Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the AudioCodes Mediant 600 with Avaya SIP Enablement Services and Avaya Communication Manager – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring the AudioCodes Mediant-600 with Avaya SIP Enablement Services and Avaya Communication Manager.

The AudioCodes Mediant 600 VoIP SIP Media Gateway serves as a gateway between analog endpoint / ISDN trunks at a branch location and a VoIP infrastructure at a main location using the Session Initiation Protocol (SIP). The Mediant-600 has 4 FXS (analog endpoint) ports and 4 BRI (ISDN trunk) ports. The BRI Ports were configured only as a failover path to the PSTN if the data WAN is unavailable and SIP calls can not be made. The ability of these BRI ports to provide local PSTN access for the branch as part of normal operation is not covered by these Application Notes.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describes the steps to configure the AudioCodes Mediant-600 with Avaya SIP Enablement Services and Avaya Communication Manager.

The Mediant-600 has 4 FXS (analog endpoint) ports and 4 BRI (ISDN trunk) ports.

Figure 1 illustrates the configuration used in these Application Notes. In the sample configuration, two sites are connected via an IP network. The main site has an Avaya SES and an Avaya S8300 Server running CM in an Avaya G350 Media Gateway. Endpoints include an Avaya 4621 IP phone (with SIP firmware) a 4620 IP phone (with H.323 firmware) and a 2410 phone.
An ISDN-PRI trunk connects the media gateway to the PSTN.

The branch site has an AudioCodes Mediant 600 with two analog endpoints and a fax machine. The branch site also has two Avaya 96xx IP Phones with SIP firmware. The Mediant-600 connects the branch site to the PSTN via one BRI (ISDN trunk) port. The other three BRI ports were not connected. At the branch site, the Mediant-600 registers all analog endpoints as SIP endpoints to the Avaya SES while all SIP telephones register directly to the Avaya SES. In the Avaya Communication Manager both types of SIP endpoints are administered as OPS stations. The PSTN trunk is registered with the SES, but does not have an OPS station administered on CM. The SIP phones and the Mediant-600 use a router in the IP network as the default gateway. In survivable mode, if the data WAN is unavailable, all SIP telephones are registered to the Mediant-600 instead.

1.1 Interoperability Compliance Testing

The following call flows were covered during interoperability compliance test:

1.1.1 Normal Mode:

1.1.1.1 96xx/FXS at branch to/from H.323 stations at CM.
96xx/FXS ←→ SES ←→ CM ←→ H.323 station

1.1.1.2 96xx/FXS at branch to/from Digital/Analog station.
96xx/FXS ←→ SES ←→ CM ←→ Avaya Media Gateway ←→ D/A station

1.1.1.3 96xx/FXS at branch to/from PSTN endpoint.
96xx/FXS ←→ SES ←→ CM ←→ Avaya Media Gateway ←→ PSTN endpoint

1.1.1.4 96xx/FXS at branch to/from 96xx/FXS at same branch.
96xx/FXS ←→ SES ←→ CM ←→ SES ←→ 96xx/FXS

1.1.2 Survivable Mode:

1.1.2.1 96xx/FXS at branch to PSTN endpoint.
96xx/FXS → AC SAS – BRI → PSTN endpoint

1.1.2.2 96xx/FXS at branch to/from 96xx/FXS at same branch.
96xx/FXS ←→ AC SAS ←→ 96xx/FXS

1.2 Support

For technical support on the Mediant-600, contact AudioCodes via the support link at www.audiocodes.com.

2. Reference Configuration

2.1. Description of the test scenario

This configuration was utilized for compliance testing.

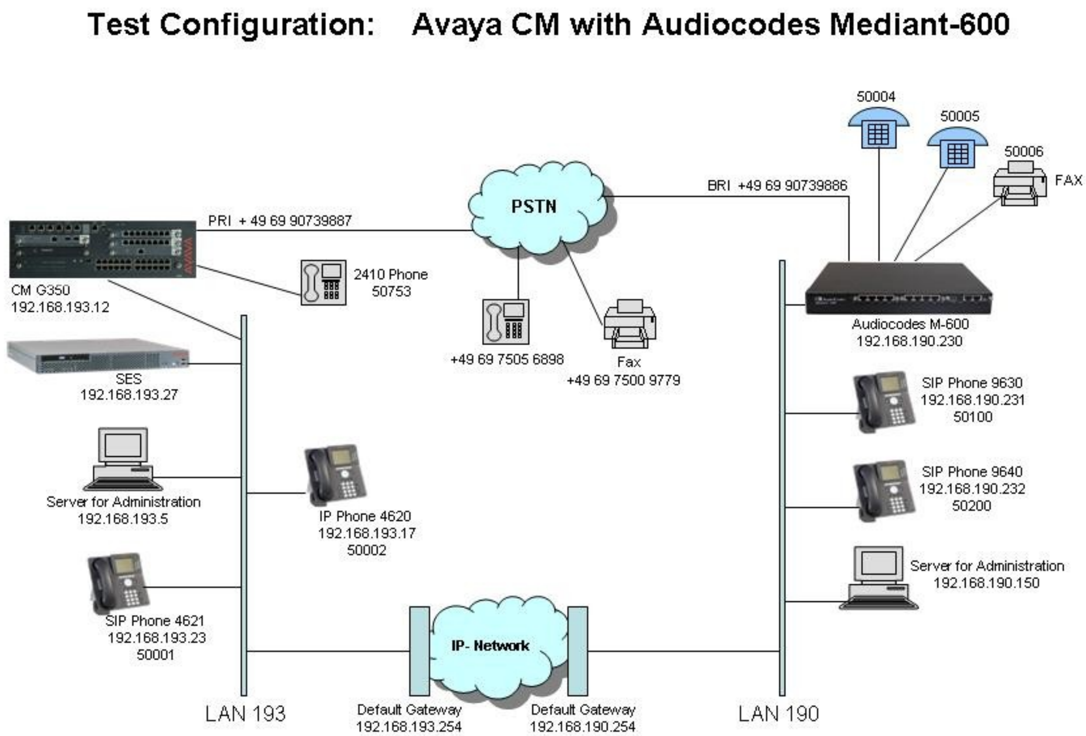


Figure 1 AudioCodes Mediant-600 Test Configuration

3. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

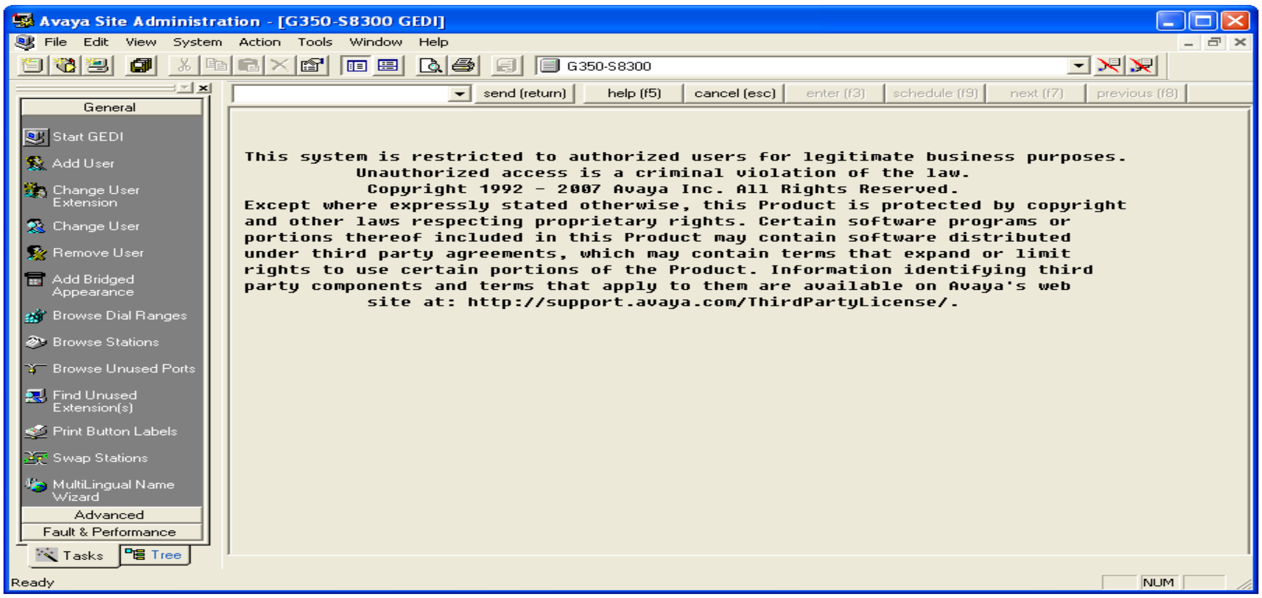
Avaya Components	
Avaya G350 Media Gateway	28.18.0 /1
Avaya S8300 Server	R015x.00.0.825.4
Avaya SES	5.1.2.0-416.4b
Avaya 4621 IP phone with SIP firmware	2.2.2
Avaya 4620 IP phone with H.323 firmware	2.887
Avaya 2410 phone	n/a
Avaya 9630 IP phone with SIP firmware	2.4.8.20
Avaya 9640 IP phone with SIP firmware	2.4.8.20
Avaya T3-Standard Analog Telephone	n/a
AudioCodes Components	
Mediant-600	5.40A.033

Table 1: Equipment and Software Tested

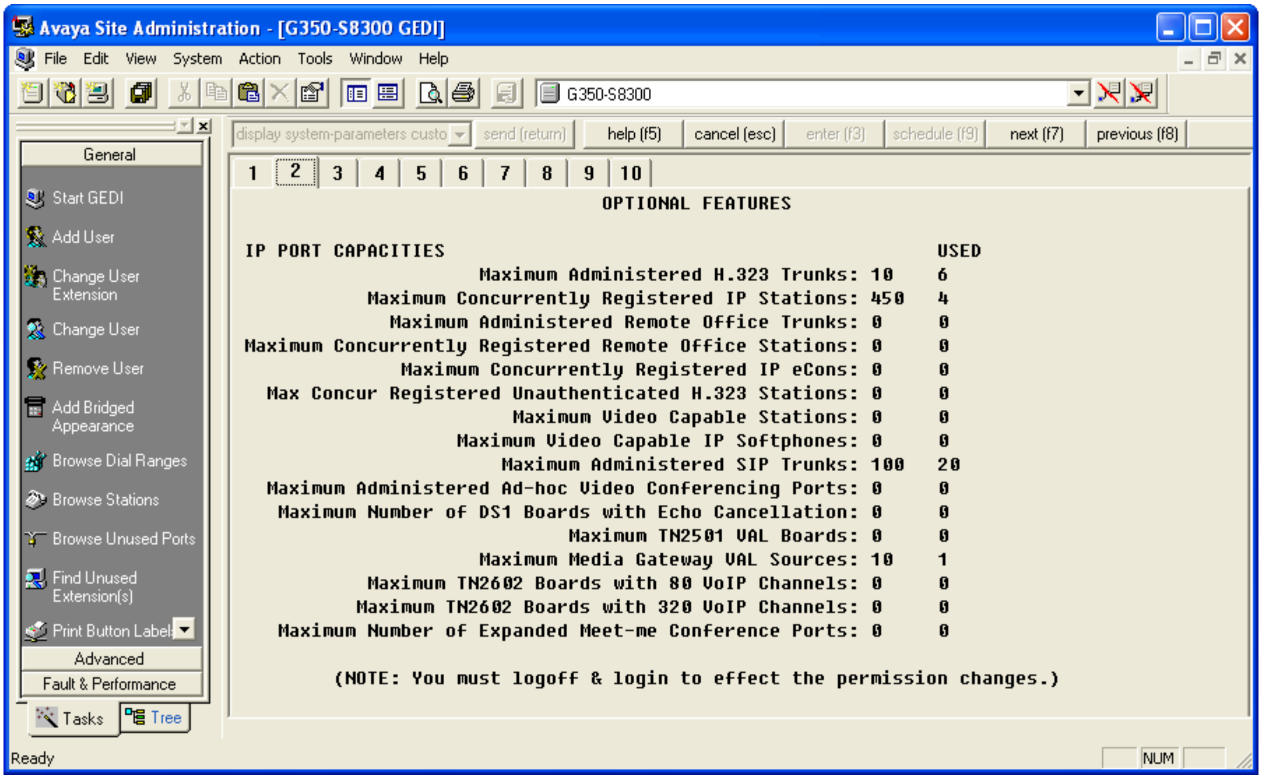
4. Configure Communication Manager

This section describes the steps for configuring the Communication Manager. Communication Manager is configured via the Avaya Site Administration (ASA).

4.1 Login to your PC and select Start → Programs → Avaya → Site Administration to launch the ASA application. Enter “Start GEDI” at the left pane and log into the Communication Manager using the appropriate credentials.

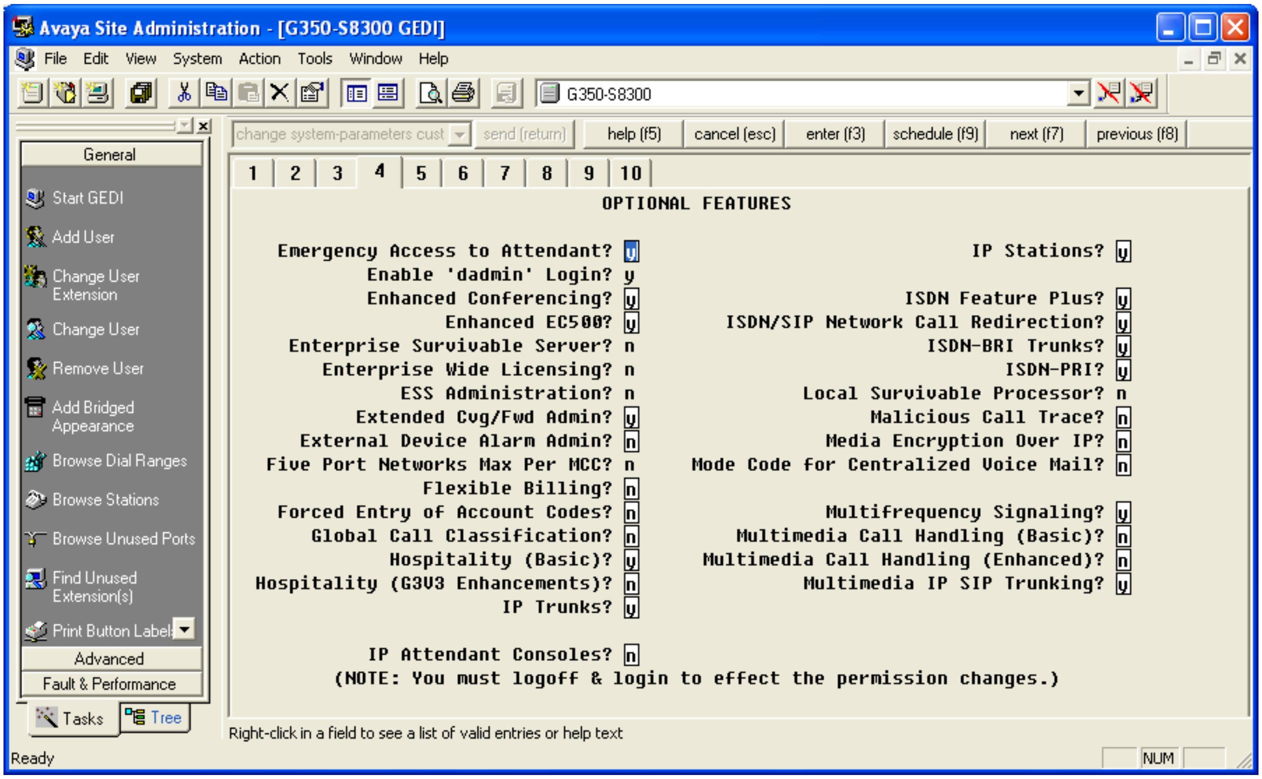


4.2 Enter the “display system-parameters customer-options” command at the top of the right pane to verify that sufficient SIP trunk capacity exists. Verify on page 2, that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

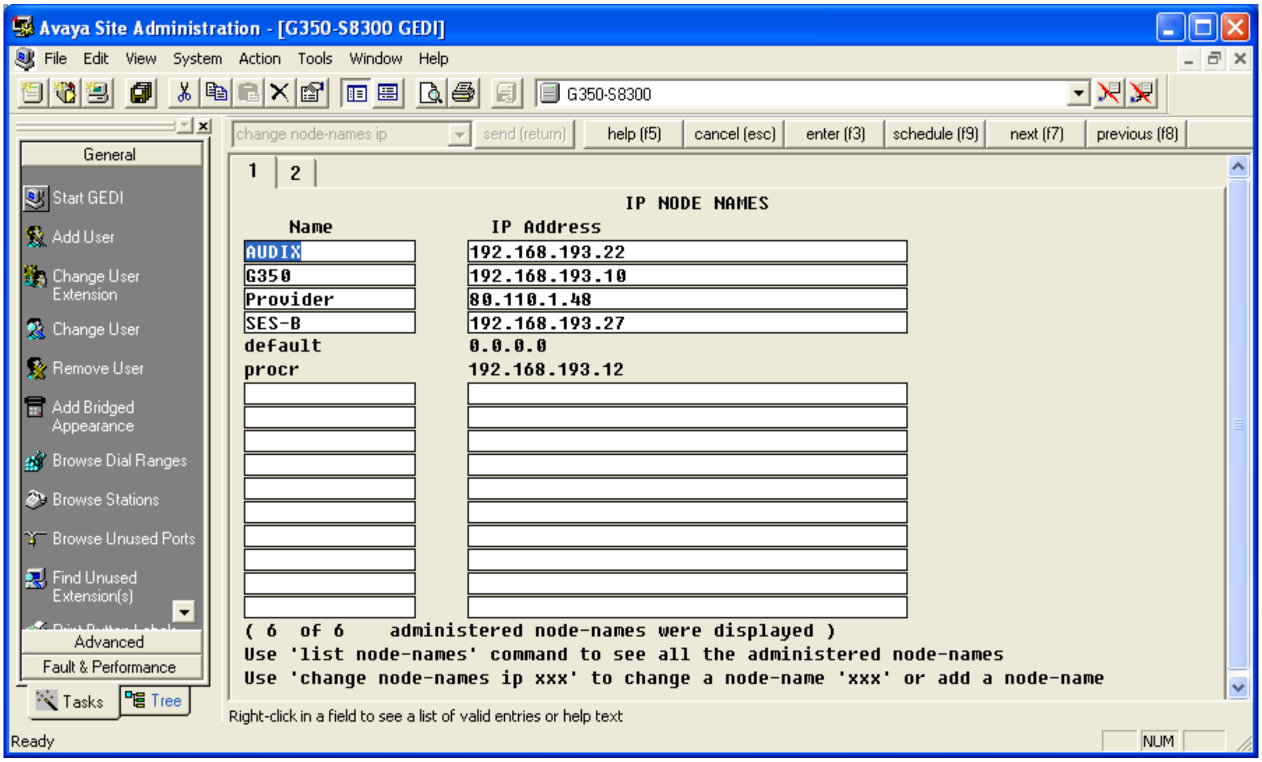


4.3 Use the “change system-parameters customer-options” command to verify that the following fields on page 4 have been set to y:

- ISDN-PRI: y
- IP trunks: y

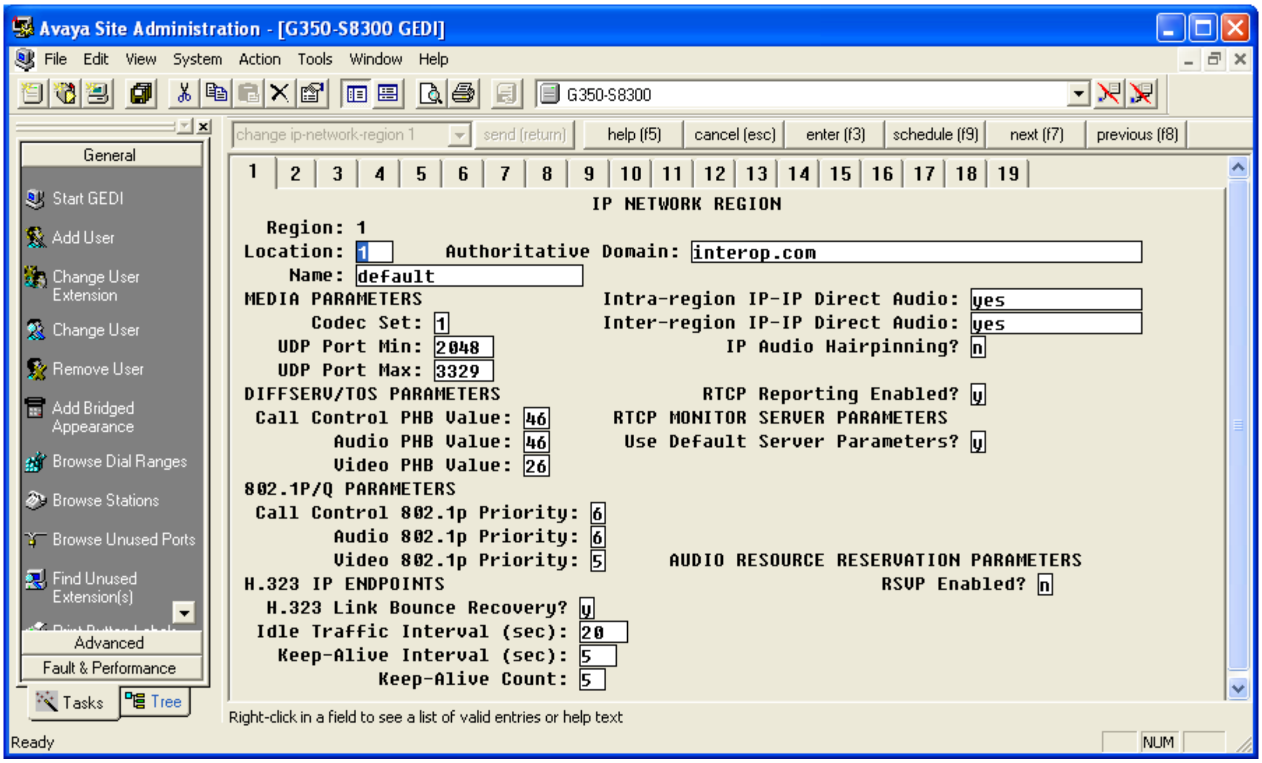


4.4 Use the “change node-names ip” command to assign the node name and IP address for the Avaya SES. In this case SES-B is used with a 192.168.193.27 IP address.



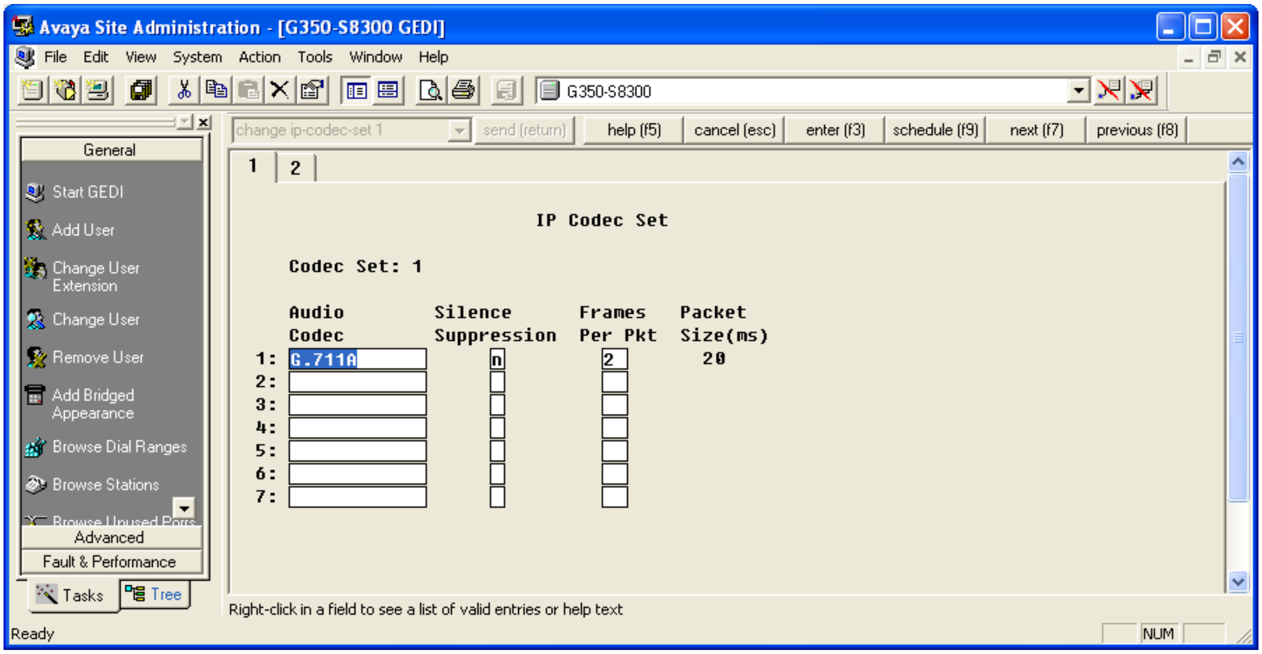
4.5 Enter the "change ip-network-region x" command, where x is the number of the region. Change the following fields on the IP Network Region form:

- The Location field is set to 1.
- The Authoritative Domain is set to interop.com. This name will appear in the From header of SIP messages.
- IP-IP Direct Audio (shuffling) is set to yes.
- The Codec Set is set to the number of the IP codec set to be used for calls.

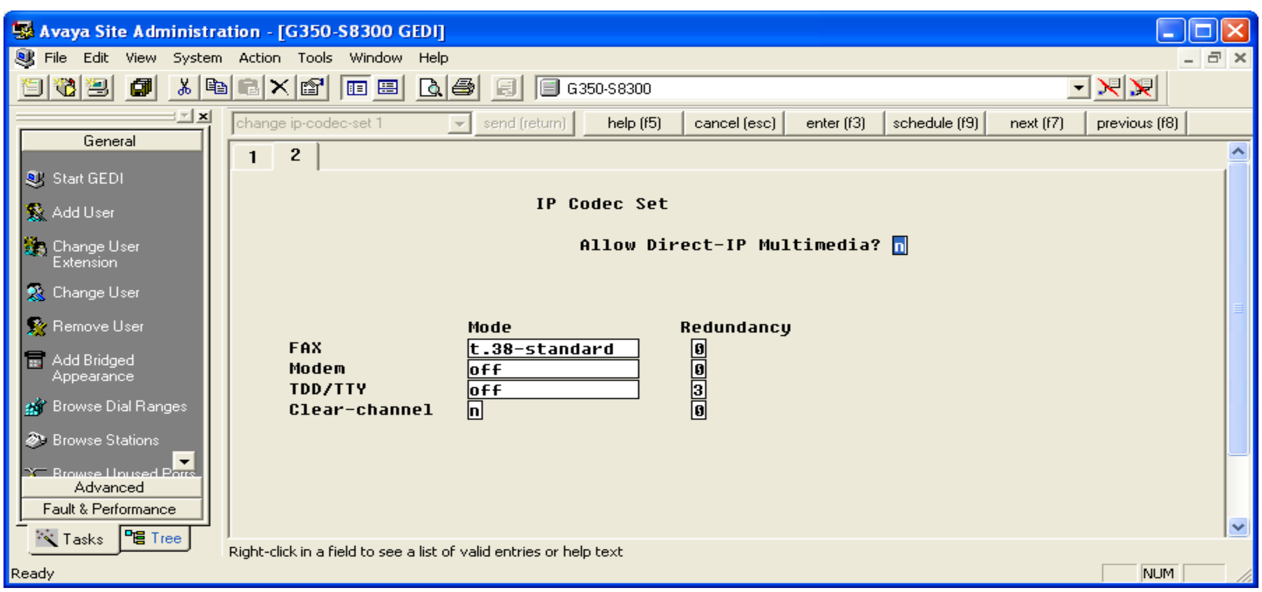


4.6 Enter the “change ip-codec-set x” command, where x is the codec set value specified in the step before.

The example below shows the values used in the test.

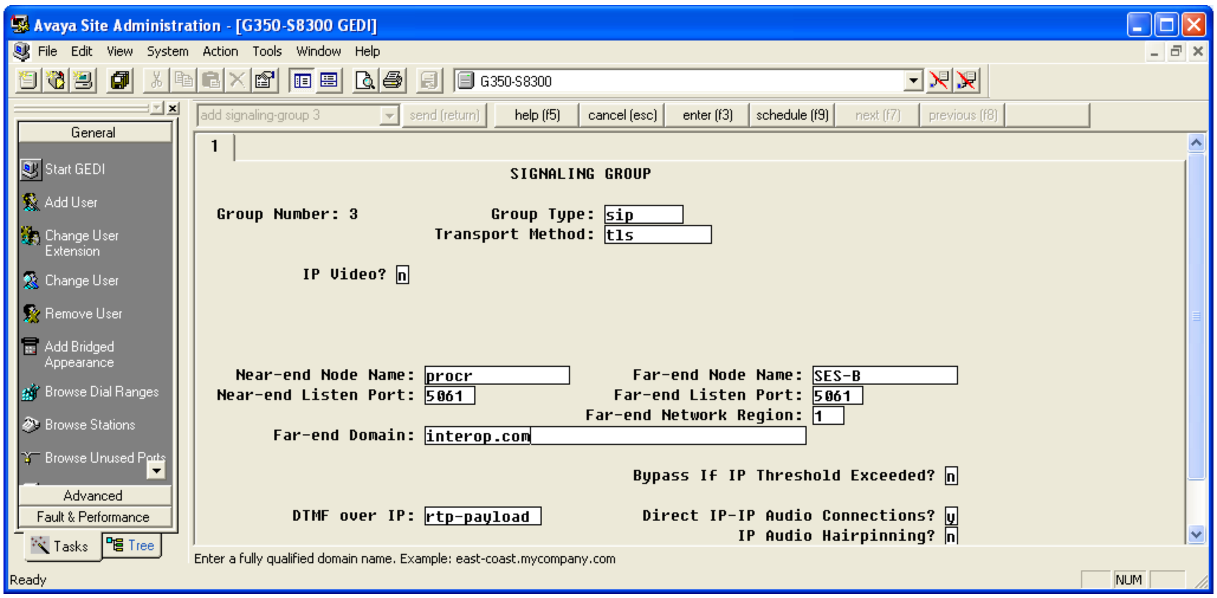


On page 2, the FAX Mode field have to be set to t.38-standard to support fax machines. The screen below shows the settings used for the fax testing.



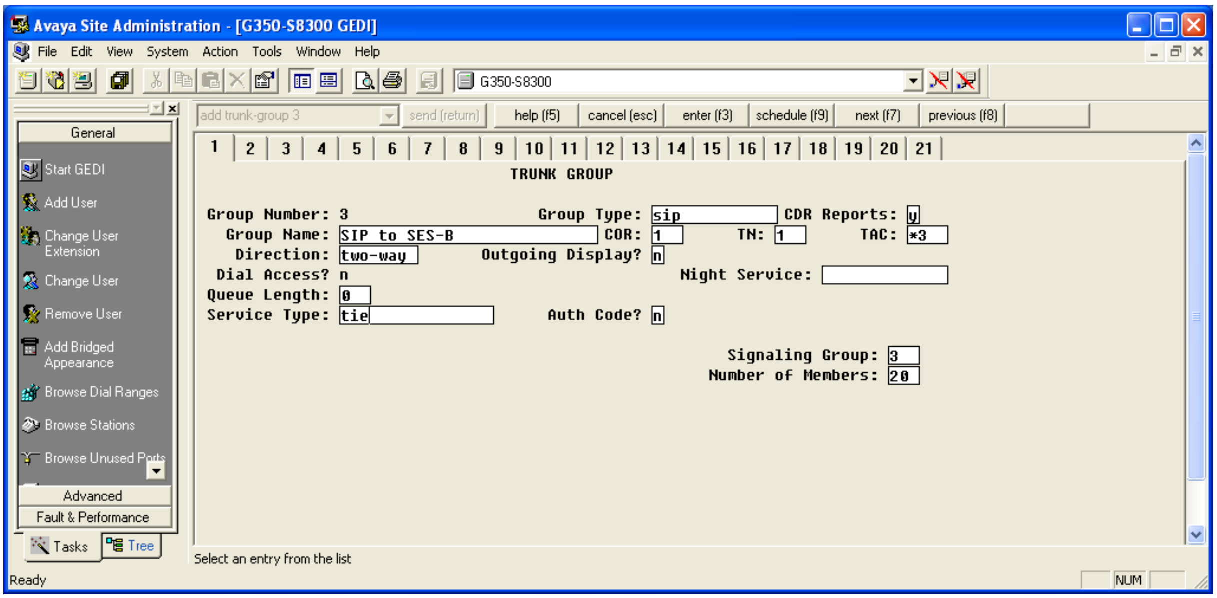
4.7 Create a SIP signalling group on your Communication Manager by entering the "add signalling-group x" command, where x is the number of the signalling group. The far-end will connect to your SES Home.

- Set the group type field to SIP.
- Set the transport method to tls.
- Set Near-end Node name to procr and the port to 5061.
- Set Far-end Node name to SES-B and the Port to 5061. This name was specified in step 3.4.
- Set Far-end Network region to 1. The region was specified in step 3.5.
- Set Far-end Domain to interop.com.
- Verify that Direct IP-IP Audio Connections is set to y.



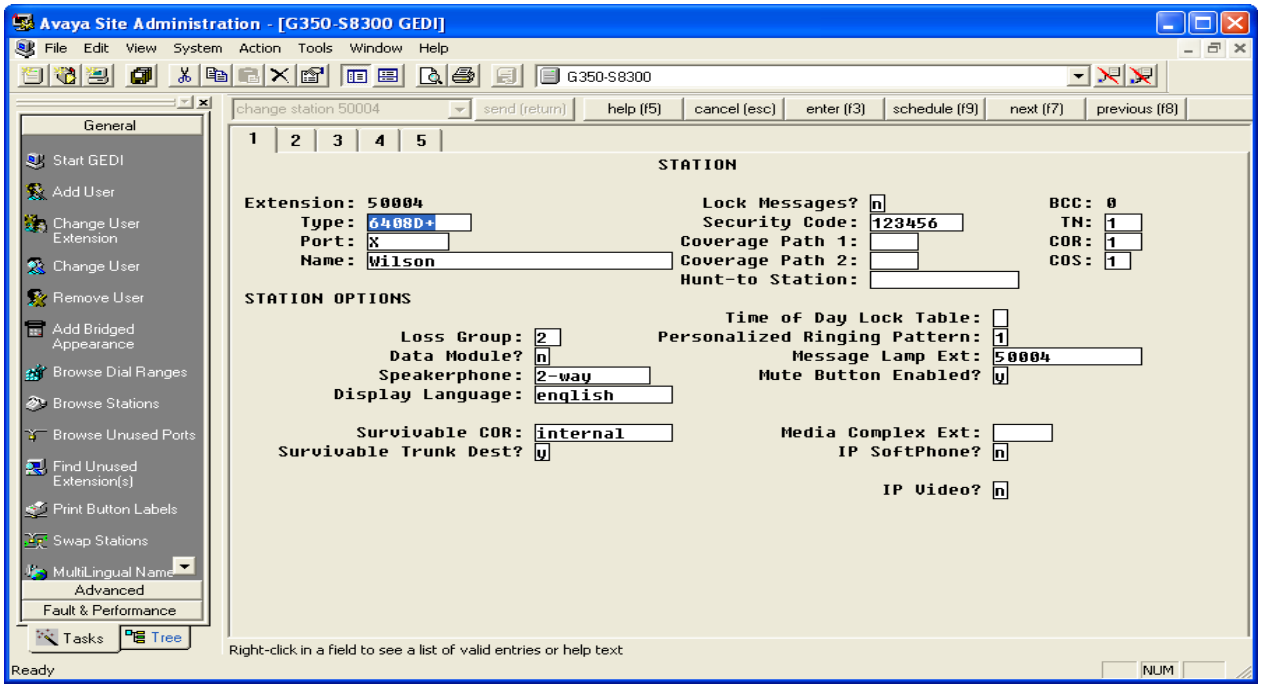
4.8 Add a SIP trunk for the signalling group added at the previous step. Enter command “add trunk-group x”, where x is the number of the group.

- Set the group type field to SIP.
- Specify a Group Name.
- Set the Service Type field to tie.
- Set the TAC field to a number (in this case *3). The format of the number was defined in the dial-plan analysis form.
- Set the Signalling Group to the value specified in the step before.



4.9 Add a 6408D+ station for each FXS Station on the M-600. Enter command "add station xxx", where xxx describes the number of the station.

- Set the Type for the station to 6408D+.
- Set the Port to X.
- Define a name for the station.
- Set the Security Code to a number you like.

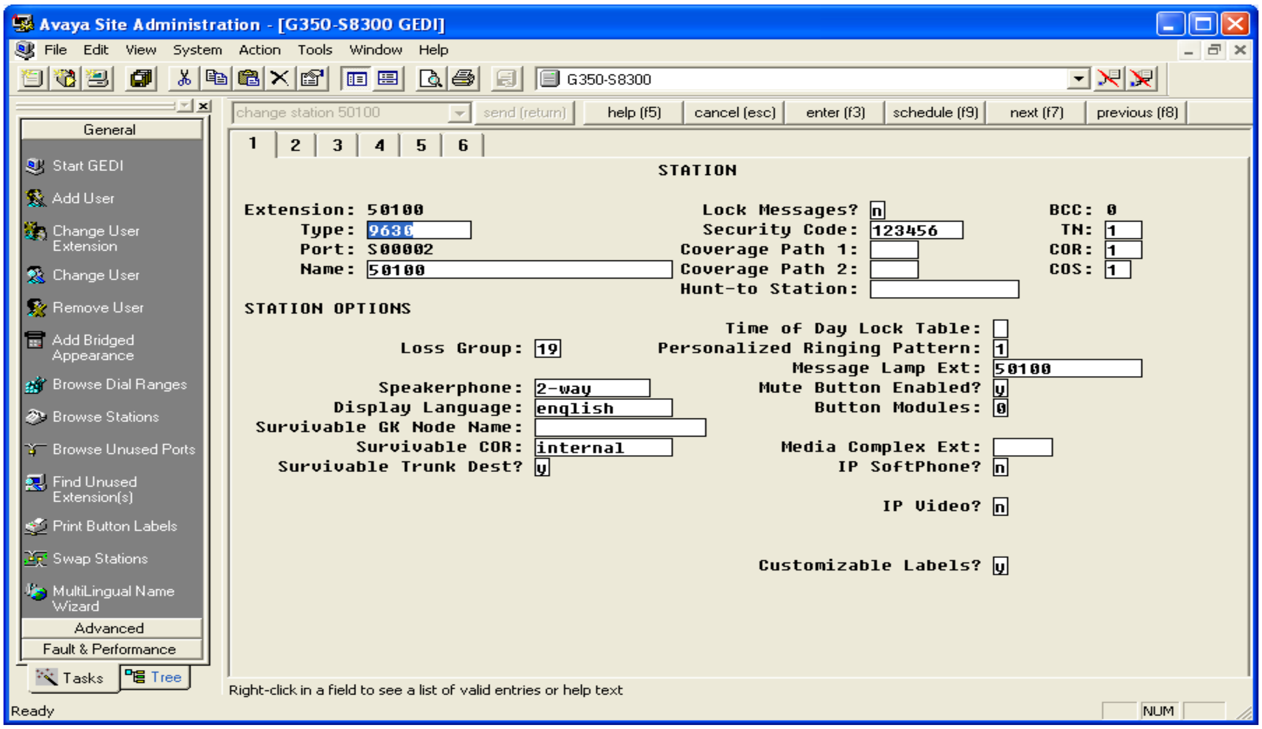


Repeat this for all the FXS stations and an additional station for the Mediant-600 Gateway.

4.10 Add any 96xx SIP stations that will be used.

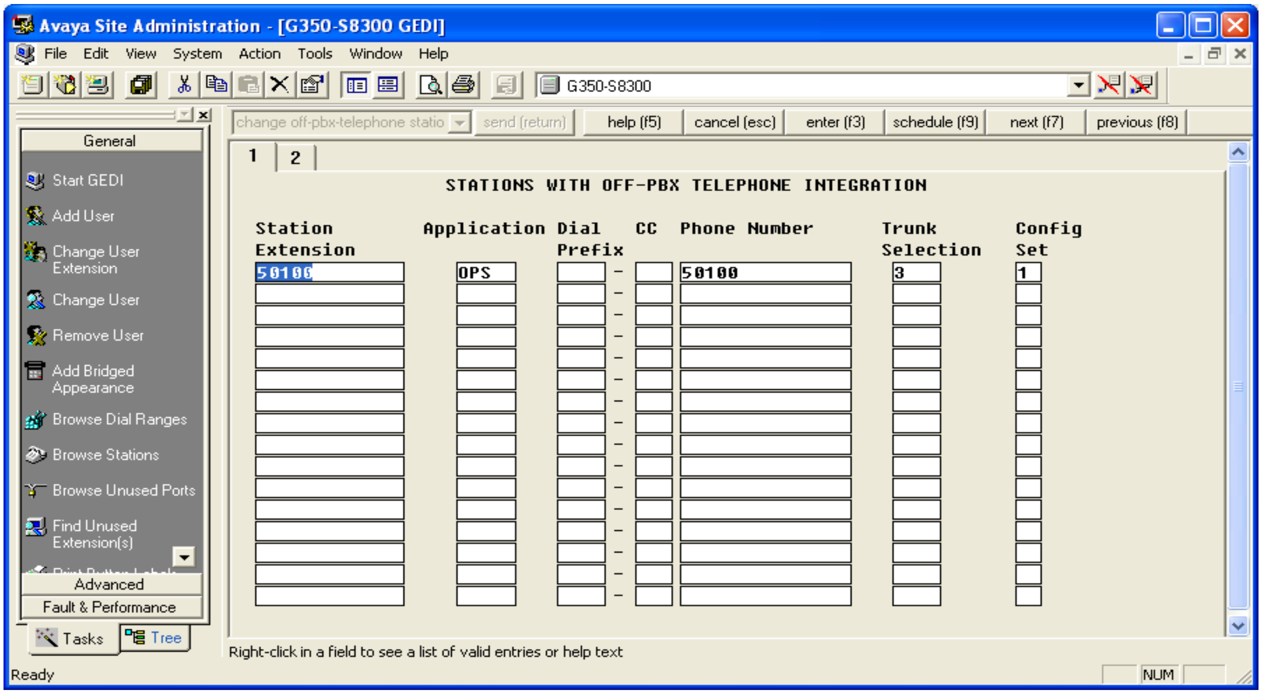
Enter command “add station xxx”, where xxx describes the number of the station.

- Set the Type for the station to 96xx (in this case 9630).
- The Port will be set automatically.
- Define a name for the station.
- Set the Security Code to a number you like.



Repeat this for all the SIP stations.

4.11 Enter the “change off-pbx-telephone station-mapping” command to add an Off-PBX-Station entry for every station created at step 3.9 and 3.10. In the Trunk Selection field you can specify aar, ars, or the trunk group number you want the call to be routed to.



4.12 Verify the synchronisation status of the MM710 card (PRI). Log in to the Avaya G350 gateway via putty. Enter the "show sync timing" command at the prompt. The status of the primary synchronisation at the MM710 card must be active. The screen below shows that the primary synchronisation is set to v2 port. (V2 is the port of the card MM710).

```
G350-001(super)# show sync timing
SOURCE     MM                  STATUS                  FAILURE
-------------  -------------------  ----------------------  ---------------
Primary        v2              Active                  None
Secondary                      Not Configured
Local          v0              Standby                 None
Active Source: v2            Sync Source Switching: Enabled
Done!
G350-001(super)#
```

If the primary synchronisation is not configured at the MM710 card (see the screen below), enter the following commands:

- Set Sync Interface Primary V2 (V2 is the port of the card MM710)
- Set Sync Source Primary

```
G350-001(super)# show sync timing
SOURCE     MM                  STATUS                  FAILURE
-------------  -------------------  ----------------------  ------------------
Primary                        Not Configured
Secondary                      Not Configured
Local          v0              Active                  None
Active Source: v0              Sync Source Switching: Enabled
Done!
G350-001(super)#
```

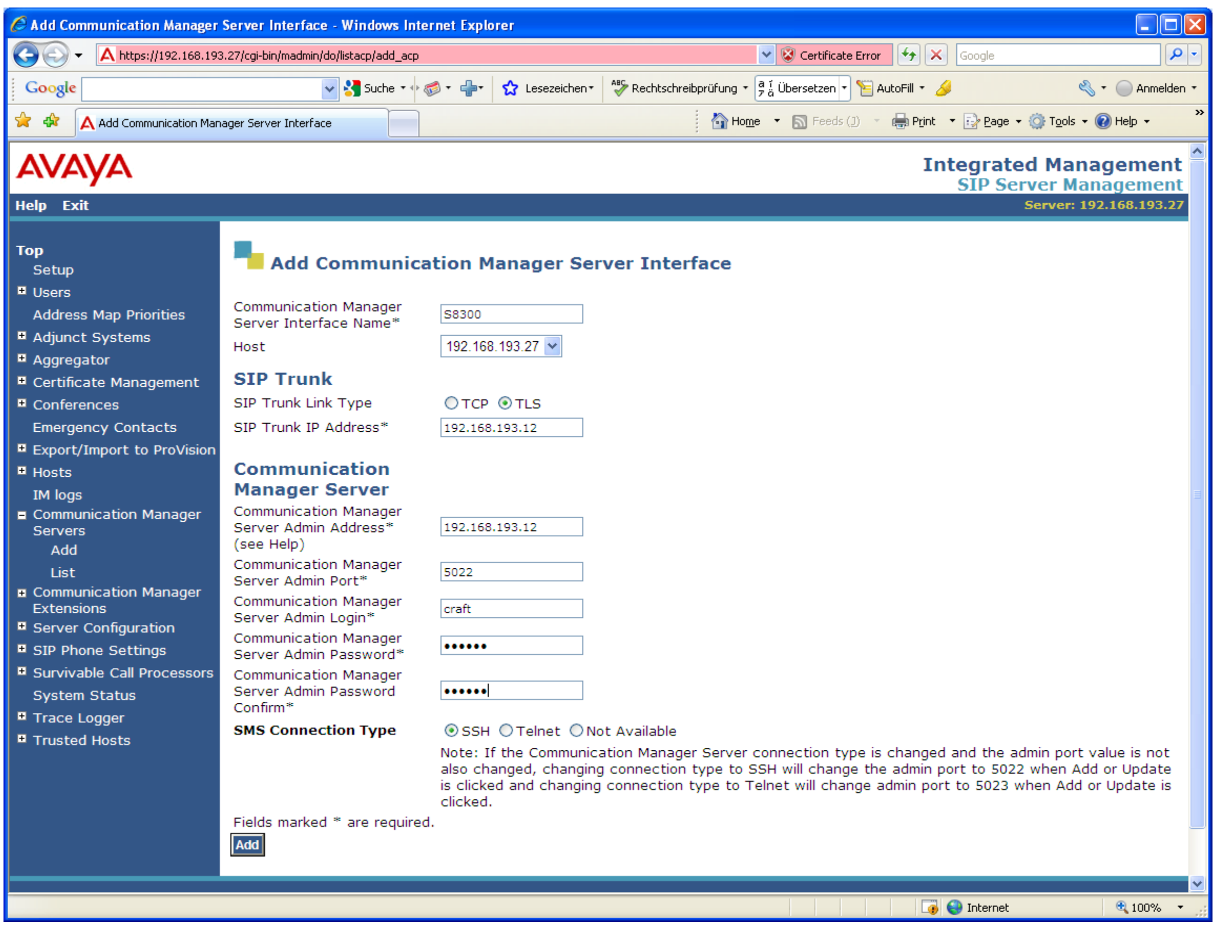
5. Configure Avaya SES

This section describes the steps for configuring the SES Edge. SES Edge is configured via a Web browser.

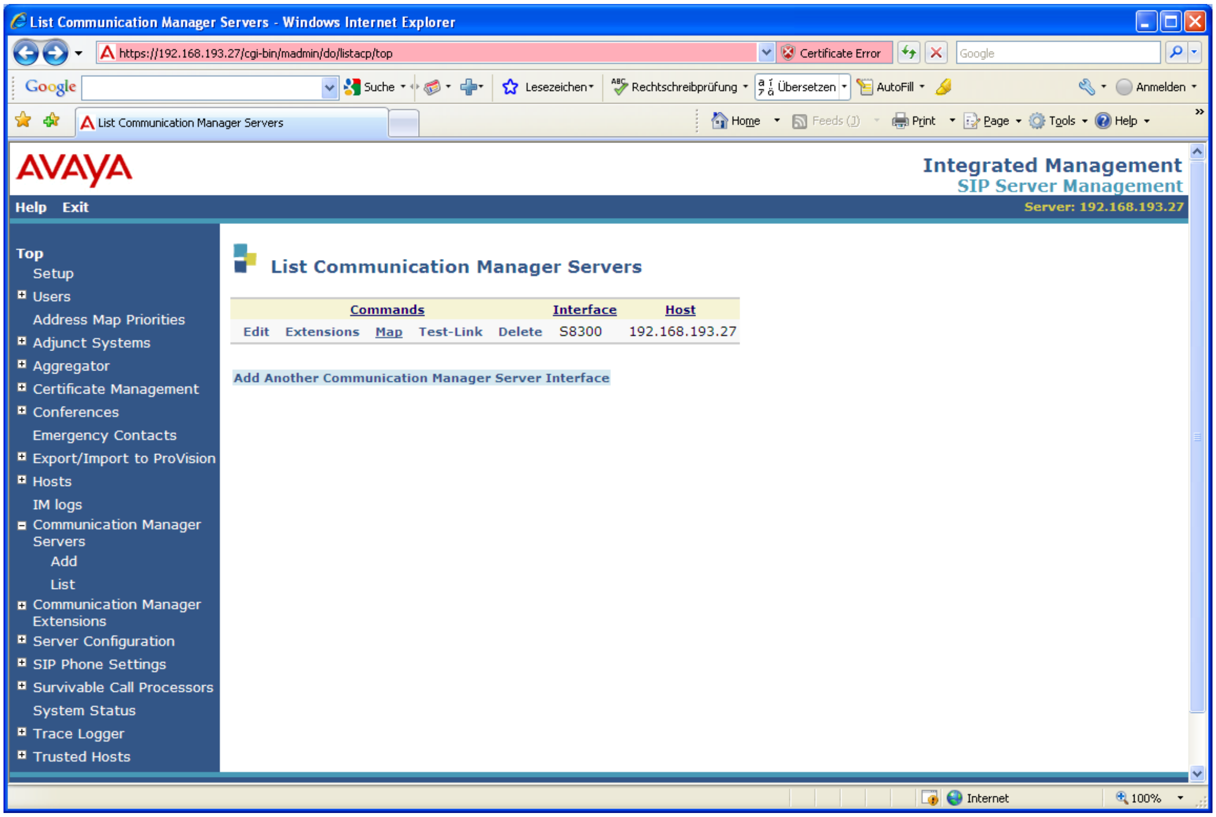
- Enter the <IP address>/admin and log in.
- Select Launch SES Administration interface.

5.1 From the left pane navigate to Communication Manager Servers – Add to bring up the Add Communication Manager Server interface screen.
Add a Communication Manager Server Connection to the SES by going through the following administration steps.

- Enter a name for Communication Manager Server Interface
- Set Host (SES IP address).
- Set SIP Trunk IP Address to the IP Address of the Avaya S8300 Server.
- Set Communication Manager Server Admin Address to the IP Address of the Avaya S8300 Server.
- Set Communication Manager Server Admin Login and Password.
- Click Add to save the changes.

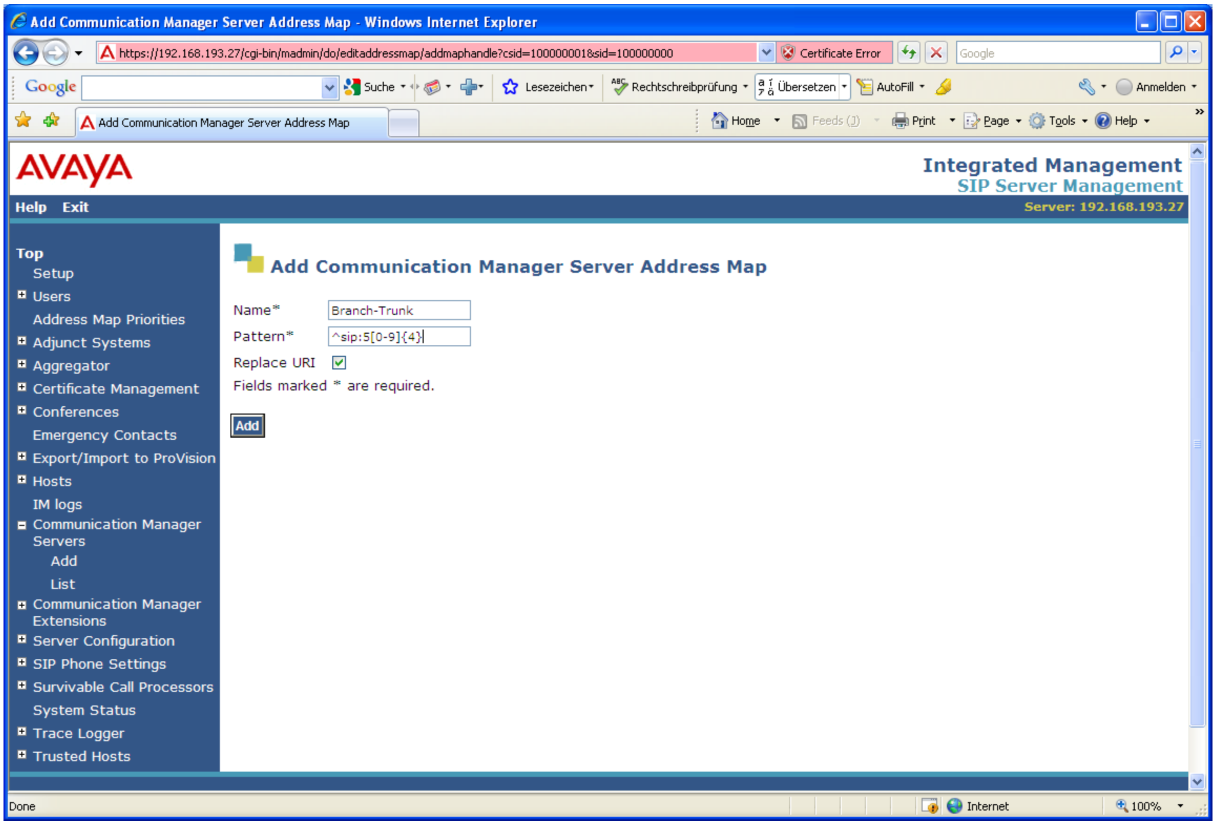


5.2 Add a Communication Manager Server Address Map.
From the left pane navigate to Communication Manager Servers – List to display the Communication Manager Servers screen.
Select the entry that was entered in step 5.1.
Click map to display the List Media Server Address Map screen



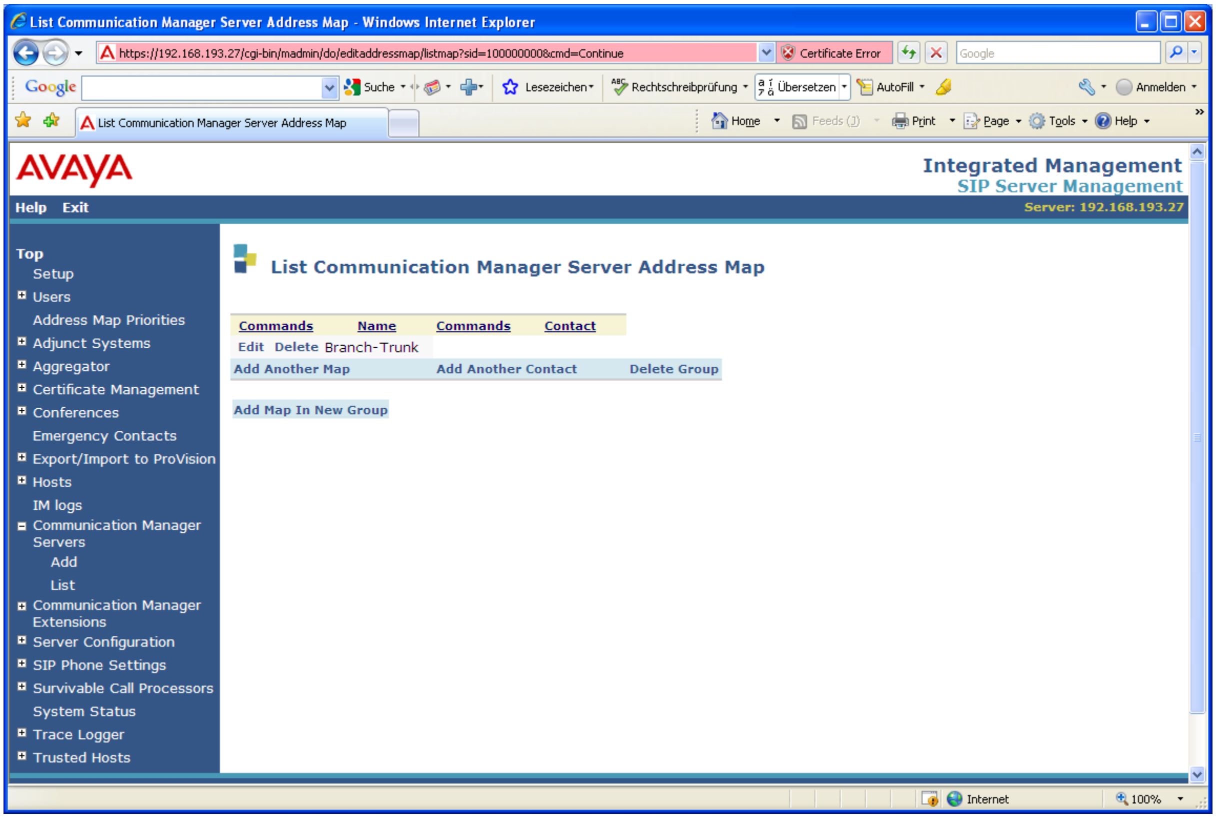
5.3 Click Map. The Add Communication Manager Server Address Map screen will be displayed.

- Enter a name for the Communication Manager Server Address Map.
- Set Pattern, in this case ^sip:5[0-9]{4}.
- Tick Replace URI.
- Click Add to save the changes.



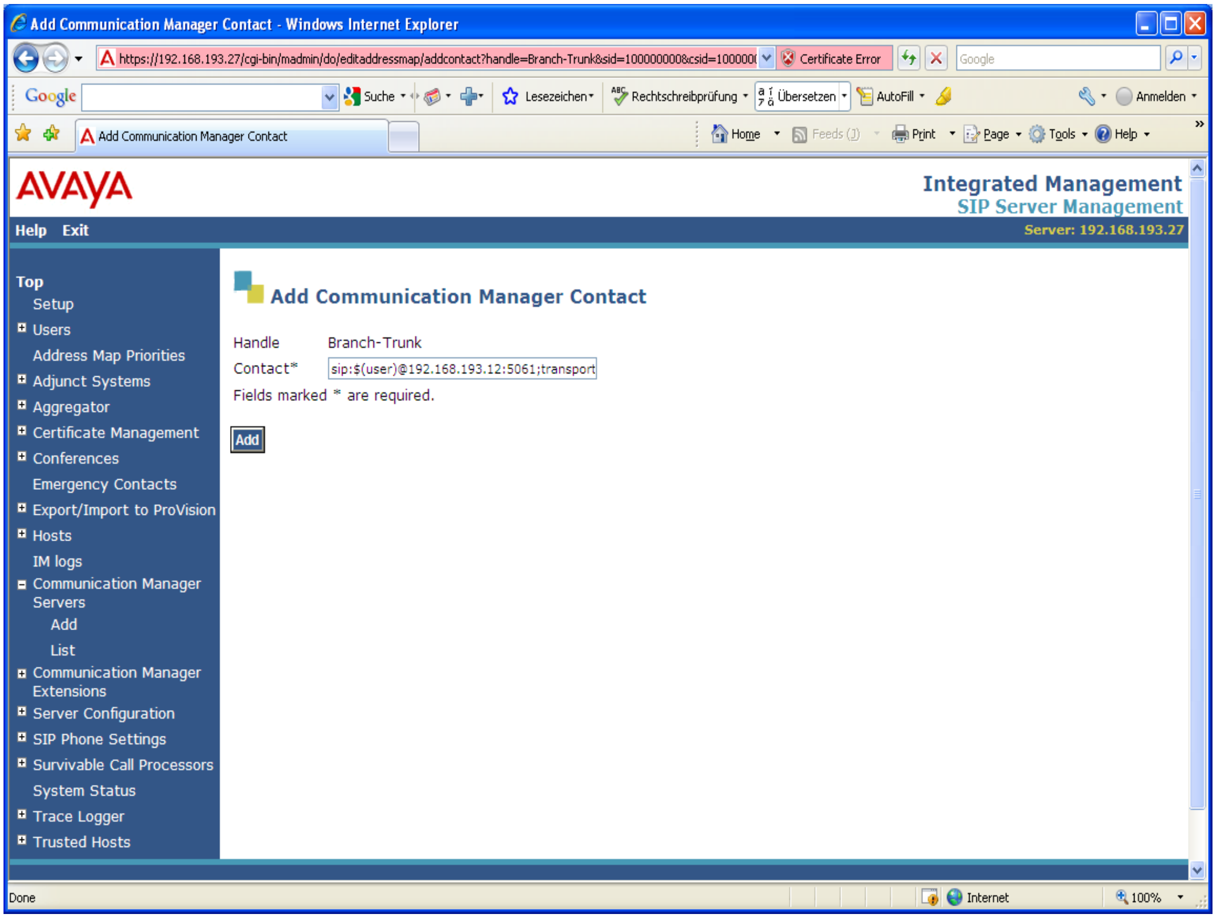
5.4 The List Communication Manager Server Address Map screen will be displayed.

- To add a contact for this address map press Add Another Contact

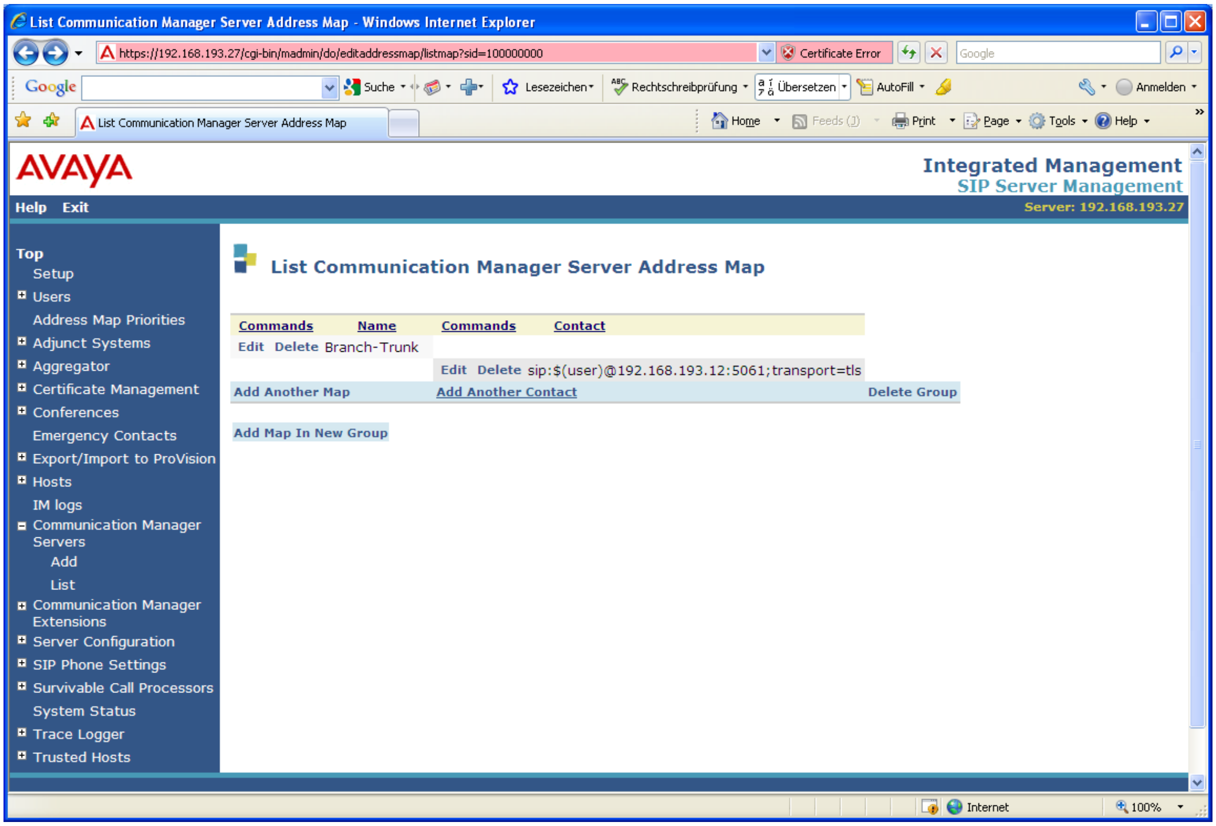


5.5 Create the first Media Server Contact and direct the calls to the IP address of the Avaya Media Server using port 5061 and TLS as the transport protocol.

- sip:$(user)@192.168.193.12:5061;transport=tls.
- Click Add to save the changes.



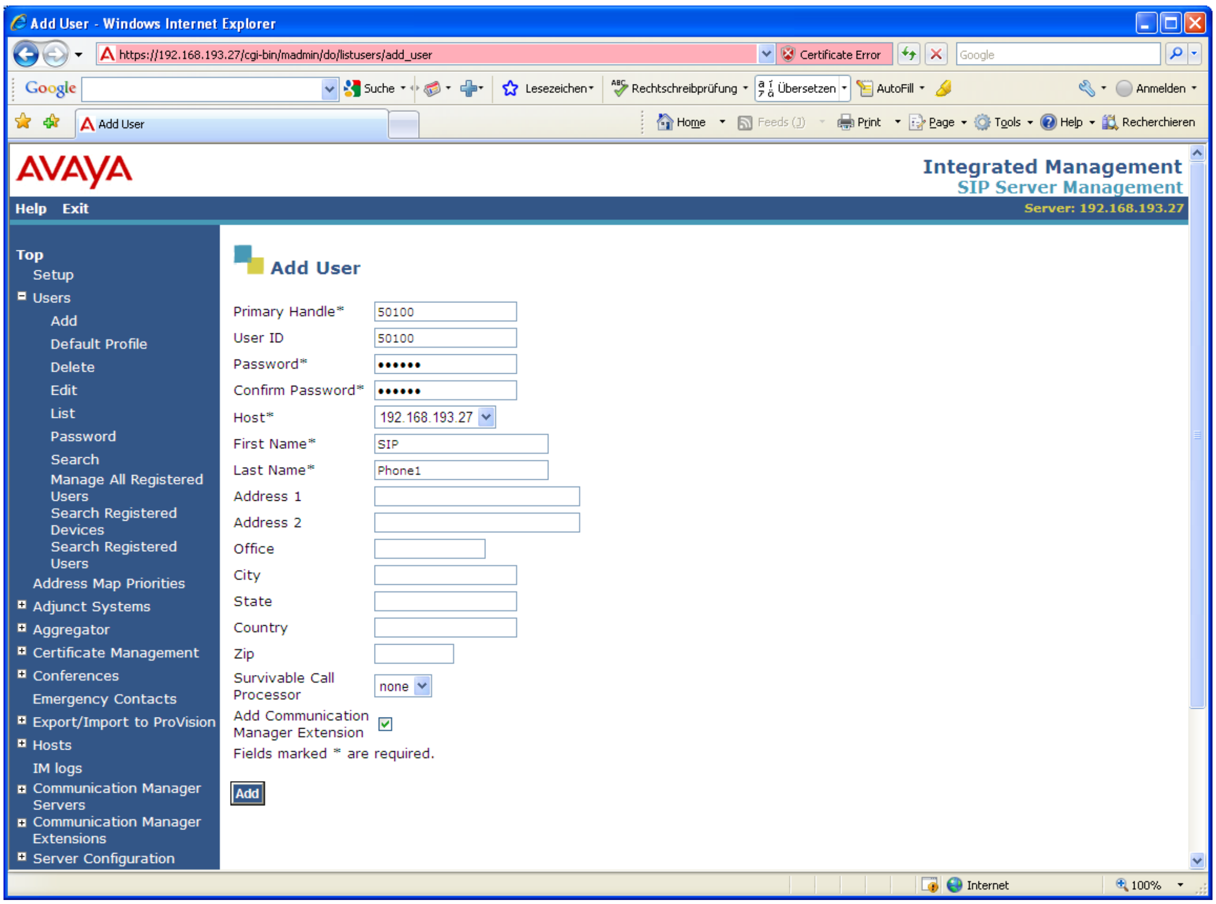
5.6 The Contact field is displayed as: sip:$(user)@192.168.193.12:5061;transport=tls.



5.7 Add a User for every FXS station, 96xx SIP station and the Mediant-600 Gateway.

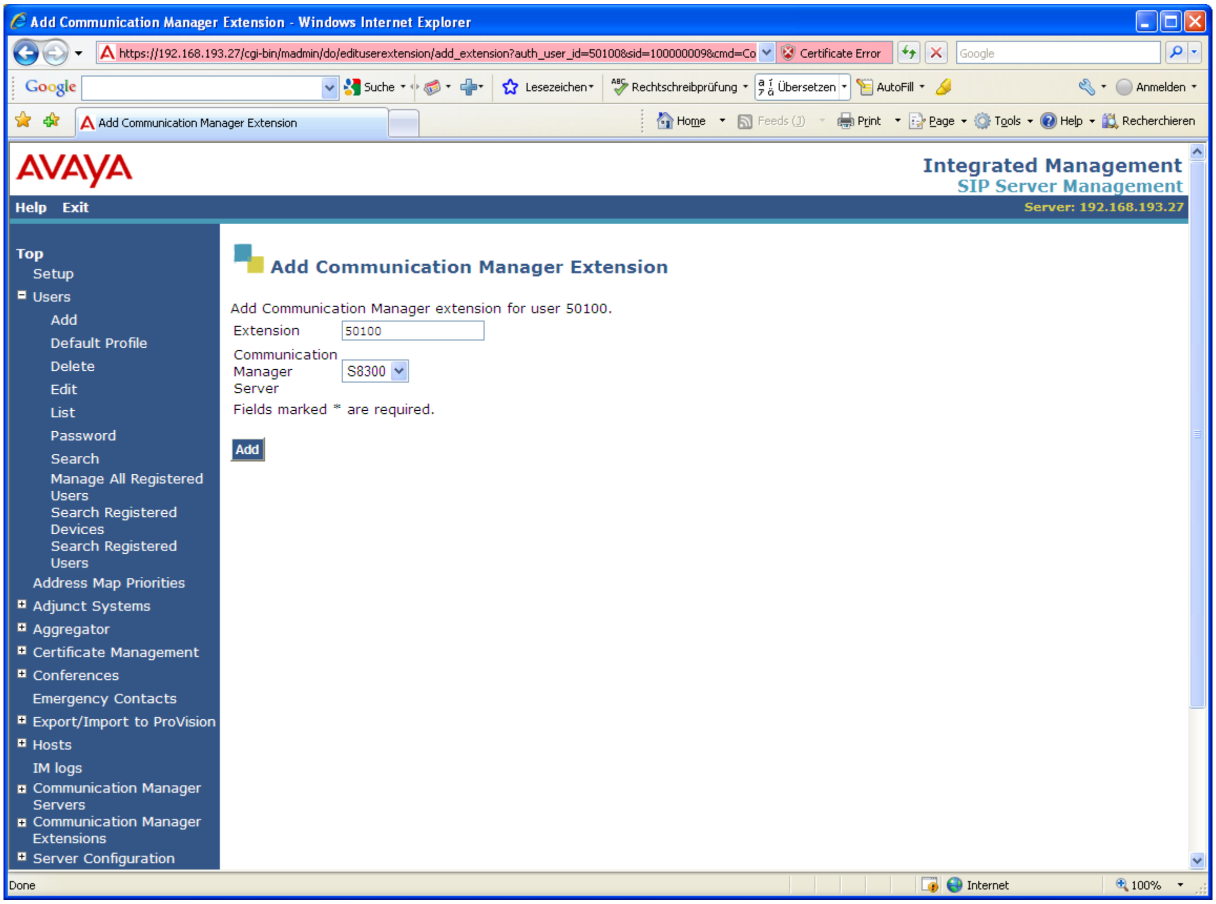
From the left pane, navigate to Users – Add. Enter the values as shown below.

- Primary Handle: Enter the extension for this user.
- User ID: Enter also the extension for this user.
- Password: Enter a password for logging into the SIP endpoint.
- Confirm Password: Re-enter the password.
- Host: Select the Avaya SES server.
- First name: Enter a name for this extension.
- Last name: Enter a name for this extension.
- Check the Add Communication Manager Extension box.



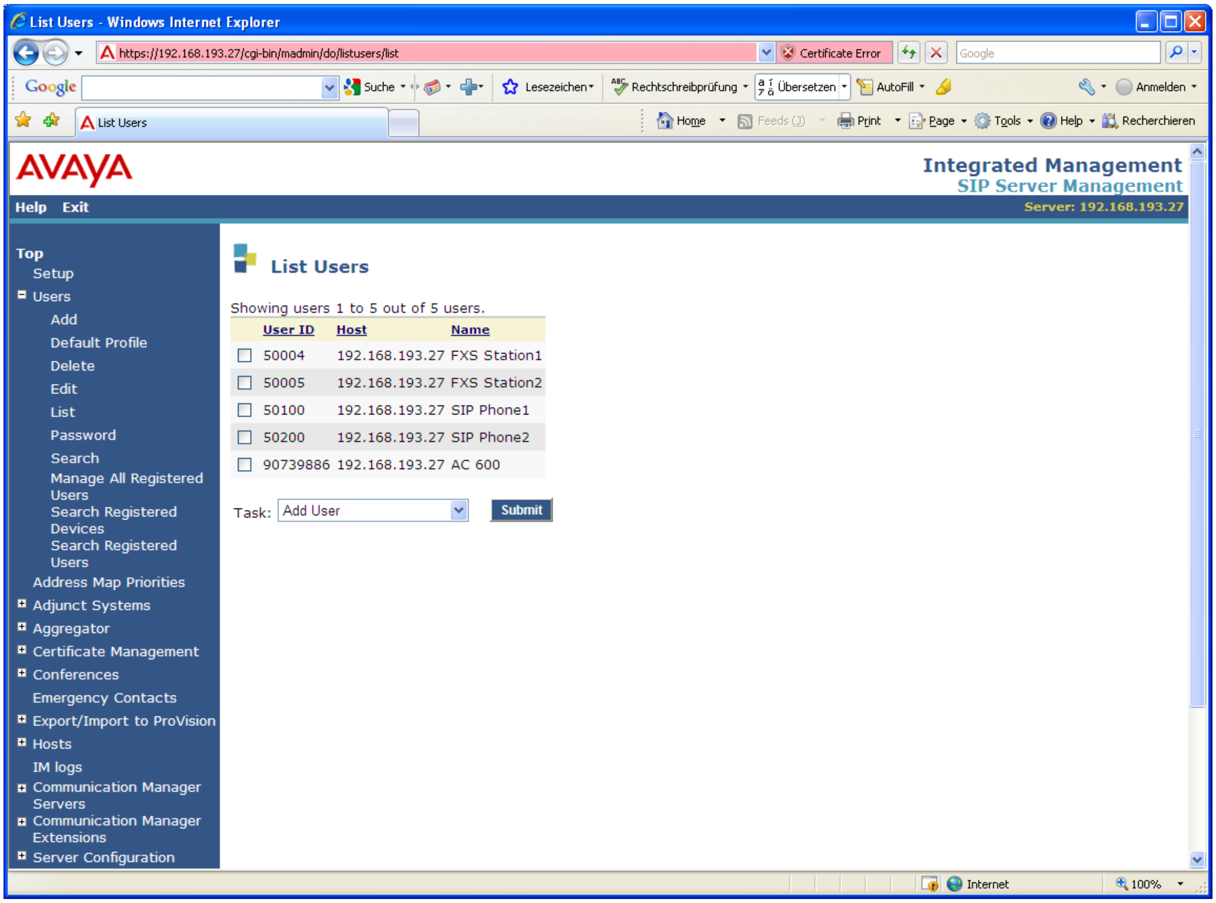
Note: The Mediant-600 Gateway user doesn't need a Communication Manager Extension. Do not check the Add Communication Manager Extension box, if the Mediant-600 Gateway user will be established.

5.8 Click the Add button. The Continue window will appear. Click the continue button. The Add Media Server Extension page will appear. In the Extension field enter the extension used on the previous screen.



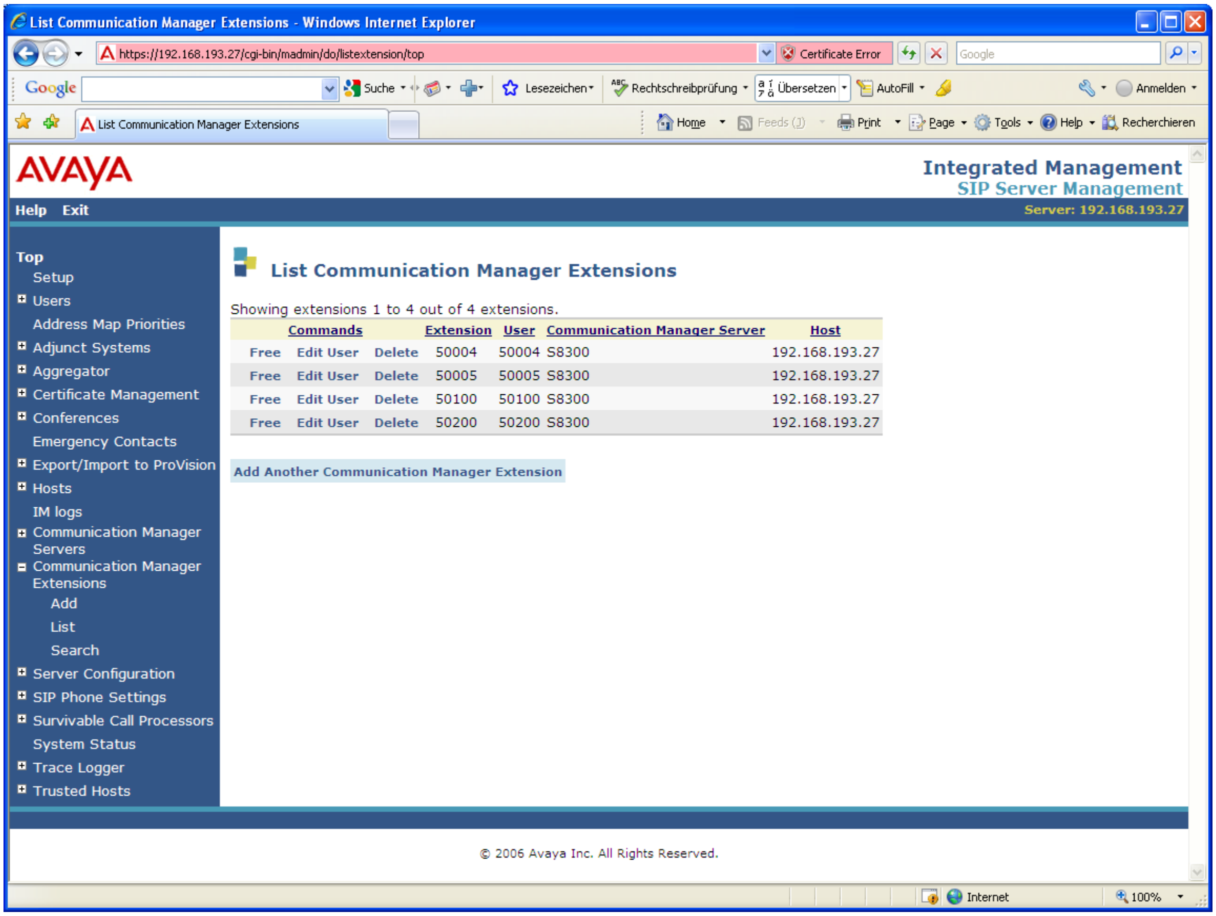
Click the **Add** button to save the changes.

5.9 List users. From the left pane, navigate to **Users – List**. The list shows all the users established.



5.10 List Communication Manager Extensions.

From the left pane, navigate to Communication Manager Extension – List. The list shows all the users established.



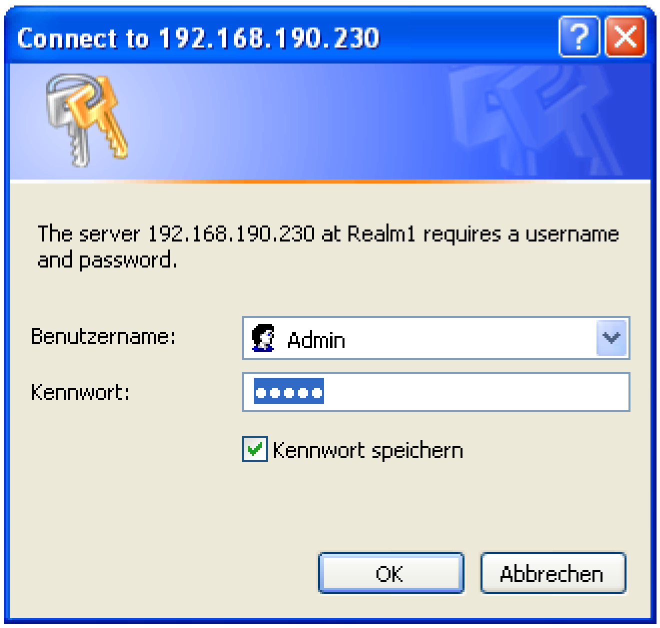
Note: The Communication Manager Extensions list doesn't contain a Mediant-600 gateway extension.

6. Configure the Mediant-600

This section describes the procedures for configuring the Mediant-600. The procedures show the configuration of all FXS and BRI ports even though not all ports were used in the test.

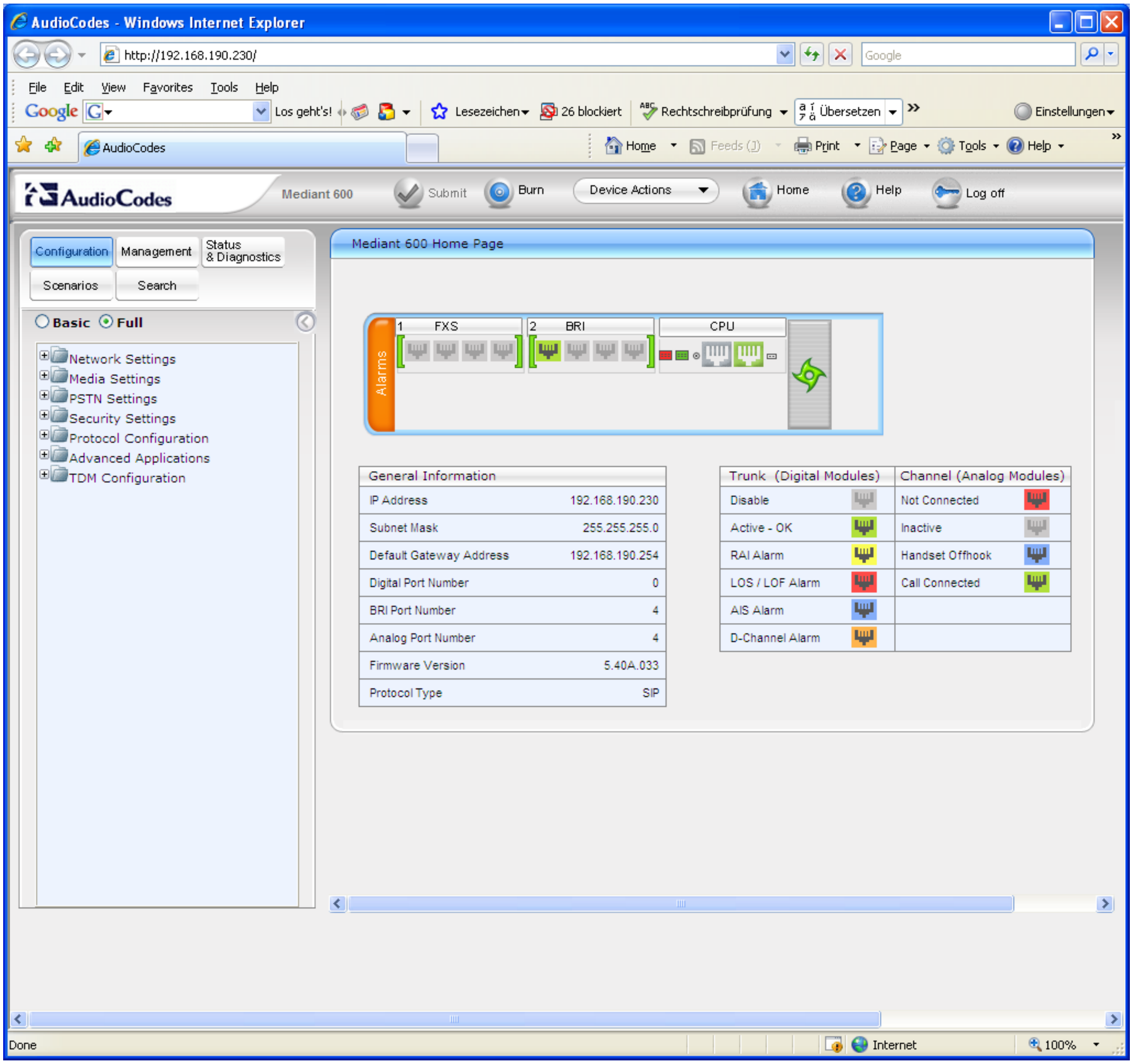
6.1 The configuration of the Mediant-600 is done via a Web browser.

- To have access enter the IP address of the device. Default address is 10.1.10.10.
- Set your PC / Laptop local IP to 10.1.10.11, if you use the default address.
- Connect your pc directly to the Mediant-600 using a cross over cable.
- Login and password are both “Admin”.



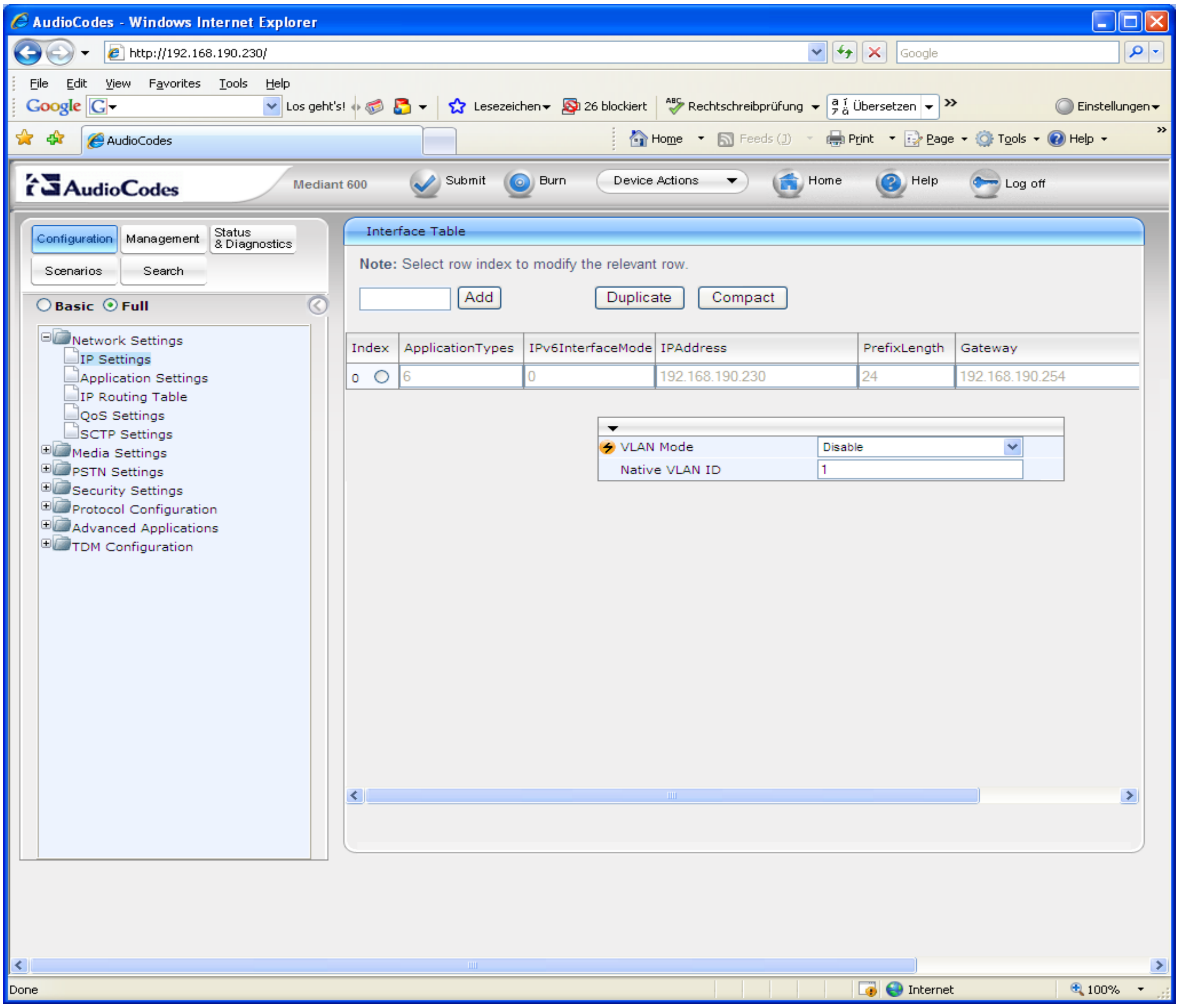
- Press ok.

The Mediant-600 main page will appear as shown below.



6.2 From the left pane, navigate to Network Settings – IP settings.

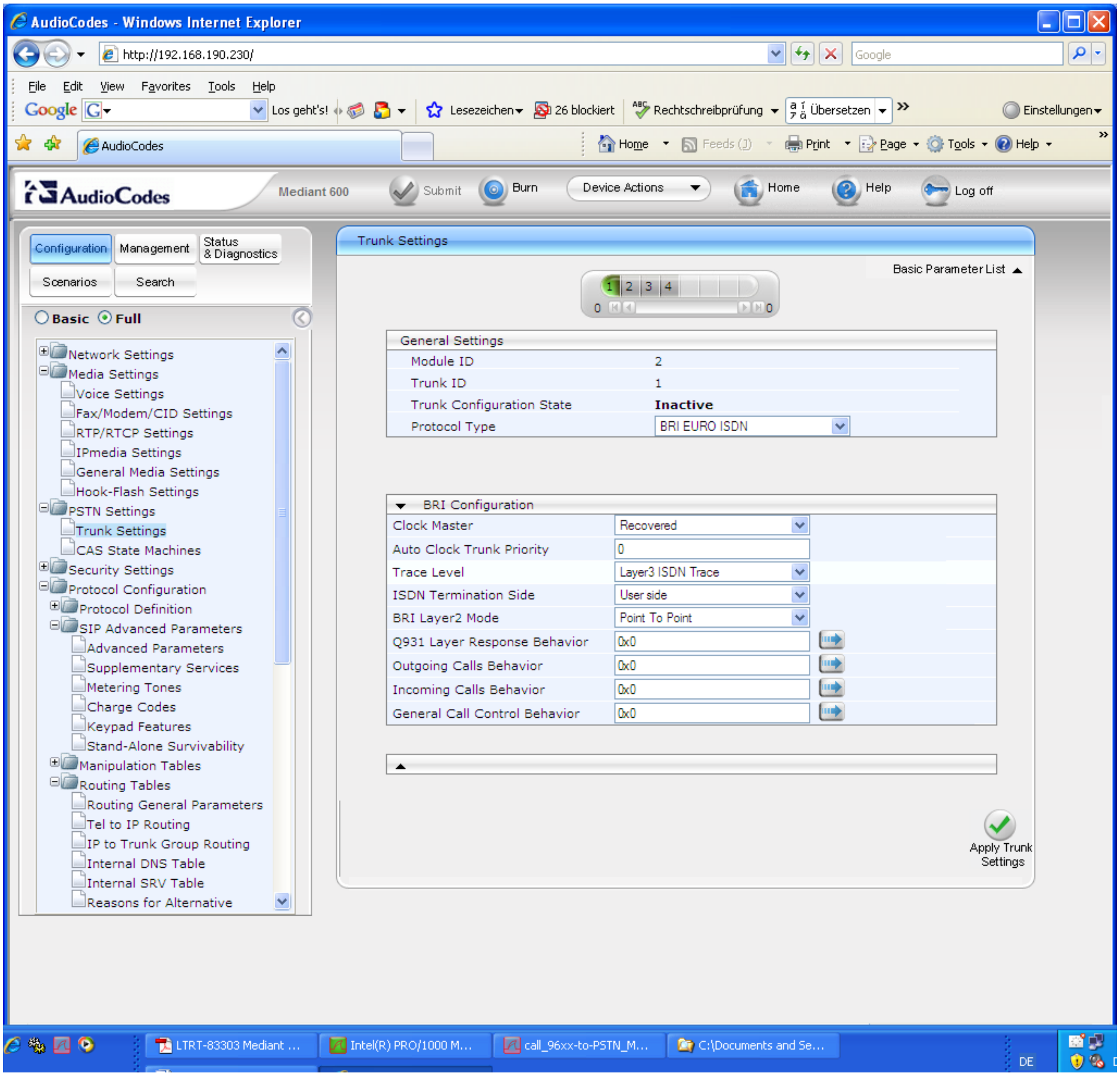
- For the test, the IP address, PrefixLength (Subnet mask) and Gateway address were set to values consistent with the test configuration shown in Figure 1.
- Click submit to set the changes.



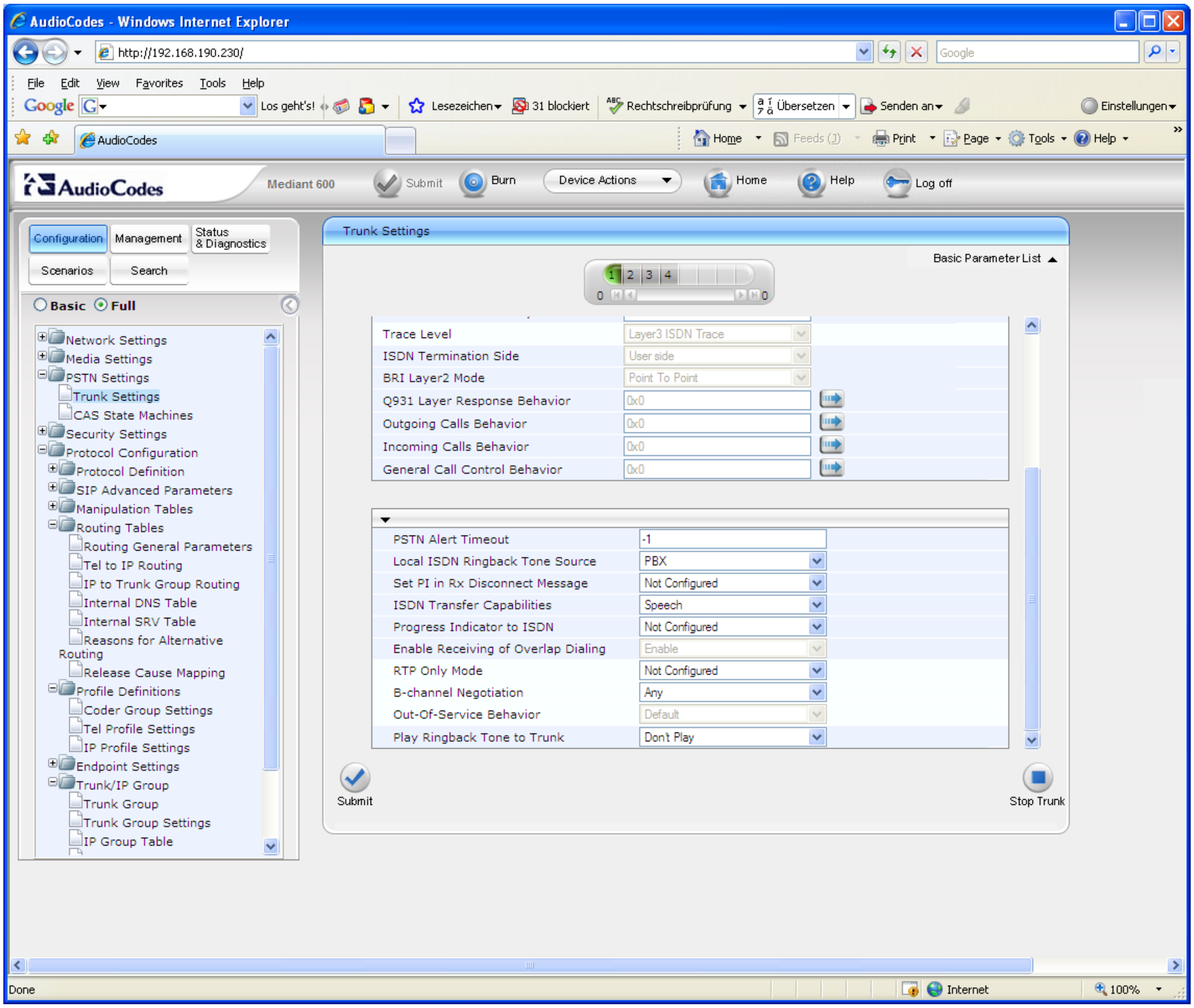
Note: In the test configuration the VLAN Mode must be disabled. This is dependent on the network configuration. In an earlier test configuration the entry "VLANMODE = 1" was deleted in the configuration file, because the entry wasn't shown on the Web screen.

6.3 From the left pane, navigate to PSTN Settings - Trunk Settings. Refer to the next two screen shots for the resulted settings.

- Set Protocol Type to BRI EURO ISDN.
- Set ISDN Termination Side to User side.
- For the BRI Layer 2 Mode field, select Point to Point.

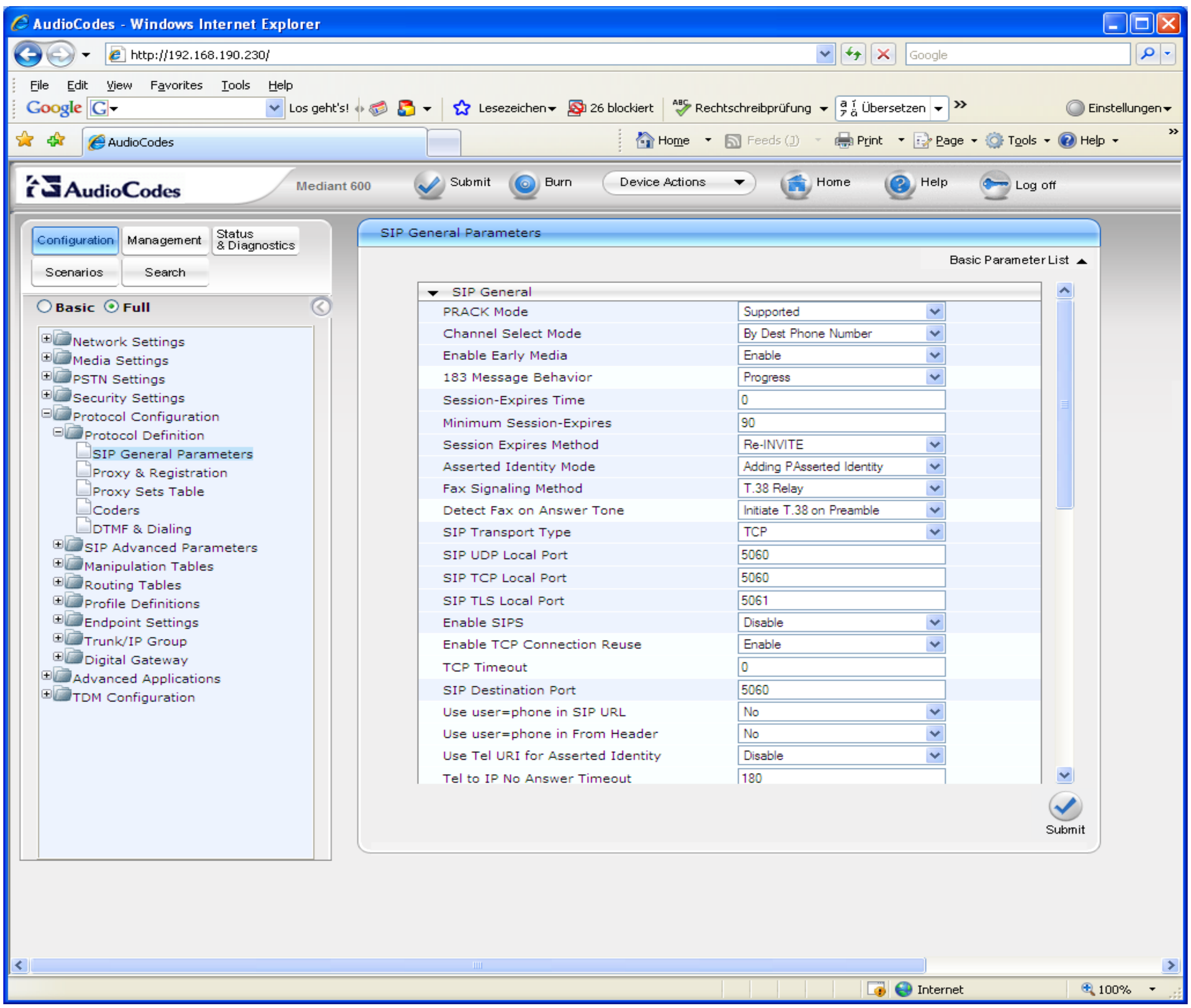


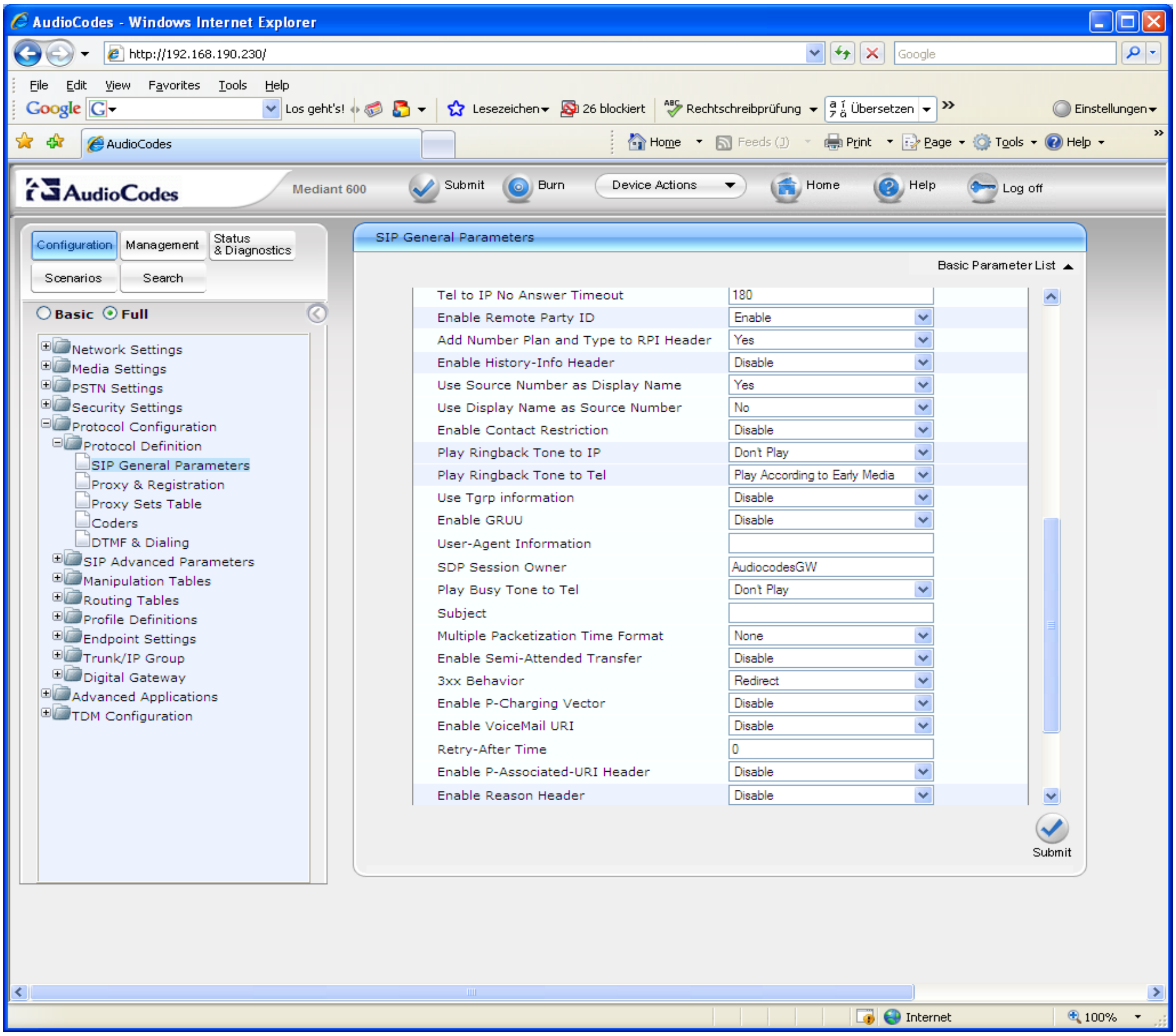
- For the ISDN Transfer Capabilities field, select Speech.
- Set B-channel Negotiation to Any.
- Set Out-Of-Service Behaviour to Default.
- Click Apply Trunk Settings.
- Click Submit.



6.4 From the left pane, navigate to Protocol Configuration - Protocol Definition – SIP General Parameters. Make sure to select the 'Full' radio button above the tree view in order to see everything. Refer to the next two screen shots for the resulted settings.

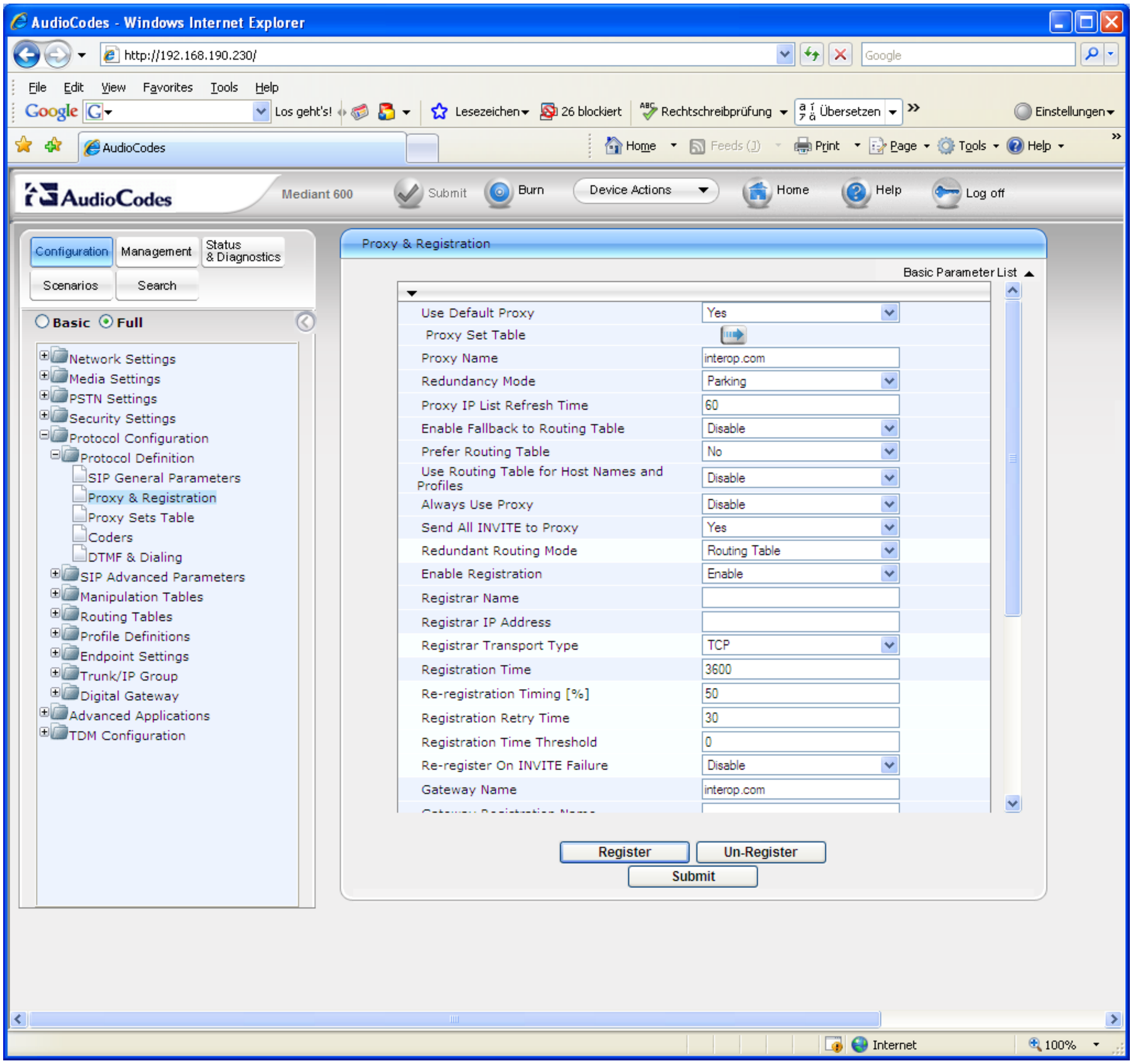
- For the Enable Early Media field, select Enable.
- Set SIP Transport Type to TCP.
- Select No for the Use user=phone in SIP URL field.
- For the Fax Signalling Method field, select T.38 Relay.
- For the Detect fax on Answer Tone field, select initiate T.38 on Preamble.
- For the Asserted Identity Mode field, select Adding PAsserted Identity.
- For the Enable Remote party ID field, select Enable.
- For the Add Number Plan and Type to RPI Header field, select Yes.
- Click Submit.

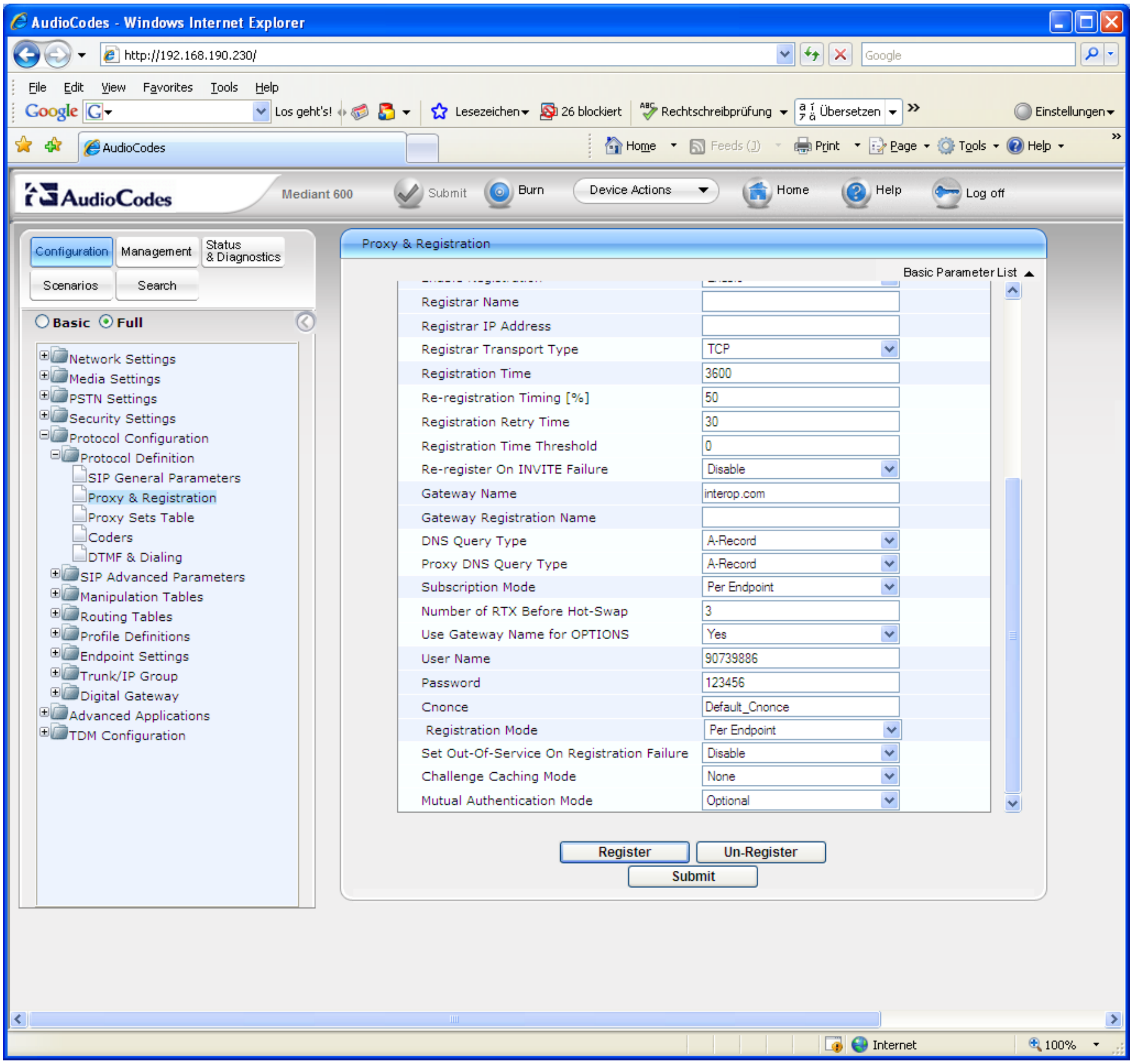



AudioCodes - Windows Internet Explorer
http://192.168.190.230/
File Edit View Favorites Tools Help
AudioCodes
Mediant 600
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Management
Status & Diagnostics
Scenarios
Search
Basic
Full
Network Settings
Media Settings
PSTN Settings
Security Settings
Protocol Configuration
Protocol Definition
SIP General Parameters
Proxy & Registration
Proxy Sets Table
Coders
DTMF & Dialing
SIP Advanced Parameters
Manipulation Tables
Routing Tables
Profile Definitions
Endpoint Settings
Trunk/IP Group
Digital Gateway
Advanced Applications
TDM Configuration
SIP General Parameters
Basic Parameter List
Tel to IP No Answer Timeout 180
Enable Remote Party ID Enable
Add Number Plan and Type to RPI Header Yes
Enable History-Info Header Disable
Use Source Number as Display Name Yes
Use Display Name as Source Number No
Enable Contact Restriction Disable
Play Ringback Tone to IP Don't Play
Play Ringback Tone to Tel Play According to Early Media
Use Tgrp information Disable
Enable GRUU Disable
User-Agent Information
SDP Session Owner AudiocodesGW
Play Busy Tone to Tel Don't Play
Subject
Multiple Packetization Time Format None
Enable Semi-Attended Transfer Disable
3xx Behavior Redirect
Enable P-Charging Vector Disable
Enable VoiceMail URI Disable
Retry-After Time 0
Enable P-Associated-URI Header Disable
Enable Reason Header Disable
Submit
Internet
100%

6.5 Navigate to Protocol Configuration – Protocol Definition – Proxy and Registration and set all the options as shown below.

Note that the Proxy Name is set to the same domain as your SES, and the Gateway Name is set to the same Far-end Domain as specified on the signalling group form on your CM.

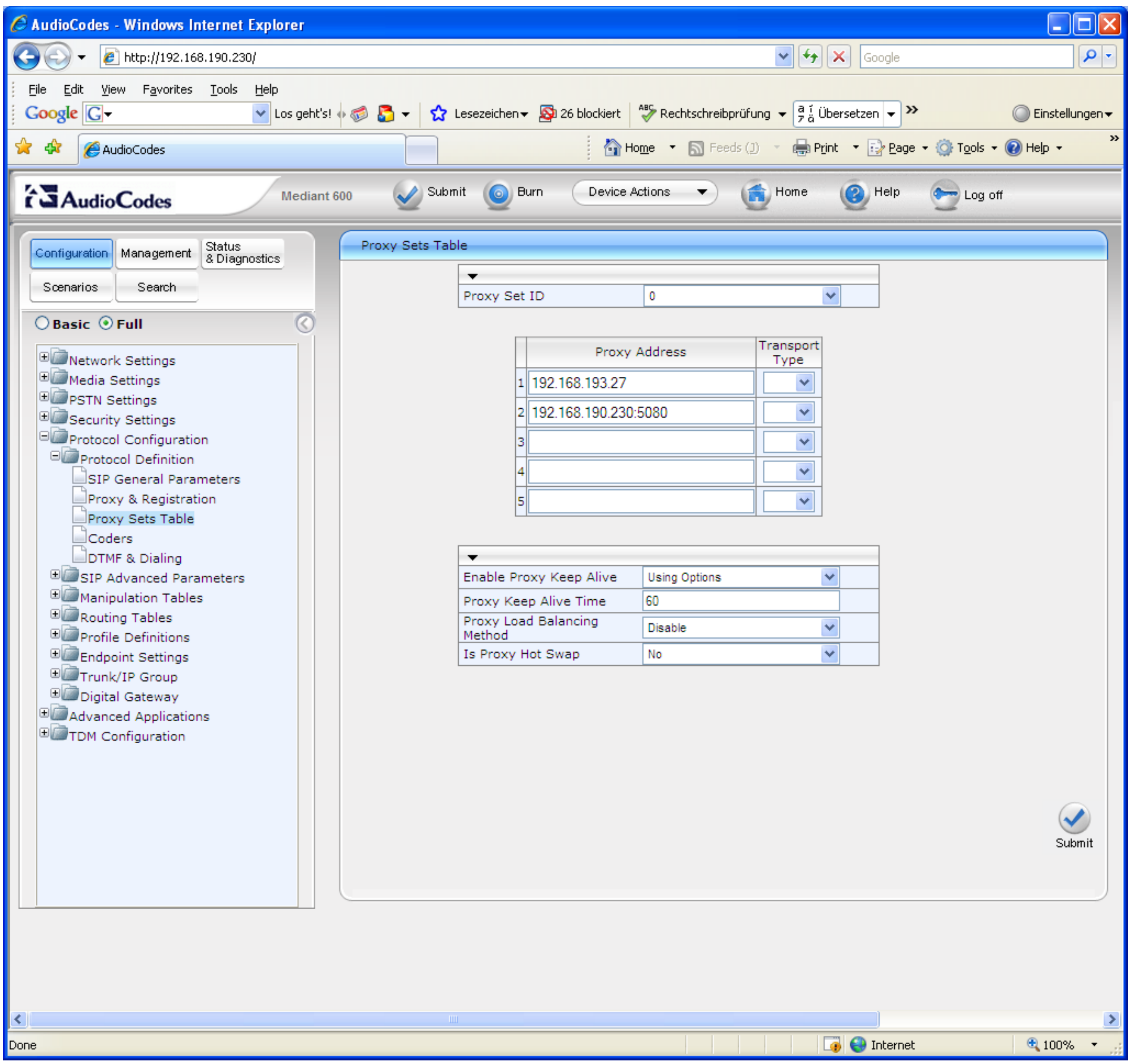



AudioCodes - Windows Internet Explorer
http://192.168.190.230/
Mediant 600
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Management
Status & Diagnostics
Scenarios
Search
Basic
Full
Network Settings
Media Settings
PSTN Settings
Security Settings
Protocol Configuration
Protocol Definition
SIP General Parameters
Proxy & Registration
Proxy Sets Table
Coders
DTMF & Dialing
SIP Advanced Parameters
Manipulation Tables
Routing Tables
Profile Definitions
Endpoint Settings
Trunk/IP Group
Digital Gateway
Advanced Applications
TDM Configuration
Proxy & Registration
Basic Parameter List
Registrar Name
Registrar IP Address
Registrar Transport Type TCP
Registration Time 3600
Re-registration Timing [%] 50
Registration Retry Time 30
Registration Time Threshold 0
Re-register On INVITE Failure Disable
Gateway Name interop.com
Gateway Registration Name
DNS Query Type A-Record
Proxy DNS Query Type A-Record
Subscription Mode Per Endpoint
Number of RTX Before Hot-Swap 3
Use Gateway Name for OPTIONS Yes
User Name 90739886
Password 123456
Cnonce Default_Cnonce
Registration Mode Per Endpoint
Set Out-Of-Service On Registration Failure Disable
Challenge Caching Mode None
Mutual Authentication Mode Optional
Register
Un-Register
Submit
Internet
100%

6.6 Navigate to Protocol Configuration – Protocol Definition – Proxy Sets Table to open the Proxy Sets Table. We need two proxy set ID's for the normal mode and fail-over mode.

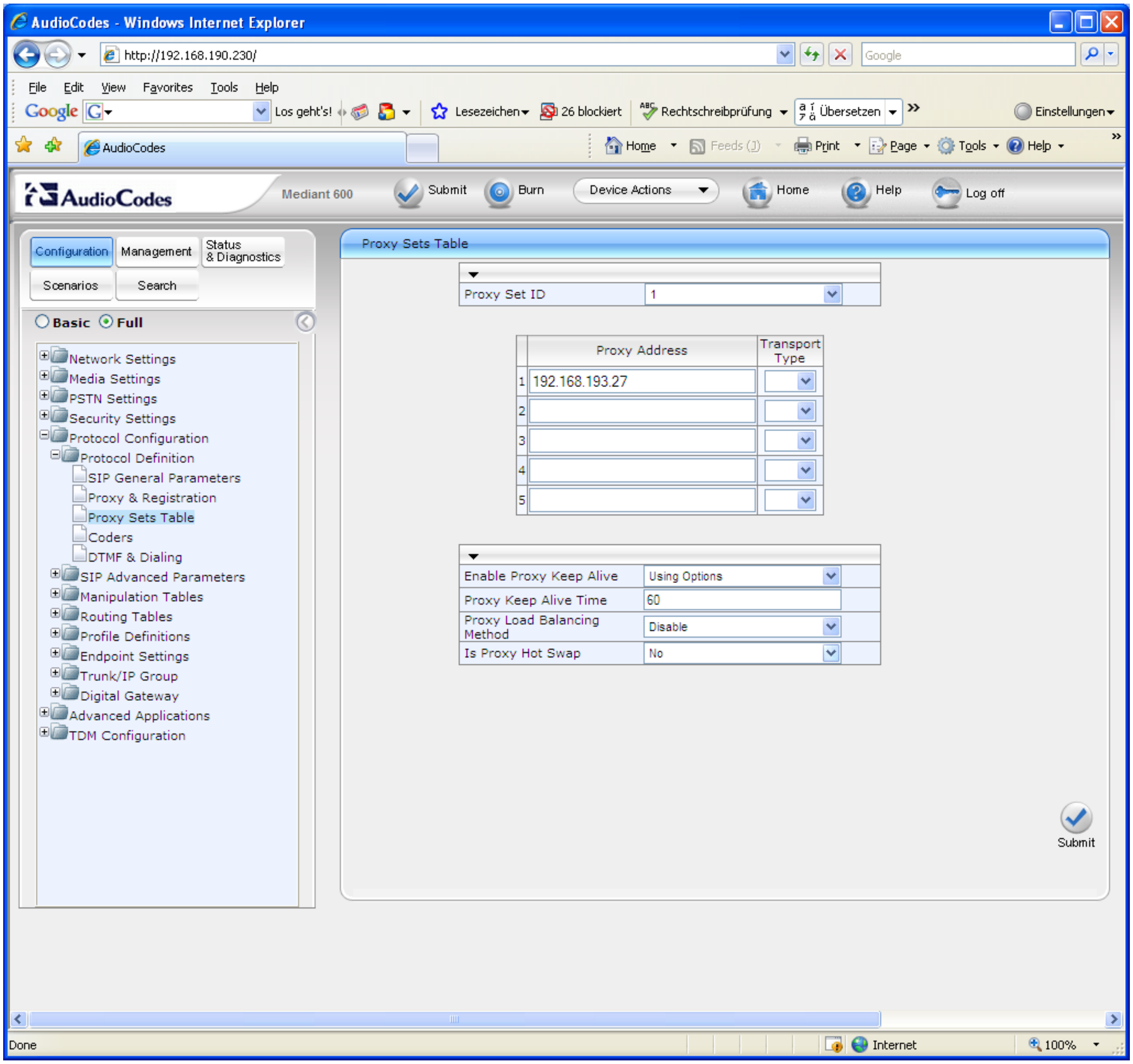
Proxy Set ID 0

- Select 0 for the Proxy Set ID.
- Set the first Proxy Address to the IP address of your SES Home.
- Set the second Proxy Address to the IP address of your Mediant-600 and port 5080.
- Set Enable Proxy Keep Alive to Using Options
- Click Submit to save the changes.



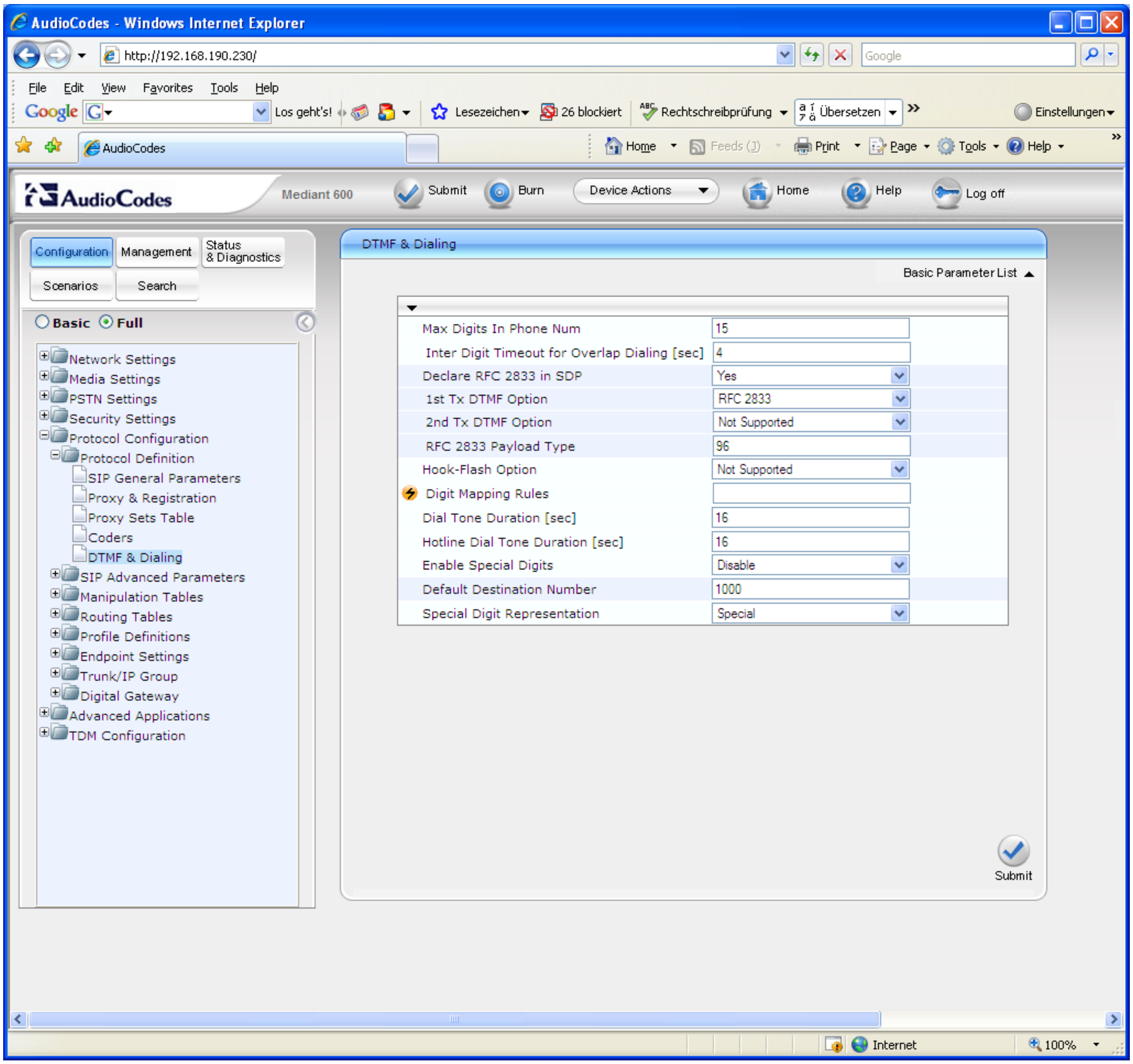
Proxy Set ID 1

- Select 1 for the Proxy Set ID.
- Set the first Proxy Address to the IP address of your SES Home.
- Set Enable Proxy Keep Alive to Using Options
- Click Submit to save the changes.



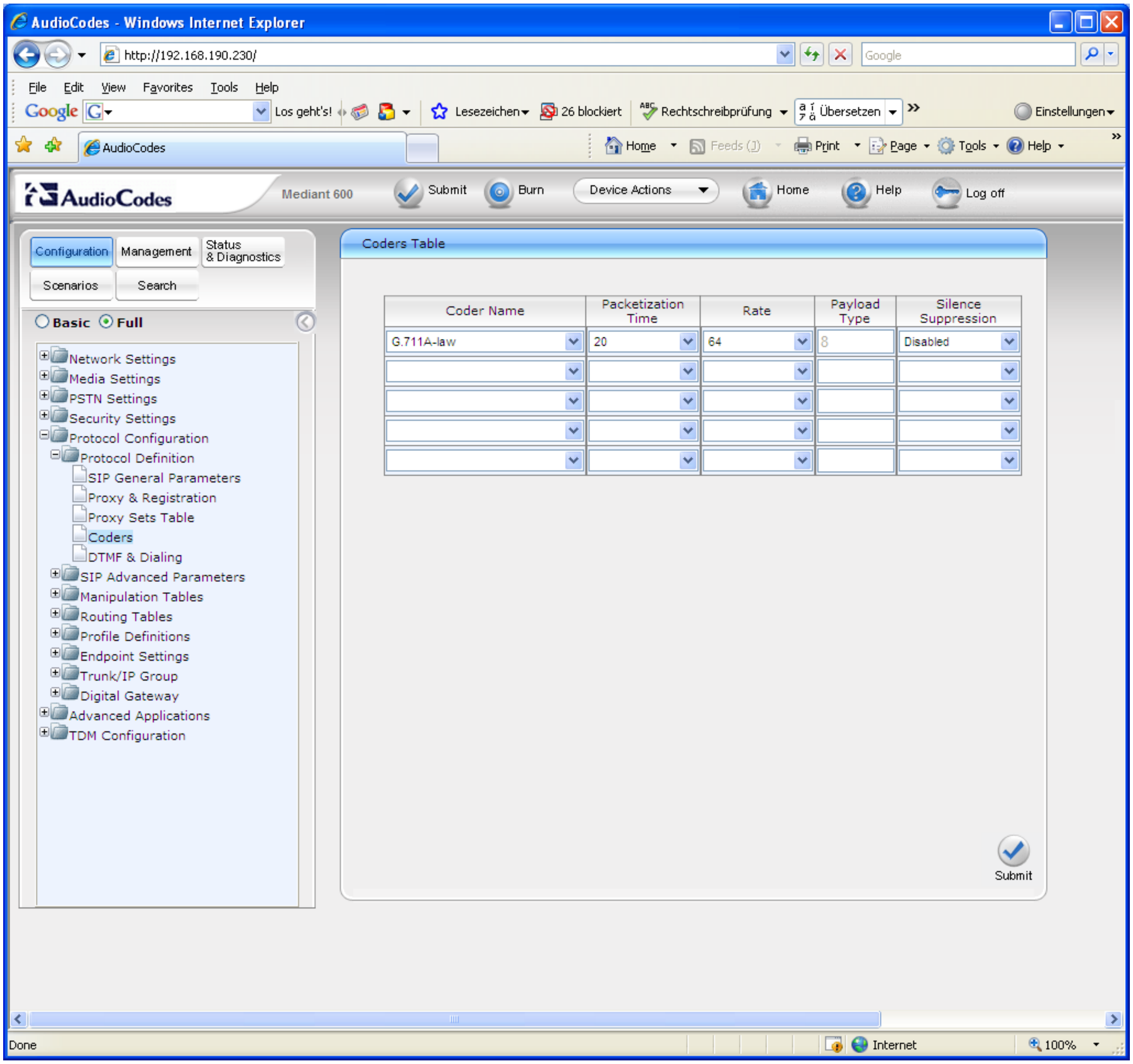
6.7 Navigate to Protocol Configuration – Protocol Definition – DTMF & Dialing.

- Set Max Digits in Phone Num to 15 or a greater number.
- Make sure that the Digit Mapping Rules field is empty.
- Click Submit to save the changes.

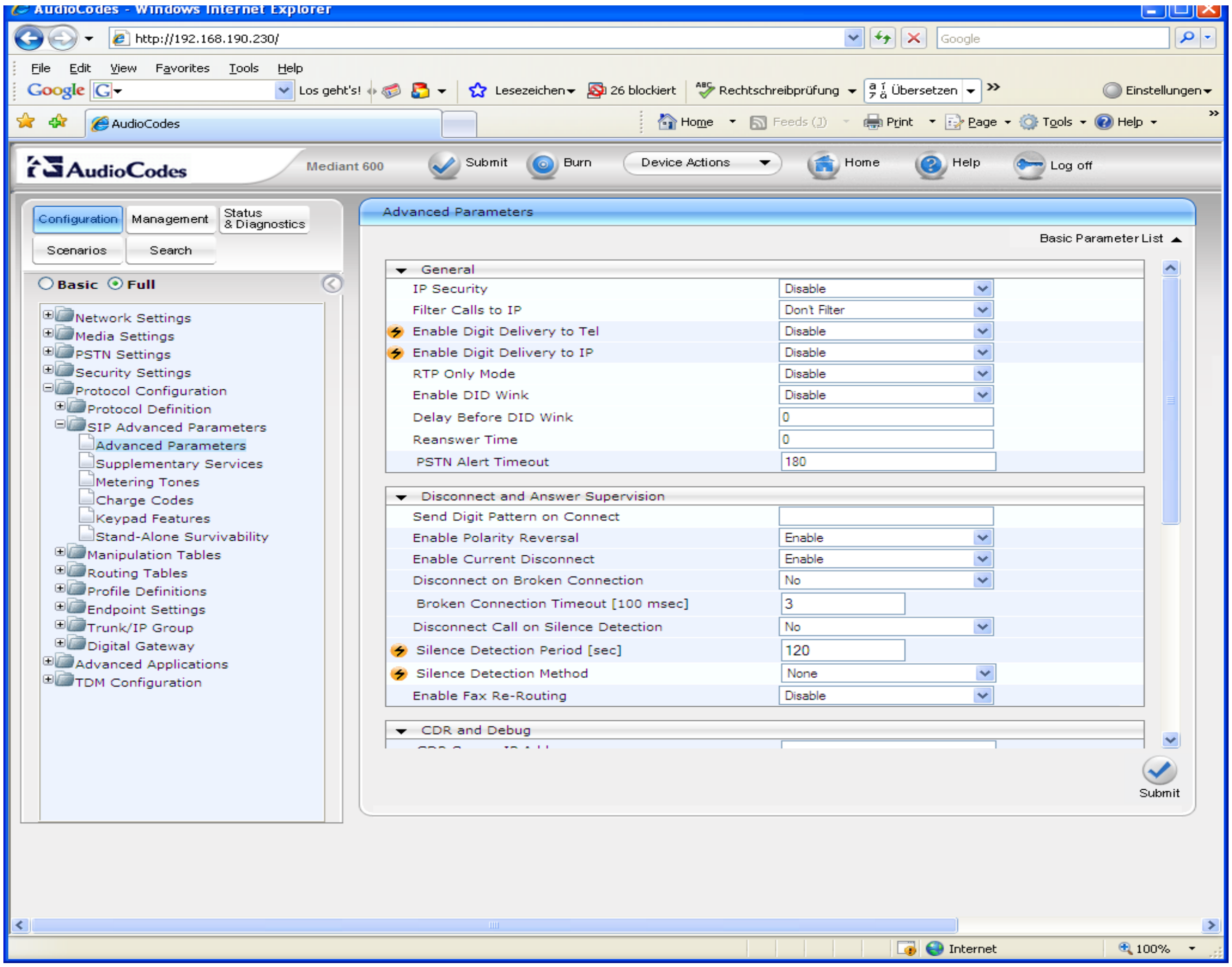


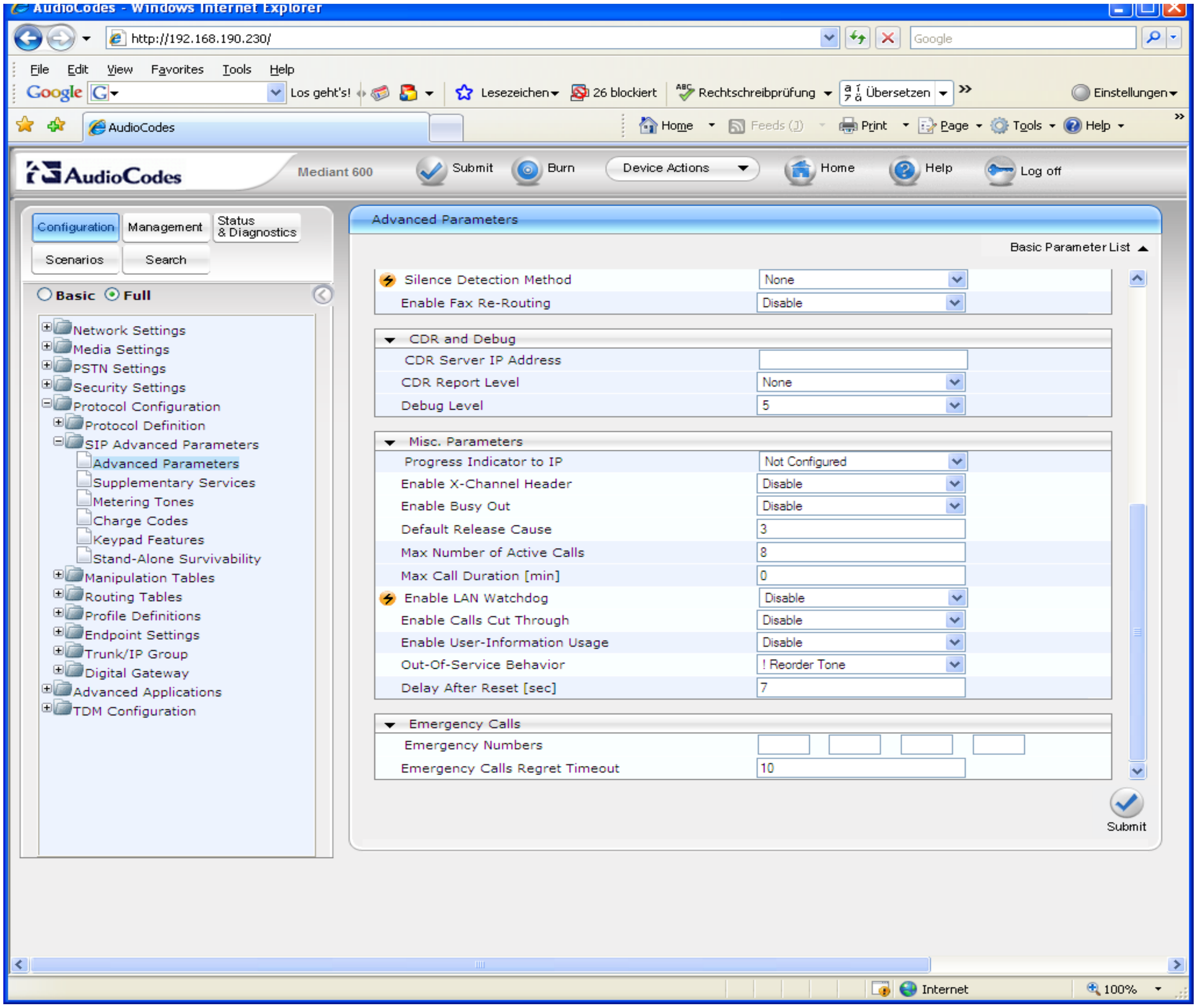
6.8 Navigate to Protocol Configuration – Protocol Definition – Coders.

- Set the preferred codec priority for your gateway.
- In test only the G.711 A-law codec was used.
- Click Submit to save the changes.



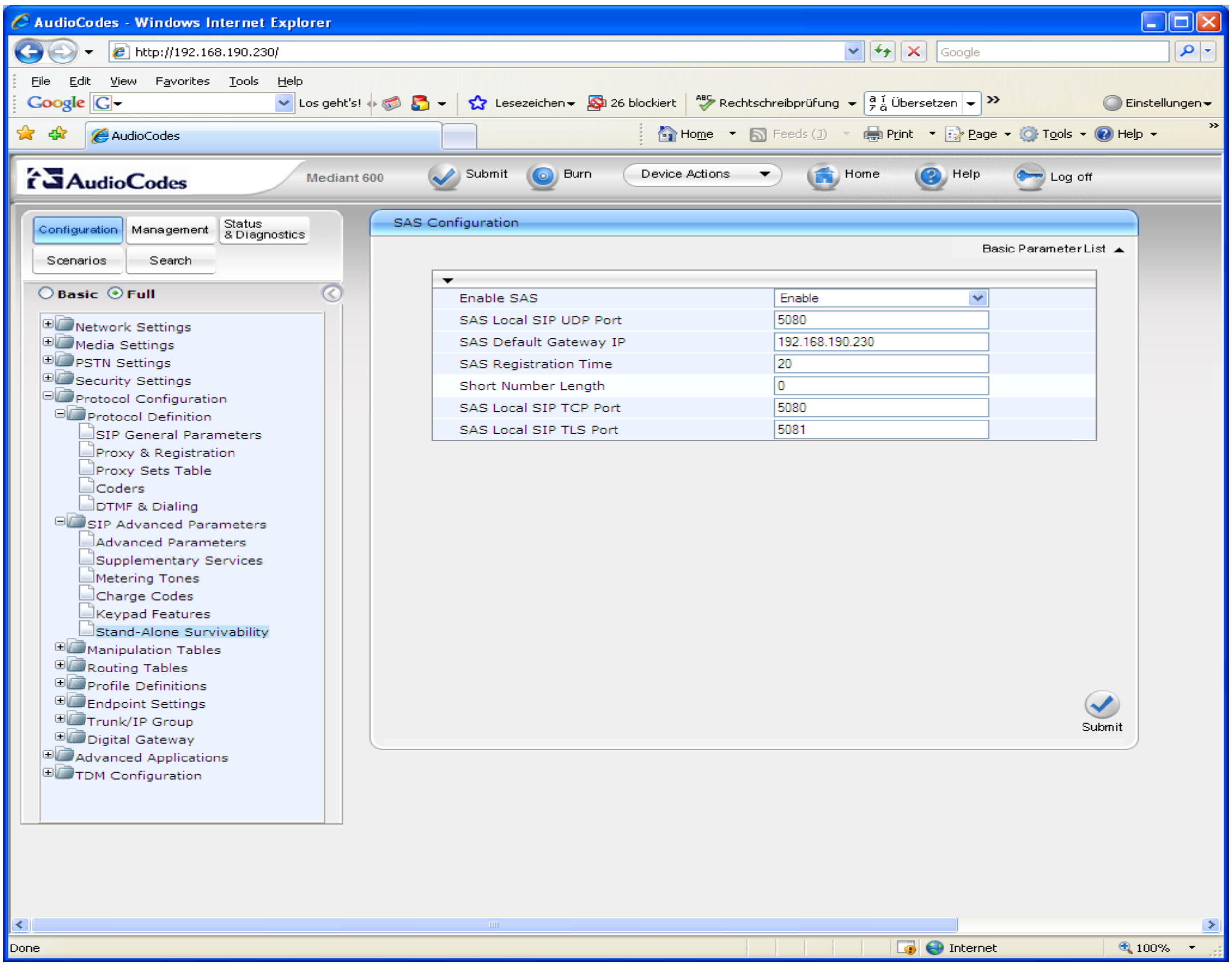
6.9 Navigate to Protocol Configuration – SIP Advanced Parameters – Advanced Parameters and set the parameters as shown in the next two screen shots.




AudioCodes - Windows Internet Explorer
http://192.168.190.230/
Mediant 600
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Management
Status & Diagnostics
Scenarios
Search
Basic
Full
Network Settings
Media Settings
PSTN Settings
Security Settings
Protocol Configuration
Protocol Definition
SIP Advanced Parameters
Advanced Parameters
Supplementary Services
Metering Tones
Charge Codes
Keypad Features
Stand-Alone Survivability
Manipulation Tables
Routing Tables
Profile Definitions
Endpoint Settings
Trunk/IP Group
Digital Gateway
Advanced Applications
TDM Configuration
Advanced Parameters
Basic Parameter List
Silence Detection Method None
Enable Fax Re-Routing Disable
CDR and Debug
CDR Server IP Address
CDR Report Level None
Debug Level 5
Misc. Parameters
Progress Indicator to IP Not Configured
Enable X-Channel Header Disable
Enable Busy Out Disable
Default Release Cause 3
Max Number of Active Calls 8
Max Call Duration [min] 0
Enable LAN Watchdog Disable
Enable Calls Cut Through Disable
Enable User-Information Usage Disable
Out-Of-Service Behavior ! Reorder Tone
Delay After Reset [sec] 7
Emergency Calls
Emergency Numbers
Emergency Calls Regret Timeout 10
Submit
Internet
100%

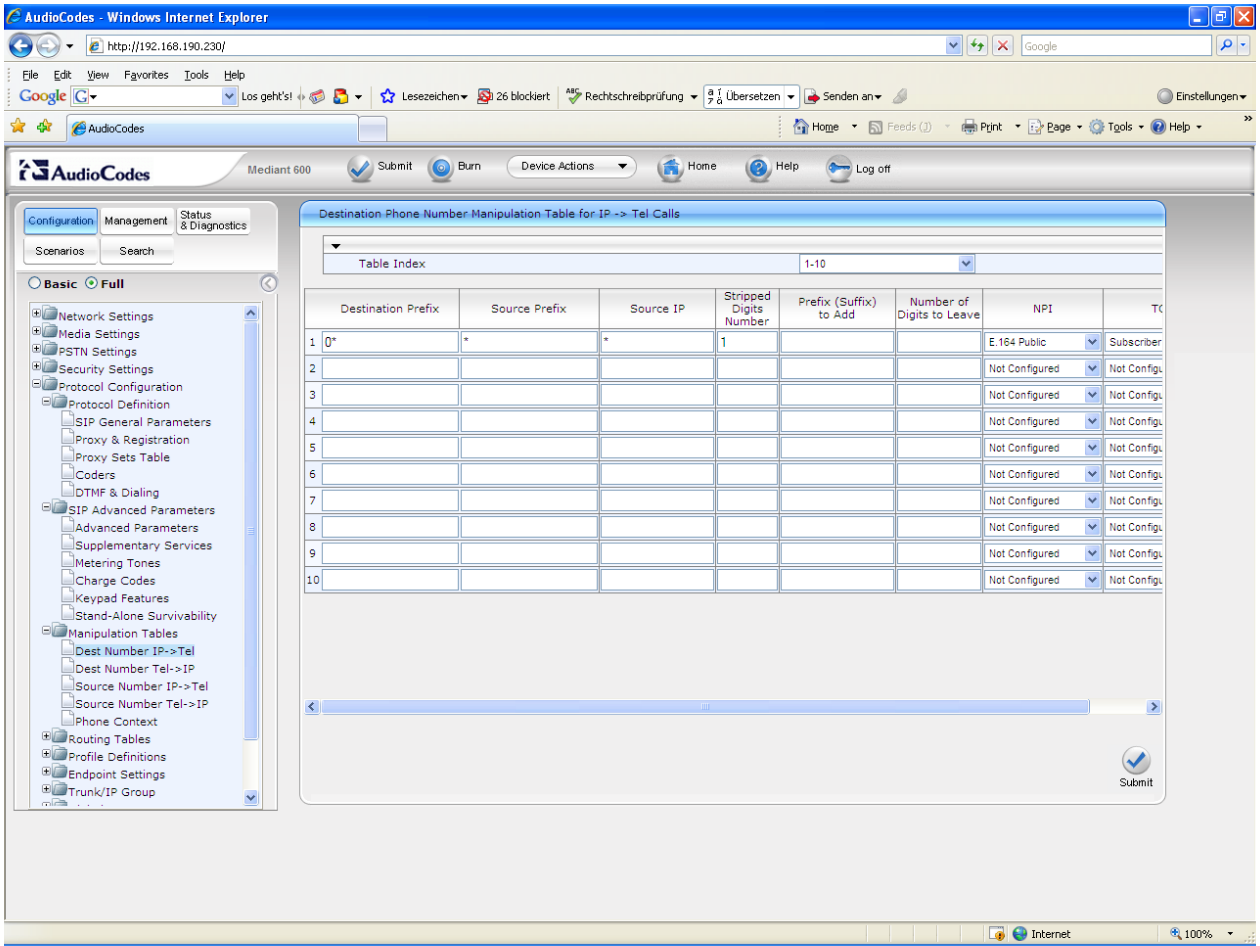
6.10 Navigate to Protocol Configuration – SIP Advanced Parameters – Stand-Alone Survivability to set the parameters of the second SIP proxy on the Mediant-600 gateway that is used in survivable mode.

- For the Enable SAS field, select Enable.
- Set the SAS Local SIP UDP Port to 5080.
- Set the SAS Default Gateway IP to the IP address of your Mediant-600.
- Set the SAS Local SIP TCP Port to 5080.
- Set the SAS Local SIP TLS Port to 5081.



6.11 Navigate to Protocol Configuration – Manipulation Tables – Dest Number IP → Tel.

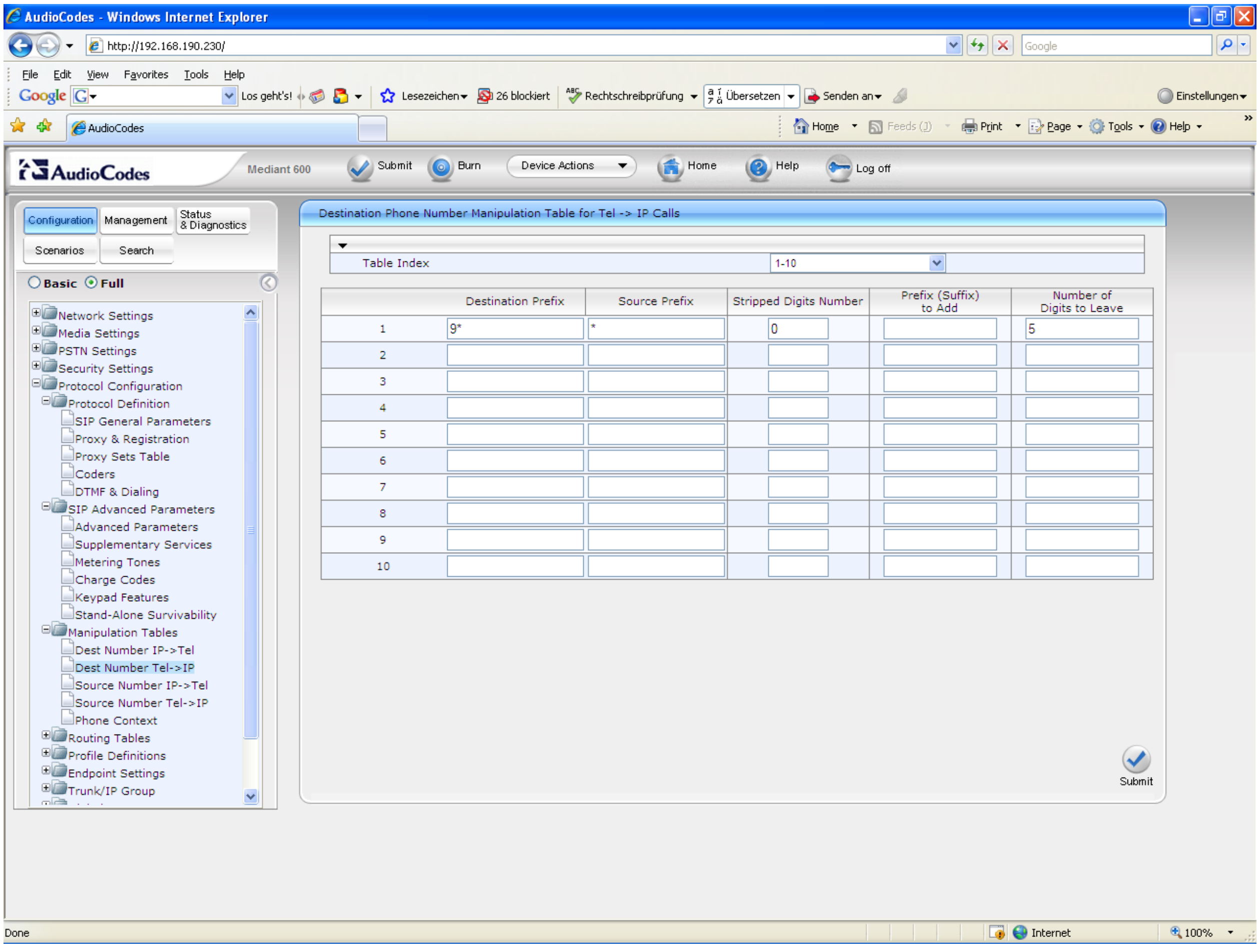
- Add an entry for an outgoing call via BRI in survivable mode.
- Set NPI (numbering plan) to E.164 Public.
- Set TON (type of number) to Subscriber.



In normal mode we need a 0 prefix for outgoing call via PRI. In survivable mode we don't need the 0 prefix. This entry will strip the 0 prefix (i.e. Stripped Digits Number = 1), so that the end user can dial the same number in normal and in fail-over mode.

6.12 Navigate to Protocol Configuration – Manipulation Tables – Dest Number Tel → IP.

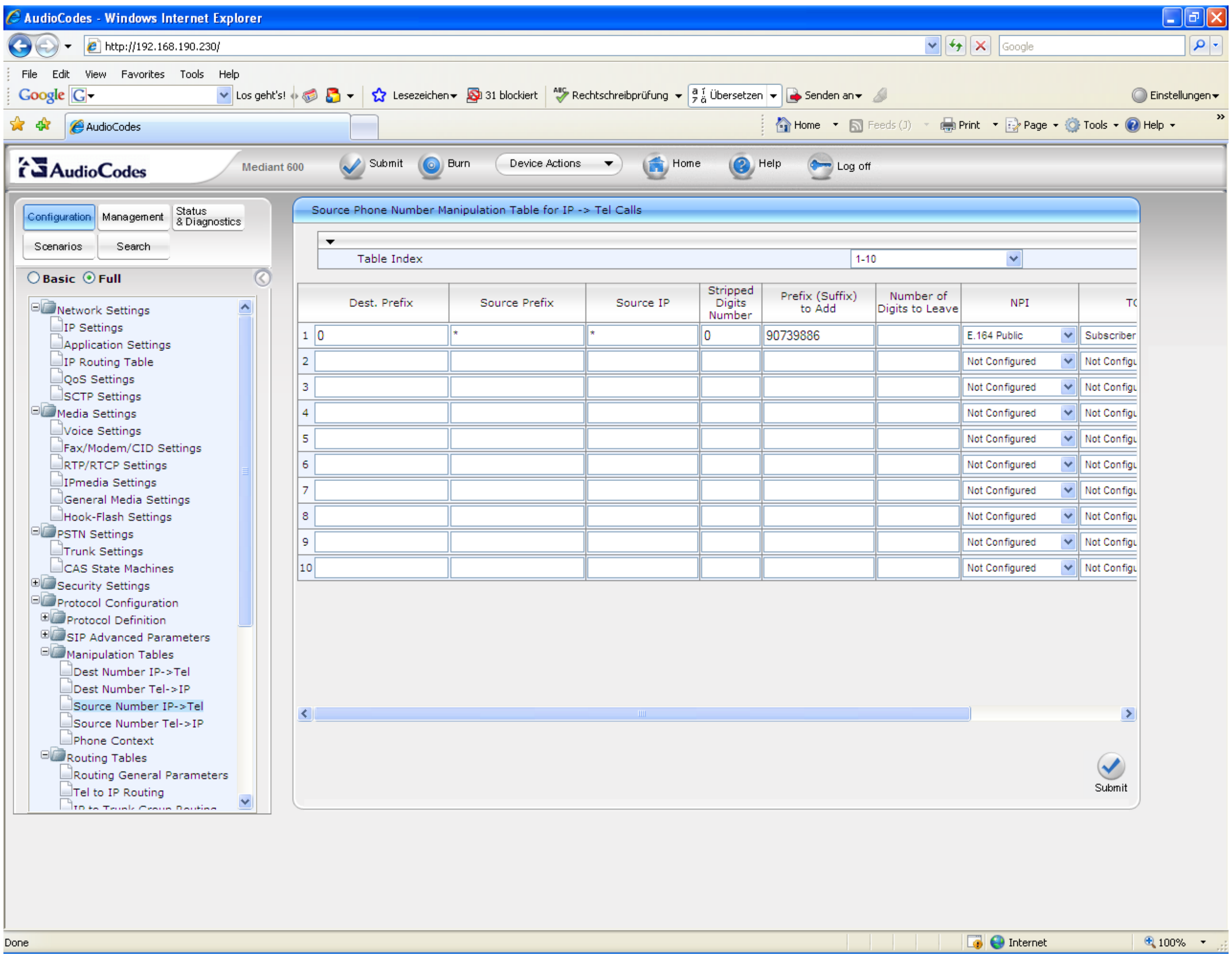
- Add an entry for an incoming call via BRI in survivable mode.



Add an entry for each incoming number to your Mediant-600 gateway that you wish to modify before routing. This is similar to the Incoming Call Handling Treatment Table on Avaya Communication Manager. The example above shows that for an inbound call to the Mediant-600 that has the called party number beginning with nine, only the last five digits will be used for routing to the local FXS station.

6.13 Navigate to Protocol Configuration – Manipulation Tables – Source number IP → Tel.

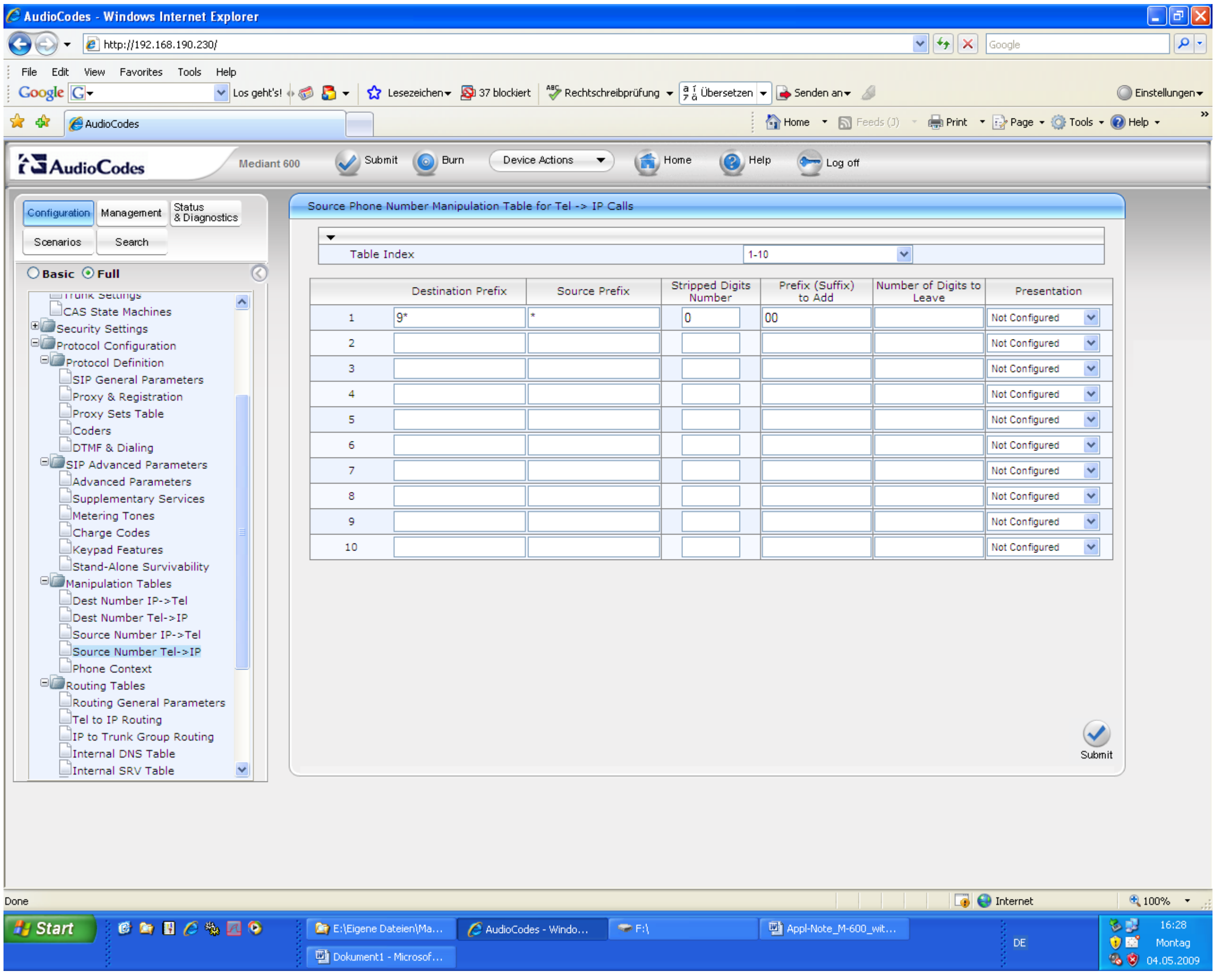
- Add an entry for an outgoing call via BRI in survivable mode.
- Set NPI (numbering plan) to E.164 Public.
- Set TON (type of number) to Subscriber.
- Set Presentation to not configured (not shown). This means that the phone can enable the CLIP/CLIR function via the keypad feature. If this field is set to allowed or restricted, the keypad function doesn't work.



This entry will add the 90739886 prefix to the calling party number and will set the numbering plan to E.164 public and the type of number to Subscriber, if the called party begins with 0. This causes the called party to display the correct number.

6.14 Navigate to Protocol Configuration – Manipulation Tables – Source number Tel → IP.

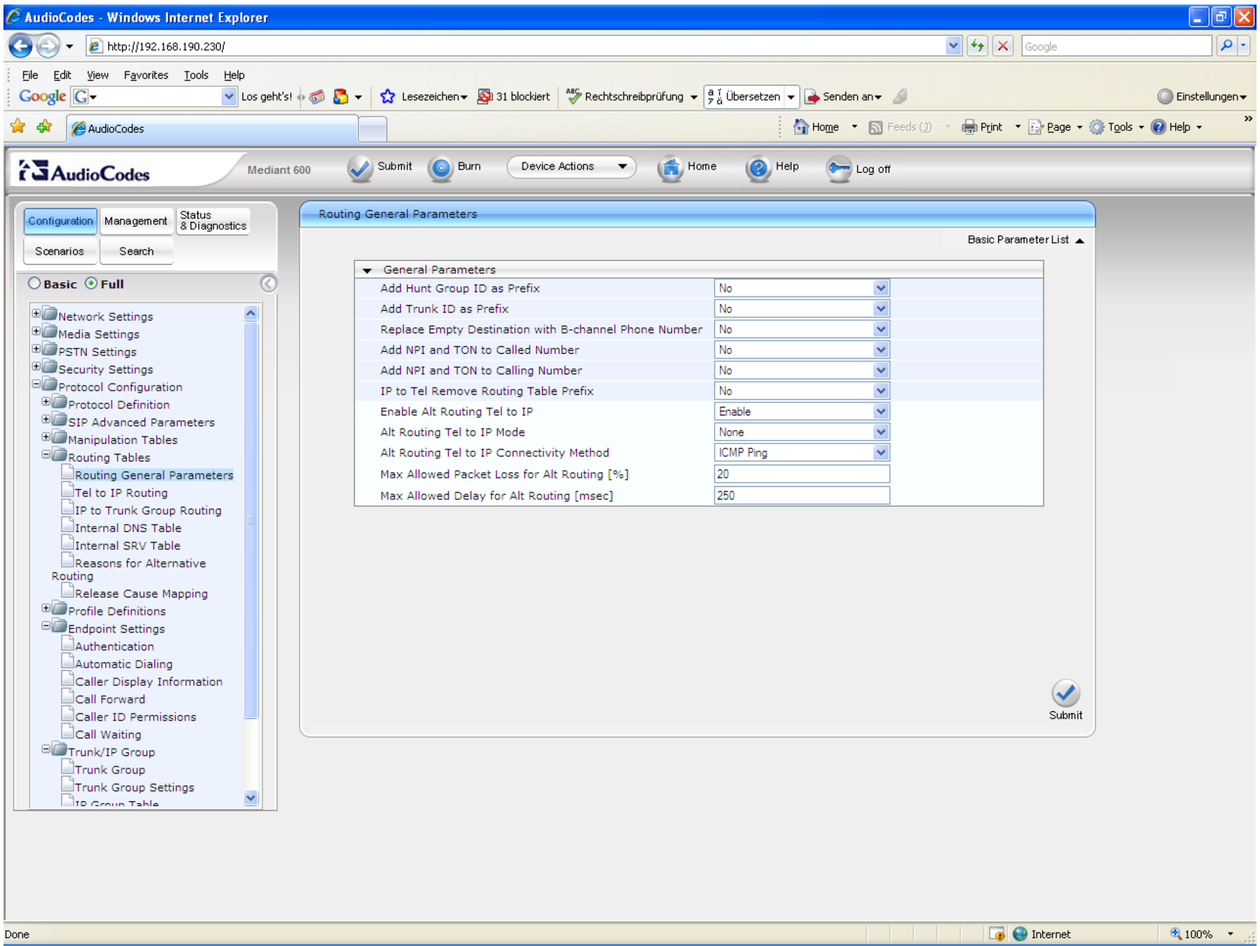
- Add an entry for an incoming call via BRI in survivable mode.



This entry will add the 00 prefix to the calling party number. This effectuates a correct entry in the call history list, so that a recall is possible from the call history list.

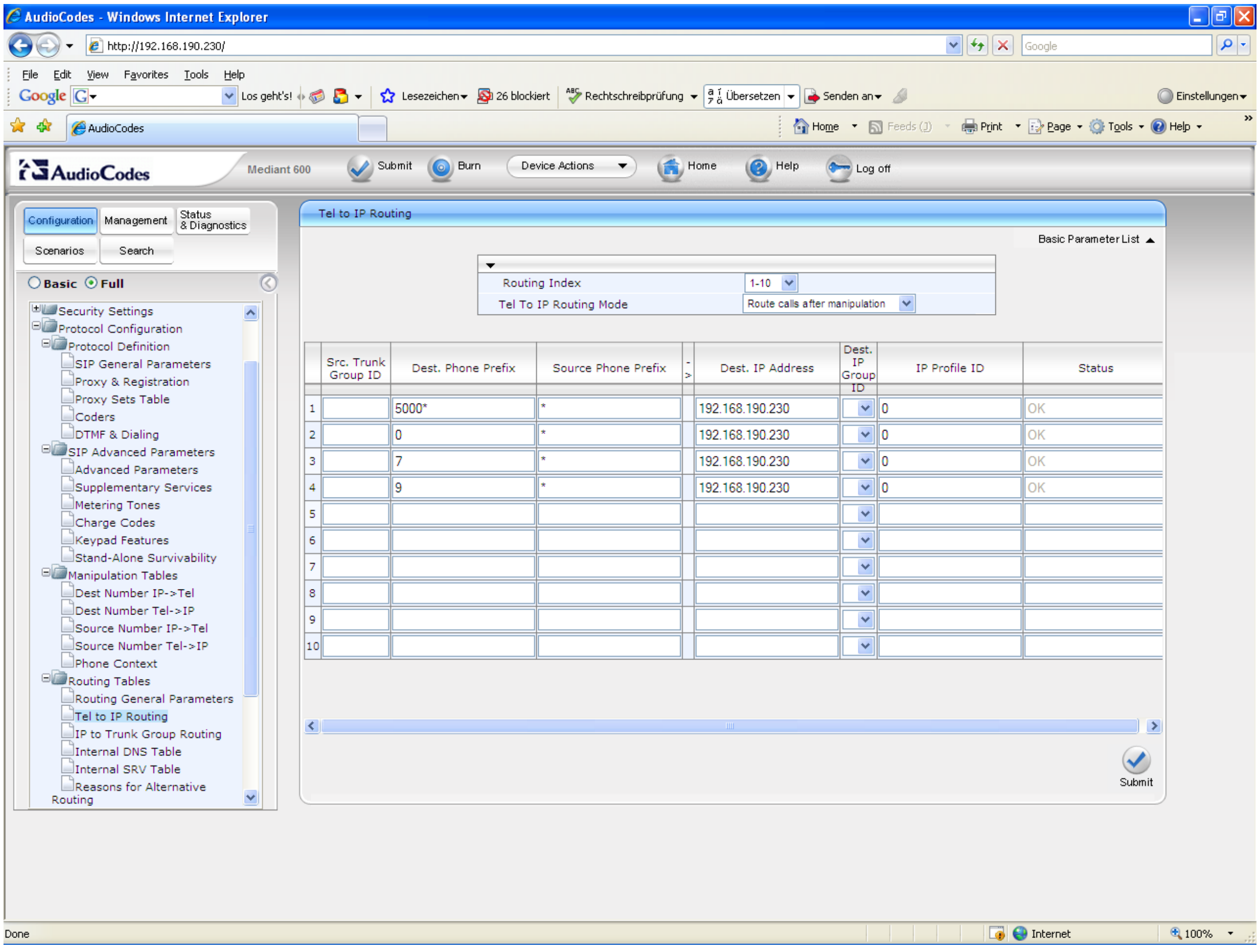
6.15 Navigate to Protocol Configuration – Routing Tables – Routing General Parameters.

- Set all the options as shown below.



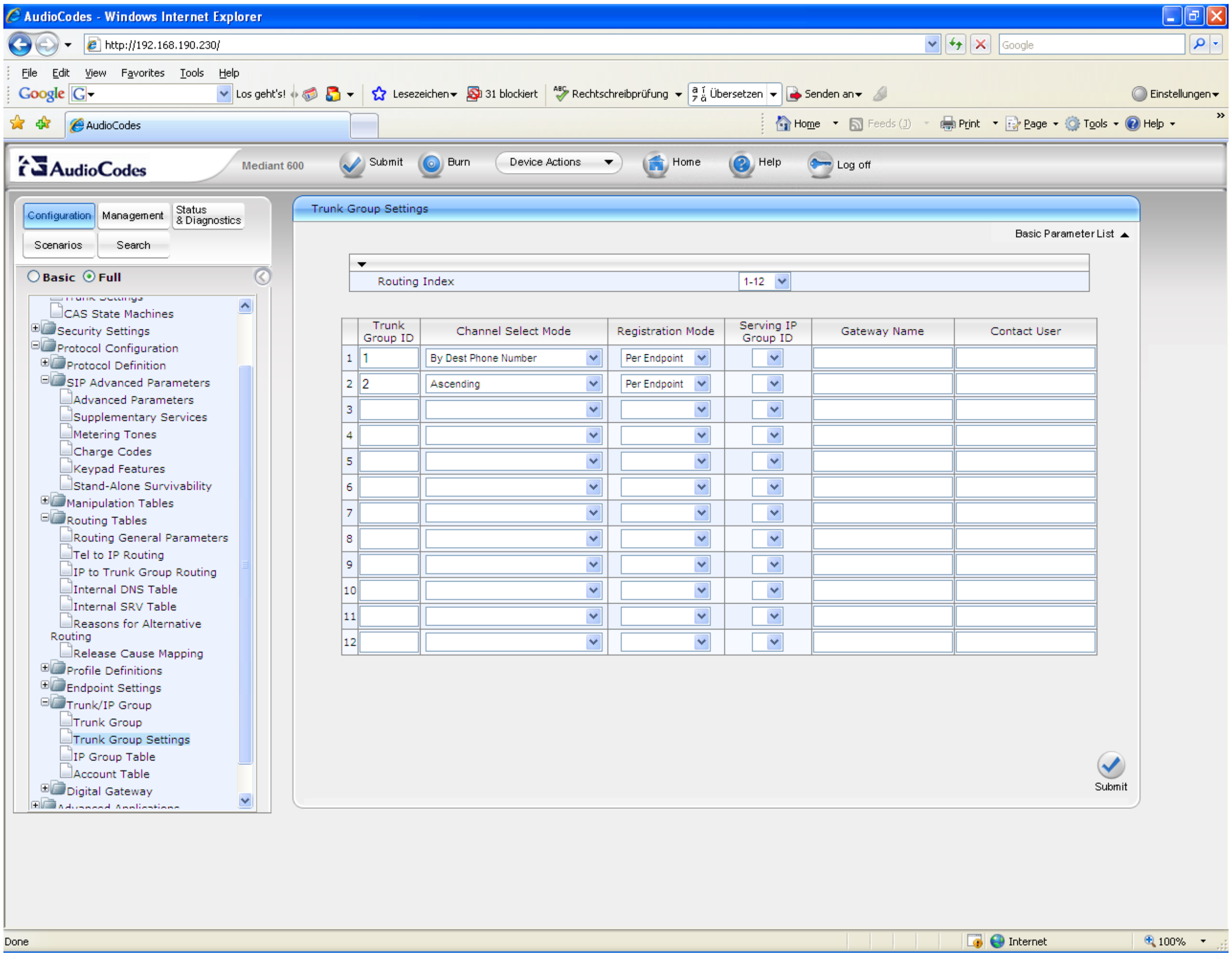
6.16 Navigate to Protocol Configuration – Routing Tables – Tel to IP Routing.

- Set the Tel to IP Routing table as shown, with the Dest. IP Address field set to the IP Address of your Mediant-600 gateway.
- The IP Profile ID is set to 0.
- The Dest. Phone Prefix field shows the extensions or the prefixes of the extensions existing in the Avaya Communication Manager.



6.17 Navigate to Protocol Configuration – Trunk/IP Group – Trunk Group Settings.

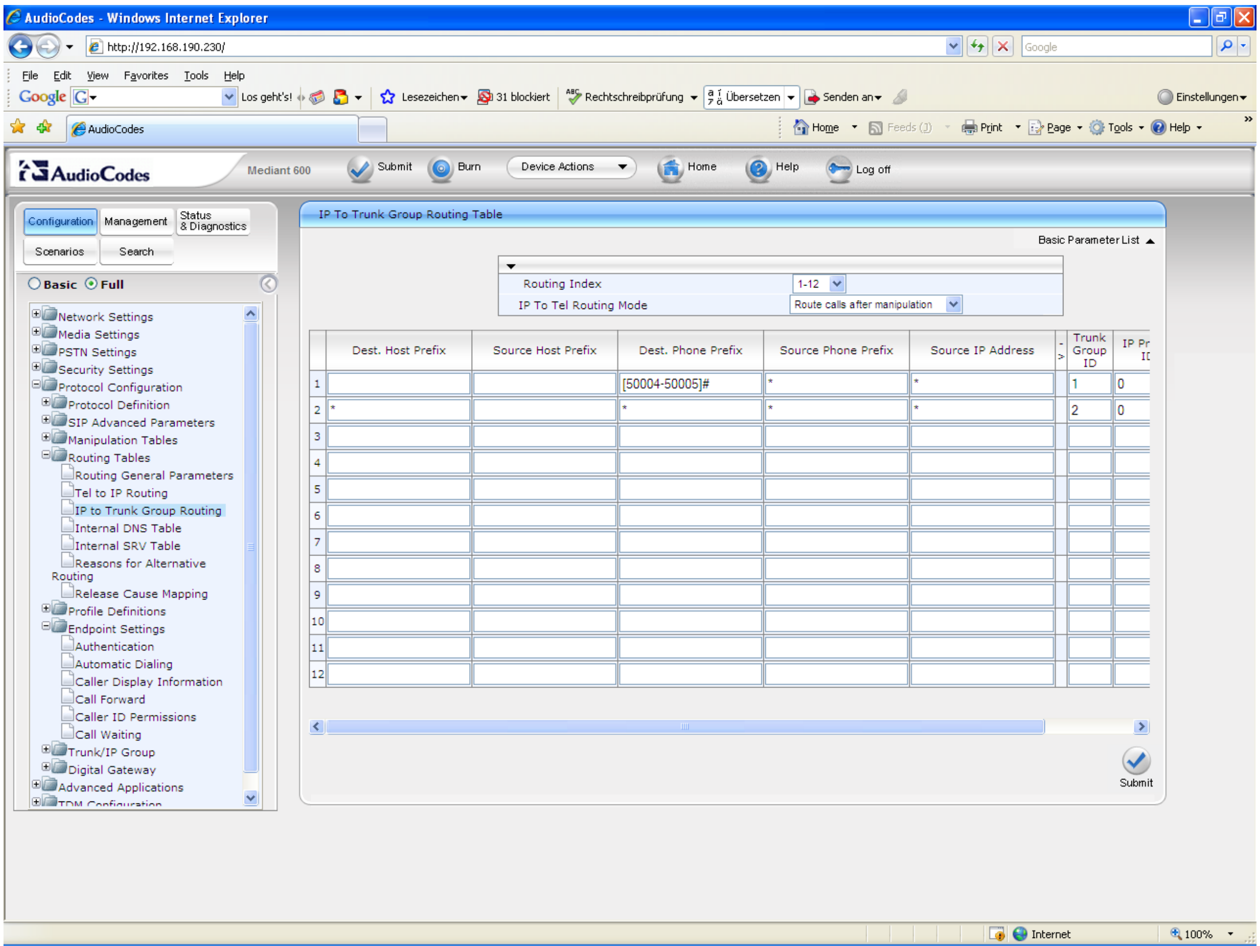
- Add two entries as shown.
- The first entry will route calls to endpoints.
- The second entry will cycle calls to the BRI trunks.
- For endpoints and BRI numbers registration, the Registration Mode field has to be set to Per Endpoint.



6.18 Go back to the IP to Trunk Group Routing Form under Protocol Configuration – Routing Tables.

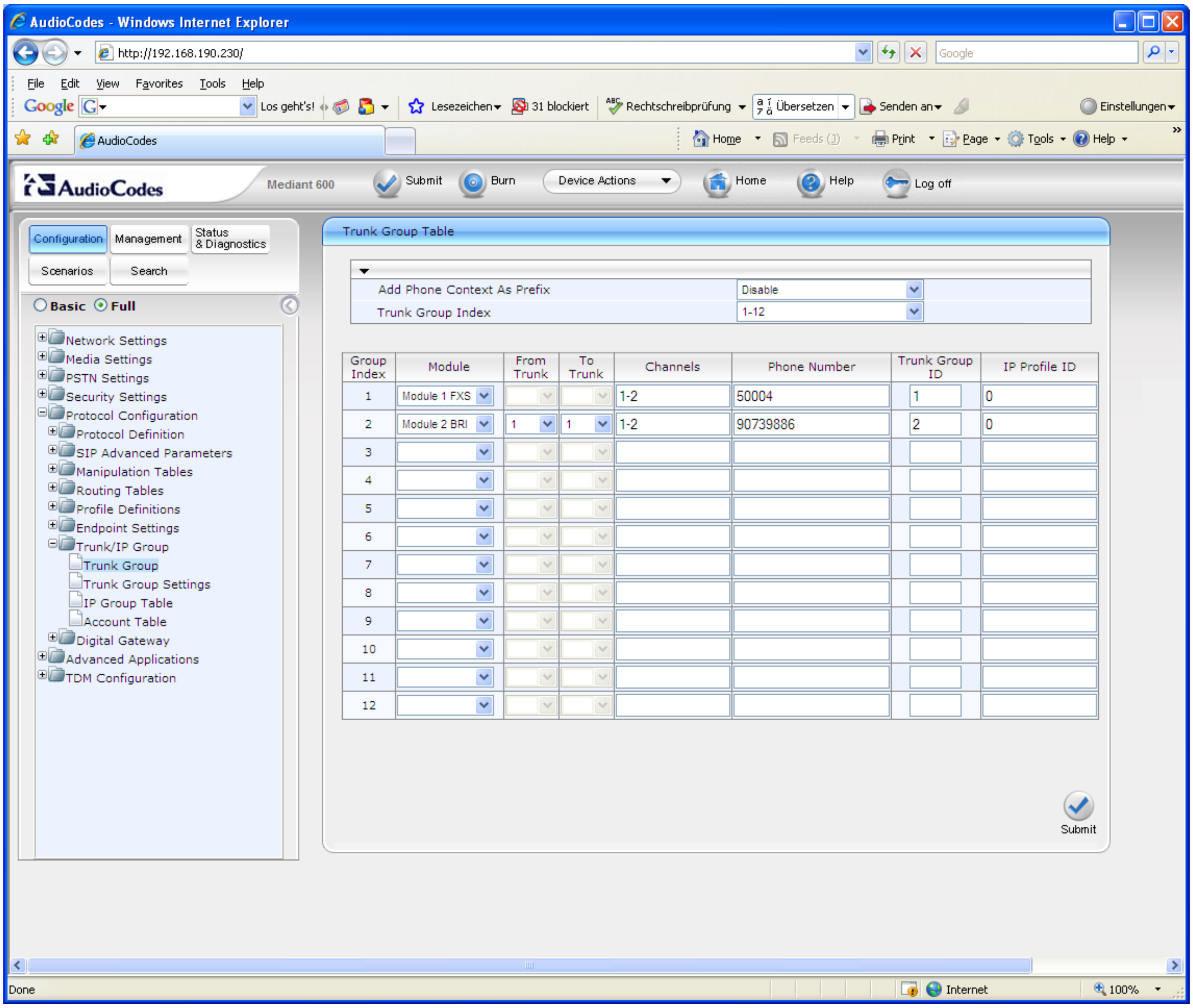
- Add an entry for local extensions in the Dest. Phone Prefix field and set it to Trunk Group 1.
- Add an entry for remote extensions and set them to Trunk Group 2.
- Source Phone Prefix and Source IP Address should be set to *.

All fields not shown above should be left blank.



6.19 Navigate to Protocol Configuration – Trunk/IP Group – Trunk Group. Each number specified in the Channels field represents a port of the Mediant-600 gateway.

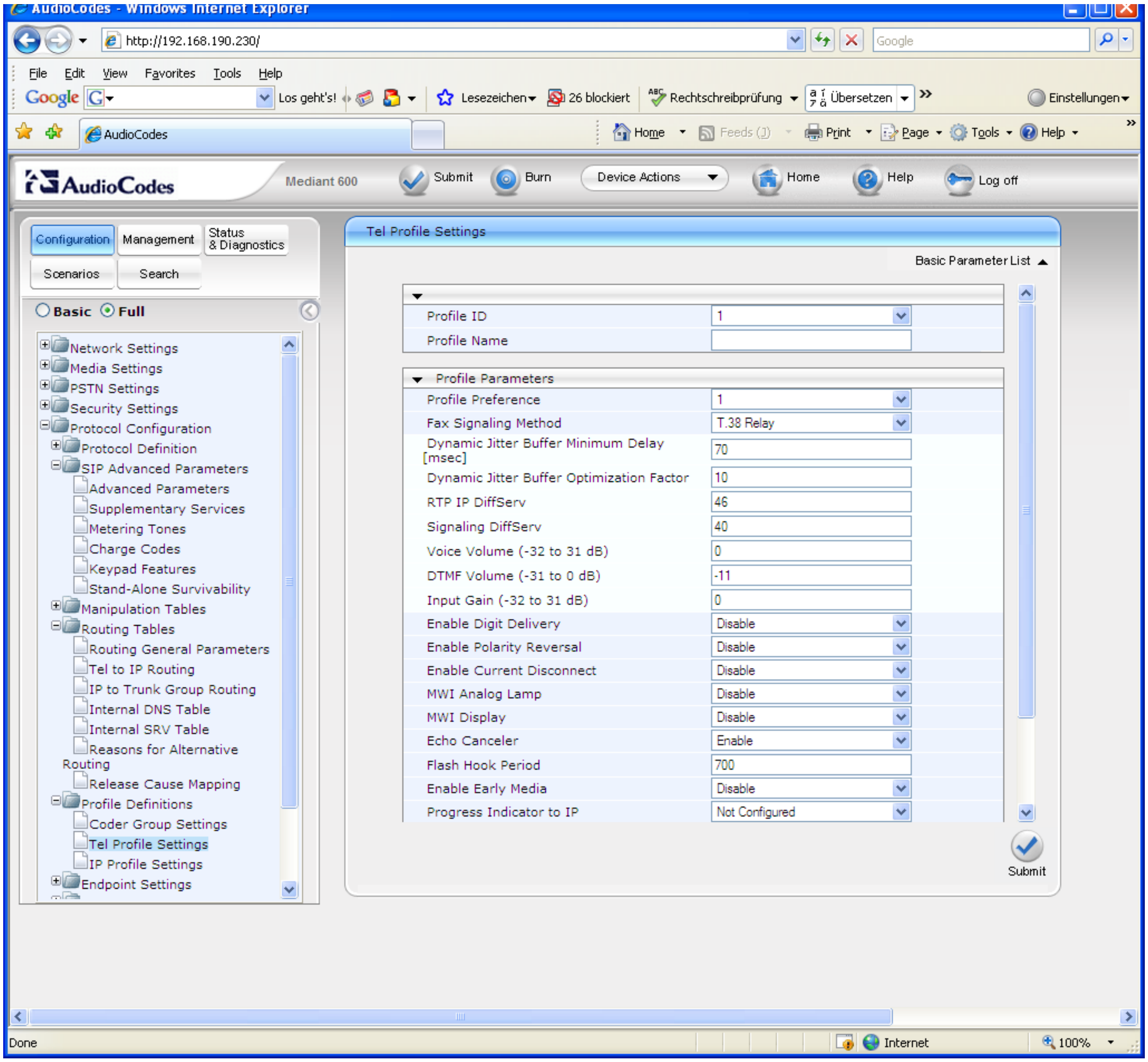
- Add two entries as shown.
- The first entry will assign Trunk Group 1 and a Phone Number to the FXS endpoints. This entry is necessary for incoming calls to the FXS stations.
- The second entry assigned Trunk Group 2 to the BRI trunks.



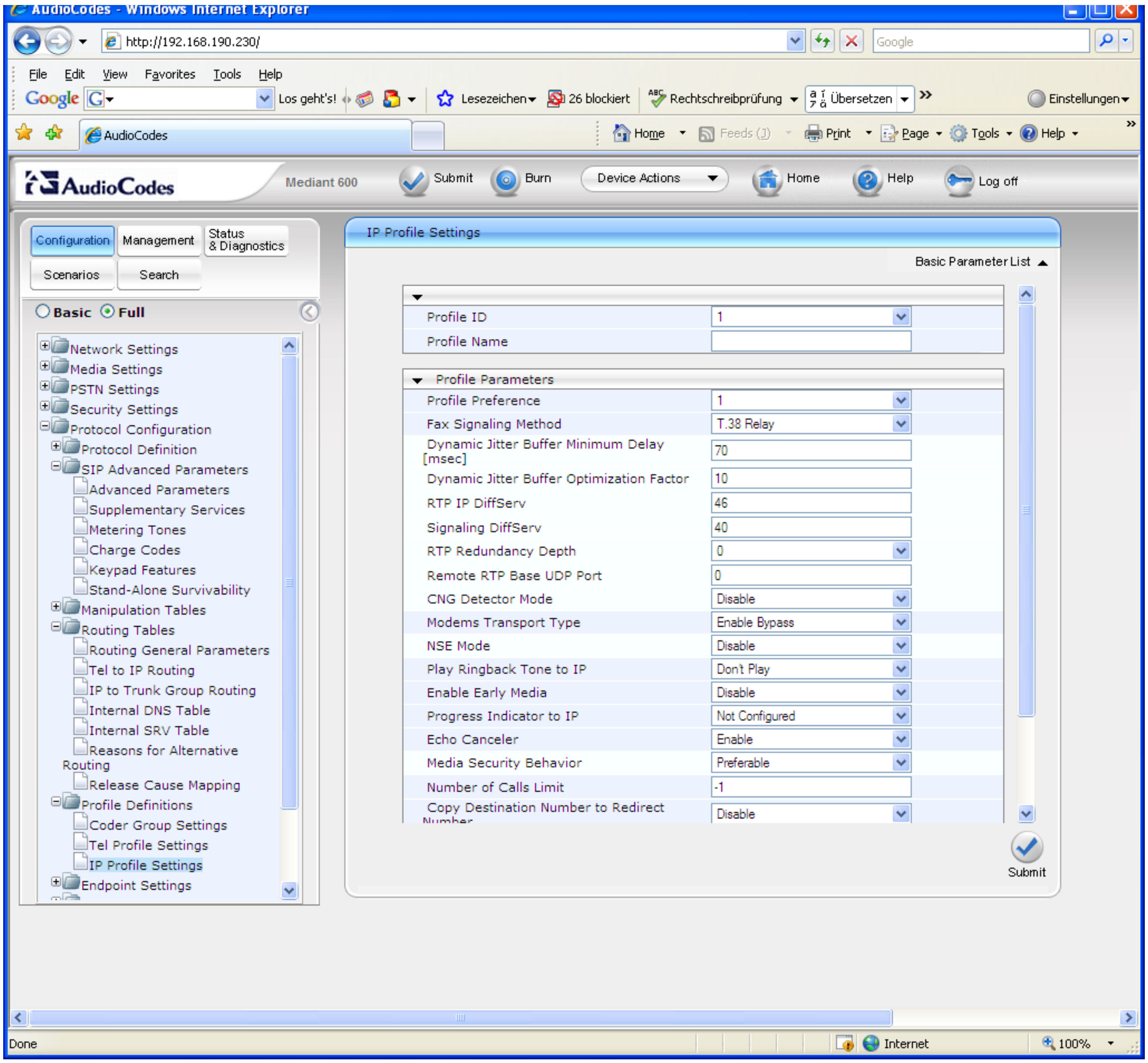
6.20 Navigate to Protocol Configuration – Profile Definitions – Tel Profile Settings / IP Profile Settings

- On both Tel Profile Settings and IP Profile Settings screens, set the Dynamic Jitter Buffer Minimum Delay to 70ms.
- On the IP Profile Settings screen set the Enable Early Media to Disable.

Tel Profile Settings

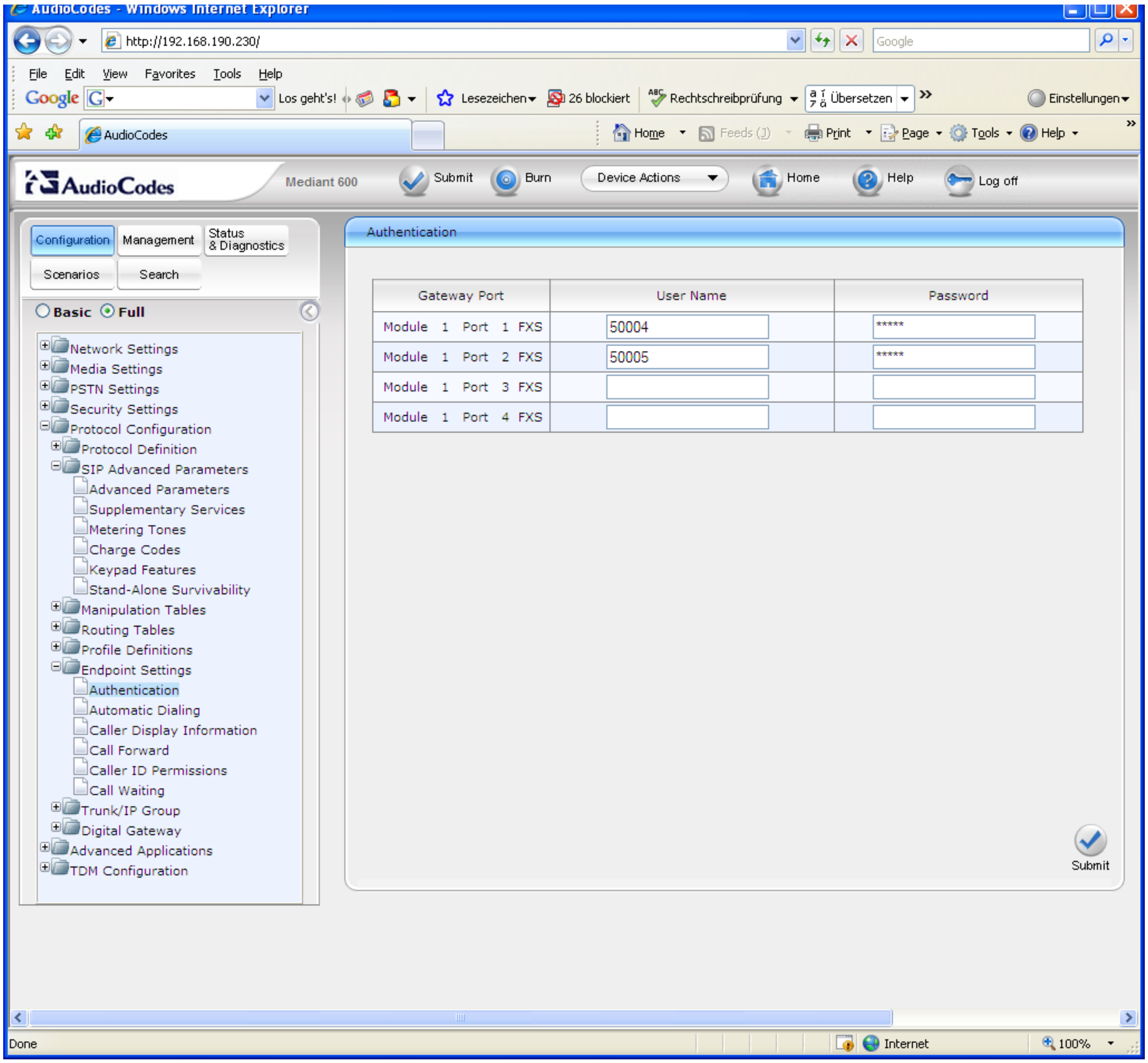


IP Profile Settings



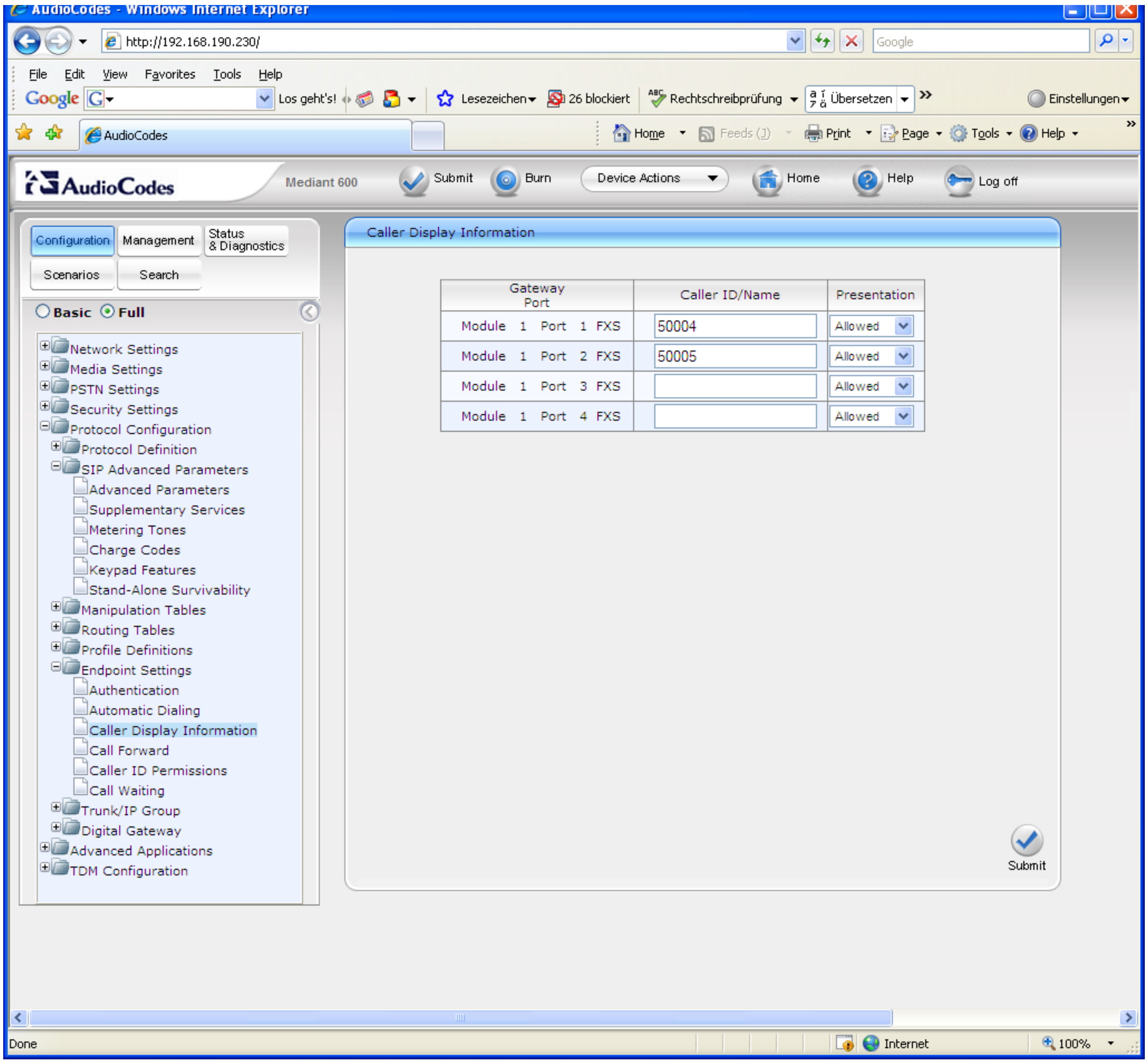
6.21 Navigate to Protocol Configuration – Endpoint Settings – Authentication.

- Assign each FXS port an User Name (extension) and Password that matches the user account setup. These will be used to register each port on the SES Home / Home-Edge.



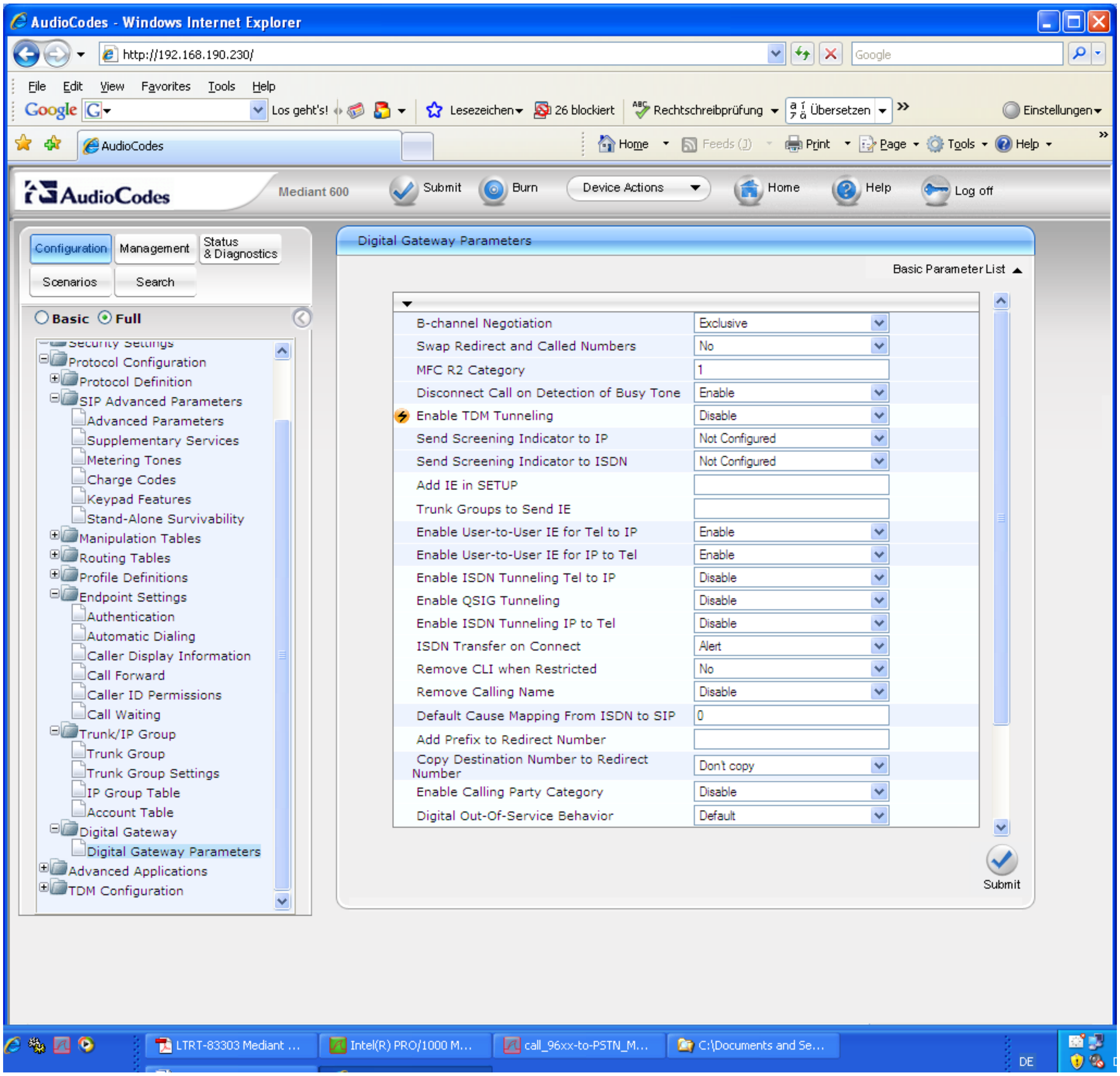
6.22 Navigate to Protocol Configuration – Endpoint Settings - Caller Display Information.

- Assign a Caller-ID to each associated port and set the Caller-ID permissions.



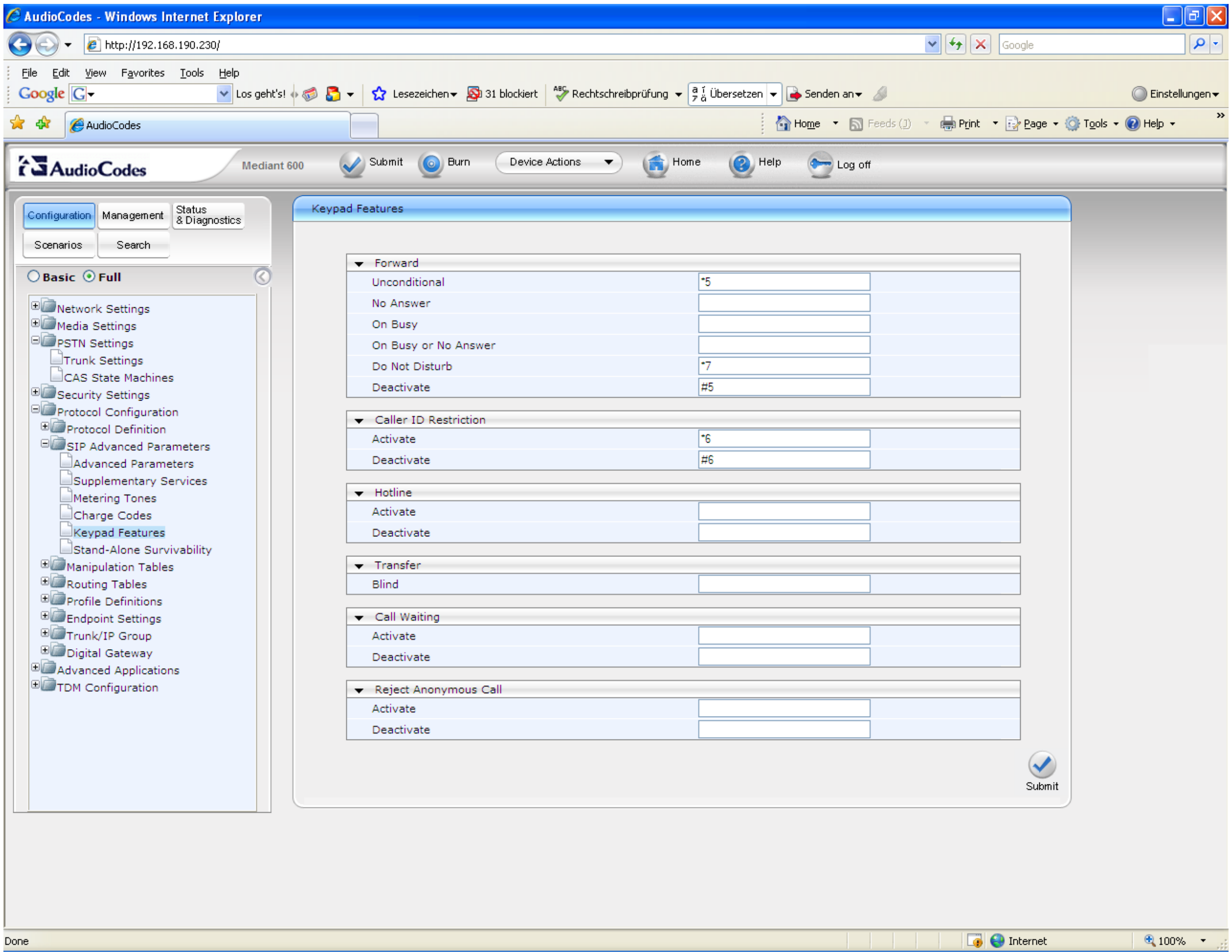
6.23 Navigate to Protocol Configuration – Digital Gateway - Digital Gateway Parameters.

- Set all the options as shown below.



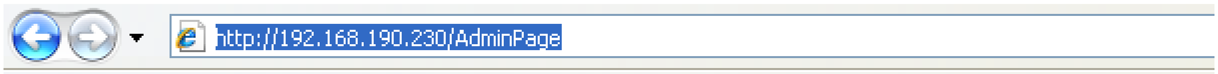
6.24 Navigate to Protocol Configuration – SIP Advanced Parameters – Keypad Features. The keypad features page enables you to activate/deactivate the following features directly from the connected phone's keypad.

- Define dial strings for features. The screen below shows examples for activation/deactivation.

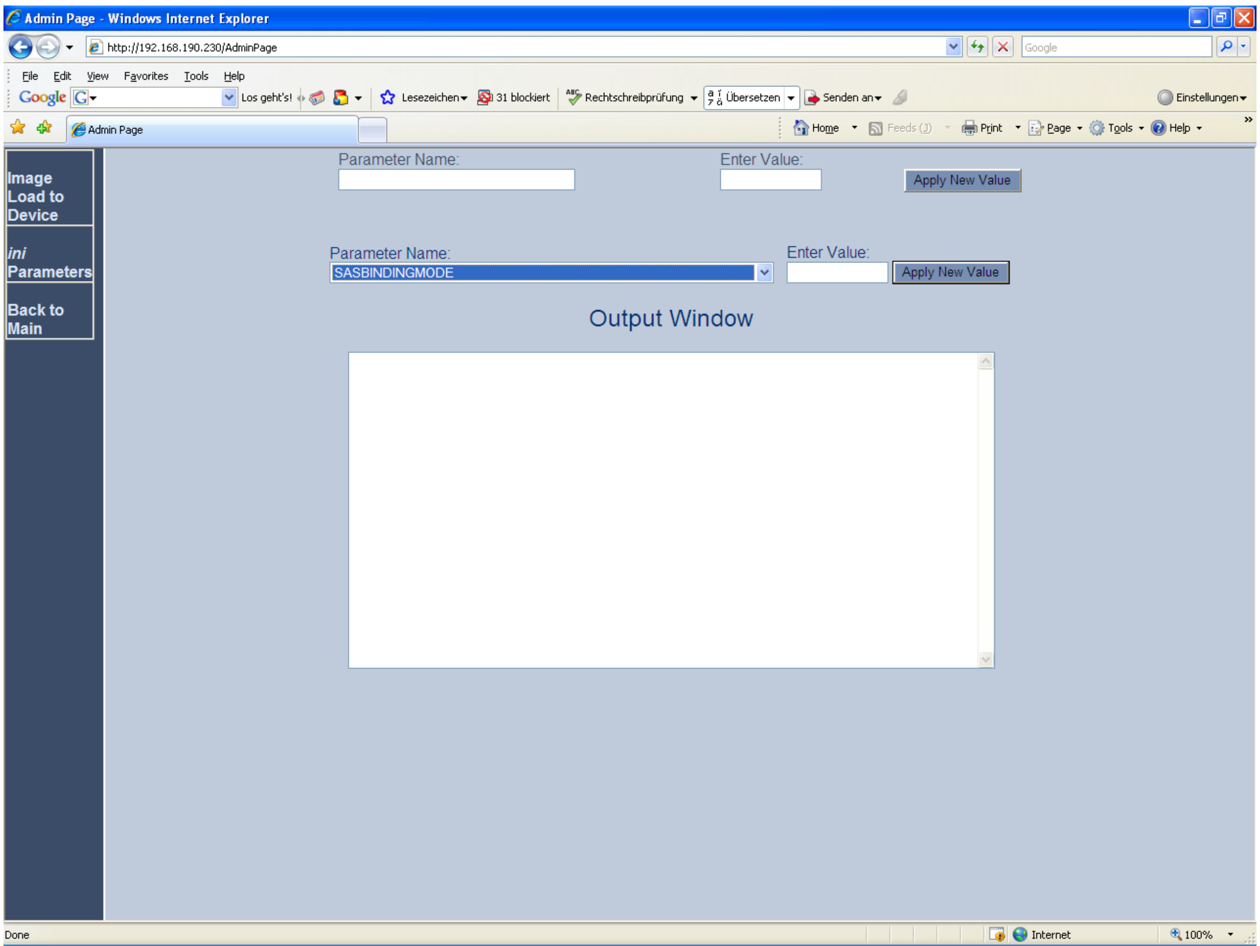


6.25 Additional Settings.

Goto http://<address of M-600>/AdminPage.



- Select 'ini Parameters' link from the left pane.



- Set SASSURVIVABILITYMODE to Value 2.
- Set RELIABLECONNECTIONPERSISTENTMODE to Value 1.
- Set CURRENTDISCONNECTDURATION to Value 600.
- Set SASBINDINGMODE to Value 1.

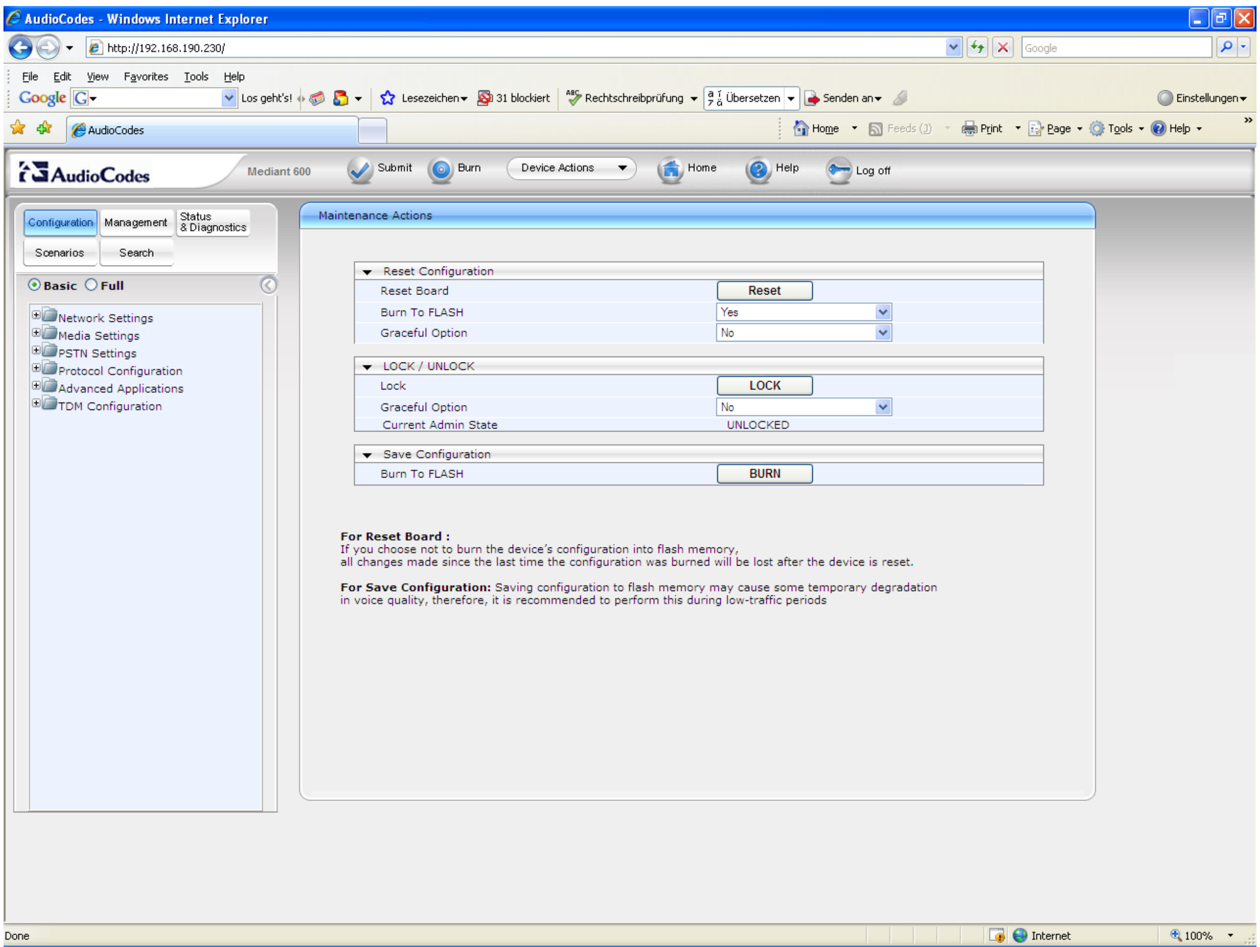
Click the Apply new value button to save changes.

6.26 Click the Burn button to make your changes permanent.

6.27 Reset the Gateway to effect all of your changes.

- Open the Device Actions pull down menu in the AudioCodes menu bar.
- Select Reset

The Reset screen is shown as below.



- Press Reset button to reset the Gateway.

7. 96xx SIP Phone Administration

Edit the 46xxsettings.txt file on your File Server. Add or edit the entries shown below. The SIP_CONTROLLER_LIST contains a comma delimited list of SIP servers that your endpoints register to. Add your SES Home / Home-Edge followed by your M-600 Gateway. Set the SIP Domain to that of you SES and the other values as shown.

- SET SIP_CONTROLLER_LIST "192.168.193.27:5060;transport=udp,192.168.190.230:5080;transport=tcp"
- SET SIPDOMAIN "interop.com"
- SET DISCOVER_AVAYA_ENVIRONMENT "1"
- SET FAILBACK_POLICY "auto"

Alteratively you can access the administration menu of your 96xx SIP endpoint by pressing the MUTE button followed by CRAFT#.

- Scroll down and select the SIP menu option.
- Select the "SIP Proxy Settings".
  The SIP server that the endpoint is currently registered will show a "√".
- Select a Proxy server. The IP address, transport type and port of the SIP Proxy server will be displayed.
  - To change the SIP Proxy Server press the "Bksp button" and enter the new IP-address.
  - To change the Transport Type scroll down to Transport Type and press the "Change button".
  - To change the SIP Port scroll down to SIP Port, press the "Bksp button" and enter the new port.
  - To save all the changes, press the "Save button".
- From the SIP menu, select the "SIP Global Settings" menu.
- The "SIP Global Settings" menu will show you the other values inherited from the 46xxsettings.txt file.
  - Select the item which you will change.
  - You can change the value by pressing the "Bksp button" and enter the new name or by pressing the "Change button".
  - To save the changes, press the "Save button".

When the endpoint fails over, it will show an "Aquiring Service" message. Once the endpoint has registered to the Mediant-600 in fail-over mode, a triangle with an apostrophe will be present.

8. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify AudioCodes Mediant-600 with Avaya SIP Enablement Services and Avaya Communication Manager. This section covers the general test approach and the test results.

8.1 General Test Approach

The following features and functionality were covered during interoperability compliance test:

- **Connectivity / Failover**
  Test cases focus on transitions of the 96xx series phones and AudioCodes Gateway to/from normal mode and survivability mode.

- **Normal Mode**
  Test cases focus on endpoint to endpoint call flows and feature invocation when the branch connectivity is in Normal Mode. Features tested include: Hold/Resume, Conference Add/Drop, Call Transfer – Attended/Un-attended, Call Waiting and special ISDN features.
    - All call routing is controlled by the centralized SES/CM.
    - CM is configured with IP-IP Direct Audio enabled.
    - All PSTN inbound/outbound calls are routed to a centralized G350 media gateway.
    - All branch 96xx and AudioCodes FXS stations are registered to the centralized SES.
    - Inbound calls via branch side aren't tested. In order to support it a route must be added in the CM/SES. This wasn't a component of the test environment.

- **Survivability Mode**
  Test cases focus on Centralized Trunking endpoint to endpoint call flows and feature invocation when the branch looses WAN connectivity and is in Survivability Mode. Features tested include: Hold/Resume, Conference Add/Drop, Call Transfer – Attended/Un-attended, Call Waiting and special ISDN features.
    - All branch 96xx phones are transitioned and registered to the AudioCodes Gateway SAS server.
    - All call routing is controlled by the local branch AudioCodes Gateway.
    - All PSTN outbound calls are routed to the AudioCodes Gateway BRI port.
    - PSTN inbound calls to branch endpoints via the local branch AudioCodes Gateway BRI interface were not tested.

8.2 Test Results

Interoperability testing of the sample configuration was completed with successful results.
The following observations were noted:

Survivable Mode

- When an external outbound call is placed, the display of the calling party (96xx) will switch the dialled number to the registered phone number which is programmed in trunk group when the called party is ringing.

- The keypad features are not applicable to 96xx phones. Thus 96xx endpoint can't use such features (e.g. call forwarding, don't disturb). To use CLIR it is only possible by changing the settings via a Web browser.

9. Verification Steps

The following steps may be used to verify the configuration:

- From the Avaya Communication Manager, use the status signalling-group command to verify that the SIP signalling group is in-service.
- From the Avaya Communication Manager, use the status trunk-group command to verify that the SIP trunk group is in-service.
- From the Avaya SES web administration interface, verify that all endpoints behind the Mediant-600 are registered with the Avaya SES.
- Verify that calls can be placed to/from analog and SIP endpoints behind the Mediant-600 and the Avaya Communication manager.
- Verify that calls can be placed to/from analog and SIP endpoints behind the Mediant-600 and the PSTN.
- Verify that calls can be placed from analog and SIP endpoints behind the Mediant-600 when a data WAN failure is introduced.

10. Conclusion

These Application Notes describe the procedures required to configure the AudioCodes Mediant-600 BRI VoIP Gateway to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager. The AudioCodes Mediant-600 successfully passed compliance testing with the observations documented in Section 8.

11. Additional References

- Feature Description and Implementation For Avaya Communication Manager, Doc # 555-245-205, Issue 6.0, January 2008.
- Administrator Guide for Avaya Communication Manager, Doc # 03-300509, Issue 4.0, January 2008
- Avaya Communication Manager Basic Administration Quick Reference, Doc # 03-300363, Issue 4.0, January 2008
- Avaya Communication Manager Advanced Administration Quick Reference, Doc # 03-300364, Issue 4.0, January 2008
- Administering SIP Enablement Services on the Avaya S8300 Server, Doc# 03-602508, Issue 1.0, January 2008
- Document LTRT-83303 Mediant 1000 and Mediant 600 SIP User's Manual Version 5.4, May 2008
- AudioCodesMP118.PDF Application Notes for Configuring the AudioCodes MP-118 Analog VoIP Gateway with Avaya SIP Enablement Services and Avaya Communication Manager, Issue 1.0, November 2007.
- Document Solution Validation Test Plan and Results for AudioCodes MP-114/118 SIP Gateway with ACM 5.1 and SES 5.1. Supporting the Avaya Communication Manager with Survivable SIP Gateway Solution, Issue 0.5, November 2008

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for AudioCodes Mediant-600 BRI VoIP Gateway products may be found at http://www.audiocodes.com.

12. APPENDIX A: Mediant-600 Ini-File

This section displays the Mediant-600 configuration.

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

;Board: Mediant 600
;Serial Number: 1490018
;Slot Number: 1
;Software Version: 5.40A.033
;DSP Software Version: 204IM => 540.17
;Board IP Address: 192.168.190.230
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 192.168.190.254
;Ram size: 128M   Flash size: 32M
;Num DSPs: 3  Num DSP channels: 12
;Profile: NONE
;Key features:;Board Type: Mediant 600;Channel Type: RTP PCI DspCh=12
;E1Trunks=0;T1Trunks=0;PSTN Protocols: ISDN CAS ;Coders: G723 G729 GSM-FR
G727 ;DSP Voice features: EC128mSec ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;Control Protocols: SIP SASurvivability
;Default features:;Coders: G711 G726;

;------- Mediant-1000 HW components-------
;
; Slot # : Module type : # of ports
;----------------------------------------
;     1 : FXS        : 4
;     2 : BRI        : 4
;----------------------------------------


[SYSTEM Params]

SyslogServerIP = 192.168.190.150
EnableSyslog = 1
VXMLFIleName = ''

[BSP Params]
```

PCMLawSelect = 1
TDMBusClockSource = 4
PREMIUMSERVICECLASSCONTROLDIFFSERV = 46
StorageServerNetworkAddress = 255.255.255.255

[ATM Params]

[Analog Params]

PolarityReversalType = 1
MinFlashHookTime = 100
FXSLoopCharacteristicsFilename = 'M1K13-1-fxs16khz.dat'
FXOLoopCharacteristicsFilename = 'M1K12-1-16khz-fxo.dat'
CurrentDisconnectDuration = 600

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]

[MEGACO Params]

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

[PSTN Params]

TraceLevel_0 = 2
TraceLevel_1 = 0
TraceLevel_2 = 0
TraceLevel_3 = 0
TDMBusPSTNAutoClockEnable = 1
ProtocolType_0 = 50
ProtocolType_1 = 0
ProtocolType_2 = 0
ProtocolType_3 = 0
ClockMaster = 0
TerminationSide = 0
FramingMethod = 0

LineCode = 0
LineBuildOut.LOSS = 0
LineBuildOut.OVERWRITE = 0
LineBuildOut.XPM0 = 0
LineBuildOut.XPM1 = 0
LineBuildOut.XPM2 = 0
DCHConfig = 0
ISDNIBehavior = 0
ISDNInCallsBehavior = 0
ISDNOutCallsBehavior_0 = 0
ISDNOutCallsBehavior_1 = 1024
ISDNOutCallsBehavior_2 = 1024
ISDNOutCallsBehavior_3 = 1024
ISDNGeneralCCBehavior = 0
ISDNNFASInterfaceID = 255
IUAInterfaceID = 4294967295
NFASGroupNumber = 0
AutoClockTrunkPriority = 0
DPNSSBehavior = 12
CasTrunkDialPlanName = ''
BriLayer2Mode = 0

[SS7 Params]

[Voice Engine Params]

VoicePromptsFileName = ''
BrokenConnectionEventTimeout = 3
CallerIDType = 1
FarEndDisconnectSilenceMethod = 0
CallProgressTonesFilename = 'usa_tones_12.dat'

[WEB Params]

HTTPSCipherString = 'RC4:EXP'

[SIP Params]

ENABLECALLERID = 1
MAXDIGITS = 15
REGISTRATIONTIME = 3600
ISPROXYUSED = 1
ISREGISTERNEEDED = 1
AUTHENTICATIONMODE = 0

ROUTEMODEIP2TEL = 1
ROUTEMODETEL2IP = 1
ENABLECURRENTDISCONNECT = 1
ENABLEREVERSALPOLARITY = 1
MAXACTIVECALLS = 8
CHANNELSELECTMODE = 0
GWDEBUGLEVEL = 5
ENABLEPROXYKEEPALIVE = 1
ENABLERPIHEADER = 1
ENABLEEARLYMEDIA = 1
ISUSERPHONE = 0
ISDNRXOVERLAP_0 = 1
ISDNRXOVERLAP_1 = 0
ISDNRXOVERLAP_2 = 0
ISDNRXOVERLAP_3 = 0
ISDNRXOVERLAP_4 = 0
ISDNRXOVERLAP_5 = 0
ISDNRXOVERLAP_6 = 0
ISDNRXOVERLAP_7 = 0
ISDNRXOVERLAP_8 = 0
ISDNRXOVERLAP_9 = 0
ISDNRXOVERLAP_10 = 0
ISDNRXOVERLAP_11 = 0
ISDNRXOVERLAP_12 = 0
ISDNRXOVERLAP_13 = 0
ISDNRXOVERLAP_14 = 0
ISDNRXOVERLAP_15 = 0
ISDNRXOVERLAP_16 = 0
ISDNRXOVERLAP_17 = 0
ISDNRXOVERLAP_18 = 0
ISDNRXOVERLAP_19 = 0
PROXYNAME = 'interop.com'
SIPGATEWAYNAME = 'interop.com'
USERNAME = '90739886'
PASSWORD = '123456'
ALTROUTINGTEL2IPENABLE = 1
ALTROUTINGTEL2IPMODE = 0
KEYCFUNCOND = '*5'
KEYCFDEACT = '#5'
KEYCLIR = '*6'
KEYCLIRDEACT = '#6'
DISCONNECTONBROKENCONNECTION = 0
WAITINGBEEPDURATION = 200
SENDINVITETOPROXY = 1
ASSERTEDIDMODE = 1

MWIDISPLAY = 1
ENABLEMWI = 1
USEGATEWAYNAMEFOROPTIONS = 1
USESOURCENUMBERASDISPLAYNAME = 1
ISFAXUSED = 1
SIPTRANSPORTTYPE = 1
KEYCFDONOTDISTURB = '*7'
ENABLEUUITEL2IP = 1
ENABLEUUIIP2TEL = 1
SIP183BEHAVIOUR = 1
ISUSETOHEADERASCALLEDNUMBER = 1
PROGRESSINDICATOR2ISDN = -1
LOCALISDNRBSOURCE = 0
ISDNTRANSFERCAPABILITY_0 = 1
ISDNTRANSFERCAPABILITY_1 = -1
ISDNTRANSFERCAPABILITY_2 = -1
ISDNTRANSFERCAPABILITY_3 = -1
ISDNTRANSFERCAPABILITY_4 = -1
ISDNTRANSFERCAPABILITY_5 = -1
ISDNTRANSFERCAPABILITY_6 = -1
ISDNTRANSFERCAPABILITY_7 = -1
ISDNTRANSFERCAPABILITY_8 = -1
ISDNTRANSFERCAPABILITY_9 = -1
ISDNTRANSFERCAPABILITY_10 = -1
ISDNTRANSFERCAPABILITY_11 = -1
ISDNTRANSFERCAPABILITY_12 = -1
ISDNTRANSFERCAPABILITY_13 = -1
ISDNTRANSFERCAPABILITY_14 = -1
ISDNTRANSFERCAPABILITY_15 = -1
ISDNTRANSFERCAPABILITY_16 = -1
ISDNTRANSFERCAPABILITY_17 = -1
ISDNTRANSFERCAPABILITY_18 = -1
ISDNTRANSFERCAPABILITY_19 = -1
PIFORDISCONNECTMSG = -1
PLAYRBTONE2TRUNK_0 = 0
PLAYRBTONE2TRUNK_1 = -1
PLAYRBTONE2TRUNK_2 = -1
PLAYRBTONE2TRUNK_3 = -1
PLAYRBTONE2TRUNK_4 = -1
PLAYRBTONE2TRUNK_5 = -1
PLAYRBTONE2TRUNK_6 = -1
PLAYRBTONE2TRUNK_7 = -1
PLAYRBTONE2TRUNK_8 = -1
PLAYRBTONE2TRUNK_9 = -1
PLAYRBTONE2TRUNK_10 = -1

PLAYRBTONE2TRUNK_11 = -1
PLAYRBTONE2TRUNK_12 = -1
PLAYRBTONE2TRUNK_13 = -1
PLAYRBTONE2TRUNK_14 = -1
PLAYRBTONE2TRUNK_15 = -1
PLAYRBTONE2TRUNK_16 = -1
PLAYRBTONE2TRUNK_17 = -1
PLAYRBTONE2TRUNK_18 = -1
PLAYRBTONE2TRUNK_19 = -1
3XXBEHAVIOR = 1
ENABLEREASONHEADER = 0
TRUNKPSTNALERTTIMEOUT_0 = -1
TRUNKPSTNALERTTIMEOUT_1 = -1
TRUNKPSTNALERTTIMEOUT_2 = -1
TRUNKPSTNALERTTIMEOUT_3 = -1
TRUNKPSTNALERTTIMEOUT_4 = -1
TRUNKPSTNALERTTIMEOUT_5 = -1
TRUNKPSTNALERTTIMEOUT_6 = -1
TRUNKPSTNALERTTIMEOUT_7 = -1
TRUNKPSTNALERTTIMEOUT_8 = -1
TRUNKPSTNALERTTIMEOUT_9 = -1
TRUNKPSTNALERTTIMEOUT_10 = -1
TRUNKPSTNALERTTIMEOUT_11 = -1
TRUNKPSTNALERTTIMEOUT_12 = -1
TRUNKPSTNALERTTIMEOUT_13 = -1
TRUNKPSTNALERTTIMEOUT_14 = -1
TRUNKPSTNALERTTIMEOUT_15 = -1
TRUNKPSTNALERTTIMEOUT_16 = -1
TRUNKPSTNALERTTIMEOUT_17 = -1
TRUNKPSTNALERTTIMEOUT_18 = -1
TRUNKPSTNALERTTIMEOUT_19 = -1
SASDEFAULTGATEWAYIP = '192.168.190.230'
ENABLESAS = 1
RTPONLYMODEFORTRUNK = -1
REGISTRARTRANSPORTTYPE = 1
BCHANNELNEGOTIATIONFORTRUNK_0 = 2
BCHANNELNEGOTIATIONFORTRUNK_1 = -1
BCHANNELNEGOTIATIONFORTRUNK_2 = -1
BCHANNELNEGOTIATIONFORTRUNK_3 = -1
BCHANNELNEGOTIATIONFORTRUNK_4 = -1
BCHANNELNEGOTIATIONFORTRUNK_5 = -1
BCHANNELNEGOTIATIONFORTRUNK_6 = -1
BCHANNELNEGOTIATIONFORTRUNK_7 = -1
BCHANNELNEGOTIATIONFORTRUNK_8 = -1
BCHANNELNEGOTIATIONFORTRUNK_9 = -1

BCHANNELNEGOTIATIONFORTRUNK_10 = -1
BCHANNELNEGOTIATIONFORTRUNK_11 = -1
BCHANNELNEGOTIATIONFORTRUNK_12 = -1
BCHANNELNEGOTIATIONFORTRUNK_13 = -1
BCHANNELNEGOTIATIONFORTRUNK_14 = -1
BCHANNELNEGOTIATIONFORTRUNK_15 = -1
BCHANNELNEGOTIATIONFORTRUNK_16 = -1
BCHANNELNEGOTIATIONFORTRUNK_17 = -1
BCHANNELNEGOTIATIONFORTRUNK_18 = -1
BCHANNELNEGOTIATIONFORTRUNK_19 = -1
DIGITALOOSBEHAVIORFORTRUNK_0 = 0
DIGITALOOSBEHAVIORFORTRUNK_1 = -1
DIGITALOOSBEHAVIORFORTRUNK_2 = -1
DIGITALOOSBEHAVIORFORTRUNK_3 = -1
DIGITALOOSBEHAVIORFORTRUNK_4 = -1
DIGITALOOSBEHAVIORFORTRUNK_5 = -1
DIGITALOOSBEHAVIORFORTRUNK_6 = -1
DIGITALOOSBEHAVIORFORTRUNK_7 = -1
DIGITALOOSBEHAVIORFORTRUNK_8 = -1
DIGITALOOSBEHAVIORFORTRUNK_9 = -1
DIGITALOOSBEHAVIORFORTRUNK_10 = -1
DIGITALOOSBEHAVIORFORTRUNK_11 = -1
DIGITALOOSBEHAVIORFORTRUNK_12 = -1
DIGITALOOSBEHAVIORFORTRUNK_13 = -1
DIGITALOOSBEHAVIORFORTRUNK_14 = -1
DIGITALOOSBEHAVIORFORTRUNK_15 = -1
DIGITALOOSBEHAVIORFORTRUNK_16 = -1
DIGITALOOSBEHAVIORFORTRUNK_17 = -1
DIGITALOOSBEHAVIORFORTRUNK_18 = -1
DIGITALOOSBEHAVIORFORTRUNK_19 = -1
SASPROXYSET = 1
SASBINDINGMODE = 1
SIPREROUTINGMODE = 1
SASSURVIVABILITYMODE = 2
RELIABLECONNECTIONPERSISTENTMODE = 1

[SCTP Params]

[VXML Params]

[IPsec Params]

[Audio Staging 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 InterfaceTable *****
;
;

[InterfaceTable]
FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes, InterfaceTable_IPv6InterfaceMode, InterfaceTable_IPAddress, InterfaceTable_PrefixLength, InterfaceTable_Gateway, InterfaceTable_VlanID, InterfaceTable_InterfaceName;
InterfaceTable 0 = 6, 0, 192.168.190.230, 24, 192.168.190.254, 1, ALL;

[\InterfaceTable]

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

[PREFIX]
FORMAT PREFIX_Index = PREFIX_DestinationPrefix, PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileId, PREFIX_MeteringCode, PREFIX_SrcIPGroupID, PREFIX_DestHostPrefix, PREFIX_DestIPGroupID, PREFIX_SrcHostPrefix, PREFIX_TransportType, PREFIX_SrcTrunkGroupID;
PREFIX 0 = 5000*, 192.168.190.230, *, 0, 255, -1, , -1, , -1, -1;
PREFIX 1 = 0, 192.168.190.230, *, 0, 255, -1, , -1, , -1, -1;
PREFIX 2 = 7, 192.168.190.230, *, 0, 255, -1, , -1, , -1, -1;
PREFIX 3 = 9, 192.168.190.230, *, 0, 255, -1, , -1, , -1, -1;

[\PREFIX]

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

```
[ CoderName ]
FORMAT CoderName_Index = CoderName_Type, CoderName_PacketInterval,
CoderName_rate, CoderName_PayloadType, CoderName_Sce;
CoderName 0 = 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 = 1, 255, 1, 2, 50004, 0, 255, 1;
TrunkGroup 1 = 2, 0, 1, 2, 90739886, 0, 0, 2;

[ \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 0 = 0*, *, *, 4, 1, 1, 0, 255, , , 255;

[ \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 0 = 9*, *, *, 255, 255, 0, 0, 5, , , 255;

[ \NumberMapTel2Ip ]

;
;  *** TABLE SourceNumberMapIp2Tel ***
;
;

[ SourceNumberMapIp2Tel ]
FORMAT SourceNumberMapIp2Tel_Index =
SourceNumberMapIp2Tel_DestinationPrefix, SourceNumberMapIp2Tel_SourcePrefix,
SourceNumberMapIp2Tel_SourceAddress, SourceNumberMapIp2Tel_NumberType,
SourceNumberMapIp2Tel_NumberPlan, SourceNumberMapIp2Tel_RemoveFromLeft,
SourceNumberMapIp2Tel_RemoveFromRight,
SourceNumberMapIp2Tel_LeaveFromRight, SourceNumberMapIp2Tel_Prefix2Add,
SourceNumberMapIp2Tel_Suffix2Add,
SourceNumberMapIp2Tel_IsPresentationRestricted;
SourceNumberMapIp2Tel 0 = 0, *, *, 4, 1, 0, 0, 255, 90739886, , 255;

[ \SourceNumberMapIp2Tel ]

;
;  *** TABLE SourceNumberMapTel2Ip ***
;
;

[ SourceNumberMapTel2Ip ]
FORMAT SourceNumberMapTel2Ip_Index =
SourceNumberMapTel2Ip_DestinationPrefix, SourceNumberMapTel2Ip_SourcePrefix,
SourceNumberMapTel2Ip_SourceAddress, SourceNumberMapTel2Ip_NumberType,
SourceNumberMapTel2Ip_NumberPlan, SourceNumberMapTel2Ip_RemoveFromLeft,
SourceNumberMapTel2Ip_RemoveFromRight,
SourceNumberMapTel2Ip_LeaveFromRight, SourceNumberMapTel2Ip_Prefix2Add,
SourceNumberMapTel2Ip_Suffix2Add,
SourceNumberMapTel2Ip_IsPresentationRestricted;
SourceNumberMapTel2Ip 0 = 9*, *, *, 255, 255, 0, 0, 255, 00, , 0;

[ \SourceNumberMapTel2Ip ]
```

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

[ PstnPrefix ]
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId,
PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = [50004-50005]#, 1, *, *, 0, -1, , ;
PstnPrefix 1 = *, 2, *, *, 0, -1, *, ;

[ \PstnPrefix ]

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

[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = 192.168.193.27, -1, 0;
ProxyIp 1 = 192.168.193.27, -1, 1;
ProxyIp 2 = 192.168.190.230:5080, -1, 0;

[ \ProxyIp ]

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

[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;
TxDtmfOption 1 = 0;

[ \TxDtmfOption ]

;
;  *** TABLE TrunkGroupSettings ***
;
;
```

[TrunkGroupSettings]
FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId, TrunkGroupSettings_ChannelSelectMode, TrunkGroupSettings_RegistrationMode, TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser, TrunkGroupSettings_ServingIPGroup;
TrunkGroupSettings 0 = 1, 0, 0, , , -1;
TrunkGroupSettings 1 = 2, 2, 0, , , -1;

[\TrunkGroupSettings]

;
; * 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 1 = , 1, 0, 1, 70, 10, 46, 40, -11, 0, 0, 0, 0, 0, 1, 0, 0, 700, 0, -1, 255;

[\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_EnableHold;
IpProfile 1 = , 1, 0, 1, 70, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0, -1, 1, 0, 0, -1, 1;

[\IpProfile]

;
; * TABLE EnableCallerId *****
;
;

[EnableCallerId]
FORMAT EnableCallerId_Index = EnableCallerId_IsEnabled, EnableCallerId_Module, EnableCallerId_Port;
EnableCallerId 0 = 1, 1, 1;
EnableCallerId 1 = 1, 1, 2;
EnableCallerId 2 = 1, 1, 3;
EnableCallerId 3 = 1, 1, 4;

[\EnableCallerId]

;
; * TABLE CallerDisplayInfo *****
;
;

[CallerDisplayInfo]
FORMAT CallerDisplayInfo_Index = CallerDisplayInfo_DisplayString, CallerDisplayInfo_IsCidRestricted, CallerDisplayInfo_Module, CallerDisplayInfo_Port;
CallerDisplayInfo 0 = 50004, 0, 1, 1;
CallerDisplayInfo 1 = 50005, 0, 1, 2;

[\CallerDisplayInfo]

;
; * TABLE Authentication *****
;
;

[Authentication]
FORMAT Authentication_Index = Authentication_UserId, Authentication_UserPassword, Authentication_Module, Authentication_Port;
Authentication 0 = 50004, 123456, 1, 1;
Authentication 1 = 50005, 123456, 1, 2;

[\Authentication]

;

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

[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap;
ProxySet 0 = 1, 60, 0, 0;
ProxySet 1 = 1, 60, 0, 0;

[ \ProxySet ]

;
;  *** TABLE SASRegistrationManipulation ***
;
;

[ SASRegistrationManipulation ]
FORMAT SASRegistrationManipulation_Index =
SASRegistrationManipulation_RemoveFromRight,
SASRegistrationManipulation_LeaveFromRight;
SASRegistrationManipulation 0 = 0, 0;

[ \SASRegistrationManipulation ]
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.