# NetVanta Unified Communications Technical Note

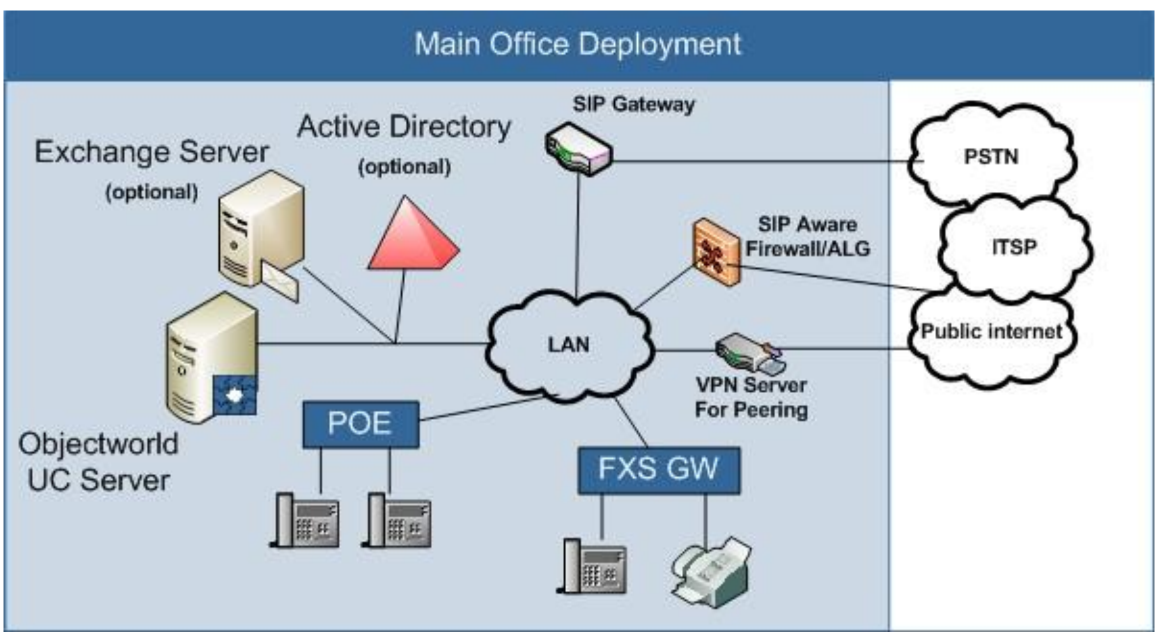
# Installing and Configuring AudioCodes MP-11x

## Introduction

The AudioCodes MP-11x is an analog gateway that can be used in UC server installations to provide a bridge between internal (SIP) phone calls and the phone network (PSTN). The MP-11x supports 4 or 8 analog trunks. 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 embodied in the NetVanta Unified Communications 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 the UC server. 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.

# Overview of Procedure

The M1K must be connected to the internal LAN (a 100 Mbps connection is recommended) and 1 to 8 trunks.

The MP-11x has two configuration options: webpage or configuration file. For convenience, this document describes configuration using the gateway's webpage. This has the advantage of providing familiarity with using the web page, which will most likely be used later for making minor changes in the gateway configuration. It has the disadvantage that an inadvertent change to one of the many parameters may cause problems. For this reason it is recommended that you make only the changes described below to a factory reset gateway. After the gateway is operational, further changes can be made as required.

The basic steps for installation and configuration are:

1. Unpack the MP-11x.
2. Mount the MP-11x.
3. Connect cables.
4. Set the IP address and subnet mask of the gateway through the serial port on MP-11x.
5. Access the MP-11x web page.
6. Configure the gateway.
7. Save the configuration and reboot the MP-11x.
8. Back up the configuration.

Steps 1 through 3 are standard for any gateway. Please follow the instructions provided by AudioCodes for the gateway. This document is intended to be a companion document to the AudioCodes documents. Use both while configuring the gateway.

The rest of this section details Steps 4 and 8 to configure the MP-11x for operation with the UC server.

# Configuring the Gateway

## 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 may result in unexpected behavior.

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

| | |
|---|---|
| Versions Version ID: | 5.00A.038.006 |
| DSP Type: | 0 |
| DSP Software Version: | 20916 |
| DSP Software Name: | 204IM |
| Flash Version: | 195 |

## Setting the IP Address and Subnet Mask

It is important that the MP-11x 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 MP-11x and a PC on the same LAN subnet.
2. Use or install from the AudioCodes CD a bootp application.
3. Power up the MP-11x.
4. Edit the IP address, subnet mask and gateway information in the bootp information window. See Figure 1 below.

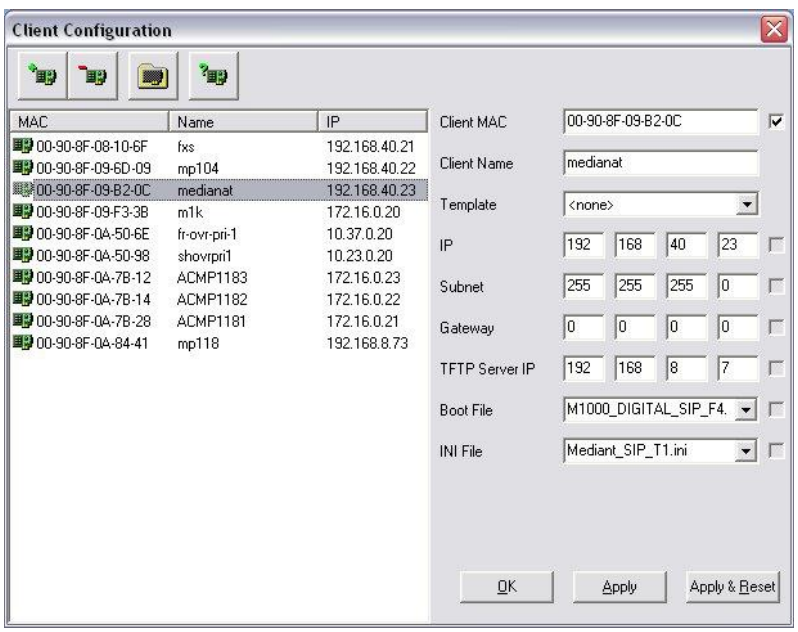



**Figure 1**

5. Access the gateway's webpage using your web-browser and the MP-11x's new IP address. The default login information is

   Username: Admin

   Password: Admin

## 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 **Gateway Name** blank.
3. Ensure that **Working with Proxy** is set to **Yes**.
4. In **Proxy IP Address** enter the IP address of the UC server.
5. Leave **Proxy Name** blank.
6. Select **Reset** and when prompted save the information.

See Figure 2 below.

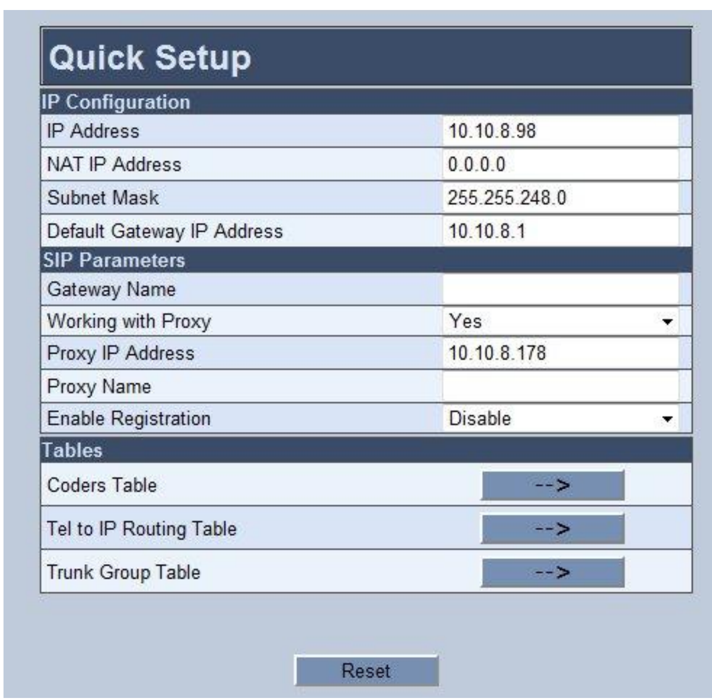



**Figure 2**

7. Go to **Protocol Management -> Routing Tables -> Tel to IP Routing**.
8. Enter a row that has **Dest. IP Address** set to the UC server IP address.
9. **Submit**.

See Figure 3 below.

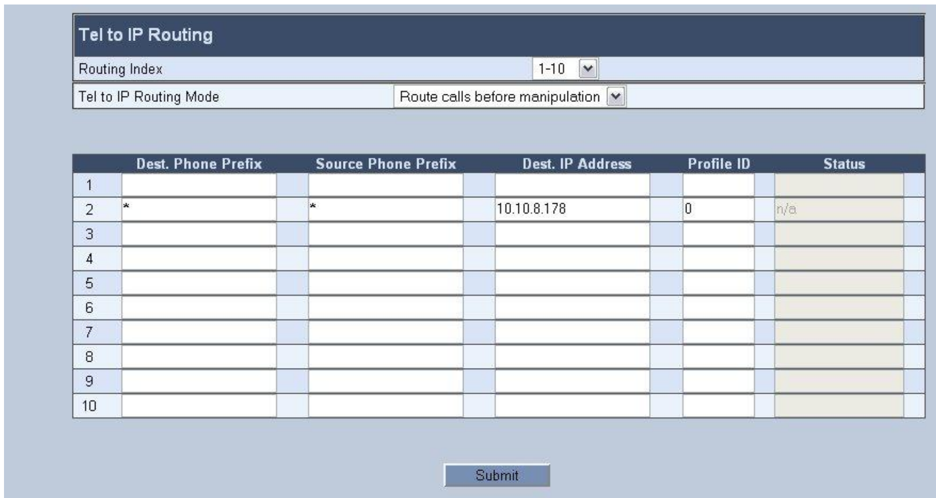



**Figure 3**

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

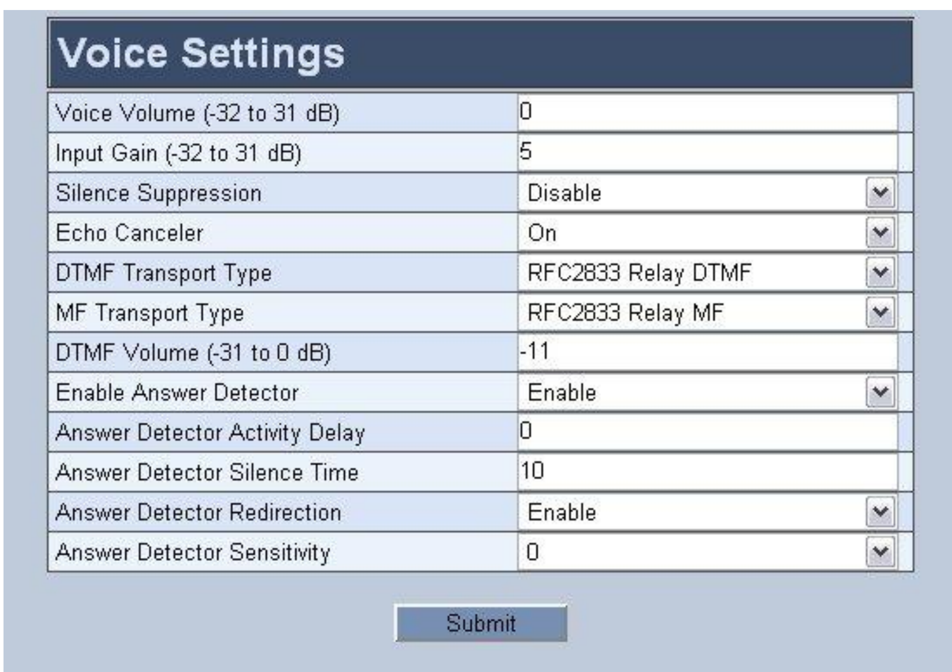



**Figure 4**

11. Go to **Advanced Configuration -> Media Settings -> Fax/Modem/CID Settings** and ensure that:

- **Fax Transport Mode** is **T.38 Relay**.
- **V.21 Modem Transport Type** is set to **Disable**.

See Figure 5 below.

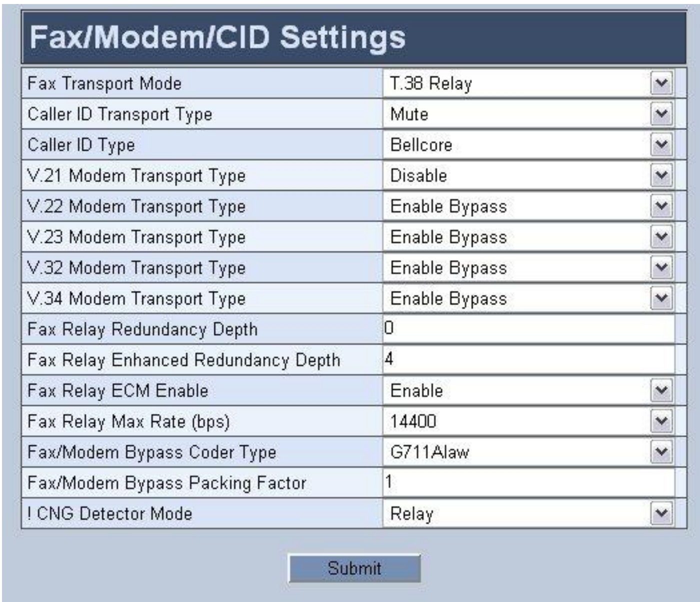



**Figure 5**

# 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.

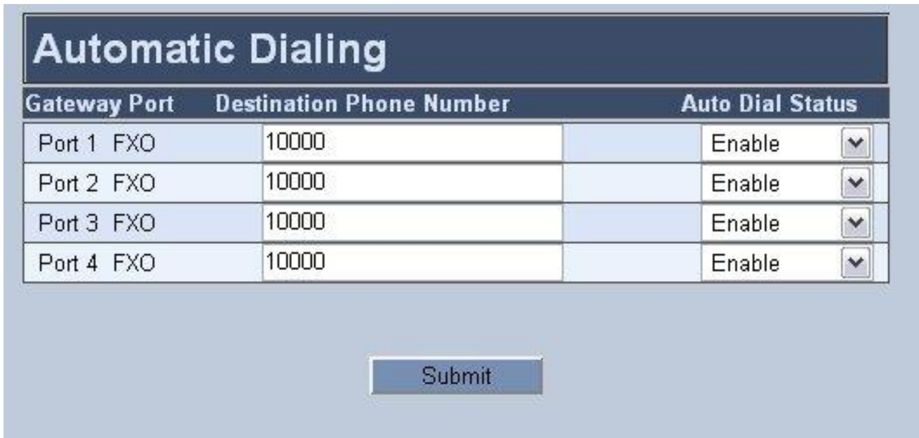
## Incoming

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.

Go to **Protocol Management -> Endpoint Settings -> Automatic Dialing** and:

1. Enable each of the trunks that are operational.
2. In **Destination Phone Number** add an entry of 10000 for each one.

See Figure 6 below.



**Automatic Dialing**

| Gateway Port | Destination Phone Number | Auto Dial Status |
|---|---|---|
| Port 1 FXO | 10000 | Enable |
| Port 2 FXO | 10000 | Enable |
| Port 3 FXO | 10000 | Enable |
| Port 4 FXO | 10000 | Enable |

Submit

**Figure 6** – With these settings any incoming call is routed to 10000.

## Outgoing

A gateway may have more physical trunks than are being used at a site. When an outgoing call is being 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 trunks that are actually in use.

If an operational trunk is busy, the gateway must 'hunt' for a free 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 -> Endpoint Phone Numbers.**
2. Assign each trunk the same **Hunt Group ID**.

See Figure 7 below.

**Endpoint Phone Number Table**

| | Channel(s) | Phone Number | Hunt Group ID | Profile ID |
|---|---|---|---|---|
| 1 | 1 | 1001 | 1 | 1 |
| 2 | 2 | 1002 | 1 | 1 |
| 3 | 3 | 1003 | 1 | 1 |
| 4 | 4 | 1004 | 1 | 1 |

Register Un-Register
Submit

**Figure 7** – The important thing here is the **Hunt Group ID.**

3. Go to **Protocol Management -> Hunt Group Settings.**
4. For the **Hunt Group ID** chosen select an appropriate **Channel Select Mode.**

See Figure 8 below.

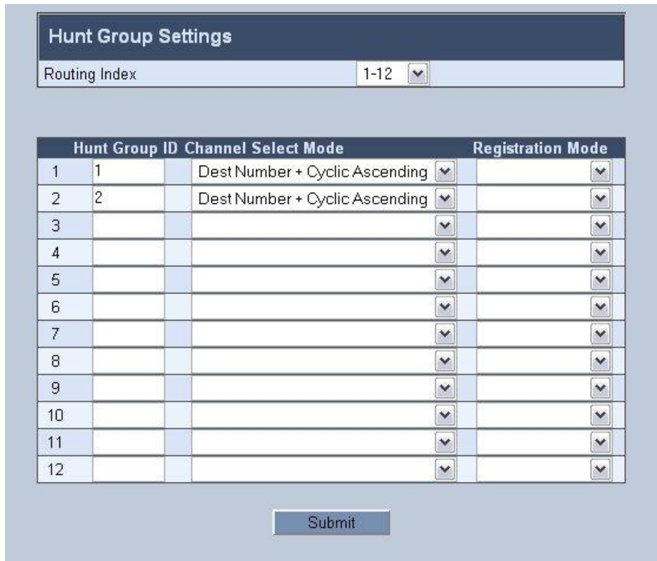



**Figure 8**

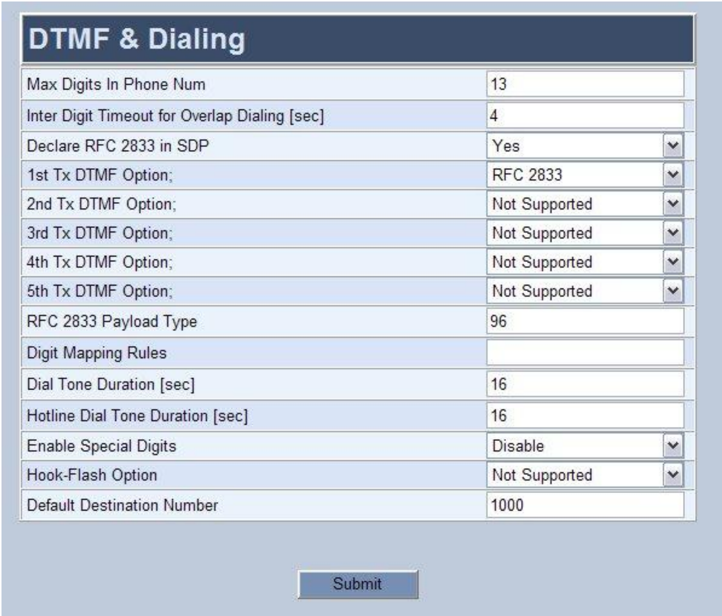
## Outgoing DID Presentation

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.

## DTMF Reception

1. Go to **Protocol Management -> Protocol Definition -> DTMF & Dialing**.
2. Change **Declare RFC 2833 in SDP** to **Yes**.
3. Click **Submit**.

See Figure 9 below.



| DTMF & Dialing | |
|---|---|
| Max Digits In Phone Num | 13 |
| Inter Digit Timeout for Overlap Dialing [sec] | 4 |
| Declare RFC 2833 in SDP | Yes |
| 1st Tx DTMF Option; | RFC 2833 |
| 2nd Tx DTMF Option; | Not Supported |
| 3rd Tx DTMF Option; | Not Supported |
| 4th Tx DTMF Option; | Not Supported |
| 5th Tx DTMF Option; | Not Supported |
| RFC 2833 Payload Type | 96 |
| Digit Mapping Rules | |
| Dial Tone Duration [sec] | 16 |
| Hotline Dial Tone Duration [sec] | 16 |
| Enable Special Digits | Disable |
| Hook-Flash Option | Not Supported |
| Default Destination Number | 1000 |

Submit

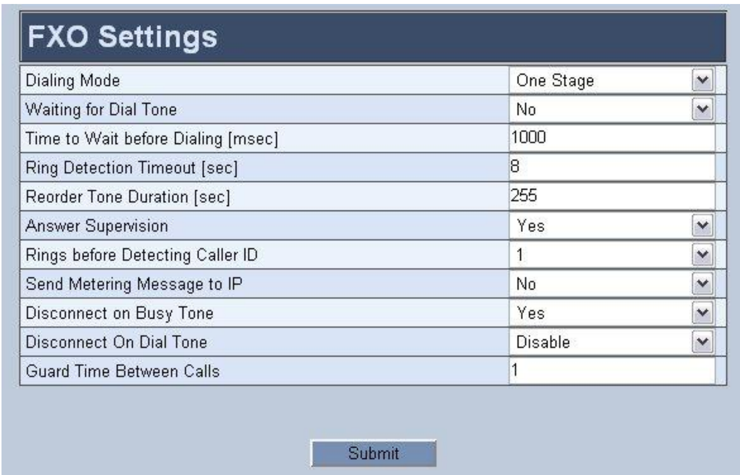
**Figure 9**

## POTS

The basic settings in the gateway should be sufficient for most cases.

1. Go to **Protocol Management -> Advanced Applications -> FXO Settings**.
2. Ensure that **Dialing Mode** is One Stage.
3. Ensure that **Answer Supervision** is set to Yes.
4. Ensure that **Rings before Detecting Caller ID** is set to 1.

See Figure 10 below.



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

Submit

**Figure 10**

5. **Submit**

# Backing up the Configuration

It is important to back up the system configuration once the gateway is configured.

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

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

# Configuring the NetVanta UC Server

After the gateway is 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 client.

## Adding a Trunk Identity

1. Go to **Identities**.
2. Right-click in the right panel and select **New Identity**.
3. In the first page of the Wizard, select an **Attendant** identity. Ensure 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). Ensure that **Default Trunk Service** is the service to be run.

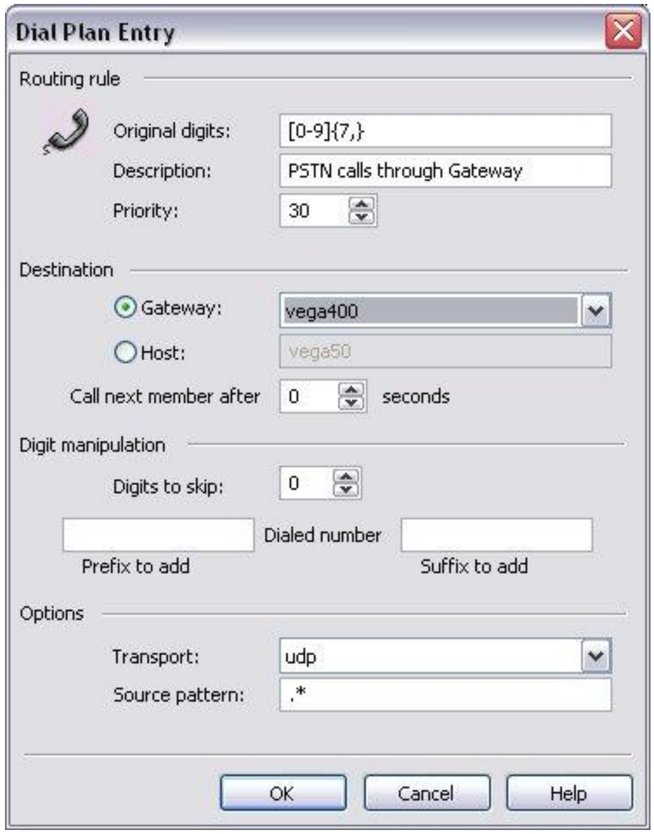
## Adding a SIP Gateway

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

## Dial Plan

Incoming calls from the PSTN are already configured by having incoming calls routed to the 10000 Trunk identity. An entry or entries must be entered in the Dial Plan for outgoing calls to the PSTN.

1. Go to **Communication Service -> UC Server -> Routing.**
2. There are many possibilities here. 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:




## Toll Restrictions

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