Integrating VoIP Phones and IP PBX's with VidyoGateway

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I. ABSTRACT

Vidyo recognizes the importance of providing a platform that integrates seamlessly with IP based voice and video communications systems that are deployed on a client network. We believe that our technology and patented architecture provides capabilities, quality, and scalability that have never been experienced with traditional solutions, however it is a high priority to enable clients to leverage their investments in existing technologies in a way that does not introduce unnecessary complexity to end users. This paper will describe in detail the VidyoGateway, a product developed specifically for this purpose, focusing on the integration of PBX systems and Voice over IP (VoIP) technologies with our solution. It will highlight the operations and configurations of the VidyoGateway, identify supported protocols, illustrate typical network topologies, and define the components involved. In addition, it will provide sample scenarios and include requirements to configure PBX systems to route calls into VidyoConferences, and provide configuration guidelines for Cisco Unified Call Manager and Trixbox.

II. VIDYOGATEWAY OVERVIEW

The VidyoGateway is an IP-based appliance that enables legacy voice and video systems to connect to VidyoConferences. Focusing this paper on voice integration with IP-based PBX systems, we’ll limit our scope to discussing the capacities, codecs, and signaling protocols that apply to this scenario. Each VidyoGateway supports up to 50 simultaneous voice only connections with additional capacity achieved by stacking multiple systems. The VidyoGateway supports G.711 and G.722 (narrowband and wideband) audio codecs commonly used by VoIP systems, as well as the SPEEX codec used on the Vidyo side. The VidyoGateway supports H.323 and SIP signaling.
as well as Vidyo’s signaling format, based on SIP but enhanced to enable additional capabilities. As calls are placed in either direction, the VidyoGateway performs transcoding between codecs, converts signaling, and enable Vidyo endpoints and VoIP phones to participate in the same conference or point to point call. In addition, the VidyoGateway supports registration with H.323 gatekeepers and SIP servers, and uses E.164 dialing methods that are familiar to most users.

III. NETWORK TOPOLOGY AND DEFINITIONS

Diagram 1a below illustrates a typical network topology for VoIP integration with VidyoConferencing. Elements include VidyoPortal, VidyoRouter, VidyoGateway, an IP PBX, analog POTS phones, VoIP soft phones, dedicated IP phones, VidyoDesktop, and VidyoRoom endpoints.

Before we discuss how these elements connect and communicate with one another, let’s define some terms:

**VidyoPortal** - Server appliance that provides web interface for Vidyo users and administrators and performs management functions for Vidyo endpoints and components. It also performs authentication and enables users to manage contact lists, control their VidyoConference sessions, and initiate outbound calls.

**VidyoRouter** – Server appliance that intelligently and dynamically routes video and audio traffic between endpoints. It is the key component that measures bandwidth and capabilities, adjusts rates and resolutions to optimize quality, and provides resiliency to enable the solution to run smoothly on general purpose networks.

**Soft Phone** - software program used to making telephone calls over IP networks using a computer, rather than using dedicated hardware. A soft phone is usually used with a headset connected to the PC and is capable of communicating with other devices, including the VidyoGateway, using the SIP signaling protocol without requiring a PBX to route calls.
**IP Phone** – Dedicated device that uses VoIP to make calls over IP networks instead of the ordinary PSTN system. The phones use control protocols such as SIP or SCCP and are typically integrated with an IP PBX. This device may include the ability to dial devices and endpoints directly using the SIP protocol.

**POTS Phone** – Analog telephone connecting to the PBX over the public switched telephone network (PSTN).

**IP PBX** – Business telephone system designed to deliver voice communications over a data network and interoperate with the PSTN.

**SIP Trunk** – There are multiple definitions and applications, however for the purposes of this document we will define a SIP trunk as a configuration on an IP PBX that enables a call to be routed from a PSTN or VoIP phone to a VidyoConferencing endpoint or Meeting Room.

IV. **CONNECTING TO VIDYOCONFERENCES FROM VOIP PHONES**

If you are using a soft phone or IP phone that supports the ability to place calls directly using a SIP dial string, it is possible to call into VidyoConferences directly without the involvement of an IP PBX. The SIP dial string must include three components: the voice-only service prefix of the VidyoGateway, the extension of the endpoint or Meeting Room you are calling, and the address of the VidyoGateway.

Here is an example: **91234@10.10.99.1**

This dial string would route a call to the VidyoGateway at IP address 10.10.99.1, on the voice only service prefix “9”, to extension “1234”. Depending on the configuration of the VidyoGateway service prefix, the call will either connect to the Vidyo user’s Meeting Room or ring the endpoint with extension “1234” directly.

Below is a screen shot of the VidyoGateway admin interface (Figure 1) where service parameters for the voice-only prefix are configured (more details on this configuration is provided in section VIII). The VidyoGateway receives the inbound call and identifies that it is a voice only call based on the service prefix “9”. It then converts the media and signaling to formats used in VidyoConferencing, removes the prefix, and communicates with the VidyoPortal and VidyoRouter to connect the call to the appropriate Vidyo Meeting Room.

![VidyoGateway Configuration Page](image)

**Figure 1**

This VidyoGateway prefix is configured for inbound voice-only calls using the G.711 voice codec at 64kbps. Note that the “Conference” option will route the call into a Vidyo Meeting Room. Selecting the “PtP” option will ring the Vidyo endpoint directly.
The alias “1234” in the example “91234@10.10.99.1” represents the user “Brett Flenniken”. This can be found in the VidyoPortal user interface (Figure 2):

![User extension in VidyoPortal](image)

Figure 2

V. CONNECTING TO VIDYOCONFERENCES VIA AN IP PBX

Strategies for Connecting to an IP PBX

IP PBX systems allow for various call routing rules that determine how calls are connected. There are many different possible permutations for setting up how calls are connected from telephone to the VidyoConference. There are two common methods for call routing on the IP PBX.

Direct Dial Number

With this method the IP PBX is configured with a dedicated public telephone number that will be used exclusively for voice participants joining a VidyoConference. This number will bring callers into an IVR which will prompt them for the Vidyo user extension. Once they have entered the extension the IP PBX will forward the call out the SIP trunk to the VidyoGateway. Ideally, the extension that users enter at the IVR will match the destination Vidyo extension and the IP PBX will prepend the VidyoGateway voice-only prefix.

Extension Dial

A second method is to treat each Vidyo user as another extension similar to a desk phone. This is done by assigning a block of extensions to the VidyoGateway trunk. When creating a user account on the Vidyo system, each user will be assigned one of the valid extensions. When a user then calls into the PBX from a telephone and has the option to enter an extension, they can enter one of the Vidyo extensions which will then route the call through the SIP trunk to the VidyoGateway. The IP PBX will be configured to route any extension in this range to the VidyoGateway with the VidyoGateway voice-only prefix pre-pended.

First, let’s look at the Direct Dial scenario on the IP PBX and what will need to be configured to integrate VoIP calls into VidyoConferences. Here are a few steps that will need to be taken:

- Configure the VidyoGateway with an inbound voice-only prefix (see section VIII)
- Configure a SIP trunk between the IP PBX and the VidyoGateway
- Configure IP PBX with a dedicated phone number that corresponds to the VidyoGateway
- Configure IP PBX to have an IVR prompt where callers enter the destination Vidyo extension
- Configure the IP PBX to prepend VidyoGateway voice-only prefix and route the call over the SIP trunk to the VidyoGateway

In this sample scenario here are our assumptions:

- We are connecting to a Vidyo Meeting Room with extension 1001
- The VidyoGateway IP address is 10.10.99.1
- The Voice Only prefix on the VidyoGateway is “9” and is configured in “Conference Mode”.
- The PBX is configured to forward all calls to 800-555-1212 to VidyoGateway
Dial String and Call Workflow:

- Telephone caller dials 800-555-1212 and enters extension “1001” at the IVR prompt.
- IP PBX converts this number to the SIP address “91001@10.10.99.1” based on its service configuration and the SIP trunk and the call is sent to the VidyoGateway.
- The VidyoGateway identifies the call as a voice only connection based on prefix “9” then strips the prefix away.
- The VidyoGateway converts G.711 or G.722 audio codec to SPEEX and converts SIP signaling to the signaling protocol used by VidyoConferencing.
- The VidyoGateway communicates with the VidyoPortal to identify where the call on prefix “1001” needs to be routed.
- The VidyoPortal provides the address of the VidyoRouter and the VidyoGateway communicates with the VidyoRouter to complete the connection.

Next, let’s look at the Extension Dial scenario on the IP PBX and what will need to be configured to integrate VoIP calls into VidyoConferences. Here are a few steps that will need to be taken:

- Configure the VidyoGateway with an inbound voice-only prefix
- Configure a SIP trunk between the IP PBX and the VidyoGateway
- Configure IP PBX with a range of extensions that match the Vidyo user extensions or wildcard rules that will be forwarded to the VidyoGateway
- Configure the IP PBX to prepend VidyoGateway voice-only prefix and route the call over the SIP trunk to the VidyoGateway in SIP format.

In this sample scenario here are our assumptions:

- We are connecting to a Vidyo Meeting Room with extension 1001
- The VidyoGateway IP address is 10.10.99.1
- The Voice Only prefix on the VidyoGateway is “9” and is configured in “Conference Mode”.
- The IP PBX is configured to forward all calls to extensions 1000-1999 to the VidyoGateway
- The IP PBX is configured to prepend VidyoGateway prefix to the alias when sending to the VidyoGateway in SIP format.

Dial String and Call Workflow:

- Telephone caller dials PBX number 800-555-1212 and enters extension “1001” at the IVR prompt.
- IP PBX converts this number to the SIP address “91001@10.10.99.1” based on its service configuration and the SIP trunk and the call is sent to the VidyoGateway.
- The VidyoGateway identifies the call as a voice only connection based on prefix “9” then strips the prefix away.
- The VidyoGateway converts G.711 or G.722 audio codec to SPEEX and converts SIP signaling to the signaling protocol used by VidyoConferencing.
- The VidyoGateway communicates with the VidyoPortal to identify where the call on prefix “1001” needs to be routed.
- The VidyoPortal provides the address of the VidyoRouter and the VidyoGateway communicates with the VidyoRouter to complete the connection.

VI. CONFIGURING CISCO UNIFIED CALL MANAGER

This appendix covers some of the specific configurations needed to enable Cisco Call Manager versions v.6.x – v.7.x to communicate with the VidyoGateway. More advanced configurations and features are possible using the Cisco Call Manager PBX. Please consult Call Manager documentation for further details.

Note: These instructions are provided as is. Vidyo cannot guarantee making these changes will alter your current PBX operation. Please consult your PBX documentation to assure this configuration will not disrupt current operation.
Configuring a Trunk
The first step is to create a SIP trunk from the Call Manager to the VidyoGateway. This will establish a way for the PBX to route calls intended for a VidyoConference to the VidyoGateway. Perform the following procedure to add a new trunk device or update an existing trunk device.

**Note:** You can configure multiple trunk devices per Cisco Unified Communications Manager cluster.

Login into the Cisco Unified Call Manager Administration interface and select Device -> Trunk. (Figure 1)

This will bring up the Find and List Trunks page. Next, click on the Add New button. (Figure 2)

On the Trunk Configuration page, under Trunk Type, select SIP Trunk. (Figure 3) NOTE: The Device Protocol will automatically fill in SIP. Click Next.
From the **Device Information** screen, under **Device Name** give your Trunk a name and fill in all other mandatory fields represented by the * leaving all settings at its defaults. (Figure 4)

**Note:** If inbound VOIP calls are unsuccessful from a Cisco IP phone, enabling “Media Termination Point Required” may resolve the problem. This step resolved such an issue with CUCM version 6.15 and VidyoGateway 2.0.4.

Next the **SIP Information** section (Figure 5) needs to be filled out. In the **Destination Address** enter in the VidyoGateway IP address. The **Destination Port** should remain the default of 5060. The **SIP Trunk Security Profile** should be set to **Non Secure SIP Trunk Profile**. Finally, the **SIP Profile** set to **Standard SIP Profile**. Finally, click **Save**.
The Cisco Call Manager still requires setting changes. The Route Configuration needs to be setup properly. These settings determine when calls will go out through the SIP trunk that was just created. Given that every Call Manager has a dial plan unique to every organization. Explanation of the Route Configuration is out of the scope of this document. Please consult your Cisco Call Manager documentation for setting up the Router Configuration properly.

**Note**: it is important to remember that the VidyoGateway requires inbound voice calls to follow a SIP format that includes VidyoGateway voice-only prefix, Vidyo extension, and VidyoGateway IP address (i.e. 91001@10.10.99.1). Follow best practices on the Cisco Call Manager to make sure the format sent to the VidyoGateway meets these requirements.

**Note**: it is recommended that you consider configuring your IP PBX in a way that will not require users to provide the VidyoGateway voice-only prefix before the extension when they are prompted by IVR. For the easiest end user experience you want users to enter only the Vidyo extension and have the Cisco Call Manager prepend the VidyoGateway prefix based on configuration rules.

VII. CONFIGURING TRIXBOX PBX

This appendix covers some of the specific configurations needed to enable Trixbox version 2.6.2.2 to communicate with the VidyoGateway. More advanced configurations and features are possible using the Trixbox PBX. Please consult Trixbox documentation for further details.

**Note**: These instructions are provided as is. Vidyo cannot guarantee making these changes will alter your current PBX operation. Please consult your PBX documentation to assure this configuration will not disrupt current operation.

**Creating a SIP Trunk**

The first step is to create a SIP trunk from the Trixbox to the VidyoGateway. This will establish a way for the PBX to route calls intended for a VidyoConference to the VidyoGateway. To setup a SIP trunk go to the PBX Configuration tool by clicking **PBX -> PBX Settings** (Figure 6).
Next select Trunks from the navigation menu on the left side of the user interface. Then select Add SIP Trunk (Figure 7).

On the SIP Trunk General Settings (Figure 8) the Outbound Call ID needs to be configured. For the Outbound Caller ID specify an outbound caller ID for your VidyoGateway.

**NOTE:** Be sure to leave the Maximum Channels setting blank.

**Add SIP Trunk**

General Settings

- Outbound Caller ID: vidyogateway/
- Never Override CallerID: [ ]
- Maximum Channels: [ ]
- Disable Trunk: [ ]
- Monitor Trunk Failures: [ ]

Next the Outgoing Settings (Figure 4) need to be configured for the SIP Trunk. This includes setting the Trunk Name as well as configuring the PEER Details.
• **Trunk Name** – This is the same entry used in the VidyoGateway SIP Configuration under User ID (Figure 9). In this example “VidyoGateway” is used as the User ID and Trunk Name.

• PEER Details - The PEER Details provide settings and authentication for connecting a gateway. Use the following settings substituting your VidyoGateway IP address for the host IP.

\[
\begin{align*}
\text{context} &= \text{from-internal} \\
\text{type} &= \text{friend} \\
\text{host} &= 10.11.12.10 \\
\text{canreinvite} &= \text{no}
\end{align*}
\]

**Outgoing Settings**

- **Trunk Name**: VidyoGateway
- **PEER Details**: context=from-internal
type=friend
host=10.11.12.10
canreinvite=no

Figure 9

Finally **Submit change** and click **Applied Configuration Changes** (Figure 10) to active the new SIP trunk settings.

Figure 10

**Setting up Outbound Routes**

Click on the **Outbound Routes** located on the left-hand navigation menu, and then **Add Route**. This is where you will specify the **Dial Pattern** so calls can be routed to the VidyoGateway.

In the example below “9.” was chosen for the **Dial Pattern**. The “9” represents a Voice Only service that was setup on the VidyoGateway. The following period “.” is the wildcard. If a number dialed starts with 9 it will be routed to the SIP trunk you specified under **Trunk Sequence** (figure 11). The “.” wildcard tells the system to match any combination of characters that follow. See next section for other wildcard patterns.

Example: If you dial 91001 all digits are sent through the VidyoGateway trunk you created earlier. Where “9” is equal to the Voice Only prefix in your VidyoGateway and 1001 represents the user extension or conference number.
This section provides some basic information about constructing dial patterns on the Trixbox. This feature can provide tremendous flexibility in determining how calls are routed. For a more complete description of Dial Patterns please consult the Trixbox documentation.

The Dial Patterns section can be used to specify what dial patterns will match for this trunk. In order to create a working pattern we need to know a few things about how to create one. The following characters can be used to create a pattern:

- **X** — The X matches any digit from 0 – 9.
- **Z** — The Z matches any digit from 1 – 9.
- **N** — The N matches any digit from 2 – 9.
- **[12347-9]** — Numbers within brackets matches a number or sequence within the brackets. This example would match on 1,2,3,4,7,8,9.
- **|** — The period acts as a wildcard character matching any combination of characters.
- **|** — The | separates the prefix from the number to be dialed. As you can see if the pre-configured outbound route uses “9|.”, it would strip off the 9 and send all remaining digits to the trunk.

**Note:** It is important to remember that the VidyoGateway requires inbound voice calls to follow a SIP format that includes VidyoGateway voice-only prefix, Vidyo extension, and VidyoGateway IP address (ie. 91001@10.10.99.1). Follow best practices on the Trixbox to make sure the format sent to the VidyoGateway meets these requirements.

**Note:** It is recommended that you consider configuring your IP PBX in a way that will not require users to provide the VidyoGateway voice-only prefix before the extension when they are prompted by IVR. For the easiest end user experience you want users to enter only the Vidyo extension and have the Trixbox prepend the VidyoGateway prefix based on configuration rules.

**VIII. VIDYOGATEWAY CONFIGURATION**

Finally the VidyoGateway needs to be configured to receive the SIP trunk for incoming voice calls. To do this the VidyoGateway must have properly configured SIP settings as well as have a SIP Voice service setup to handle the voice calls.

**SIP Configuration**

Log into the VidyoGateway and select the **Config** tab from the top. Then scroll down to the SIP Configuration section (Figure 12). Fill out the information for this section.

**SIP Server Address:** - This field represents your IP PBX IP address or URL.
**SIP Server Port:** - This port is 5060 by default which is also the standard for SIP signaling. Do not change this unless you are certain the IP PBX is using a non-standard port.

**Register** – Leave this setting unchecked.

**User ID:** - This is the SIP user assigned to the VidyoGateway by the IP PBX. In this example the User ID is “vidyogateway.”

**Password:** - Is the password associated with the User ID.

![Figure 12](image)

Once the VidyoGateway has a proper SIP configuration it next needs to have a Service setup to handle the incoming voice call. To add a voice service, click on the **Services** tab and then click on **Add Service** along the left side of the user interface. This will open the Add Service page (Figure 13).

**Adding Gateway Service**

![Figure 13](image)

Configure the following settings:

**Prefix** – The prefix is the portion of the dial strings that tells the VidyoGateway what type of call this will be. In this example the prefix “01” is used.

**Direction** – This defines whether the call is inbound to the VidyoGateway (From Legacy) or outbound from the VidyoGateway (To Legacy). Set the **Direction** to “From Legacy.”

**Video Codec** – Since this is for a voice call, select “Voice Only.”
Audio Codec – The audio codec determines how the audio is encoded and decoded. This is dependent upon the audio devices calling the VidyoGateway. The VidyoGateway supports both G.711 and G.722. The G.711 is the most common audio codec, however G.722 offers higher quality audio. By selecting “G722/G711” the VidyoGateway will try G.722 first and then revert to G.711 if the other side isn’t capable of supporting it.

Call Type – This setting determines if the inbound call is routed to a Vidyo user’s Meeting Room (Conference) or if it will ring their Vidyo endpoint directly in a point to point call (P2P).

Description – This field is information for the administrator to distinguish between services on the VidyoGateway. Provide a name that is meaningful.

Status – Make sure that the service is Active.

Click Save to save the newly created service. Now click on the Config tab and scroll to the bottom. To apply the new service click on the Save and Apply button. The VidyoGateway is now configured to receive inbound voice calls.