# VU TelePresence® Interoperability Technical Guide

With the release of the VU TelePresence® Interoperability as part of the No. 7 service pack, VU TelePresence® users will be able to connect to other systems that are SIP enabled and support H.264 baseline profile. This guide describes the steps required to enable and use the VU TelePresence® Interoperability pack that has been downloaded onto your systems automatically.

The key requirements for interoperability to work are:

- SIP enabled endpoint
- Support for H.264 baseline profile
- Support for G.711 u-LAW and a-LAW audio codec

## Set-up Procedure to Receive and Make SIP Calls

Direct Mode

SIP can be used in a direct mode where one SIP device talks to the other using an IP address or host name, and without the existence of any proxy.

**Placing a Call**

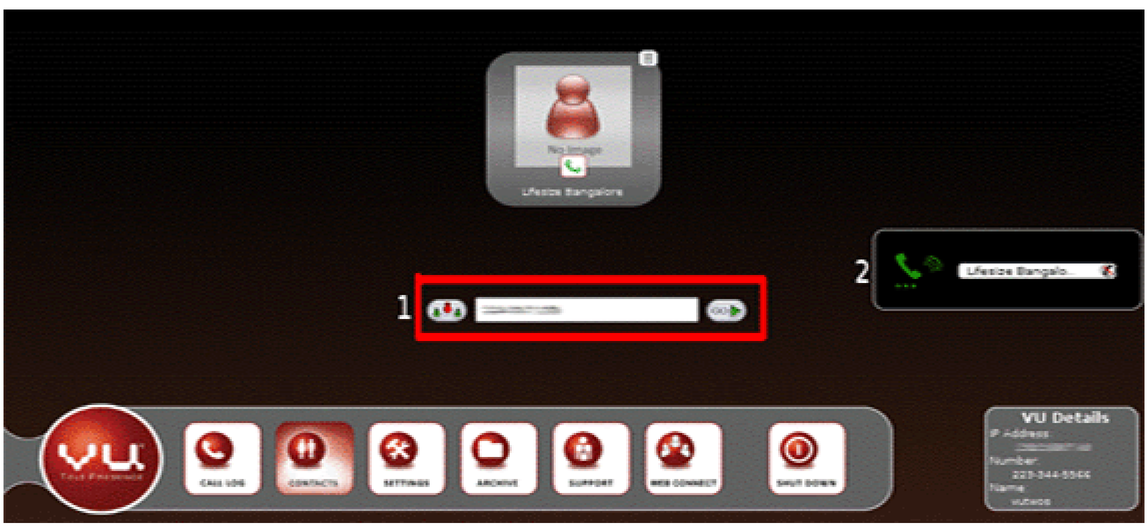


*Figure A – VU TelePresence® Home Screen User Interface*

a) Enter the IP address or Host name of the SIP endpoint in the search box **(#1. - Figure A)** in the figure above and click on GO.

b) The remote endpoint will receive the call. **(#2. - Figure A)**
c) If a call is established successfully, a SIP contact is automatically added to the Contact List. **(Figure B)**
d) Further calls can be made by directly clicking on the contact in the Contact List.

<u>Receiving a Call</u>

a) Enter the IP address or Host name of the VU System in the remote unit and specify SIP as the protocol for dialing (SIP:Vu.xyz.com).
b) The VU System will display the Incoming call pop up and the user can accept or reject the call.
c) If a call is established successfully an SIP contact is automatically added to the Contact List.
d) Further calls can be made by directly clicking on the contact in the Contact List.

*Figure B – Contact List*

## SIP Proxy Mode

In this mode, the VU unit needs to register with a SIP Proxy and use the proxy for making calls. To register with a SIP proxy, please follow the instructions below:

Navigate the menu Settings → SIP Settings. *(Note: You will be prompted for a password to edit the settings. Enter the password if you have set one.)* **(Figure C)**


*Figure C – SIP Settings Screen*

Enter the details of your SIP Proxy in the form shown in **Figure C**:

- SIP User Name: xyz@hostname.com <The SIP URI assigned to the unit>
- Authorization Name: xyz <username for authentication>
- Authorization Password: abcefg <Password>
- SIP Registration: Through Proxy / Direct < Proxy will be used for all communication to other end points>
- Proxy Server: proxy.hostname.com <Proxy IP or hostname>
- Proxy Port: 5060 <Proxy port to send messages>

Once you fill in the information and click on the Submit button, the system will reset the menus and come up with the registration.

You are now ready to make calls.

<u>Placing a Call</u>

a) Enter the SIP URL of the remote SIP endpoint in the form SIP: username@proxy in the search box and click on GO.
b) The remote endpoint will receive the call.
c) If a call is established successfully, a SIP contact is automatically added to the Contact List.
d) Further calls can be made by directly clicking on the contact in the Contact List.

<u>Receiving a Call</u>

a) Enter the username of the VU System in the remote unit and specify SIP as the protocol for dialing (SIP: Vuuser@proxy).
b) The VU System will display the Incoming call pop-up and the user can accept or reject the call.
c) If a call is established successfully an SIP contact is automatically added to the Contact List.
d) Further calls can be made by directly clicking on the contact in the Contact List.

## Add/Edit SIP Contacts


*Figure D – Add/Edit Contact Screen*

It is also possible to add and edit SIP contacts directly.

1) Click on the Contacts Button -> Add Contact.
2) Enter the SIP URL / IP address in the first field.
3) Enter the Display Name, and if you what to add as a favorite click the check box.

# Additional Port Information for Interoperability Patch

In addition to the ports used normally with your VU system, the following ports need to be opened:

- Port 5060,5061-TCP and UDP for SIP
- Port 7000,7001,7500,7501 UDP for media traffic.

## Multi-Party Calling with SIP

VU supports multi-party calling for 5 (4+1) parties in a native VU to VU mode. In interoperability mode, conferences will be point-to-point unless the call uses a central Mixer or Bridge. VU supports interoperability with most standards compliant mixers both on premise or hosted in the cloud such as Vidtel.com. The Mixer from a service like Vidtel allows multi-party conferences with multiple VU and Non VU end points.

Please visit http://www.vidtel.com for more information on their mixer services. Vu does not specifically endorse Vidtel's services and customers have to evaluate such services on their own.

## Bandwidth and Rate Control

One of the important features available in this latest service pack is a way to control and tune the bandwidth so that you can get the best possible VU experience. You can set the bandwidth by following the step listed below:

1. Navigate the on screen menu to Settings ->User Settings. Edit the Max Allowed Bandwidth (300 – 3000 kbps) **(Figure E)**


*Figure E – User Settings Screen*

The maximum video bandwidth is set to a value that is the difference of the Max Bandwidth set and the Audio Bandwidth required, which with overhead of the protocol headers is 80 kbps.

# Frequently Asked Questions (FAQs)

1. **I called my unit but it's not ringing?**
   First check if SIP is enabled on the unit you are trying to call. Most legacy units have support for SIP disabled and the default is H.323. Please make sure you enable SIP on the device and call the unit.

   Also check if the unit supports H.264. VU currently does not support H.263 or H.263+. This is planned for a future release.

2. **Do you support H.323?**
   VU currently does not support H.323 and the family of protocols. This is currently in the works and updates regarding the same will be provided shortly.

3. **Is recording supported for SIP calls?**
   Yes it is supported. You can record the whole call or parts of a call just as you would with normal VU based calls.

4. **How can I connect to my H.323 or ISDN endpoint?**
   Currently you can use a hosted Bridge provider such as Vidtel which is certified to bridge calls with non IP and endpoints that do not have SIP enabled.

5. **How do I control the Bandwidth?**
   See Bandwidth and Rate Control section.

6. **VU does not work with my system?**
   Please contact support. We will help resolve the issue if the conditions listed above are met.

7. **Does the VU unit still require Internet connectivity to work?**
   Yes. The unit still registers with the VU Cloud to allow native operation for ease of configuration. A server-less (or in-premise server) mode of operation is being evaluated and our product roadmap. More information will be released shortly.