NetVanta Unified Communications Technical Note

Supporting Multiple PBXs in Hybrid Deployment Models

Application Note for Hybrid Deployments

The goal of this technical note is to ensure that you can leverage the technology that is available within the current PBX environment, as cost effectively as possible.

The UC server can be deployed in an environment where customers want to leverage their existing investment in traditional PBXs *and* want to migrate users to a next-generation UC server SIP architecture.

The NetVanta Unified Communications Server has the native capability to connect to additional PBXs while simultaneously supporting SIP services. The UC server can be deployed as a voicemail server, unified message server, database IVR platform and fax server behind traditional PBXs while simultaneously supporting SIP Telephony. UC server PBX users can still take advantage of Active Directory integration for user management; however, the telephony features of the traditional PBX are managed through its PBXs configuration interface.

Native PBXs currently supported:

- Avaya Merlin Legend/Partner/Magix
- Avaya Definity/Communications Manager
- Avaya IP Office
- Cisco Call Manager
- Mitel SX-200 D/ML/EL, SX-200icp
- Mitel 3300
- Nortel Norstar
- Generic analog integration

One of the benefits of a hybrid integration is that users on either system can take advantage of UC server services: voicemail, auto-attendant, unified messaging, inbound and outbound fax and IVR.

In the case where users have an extension of the traditional PBX and identities on the UC server, those users can access their unified messaging account from any of their configured telephones. In addition, when a voice or fax message is left for a user, all their extensions display a message waiting indication. If a user has an extension on the traditional PBX and on the UC server SIP PBX, both extensions have their message waiting lights toggled.

Ensuring seamless calling between traditional PBX users and UC server SIP users means that all users can place and receive calls between each other.

PBX Voicemail and Unified Messaging Integration

Integrating the UC server with traditional PBXs for unified messaging, voice messaging, auto-attendant, inbound faxing and outbound faxing is done by using a combination of PBX Integration boards, TAPI/Wave, and CTI enablement technologies. Consult the current integration technical notes for a list of supported PBXs.

Inter-PBX Connectivity Technology Choices

There are a variety of technology choices to implement private networking between two disparate communications systems.

Choose any one of the following:

- Digital T1/E1/PRI links
- Analog gateways (Using a combination of FXO and FXS)
- Analog E&M tie lines

The benefits and disadvantages of each method are described below.

Digital T1/E1/PRI Gateway

Digital T1/PRI gateways provide bi-directional, high-density facilities that allow public connection to the central office, as well as communications system interconnectivity.

Pros:

- Calling name and number transparency
- Fast call setup and tear down
- Bi-directional call directions

Cons:

- High cost in the PBX signaling interface card
- High cost in SIP T1/PRI Gateway

Analog Gateway(s) – FXO and FXS

Analog interfaces provide a low-cost alternative to connecting two communications systems. In order to provide bi-directional calling capabilities, two interface types must be used. An FXO gateway terminates on an analog station card. An FXS gateway terminates on a Loop Start trunk card.

Pros:

- Cost effective
- Uses interfaces that are more commonly available on existing communications systems

Cons:

- Caller name and number presentation is not available
- PBX must be able to provide disconnect supervision
- Must provision for simultaneous calls in each direction

Analog E&M – 2 wire and 4 wire

An analog E&M interface provides bi-directional, low-density facilities that allow private network trunking facilities between communications systems.

Pros:

- Bi-directional trunking facilities

Cons:

- Not typically provisioned with a traditional PBX. To use analog E&M trunks, you might have to invest in analog E&M trunk cards for your PBX.

Inter-PBX Connectivity Deployment Architectures

This section explains the different deployment architectures based on the type of technology that you want.

T1/PRI Gateway – "Monkey in the middle"

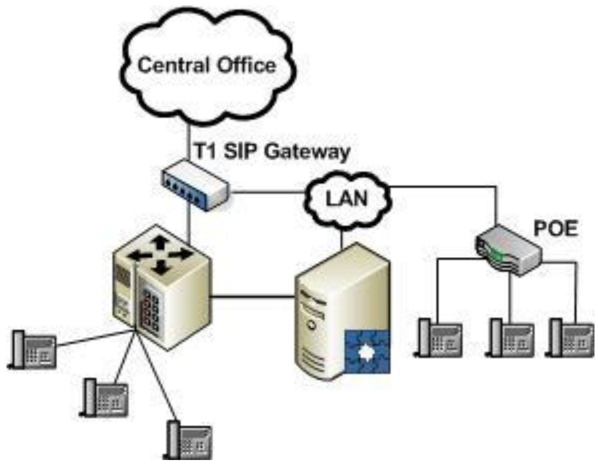
If the customer has an existing T1/PRI service connected to the central office, the gateway can be placed between the PBX T1 connection and the central office. This option is preferred if they do not have an extra T1/PRI interface card on the PBX. This placement allows seamless integration between the existing PBX and the UC server. Since the existing PBX T1/PRI connection is used, there is no additional investment in PBX equipment required.

However, the number of trunks that are configured is determined by the sum of both the number of PSTN lines and the inter-PBX lines that are required to satisfy business requirements. If you currently deploy 23 digital trunks to the PSTN, and you require an additional 6 lines for inter-PBX communications, you have to provide additional trunking facilities.

Use the T1/PRI SIP gateway for the following purposes:

- Continue incoming and outgoing calls between the Central Office and existing PBX. Enable migration of incoming numbers from the CO to terminate on the UC server.
- Allow transparent bi-directional calls between phones on the existing PBX and the UC server. Support for calling name and number is possible in both directions.

- Allow UC server users to access the central office directly through the SIP gateway without having to transit through the existing PBX
- Allow incoming calls to be diverted through the T1 SIP gateway directly to UC server users.
- Eventual phase-out of PBX.



T1/PRI Gateway – "On the side"

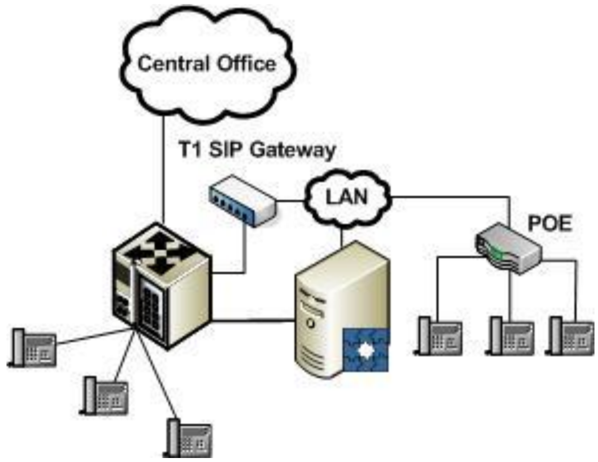
If the customer has an extra T1/PRI interface card on the PBX, this is a good alternative. This method is preferable if the combination of PSTN trunks and inter-PBX trunks exceed that of a single T1/PRI gateway.

Use the T1/PRI SIP gateway for the following purposes:

- Allow transparent bi-directional calls between phones on the existing PBX and UC server. support for calling name and number is possible in both directions.
- Allow UC server users to access the central office, using the existing PBX as a transit. Some capabilities might be restricted when the PBX is the transit gateway. The media "trombones" through the existing PBX.

This method is a good alternative if the goal is to provide physical separation between the T1/PRI services and the inter-connecting trunks. However, as a drawback, all calls between the PBX and UC server SIP PBX "trombone" through the PBX.

In this configuration, the PBX acts as a transit node to access the PSTN. All calls originating from the UC server to the PSTN have to bridge through the PBX. This configuration maintains a distinction between PSTN trunks and inter-PBX trunks. Each call that is made from UC server users transits through the gateway. This includes both local extension-to-extension calling and calls that are intended to terminate on the PSTN.



Analog Gateways – FXO and FXS Gateways

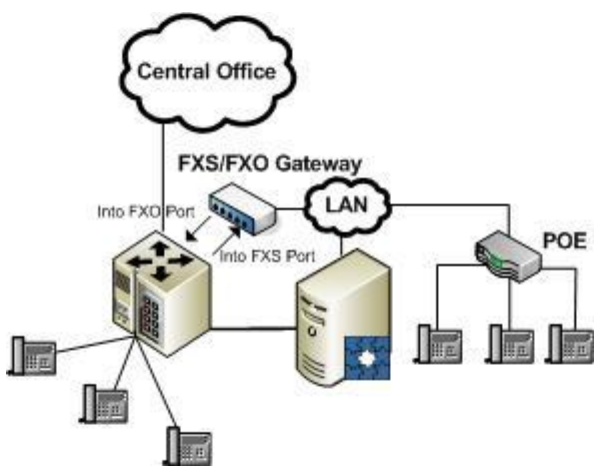
Use analog facilities to connect the PBX to the UC server and allow dialing between communications systems. The analog facilities take advantage of available analog loop start and analog station ports available. The analog facilities do not have the capability of sending the calling name and number information.

Use the FXO/FXS SIP gateway for the following purposes:

- Allow transparent bi-directional calls between phones on the existing PBX and the UC server. Support for calling name and number is possible in both directions.
- Allow UC server users to access the Central Office, using the existing PBX as a transit. Some capabilities might be restricted when the PBX is the transit gateway. The media “trombones” through the existing PBX.

In this configuration, the PBX acts as a transit node to access the PSTN. All calls originating from the UC server to the PSTN have to bridge through the PBX.

NOTE: *Make sure that the PBX provides proper disconnect supervision – the ability for the facility to determine when one of the parties has disconnected. If disconnect supervision is not provided, there is a chance that the trunks might “lock-up,” which must be manually cleared.*



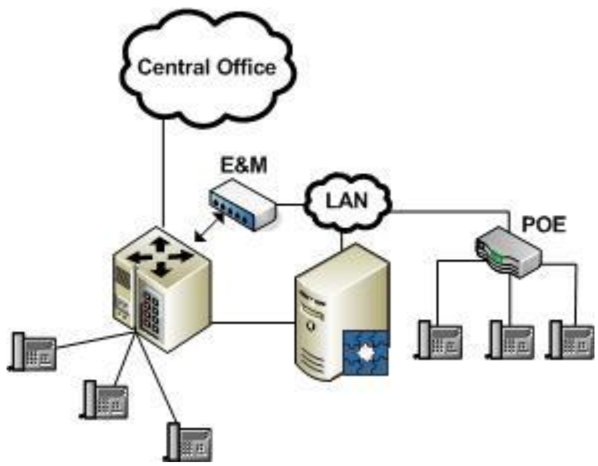
Analog Gateways – E&M Gateways

Use analog E&M facilities to connect the PBX to the UC server and allow dialing between communications systems. The analog facilities take advantage of available analog loop-start and analog station ports available. The analog facilities do not have the capability of sending the calling name and number information.

Use the E&M SIP gateway for the following purposes:

- Allow transparent bi-directional calls between phones on the existing PBX and the UC server. Support for calling name and number is not possible.
- Allow UC server users to access the central office, using the existing PBX as a transit. Some capabilities might be restricted when the PBX is the transit gateway. The media “trombones” through the existing PBX.

In this configuration, the PBX acts as a transit node to access the PSTN. All calls originating from the UC server to the PSTN have to bridge through the PBX.



PBX Considerations

This section details general considerations and best practices when configuring the PBX to support the UC server in an adjunct mode.

Uniform dialing plan

Generally speaking, a uniform dialing plan is required to allow proper configuration of dialing rules between PBXs. The PBX administrator creates a rule so that when a UC server user’s extension is dialed, the call exits the PBX through the configured gateway.

Because you must be able to dial between each office, a uniform dialing plan is a good strategy when connecting systems together. A uniform dialing plan ensures that you have unique extension numbers in all locations. For example, extension range 2000-2999 on one PBX and extension range 3000-3999 on the UC server SIP PBX.

Automated attendants

In order to transfer from an auto-attendant that is answered on a legacy PBX to the UC server, you must include dialing plan entries on both the traditional PBX and the UC server SIP PBX. This ensures that you can dial and transfer from an automated attendant. The dialing plan must include any prefix codes or special dialing entries that ensure one can uniquely transfer to the other.

The UC Administrator must also decide which PBX will answer incoming automated attendant calls. The following table recommends the auto-attendant placement.

Deployment model	Auto-attendant placement	Reason
T1/PRI - Monkey in the middle	UC server	No tandem action. Media is connected through the gateway
T1/PRI - On the Side	The PBX with the most UC server Users	Minimize “tromboning” through PBX.
Analog – FXO and FXS	The PBX with the most UC server users	Minimize “tromboning” through PBX.
Analog E&M	The PBX with the most UC server users	Minimize “tromboning” through PBX.

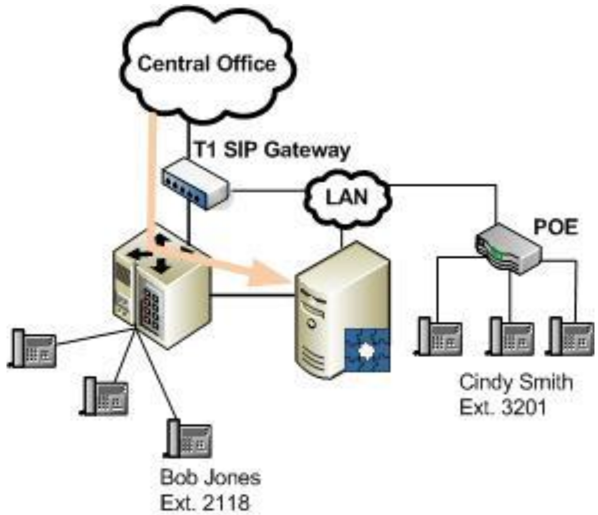
UC Server Configuration

The UC server can be configured to transparently reach users that exist on either PBX in a customer’s environment.

By default, the UC server limits the extension directory and the dial-by-name directory to identities that are configured for a specific PBX. If a user is configured to have an identity on the UC server SIP PBX, but a caller is terminated from an auto-attendant that uses one of the UC server PBX ports, the caller is not able to transfer to an extension by either using dial-by-name or by dialing the extension number of any extension on the UC server PBX.

To overcome this restriction, additional attendant Identities can be created for each of the users to allow dialing from the auto-attendant. The UC server administrator can create attendant identities, which exist on the PBX that callers dial from.

As an example, if the customer's environment includes a UC server SIP PBX and a traditional PBX (the UC server is acting as the unified messaging and auto-attendant server for the traditional PBX) and the following is configured.



An external caller is routed through the PBX and terminates on an auto-attendant connected to the PBX (the UC server port connected to the PBX). The caller wants to dial Cindy Smith at extension 3201 on the UC server SIP PBX.

The UC server administrator must create two identities for Cindy Smith: a user identity on the UC server SIP PBX (3201), and an attendant identity (3201) configured on the traditional PBX. For users on the PBX, the opposite must be configured; Bob Jones would have a user identity (2118) on the PBX and an attendant identity (2118) on the UC server PBX. The following dialog box shows this configuration. For users on the PBX, the opposite must be configured; Bob Jones would have a user identity (2118) on the PBX and an attendant Identity (2118) on the UC server PBX.