



NetVanta Unified Communications Technical Note

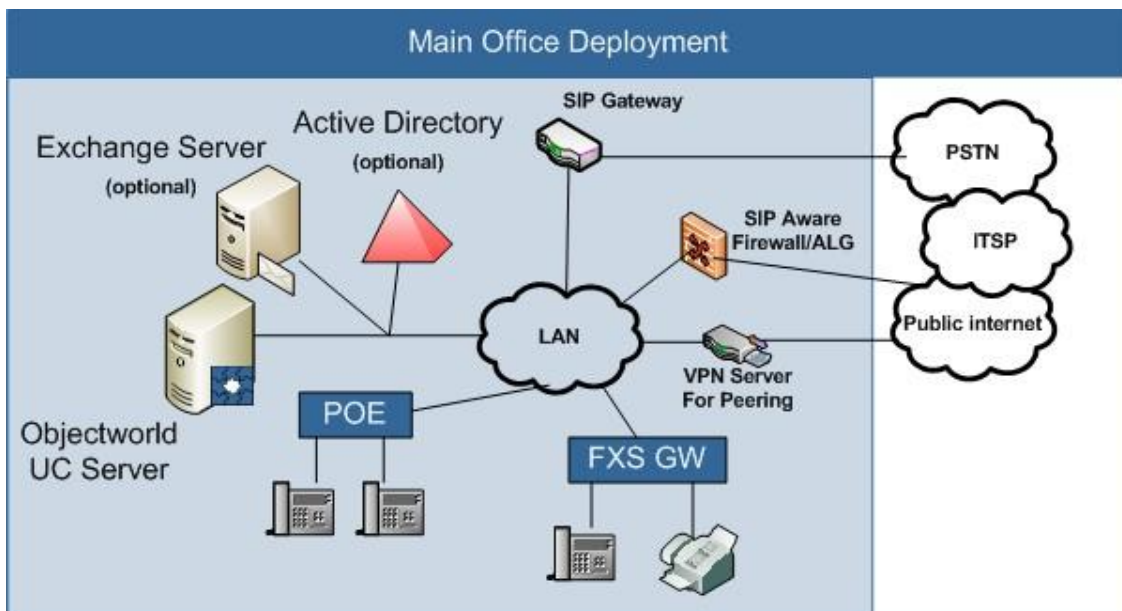
Installing and Configuring the PSTN Gateway

Introduction

A PSTN gateway can be used in NetVanta Unified Communications Server installations to provide a bridge between internal (SIP) phone calls and the phone network (PSTN). A gateway can support analog trunk facilities, digital trunk facilities, or both. In general however a gateway supports one or the other – analog or digital.

A gateway works in conjunction with the SIP Proxy and Registrar embodied in the UC server. All telephony services are provided by the SIP Gateway, SIP Telephones, SIP Proxy and the Core Application Services, which all work together.

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



This document provides an overview of the general setup required for a gateway. Details of how to configure a specific gateway are provided in other technical notes. ADTRAN's documentation 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.

Definition of Terms

PSTN

The **Public Switched Telephone Network** is best thought of as the telephone network. It used to be exclusively provided by Bell but is now an interconnected network of a large number of traditional and cell phone carriers.

PSTN Gateway

A SIP to PSTN gateway is a SIP device on one side and a traditional telephony device on the other. Incoming traditional phone calls are received by the trunk side of the gateway and sent out as SIP calls over the Ethernet interface of the gateway. The reverse happens for calls originating on an internal SIP device directed to the outside PSTN network.

Trunks

A standard analog telephone connects to the carrier's network over a trunk. A key distinction, in the traditional analog case, is that the carrier generates a ringing voltage to the device on the other end. This causes the 'bell' to ring when a call is initiated. The port on a gateway that connects to the network in this way is called a Foreign Exchange Office or **FXO** port. It must *accept* ringing from the carrier.

A port on a gateway that performs the reverse function is called a Foreign Exchange Subscriber or **FXS** port. This port provides the same functions as the carrier. A traditional analog phone, when plugged into this port, in conjunction with the gateway, is converted into a SIP device. A key distinction is that an FXS port must *generate* ringing to the locally attached phone.

Loop Start Analog Trunk

A type of 2-wire analog trunk where on-hook and off-hook signaling is implemented by modifying the voltage on the line.

Ground Start Analog Trunk

A type of 2-wire analog trunk where on-hook and off-hook signaling is implemented by temporarily grounding the line.

ISDN

An **Integrated Services Digital Network** is a digital network where call signaling is carried on a separate channel called the D or data channel. The signaling is on behalf of a number of B, or bearer, channels that carry the voice conversations. There may be any number of B channels but commonly there are either 2 for BRI or 23 for T1 PRI.

T1 Digital Trunk

A T1 digital trunk is a 4-wire trunk that provides 1.544 Mbps of digital bandwidth. A T1 trunk provides 24 channels, each of which carries a 64Kbps voice conversation.

Network Side

A T1 trunk has a network side and a customer premises side. This is purely a convention. Normally the carrier equipment is configured at the network side and provides the physical clocking for the link.

Customer Side

The customer side of a T1 link normally does not provide a clock and is slave to the network side.

CAS

A T1 trunk may use **Channel Associated Signaling** where call signaling is performed in band for each channel. In effect, each of the 24 T1 channels operates as a separate trunk.

PRI

A T1 trunk used in a 23 B + D ISDN configuration.

BRI

A **Basic Rate Interface** is a type of ISDN interface where there is 1 D channel and 2 B channels. It is not commonly used in North America and is popular in a number of European countries.

SIP

The **Session Initiation Protocol** is a set of protocols that implements a type of Voice over IP communication. Telephones and PSTN gateways supported by the UC server are SIP devices.

PoE

Power over Ethernet is a method of providing power to Ethernet cables, which eliminates the need to power the device from standard electrical sources.

Codec

A **Coder-Decoder** is a method of converting voice conversations to analog from digital and back. There are a number of codecs with different characteristics. The most important ones for the UC server in North America are the G.711 μ Law and G.729 codecs.

T.38 Fax Transmission

G3 and Super G3 faxes communicate using standard phone lines by transmitting their data using modem techniques. The standard for fax transmission is T.30. This transmission is sensitive to delay and for this reason T.30 cannot be reliably sent over an IP network. T.38 is a standard that encapsulates T.30 into IP packets where each end provides the appropriate local signals to its fax machine (spoofing) to ensure that communication continues. There are also transmission redundancy techniques in T.38 to ensure the reliable delivery of the fax data.

Synopsis

The following is a list of considerations when installing a new PSTN gateway. Explanatory details are provided in the sections below.

- The gateway requires a **static** or pseudo static LAN IP address.
- The gateway **does not** have to register with an identity on the UC server.
- DTMF tones **must** be carried out of band using RFC 2833.
- The gateway **must** send and receive faxes with the UC server using the T.38 fax protocol.

- The gateway should direct all PSTN incoming calls to a defined identity on the UC server. This is usually identity 10000.
- A UC server gateway object must be configured with a dial plan entry directing outgoing PSTN calls to the gateway.
- T1 PRI interfaces in North America are **most likely** to be one of the following:
 - T1
 - DMS, ATT or NI signaling
 - ESF framing
 - B8ZS line encoding
 - User or CPE network termination
 - Receive clock from the network
 - Use the G.711 μ Law codec for line transmission.
- Analog FXO trunks should have:
 - One stage dialing
 - Answer supervision
 - Detect incoming caller ID
 - Nominal 600 Ω impedance.
 - Factory default loss and level settings.

Overview of Installation Procedure

To provide its functionality, the gateway must be connected to the internal LAN (a 100 Mbps connection is recommended) and to the PSTN through either analog or digital trunks.

Gateways usually have a number of possible configuration methods. For convenience, ADTRAN provides configuration instructions using the gateway's Web page. 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 might cause problems. **For this reason it is recommended that initially only the changes given by ADTRAN 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 gateway.
2. Mount the gateway.
3. Connect cables.
4. Set the IP address and subnet mask of the gateway.

5. Access the gateway's Web page.
6. Configure the gateway.
7. Save the configuration and reboot the gateway.
8. Back up the configuration.

Documents provided by ADTRAN are intended to be a companion to those provided by the gateway vendor. Both should be used while configuring the gateway.

Gateway Configuration

Software Level

ADTRAN's documentation is always based on the gateway software used in the latest interoperability testing. It is important that the software revision in the gateway matches that tested by ADTRAN. If it does not match, contact the manufacturer's website for the version specified. Failure to do so might result in unexpected behavior.

Set IP Address and Subnet Mask

The gateway must have a LAN IP address that does not change; otherwise the UC server does not know where to direct outgoing calls to the PSTN. The gateway should be configured to have a static IP address or a 'pseudo' static IP address where the gateway uses DHCP but it is always provided with the same address.

SIP Configuration

The gateway must know where to send a call that it receives from the PSTN. 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 can 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 voice path. This is implemented using T.38 fax support. This must be enabled on the gateway.

Analog Gateways

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 be an answering point defined on the UC server for incoming PSTN calls. A ‘standard’ configuration uses a UC server trunk identity of 10000. A general routing entry is required 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, use that extension instead of 10000.

Incoming Call Routing

The analog trunks might each have their own phone number or might be part of an incoming hunt group. If there are separate phone numbers, the call can be routed based on the port.

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 choose a trunk that is there and in service. Only trunks that are actually in use should be enabled.

Outgoing Name and Number Presentation

There is no ability to change the ID on a call by call basis for analog trunks. The number is provided by the carrier.

Trunk Settings

Fortunately most analog FXO gateways are shipped pre-configured with the parameters required for North America. Unfortunately, in terms of parameters to be checked, gateway vendors use different terms to describe the same capability.

In general:

- Ensure that one-stage dialing is used. If two-stage dialing is used, a user making a call first receives dial tone from the trunk at which point they can enter the digits for where they want to call.
- Ensure that answer supervision is enabled. There are many answer supervision methods including loop current detection, loop reversal and voice energy detection.
- Ensure that the gateway is configured to detect incoming caller ID.
- Impedance should be nominally 600Ω.

The trunk loss and level settings should be left at factory default levels. If there are echo or audio level issues, the levels should be adjusted based on recommendations from the vendor and ADTRAN.

Digital Gateways

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 Call Routing

There must be an answering point defined on the UC server for incoming PSTN calls. A ‘standard’ configuration uses 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, it should be used instead of 10000.

Incoming DID Call Routing

A PRI gateway 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-digit number but the last 4 digits of the number being called. In either case, that number can be used to route the call to a different destination.

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. Only interfaces and trunks that are actually in use should be enabled.

Outgoing Name and Number Presentation

When an outgoing call to the PSTN is made over an analog trunk, the calling name and number are generated by the carrier. There is no ability to change these on a call by call basis. A PRI allows the calling party to specify the calling name and number for each call. All digital gateways can be configured to present a calling number based on which extension or identity is calling. Generally, it is entered into a dial plan or a digit modification table.

Some but not all digital gateways also allow the calling name to be changed based on the calling extension or identity. If this is not possible in the digital gateway, there are workarounds where the UC server can provide a name substitution.

Trunk Settings

The DSL / PRI settings are site- and carrier-specific. ADTRAN’s documentation covers the most likely settings for North America.

The following settings must be considered:

- Ensure that Network Topology is set to T1 (most likely for North America).
- Ensure that only the T1 spans that are active are enabled.
- The network type should be set as defined by your carrier. DMS, ATT and NI are the most common settings.
- Framing should be set as defined by your carrier. ESF is the most common.
- Line Encoding should be set as defined by your carrier. B8ZS is the most common.
- Network termination should be set as defined by your carrier. Most likely it should be set to User or CPE side.

- The clock master source should be set as defined by your carrier. Most likely the gateway should not provide the clock source. If the clock source is not configured properly clicks and pops might be heard during a voice call.
- In some cases where there are multiple T1 interfaces on the gateway, a bus master must be chosen. This is a complicated issue and beyond the scope of this document. Further information is provided by the gateway vendor on how to configure this option, if required.
- If a layer 1 or codec is required, it should be set to G.711 µLaw.

Media Configuration

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

LAN Configuration

In general the LAN should be configured correctly already or configuration through the Web page would not be possible. However, if required, ensure that all SIP traffic is routed to the correct LAN port.

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 (July 2007) be disabled if the gateway is a Vegastream gateway.

Backup Configuration

Ensure that the configuration changes are saved on the gateway and that the configuration file is saved in a secure place.

UC Server Configuration

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.

To add 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.

To add a SIP gateway

1. Select **Gateways**.
2. Right-click the right-hand panel and select **New Gateway**.
3. Choose **Public Switched Telephone Network (PSTN)** from the gateway list.
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. There are many possibilities in the routing area. 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 gateway. For example:

Dial Plan Entry

Routing rule

Original digits: [0-9]{7,}

Description: PSTN calls through Gateway

Priority: 30

Destination

Gateway: vega400

Host: vega50

Call next member after: 0 seconds

Digit manipulation

Digits to skip: 0

Prefix to add:

Dialed number:

Suffix to add:

Options

Transport: udp

Source pattern: .*

OK Cancel Help

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. Toll restrictions are explained in detail in the *Managing PBX Configuration Categories > Routing—Toll Restrictions* section.

Reliability and Performance Considerations

UPS

Given the enterprise critical nature of voice communications, ADTRAN recommends that PSTN gateways be powered using an Uninterruptable Power Supply with sufficient hold-over power for the majority of power outages. For this to be effective, the UC server, SIP telephones and the LAN must also be protected. The most convenient way of achieving this for telephones is to provide power using a UPS protected PoE switch.

Traffic Engineering

When two SIP devices communicate with each other they negotiate an agreed codec that is used to carry the voice conversation. In most cases this is the G.711 codec, which uses approximately 100 Kbps in each direction. In a simple installation, this traffic is all intra-LAN between SIP phones and gateways. The LAN must provide sufficient bandwidth to carry the number of simultaneous calls that are handled by the system. A quick calculation shows that this is not an issue for the majority of situations using a 100 Mbps LAN.

The situation might be different when some of the voice conversations are carried inter-LAN, for example over the Internet. In these situations, G.729 can be used, which requires about 20 Kbps in both directions. Note that UC server services, such as auto attendant and voice mail, only communicate using G.711. So G.711 must always be one of the allowed codecs for SIP devices supported by the UC server.

Quality of Service

Having adequate bandwidth is necessary for providing good quality voice conversations; however adequate bandwidth alone might not be enough. Prioritization of voice traffic might be required if other data streams, such as file transfers, delay the voice traffic.

When a voice call is established, communication is end-to-end between the two SIP devices. Quality of service can be provided by having the SIP devices mark the voice packets for priority treatment and / or having the IP switches prioritize the voice data from the SIP end points. There are a number of techniques available, including DiffServ, 802.1P and VLAN.