NetVanta Unified Communications Technical Note

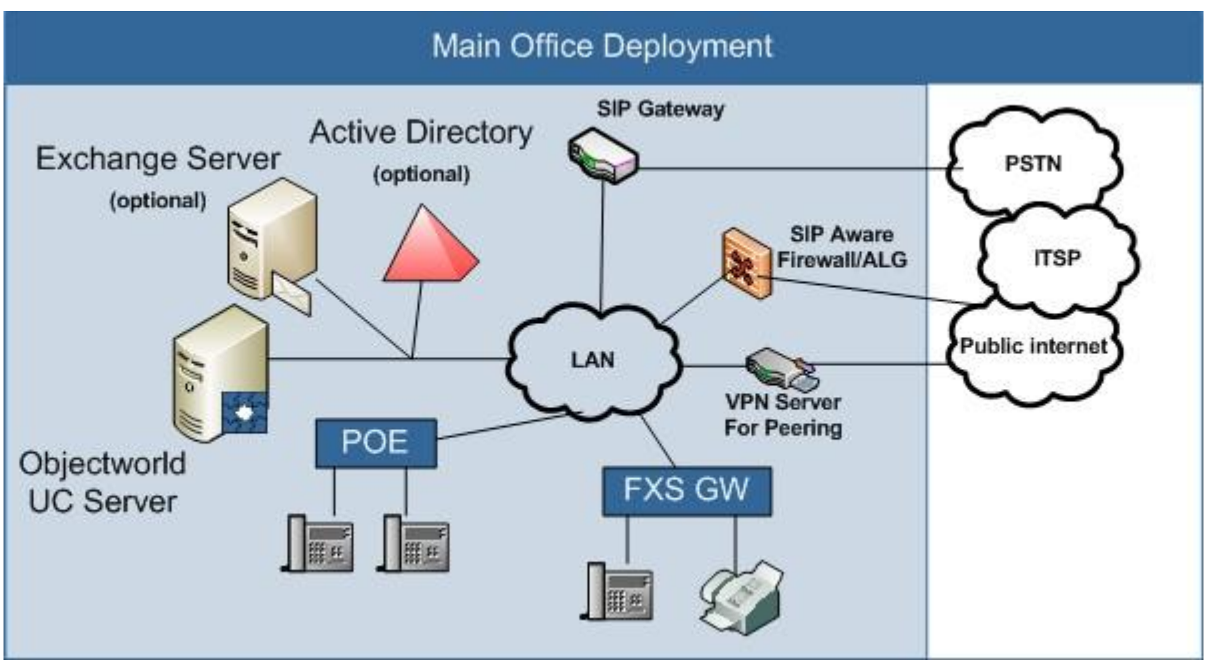
Installing and Configuring AudioCodes Mediant 1000

1 Introduction

The AudioCodes Mediant 1000 (M1K) is a T1 digital gateway that can be used in NetVanta Unified Communications Server installations to provide a bridge between internal (SIP) phone calls and the phone network (PSTN). The M1K supports 1, 2 or 4 T1 spans. It bridges SIP VoIP phones on the Local Area Network (LAN) and the traditional TDM voice network (PSTN).

A gateway works in conjunction with the SIP Proxy and Registrar that are part of the UC server. All telephony services are provided through the mutual co-operation of the SIP Gateway, SIP Telephones, SIP Proxy and the Core Application Services.

The following diagram illustrates the UC server SIP Architecture and its relationship with other components in a typical customer network.



This document is a quick start guide to integrating the gateway with UC Serer. It is not a replacement for the manufacturer's documentation, and specific customer requirements might fall outside the guidelines of this document.

It is recommended that skilled technicians obtain training directly from the manufacturer for gateway configuration and deployment.

2 Overview of Procedure

To provide its functionality, the M1K must be connected to the internal LAN (a 100 Mbps connection is recommended) and 1, 2 or 4 T1 digital spans.

The M1K has two configuration methods: Web page or configuration file. This document describes configuration using the gateway's Web page. This has the advantage of providing familiarity with the We page method, which will most likely be used later for making minor changes in the gateway configuration. It has the disadvantage that an inadvertent change in one of the many parameters might cause problems. **For this reason it is recommended that only the changes described below are made to a factory reset gateway**. After the gateway is operational, further changes may be made as required.

The basic steps for installation and configuration are:

1. Unpack the M1K.
2. Mount the M1K.
3. Connect cables.
4. Set the IP address and subnet mask of the gateway through the serial port on M1K.
5. Access the M1K Web page.
6. Configure the gateway.
7. Save the configuration and reboot the M1K.
8. Back up the configuration.

Steps 1-3 are standard for any gateway. Please follow the instructions provided by AudioCodes for the gateway. AudioCodes provides comprehensive information on the use and configuration of the gateway. This document is intended to be a companion document to the AudioCodes documents, which should be used while configuring the gateway.

The rest of this section details Steps 4 and 8 to configure the M1K for operation with the UC server.

3 Configuring the Gateway

3.1 Software Level

This document is based on the gateway software from the last interoperability testing conducted by ADTRAN. If the software revision does not match, contact the manufacturer's website for the version that is shown below. Failure to do so might result in unexpected behavior.

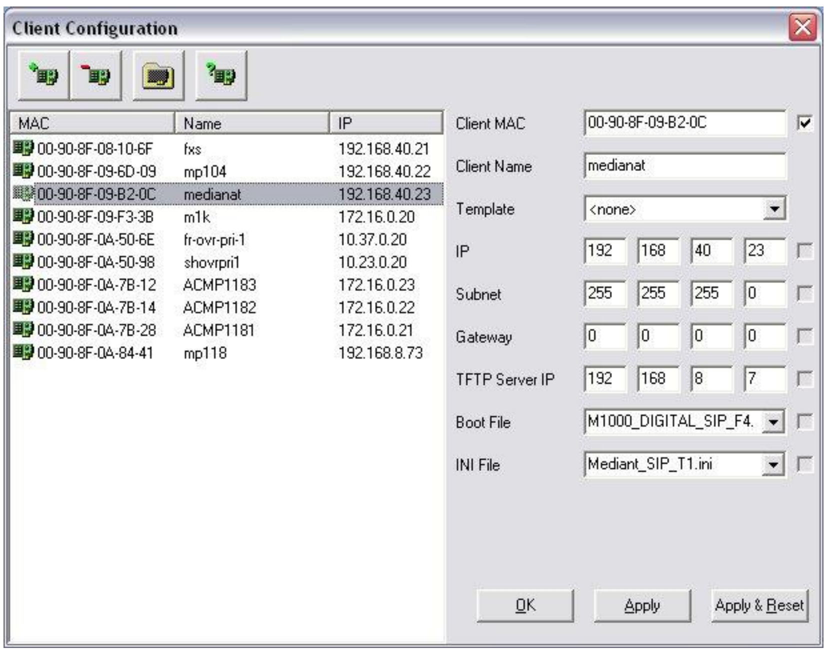
To confirm the software revision for your gateway, navigate to **Status and Diagnostics** -> **Device Information** and make sure that the gateway software matches the information below before proceeding.

Versions Version ID: 5.20A.047.003

3.2 Setting the IP Address and Subnet Mask

It is important that the M1K have a LAN IP address that does not change. This can be set using a static IP address and subnet mask compatible with the on-site LAN. Please follow the instructions provided by AudioCodes. A summary of the procedure is:

1. Connect the M1K and a PC on the same LAN subnet.
2. Use or install from the AudioCodes CD a bootp application.
3. Power up the M1K.
4. Edit the IP address, subnet mask, and gateway information in the bootp information window. See Figure 1 below.



5. Now access the gateway's Web page using your browser and the M1K's new IP address. The default login information is the following:

   Username: **Admin**

   Password: **Admin**

3.3 Configuring SIP

When the gateway receives a call from the PSTN it must know where to send that call. In the reverse direction, the gateway must accept SIP calls from the UC server and direct those calls out to the PSTN. This is generally configured by providing SIP and Dial Plan configuration information.

The UC server IP address or name within the enterprise domain is the SIP local domain for the gateway. Standard UC server configurations for gateways do not require that the gateway register on the UC server as a SIP identity.

Calls between PSTN devices and services on the UC server may make use of DTMF tones, for example, voice mail and auto attendant functions. The DTMF digits must be transported outside the voice stream to the UC server. This is done by enabling DTMF Transport using rfc2833.

Similarly, faxes that are sent or received by the UC server must be supported by transmitting the fax information outside the TDM voice path. This is implemented using T.38 fax support. This must be enabled on the gateway by doing the following:

1. Navigate to **Quick Setup**.
2. Leave the Gateway Name blank.
3. Make sure that **Working with Proxy** is set to Yes.
4. In **Proxy IP Address** enter the IP address of the UC server.
5. Leave the **Proxy Name** blank.
6. Click **Reset** and when prompted save the information.

See Figure 2 below.

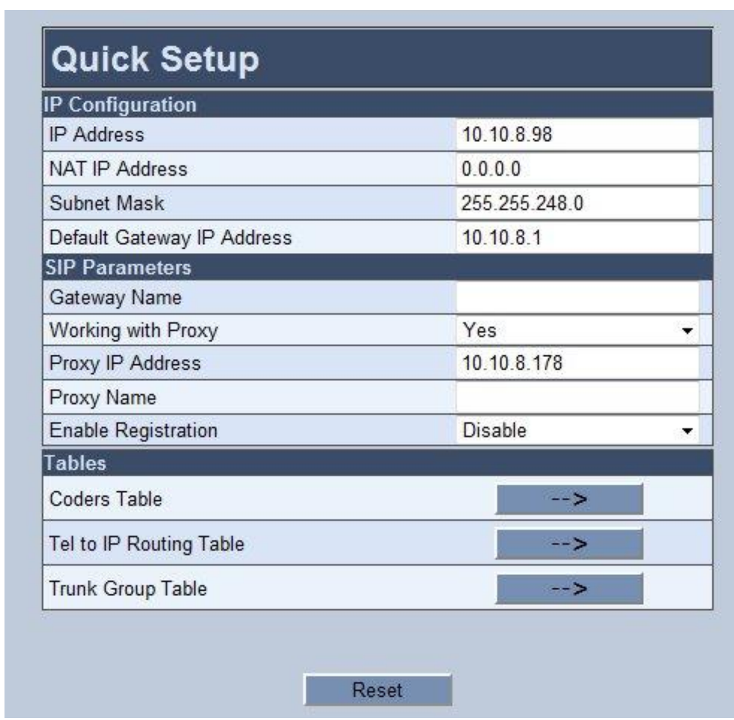


Figure 2

7. Go to **Protocol Management -> Routing Tables -> Tel to IP Routing**.

8. Enter a row that has **Dest. IP Address** set to the the UC server IP address.
9. Click **Submit**.

See Figure 3 below.

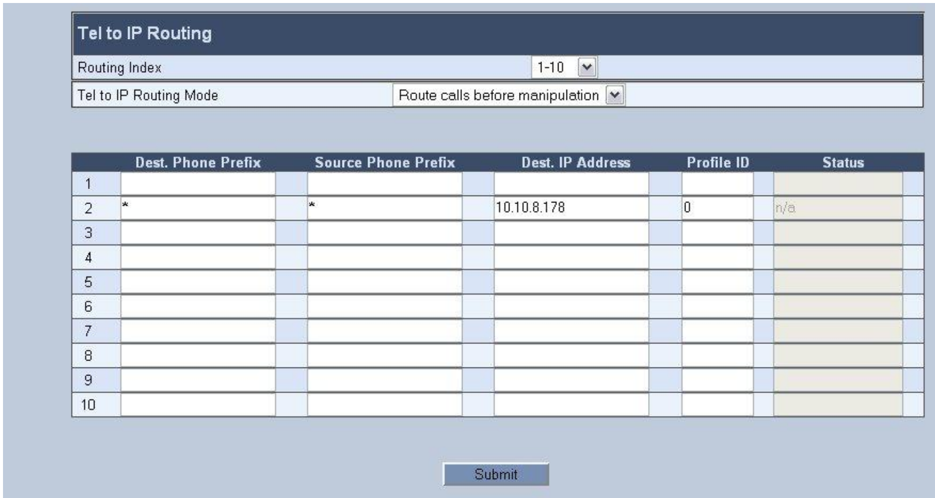


Figure 3

10. Go to **Advanced Configuration -> Media Settings -> Voice Settings** and make sure that **DTMF Transport Type** is set to RFC2833 Relay DTMF. See Figure 4 below.

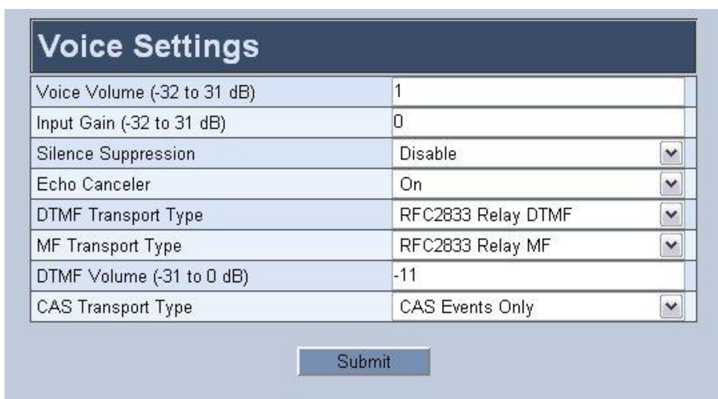


Figure 4

11. Go to **Advanced Configuration -> Media Settings -> Fax/Modem/CID Settings** and make sure that:
    - **Fax Transport Mode** is T.38 Relay.
    - **V.21 Modem Transport Type** is set to Disable.

See **Figure 5** below.

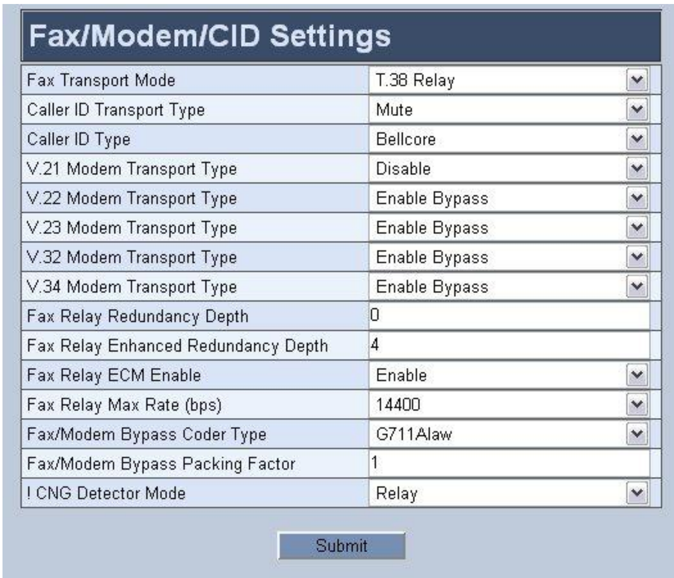


Figure 5

12. Go to **Protocol Management -> Protocol Definition -> Coders** and make sure that G.711U-law and G.729 codecs are chosen (for North America).

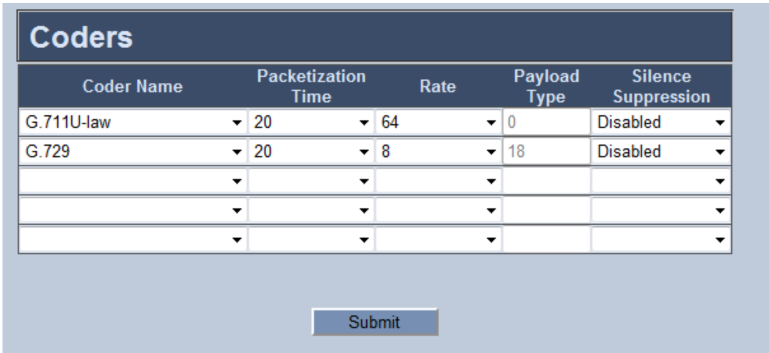


Figure 6

13. Go to **Protocol Management -> Protocol Definition -> General** and ensure that **Enable Early Media** is set to **Enable.**

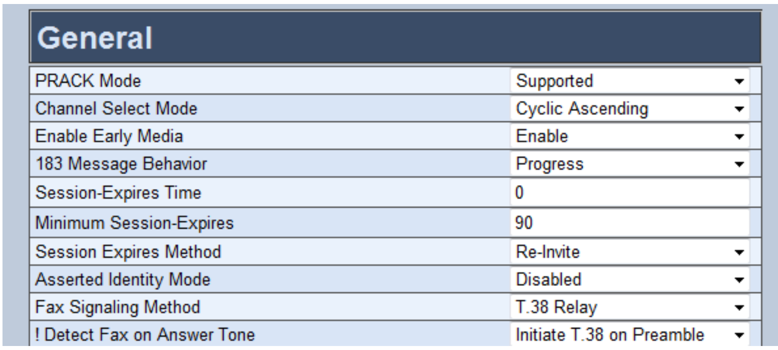


Figure 7

3.4 Dial Plan

Routing or dial plan entries must be configured to route calls in from the PSTN to the UC server, and out from the UC server to the PSTN.

3.4.1 Incoming

There must also be an answering point defined on the UC server for incoming PSTN calls. A standard configuration has a UC server trunk identity of 10000. Enter a general routing entry on the gateway so that all incoming calls are sent to 10000 on the UC server.

When the call is presented from the PSTN, the carrier provides the called party information. Often this is not the full 10-digit number, but is the last 4 digits of the number being called. In either case that number may be used to route the call to a destination.

Go to **Protocol Management -> Manipulation Tables -> Tel -> IP Destination Numbers** and:

1. Add an entry with:
    - **Destination Prefix** set to *.
    - **Source Prefix** set to *.
    - **Number of stripped Digits** set to the number of incoming digits given by the carrier.
    - **Prefix (Suffix) to Add** set to the identity that should answer the call (for example, 10000).
2. Click **Submit.**

See **Figure 8** below.

3.4.2 Incoming DID Call Routing

The entry routes all incoming PSTN calls to identity 10000 no matter what number is called. If there are multiple incoming DIDs, more routing entries must be added.

	Destination Prefix	Source Prefix	Number of stripped Digits	Prefix (Suffix) to Add	Number of Digits to Leave
1	2180	*	4	707	
2	2171	*	4	222	
3	2172	*	4	235	
4	2173	*	4	228	
5	2174	*	4	299	
6	2175	*	4	333	
7	2170	*	4	2170	
8	2179	*	4	151000	
9	9698	*	4	10000	
10	2177	*	4	2177	

Submit

Figure 8 – in this example, the main DID is 9698 and it is routed to the trunk identity 10000. There are a number of other incoming DIDs. An example is 2180, which is routed to the 707 identity. The first rule that matches is the one that is used, so it is important to have any catch-all rules (for example, a destination of *) at the bottom of the tables.

3.4.3 Outgoing

A gateway may have more physical and logical trunks than are being used at a site. When an outgoing call is made to the PSTN it is important that the gateway not choose a trunk that is not there or not in service. Enable only those physical interfaces and trunks that are actually in use.

If an operational trunk is busy, the gateway must 'hunt' for an unoccupied trunk to complete the call. Configure a trunk group that specifies which trunks are to be used for outgoing calls and what the hunting algorithm should be, for example, ascending or descending.

1. Go to **Protocol Management -> Trunk Group**
2. Create a **Trunk Group ID** (for example, 1) that includes all trunks and channels used for outgoing calls.

The phone number assignment is arbitrary. A number should be assigned but in general will not be used for most installations.

Figure 9 – In this example there are two trunk groups, 1 and 2. Trunk group ID 1 is using the first T1 span and has channels 1-8 enabled. If a call is presented to that group, those channels are used.

3. Go to **Protocol Management -> Trunk Groups Settings**
4. Choose the type of outgoing trunk behavior required, for example, Descending.

Trunk Group Settings

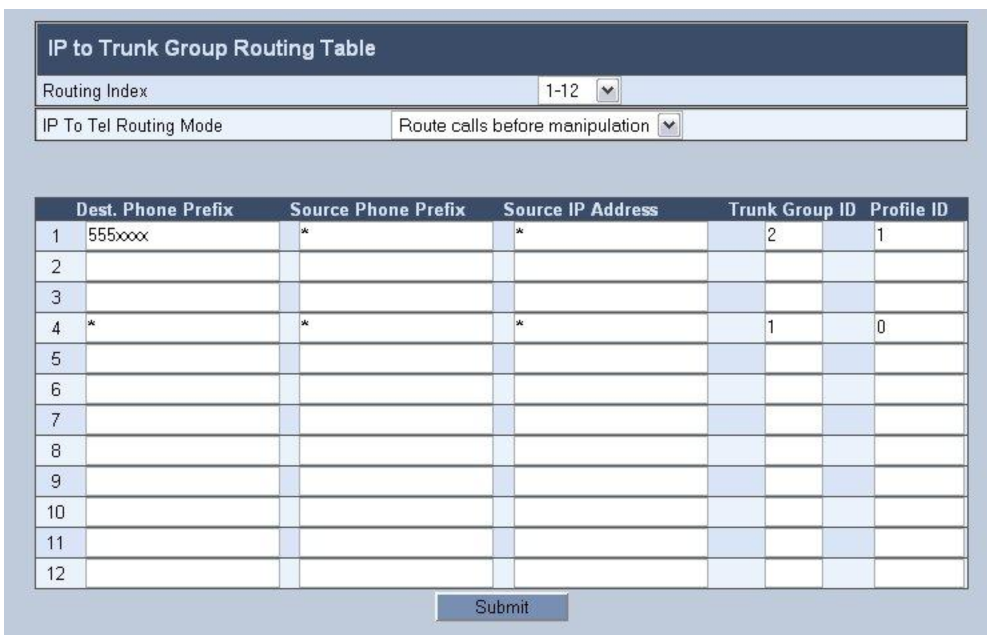
Routing Index: 1-12

	Trunk Group ID	Channel Select Mode
1	1	Descending
2		
3		
4		
5		
6		
7		
8		
9		
10		
11		
12		

Submit

Figure 10

5. Go to **Protocol Management -> Routing Tables -> IP to Hunt Group Routing.**
6. Add an entry that maps all outgoing calls to the trunk group defined.



IP to Trunk Group Routing Table

Routing Index: 1-12

IP To Tel Routing Mode: Route calls before manipulation

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID	Profile ID
1	555xxxx	*	*	2	1
2					
3					
4	*	*	*	1	0
5					
6					
7					
8					
9					
10					
11					
12					

Submit

Figure 11 – the last entry is the default route that takes any number and sends it to Trunk Group ID 1.

3.4.4 Outgoing DID Presentation

When an outgoing call to the PSTN is made over an analog trunk, the caller ID is generated by the carrier. There is no ability to change the ID on a call-by-call basis. A PRI allows the calling party to specify the caller ID.

1. Go to **Protocol Management -> Manipulation Tables -> IP -> Tel Source Numbers**.
2. If an enterprise-wide caller ID is required, add an entry that maps any source and destination number to the phone number of the enterprise.
3. If other outgoing phones are desired for some phone number, add table entries for those above the catch-all entry.

	Dest. Prefix	Source Prefix	Source IP	Number of Stripped Digits	Prefix (Suffix) to Add	Number of Digits to Leave	NPI	TON	Presentation
1							Not Configured	Not Configured	Not Configured
2	*	222x	*	4	6132712171		E.164 Public	National	Allowed
3	*	235x	*	4	6132712172		E.164 Public	National	Allowed
4	*	228x	*	4	6132712173		E.164 Public	National	Allowed
5	*	299x	*	4	6132712174		E.164 Public	National	Allowed
6	*	333x	*	4	6132712175		E.164 Public	National	Allowed
7	*	707x	*	4	6132712180		E.164 Public	National	Allowed
8	*	xxxxxxx	*	0			E.164 Public	National	Allowed
9	*	xxxxxxxxxx	*	0			E.164 Public	National	Allowed
10	*	*	*	4	6135999698		E.164 Public	National	Allowed

Submit

Figure 12 – The last entry is the catch-all that takes any destination and source and presents 6135999698 as the calling number. The second entry, as an example, means that anytime an extension 2220-2229 calls any number, the presented phone number is 6132712171.

3.5 DSL

The DSL / PRI settings are site- and carrier-specific. This section covers the most likely settings for North America.

1. Go to **Advanced Configuration -> PSTN Settings -> Trunk Settings**.
2. Select the trunk to configure by clicking the **Trunk Status** icon at the top of the page. If the trunk cannot be selected, it is because it is active. If that is the case, go to the bottom of the page and stop the trunk.

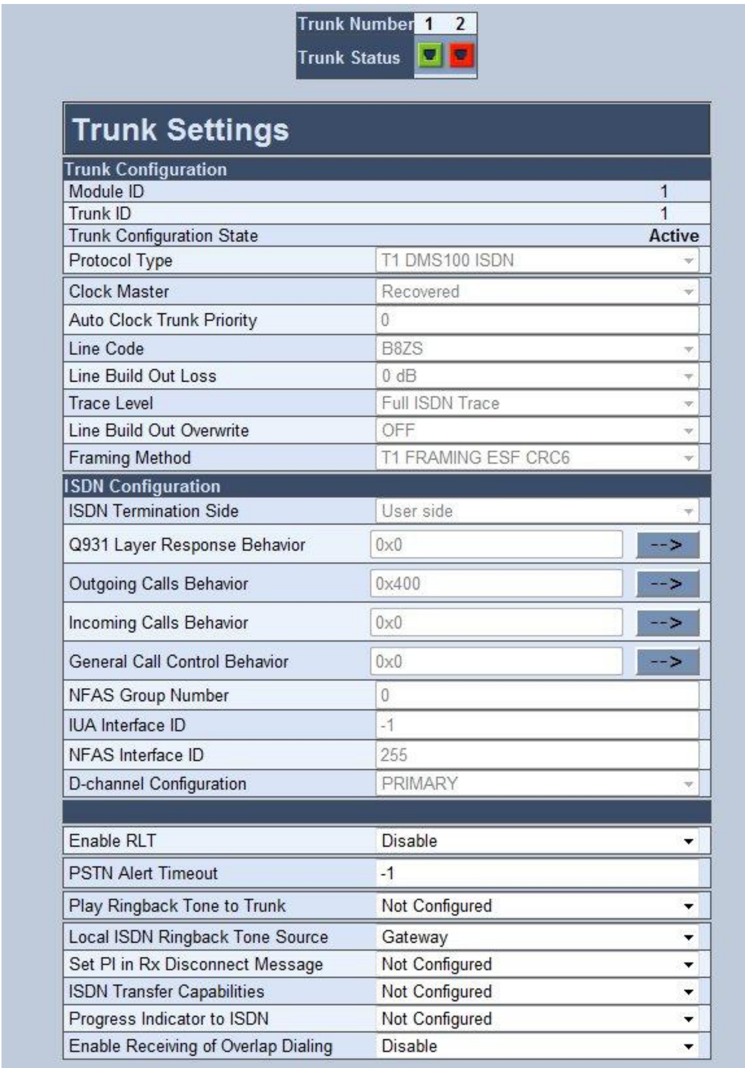


Figure 13

3. **Protocol Type** should be set as defined by your carrier. DMS, ATT and NI are the most common settings.
4. **Clock Master** should be set as defined by your carrier. Recovered is the most common.
5. **Line Code** should be set as defined by your carrier. B8ZS is the most common.
6. **Framing Method** should be set as defined by your carrier. T1 Framing ESF CRC6 is the most common.
7. **ISDN Termination Side** should be set as defined by your carrier. User side is the most common.
8. **Local ISDN Ringback Tone Source** should be set as defined by your carrier. Gateway is the most common.
9. **Apply Trunk Settings.**
10. Click **Submit.**

3.6 Backing up the Configuration

1. Go to **Maintenance**. Click **Burn** under **Save Configuration** to save all the changes.

2. Go to **Advanced Configuration -> Configuration File** and get the .ini file. Store the configuration file in a safe place.

4 Configuring the UC Server

After the gateway has been added to your network, the UC server must be configured to handle incoming and outgoing phone calls.

These instructions are for Release 4.1 of the UC server. Start the UC server admin client.

4.1 Adding a Trunk Identity

1. Go to **Identities**.
2. Right-click the right panel and select **New Identity**.
3. In the first page of the Wizard, select an **Attendant** identity. Make sure that the Identity is associated with the Admin profile.
4. On the following page, enter a descriptive name and enter 10000 for the **Address** (assuming a standard configuration). Make sure that **Default Trunk Service** is the service to be run.

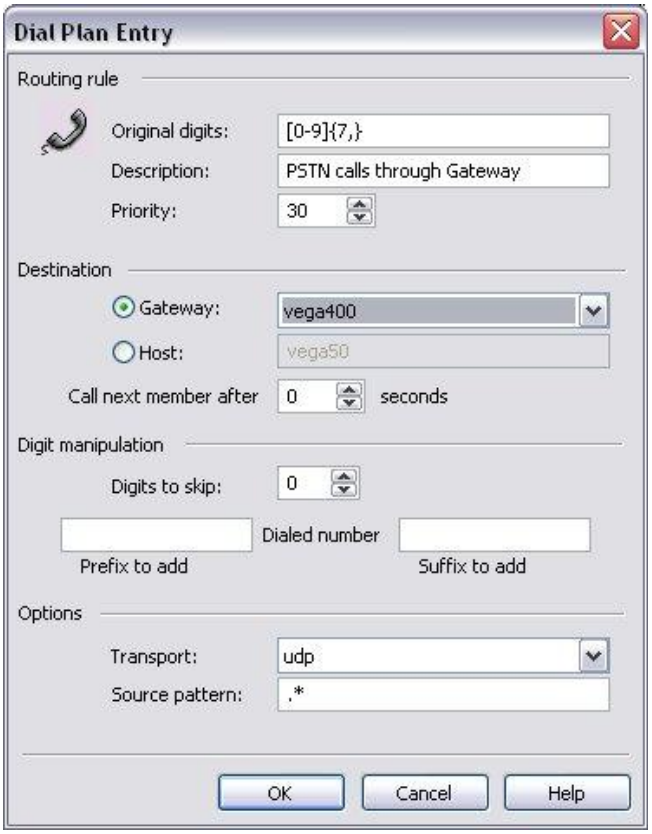
4.2 Adding a SIP Gateway

1. Click **Gateways**.
2. Right-click the right-hand panel and select **New Gateway**.
3. Choose **Public Switched Telephone Network (PSTN)** from the list of gateways.
4. In the **Host** name field, enter the IP address of the gateway.
5. Enter a descriptive name for the gateway.
6. Click **Save**.

4.3 Dial Plan

Incoming calls from the PSTN are already configured by routing incoming calls to the 10000 Trunk identity. An entry or entries must be entered in the Dial Plan for outgoing calls to the PSTN. There are many possibilities here, and the procedure below presents one possibility.

1. Go to **Communication Service -> UC Server -> Routing.**
2. If regular PSTN calls are to be routed out the gateway, add or modify an entry where the **Original Digits** are [0-9]{7,} and select the Vega gateway. For example:



4.4 Toll Restrictions

Configure the toll restrictions to match the requirements of your organization. Consult the UC Server Administration manual for the correct usage of regular expressions in the toll restrictions to enforce corporate dialing policy. Toll restrictions are explained in detail in the *Managing PBX Configuration Categories > Routing—Toll Restrictions* section of the *NetVanta Unified Communications Server Administrator Guide*, available online at http://kb.adtran.com.