ADTRAN®

NetVanta Unified Communications Technical Note

Integrating the Cisco CallManager Fax

Introduction

This technical note describes the fax integration between the NetVanta Unified Communications Server and Cisco CallManager. The UC server integrates with Cisco CallManager using SIP interoperability and T.38 Fax Relay.

The UC server can support the following applications of fax in a Cisco CallManager environment:

- Separate incoming voice and fax numbers (DID Fax)

- Shared incoming voice and fax application

- Outgoing faxing

Summary of fax deployment models:

- Direct SIP Integration using T.38 Relay

- Direct SIP Integration using T.38 Relay with CNG detection

- Direct T1/E1 Connection to Central Office

- Direct T1/E1 Connection via MGCP Gateway

The following table summarizes the deployment technologies and associated capabilities.

Capabilities	Dedicated DID fax numbers	Shared voice and fax numbers	Outgoing fax
Direct SIP Integration using T.38 relay	✓		✓
Direct SIP Integration using T.38 relay with CNG Detection		✓	✓
Direct T1/E1 Connection to Central Office	✓	✓	✓
Direct T1/E1 Connection via MGCP Gateway	✓	✓	✓

Shared incoming voice and fax application requires calls to be pre-answered for a brief period of time to determine if an incoming call is a fax call. The UC server automated attendant pre-answering service requires services to be added to the UC server and configured on an answering identity for each user. The UC server answering point plays ring back (optionally an audio prompt) to allow the system to determine

if the call is a fax call. If the call is not a fax call, the call will be transferred to the associated extension number using a SIP trunk configured on Cisco CallManager.

The diagram associated with each deployment technique will also show an optional voice mail/unified messaging integration that can be deployed. Voice messaging, auto-attendant and Unified Messaging instructions for Cisco CallManager can be accomplished using one of the following techniques:

- TAPI/Wave: Refer to *Cisco CallManager Integration using TAPI/Wave* technical note, available online at *http://kb.adtran.com*.

- Direct SIP: Refer to *Integrating Cisco CallManager Integration using SIP* technical note, available online at *http://kb.adtran.com*.

Cisco does not support fax on TAPI/Wave or direct SIP connectivity.

Required Components

The following are minimum requirements:

1. NetVanta Enterprise Communications Server release 4.4 or higher. The Enterprise Communications edition is required to take advantage of advanced routing rules to support fax integration with Cisco CallManager.

2. A user requiring pre-answering to determine if a call is a voice call or a fax call is recommended to be implemented as Attendant Identity.

3. Cisco CallManager 5.1 or higher.

4. Cisco IOS Minimum released version that supports SIP and SIP T.38 Fax. Information is available on the Cisco website.

Additional References

This document assumes that the reader has knowledge of Cisco CallManager and Cisco IOS Gateways.

It is advisable to also consult the following references for a detailed understanding of fax interoperability with Cisco CallManager.

1. Cisco website: *Cisco Fax Services over IP Application Guide*

2. Cisco website: *TCL IVR API Version 2.0 Programming Guide*

3. *Cisco CallManager Integration using TAPI/Wave* technical note, available online at *http://kb.adtran.com*.

4. *Integrating Cisco CallManager Integration using SIP* technical note, available online at *http://kb.adtran.com*.

Deployment Models

The UC server can integrate with Cisco CallManager in a variety of ways, depending on the desired capabilities and the current voice and data environment.

Direct SIP Integration Using T.38 Relay

Direct SIP Integration using T.38 relay provides direct T.38 fax integration with dedicated DID numbers routed via SIP to the UC server. Cisco IOS Gateways do not provide CNG Detection in their default configurations. The Cisco gateway must be configured as a SIP dial peer with the dial peer routing configured to route incoming DID fax calls directly to the UC server while routing voice calls to Cisco CallManager. The UC server can also provide voice mail and unified messaging integration using a separate direct SIP connection.

A Cisco gateway is used to terminate calls from the central office (PSTN connection via T1/E1 PRI, BRI or FXO interfaces) and selectively redirects incoming fax telephone numbers to the UC server. The called number (DID) will be used to route calls to either Cisco CallManager for voice or to the UC server for fax. The Cisco IOS gateway must be configured to use SIP dial-peer routing. MGCP controlled gateways are not supported for faxing applications. Fax calls that terminate on the UC server must have both the G.711ulaw and T.38 fax codec enabled.
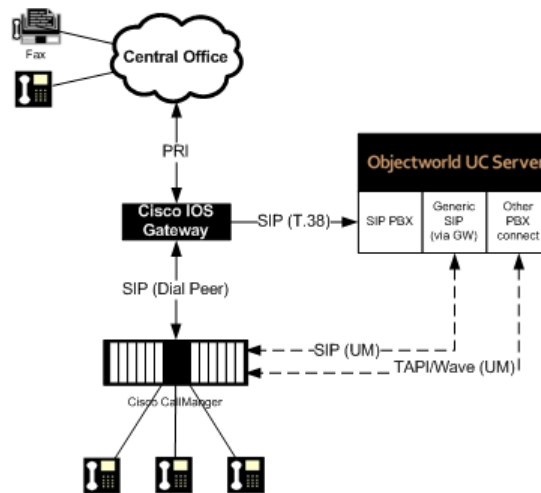
Configuration

Configuring Cisco IOS Inbound POTS Dial Peer for Separate Voice and Fax Calls

Perform the following steps on the Cisco gateway to accept incoming calls on the PSTN interface.

1. Log onto the Cisco IOS Gateway

2. Enter configuration mode

3. **dial-peer voice** <tag> **pots**

4. **direct-inward-dial**

5. **incoming called-number** *<string>*

6. exit

Configuring Cisco IOS Outbound VoIP dial Peer for Voice Calls

Perform the following steps on the Cisco gateway to route calls to the UC server pre-answering service.

1. Log onto the Cisco Gateway

2. Enter configuration mode

3. **destination pattern** *<voice routing prefix>….*

The configured voice routing prefix must match a routing pattern in Cisco CallManager.

4. **session protocol sipv2**

5. **session target ipv4**:*<IP address of Cisco CallManager>*

6. **dtmf-relay rtp-nte**

7. **codec g711ulaw**

8. **fax rate disable**

9. exit

Configuring Cisco IOS Outbound VoIP Dial Peer for Fax Calls

Perform the following steps on the Cisco gateway to route calls to the UC server fax answering behavior.

1. Log onto the Cisco Gateway

2. Enter configuration mode

3. **destination pattern** *<fax routing prefix>….*

The configured fax routing prefix must match an identity as created in *Creating an Identity for the Fax Service on page 17*. The voice routing prefix will be associated with an identity on the UC server.

4. **session protocol sipv2**

5. **session target ipv4**:*<IP address of the UC server>*

6. **dtmf-relay rtp-nte**

7. **fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none**

8. exit

Example – Minimum Configuration Settings

The following example shows an incoming 4-digit DID range starting with a digit 7. Voice DID numbers start with 72 whereas dedicated DID fax numbers start with 71.

This example uses the following information

- Incoming DID called number pattern: 7xxx

- UC server IP address: 192.168.8.54

- Cisco CallManager IP address: 192.168.8.129

- Voice routing prefix: 71

- Fax routing prefix: 329

```
dial-peer voice 2850 pots

incoming called-number 7...

 direct-inward-dial

!

dial-peer voice 2851 voip

description Incoming Voice calls to CCM

 destination-pattern 71..

 session protocol sipv2

 session target ipv4:192.168.8.129

 codec g711ulaw

!

dial-peer voice 2852 voip

description Incoming Fax Calls to UC Server

 destination-pattern 329..

 session protocol sipv2

 session target ipv4:192.168.8.54

 codec g711ulaw

 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none

!

sip-ua

!
```

Direct SIP Integration using T.38 Relay with CNG Detection

Direct SIP Integration using T.38 relay with CNG detection allows a single incoming telephone number to receive voice and fax calls. All incoming calls will be pre-answered to determine if the incoming caller is a fax machine that generated CNG tones. If CNG tone is not detected, the caller will be transferred to

the associated user's extension over a configured SIP trunk between the UC server and Cisco CallManager. The pre-answer service requires the database integration feature to be enabled on the UC server and for the UC server administrator to associate incoming DID called numbers to an internal extension on Cisco CallManager.

In order for a Cisco IOS gateway to detect CNG tones an application (TCL application script) must be loaded onto the Cisco gateway. TCL is a Cisco IVR programming language. The TCL Script that will perform the CNG detection does the following high level functions.

1. Direct incoming POTS call an application service in the UC server.

2. The incoming call is redirected by the CNG-detect TCL script to a UC server pre-answering service. The TCL script modifies the original called number (DNIS) by adding some preceding routing digits in front of the original called number and then directs it to the UC server using a SIP VoIP dial peer. This call is then answered by the UC server by an application service that plays a ring back sound (optionally it can be configured to play an audio prompt) for a short period of time. After the audio is played the service will then perform a blind transfer to the associated Cisco telephone extension.

3. If CNG tone is detected by the Cisco gateway, the call to the pre-answering service is terminated and another call is made to the fax answering service. The Cisco gateway will direct the call to a SIP VoIP dial peer by inserting routing digits in front of the original called number.

4. The UC server will then route the call to a user's fax identity that is associated with a UC server user.
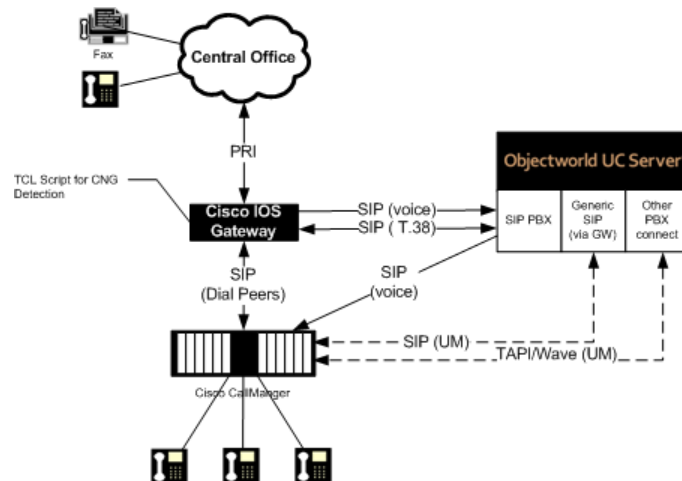
Configuration

Loading and Configuring the Call Application on the IOS Gateway

Perform the following steps on the UC server computer platform:

1. Locate the TCL script *app_fax_detect.2.1.2.3.tcl* available on the UC server installation media. The original tcl script is available from Cisco's website.

2. Copy the *app_fax_detect.2.1.2.3.tcl* file to the following directory '*C:\Program Files\ADTRAN\NetVanta UC Server\Data\TFTP*'

Perform the following steps on the Cisco gateway:

1. Log onto the Cisco IOS Gateway

2. Enter configuration mode

3. **call application voice *application-name location*.** (e.g. call application voice cng-detect tftp://192.168.8.54/ *app_fax_detect.2.1.2.3.tcl*)

4. **application service CNG-detect** tftp://*<IP address of UC Server>*/ *app_fax_detect.2.1.2.3.tcl*

5. **application service CNG-detect param mode** connect-first

   The connect-first mode instructs the Cisco IOS Gateway to route the call to the routed destination and analyze for the presence of CNG tone after the call is answered. If CNG Tone is detected, the call will be disconnected and then a new call will be re-initiated to the fax answering behavior.

6. **application service CNG-detect param fax-prefix** *<fax routing prefix>*

   If the Cisco gateway detects fax, the original voice call will be terminated and a new call will be initiated to the fax destination. The *fax routing prefix* is used to route the call to individual user's fax answering behavior.

Configuring Cisco IOS Inbound POTS Dial Peer

Perform the following steps on the Cisco gateway:

1. Log onto the Cisco IOS Gateway

2. Enter configuration mode

3. **dial-peer voice** <tag> **pots**

4. **service** *cng-detect*

5. **direct-inward-dial**

6. **incoming called-number** *<string>*

   Where string is a Cisco routing character that specifically identifies either a single number or multiple numbers that will be answered the UC server's pre-answering attendant identity.

7. exit

Configuring Cisco IOS Outbound VoIP Dial Peer for Pre-answer

Perform the following steps on the Cisco gateway.

1. Log onto the Cisco Gateway

2. Enter configuration mode

3. **destination pattern** *<voice routing prefix>*

The configured *voice routing prefix* must match an identity as created in *Creating an Identity for the Pre-Answer Service on page 17*. The voice routing prefix will be associated with an identity on the UC server.

4. **session protocol sipv2**

5. **session target ipv4***:<IP address of the UC server>*

6. **dtmf-relay rtp-nte**

7. **codec g711ulaw**

8. **fax rate disable**

Configuring Cisco IOS outbOund VoIP Dial Peer for Fax Answer

Perform the following steps on the Cisco gateway.

1. Log onto the Cisco Gateway

2. **destination pattern** *<fax routing prefix>….*

The configured fax routing prefix must match an identity as created in *Creating an Identity for the Fax Service on page 17*. The voice routing prefix will be associated with an identity on the UC server..

3. **session protocol sipv2**

4. **session target ipv4***:<IP address of the UC server>*

5. **dtmf-relay rtp-nte**

6. **fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none**

7. exit

Configuring Cisco IOS default Outbound VoIP Dial Peer for Transfer

Perform the following steps on the Cisco gateway.

1. Log onto the Cisco Gateway

2. **destination-pattern** .

The configured *voice routing prefix* must match an identity as created in *Creating an Identity for the Pre-Answer Service on page 17*. The voice routing prefix will be associated with an identity on the UC server.

3. **session protocol sipv2**

4. **session target ipv4***:<IP address of the UC server>*

5. **dtmf-relay rtp-nte**

6. **fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none**

7. exit

Example

This example uses the following information:

- Incoming DID called number pattern: 28xx (2800 to 2899)

- UC server IP address: 192.168.8.54

- Voice routing prefix: 46

- Fax routing prefix: 329

```
dial-peer voice 2800 pots
  description POTS Calls to Datsun
   service cng-detect
   destination-pattern .
!
dial-peer voice 2850 voip
  description Default pattern for transfers
     destination-pattern .
     session protocol sipv2
     session target ipv4:192.168.8.54
     dtmf-relay rtp-nte
     codec g711ulaw
!
dial-peer voice 2851 voip
   description SIP to DATSUN
   destination-pattern 46....
   session protocol sipv2
   session target ipv4:192.168.8.54
   dtmf-relay rtp-nte
   codec g711ulaw
   fax rate disable
!
dial-peer voice 2852 voip
   description FAX: SIP to Datsun
   destination-pattern 329....
   session protocol sipv2
   session target ipv4:192.168.8.54
   codec g711ulaw
   fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
!

sip-ua
!
```

Direct T1/E1 PRI Integration to Central Office

Direct T1/E1 PRI integration to a central office leverages dedicated facilities terminating on an interoperable SIP PSTN Gateway. This integration can provide both dedicated fax integration (DID fax) as well as combined voice and fax services. It is also possible to route voice calls through the UC server SIP PBX using SIP trunks between the UC server and Cisco CallManager. Fax identities are created on the UC server to terminate voice and fax calls.
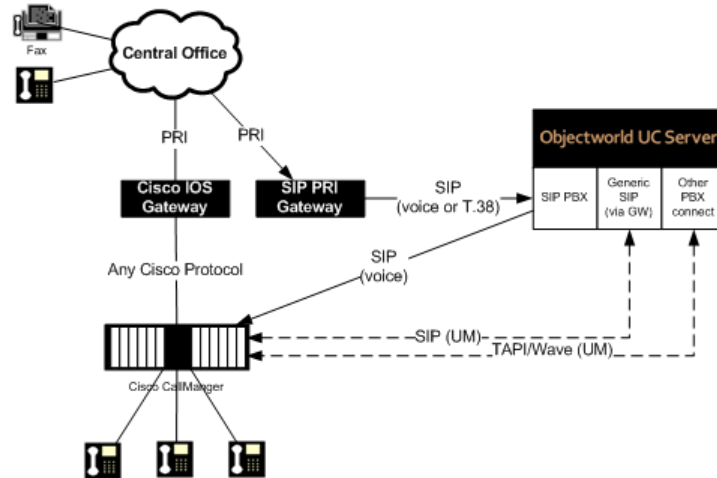
Configuration

This integration technique does not require Cisco IOS Gateway configuration. Two configuration options are available depending on whether or not you are planning to use a single number for voice and fax or are planning to use a dedicated incoming number to associate with a fax behavior.

The high level configuration steps are:

1. Configure the SIP PRI Gateway in accordance to NetVanta guidelines. Technical notes are available online at *http://kb.adtran.com* or on the media that is shipped with the product.

If a single number is used for voice and fax:

2. *Creating a Database in the UC Server – Mapping Pre-answer Numbers to Target Extensions on page 14*.

3. *Creating a Fax Pre-answer Service on page 16*.

4. *Creating an Identity for the Pre-Answer Service on page 17*.

5. *Creating an Identity for the Fax Service on page 17*.

If a dedicated incoming number is used for fax:

2. *Creating an Identity for the Fax Service on page 17*.

Direct T1/E1 Connection via MGCP Gateway

Direct T1/E1 PRI Integration via MGCP Gateway is recommended if the Cisco IOS platform cannot be reconfigured to support SIP. This integration requires additional T1/E1 PRI gateways (Cisco IOS Gateway T1/E1 PRI Span and a corresponding SIP PSTN Gateway). This integration can provide both dedicated fax integration (DID fax) as well as combined voice and fax services. SIP Trunks between the UC server and Cisco CallManager are used to avoid hair-pinning (using two channels on the PRI for a single call) calls being redirected through the SIP PRI gateway.
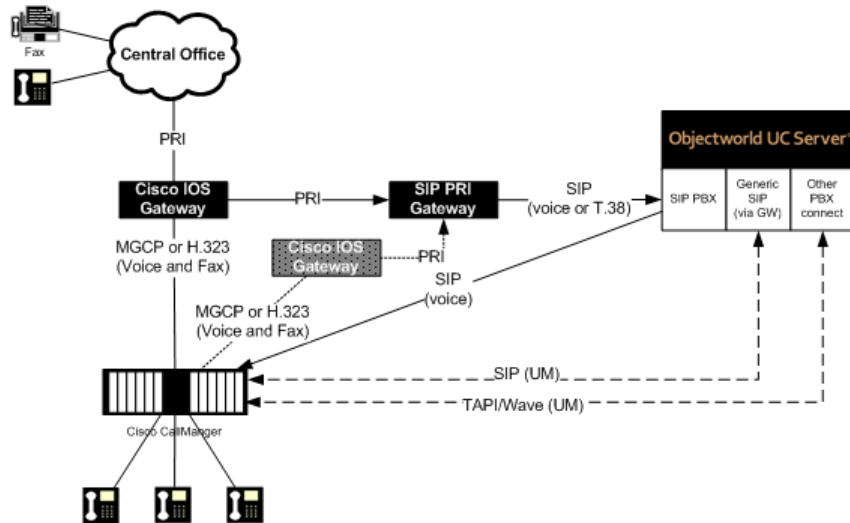
Configuration

The high level configuration steps are:

1. Configure the SIP PRI Gateway in accordance to NetVanta guidelines. Technical notes are available at *http://kb.adtran.com* or on the media that is shipped with the product.

2. Configure the Cisco IOS Route according to Cisco recommendations to ensure that faxes can be relayed between trunk facilities that are between Cisco IOS Gateways. It is beyond the scope of this document to provide instructions on how to configure T.38 relay for non-SIP Cisco IOS Gateways.

If a single number is used for voice and fax:

3. *Creating a Database in the UC Server – Mapping Pre-answer Numbers to Target Extensions on page 14*.

4. *Creating a Fax Pre-answer Service on page 16*.

5. *Creating an Identity for the Pre-Answer Service on page 17*.

6. *Creating an Identity for the Fax Service on page 17*.

If a dedicated incoming number is used for fax

3. *Creating an Identity for the Fax Service on page 17*.

Configuring Cisco CallManager

This section highlights specific Cisco configuration to provide transparent dialing between CallManager and the UC server.

Creating a SIP Trunk to Route Calls from Cisco CallManager to the UC Server

To create a SIP Trunk in Cisco CallManager to route calls to the UC server

1. Log into Cisco Unified CM Administration.

2. Navigate to **Device > Trunk**.

3. Select **Add New**.

4. Select **SIP Trunk** from the **Trunk Type** drop down list.

5. Select **SIP** from the **Device Protocol** drop down list.

6. Select **Next**.

7. Enter a **Device Name** such as "UCServer_SIP_Trunk".

8. Select a **Device Pool** from the drop down list.

9. Select a media resource group list from the **Media Resource Group List** drop down list.

10. Select a location from the **Location** drop down list.

11. Ensure **Media Termination Point Required** is checked.

12. Ensure **Redirecting Diversion Header Delivery – Inbound** is not checked.

13. Select **Originator** from the **Calling Party Selection** drop down list.

14. Ensure **Redirecting Diversion Header Delivery – Outbound** is not checked.

15. Enter the UC server's IP address or FQDN for the **Destination Address**.

16. Enter 5060 for the **Destination Port**.

17. Select **711uLaw** from the **Preferred Originating Codec** drop down list.

18. Select a SIP trunk security profile from the **SIP Trunk Security Profile** drop down list.

19. Select a SIP profile from the **SIP Profile** drop down list.

20. Select **RFC 2833** from the **DTMF Signaling Method** drop down list.
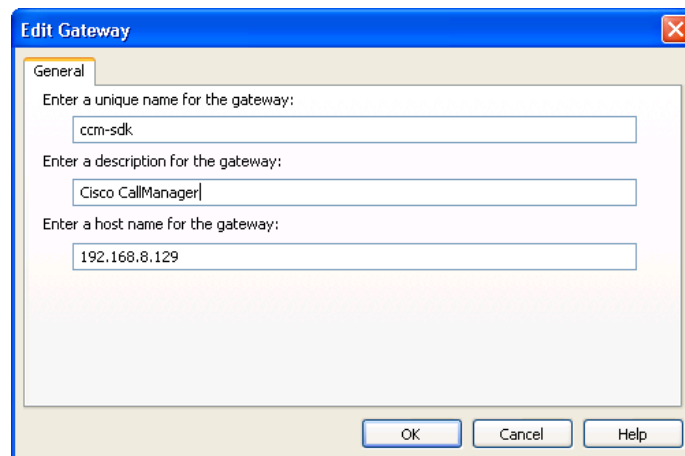
21. Select **Save**.

Configuring the UC Server

This section highlights the configuration steps that can be performed on the UC server. Some of the detailed steps may be optional. The subsections highlighted in this section are referenced in earlier sections and are not intended to be followed in any sequence except as outlined in the above instructions.

Creating SIP Routing Rule to Cisco CallManager

Adding a gateway

To add a gateway

1. From the Administration Navigation pane, select **Gateways**.

2. Right-click in the Content pane and select **New Gateway**.

3. Follow the wizard instructions and include the following information:

    a. Select SIP Peer

    b. Hostname or IP address of Cisco CallManager

    c. Unique Name of the gateway

    d. Select **UC Server** as the associated gateway

    e. Enter an appropriate description

Modifying the Dial Plan
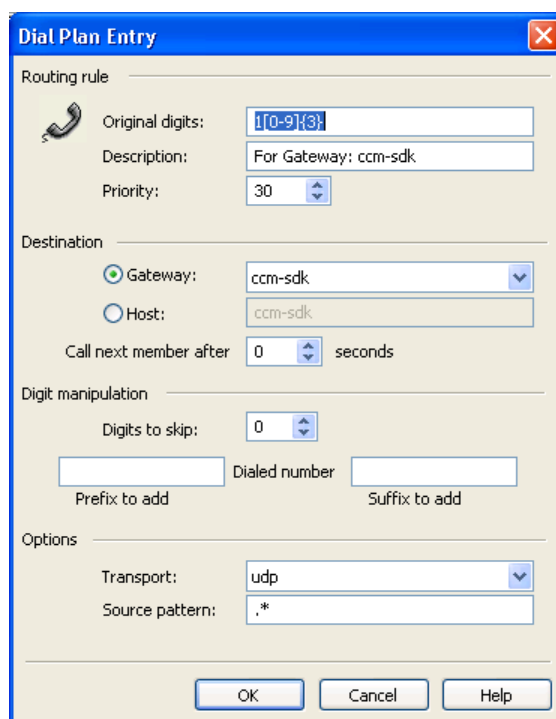
The UC server dial plan will be modified to allow calls that are transferred from the UC server to terminate on Cisco CallManager configured telephones.

Creating a Dial Plan Entry to Route Calls Between the UC Server and Cisco CallManager

The following dial plan entry routes incoming call where the destination is a 4 digit numbers that start with the digit 1 to Cisco CallManager. This example assumes that Cisco CallManager telephone extensions are in the 1xxx range.

To create a dial plan entry:

1. From the Administration Navigation pane, select **Communications Systems > UC Server > Routing**.

2. Right-click in the Contents Pane and select **Add.**

Creating a Database in the UC Server – Mapping Pre-answer Numbers to Target Extensions

Follow the instructions below to create a new data source that allows an incoming DID number to be mapped to extension numbers. This table will define a mapping between the incoming called number and the associated extension. If the call is not a fax – no CNG tone is detected – then the call will be transferred to the associated extension using the SIP trunk.

Creating a Data Source on the UC Server Platform

A database is used to match incoming pre-answer called numbers and match to a target extension number. The database in this technical note is using personal tables; however, an ODBC database can also be used.

To create the database:

1. Create a User Data Table in the UC server administrator. You can consult the *NetVanta Unified Communications Server Administrator Guide*, available online at *http://kb.adtran.com*, for specific instructions. The following fields and names are referenced in the service:

   - Called Number

   - Transfer Extension

   - Fax Extension

2. The UC server administrator can manage the incoming called number and associated transfer telephone numbers by editing the Fax Pre-Answer User Data Source.
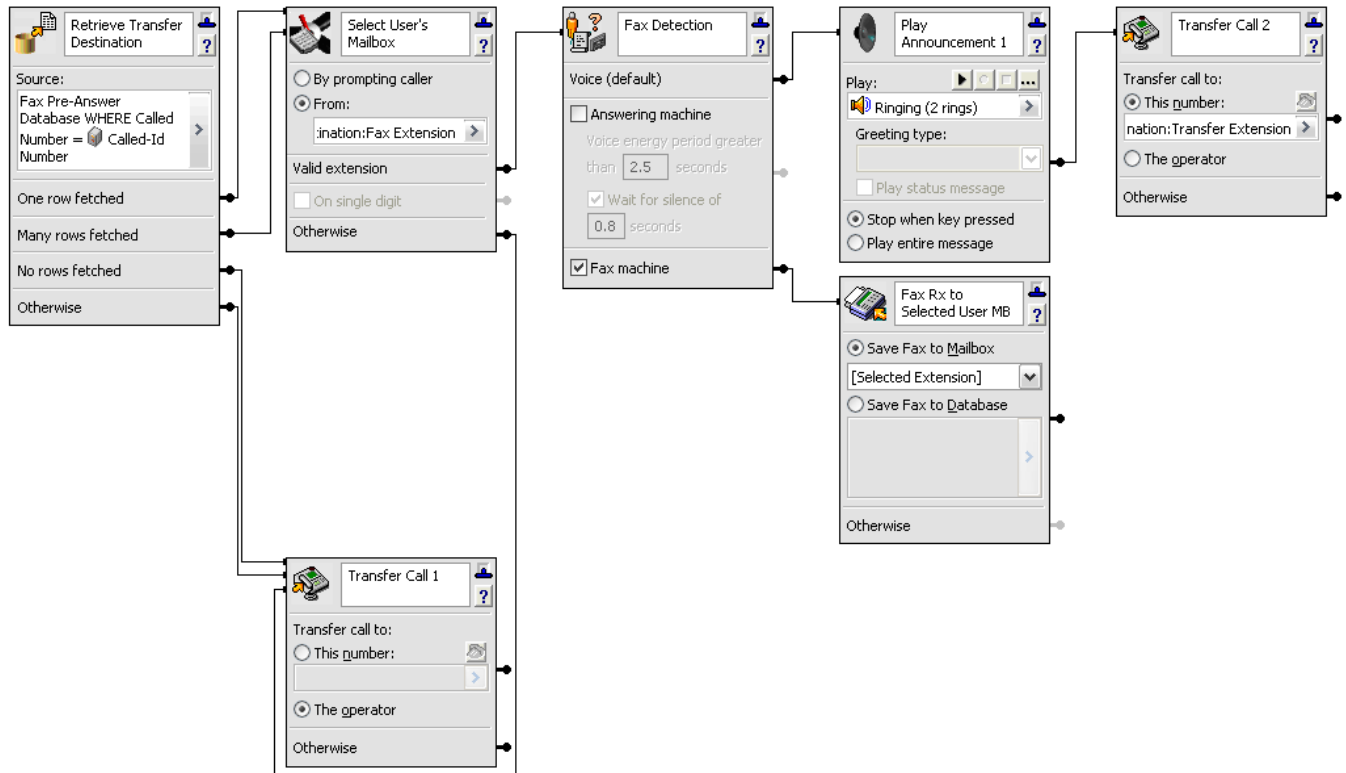
Creating a Fax Pre-answer Service

The fax pre-answer service is only required if you want to provide voice and fax on the same incoming called number. You can skip this step if you have separate incoming numbers for voice and fax.

The fax pre-answer service performs a database 'fetch data' that matches the original called number (represented as Caller-ID Number) to an associated extension. If there is nothing matching the called number in the database, the service will transfer the call to the operator. The "Ringing (2 rings).wav" audio file is available as part of this integration note.

To create the Pre-answer service

1. Copy the "Ringing (2 rings).wav" to the
   C:\Program Files\ADTRAN\NetVanta UC Server\Data\Users\00000000\Announcements\Shared
   directory on the UC server platform.

2. From the Administration Services Navigation pane, navigate to the **Shared** services folder.

3. Right-click and select **New > Service**

4. Name the service *Fax Pre-answer*

5. Double-click on the newly created service to open the Service Editor

6. Create the service as defined in the picture above.

Creating an Identity for the Pre-Answer Service

Each user that requires a single number for both voice and fax will require an attendant identity associated on the UC server – SIP PBX with the call answering behavior configured to service created in *Creating a Fax Pre-answer Service on page 16*.

To create an identity for pre-answer service:

1. From the Administration Navigation pane, select **Identities**.

2. Right-click the Content pane and select **New Identity**

3. The new identity wizard will be launched.

4. Select the **UC Server** communications system.

5. Select **Attendant Service** and select Next.

6. Configure a Name and Address. The address that you enter is the *<Voice Routing Prefix><User's Called Number>*.

7. Check the **Run Service** radio button, and navigate to the shared service that was created in *Creating a Fax Pre-answer Service on page 16.*

8. Select Finish.

Creating an Identity for the Fax Service

Creating an identity for the fax answering behavior is required for every user that needs to receive faxes.

To create an identity for the pre-answer service:

1. From the Administration Navigation pane, select **Users**.

2. Right-click the user in the Content pane that you want to create Receive fax behavior and select **Create Identity.**

3. Select the **UC Server** communications system.

4. Select **Attendant Service** and select Next.

5. Configure a Name and Address.

   a. If this user has a dedicated DID fax number, the address that you enter is the *<Fax Routing Prefix><User's Called Number>*.

   b. If the user has a shared voice and fax number, the address that you enter is the *<Different Prefix><User's Called Number>*.

6. Check the **Receive Fax** radio button.

7. Select Finish.