NetVanta Unified Communications Technical Note

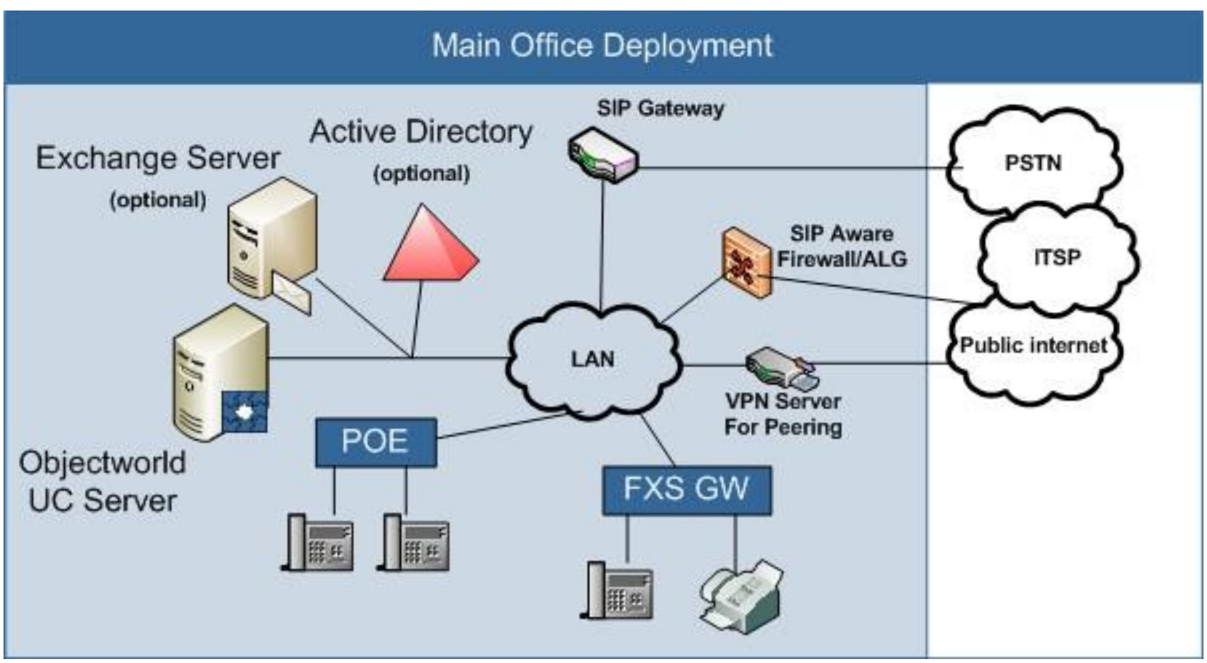
Installing and Configuring Vegastream Vega 50 6x4

1 Introduction

The Vegastream Vega 50 6x4 is an analog 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 50 6x4 supports up to 6 interface modules with 4 trunks per module allowing for a total of 24 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 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.



2 Overview of Procedure

To provide its functionality the Vega 50 must be connected to the internal LAN (a 100 Mbps connection is recommended) and from 1 to 24 analog trunks.

The Vega 50 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. Backup the configuration.

Steps 1-3 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*). This 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 50 for operation with the UC server.

3 Configuration

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 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.

```
[Partition 1]: *ACTIVE*
   Physical Data     : Flash Page 1, upper 7.5 MB, v-file FLASH:IMAGE1
   Binary File Name  : VEGA-6x4_R080S020
   Release Date      : Apr 16 2007 11:41:51
   Versioning Info   : SIP Firmware Rev 08.00 for H/W Type 11
   Boot Requirements : Boot Loader 02.00
```

3.2 Set IP Address and Subnet Mask

Connect a serial cable to the RS-232 port of the Vega 50. 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 login using username: admin and password: admin.

It is important that the Vega 50 have a LAN IP address that does change. This can be set using a static IP address and subnet mask compatible with the onsite LAN. Once that address has been chosen enter the following commands:

1. set .lan.gateway.dhcp_if="1"
2. set .lan.if.1.ip="<<*IP address of Vega 50*>>"
3. set .lan.if.1.subnet="<<*subnet mask for Vega 50>>"*
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 web-browser and the Vega's new IP address. The default login information is

1. Username: admin
2. Password: admin.

3.3 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. 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 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 UC Sever.
3. Submit and Save.

See Figure 1 below.

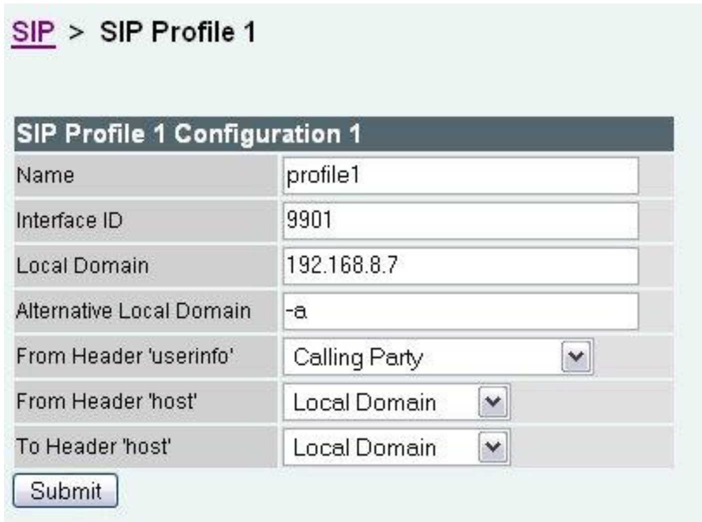


Figure 1

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

See Figure 2 below.

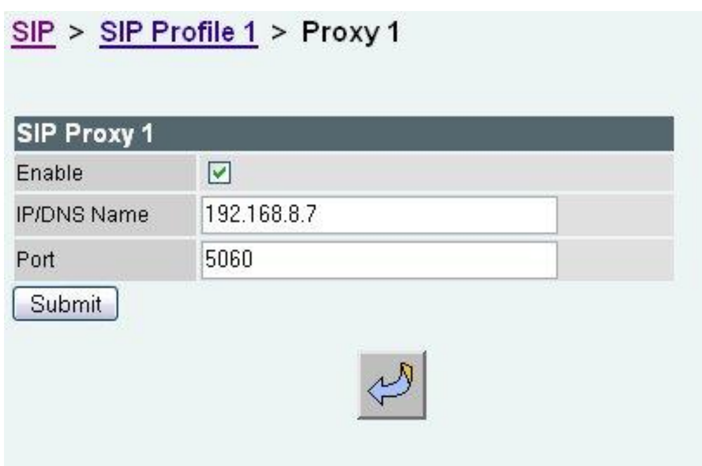


Figure 2

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

See Figure 3 below.

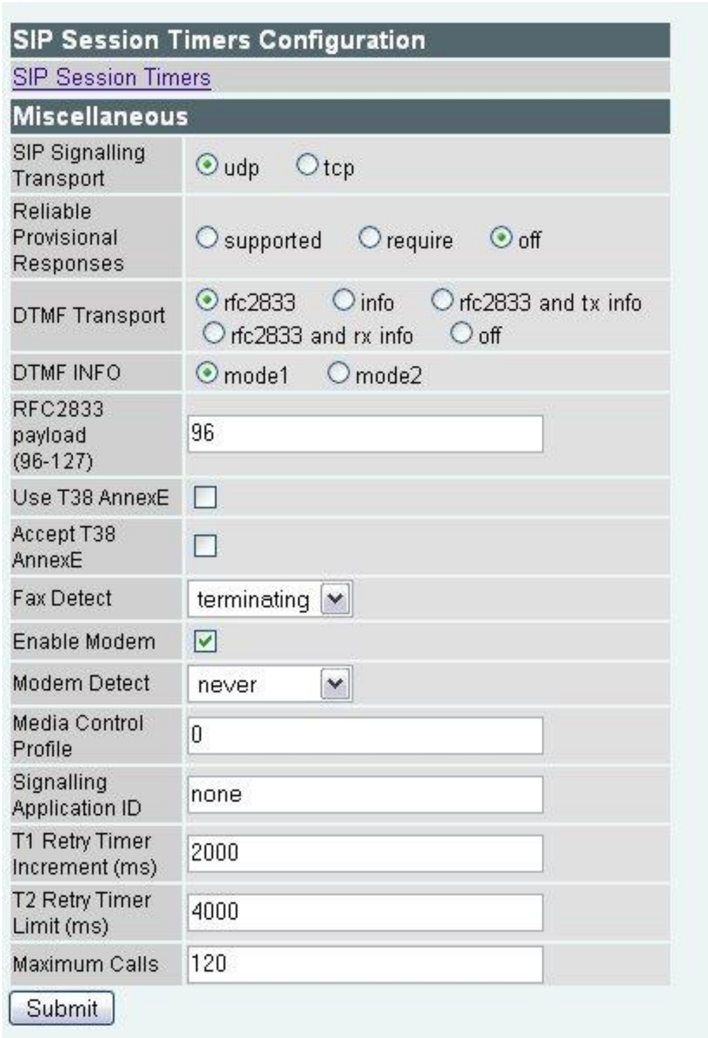


Figure 3

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.

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

3.4.1 Incoming

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

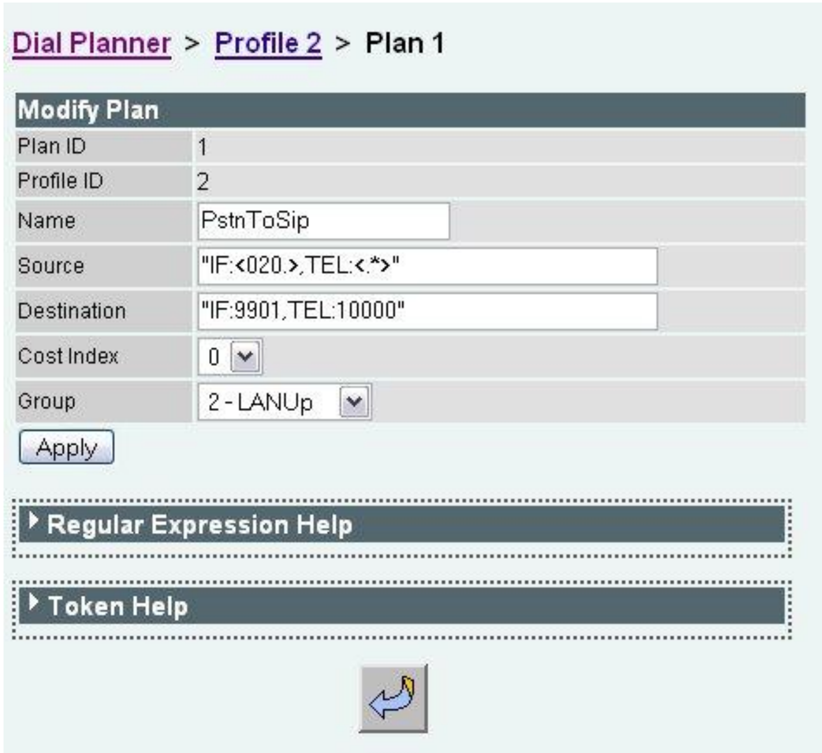


Figure 4 - This routes any call from any of the trunks to identity 10000 on the SIP UC server. This assumes a ‘standard’ configuration where a trunk attendant identity of 10000 is created on the UC server (which will be covered later in this document). If, for example, all calls are to be directed to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

3.4.2 Incoming DID Call Routing

The entry above routes all incoming PSTN calls to identity 10000 no matter what number is called. In infrequent cases there may be a requirement to route calls from one trunk or trunks to one identity and from another group to a second identity. This may be done by adding additional plan entries.

When the S**ource** criteria are matched, the call is routed to the **Destination**. If there is a tie then the **Cost Index** is used as a tie breaker. So assuming a catch all routing to 10000 the **Cost Index** above should be increased to allow room for other entries. A new entry can then be added which is a specific rule for a trunk with a **Cost Index** less (higher priority) than the catch all.

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

Source: “IF:020x, TEL:<.*>

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

Where the x in 020x identifies which trunk port is being selected and **<<Destination Identity>>** is the identity that the call should be directed to.

3.4.3 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 physical interfaces that are 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. Now create/modify a new dial plan entry and configure it as shown in Figure 5b.
2. Apply and Save.

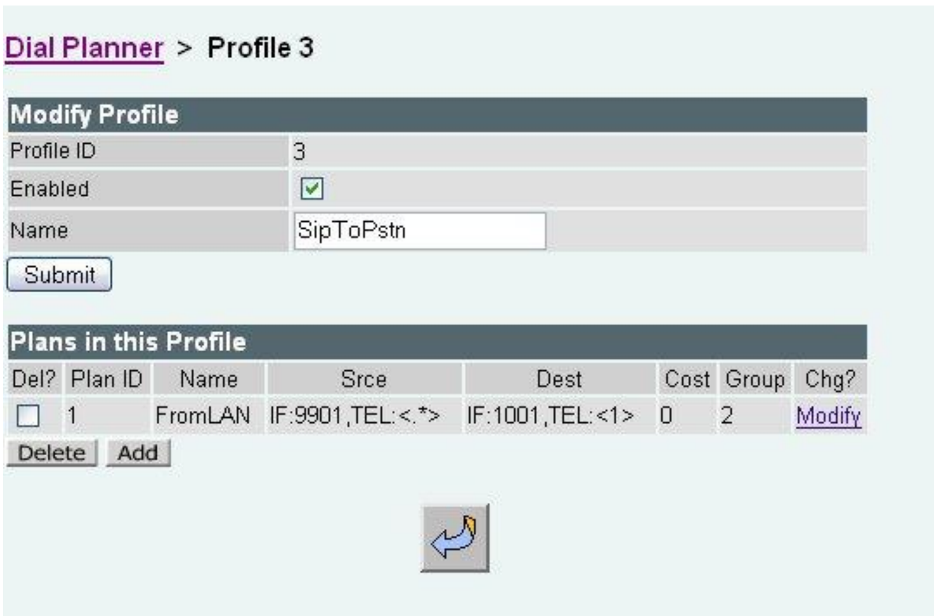


Figure 5a

Figure 5b - This routes all calls from interface 9901 (SIP / UC server) to the Call Presentation Group 1001 which is a collection of trunks.

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 for analog trunks unlike PRI.

3.4.5 Outgoing Call Presentation Group

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

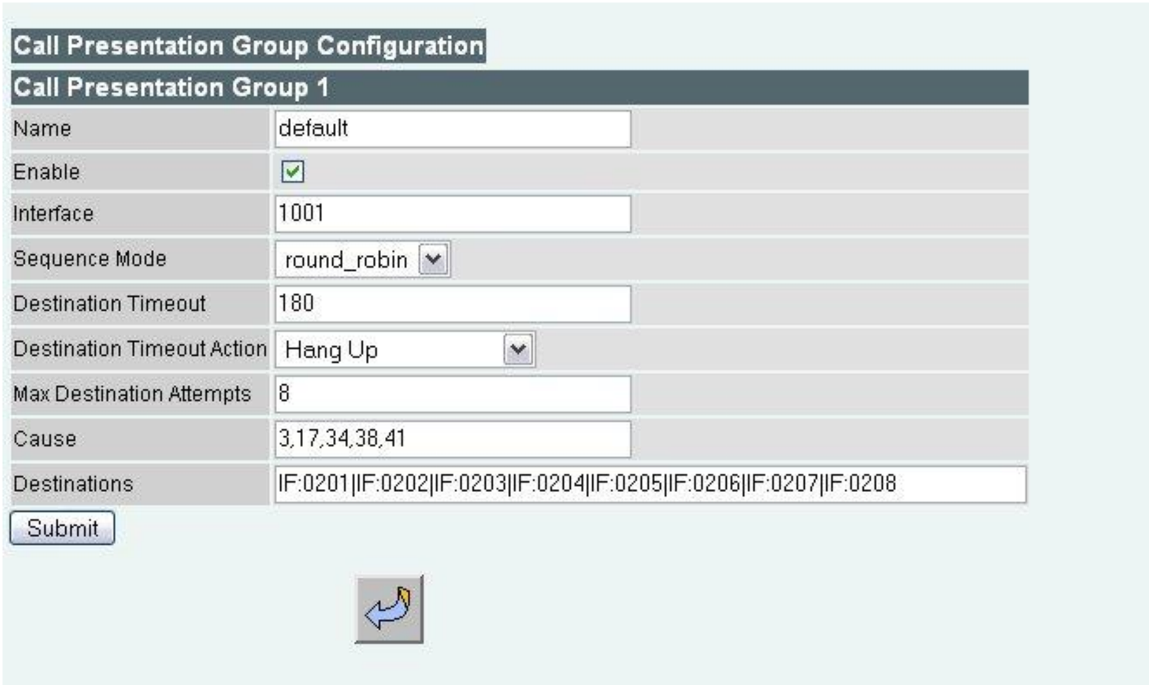
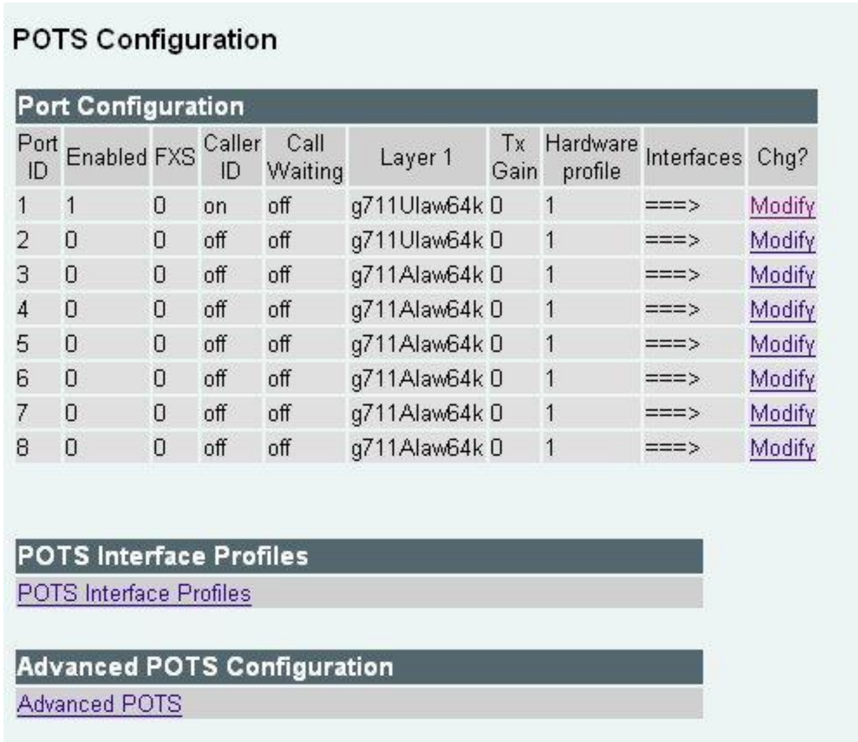


Figure 6 - In this case there may be a number of site specific changes. This screen defines a logical interface 1001 which contains 8 trunks. Generally all physical trunks can be in the **Destinations** field since they will not be selected if they have not been enabled. However if a trunk is only being used for incoming calls, for example incoming faxes, 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. The **Sequence Mode** determines the roll over behavior or how the next outgoing trunk is chosen. The modes include: linear up, round robin and random. If the trunks are configured as an incoming hunt group by the carrier then it is generally best to hunt in the opposite order for outgoing calls.

3.5 POTS

The factory default settings in general do not need changing.

1. Ensure that the Layer 1 settings are set for North American as opposed to Europe: g711Ulaw64k. See Figure 7b.
2. Submit and Save.



POTS Configuration

Port Configuration

Port ID	Enabled	FXS	Caller ID	Call Waiting	Layer 1	Tx Gain	Hardware profile	Interfaces	Chg?
1	1	0	on	off	g711Ulaw64k	0	1	===>	Modify
2	0	0	off	off	g711Ulaw64k	0	1	===>	Modify
3	0	0	off	off	g711Alaw64k	0	1	===>	Modify
4	0	0	off	off	g711Alaw64k	0	1	===>	Modify
5	0	0	off	off	g711Alaw64k	0	1	===>	Modify
6	0	0	off	off	g711Alaw64k	0	1	===>	Modify
7	0	0	off	off	g711Alaw64k	0	1	===>	Modify
8	0	0	off	off	g711Alaw64k	0	1	===>	Modify

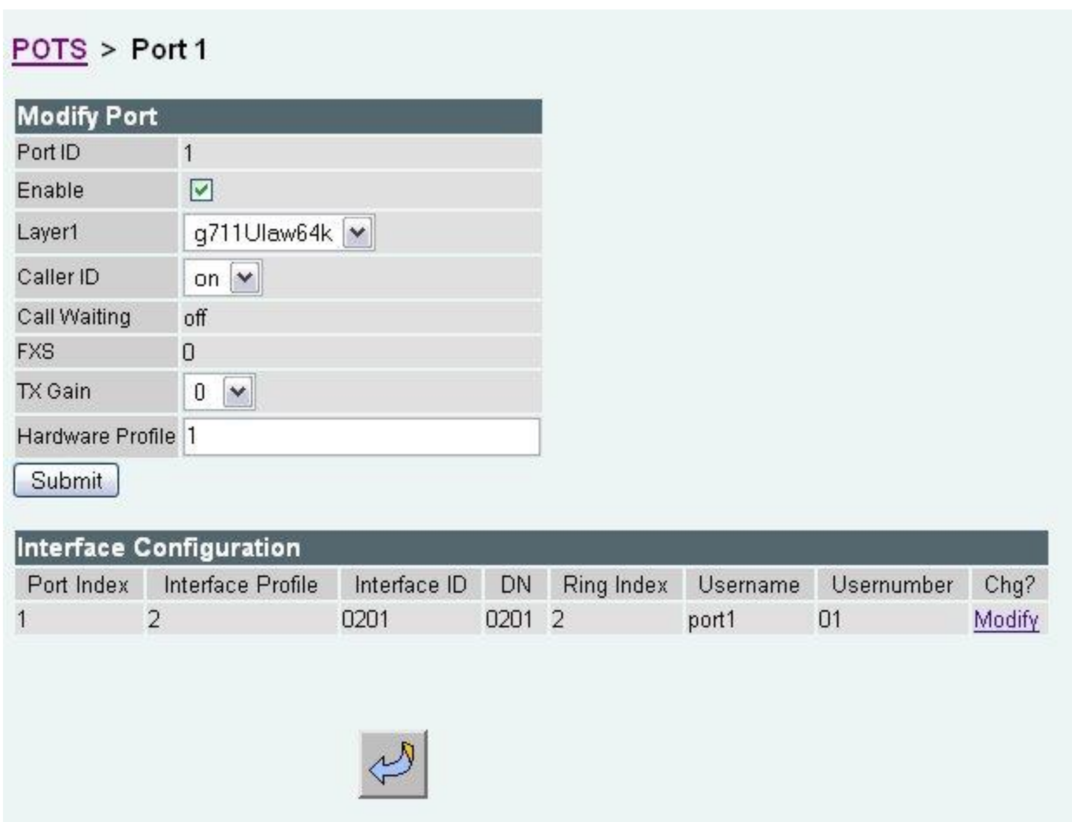
POTS Interface Profiles

POTS Interface Profiles

Advanced POTS Configuration

Advanced POTS

Figure 7a **-** In this example only trunk/port 1 is enabled.



POTS > Port 1

Modify Port

Port ID	1
Enable	☑
Layer1	g711Ulaw64k
Caller ID	on
Call Waiting	off
FXS	0
TX Gain	0
Hardware Profile	1

Submit

Interface Configuration

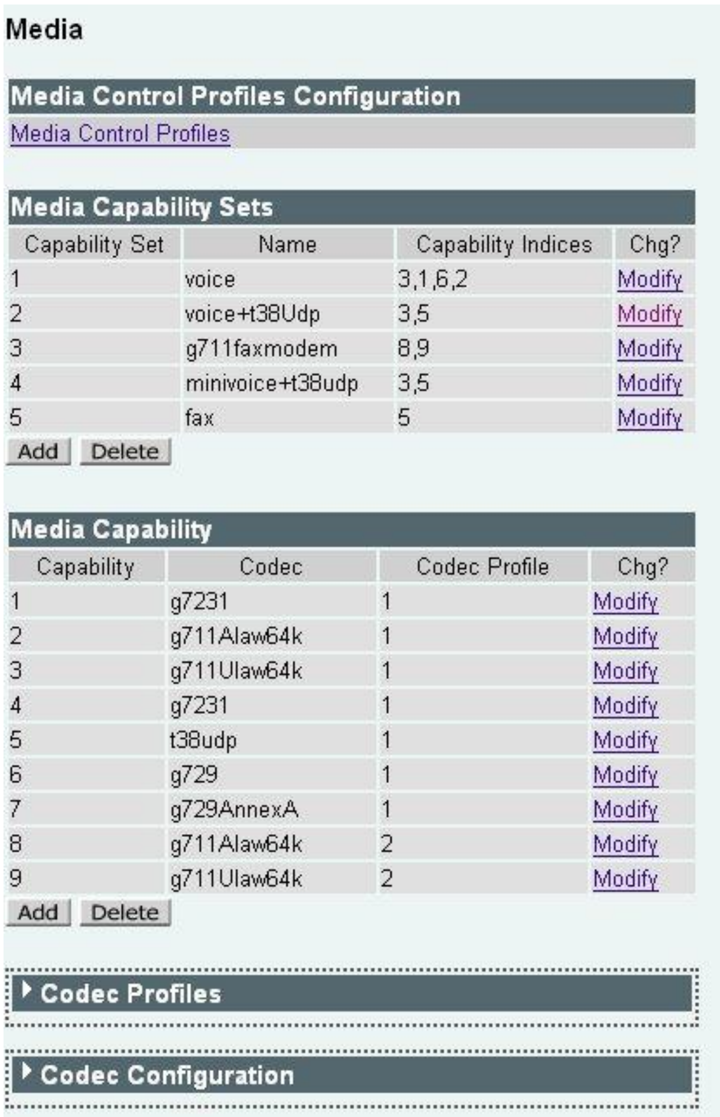
Port Index	Interface Profile	Interface ID	DN	Ring Index	Username	Usernumber	Chg?
1	2	0201	0201	2	port1	01	Modify

Figure 7b

3.6 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 8.



Media

Media Control Profiles Configuration

Media Control Profiles

Media Capability Sets

Capability Set	Name	Capability Indices	Chg?
1	voice	3,1,6,2	Modify
2	voice+t38Udp	3,5	Modify
3	g711faxmodem	8,9	Modify
4	minivoice+t38udp	3,5	Modify
5	fax	5	Modify

Add Delete

Media Capability

Capability	Codec	Codec Profile	Chg?
1	g7231	1	Modify
2	g711Alaw64k	1	Modify
3	g711Ulaw64k	1	Modify
4	g7231	1	Modify
5	t38udp	1	Modify
6	g729	1	Modify
7	g729AnnexA	1	Modify
8	g711Alaw64k	2	Modify
9	g711Ulaw64k	2	Modify

Add Delete

Codec Profiles

Codec Configuration

Figure 8

4. Now go back to **SIP -> Media** and ensure that voice+t38Udp is selected in the drop down box. See Figure 9.

Figure 9

3.7 LAN Configuration

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

1. Go to **LAN** and in the **Calls** section select the correct LAN interface (most likely **LAN 1**) in the drop down menu. See Figure 10.
2. Save and Submit.

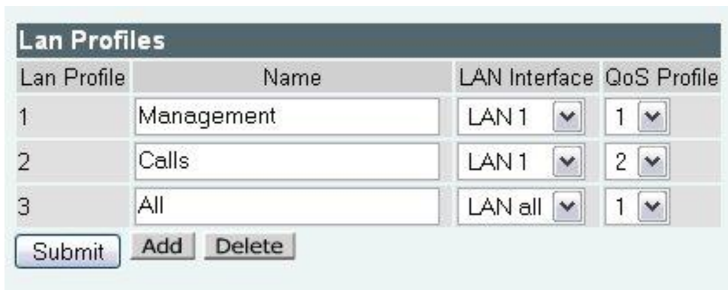


Figure 10

3.8 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)**.

3.9 Backup Configuration

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

4 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 server admin client.

4.1 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.

4.2 Add a SIP Gateway

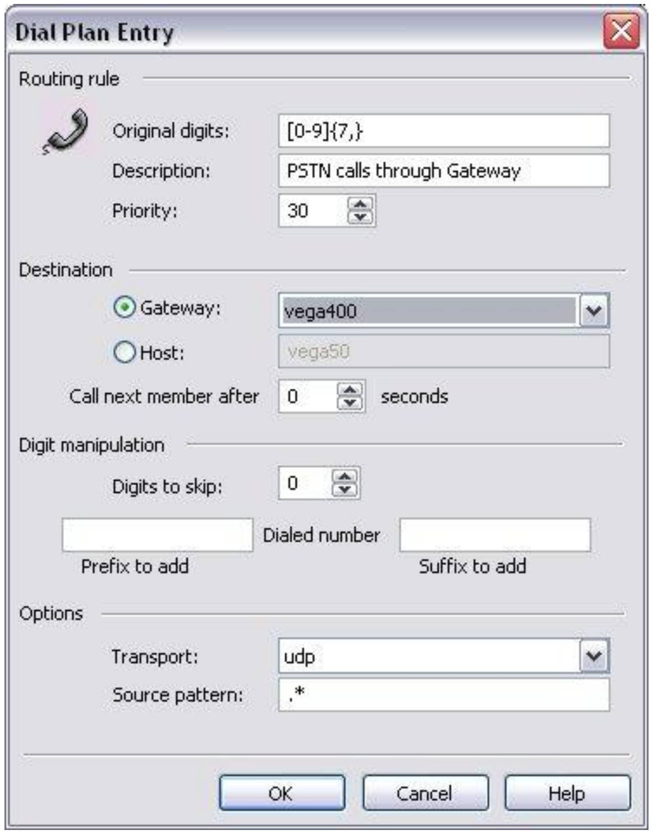
1. Click on **Gateways**.
2. Right click on the right hand panel and select **New Gateway**.
3. Choose **Public Switched Telephone Network (PSTN)** from the list of gateway choices.
4. In the **Host** name field enter the IP address of the Vega gateway.
5. Enter a descriptive name for the gateway.
6.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

4.3 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:



4.4 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.