



## NetVanta Unified Communications Technical Note

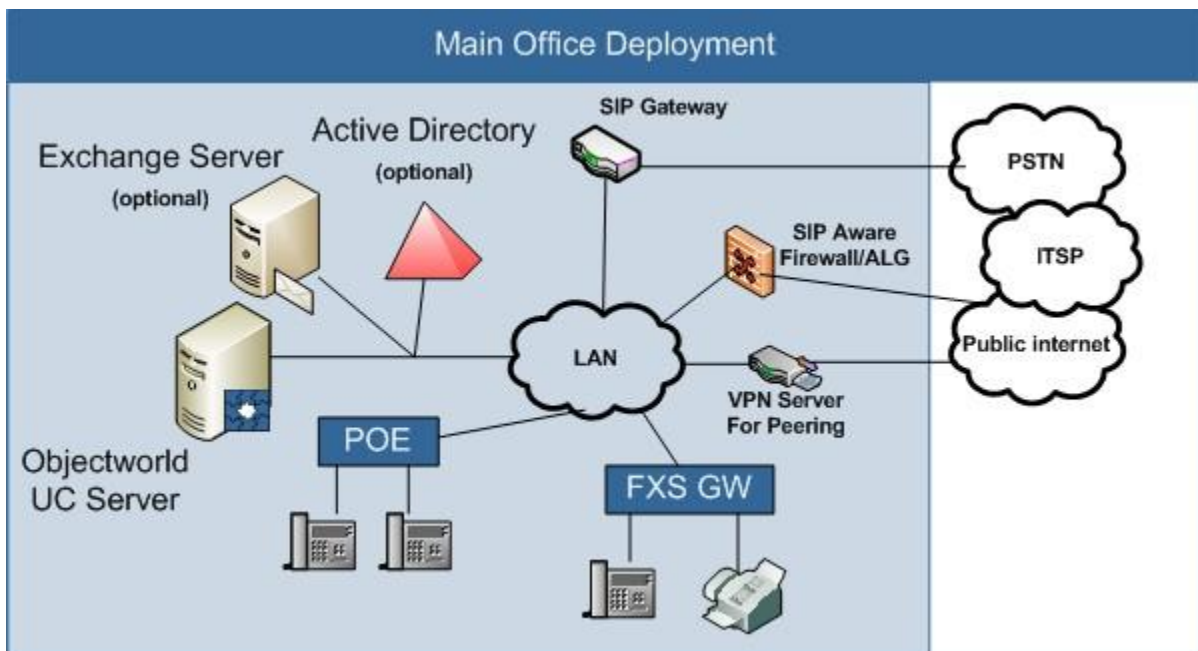
# Installing and Configuring Vegastream Vega 400

## Introduction

The Vegastream Vega 400 is a T1 digital gateway which may be used in NetVanta Unified Communications Server installations to provide a bridge between internal (SIP) phone calls and the phone network (PSTN). The Vega 400 supports up to 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 embodied in the UC server. All telephony services are provided through the mutual cooperation 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 may fall outside the guidelines of this document.

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

## Overview of Procedure

To provide its functionality, the Vega 400 must be connected to the internal LAN (a 100 Mbps connection is recommended) and from 1 to 4 T1 digital spans.

The Vega 400 has two configuration methods: webpage or command line interface (CLI). For convenience this document describes configuration using the gateway's webpage. This has the advantage of providing familiarity with the web 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 may cause problems. **For this reason it is recommended that only the changes described below are made and to a factory reset gateway.** Once the gateway is operational further changes may be made as required.

The basic steps for installation and configuration are:

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

Steps 1 through 3 are standard for any gateway. Please follow the instructions provided by Vegastream for the gateway (See the Vega Primer at [http://www.vegaassist.com/tech\\_docs.php?ProdId=1](http://www.vegaassist.com/tech_docs.php?ProdId=1)). The Vega Primer provides comprehensive information on the use and configuration of the gateway. This document is intended to be a companion document to the Vega Primer. Both should be used while configuring the gateway.

The rest of this section details Steps 4 and 8 to configure the Vega 400 for operation with the UC server.

## Gateway Configuration

### Software Level

This document is based on the gateway software from interoperability testing conducted in June 2007. If the software revision does not match, contact the manufacture'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 **Maintenance -> Show Version** and ensure that the gateway software matches the information below before proceeding.

Physical Data : Flash Page 1, upper 7.5 MB, v-file FLASH:IMAGE1

Binary File Name : VEGA400\_R080S020

Release Date : Apr 16 2007 11:15:04

Versioning Info : SIP Firmware Rev 08.00 for H/W Type 10

Boot Requirements : Boot Loader 02.00

## Set IP Address and Subnet Mask

Connect a serial cable to the RS-232 port of the Vega 400. Configure a terminal program such as HyperTerminal for 115,200 bits per second, 8 data bits, 1 stop bit, No parity.

Hit enter and then log in using username: admin and password: admin.

It is important that the Vega 400 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. Once that address has been chosen enter the following commands using your terminal program:

1. set .lan.gateway.dhcp\_if="1"
2. set .lan.if.1.ip="<<IP address of Vega 400>>"
3. set .lan.if.1.subnet="<<subnet mask for Vega 400>>"
4. set .lan.gateway.ip==<<gateway address of LAN gateway>>
5. Save
6. Reboot system

This assumes LAN Interface 1 is the one being used. Substitute 1 for 2 if the other interface is to be used.

**Note:** if 2 is used then in any sections below that reference LAN 1 it should be substituted with LAN 2.

Now access the gateway's webpage using your browser and the Vega's new IP address. The default login information is:

1. Username: admin
2. Password: admin

## SIP Configuration

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 will be 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 e.g. voicemail 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 of the TDM voice path. This is implemented using T.38 fax support. This must be enabled on the gateway.

1. Navigate to **SIP -> SIP Profiles -> Modify** for profile1.
2. In the **Local Domain** field enter the IP address for the UC server.
3. Submit and Save

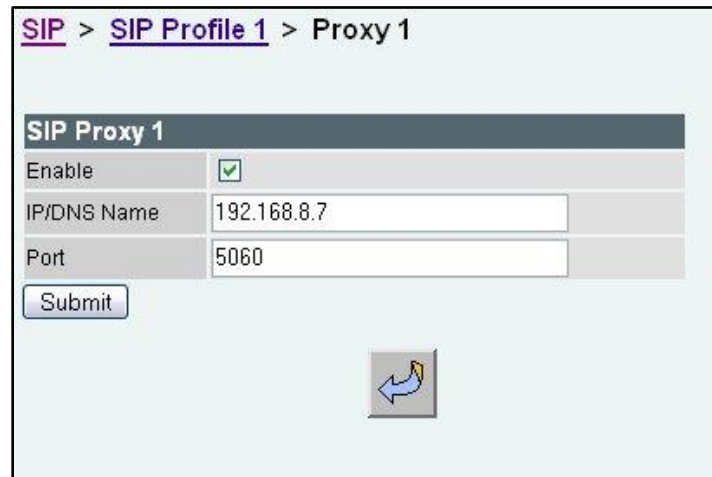
See Figure 1 below.

SIP Profile 1 Configuration 1	
Name	profile1
Interface ID	9901
Local Domain	192.168.8.7
Alternative Local Domain	-a
From Header 'userinfo'	Calling Party
From Header 'host'	Local Domain
To Header 'host'	Local Domain
<input type="button" value="Submit"/>	

**Figure 1**

4. Go to **SIP Proxy -> Chg? -> Modify**.
5. Enter the UC server IP address or domain name in the IP/DNS Name. Make sure that **Enable** is checked.
6. **Submit and Save**.

See Figure 2 below.



The screenshot shows a web interface for configuring SIP Proxy 1. The breadcrumb navigation at the top reads "SIP > SIP Profile 1 > Proxy 1". Below this is a form titled "SIP Proxy 1" with the following fields:

SIP Proxy 1	
Enable	<input checked="" type="checkbox"/>
IP/DNS Name	<input type="text" value="192.168.8.7"/>
Port	<input type="text" value="5060"/>

Below the form is a "Submit" button and a refresh icon (a square with a circular arrow).

**Figure 2**

7. Go to **Miscellaneous**. Ensure that
  - **DTMF Transport -> DTMF Transport -> rfc2833** is checked.
  - **Fax Detect** is terminating.
  - **Enable Modem** is checked.
  - **Modem Detect** is never.

See Figure 3 below.

SIP Session Timers Configuration	
<a href="#">SIP Session Timers</a>	
Miscellaneous	
SIP Signalling Transport	<input checked="" type="radio"/> udp <input type="radio"/> tcp
Reliable Provisional Responses	<input type="radio"/> supported <input type="radio"/> require <input checked="" type="radio"/> off
DTMF Transport	<input checked="" type="radio"/> rfc2833 <input type="radio"/> info <input type="radio"/> rfc2833 and tx info <input type="radio"/> rfc2833 and rx info <input type="radio"/> off
DTMF INFO	<input checked="" type="radio"/> mode1 <input type="radio"/> mode2
RFC2833 payload (96-127)	<input type="text" value="96"/>
Use T38 AnnexE	<input type="checkbox"/>
Accept T38 AnnexE	<input type="checkbox"/>
Fax Detect	<input type="text" value="terminating"/>
Enable Modem	<input checked="" type="checkbox"/>
Modem Detect	<input type="text" value="never"/>
Media Control Profile	<input type="text" value="0"/>
Signalling Application ID	<input type="text" value="none"/>
T1 Retry Timer Increment (ms)	<input type="text" value="2000"/>
T2 Retry Timer Limit (ms)	<input type="text" value="4000"/>
Maximum Calls	<input type="text" value="120"/>
<input type="button" value="Submit"/>	

Figure 3

## 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.

Go to **Dial Plan -> Profiles** and for the default profile click **Modify**. In Plans in the Profile we will create and enable two entries: PstnToSip and SipToPstn.

## 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.

If for any reason calls are to be directed to a different extension then it should be used instead of 10000.

1. Add an entry and configure it as shown in Figure 4.
2. Apply and Save.

Dial Planner > Profile 1 > Plan 1


**Modify Plan**

Plan ID	1
Profile ID	1
Name	PstnToSip
Source	"IF:0[1-4]...TEL:<*>"
Destination	"IF:9901,TEL:10000"
Cost Index	0
Group	1 - default

Apply

▶ Regular Expression Help

▶ Token Help



**Figure 4** - This routes a call from any of the T1 spans to identity 10000 on the SIP UC server. This assumes a ‘standard’ configuration where a trunk attendant identity of 10000 is created on the UC server (which will be covered later in this document). If, for example, all calls are to be directed to an extension other than 10000 then it should be replaced with that number.

## Incoming DID Call Routing

The entry above routes all incoming PSTN calls to identity 10000 no matter what number is called. A PRI may support many DIDs. When the call is presented from the PSTN the carrier provides the called party information. Often this is not the full 10 digits 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 different destination.

When the criteria in **Source** is matched the call is routed to the **Destination**. If there is a tie the **Cost Index** is used as a tie breaker. So assuming a catch all routing to 10000 is desired its **Cost Index** should be increased to allow room for other entries. A new, higher priority, entry can then be added with a specific rule for a DID with a **Cost Index** less than the catch all.

The new entry for the DID entry would then have the form:

**Source:** “IF:0[1-4]...TEL:<<DID number>>”

**Destination:** “IF:9901,TEL:<<Destination Identity>>”

Where <<DID number>>” is replaced by the incoming DID string and <<Destination Identity>> is the identity that the call should be directed to.

## Outgoing

A gateway may have more physical and logical 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 physical interfaces and trunks that 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 e.g. ascending or descending.

1. Create a new dial plan entry and configure it as shown in Figure 5.
2. Apply and Save.

Dial Planner > Profile 1 > Plan 2


**Modify Plan**

Plan ID	2
Profile ID	1
Name	SipToPstn
Source	"IF:9901,TEL:<*>"
Destination	"IF:1001,TEL:<1>"
Cost Index	0
Group	1 - default

Apply

▶ Regular Expression Help

▶ Token Help



**Figure 5** - This routes all calls from interface 9901 (SIP / UC Server) to the Call Presentation Group 1001 which is a collection of T1 interfaces and channels.



## 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. If an enterprise wide caller ID is required then for the dial plan entry above the **Destination** line should be changed to:

```
"IF:1001,TEL:<1>,TELC:<<Caller ID>>"
```

Where <<Caller ID>> is replaced by the desired caller ID. TELC specifies the caller as opposed to TEL which is the called number.

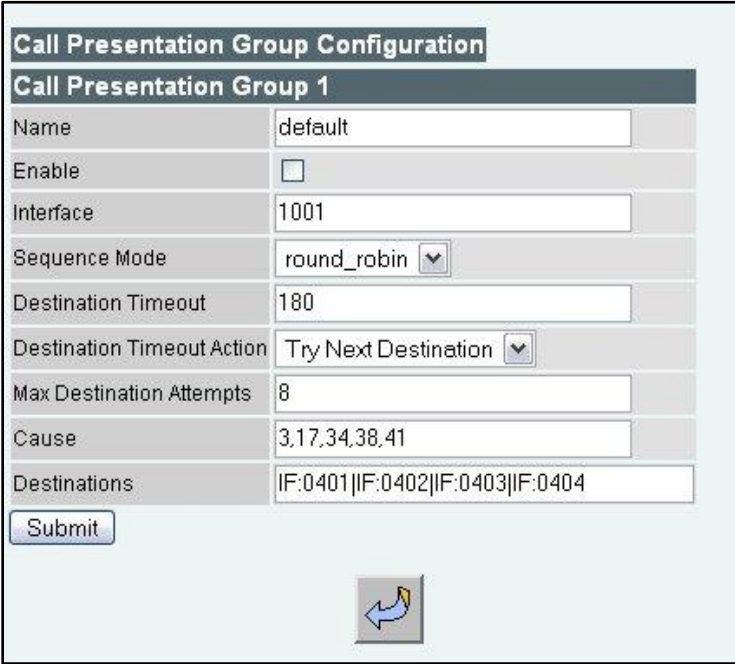
If a different caller ID is desired depending on who is calling then another dial plan entry should be added where the **Source** matching is like:

```
"IF:9901,TEL:<. *>,TELC:<<caller>>"
```

Where <<caller>> is replaced by the desired calling number.

## Outgoing Call Presentation Group

1. Go to **Dial Plan -> Call Presentation Groups**.
2. Ensure that it is configured as shown in Figure 6.



The screenshot shows a configuration window titled "Call Presentation Group Configuration" for "Call Presentation Group 1". The configuration includes the following fields:

Field	Value
Name	default
Enable	<input type="checkbox"/>
Interface	1001
Sequence Mode	round_robin
Destination Timeout	180
Destination Timeout Action	Try Next Destination
Max Destination Attempts	8
Cause	3,17,34,38,41
Destinations	IF:0401  IF:0402  IF:0403  IF:0404

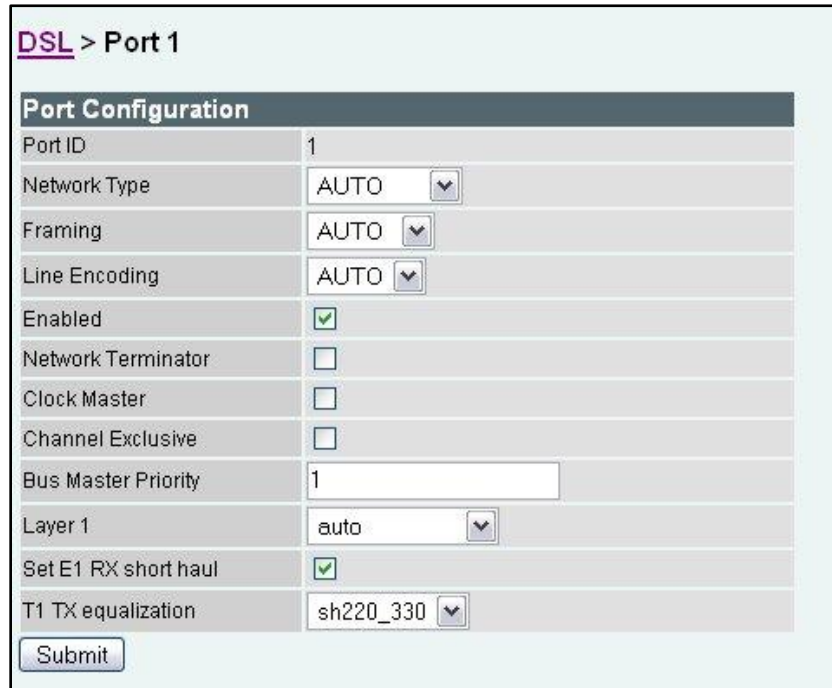
At the bottom of the form, there is a "Submit" button and a refresh icon.

**Figure 6** - This defines a logical interface 1001 which contains all 4 T1 spans for outgoing calls. Generally there is no need to remove spans or individual channels that are not being used here since they will be disabled from a physical and logical perspective later. However if, for example, a span is only being used for incoming calls then it should be excluded from the **Destinations** list. It is important that the **Cause** codes be accurate to ensure the appropriate roll over behavior.

## DSL

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

1. Go to **DSL**.
2. Ensure that Network Topology is set to T1 (most likely for North America).
3. In **Port Configuration** only the T1 spans that are active should be **Enabled**.
4. Click **Modify** for an individual span (see Figure 7):



The screenshot shows a web interface for configuring a DSL port. The title is "DSL > Port 1". Below the title is a "Port Configuration" section with the following fields:

Port ID	1
Network Type	AUTO
Framing	AUTO
Line Encoding	AUTO
Enabled	<input checked="" type="checkbox"/>
Network Terminator	<input type="checkbox"/>
Clock Master	<input type="checkbox"/>
Channel Exclusive	<input type="checkbox"/>
Bus Master Priority	1
Layer 1	auto
Set E1 RX short haul	<input checked="" type="checkbox"/>
T1 TX equalization	sh220_330

At the bottom of the form is a "Submit" button.

**Figure 7**


5. **Network Type** should be set as defined by your carrier. DMS, ATT and NI are the most common settings.
6. **Framing** should be set as defined by your carrier. ESF is the most common.
7. **Line Encoding** should be set as defined by your carrier. B8ZS is the most common.
8. **Enabled** should be checked.
9. **Network Terminator** should be set as defined by your carrier. Mostly likely it should not be checked.
10. **Clock Master** should be set as defined by your carrier. Most likely it should not be checked.
11. **Bus Master Priority** should most likely be set to 1. If this is a multi T1 span installation this area may be complicated and is beyond the scope of this document. Further information is provided at [http://www.vegaassist.com/tech\\_docs.php?ProdId=1](http://www.vegaassist.com/tech_docs.php?ProdId=1) in the "ISDN Clocks" document.
12. **Layer 1** should most likely be set to g711Ulaw64k.

### 13. Submit and Save.

Now go to the bottom of this page at **Groups Chg ? -> Modify** (see Figure 8).

Modify Port Group	
Group ID	1
Port ID	1
Interface ID	0401
DN	*
First Channel	1
Last Channel	auto
Alloc Channel	Default
Tunnel Mode	Off
Tunnel IEs Only	<input type="checkbox"/>

Submit



**Figure 8**

1. In **First Channel** and **Last Channel** specify the D channels that are being used as specified by your carrier. This ensures that only active channels are used.
2. Specify in **Alloc Channel** a hunting mode (which may be specified by your carrier). Often Linear Down is a good choice.
3. Submit and Save.

### PSTN Ringback

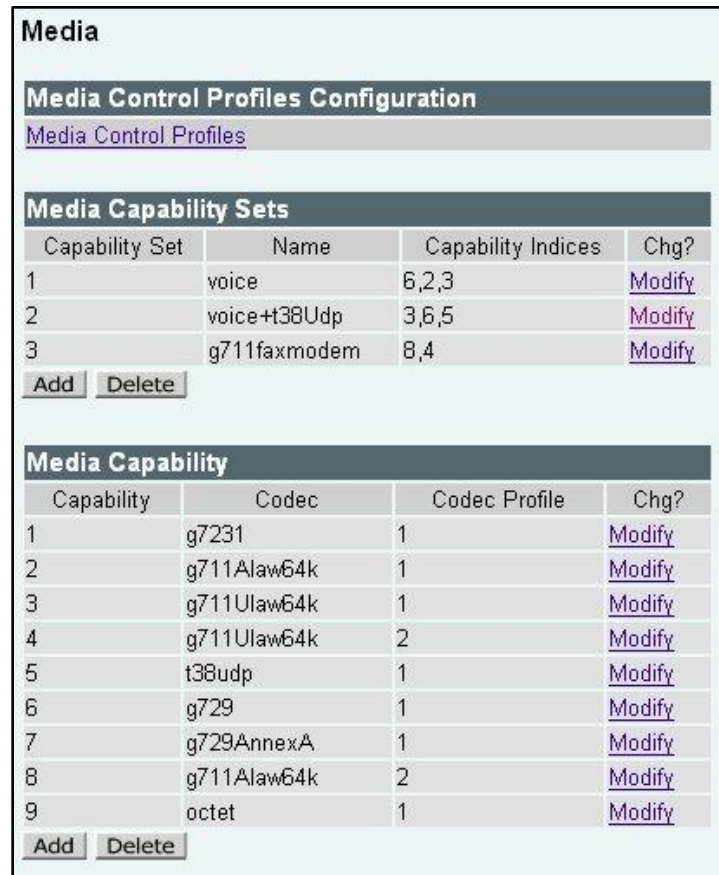
If the Vega 400 is not the Network Terminator (the usual mode for customer premises equipment) then its default is that it will not generate a ringback tone when a call is received from the PSTN. If the carrier providing the PRI service does not provide the ringback tone (if a caller from the PSTN hears ringback when the gateway is called then the carrier is providing ringback) then it must be enabled on the gateway. If required:

1. Go to the **Advanced** section of the webpage.
2. In the **CLI** section type: set \_advanced.isdn.user\_progress=1
3. **Submit** the change.
4. **Apply** and **Save** the change.

## Media Configuration

SIP calls utilize a codec for voice communication and the T.38 protocol for faxes. The Vega gateway allows for the selection of which codecs to use. In general G.711 and T.38 need to be allowed options on the gateway.

1. Go to **Media**.
2. Create or ensure that there is a **Media Capability Set** called voice+t38udp.
3. Edit the set and ensure that **Capability Indices** 3 and 5 are in it (North American settings). See Figure 9.



The screenshot displays the 'Media' configuration page. It features a 'Media Control Profiles Configuration' section with a link to 'Media Control Profiles'. Below this is the 'Media Capability Sets' section, which contains a table with three rows of capability sets. Each row includes a 'Capability Set' ID, a 'Name', 'Capability Indices', and a 'Chg?' link. Below the table are 'Add' and 'Delete' buttons. The second section is 'Media Capability', which contains a table with nine rows of capabilities. Each row includes a 'Capability' ID, a 'Codec', a 'Codec Profile', and a 'Chg?' link. Below this table are also 'Add' and 'Delete' buttons.

Capability Set	Name	Capability Indices	Chg?
1	voice	6,2,3	<a href="#">Modify</a>
2	voice+t38Udp	3,6,5	<a href="#">Modify</a>
3	g711faxmodem	8,4	<a href="#">Modify</a>

[Add](#) [Delete](#)

Capability	Codec	Codec Profile	Chg?
1	g7231	1	<a href="#">Modify</a>
2	g711Alaw64k	1	<a href="#">Modify</a>
3	g711Ulaw64k	1	<a href="#">Modify</a>
4	g711Ulaw64k	2	<a href="#">Modify</a>
5	t38udp	1	<a href="#">Modify</a>
6	g729	1	<a href="#">Modify</a>
7	g729AnnexA	1	<a href="#">Modify</a>
8	g711Alaw64k	2	<a href="#">Modify</a>
9	octet	1	<a href="#">Modify</a>

[Add](#) [Delete](#)

**Figure 9**

4. Submit and Save.
5. Now go back to **SIP -> Media** and ensure that voice+t38Udp is selected in the dropdown box. See Figure 10.

**SIP Configuration**

**General**

Local SIP Port: 5060

Accept Non-Proxy Invites:

Submit

**SIP Profiles**

SIP Profile	Name	Interface ID	Other SIP Profile Parameters	Chg?
1	profile1	9901	====>	<a href="#">Modify</a>

Add Delete

**Media**

Capability Set: 2 - voice+t38Udp

Submit

**Figure 10**

## LAN Configuration

There should be no other changes required for the LAN. However, ensure that all SIP traffic is routed to the correct LAN port.

1. Go to **LAN** and in the **Calls** section select LAN 1 in the drop down menu (See Figure 11).
2. Save and Submit.

**Calls**

LAN Profile: 1-LAN\_1

Submit

**Figure 11**

## ECM Modification

When faxes are sent or received by the UC server the T.38 protocol is used to communicate with the gateway. An Error Correction Mode (ECM) is used that must currently be disabled on the gateway. This configuration can only be done via the command line as follows:

1. Telnet to IP address of the Vega.
2. Log in using your username and password (Default: admin / admin)
3. Type : set \_advanced.t38.allow\_ecm=0
4. Type : save
5. Reboot the system as requested.

If for any reason the gateway was not rebooted above please reboot the gateway through the webpage interface (**Main Page -> Reboot System**).

## Backup Configuration

Go to **Maintenance -> Upload/Download Configuration -> Receive File From Vega -> Download**. Store the configuration file in a safe place.

## UC Server Configuration

Once 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 client.

### Add a Trunk Identity

1. Go to **Identities**.
2. Right click on 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.

### Add a SIP Gateway

1. Click on **Gateways**.
2. Right click on the right hand 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 Vega gateway.
5. Enter a descriptive name for the gateway.
6. **Save**.

### Dial Plan

Incoming calls from the PSTN have been configured already 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 then 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 usage of regular expressions in the toll restrictions to enforce corporate dialing policy. It will be explained in detail in the “Managing PBX Configuration Categories > Routing—Toll Restrictions” section.