6.40A – AudioCodes Mediant 800 MSBG

1. Important Notes

- Check the *SIP 3rd Party Validation Website* for current validation status. The *SIP 3rd party Validation Website can be viewed at:* http://testlab.inin.com

- Unlike other Mediant devices, no INI files will be provided with this document. The Mediant 800 requires two INIs to have a complete configuration. These INI files are interdependent and editing the DATA.INI file is not supported by AudioCodes.

- As this unit is a critical piece of the SIP infrastructure, it is highly recommended that DHCP not be used. A static IP address is the preferred method and is how the unit was configured during validation.

- The M800 will require 3 IP Addresses as a base configuration. One for the WAN interface and two for the internal interfaces (VoIP & Data).

- After a factory reset, the Mediant 800's default configuration has DHCP enabled, and the VOIP & Data IP addresses will be 192.168.0.1 and 192.168.0.2 respectively. The IP Address of the VoIP interface is the address of the Web Interface.

- If a configuration change is made that causes errors, the Mediant 800 will revert to the default IP & DHCP configuration. If this happens connect a PC to the Mediant 800 via crossover cable and undo any changes made.

- The Mediant 800 MSBG has two configuration sections that are controlled by two different INI files. Board.INI contains the VoIP/Gateway configuration. Data.ini contains the Firewall/Routing configuration.

- Any time you see -1 in a configuration this is mean "not configured".

- While this device supports IPv6, IPv4 was used in the certification tests.

2. Vendor Documentation

Documentation can be found on the CD shipped with the Mediant 800.

3. Validated Firmware Version

6.40A.037.009

Initial Setup - Network Settings

4. Install

Download the Mediant 800 files form the *3rd Party Validation Website*:

http://testlab.inin.com

Contained in the zip file will be the validated version of firmware (.cmp), as well as any supplemental configuration files.

5. Configuration

Methods:

- Manipulation of the supplied .ini file(s), then uploading it via the Web interface.
  - o This method is not supported for the Mediant 800. The Mediant 800 requires two separate INI files that are interdependent and modifying the DATA.INI file manually is not supported by AudioCodes. As a result, no INI files have been provided with this document.
  - o Backing up of the configuration by downloading the INIs and restoring them if the device fails is still supported.
- Web Interface
  - o This method is the preferred method for all configuration needs.
  - o **Please note:** Caution should be exercised and the AudioCodes documentation should always be referenced when using the Web interface configuration option.
- TFTP
  - o This method of configuring the Mediant 800 has not been tested.
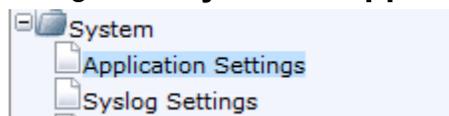
Contents

Initial Setup

Network Settings

1. Connect a Laptop/Desktop with its NIC configured for DHCP to the Mediant 800 via a crossover cable or a switch
2. Open a browser and navigate to 192.168.0.1 entering the default username and password
    a. User: Admin
    b. Pass: Admin
3. After logging in, always make sure the radio button next to **Full** is selected. Without this radio button selected you do not have access to all the possible configuration options
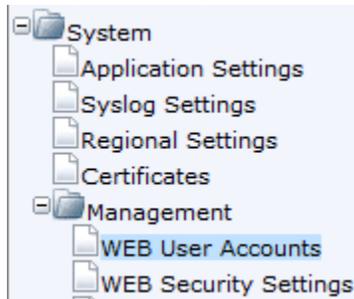

4. Navigate to **System** > **Application Settings**

5. Enter the appropriate information for your network

6. Navigate to **System** > **WEB User Accounts**

7. Change the Admin user password and, if necessary, the user name

8. Navigate to **Data** > **WAN Access** > **Settings**

9. Change **Connection Type** to **Manual IP Address Ethernet Connection**
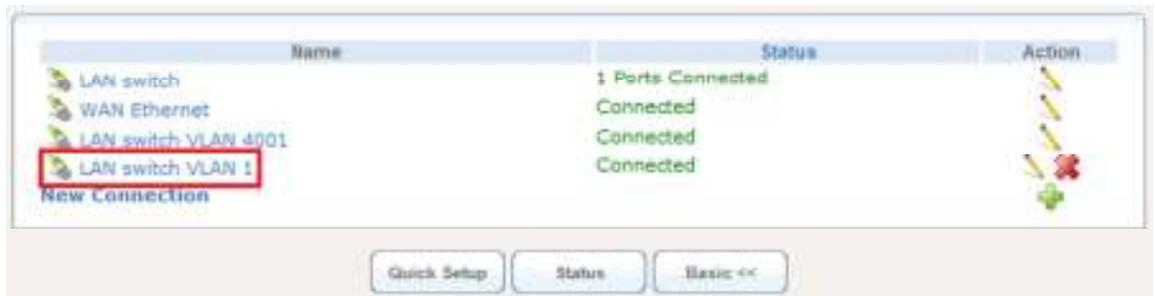

10. Enter the settings necessary to fit your network design.

11. Navigate to **Data** > **Data System** > **Connections**

12. Select **LAN Switch VLAN 1**

13. Select **Settings**

14. Enter the settings necessary to fit your network design.

15. Click **OK** on this and the next page

16. Navigate to **Data** > **Data Services** > **DHCP Server**
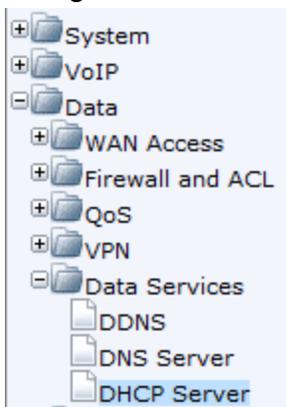
17.   Select **LAN switch VLAN 1**

18.   If you do not wish to use the Mediant 800's DHCP server then change the highlighted (see below) drop down to **Disabled** and click **OK**. Otherwise, modify the settings to fit your network design.

19.   Navigate to **VoIP** > **Network** > **IP Settings**
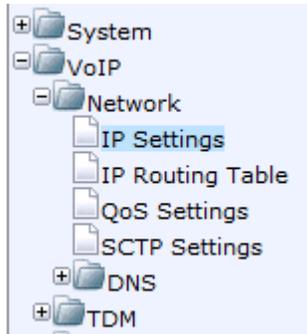
20.   Click the radio button next to **Index 0**

21.   Click the **Edit** button that appears above the IP settings table

22.   Enter the information necessary to fit your network design. Also ensure that the **WAN Interface Name** has the proper

Interface selected in the drop down. Click **Apply**.
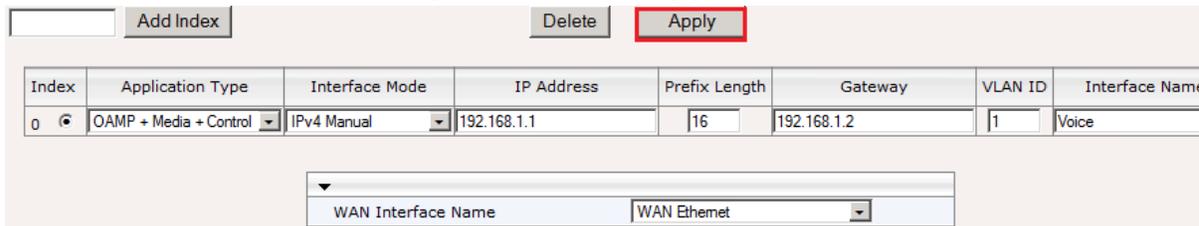
23.   Click **Burn** & then **Yes** on the following dialog prompt.

24.   Now that the configuration has been saved to memory and the
Mediant 800 is configured to be on your LAN, put it in place
and we'll continue in the next section with the configuration.

VoIP Settings

In this section we'll begin configuring the Mediant 800 to accept & route calls. The Mediant 800 uses an organizational structure where groups of settings are contained in a Signaling Routing Domain (SRD). For further description please see the **Configuring SRD Table** section of the **Mediant 800 MSBG User's Manual**.

There are several pieces that will need to be configured to construct the SRD. They are:

1) SIP Interface
2) Media Realm
3) IP Group
4) IP Profile
5) Proxy Set

After the Mediant 800 has been connected to your LAN and you have established connectivity, continue below. The MSBG has built in diagnostic tools (PING, ARP, & Traceroute) you can use to verify successful configuration. They are located under **Status & Diagnostics** > **Data Status** > **Diagnostics**.

General VoIP Settings

These settings are for any of the M800 models and will be needed for SBC, TDM, or FXO/FXS functionality.

1. Navigate to **VoIP** > **Media** > **Media Realm Configuration**
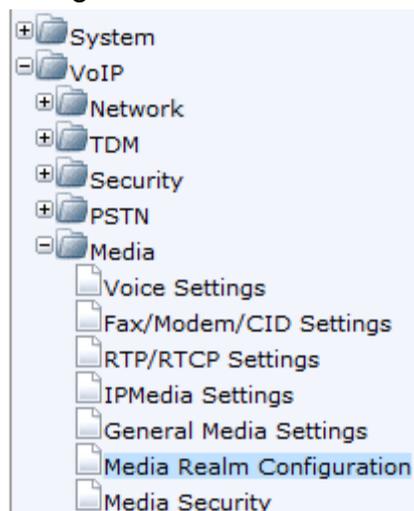
3. Click **Add**. For the purposes of this document, we will be considering Voice indexes to be between 1 & 10 and WAN indexes to be between 11 & 20.
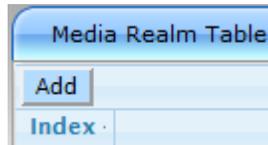
Note: The default configuration has no media realms configured. You will need at least one Media Realm for TDM. If using SBC functionality, you will need another Media Realm for the WAN.

4. Enter the following settings as shown:
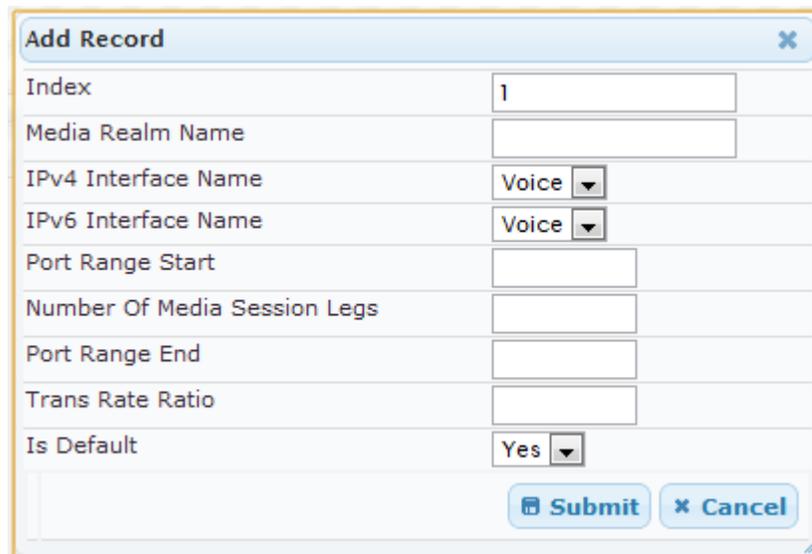
5. Click **Submit**. A dialog box appears telling you that you've configured an offline parameter. Ignore this for now as we will be rebooting later.

6. Navigate to **VoIP** > **Media** > **IPMedia Settings**
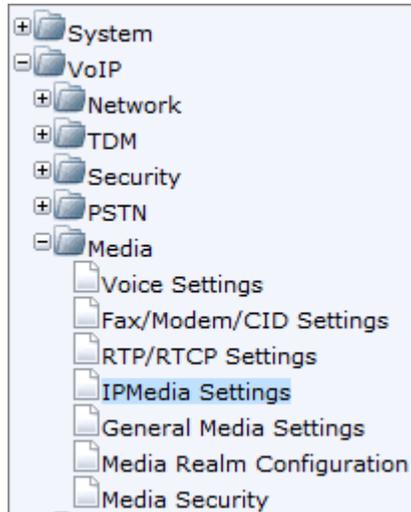
7. Enter in the **Number of Media Channels** necessary to meet the configuration needs. The number of channels should be double the maximum number of projected concurrent calls.

8. Click **Submit**

9. Navigate to **VoIP** > **Control Network** > **SIP Interface Table**

10. Enter 1 in the box next to **Add** and then click **Add**.

11. Enter a **Network Interface** name, **SIP Ports**, **SRD**, and choose an **Application Type** for the SIP Interface. Click **Apply**.
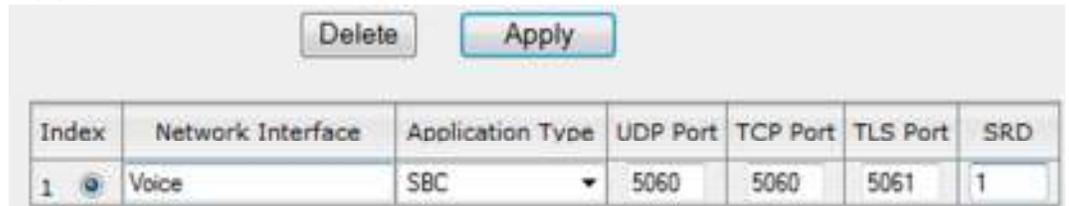
Note: The Network Interface Name has to match, precisely, the name given to the VoIP Interface in **VoIP > Network > IP Settings**.

12. Navigate to **VoIP > Control Network > IP Group Table**
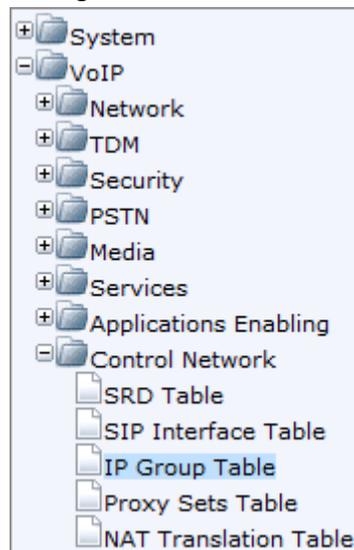
13. In the **IP Group Table** configure the following values for the LAN Group clicking **Submit** when settings are entered.

Index	1

Common Parameters

Type	SERVER
Description	Voice
Proxy Set ID	1
SIP Group Name	
Contact User	
Domain Name in Contact	
⚡ SRD	0
⚡ Media Realm	Voice
IP Profile ID	0

Gateway Parameters

Always Use Route Table	No
Routing Mode	Not Configured
SIP Re-Routing Mode	Standard

SBC Parameters

Classify By Proxy Set	Enable
Max Number Of Registered Users	-1
Inbound Message Manipulation Set	-1
Outbound Message Manipulation Set	-1
Registration Mode	User initiates registrations
Authentication Mode	User authenticates
Authentication Method List	
Enable SBC Client Forking	No

14. Navigate to **VoIP** > **Control Network** > **Proxy Sets**

15. You will need to create a proxy set for the IC Server(s). Configure the following values for the LAN Proxy Set clicking **Submit** when done.

Proxy Set ID	1

	Proxy Address	Transport Type
1	192.168.1.11:5060	TCP
2	192.168.1.13:5060	TCP
3		
4		
5		

Enable Proxy Keep Alive	Using Options
Proxy Keep Alive Time	60
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	Yes
Proxy Redundancy Mode	Homing
⚡SRD Index	1
Classification Input	IP only

16. Navigate to **VoIP** > **SIP Definitions** > **General Parameters**
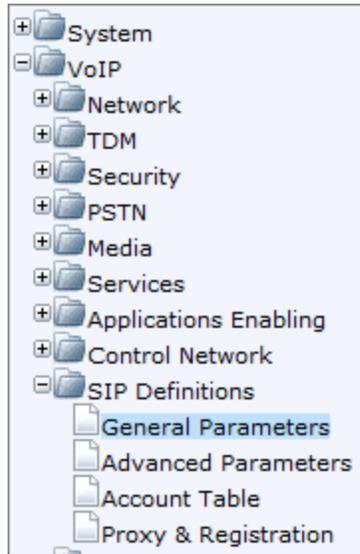
17. There are many different combinations of settings you can use here. The main things that need configured are:
    Enable Early Media: Configure as needed
    Fax Signaling Method: T.38 Relay
    SIP Transport Type: Configure as needed
    SIP UDP Local Port: Same as Voice **SIP Interface**
    SIP TCP Local Port: Same as Voice **SIP Interface**
    SIP TLS Local Port: Same as Voice **SIP Interface**
    Enable Sips: Please see the section on Enabling TLS.
    SIP Destination Port: Matches protocol for **SIP Transport Type**

18. Once your changes are complete, click **Submit**

19. Navigate to **VoIP** > **Coders and Profiles** > **IP Profile Settings**

20. Here you can configure various settings to handle variations of the **SIP General Parameters**. You then assign these Profiles to their respective **IP Groups**.

    These profiles can be used to handle:
    a. Differences in DSCP from one network to another
    b. If one network uses SRTP and another uses RTP if Media Security is Enabled
    c. If one network uses T.38 Relay and another does not allow faxing
    d. If you set up Coder Groups you can assign them to the IP Profile and then assign the IP Profile to the proper IP Groups.
    e. Etc.

21. Once your configurations are complete, press **Submit** and, if necessary, assign them to the proper **IP Groups**.

22. Navigate to **VoIP** > **Control Network** > **SRD Table**
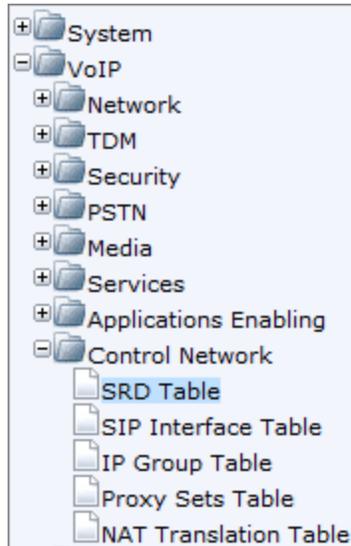
23. Select **SRD Index**: 1 – Not Exist
24. Configure the SRD to match the following image
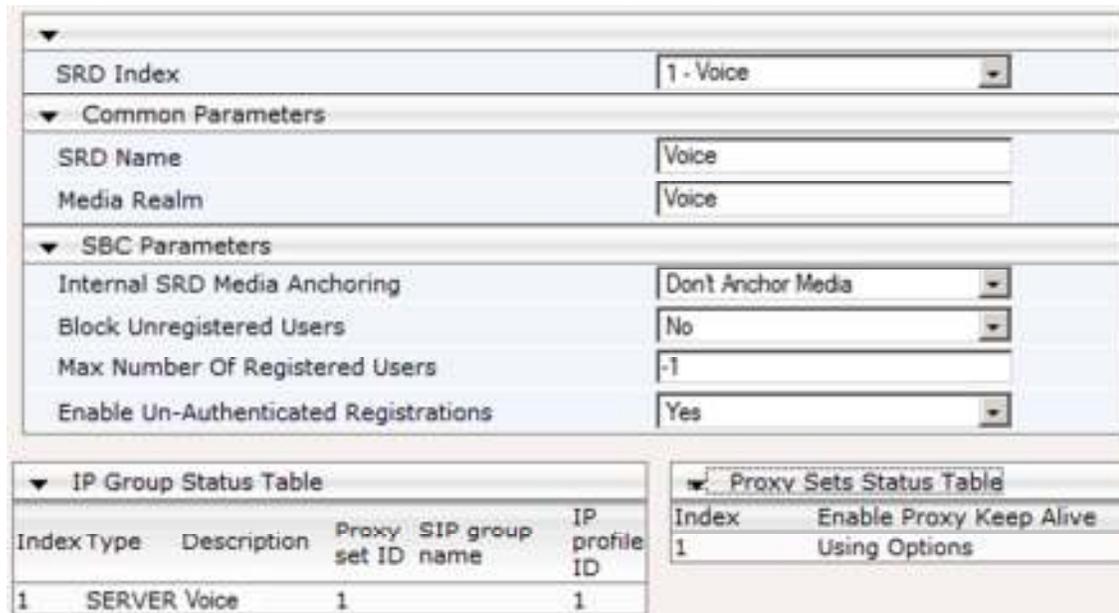
25. Navigate to **Device Actions** > **Reset**

26. Ensure **Burn to Flash** is set to **Yes** and then click **Reset**
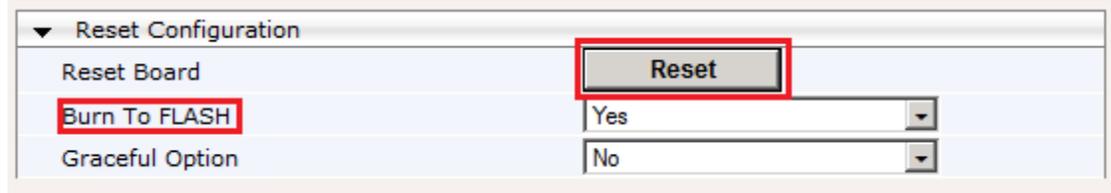
SBC VoIP Settings

1. Navigate to **VoIP** > **Applications Enabling** >
    Applications Enabling
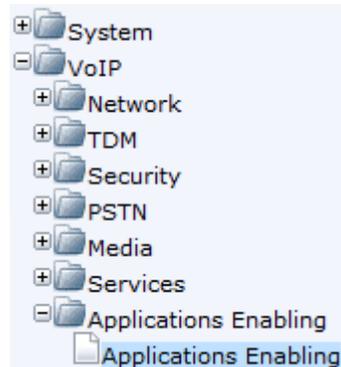
2. Change **Enable SBC Application** to **Enable**
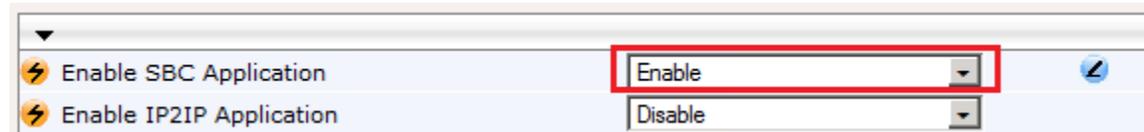
3. Click **Submit**

4. Navigate to **VoIP** > **Media** > **Media Realm Configuration**
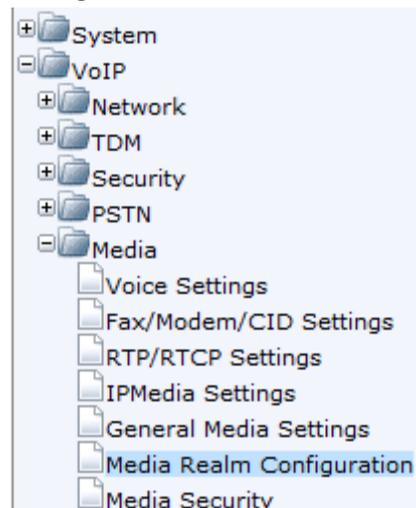
5. Click **Add**. For the purposes of this document, we will be
    considering Voice indexes to be between 1 & 10 and WAN

indexes to be between 11 & 20.

Media Realm Table

Add

Index ·
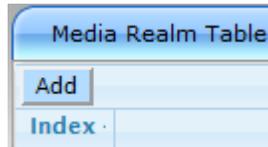
6. Enter the settings as shown:

Add Record ✖

Index	11
Media Realm Name	WAN
IPv4 Interface Name	WAN ▼
IPv6 Interface Name	WAN ▼
Port Range Start	
Number Of Media Session Legs	
Port Range End	
Trans Rate Ratio	
Is Default	No ▼

🖫 Submit   ✖ Cancel
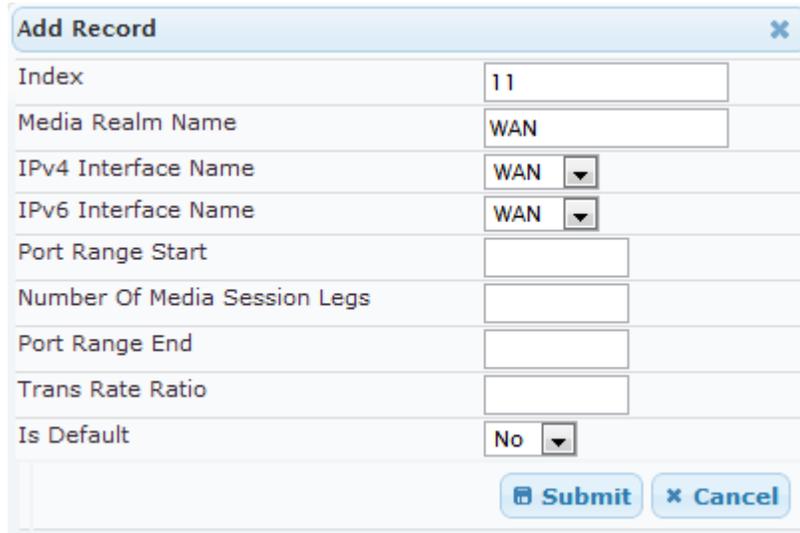
7. Click **Submit**. A dialog box appears telling you that you've configured and offline parameter. Ignore this for now as we will be rebooting later.

🖫 Submit   ✖ Cancel

8. Navigate to **VoIP** > **Control Network** > **SIP Interface Table**

System
VoIP
  Network
  Security
  Media
  Services
  Applications Enabling
  Control Network
    SRD Table
    SIP Interface Table
    IP Group Table
    Proxy Sets Table
    NAT Translation Table

9. Enter 11 in the box next to **Add** and then click **Add**.

11   Add

10. Enter a Network Interface name, SIP Ports, SRD, and choose an application type (SBC) for the SIP Interface. Click **Apply**.

Index	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	SRD
1	Voice	SBC	5060	5060	5061	1
11	WAN	SBC	5060	5060	5061	11

Note: WAN is the default Network Interface name for the WAN port. It is not recommended to try changing this.

11. Navigate to **VoIP** > **Control Network** > **IP Group Table**

System
VoIP
  Network
  TDM
  Security
  PSTN
  Media
  Services
  Applications Enabling
  Control Network
    SRD Table
    SIP Interface Table
    IP Group Table
    Proxy Sets Table
    NAT Translation Table
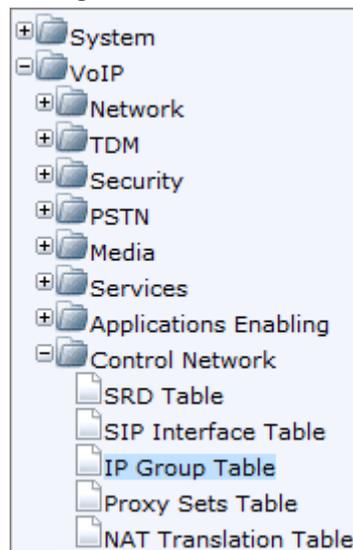
12. In the IP Group Table select and/or enter the following values for the WAN Group clicking Submit when settings

are entered.

Index	11

▼ Common Parameters

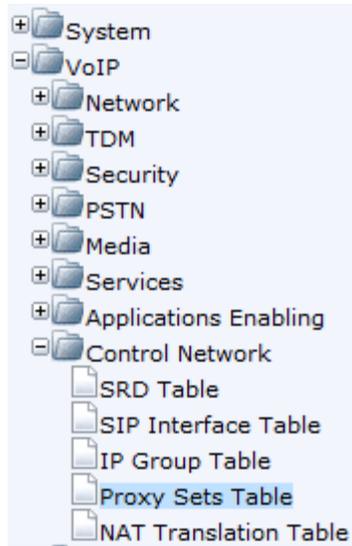| Type	SERVER
Description	WAN
Proxy Set ID	11
SIP Group Name	
Contact User	
Domain Name in Contact	
⚡ SRD	0
⚡ Media Realm	WAN
IP Profile ID	2

▼ Gateway Parameters

Always Use Route Table	No
Routing Mode	Not Configured
SIP Re-Routing Mode	Standard

▼ SBC Parameters

Classify By Proxy Set	Enable
Max Number Of Registered Users	-1
Inbound Message Manipulation Set	-1
Outbound Message Manipulation Set	-1
Registration Mode	User initiates registrations
Authentication Mode	User authenticates
Authentication Method List	
Enable SBC Client Forking	No

14. Navigate to **VoIP** > **Control Network** > **Proxy Sets**

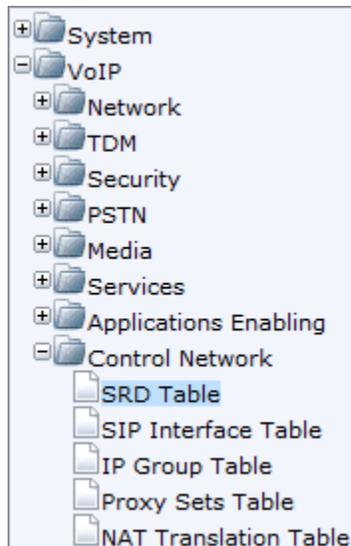15. You will need to create a proxy set for the Remote Endpoint(s). Configure the values for the WAN Proxy Set, clicking **Submit** when done.

	Proxy Address	Transport Type
Proxy Set ID	11	
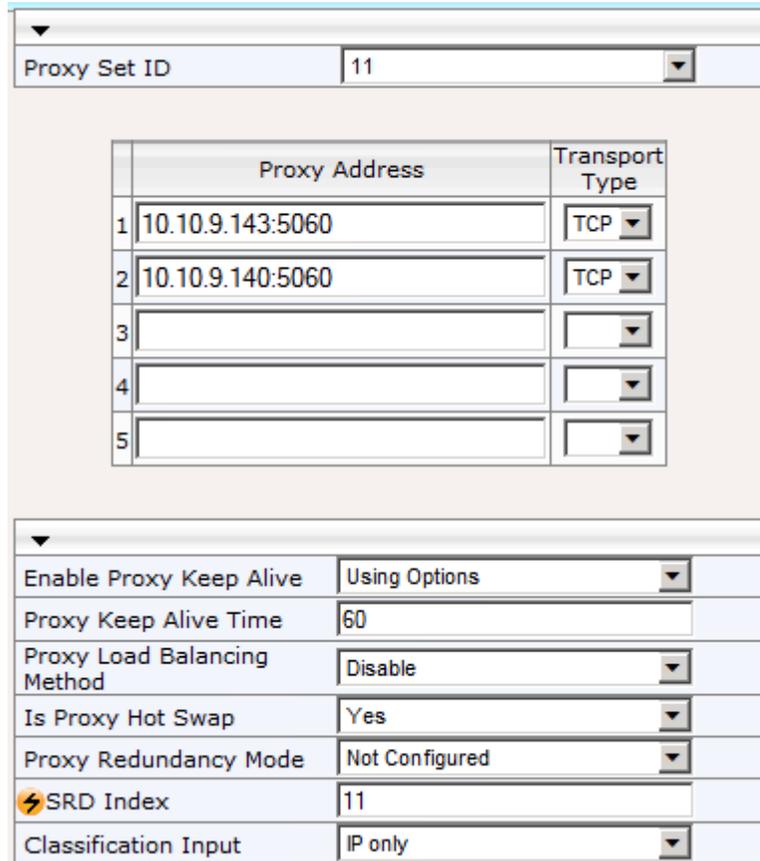
	Proxy Address	Transport Type
1	10.10.9.143:5060	TCP
2	10.10.9.140:5060	TCP
3		
4		
5		

Enable Proxy Keep Alive	Using Options
Proxy Keep Alive Time	60
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	Yes
Proxy Redundancy Mode	Not Configured
⚡ SRD Index	11
Classification Input	IP only

16. Navigate to **VoIP** > **Control Network** > **SRD Table**

- ⊞ System
- ⊟ VoIP
  - ⊞ Network
  - ⊞ TDM
  - ⊞ Security
  - ⊞ PSTN
  - ⊞ Media
  - ⊞ Services
  - ⊞ Applications Enabling
  - ⊟ Control Network
    - SRD Table
    - SIP Interface Table
    - IP Group Table
    - Proxy Sets Table
    - NAT Translation Table

17. Select **SRD Index**: 11 – Not exist

18. Expand all sections by clicking the ^s next to their names.
Configure as follows:

19. Navigate to **Device Actions** > **Reset**

20. Ensure **Burn to Flash** is set to **Yes** and then click **Reset**

TDM Configuration

Now that the SRDs & IP Groups have been configured we can move on to configuring the TDM Settings. For more details see the **GW and IP to IP** section of the **Mediant 800 MSBG SP User's Manual**.

1. Navigate to **VoIP > PSTN > Trunk Settings**
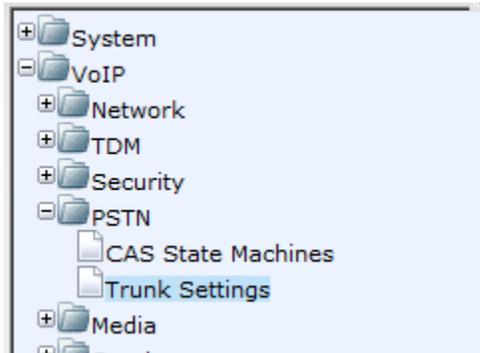
2. Choose the appropriate **Protocol Type**. NI2 will be used for this example.

3. Choose the appropriate **Clock Master**, **ISDN Termination Side**, and any other settings necessary to match your **Carrier**
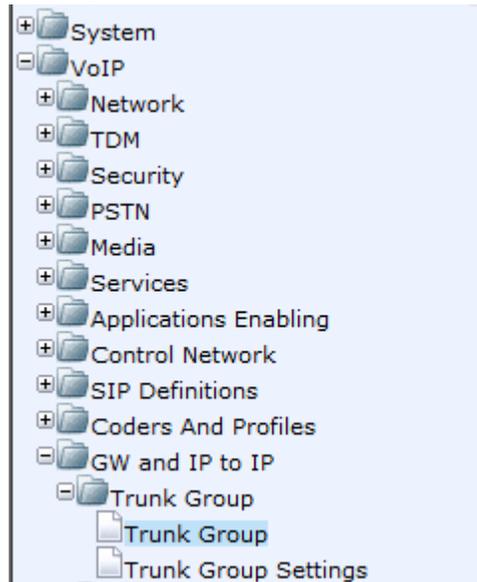
Requirements

Trunk Configuration	
Clock Master	Recovered
Auto Clock Trunk Priority	0
Line Code	B8ZS
Line Build Out Loss	0 dB
Trace Level	Full ISDN Trace
Line Build Out Overwrite	OFF
Framing Method	T1 FRAMING ESF CRC6

ISDN Configuration	
ISDN Termination Side	User side
Q931 Layer Response Behavior	0x0
Outgoing Calls Behavior	0x400
Incoming Calls Behavior	0x11000
General Call Control Behavior	0x0
ISDN NS Behaviour 2	0x8
NFAS Group Number	0
IUA Interface ID	-1
NFAS Interface ID	255
D-channel Configuration	PRIMARY

4. Click **Apply Trunk Settings**

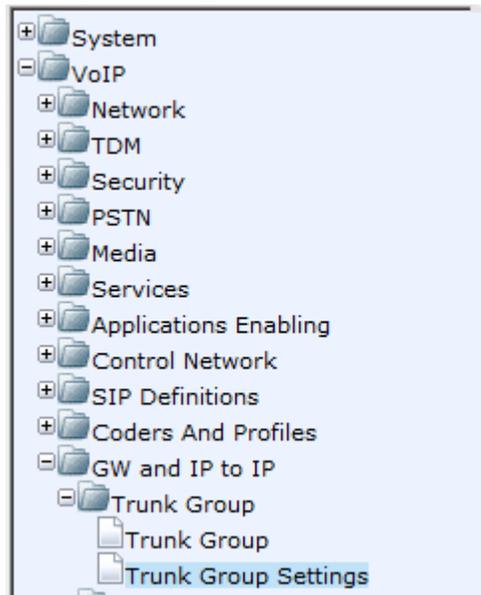5. Navigate to **VoIP > GW and IP to IP > Trunk Group > Trunk Group**

6. Enter the settings either as shown, or per **Carrier Requirements**:

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Tel Profi
1	Module 1 PRI	1	1	1-23	5555555	1	0

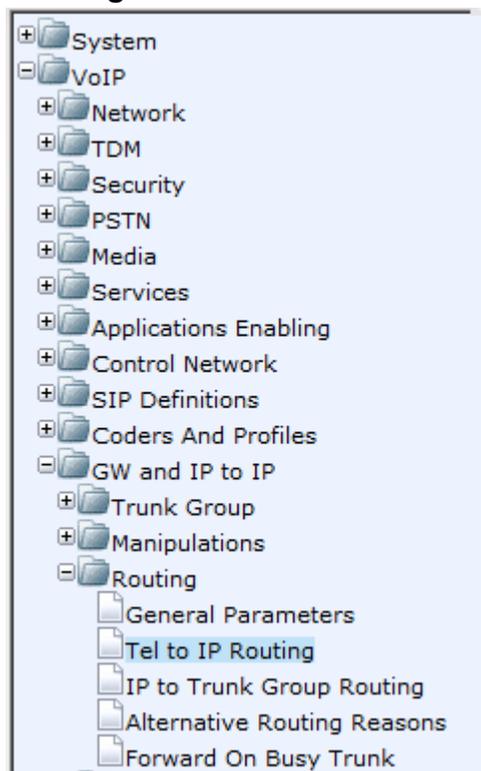7. Click **Submit**

8.   Navigate to **VoIP > GW and IP to IP > Trunk Group > Trunk
      Group Settings**

9.   Enter the appropriate settings in the fields as shown:

Trunk Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID	Gateway Name	Contact User
1 1	Cyclic Ascending				

10.   Navigate to **VoIP > GW and IP to IP > Routing > Tel to IP
      Routing**

11. Configure records in the **Tel to IP Routing** table as necessary.
    If all calls are going to the **SRD/IP Group** that were created in
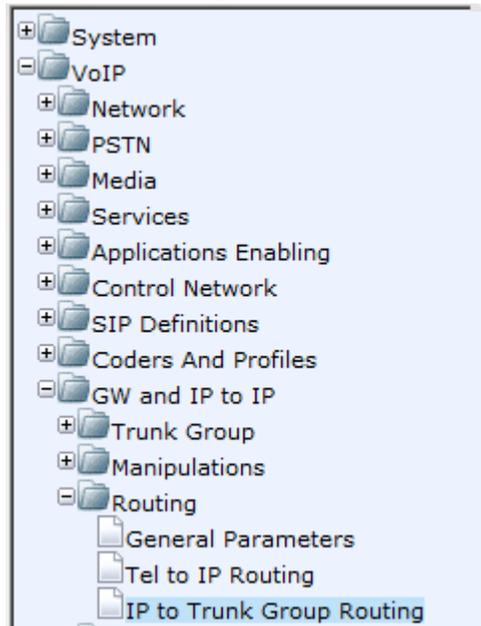   earlier sections, configure as shown.

	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	>	Dest. IP Address	Port	Transport Type	Dest
1	*	*	*				Not Configured ▾	1

	* Type	Dest. IP Group ID	Dest. SRD	IP Profile ID	Cost Group ID
	...ured ▾	1	1	-1	None ▾

12. Click **Submit**

13. Navigate to **VoIP > GW and IP to IP > Routing > IP to Trunk
    Group Routing**

14. Configure the **IP to Trunk Group Routing** to route all SIP calls
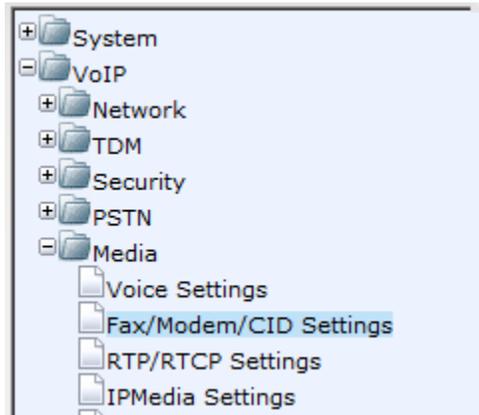   to the configured **Trunk Group**, as shown

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Trunk Group ID	IP Profile ID
1	*	*	*		1	0

15. Click **Submit**

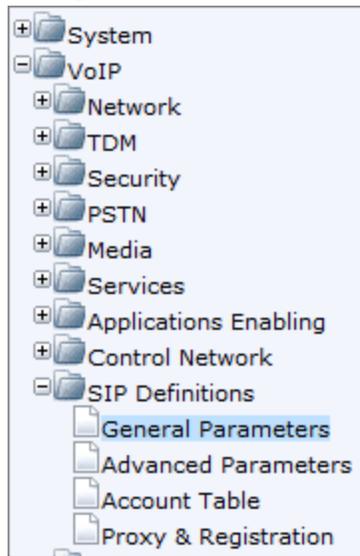### TDM Fax Configuration

1. Navigate to **VoIP > Media > Fax/Modem/CID Settings**
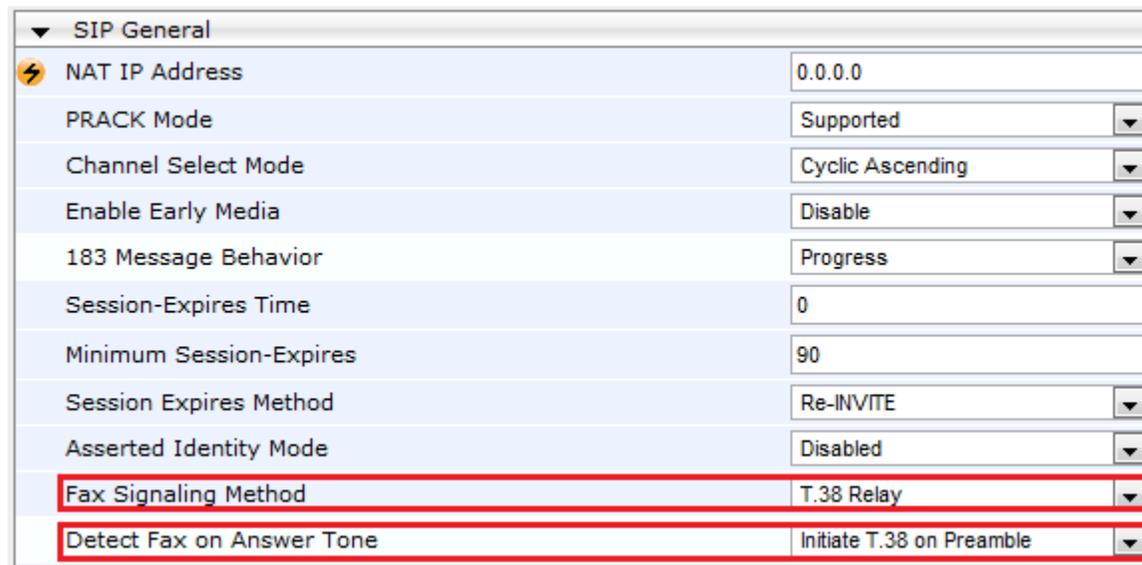

1. Configure the settings to match your needs. The following image contains the recommended settings for faxing.

2. Navigate to **VoIP** > **SIP Definitions** > **General Parameters**

System
VoIP
   Network
   TDM
   Security
   PSTN
   Media
   Services
   Applications Enabling
   Control Network
   SIP Definitions
      General Parameters
      Advanced Parameters
      Account Table
      Proxy & Registration

3. Set **Fax Signaling Method** to **T.38 Relay** and set **Detect Fax on Answer Tone** to **Initiate T.38 on Preamble**

SIP General	
NAT IP Address	0.0.0.0
PRACK Mode	Supported
Channel Select Mode	Cyclic Ascending
Enable Early Media	Disable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 on Preamble

FXO/FXS Configuration

In this section we will configure the Mediant 800 to accept & route calls using the FXO/FXS ports. Refer to FXO/FXS section of **Mediant 800 MSBG SIP User's Manual**.

1. Navigate to **GW and IP to IP > Analog Gateway > FXO Settings**

2. Set **Dialing Mode** to **One Stage**.
   NOTE: Two-Stage dialing will present a dial tone when the line is

opened and does not forward the dialed call.

Dialing Mode	One Stage
Waiting for Dial Tone	Yes
Time to Wait before Dialing [msec]	1000
Ring Detection Timeout [sec]	8
Reorder Tone Duration [sec]	255
Answer Supervision	No
Rings before Detecting Caller ID	1
Send Metering Message to IP	No
Disconnect Call on Busy Tone Detection (CAS)	Disable
Disconnect On Dial Tone	Disable
Guard Time Between Calls	1
FXO AutoDial Play BusyTone	Disable

3. Click **Submit**

Submit

4. Navigate to **VoIP > Coders And Profiles > Tel Profile Settings**

```
VoIP
  Network
  Security
  Media
  Services
  Applications Enabling
  Control Network
  SIP Definitions
  Coders And Profiles
    Coders
    Coders Group Settings
    Tel Profile Settings
    IP Profile Settings
  GW and IP to IP
```

5. Enable **Current Disconnect**

Enable Digit Delivery	Disable
Enable Polarity Reversal	Disable
Enable Current Disconnect	Enable
MWI Analog Lamp	Disable
MWI Display	Disable

6. Click **Submit**

7. Navigate to **VoIP > GW and IP to IP > DTMF and Supplementary > DTMF & Dialing**

8. Set **Max Digits In Phone Num** to meet your regional needs.
   NOTE: Example > To Dial Interactive Intelligence 18002671364, max digits eleven (11) to meet carrier/regional requirements.

Max Digits In Phone Num	11
Inter Digit Timeout [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	
RFC 2833 Payload Type	101
Hook-Flash Option	Not Supported
Digit Mapping Rules	
Dial Plan Index	-1
Dial Tone Duration [sec]	16
Hotline Dial Tone Duration [sec]	16
Enable Special Digits	Disable
Default Destination Number	1000
Special Digit Representation	Special

9. Click **Submit**

10. Navigate to **VoIP > GW and IP to IP > Hunt Group**

11. Select the **Module** you wish to configure eg **Module 1 FXS**

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Tel Profile ID
1	Module 1 FXS ▼	▼	▼	1-4	2101	1	1
2	Module 2 FXO ▼	▼	▼	1	3177154222	1	1

12. Configure the **Channels** you wish to use eg **1-4**
    *See diagram step 11.

13. For an FXS enter the First Station Extension to be used by the FXS
    devices (eg 2101) in the **Phone Number** field. For an FXO enter
    the external Phone Number in the **Phone Number** field.
    *See diagram step 11.

    Note: The Device will automatically increment each Station
    Extension. If channels 1-3 are selected and 2101 is enter for the
    Phone Number, then Port 1 will be 2101, Port 2 will be 2102, etc. If
    the Station Extensions to be assigned are not sequential, a
    separate record for each port will need to be configured. This also
    applies to the FXO.

14. Enter the appropriate Trunk Group ID to meet your configuration
    needs.

Phone Number	Trunk Group ID	Tel Profile ID
2101	1	1
3177154222	1	1

Note: Multiple entries in the table can have the same Trunk Group
ID.

15. Navigate to **VoIP > GW and IP to IP > Hunt Group Settings**

16. Enter a **Hunt Group ID** configured in step 14 in the Hunt Group ID field.

	Hunt Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID
1	1	By Dest Phone Number	Per Endpoint	1
2	2	Cyclic Descending		1

17. Choose the **Channel Select Mode**:
    NOTE: For an FXS this should be By Dest Phone Number.

18. For an FXS, configure Registration Mode as Per Endpoint if the station needs to register with IC. ***See diagram step 16.**

19. For an FXS, configure Servicing IP Group ID to be the IP Group of the IC Server(s). ***See diagram step 16.**

20. To route calls to the FXS, enter records to match the Destination Phone number.

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID
1	*	*	210[1-2]#	*	*	3
2	*	*	*	*	*	2
3	*	*	*	*	*	1

21. Click **Submit**

SBC Routing

Now that the SRDs & IP Groups have been configured we can move on to configuring routing between IP Groups. For more details see the Configuring **SBC IP-to-IP Routing** section of the **Mediant 800 MSBG SIP User's Manual**.

1. Navigate to **VoIP** > **SBC** > **General Settings**

2. Enter the WAN IP address configured earlier in the **WAN IP Address** field

3. Click **Submit**

4. Navigate to **VoIP** > **SBC** > **Routing SBC** > **IP to IP Routing Table**

5. Enter a 10 in the text box next to Add. Click **Add**
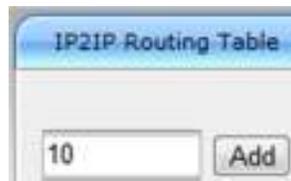
6. Enter the following settings:
   Source IP Group ID: 1
   Destination IP Group ID: 11

Index	Source IP Group ID	Source Username Prefix	Source Host	Destination Username Prefix	Destination Host	RequestType	Destination Type
10	1	*	*	*	*	All	IP Group

7. Click **Apply**

8. Enter 20 in the text box next to Add. Click **Add**
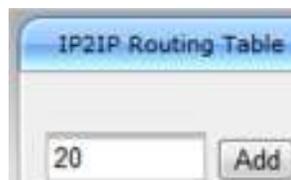
9. Enter the following settings:
   Source IP Group ID: 11

Destination IP Group ID: 1

Index	Source IP Group ID	Source Username Prefix	Source Host	Destination Username Prefix	Destination Host	RequestType	Destination Type
10	1	*	*	*	*	All	IP Group
20	11	*	*	*	*	All	IP Group

10. Click **Apply**

11. Select **Device Actions** > **Reset**

12. Ensure **Burn to Flash** is set to **Yes** and then click **Reset**
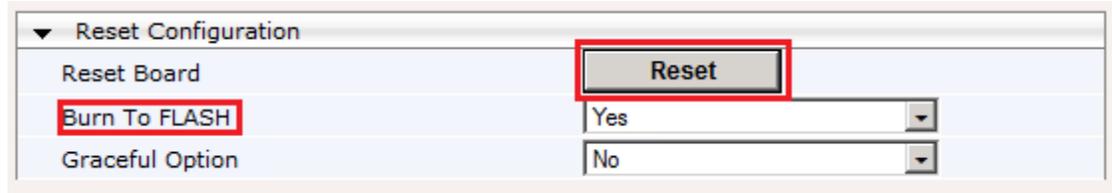
Advanced Configuration

TLS

Please note:

- If your traffic is moving between TLS & Non-TLS networks it is very important to create two IP Profiles, one for TCP/UDP and the other for TLS.

4. Navigate to **System** > **Certificates**
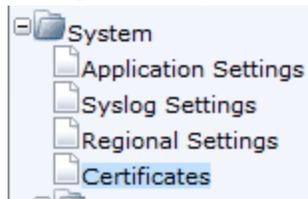
5. Enter the FQDN of the Mediant 800 gateway in **Subject Name**. Click **Generate CSR**

Note: It is necessary to use a **Subject Name** that is resolvable by all network elements, both Internal and External.

6. Copy all text from **-----BEGIN CERTIFICATE REQUEST----- to -----END CERTIFICATE REQUEST-----** including the section headers.

7. Open **Interaction Administrator** and navigate to the
    System Configuration container:
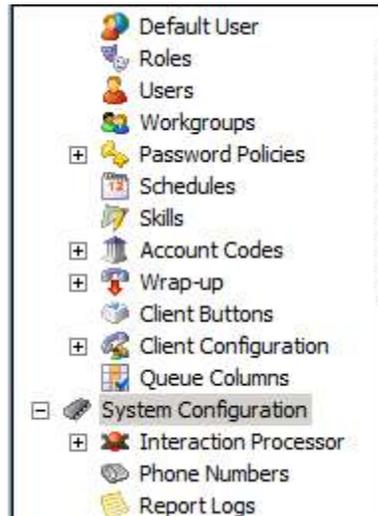
8. On right hand side double-click on **Configuration**
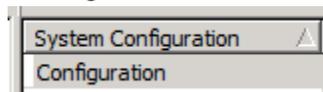
9. Open the **Certificate Management** tab and click the
    Modify… button for the **SIP/TLS Line Certificates**

Configuration
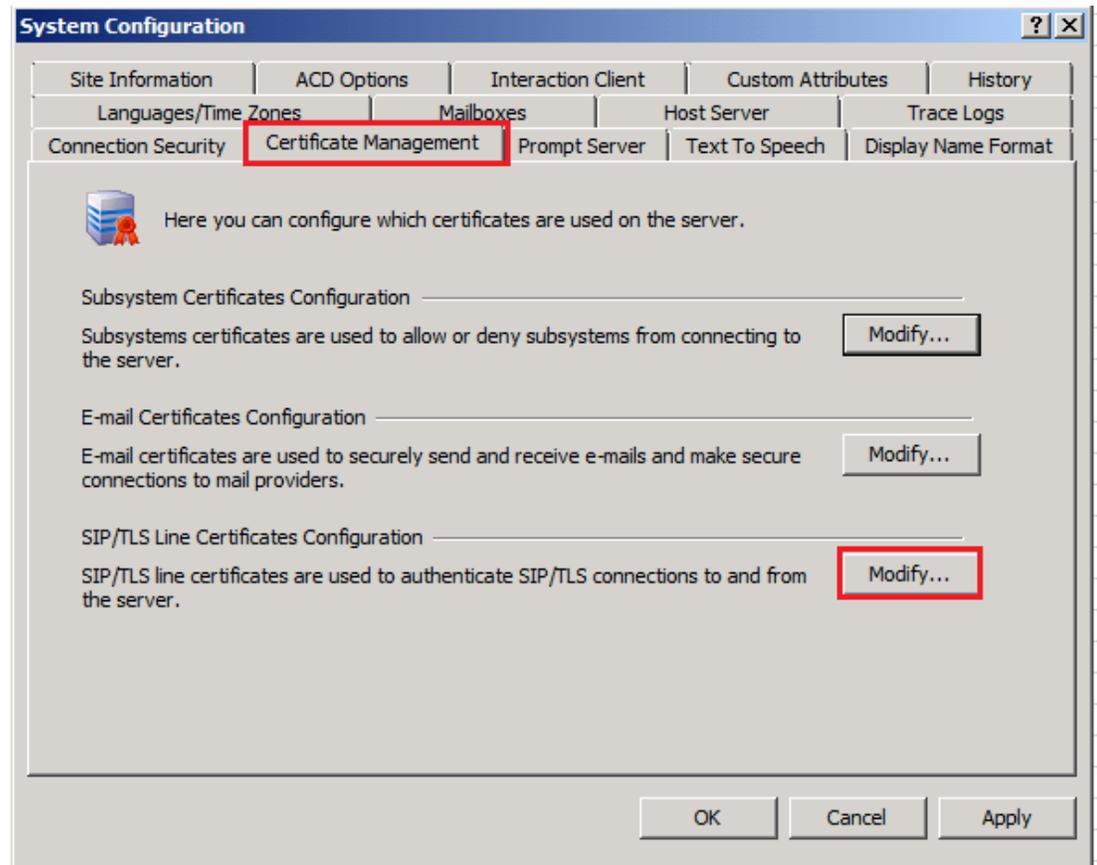
10. Click the **Port-To-Certificate Mappings** tab.
11. Remove all entries in the **Port-To-Certificate Mappings** box
12. Enter 5061 in the **Port** input box and choose **<Default Line Certificate>** in the **Line Certificate** drop down.

13. Click **Add** and then do the same for **8061**.

15. Click **Third Party Certificate Signing**
16. Paste the CSR generated at step 3 into the **Certificate to Sign** input box.
17. Click **Sign**

18. You will need to save both the **Signed** and **Signing Authority** certificates for importing to the Mediant 800.
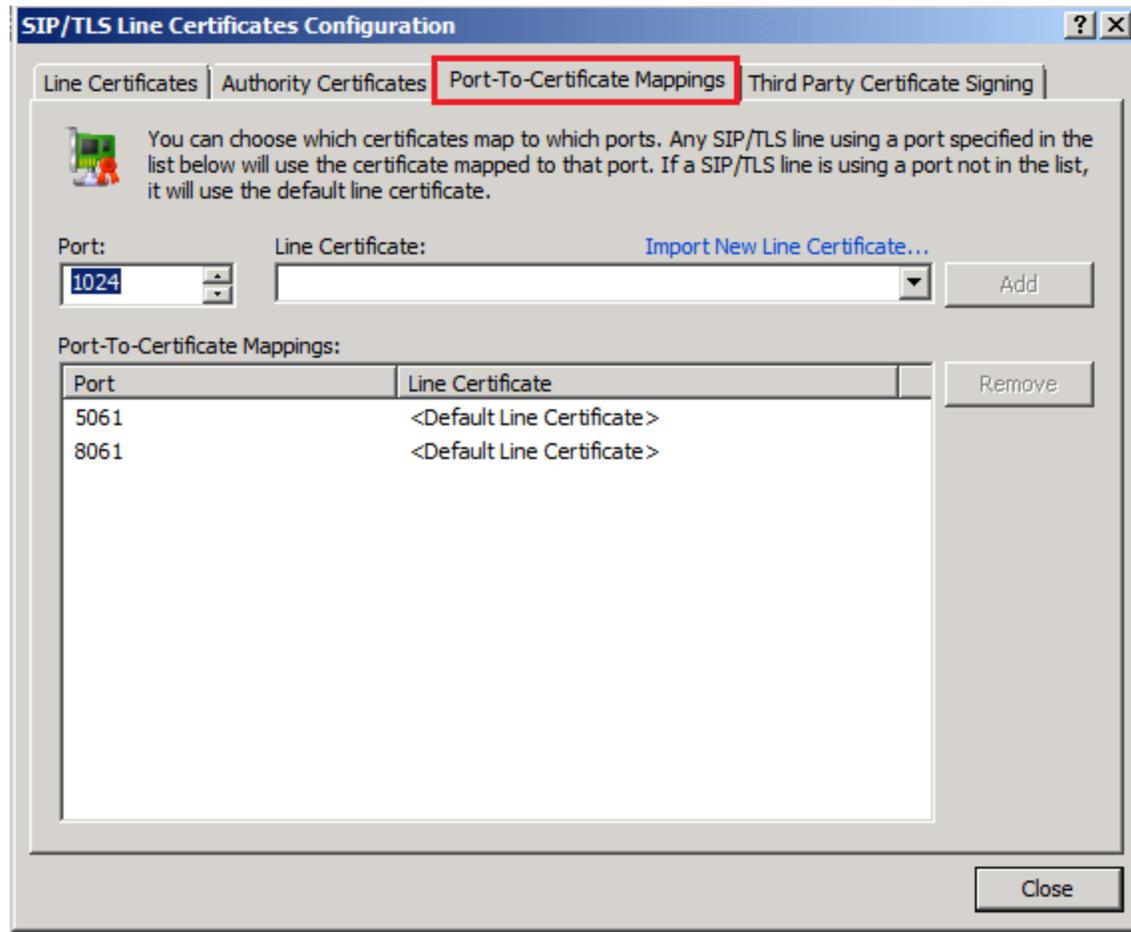
19. Click **Save As…** and save both certificates then return to the Mediant 800 Web Interface.

20. Return to **System** > **Certificates** and click **Browse…** under the **Trusted Root Certificate Store** section of **Certificate Files**. Once you've selected the **Signing Authority Certificate** you saved from Interaction Administrator, click **Send File**

21. Once the file has been uploaded, do the same for **Device Certificate**. This is the **Signed Certificate**.

Note: It is exceedingly important that you upload the certificates in the specified order.

22. Once the file has been uploaded go to **VoIP** > **Media** > **Media Security**
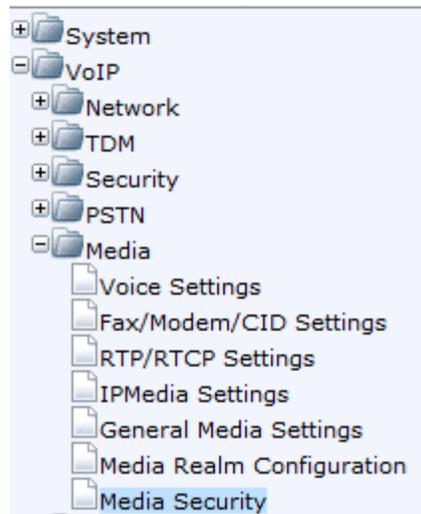
23. Change **Media Security** to **Enable** and put a check next to all applicable cipher suites in **SRTP offered Suites**

24. Click **Submit**

25. Click OK on the following prompt notifying you of the offline parameter change.

26. Navigate to **VoIP** > **SIP Definitions** > **General Parameters**
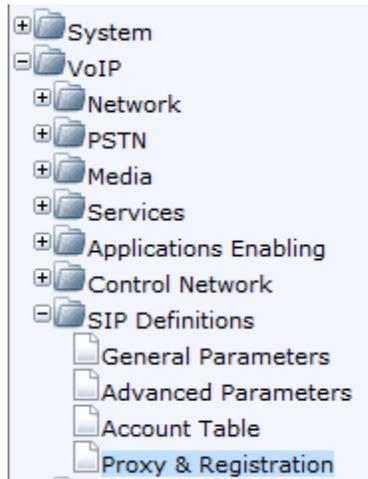

27. Change the following settings:

28. Click **Submit**

Advanced Configuration - TLS

29. Navigate to **VoIP** > **SIP Definitions** > **Proxy & Registration**

30. Enter the **Subject Name** used in generating the **CSR** in the **Gateway Name** & **Proxy Name** fields

31. Click **Submit**

32. Change your **Proxy Set** to use **TLS**
33. Select **Device Actions** > **Reset**

34. Ensure **Burn to Flash** is set to **Yes** and then click **Reset**

35. Log back in after reboot
36. Navigate to **VoIP** > **Control Network** > **Proxy Sets Table**
37. Ensure that the proper proxy set has the **Transport Type** set to **TLS** for each entry in its **Proxy Address** table. Also put the **SIPS** communication port on the end of each IP Address.
38. If one of your proxy sets does not communicate using TLS ensure that that **Proxy Set** has the **SIP** communication port on the end of each of its **IP Addresses**.

39. Navigate to **VoIP** > **Coders & Profiles** > **IP Profile
    Settings**

40. Create one **IP Profile** for **TLS** and one for **TCP/UDP**.
    a. For **TCP/UDP** set:
       Gateway Parameters > **Media Security Behavior**
       to **Disable**
       SBC > **Media Security Behavior** to **RTP**
    b. For **TLS** set:
       Gateway Parameters > **Media Security Behavior**
       to **Preferrable**
       SBC > **Media Security Behavior** to **SRTP**

41. Assign the **IP Profiles** to **IP Groups** as needed to control
   their **Media Security** behavior.

42. Click **Submit**

43. Click **Burn**

External User Agents

If you wish to have VoIP stations outside your network that connect back to IC and place calls without using VPN, there are several things you will need to configure. Primarily you cannot use the Mediant 800 for your primary Data router. You will instead need a Secondary router that will handle the HTTP(S) traffic from the WAN Phones. See the diagram below:
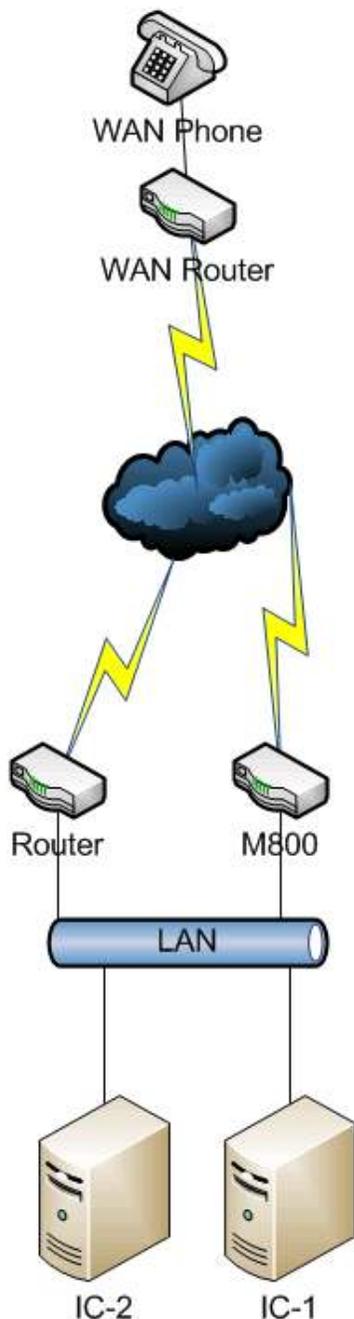
For IC to function properly with a WAN Phone & Switchover, all HTTP(S) traffic will need to be routed through a separate device. Ports 8088 and 8089 will need to be sent through your firewall using port forwarding, static NATs, or a separate method of your choosing.

The end result being that a WAN phone can contact either server for its configuration files, depending on which server is Active, and that the HTTP(S) traffic does not go through the M800.

The reason for this is due to a limitation of the Mediant 800. If a DNS record returns more than one IP, the first IP returned is the only IP Address used.

One other thing to note is that this configuration disallows provisioning managed phones by end users, and rebooting the phones remotely. All phones will need their MAC addresses entered at the time of creation.

There will need to be several changes in Interaction Administrator before changes are made to the Mediant 800. There will need to be:

- A line for phones to register on, either 5060 or 5061 depending on the usage of TLS
- A registration group dedicated to WAN phones
- A location dedicated to WAN phones

1. Open **Interaction Administrator** and navigate to the **Lines** Container
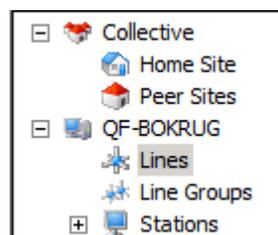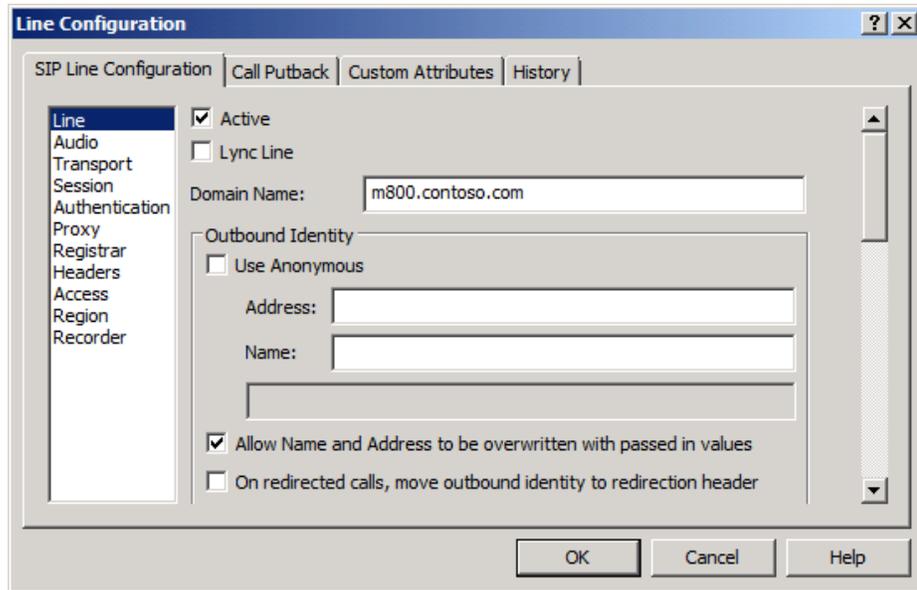

2. Click **File** > **New** and enter a meaningful name for the Line
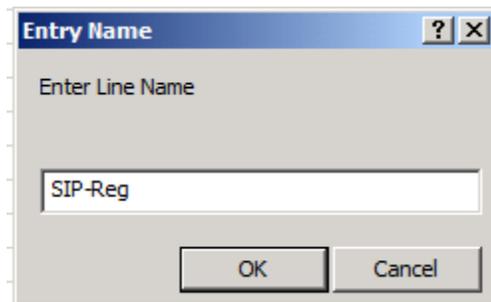
3. Enter the FQDN the WAN Phones will use to contact IC in the **Domain Name** field. Enter something in the **Address**

field. This can be your main DID or another number.

4. Change any settings necessary in the **Audio** section
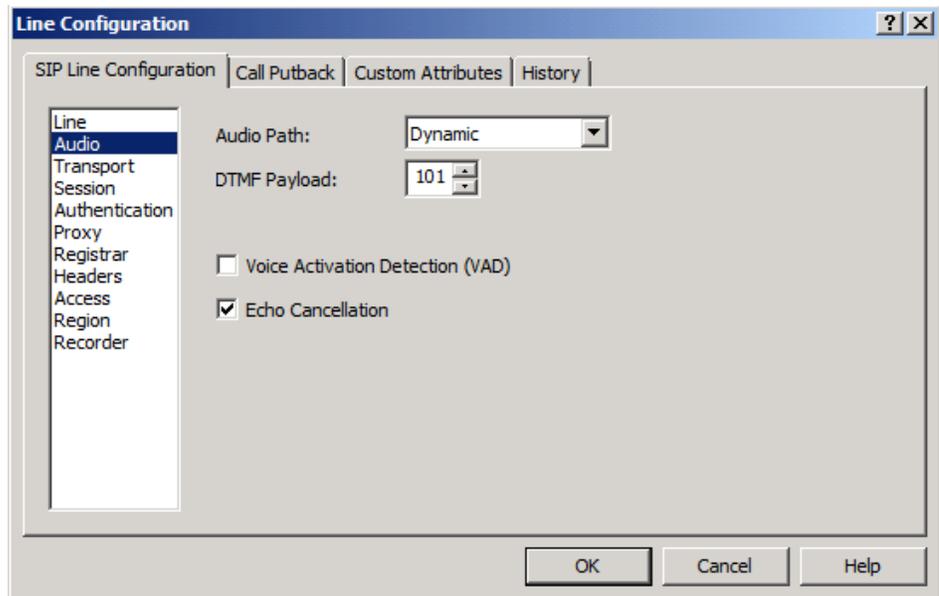
5. Change any settings necessary in the **Transport** section

6. Click **OK**
7. Click **[Server Name]** > **Managed IP Phones** > **Registration Groups**
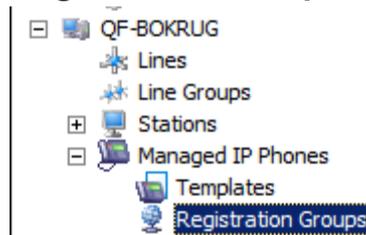
8. Click **File** > **New** and enter a name in the **New Registration Group** window

9. Click **Add..** in the **Registrations** section of the **Registration Group Configuration** window

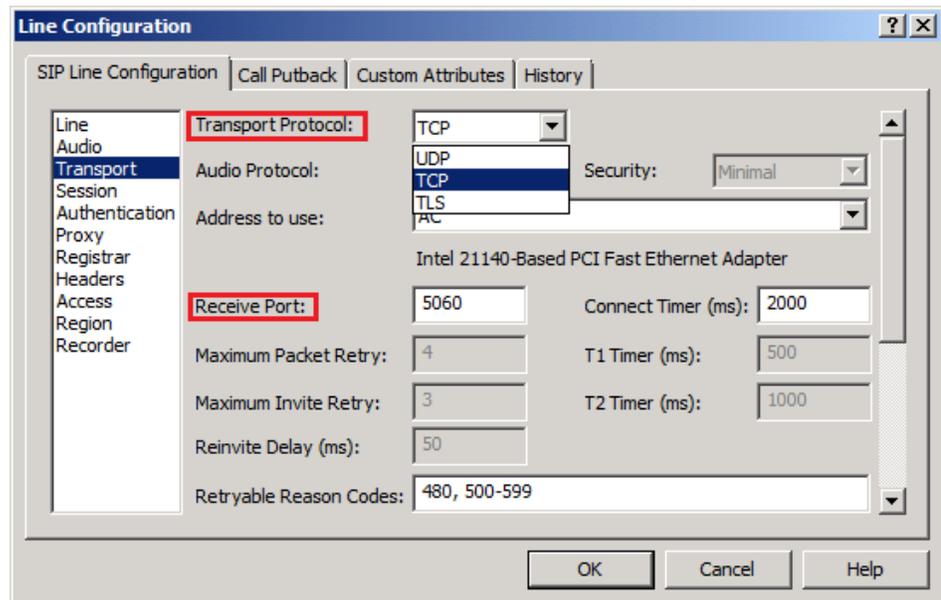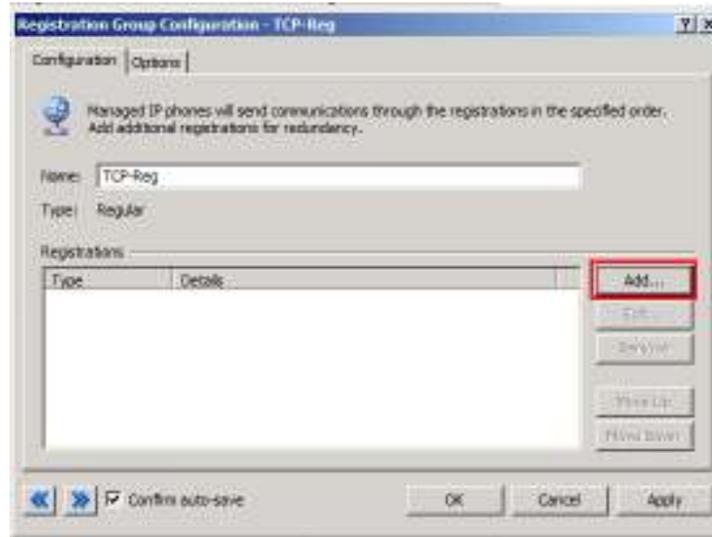10. Click the **Use the following registration settings:** radio button.
11. Enter the external FQDN of your Mediant 800, the name entered in the **Subject Name** of the CSR, and change **Port** and **Transport Protocol** as necessary.
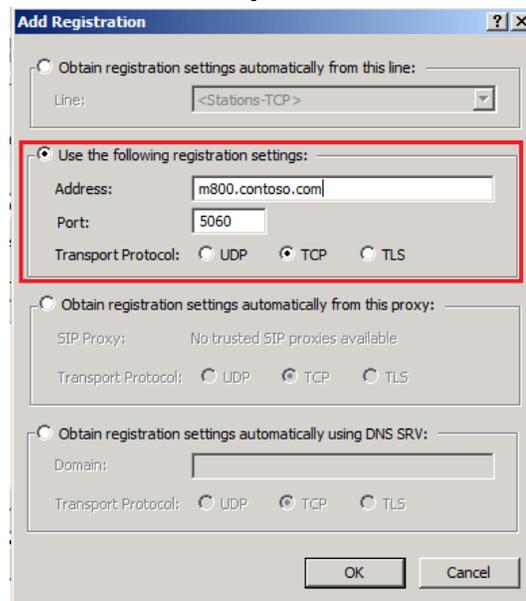
12. Click **OK**

13. Click **[Server Name]** > **Regionalization** > **Locations**

> Collective
>   Home Site
>   Peer Sites
> QF-BOKRUG
>   Lines
>   Line Groups
>   Stations
>   Managed IP Phones
>   Audio Sources
>   Server Parameters
>   Structured Parameters
>   Regionalization
>   Locations
>   Default Regionalizatic

14. Click **File** > **New**
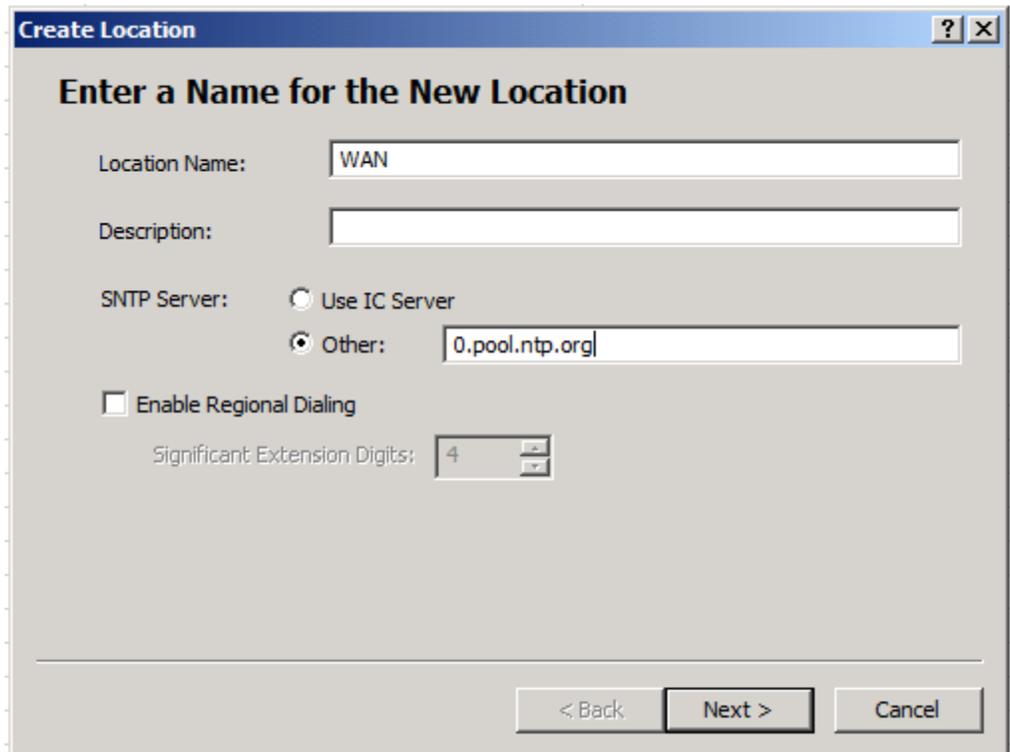
15. Enter a meaningful **Location Name** and, if there is no way to route NTP back to your network from the WAN, choose **Other** for **SNTP Server** and enter the IP or FQDN of a public NTP server.

Create Location   ? X

Enter a Name for the New Location

Location Name: WAN

Description:

SNTP Server: ○ Use IC Server
                  ● Other: 0.pool.ntp.org

□ Enable Regional Dialing

   Significant Extension Digits: 4

                                    < Back   Next >   Cancel

16. Click **Next**

17. Select the other **Location**(**s**) this **Location** needs to be able to communicate with

Create Location

Select Location Communication Mappings
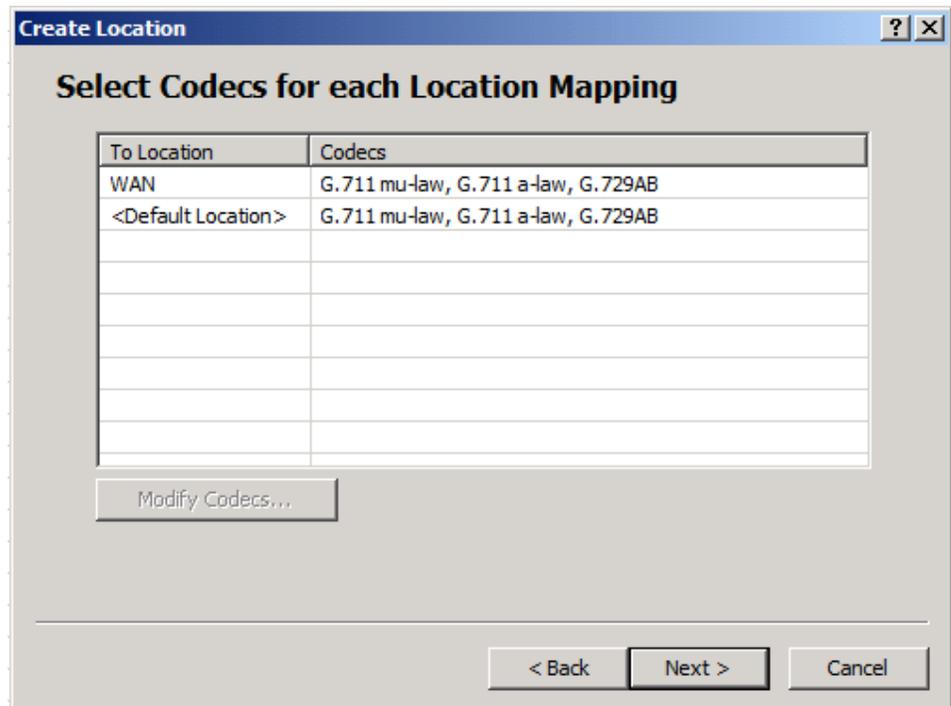
Select locations in which devices are allowed to communicate with devices in this location:

☑ WAN - This Location
☑ <Default Location>

< Back   Next >   Cancel
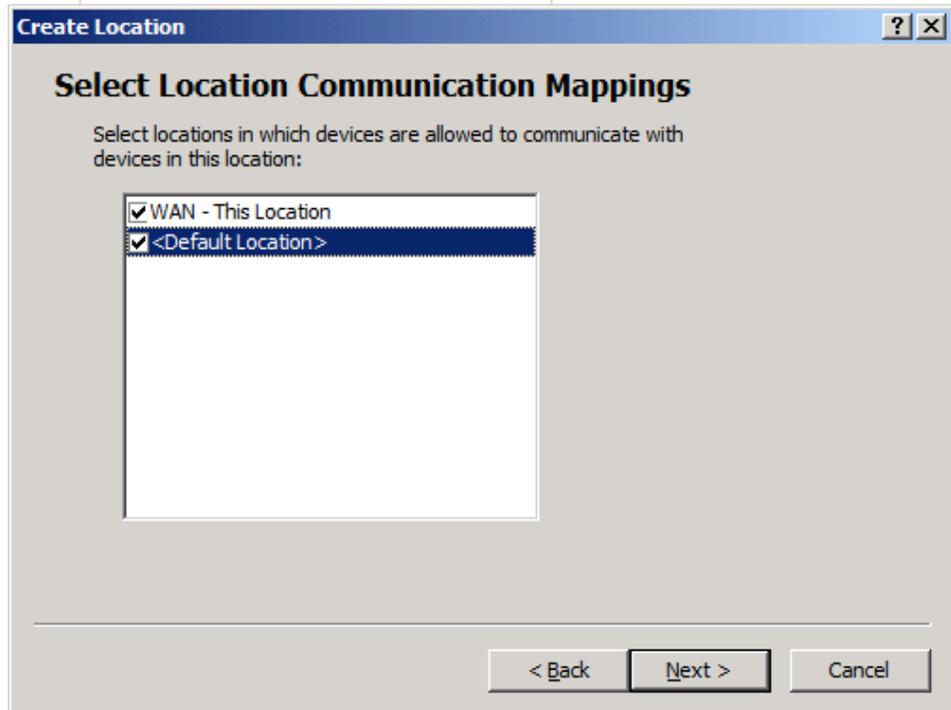
18. Click **Next**
19. Choose the **Codecs** available at each **Location**

Create Location

Select Codecs for each Location Mapping

To Location	Codecs
WAN	G.711 mu-law, G.711 a-law, G.729AB
<Default Location>	G.711 mu-law, G.711 a-law, G.729AB

Modify Codecs...

< Back   Next >   Cancel

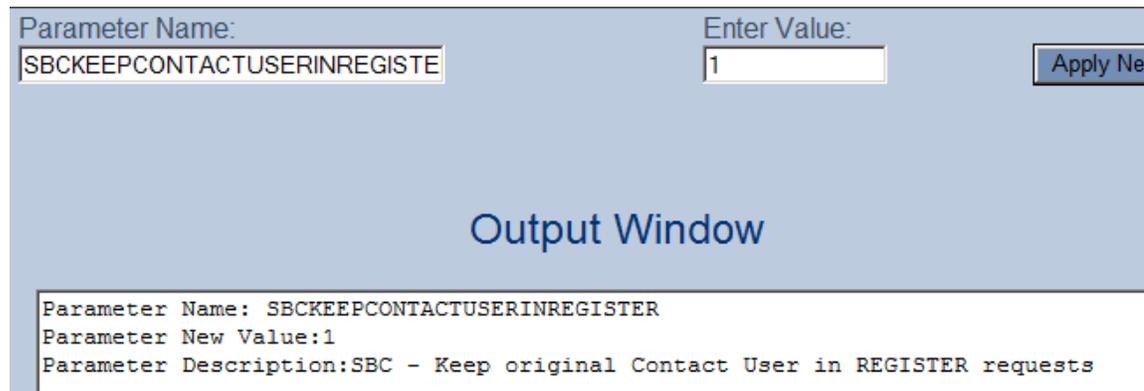20. Click **Next**
21. Click **Finish**

22. Create some Managed Phones/Stations and place them in the WAN region.
23. Open the Mediant 800's WEB Interface
24. Change the URL replacing **Index** with **AdminPage** (this is case sensitive)

http://192.168.1.1/AdminPage
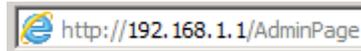
25. Select *ini* **Parameters**

Image
Load to
Device

ini
Parameters

Back to
Main

26. Enter **SBCKeepContactUserInRegister** in the **Parameter Name** field and **1** in the **Enter Value** field, click **Apply New Value**

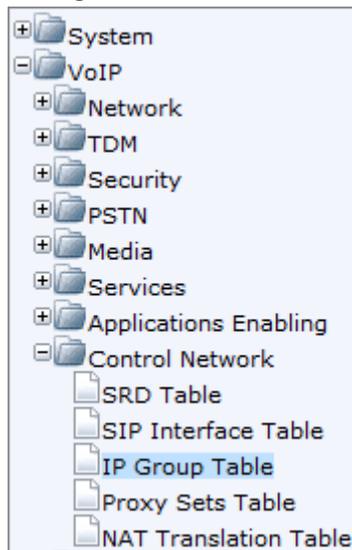Parameter Name:
SBCKEEPCONTACTUSERINREGISTE

Enter Value:
1

Apply Ne

Output Window

```
Parameter Name: SBCKEEPCONTACTUSERINREGISTER
Parameter New Value:1
Parameter Description:SBC - Keep original Contact User in REGISTER requests
```

27. Click **Back to Main** to return to the regular configuration page.

Image
Load to
Device

ini
Parameters

Back to
Main

28. Navigate to **VoIP** > **Control Network** > **IP Group Table**

System
VoIP
  Network
  TDM
  Security
  PSTN
  Media
  Services
  Applications Enabling
  Control Network
    SRD Table
    SIP Interface Table
    IP Group Table
    Proxy Sets Table
    NAT Translation Table

29. Select **Index 12** and enter the following settings:

> **Type**: USER
> **Description**: WAN Users (or something more meaningful)
> **SRD**: 11
> **Media Realm**: WAN
> **IP Profile ID**: Choose the appropriate IP Profile unless none used

Index	12

Common Parameters

Type	USER
Description	WAN_Users
Proxy Set ID	
SIP Group Name	
Contact User	N/A
⚡ SRD	11
⚡ Media Realm	WAN
IP Profile ID	0

Gateway Parameters

Always Use Route Table	No
Routing Mode	Not Configured
SIP Re-Routing Mode	Standard

SBC Parameters

Classify By Proxy Set	Enable
Max Number Of Registered Users	-1
Inbound Message Manipulation Set	-1
Outbound Message Manipulation Set	-1

30. Click **Submit**

31. Navigate to **VoIP** > **SBC** > **Routing SBC** > **Classification Table**

32. Enter 12 in the text box next to **Add** and click **Add**

33. Enter
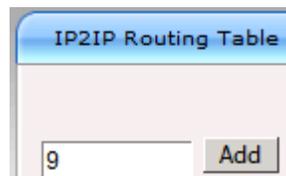   a. **Source SRD ID**: 11
    b. **Source IP Group ID**: 12

Index	Source SRD ID	Source IP Address	Source Username Prefix	Source Host Prefix	Destination Username Prefix	Destination Host Pr
12	11		-	-	-	-

34. Click **Apply**

35. Navigate to **VoIP** > **SBC** > **Routing SBC** > **IP to IP Routing Table**

36. Enter a **9** in the box next to **Add**

37. Enter the following Settings:

> **Source IP Group ID**: 1
>
> **Destination Username Prefix**: WANPhoneXXX
>
> > i. A strict naming convention will have to adopted for all phones that are going to be on the WAN so that this naming mask will be effective.
> >
> > ii. This naming mask will handle any phone named WANPHONE(001-999) as X is wild card for a single digit, 0-9.
>
> **Destination IP Group:** 12

Index	Source IP Group ID	Source Username Prefix	Source Host	Destination Username Prefix	Destination Host	RequestType	Destination Type
9	1	*	*	WANPhoneXXX	*	All	IP Group
10	1	*	*	*	*	All	IP Group
20	11	*	*	*	*	All	IP Group

38. Click **Apply**

39. Enter a **19** in the box next to **Add**

40. Enter the following Settings:
    a. **Source IP Group ID**: 12
    b. **Destination IP Group**: 1

41. Click **Apply**
42. Select **Device Actions** > **Reset**

43. Ensure **Burn to Flash** is set to **Yes** and then click **Reset**