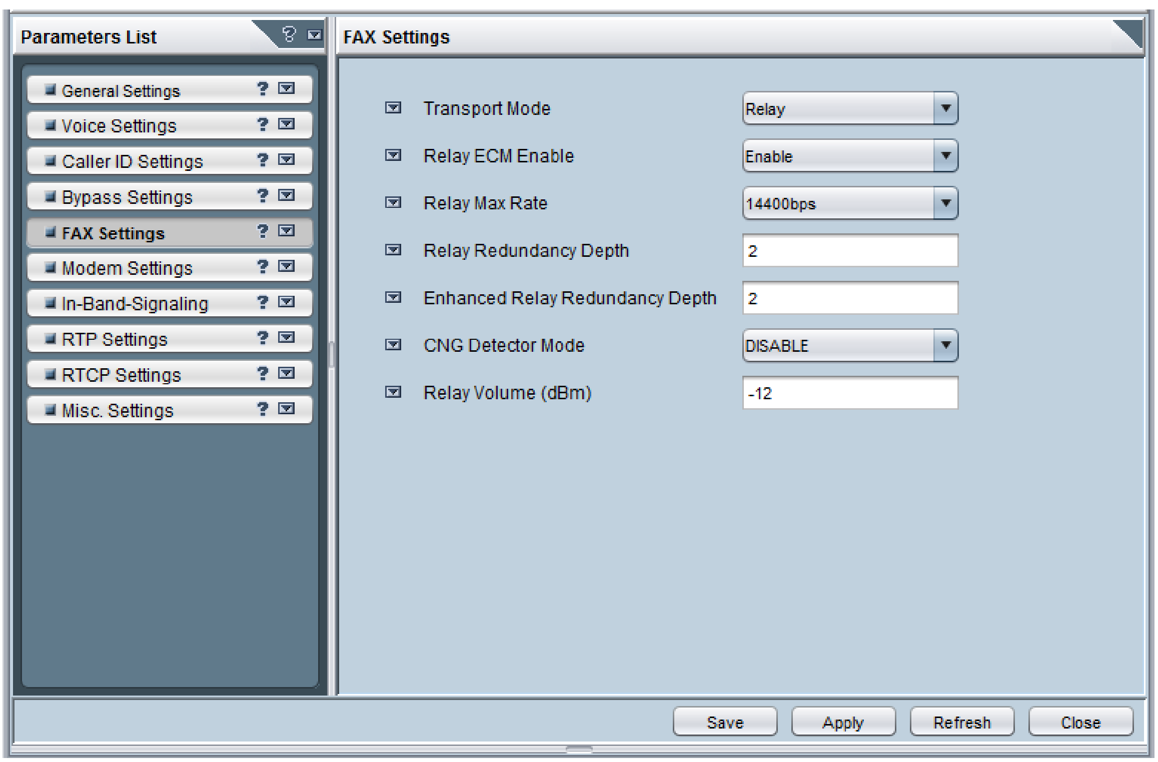

Figure 8 – EMS View FAX settings (T.38 Relay Selected)

For T.38 signaling in the EMS, under **Configuration**, at the bottom, choose **Media Settings**. In the **FAX Settings** frame, refer to the following parameter description and use the settings in the example.

Fax Transport Mode Sets the Fax transport
0 = disable (transparent mode)
1 = relay, (default, to be used for T.38)
2 = bypass.

```
FaxTransportMode = 1
```

Fax Relay Enhanced Redundancy Depth
 0 to 4 (default =0)
Number of repetitions to be applied to each fax control packet

```
FaxRelayEnhancedRedundancyDepth = 2
```

Fax Relay Redundancy Depth
 0 to 2 (default =0)
Number of repetitions to be applied to each fax relay payload when transmitting to network

```
FaxRelayRedundancyDepth = 2
```

Fax Relay Max Rate
 Limits the maximum rate at which fax messages are transmitted.
 0 = 2.4 kbps
1 = 4.8 kbps
2 = 7.2 kbps
3 = 9.6 kbps
4 = 12.0 kbps
5 = 14.4 kbps, (default)

```
FaxRelayMaxRate = 5
```

Fax Relay ECM Enable
0 = Disable using ECM mode during Fax Relay
1 = Enable using ECM mode during Fax Relay. (default)

```
FaxRelayECMEnable = 1
```

CNG Detector Mode
The terminating gateway sends T.38 packets immediately after the T.38 capabilities are negotiated in SIP. However, the originating device by default, sends T.38 (assuming the T.38 capabilities are negotiated in SIP) only after it receives T.38 packets from the remote device. This default behavior cannot be used when the originating device is located behind a firewall that blocks incoming T.38 packets on ports that have not yet

received T.38 packets from the internal network. To resolve this problem, the device should be configured to send CNG packets in T.38 upon CNG signal detection

0 – don't use CNG detector mode (default)
1 – Relay
2 - T.38 will start after CNG detection (applicable only for SIP and not recommended)

```
CNGDetectorMode = 0
```

IMPORTANT NOTE: The INI file only shows a parameter value when it differs from the `default value`. When a parameter is not found in the INI file, it can be assumed to be at the `default value`.

2.3 Related Settings

Dynamic Jitter Buffer Minimum Delay
0 to 150 msec (default = 70)
The Dynamic Jitter Buffer Minimum Delay could vary widely based upon the quality of the IP network. A setting of 10 is recommended as a starting point with the possibility that increases (in steps of 10 msec) may be necessary if packet loss, delays or bursting cause issues in the voice stream.

```
DJBufMinDelay = 10
```

Dynamic Jitter Buffer Optimization Factor
0 to 13 (default = 7)
Dynamic jitter buffer frame error/delay optimization.

```
DJBufOptFactor = 10
```

Note that this setting dynamically shifts to 13 during fax and modem calls.

2.4 Using Profiles with FAX

Profile configuration parameters are used to create the IP Profile table. Each IP Profile can be referenced by its ID to override those which are typically configured separately using their individual "global" parameter settings. This allows you to assign these IP Profiles to **Tel-to-IP Routing** rules, **IP-to-Tel Routing** rules, or **IP Groups** to provide different behavior for different phone numbers, prefixes, or IP destinations. In simple configurations, these profiles are not used and the global settings (described above) will be in effect (and the Profile ID will be zero). IF THERE IS NO NEED TO HAVE DIFFERING BEHAVIOR FOR DIFFERENT IP DESTINATIONS, YOU SHOULD NOT NEED TO USE PROFILES. In these cases, this section can be disregarded.

When using non-zero profiles, take care that you make the correct settings in each profile to enable T.38. As example, in the **IP to Trunk Group Routing** table, we may begin to define rows which have non-zero **IP Profile ID**. IpProfile_IsFaxUsed should be duplicated with the parm setting "IsFaxUsed". This is becuase the Profile setting takes precedence over the "global" settings for calls matching the profile. If users have set the "global" parameter correctly, it can be difficult to determine that the profile setting overwrote the IsFaxUsed parameter setting to another value since the call in question used a profile (although there will be a SYSLOG for this if the debug level is set). This is true for other parameters duplicated in profiles, where the profiles setting take precedence over the single parm setting in the board.ini file.

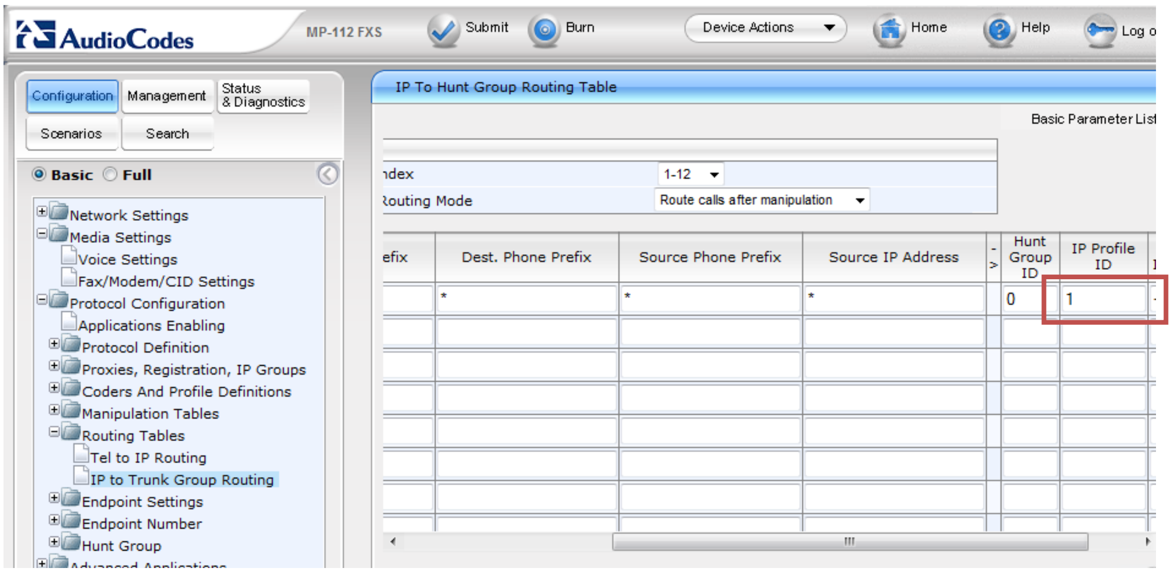

Figure 9 – Web View IP to Trunk Group Routing (non-zero IP Profile ID)

In this case, make sure to duplicate the desired Fax settings for each Profile. Under **Coders and Profile Definitions**, choose **IP Profile Settings**, set the **Profile ID** (in this example 1), and set the **Fax Signaling Method** (`IpProfile_IsFaxUsed`) to "T.38 Relay". Note that many other parameters require similar attention (notably `IpProfile_JitterBufMinDelay` and `IpProfile_JitterBufOptFactor`).

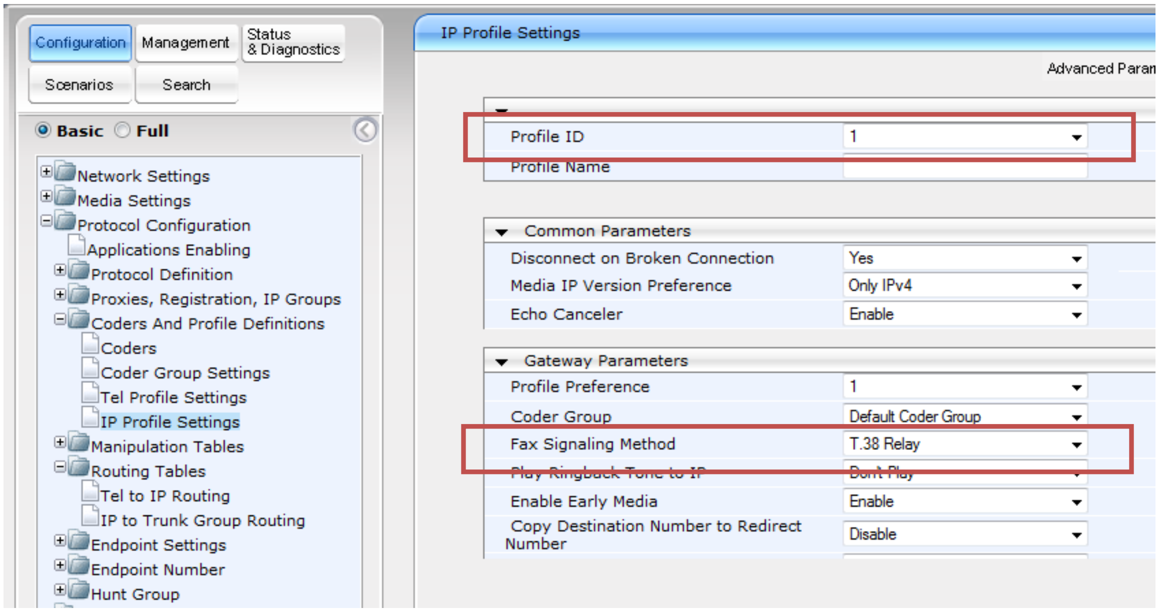

Figure 10 – Web View IP Profile Settings (Profile ID 1 shown with T,38 relay)

Repeat this step for each **IP Profile ID** used.

2.5 Advanced FAX Settings

Fax Transmission behind NAT:

In version 6.2 and above, a new feature provides support for transmission from fax machines (connected to the device) located inside a Network Address Translation (NAT). Generally, the firewall blocks T.38 (and other) packets received from the WAN, unless the device behind the NAT sends at least one IP packet from the LAN to the WAN through the firewall. If the firewall blocks T.38 packets sent from the termination IP fax, the fax fails.

To overcome this, the device sends No-Op (“no-signal”) packets to open a pinhole in the NAT for the answering fax machine. The originating fax does not wait for an answer, but immediately starts sending T.38 packets to the terminating fax machine.

This feature is enabled using the new parameter, **T38FaxSessionImmediateStart**. The No-Op packets are enabled using the existing **NoOpEnable** and **NoOpInterval** parameters.

2.6 Known Constraints

Note that the fax counters (Attempted Fax Calls Counter and Successful Fax Calls Counter) in the 'Status & Diagnostics' page do not function correctly in 6.2 versions (refer to the Release Notes for more information).

3. Troubleshooting

3.1 AudioCodes Syslog Application

AudioCodes also provides a Windows based syslog application in the Utilities section of the installation disk. This is described in further detail in the Troubleshooting section at the end of this document. It is mentioned in this section since it can be useful during the installation to confirm the basic IP configuration of the gateway.

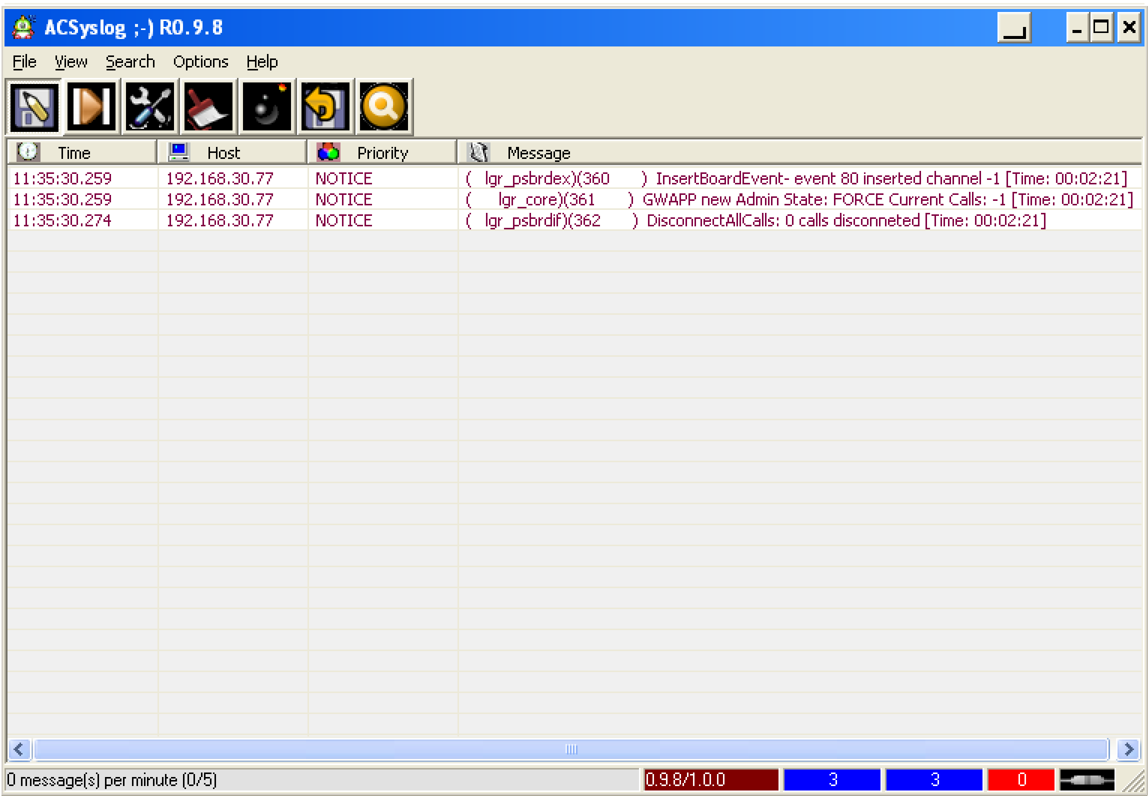

Figure 11 - AudioCodes Syslog Application

The following INI file configuration parameters are used:

```
SyslogServerIP = 192.168.10.57
```

~ SyslogServerIP: This parameter defines the IP address in dotted format notation. e.g. 192.10.1.255, Range = Legal IP address.

```
EnableSyslog = 1
```

~ EnableSyslog: This parameter is used to enable the Syslog protocol log.
1 = Activate, 0 = Deactivate.

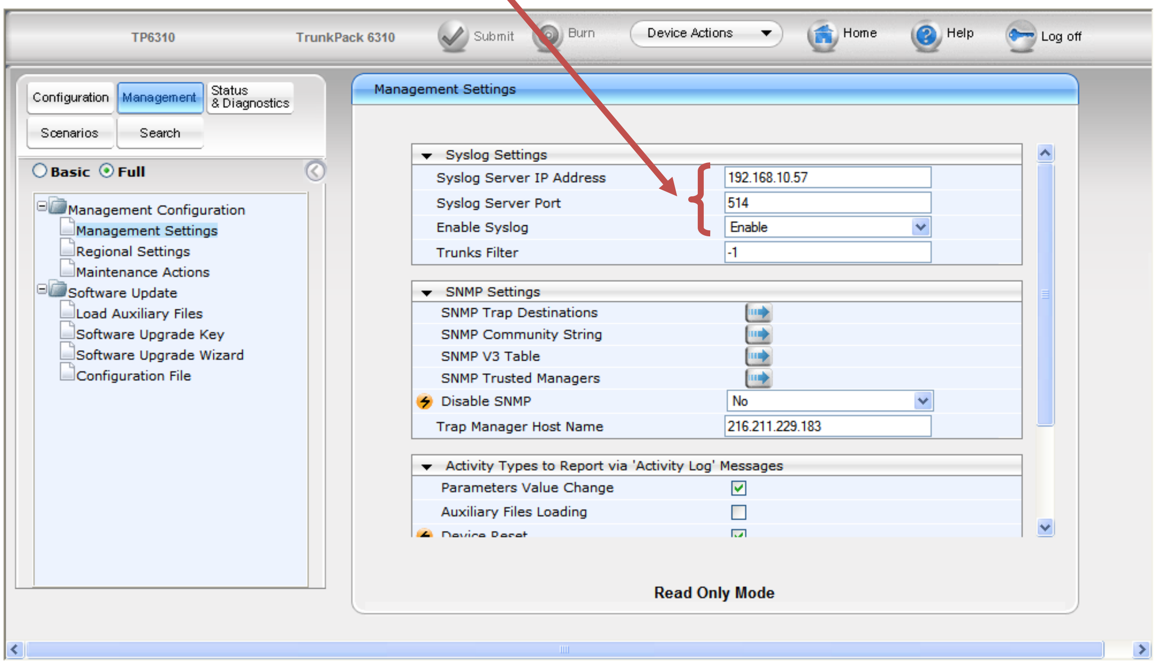

Figure 12 - Syslog Server IP Address Configuration by web

3.2 Debug Mode Syslogs

During initial configuration and during troubleshooting sessions, it can be helpful to enable Debug Mode level syslogs. Logging can be modified between five normal debug levels. In practice, most users either disable the feature or use level 5 (not using intermediate levels).

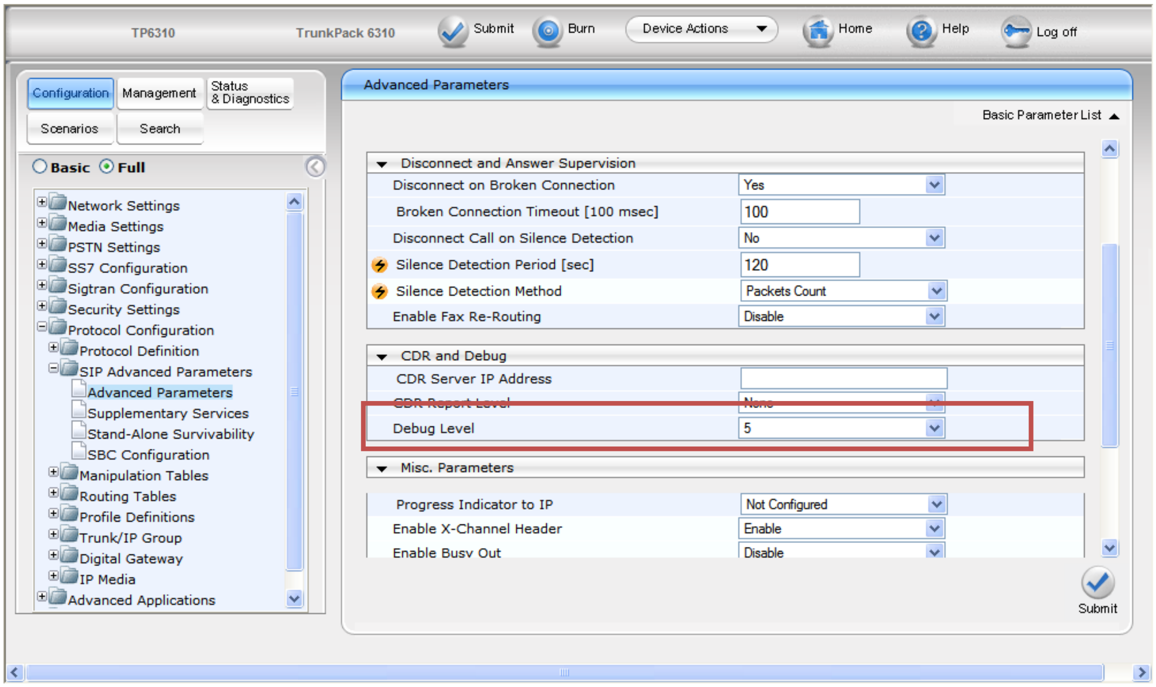

Figure 13 - Web View of Debug Level

```
GwDebugLevel = 5
```

~ Debug Level - Syslog debug logging level.

[0] 0 = Debug is disabled (default).

[1] 1 = Flow debugging is enabled.

[5] 5 = Flow, device interface, stack interface, session manager, and device interface expanded debugging are enabled (including SIP messaging).

[7] 7 = This option is recommended when the device is running under "heavy" traffic. In this mode: The Syslog debug level automatically changes between level 5, level 1, and level 0, depending on the device's CPU consumption so that VoIP traffic isn't affected.

Syslog messages are bundled into a single UDP packet, after which they are sent to a Syslog server (bundling size is determined by the MaxBundleSyslogLength parameter). Bundling reduces the number of UDP Syslog packets, thereby improving CPU utilization.

Note that when this option is used, in order to read Syslog messages with Wireshark, a special plug-in (i.e., acsyslog.dll) must be used. Once the plug-in is installed, the Syslog messages are decoded as "AC SYSLOG" and are dispalyed using the ‘acsyslog’ filter instead of the regular ‘syslog’ filter.

Notes:
This parameter is typically set to 5 if debug traces are required.
However, in cases of heavy traffic, option 7 is recommended.
Options 2, 3, 4, and 6 are not recommended.

When working with the Gateway Debug Level, be sure to return the setting to DISABLED. The volume of syslog messaging in Debug Mode can affect call processing at high levels of traffic.

```
GwDebugLevel = 0
```

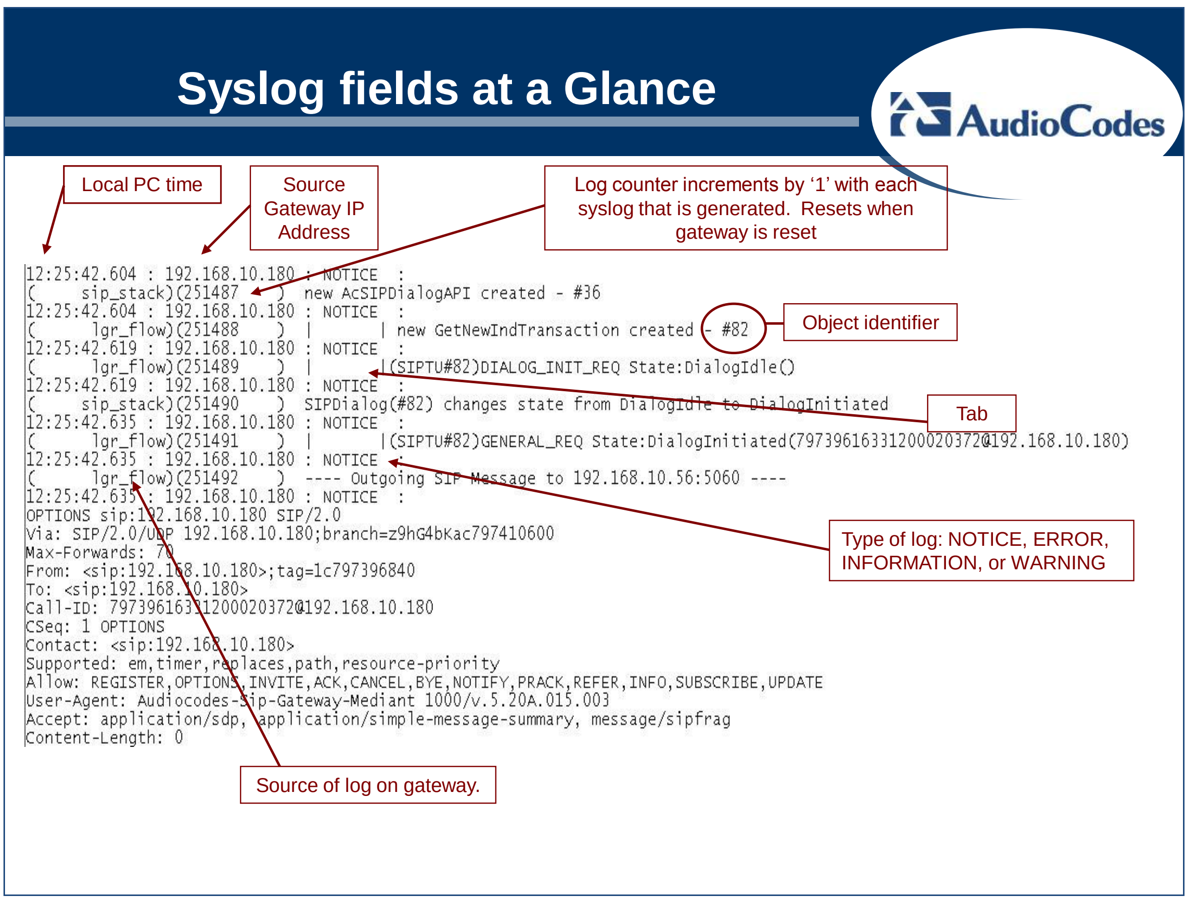

Figure 14 - Syslog fields at a glance (Debug Mode 5)

3.3 Debug Recording Overview

AudioCodes TP Board Debug Recording traces are a powerful debugging tool that provides ability to capture traffic being handled by a specific Media Gateway board.

They may be used for analyzing different inter-op scenarios or specific Media Gateway board malfunctions:

- Voice quality issues
- TDM connectivity issues
- Signaling issues

The tool enables forwarding of the specific packets being handled by the Media Gateway board to the user-specified remote IP address. The remote IP address may belong to one of the following:

- External PC (with Wireshark)
- EMS (using snoop)

Flexible user-defined filtering rules apply to selected traffic to be forwarded. For example, it is possible to select traffic between the Media Gateway board and specific remote Media Gateway. Alternatively, select traffic that belongs to the specific call. Multiple filtering rules may be applied simultaneously.
When Debug Recording Traces are forwarded to the External PC, simply use a packet capturing application (for example Ethereal or Wireshark) for trace collection. An AudioCodes plug-in for Wireshark is supplied with the gateway on the distribution media (CD/DVD) which is used for analyzing the trace (or contact an AudioCodes support representative).

Appendix A Sample Configuration File

The following INI file is provided as an example configuration:

```
;**************
;** Ini File **
;**************

;Board: MP-112 FXS
;Serial Number: 3224615
;Slot Number: 1
;Software Version:
;DSP Software Version:
;Board IP Address:
;Board Subnet Mask:
;Board Default Gateway:
;Ram size: 32M   Flash size: 8M
;Num of DSP Cores: 1  Num DSP Channels: 4
;Profile: NONE
;------------------------------------------


[SYSTEM Params]

DNSPriServerIP = 10.0.1.4
DNSSecServerIP = 0.0.0.0
SyslogServerIP = 10.1.1.89
VXMLFIleName = ''
NTPServerIP = 0.0.0.0
NTPServerUTCOffset = 0
NTPUpdateInterval = 86400

[BSP Params]

PCMLawSelect = 3
RoutingTableHopsCountColumn = 0, 0, 0, 0, 0, 0, 0, 0, 0, 0,
0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0

[Analog Params]

PolarityReversalType = 1
MinFlashHookTime = 100
FXSLoopCharacteristicsFilename = 'MP11x-02-1-FXS_16KHZ.dat'

[ControlProtocols Params]
```

```
[MGCP Params]


[MEGACO Params]

DIGITMAPPING = ''
EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 0
EP_Num_3 = 0
EP_Num_4 = 0

[Voice Engine Params]

CallProgressTonesFilename = 'usa_tones_13.dat'
EnableEchoCanceller = 1
DTMFTransportType = 2
FaxTransportMode = 1
V21ModemTransportType = 0
FaxRelayRedundancyDepth = 2
FaxRelayEnhancedRedundancyDepth = 2
FaxModemBypassCoderType = 1
CNGDetectorMode = 0
RFC2833TxPayloadType = 101
RFC2833RxPayloadType = 101
DTMFDetectorSensitivity = 1
NTEMaxDuration = -1

[WEB Params]

LogoWidth = '145'

[SIP Params]

MAXDIGITS = 12
ALWAYSUSEROUTETABLE = 0
TIMEBETWEENDIGITS = 4
ISPROXYUSED = 1
ISREGISTERNEEDED = 1
AUTHENTICATIONMODE = 0
ROUTEMODEIP2TEL = 1
CHANNELSELECTMODE = 0
GWDEBUGLEVEL = 5
ENABLEEARLYMEDIA = 1
ISUSERPHONE = 0
PROXYNAME = 'msrn.audiocodes.com'
REGISTRARIP = 'msrn.audiocodes.com'
```

```
SIPGATEWAYNAME = 'msrn.audiocodes.com'
RXDTMFOPTION = 0
ASSERTEDIDMODE = 0
ISUSERPHONEINFROM = 0
ADDTON2RPI = 0
USESOURCENUMBERASDISPLAYNAME = 0
ISFAXUSED = 1
SUBSCRIPTIONMODE = 0
SIPTRANSPORTTYPE = 0
GWREGISTRATIONNAME = 'msrn.audiocodes.com'
REGISTRARNAME = 'msrn.audiocodes.com'
USEDISPLAYNAMEASSOURCENUMBER = 0
USETELURIFORASSERTEDID = 1
DNSQUERYTYPE = 0
PROXYDNSQUERYTYPE = 0
ADDPHONECONTEXTASPREFIX = 1
REDUNDANTROUTINGMODE = 0
REGISTRARTRANSPORTTYPE = 0

[IPsec Params]


[SNMP Params]


;
;  *** TABLE DspTemplates ***
; This table contains hidden elements and will not be
exposed.
; This table exists on board and will be saved during
restarts
;

;
;  *** TABLE PREFIX ***
;
;

[ PREFIX ]
FORMAT PREFIX_Index = PREFIX_DestinationPrefix,
PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileId,
PREFIX_MeteringCode, PREFIX_DestPort, PREFIX_SrcIPGroupID,
PREFIX_DestHostPrefix, PREFIX_DestIPGroupID,
PREFIX_SrcHostPrefix, PREFIX_TransportType,
PREFIX_SrcTrunkGroupID;
PREFIX 0 = 10, 10.1.10.10, *, 0, 255, 0, -1, , -1, , -1, -
1;
```

```
PREFIX 1 = 20, 10.1.10.11, *, 0, 255, 0, -1, , -1, , -1, -
1;

[ \PREFIX ]

;
;  *** TABLE CoderName ***
;
;

[ CoderName ]
; ** NOTE: Changes were made to active configuration.
; **       The data below is different from current values.
FORMAT CoderName_Index = CoderName_Type,
CoderName_PacketInterval, CoderName_rate,
CoderName_PayloadType, CoderName_Sce;
CoderName 0 = g711Ulaw64k, 20, 0, 255, 0;
CoderName 1 = g711Alaw64k, 20, 0, 255, 0;

[ \CoderName ]

;
;  *** TABLE TrunkGroup ***
;
;

[ TrunkGroup ]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum,
TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel,
TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber,
TrunkGroup_ProfileId, TrunkGroup_LastTrunkId,
TrunkGroup_Module;
TrunkGroup 0 = 0, 255, 1, 1, 2222277, 1, 255, 255;

[ \TrunkGroup ]

;
;  *** TABLE NumberMapIp2Tel ***
;
;

[ NumberMapIp2Tel ]
FORMAT NumberMapIp2Tel_Index =
NumberMapIp2Tel_DestinationPrefix,
NumberMapIp2Tel_SourcePrefix,
NumberMapIp2Tel_SourceAddress, NumberMapIp2Tel_NumberType,
NumberMapIp2Tel_NumberPlan, NumberMapIp2Tel_RemoveFromLeft,
```

```
NumberMapIp2Tel_RemoveFromRight,
NumberMapIp2Tel_LeaveFromRight, NumberMapIp2Tel_Prefix2Add,
NumberMapIp2Tel_Suffix2Add,
NumberMapIp2Tel_IsPresentationRestricted,
NumberMapIp2Tel_SrcTrunkGroupID,
NumberMapIp2Tel_SrcIPGroupID;
NumberMapIp2Tel 1 = +, *, *, 255, 255, 1, 0, 7, , , 255, -
1, -1;

[ \NumberMapIp2Tel ]

;
;  *** TABLE NumberMapTel2Ip ***
;
;

[ NumberMapTel2Ip ]
FORMAT NumberMapTel2Ip_Index =
NumberMapTel2Ip_DestinationPrefix,
NumberMapTel2Ip_SourcePrefix,
NumberMapTel2Ip_SourceAddress, NumberMapTel2Ip_NumberType,
NumberMapTel2Ip_NumberPlan, NumberMapTel2Ip_RemoveFromLeft,
NumberMapTel2Ip_RemoveFromRight,
NumberMapTel2Ip_LeaveFromRight, NumberMapTel2Ip_Prefix2Add,
NumberMapTel2Ip_Suffix2Add,
NumberMapTel2Ip_IsPresentationRestricted,
NumberMapTel2Ip_SrcTrunkGroupID,
NumberMapTel2Ip_SrcIPGroupID;
NumberMapTel2Ip 1 = 222, *, *, 255, 255, 0, 0, 255, +, ,
255, -1, -1;
NumberMapTel2Ip 2 = 9, *, *, 255, 255, 0, 0, 255, +, , 255,
-1, -1;

[ \NumberMapTel2Ip ]

;
;  *** TABLE PstnPrefix ***
;
;

[ PstnPrefix ]
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix,
PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix,
PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix,
PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = *, 0, *, *, 1, -1, , ;
```

```
[ \PstnPrefix ]

;
;  *** TABLE Dns2Ip ***
;
;

[ Dns2Ip ]
FORMAT Dns2Ip_Index = Dns2Ip_DomainName,
Dns2Ip_FirstIpAddress, Dns2Ip_SecondIpAddress,
Dns2Ip_ThirdIpAddress, Dns2Ip_FourthIpAddress;
Dns2Ip 0 = msrn.audiocodes.com, 172.16.3.103, 0.0.0.0,
0.0.0.0, 0.0.0.0;

[ \Dns2Ip ]

;
;  *** TABLE ProxyIp ***
;
;

[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress,
ProxyIp_TransportType, ProxyIp_ProxySetId;
ProxyIp 0 = msrn.audiocodes.com, 0, 0;

[ \ProxyIp ]

;
;  *** TABLE TxDtmfOption ***
;
;

[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]

;
;  *** TABLE TelProfile ***
;
;

[ TelProfile ]
```

```
FORMAT TelProfile_Index = TelProfile_ProfileName,
TelProfile_TelPreference, TelProfile_CodersGroupID,
TelProfile_IsFaxUsed, TelProfile_JitterBufMinDelay,
TelProfile_JitterBufOptFactor, TelProfile_IPDiffServ,
TelProfile_SigIPDiffServ, TelProfile_DtmfVolume,
TelProfile_InputGain, TelProfile_VoiceVolume,
TelProfile_EnableReversePolarity,
TelProfile_EnableCurrentDisconnect,
TelProfile_EnableDigitDelivery, TelProfile_EnableEC,
TelProfile_MWIAnalog, TelProfile_MWIDisplay,
TelProfile_FlashHookPeriod, TelProfile_EnableEarlyMedia,
TelProfile_ProgressIndicator2IP,
TelProfile_TimeForReorderTone, TelProfile_EnableDIDWink,
TelProfile_IsTwoStageDial, TelProfile_DisconnectOnBusyTone,
TelProfile_EnableVoiceMailDelay, TelProfile_DialPlanIndex;
TelProfile 1 = , 1, 0, 1, 10, 10, 46, 40, -11, 0, 0, 0, 0,
0, 1, 0, 0, 700, 1, -1, 255, 0, 1, 1, 1, -1;

[ \TelProfile ]

;
;  *** TABLE IpProfile ***
;
;

[ IpProfile ]
FORMAT IpProfile_Index = IpProfile_ProfileName,
IpProfile_IpPreference, IpProfile_CodersGroupID,
IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay,
IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ,
IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType,
IpProfile_NSEMode, IpProfile_IsDTMFUsed,
IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia,
IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller,
IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection,
IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption,
IpProfile_RxDTMFOption, IpProfile_EnableHold,
IpProfile_InputGain, IpProfile_VoiceVolume,
IpProfile_AddIEInSetup,
IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference,
IpProfile_TranscodingMode;
```

```
IpProfile 1 = , 1, 0, 1, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0,
0, 0, 1, -1, 1, 0, 0, -1, 1, 0, 0, 0, 1, 0, 0, , -1, 0, 0;

[ \IpProfile ]

;
;  *** TABLE ProxySet ***
;
;

[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRD;
ProxySet 0 = 0, 60, 0, 0, 0;

[ \ProxySet ]

;
;  *** TABLE InterfaceTable ***
;
;

[ InterfaceTable ]
FORMAT InterfaceTable_Index =
InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName;
InterfaceTable 0 = 6, 10, 172.16.3.115, 24, 172.16.3.1, 1,
O+M+C;

[ \InterfaceTable ]
```

Appendix B Sample Debug SYSLOG Receiving FAX

In the sample SYSLOG below (debug level 5), an MP-114 is at 65.246.127.150. Some of the more significant entries are highlighted in **BOLD** font.

The voice call comes into the MP-114. The voice call is set up as G726 (codec 2) at 12:14:51.624 after the familiar INVITE, TRYING, RINGING, PRACK, OK-PRACK, OK-INVITE.

At time 12:14:54.148, we can see that the MP-114 detects the FAX tones.

The MP-114 sends a RE-INVITE to shift the call to t38 at a speed of 14400. The RTP stream shifts from the voice ports, in this example 6000, to the t38 ports, 6002. This N+2 pattern with the voice port having a power of 10 (that is 6000, 6010, 6020, etc.) and FAX ports having N+2 (6002, 6012, 6022) is typical of the Audiocodes MP-1xx.

The rest of the call, OK- RE-INVITE, ACK, BYE, OK-BYE, ACK have been removed.

```
12:14:50.582 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209936    )  ---- Incoming SIP Message from
192.168.193.135:5060 to SIPInterface #0 ----
12:14:50.582 : 65.246.127.150 : NOTICE  :

INVITE sip:6046981019@65.246.127.150:5060 SIP/2.0
Via: SIP/2.0/UDP
192.168.193.135:5060;branch=z9hG4bKb0j1d8109020oe0q24i1.1
Allow-Events: message-summary, refer, dialog, line-seize,
presence, call-info, as-feature-event
Max-Forwards: 69
Call-ID: E9A1CE21@10.220.1.1
From: "AUDCM8k" <sip:6046986613@sbc-
msrn.audiocodes.com>;tag=10.220.1.1+1+150601+4cb94dbf
To: <sip:6046981019@sbc-msrn.audiocodes.com>
CSeq: 238787439 INVITE
Expires: 180
Organization: verizon
Supported: 100rel, resource-priority
Content-Length: 196
Content-Type: application/sdp
Contact: "AUDCM8k"
<sip:6046986613+e4an8hs1pfpt8@192.168.193.135:5060;transport=udp>
v=0
o=- 3729920238 3729920238 IN IP4 192.168.193.135
s=-
c=IN IP4 192.168.193.135
t=0 0
m=audio 20052 RTP/AVP 2 0 101
a=rtpmap:101 telephone-event/8000
a=ptime:20
```

```
a=silenceSupp:off - - - -

12:14:50.592 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209938    )  new AcSIPCallAPI created - #3
12:14:50.592 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209939    )  |       | new GetNewSIPCall created
- #8
12:14:50.592 : 65.246.127.150 : NOTICE  :
(  lgr_stk_mngr)(209940    )  Resource StackSession <#3>
Allocated
12:14:50.592 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209941    )  |       |(SIPTU#8)INVITE
State:Idle()
12:14:50.592 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209942    )  SIPCall(#8) changes state from Idle
to Invited
12:14:50.592 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209943    )  |       |       |
#3:SIP_SETUP_EV(E9A1CE21@10.220.1.1)
12:14:50.592 : 65.246.127.150 : NOTICE  :
(      lgr_call)(209944    )  Call Allocated ResourceID: 3
12:14:50.592 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(209945    )  SIPStackSession::HandleStackSetupEV
- NEWCALL: SrcPN=0
12:14:50.592 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(209946    )  <SESSION #3> SendToCall - event:
NEW_CALL_EV  m_Call#3
12:14:50.602 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209947    )  |       |
#3:NEW_CALL_EV:(E9A1CE21@10.220.1.1)
12:14:50.602 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209948    )  |       |       #3:Call changing
states from:IdleState to:NewCallState_IP2Tel
12:14:50.602 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209949    )  ServicesMngr::GetEndPoint PhoneNum
= 6046981019
12:14:50.602 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(209950    )  MotherBoard::GetTrunkGroupId - No
entry found for: DstNum:6046981019 SrcNum:6046986613
SrcIp:d02ec187 go to default
12:14:50.602 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(209951    )  QueryOnHookPortStatus
(ChannelNum=2), status = 0 Polarity = 0
12:14:50.602 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(209952    )  QueryOnHookPortStatus
(ChannelNum=3), status = 0 Polarity = 0
12:14:50.602 : 65.246.127.150 : NOTICE  :
(      lgr_call)(209953    )  Call::SetCoderListForCall #3 Found
2 Common Coders For Call
12:14:50.602 : 65.246.127.150 : NOTICE  :
(      lgr_call)(209954    )  <Call #3> Coder g726r3220 : 20
12:14:50.602 : 65.246.127.150 : NOTICE  :
```

```
(      lgr_call)(209955    )  <Call #3> Coder g72920 : 20
12:14:50.602 : 65.246.127.150 : NOTICE  :
( lgr_profiling)(209956    )  <Call 3> Profiled<Tel=0,Ip=0>:
JBMinDel=10 JBOptF=10 EEarlyM=1 FaxTM=1 IPDS=46 IsFaxU=1 PI2IP=-1
SigIPDF=40 CNGMode=0 DTMFUsed=0 NSEMode=0 PlayRBTone2IP=0
RBUdpPort=0 RTPRD=0 SCE=1 VxxTT=0 Dst2Rdrt=0 DTMFVol=20 ECE=1
ECurDis=0 EDigDel=0 ERevP=0 FHPer=700 InG=32 MWIA=0 MWID=0
VVol=33 ReorderTime=255 DIDWink=0 2StageDial=1 DiscOnBusyT=1
DiscOnBrok=0 DPInd=255 AGC=0 NLP=0
12:14:50.602 : 65.246.127.150 : NOTICE  :
(      lgr_call)(209957    )  |       |
#3GetNextUI:GlobalUI=359392928, mACAddrLsb=850302
12:14:50.602 : 65.246.127.150 : NOTICE  :
(      lgr_call)(209958    )  |       |
#3GetNextUI:GlobalUI=359392929
12:14:50.602 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209959    )  |       #1:NEW_CALL_EV   :
(E9A1CE21@10.220.1.1)
12:14:50.612 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209960    )
EndPoint::MediaResourceList::AllocateMediaIpPortsByMediaRealmID
Perform NEW allocation of Media ports for RealmIndex(0)
port(6000) current allocations are:(1)
12:14:50.612 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209961    )  SIPSDPSession#3 - Changing state
from SIP_MEDIA_IDLE to SIP_MEDIA_OFFERED
12:14:50.612 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209962    )  |       |(SIPTU#8)TRYING_REQ
State:Invited(E9A1CE21@10.220.1.1)
12:14:50.612 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209963    )  New SIPMessage created - #6
12:14:50.612 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209964    )  ---- Outgoing SIP Message to
192.168.193.135:5060 from SIPInterface #0 ----
12:14:50.612 : 65.246.127.150 : NOTICE  :

SIP/2.0 100 Trying
Via: SIP/2.0/UDP
192.168.193.135:5060;branch=z9hG4bKb0j1d8109020oe0q24i1.1
From: "AUDCM8k" <sip:6046986613@sbc-
msrn.audiocodes.com>;tag=10.220.1.1+1+150601+4cb94dbf
To: <sip:6046981019@sbc-msrn.audiocodes.com>;tag=1c817706079
Call-ID: E9A1CE21@10.220.1.1
CSeq: 238787439 INVITE
Supported: em,timer,replaces,path,early-session,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SU
BSCRIBE,UPDATE
Server: Audiocodes-Sip-Gateway-MP-114 FXS_FXO/v.6.00A.042.005
Content-Length: 0

12:14:50.612 : 65.246.127.150 : NOTICE  :
```

```
(     sip_stack)(209966     )  Resource SIPMessage deleted - #6
12:14:50.612 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(209967     )  SIPStackSession::HandleStackSetupEV
- SETUP: SrcPN=0
12:14:50.612 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(209968     )  <SESSION #3> SendToCall - event:
SETUP_EV  m_Call#3
12:14:50.612 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209969     )  |       |       #3:SETUP
(TO:6046981019, FROM:6046986613):(E9A1CE21@10.220.1.1)
12:14:50.622 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209970     )  |       |       #3:Call changing
states from:NewCallState_IP2Tel to:InitiatedState_IP2Tel
12:14:50.622 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209971     )  |       #1:SETUP_EV   :
(E9A1CE21@10.220.1.1)
12:14:50.622 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(209972     )  UpdateChannelParams, Channel 1
12:14:50.622 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(209973     )  #1:ConfigFaxModemChannelParams
NSEMode=0, CNGDetMode=0, FAXTranType=1, VxxTranType=0, VoiceVol=
1, DTMFVol=-11, InGain=0, RTPRedDepth=0, ECE=1, SCE=3,
ECNlpMode=0, DJBufMinDelay=10, DJBufOptFac=10, Result=1)
12:14:50.622 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(209974     )  #1:GenerateRing: ChannelNum=1
RingType:0
12:14:50.622 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(209975     )  Send::CallerID Name=AUDCM8k,
Number=6046986613, Time=0053, Date=0121
12:14:50.622 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209976     )  |       #1:ALERT_EV (send)   :
(E9A1CE21@10.220.1.1)
12:14:50.622 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209977     )  |       |
#3:ALERT_EV:(E9A1CE21@10.220.1.1)
12:14:50.622 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209978     )  |       |       #3:Call changing
states from:InitiatedState_IP2Tel to:AlertingState_IP2Tel
12:14:50.622 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209979     )  |       |       |
#3:ALERT_EV(E9A1CE21@10.220.1.1)
12:14:50.622 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209980     )  New SIPMessage created - #9
12:14:50.622 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209981     )  |       |(SIPTU#8)ALERT_REQ
State:Invited(E9A1CE21@10.220.1.1)
12:14:50.632 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209982     )  ---- Outgoing SIP Message to
192.168.193.135:5060 from SIPInterface #0 ----
12:14:50.632 : 65.246.127.150 : NOTICE  :
```

SIP/2.0 180 Ringing

```
Via: SIP/2.0/UDP
192.168.193.135:5060;branch=z9hG4bKb0j1d8109020oe0q24i1.1
From: "AUDCM8k" <sip:6046986613@sbc-
msrn.audiocodes.com>;tag=10.220.1.1+1+150601+4cb94dbf
To: <sip:6046981019@sbc-msrn.audiocodes.com>;tag=1c817706079
Call-ID: E9A1CE21@10.220.1.1
CSeq: 238787439 INVITE
Contact: <sip:6046981019@65.246.127.150:5060>

Supported: em,timer,replaces,path,early-session,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SU
BSCRIBE,UPDATE
Require: 100rel
RSeq: 1
Server: Audiocodes-Sip-Gateway-MP-114 FXS_FXO/v.6.00A.042.005
Content-Length: 0
12:14:50.632 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209984    )
UdpRtxMngr::Transmit 180 Response 238787439 INVITE Rtx Left: 6
Dest: 192.168.193.135:5060 CallID: (E9A1CE21@10.220.1.1)
12:14:50.632 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209985    )  Resource SIPMessage deleted - #9
12:14:50.632 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209986    )  Resource SIPMessage deleted - #8
12:14:50.763 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209987    )  New SIPMessage created - #12
12:14:50.763 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209988    )  ---- Incoming SIP Message from
192.168.193.135:5060 to SIPInterface #0 ----
12:14:50.763 : 65.246.127.150 : NOTICE  :

PRACK sip:6046981019@65.246.127.150:5060 SIP/2.0
Via: SIP/2.0/UDP
192.168.193.135:5060;branch=z9hG4bKd8041q10c86gqe8tq240.1
Allow-Events: message-summary, refer, dialog, line-seize,
presence, call-info, as-feature-event
Max-Forwards: 69
Call-ID: E9A1CE21@10.220.1.1
From: "AUDCM8k" <sip:6046986613@sbc-
msrn.audiocodes.com>;tag=10.220.1.1+1+150601+4cb94dbf
To: <sip:6046981019@sbc-msrn.audiocodes.com>;tag=1c817706079
CSeq: 238787440 PRACK
RAck: 1 238787439 INVITE
Organization: verizon
Supported: 100rel, resource-priority
Content-Length: 0

12:14:50.763 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209990    )  |        |(SIPTU#8)PRACK
State:Invited(E9A1CE21@10.220.1.1)
12:14:50.763 : 65.246.127.150 : NOTICE  :
```

```
(     sip_stack)(209991     )  New SIPMessage created - #11
12:14:50.763 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209992     )  UdpRtxMngr::Remove 180 Response
238787439 INVITE
12:14:50.763 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209993     )  Resource SIPMessage deleted - #11
12:14:50.763 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209994     )  |       |       |
#3:SIP_PRACK_EV(E9A1CE21@10.220.1.1)
12:14:50.763 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209995     )  New SIPMessage created - #0
12:14:50.773 : 65.246.127.150 : NOTICE  :
(     sip_stack)(209996     )  New SIPMessage created - #10
12:14:50.773 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209997     )  |       |(SIPTU#8)PRACK_RESPONSE
State:Invited(E9A1CE21@10.220.1.1)
12:14:50.773 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(209998     )  ---- Outgoing SIP Message to
192.168.193.135:5060 from SIPInterface #0 ----
12:14:50.773 : 65.246.127.150 : NOTICE  :

SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.193.135:5060;branch=z9hG4bKd8041q10c86gqe8tq240.1
From: "AUDCM8k" <sip:6046986613@sbc-
msrn.audiocodes.com>;tag=10.220.1.1+1+150601+4cb94dbf
To: <sip:6046981019@sbc-msrn.audiocodes.com>;tag=1c817706079
Call-ID: E9A1CE21@10.220.1.1
CSeq: 238787440 PRACK
Contact: <sip:6046981019@65.246.127.150:5060>
Supported: em,timer,replaces,path,early-session,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SU
BSCRIBE,UPDATE
Server: Audiocodes-Sip-Gateway-MP-114 FXS_FXO/v.6.00A.042.005
Content-Length: 0

12:14:50.773 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210000     )  Resource SIPMessage deleted - #10
12:14:50.773 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210001     )  Resource SIPMessage deleted - #0
12:14:50.773 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210002     )  Resource SIPMessage deleted - #12
12:14:51.604 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdex)(210003     )  recv <-- OFF_HOOK Ch:1
12:14:51.614 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210004     )  #1:OFF_HOOK_EV
12:14:51.614 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210005     )  |       #1:OFF_HOOK_EV
12:14:51.614 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210006     )  |       #1:OPEN_VOICE-NOT_READY
(MS:0, HS:0, Ready:1)
```

```
12:14:51.614 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210007    )
#1:PSOSBoardInterface::StopPlayTone- Called
12:14:51.614 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210008    )  |       #1:CONNECT_EV (send)  :
(E9A1CE21@10.220.1.1)
12:14:51.614 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210009    )  |       |
#3:CONNECT_EV:(E9A1CE21@10.220.1.1)
12:14:51.614 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210010    )  |       |       #3:Call changing
states from:AlertingState_IP2Tel to:ConnectedState
12:14:51.614 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210011    )  |       |       |
#3:CONNECT_EV(E9A1CE21@10.220.1.1)
12:14:51.614 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210012    )  New SIPMessage created - #2
12:14:51.614 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210013    )  SIPSDPSession#3 - Changing state
from SIP_MEDIA_OFFERED to SIP_MEDIA_COMPLETED
12:14:51.614 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210014    )  |       |(SIPTU#8)CONNECT_REQ
State:Invited(E9A1CE21@10.220.1.1)
12:14:51.624 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210015    )  ---- Outgoing SIP Message to
192.168.193.135:5060 from SIPInterface #0 ----
12:14:51.624 : 65.246.127.150 : NOTICE  :

SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.193.135:5060;branch=z9hG4bKb0j1d8109020oe0q24i1.1
From: "AUDCM8k" <sip:6046986613@sbc-
msrn.audiocodes.com>;tag=10.220.1.1+1+150601+4cb94dbf
To: <sip:6046981019@sbc-msrn.audiocodes.com>;tag=1c817706079
Call-ID: E9A1CE21@10.220.1.1
CSeq: 238787439 INVITE
Contact: <sip:6046981019@65.246.127.150:5060>
Supported: em,timer,replaces,path,early-session,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SU
BSCRIBE,UPDATE
Server: Audiocodes-Sip-Gateway-MP-114 FXS_FXO/v.6.00A.042.005
Content-Type: application/sdp
Content-Length: 263
v=0
o=AudiocodesGW 817752412 817752279 IN IP4 65.246.127.150
s=Phone-Call
c=IN IP4 65.246.127.150
t=0 0
m=audio 6000 RTP/AVP 2 101
c=IN IP4 65.246.127.150
a=rtpmap:2 G726-32/8000
```

```
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv

12:14:51.624 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210017    )
UdpRtxMngr::Transmit 200 Response 238787439 INVITE Rtx Left: 6
Dest: 192.168.193.135:5060 CallID: (E9A1CE21@10.220.1.1)
12:14:51.624 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210018    )  SIPCall(#8) changes state from
Invited to LocalAccepted
12:14:51.624 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(210019    )  DtmfCapNegotiationAlgorithm ::
TxDtmfMethod = DTMF_RFC2833_SUPPORTED
12:14:51.624 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(210020    )  DtmfCapNegotiationAlgorithm ::
TxRtpRfc2833Payload = 101
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(210021    )  <SESSION #3> SendToCall - event:
DTMF_CONTROL_EV  m_Call#3
12:14:51.634 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210022    )  |       |
#3:DTMF_CONTROL_EV:(E9A1CE21@10.220.1.1)
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(210023    )  <SESSION #3> SendToCall - event:
OPEN_LOGICAL_CHANNEL_ACK_EV  m_Call#3
12:14:51.634 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210024    )  |       |
#3:OPEN_LOGICAL_CHANNEL_ACK_EV:(E9A1CE21@10.220.1.1)
12:14:51.634 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210025    )  |
#1:OPEN_LOGICAL_CHANNEL_ACK_EV    : (E9A1CE21@10.220.1.1)
12:14:51.634 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210026    )  |       #1:OPEN_VOICE
(IP:192.168.193.135, RTP:20052, RTCP:0, VoiceCoder:g726r3220,
VbdCoder: InvalidCoder255, Dtmf:gwRFC2833RalayDTMF,Rx
payload:101,Tx payload:101 ,RTPmode:1, FaxTransportType: 1,
AVoIPMediaType: gwMediaTypeAudioOnly, T38Version:N/A)
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210027    )  activate channel local rtp
port:6000, port=20052, BChannel:1, local ip:65.246.127.150,
ip=192.168.193.135 (Voice:1,Vbd:0,T38:0,Video:0)
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210028    )  #1:ActivateChannel: Socks=11 CID=1
Trunk:-1 BChannel:1 RemoteIP=192.168.193.135 RemotePort=20052
RemoteT38IP= RemoteT38Port=0 RemoteRTCPIP= RemoteRTCPPort=0
FaxModemDet=NO_FAX_MODEM_DETECTED
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210029    )  Open channel: IsVoiceOn: 1,
IsT38On: 0, IsVbdOn: 0, IsVideoOn: 0
12:14:51.634 : 65.246.127.150 : NOTICE  :
```

```
(   lgr_psbrdif)(210030    )  #1:OpenChannel:on Trunk -1
BChannel:1 CID=1 with VoiceCoder: g726r3220 VbdCoder:
InvalidCoder255 DetectorSide: 0 FaxModemDet NO_FAX_MODEM_DETECTED
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210031    )  #1:OpenChannel VoiceVolume= 1,
DTMFVolume = -11, InputGain = 0, RTPRedundancyDepth = 0
FlashHookPeriod = 700 AgcCmd = 0x13180000
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210032    )  RFC2833RTPPayloadType: Rx=101
Tx=101
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210033    )  OpenChannel, CoderType = 4,
Interval = 3, M = 1
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210034    )  #1:FAXTransportType = 1
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210035    )  #1:ConfigFaxModemChannelParams
NSEMode=0, CNGDetMode=0, FAXTranType=1, VxxTranType=0, VoiceVol=
1, DTMFVol=-11, InGain=0, RTPRedDepth=0, ECE=1, SCE=0,
ECNlpMode=0, DJBufMinDelay=10, DJBufOptFac=10, Result=1)
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210036    )  Detectors: Amd:On=0,Diretion=0,
Ans:On=0,Direction=0 En:On=0,Direction=0 Board IBScmd:0xa1
12:14:51.634 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210037    )  #1:Channel will be open WITH DSP
12:14:51.644 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdex)(210038    )  PCIIFChangeChannelParams failed
RTP_Interval Coder
12:14:51.644 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210039    )  Setting
ActivateRTP_RTCPCmd.Cmd.IpTosFieldInUdpPacket to 184
12:14:51.644 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210040    )  #1:ActivateChannel:RtpPayload: 2
12:14:51.644 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210041    )  Resource SIPMessage deleted - #2
12:14:51.784 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210042    )  New SIPMessage created - #20
12:14:51.784 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210043    )  ---- Incoming SIP Message from
192.168.193.135:5060 to SIPInterface #0 ----
12:14:51.784 : 65.246.127.150 : NOTICE  :

ACK sip:6046981019@65.246.127.150:5060 SIP/2.0
Via: SIP/2.0/UDP
192.168.193.135:5060;branch=z9hG4bKdo67j320c0j0nesp43g1.1
Allow-Events: message-summary, refer, dialog, line-seize,
presence, call-info, as-feature-event
Max-Forwards: 69
Call-ID: E9A1CE21@10.220.1.1
From: "AUDCM8k" <sip:6046986613@sbc-
msrn.audiocodes.com>;tag=10.220.1.1+1+150601+4cb94dbf
To: <sip:6046981019@sbc-msrn.audiocodes.com>;tag=1c817706079
```

```
CSeq: 238787439 ACK
Contact: "AUDCM8k"
<sip:6046986613+e4an8hs1pfpt8@192.168.193.135:5060;transport=udp>
Organization: verizon
Content-Length: 0

12:14:51.784 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210045    )  UdpRtxMngr::Remove 200 Response
238787439 INVITE
12:14:51.784 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210046    )  |       |(SIPTU#8)ACK
State:LocalAccepted(E9A1CE21@10.220.1.1)
12:14:51.784 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210047    )  SIPCall(#8) changes state from
LocalAccepted to Connected
12:14:51.794 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210048    )  |       |       |
#3:SIP_ACK_EV(E9A1CE21@10.220.1.1)
12:14:51.794 : 65.246.127.150 : NOTICE  :
(   lgr_stk_ses)(210049    )  <SESSION #3> SendToCall - event:
CONNECT_ACK_EV  m_Call#3
12:14:51.794 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210050    )  |       |
#3:CONNECT_ACK_EV:(E9A1CE21@10.220.1.1)
12:14:51.794 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210051    )  |       #1:CONNECT_ACK_EV   :
(E9A1CE21@10.220.1.1)
12:14:51.794 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210052    )  Resource SIPMessage deleted - #20
12:14:54.148 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdex)(210053    )  recv <-- DETECT_FAX_ANSWER_TONE
Ch:1, AnswerToneDetectionOrigin=0, AnswerToneDetectionDirection=0
12:14:54.158 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210054    )  #1:DETECT_ANSWER_TONE_EV
12:14:54.158 : 65.246.127.150 : NOTICE  :
(      lgr_flow)(210055    )  |       #1:DETECT_ANSWER_TONE_EV
12:14:54.158 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210056    )
#1:PSOSBoardInterface::ConfigureFaxModemChannelParams G711 Coder
Required
12:14:54.158 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210057    )  UpdateChannelParams, Channel 1
12:14:54.168 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210058    )  #1:ConfigFaxModemChannelParams
NSEMode=0, CNGDetMode=0, FAXTranType=1, VxxTranType=0, VoiceVol=
1, DTMFVol=-11, InGain=0, RTPRedDepth=0, ECE=1, SCE=0,
ECNlpMode=0, DJBufMinDelay=10, DJBufOptFac=10, Result=0)
12:14:54.168 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdif)(210059    )  No need to update channel (CID=1,
Error=0)
12:14:54.168 : 65.246.127.150 : NOTICE  :
```

```
(       lgr_flow)(210060     )  |        #1:DETECT_MODEM : Not
Handled
12:14:56.391 : 65.246.127.150 : NOTICE  :
(   lgr_psbrdex)(210061     )  recv <-- DETECT_FAX Ch:1,
Calling=0, Relay=1, Origin=1
12:14:56.411 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210062     )  #1:FAX_EV
12:14:56.411 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210063     )  |        #1:FAX_EV
12:14:56.411 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210064     )  |
#1:ANSWERING_SIDE_DETECT_FAX. Detected : PREAMBLE (V.21)
12:14:56.411 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210065     )  |        #1:FAX_DETECTED_EV (send)
: (E9A1CE21@10.220.1.1)
12:14:56.411 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210066     )  |        |
#3:FAX_DETECTED_EV:(E9A1CE21@10.220.1.1)
12:14:56.411 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210067     )  |        |        |
#3:FAX_DETECTED_EV(E9A1CE21@10.220.1.1)
12:14:56.411 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210068     )  FaxState Changed from
FAX_MODEM_STATE_IDLE  to  FAX_MODEM_STATE_T38_OFFERING
12:14:56.411 : 65.246.127.150 : NOTICE  :
(      sip_stack)(210069     )  New SIPMessage created - #15
12:14:56.411 : 65.246.127.150 : NOTICE  :
(      sip_stack)(210070     )  SIPSDPSession#3 - Changing state
from SIP_MEDIA_COMPLETED to SIP_MEDIA_OFFERING
12:14:56.411 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210071     )  |        |(SIPTU#8)REINVITE_REQ
State:Connected(E9A1CE21@10.220.1.1)
12:14:56.421 : 65.246.127.150 : NOTICE  :
(       lgr_flow)(210072     )  ---- Outgoing SIP Message to
192.168.193.135:5060 from SIPInterface #0 ----
12:14:56.421 : 65.246.127.150 : NOTICE  :

INVITE sip:6046986613+e4an8hs1pfpt8@192.168.193.135:5060 SIP/2.0
Via: SIP/2.0/UDP 65.246.127.150;branch=z9hG4bKac832267398
Max-Forwards: 70
From: <sip:6046981019@sbc-msrn.audiocodes.com>;tag=1c817706079
To: "AUDCM8k" <sip:6046986613@sbc-
msrn.audiocodes.com>;tag=10.220.1.1+1+150601+4cb94dbf
Call-ID: E9A1CE21@10.220.1.1
CSeq: 1 INVITE
Contact: <sip:6046981019@65.246.127.150:5060>
Supported: em,timer,replaces,path,early-session,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SU
BSCRIBE,UPDATE
User-Agent: Audiocodes-Sip-Gateway-MP-114 FXS_FXO/v.6.00A.042.005
Content-Type: application/sdp
```

```
Content-Length: 320
v=0
o=AudiocodesGW 817752412 817752280 IN IP4 65.246.127.150
s=Phone-Call
c=IN IP4 65.246.127.150
t=0 0
m=image 6002 udptl t38
c=IN IP4 65.246.127.150
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxMaxBuffer:1024
a=T38FaxMaxDatagram:238
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPRedundancy

12:14:56.431 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210074    )
UdpRtxMngr::Transmit 1 INVITE Rtx Left: 6 Dest:
192.168.193.135:5060 CallID: (E9A1CE21@10.220.1.1)
12:14:56.431 : 65.246.127.150 : NOTICE  :
(     sip_stack)(210075    )  Resource SIPMessage deleted - #15
```