Citel. The VoIP Migration Company™

12 and 24 Port SIP Handset Gateway

Softswitch Integration Guide

Document Number: G-SIP3DS-RUC **Version: AB**

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Trademark Information

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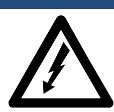
Symbols and Conventions

Important symbols and conventions used throughout this guide are shown below.

Icon	Description
	Important safety information. Ignoring this information may lead to physical danger to people.
	Information alerting you to potential loss of data or damage to an application, system or device.
	Highlights important information.

Contacting Citel Technologies

Mail	Citel Technologies 3131 Elliott Avenue Suite 250 Seattle WA 98121	Citel Technologies 4 Wheatcroft Business Park Landmere Lane Edwalton Nottinghamshire NG12 4DG UK
Email	support@citel.com	
Web	www.citel.com	
Telephone	+1 888 454 5828 – select the support option	

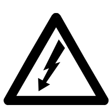
Product model numbers

The information contained within this document refers to the following products:

Part Number	Description
E-SIP3D-RUC12	Handset Gateway 12 port (digital only)
E-SIP3D-RUC1241	Handset Gateway 12 port (digital only)*
E-SIP3D-RUC24	Handset Gateway 24 port (digital only)
E-SIP3D-RUC2441	Handset Gateway 24 port (digital only)*
E-SIP3DZ-RUC12	Handset Gateway 12 port (digital) with FXO port
E-SIP3DZ-RUC1241	Handset Gateway 12 port (digital) with FXO port*
E-SIP3DZ-RUC24	Handset Gateway 24 port (digital) with FXO port
E-SIP3DZ-RUC2441	Handset Gateway 24 port (digital) with FXO port*
E-SIP3DY-RUC12	Handset Gateway 12 port (digital) with 2 FXO ports
E-SIP3DY-RUC1241	Handset Gateway 12 port (digital) with 2 FXO ports*
E-SIP3DY-RUC24	Handset Gateway 24 port (digital) with 2 FXO ports
E-SIP3DY-RUC2441	Handset Gateway 24 port (digital) with 2 FXO ports*
E-SIP3P-RUC24	Handset Gateway 24 port (Pphone/analog only)
E-SIP3P-RUC2441	Handset Gateway 24 port (Pphone/analog only)*
E-SIP3PZ-RUC24	Handset Gateway 24 port (Pphone/analog) FXO port
E-SIP3PZ-RUC2441	Handset Gateway 24 port (Pphone/analog) FXO port*
E-SIP3PY-RUC24	Handset Gateway 24 port (Pphone/analog) 2 FXO ports
E-SIP3PY-RUC2441	Handset Gateway 24 port (Pphone/analog) 2 FXO ports*

* European version

Safety Information



Important

This guide and warranty and liability details are published in the 'downloads' area at www.citel.com/service_support. Any questions regarding the use of this guide may be directed to support@citel.com.

Before you start to install the product, make sure you have read and complied with all instructions, including the Safety Information in the 'Installation and Configuration Guide'.

You must communicate the safety information to the users and administrators of the telephone system in which a Citel SIP Handset Gateway is operating.

Only trained, qualified service personnel shall install or maintain this product.

Failure to follow all instructions may result in improper equipment operation or risk of electrical shock.

Changes or modifications not expressly approved by Citel could void the user's authority to operate the equipment.

Power Surges

Sudden surges in electrical current can damage sensitive equipment. To reduce the risk of damage to your equipment, for example caused by lightening strikes, install a surge protector between your equipment and both the AC power outlet and the telephone line.

Contents

1 INTRODUCTION

Purpose of This Document

This document contains an overview of the steps required to configure the Citel Gateway to implement conferencing and call-sharing options provided by softswitch manufacturers.

The softswitches included in this document are:

- Asterisk
- BroadSoft
- Sylantro

Support for additional softswitches is added on an ongoing basis. If your softswitch is not listed above, please contact Citel to see if it is currently supported.

This document contains basic instructions, based on examples, for configuring the Citel Gateway to work with the listed softswitches.

It is not intended to be a comprehensive guide to your softswitch. Please read the documentation provided with your softswitch for full configuration instructions.

Who Should Read This Document?

This document should be read by those installing the Citel SIP Gateway and those responsible for managing it.

Which Sections Should You Read?

The rest of this document is divided into chapters, one for each of the different types of softswitch. You only need to read the chapter for the softswitch you are using.

Other Documentation

You should read the Citel SIP Gateway Installation and Configuration Guide, especially chapter 3 (Installation) and chapter 4 (Initial Configuration).

You should locate the pages in the Phone User Guide containing the default button mappings for the telephones in use at your location and update them to show which have been allocated the functions described in this document.

2 ASTERISK

Conference calls can be made using functionality provided by the Citel Gateway, without the need for additional configuration. Making conference calls locally in this way also reduces the network traffic. To make local conference calls, you need to leave the Conference URI (see Figure 1 for the location of the Conference URI) blank and map a button on the telephone to the conference feature.

Conference calls using the Asterisk softswitch can be made by joining a conference room, which can be insecure as anyone with the correct codes can enter the conference, or by using 'ad-hoc' conferencing, which limits the conference call to three people but which is more secure.

If you are using an interface to your Asterisk softswitch, you may need to make the changes to comply with the requirements of the interface. For example, you may need to edit custom versions of the files, which are managed by the interface.

'Ad-hoc' Conferencing Method

This feature enables users to create conferences, using the designated Conference buttons on their phones, which two other parties can join in.

Before editing any configuration file take a safe copy of it in case of any problems. Document the changes you make. Remember that changes may affect the entire telephone system.

Asterisk Configuration

1. Edit the **`extensions.conf`** file.

   Add a rule in the **`extensions.conf`** file to send calls to the Asterisk MeetMe Application. An example of the change needed is given below.

   ```
   exten => conf,1,Meetme(${CALLERIDNUM}|dq) exten => conf,2,Hangup
   ```

   - The first line matches all calls to conf and invokes the Asterisk MeetMe Application. The options to this are the **CallerID** (conference number), and **d** and **q**.
   - By using the **CallerID** as the conference number, all phones get a unique conference room to use for ad-hoc conferencing.
   - The **d** option creates the conference dynamically if it does not already exist. This eliminates the need to provide many conference rooms ahead of time.
   - The **q** option suppresses audio prompts when people enter and leave the conference.

2. Restart Asterisk.

Citel Gateway Configuration

For each phone you want to be able to use on conference calls, set the Conference URI to conf as part of the line configuration. See *Figure 1*.

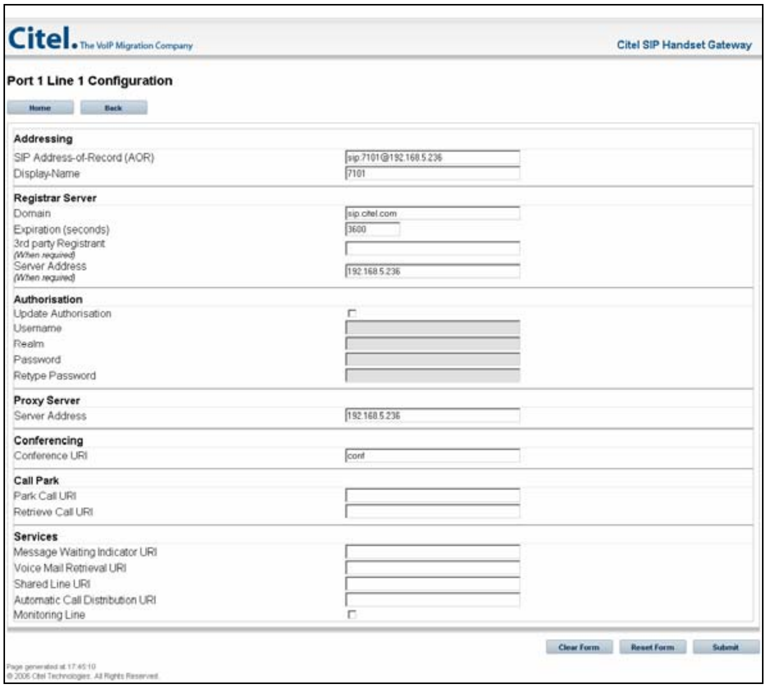


Figure 1: Setting the conference URI

For each phone, set one of the keys to be the **Conference** button, as shown in *Figure 2*.

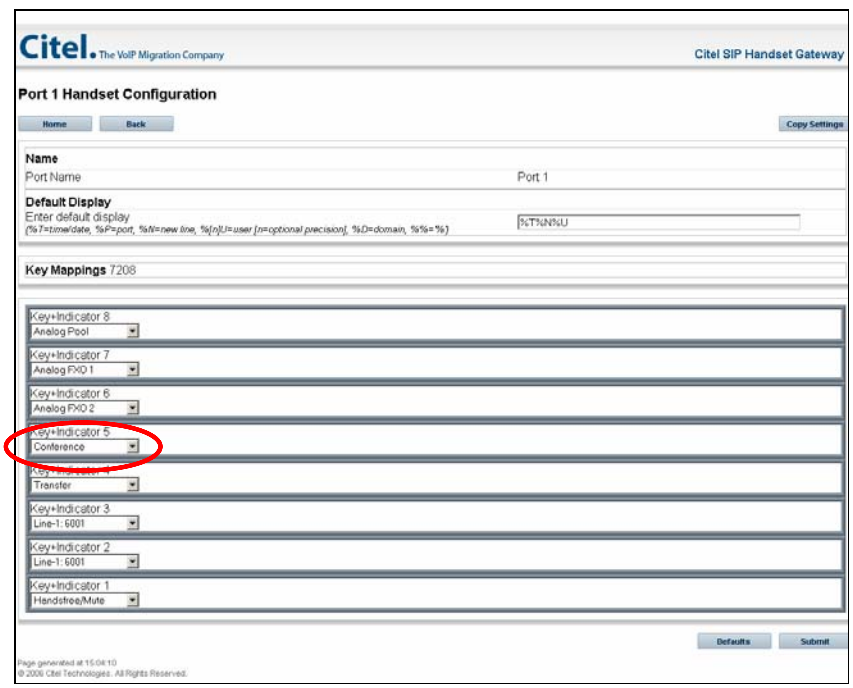


Figure 2: Setting a key as the conference button

Conference Room Method

All the callers dial a conference room number and then enter a PIN. The conference room number and PIN must be set up by the Systems Administrator editing the relevant configuration file.

Before editing any configuration file take a safe copy of it in case of any problems. Document the changes you make. Remember that changes may affect the entire telephone system.

Asterisk MeetMe Simple Conference Rooms

Conferencing is achieved by using the Asterisk MeetMe Application. This requires a timer device such as a Zaptel.

1. Edit the **`meetme.conf`** file

   The example below demonstrates how a conference room is configured using the **`meetme.conf`** file, where **`1234`** is the conference room and **`6789`** is the PIN required to access it.

   ```
   [rooms]
   ;
   ; Usage is conf => confno[,pin]
   ;
   conf => 1234,6789
   ```

2. Edit the **`extensions.conf`** file

   The dial plan is configured in the **`extensions.conf`** file. It controls how all calls are routed and handled and controls the behavior of all connections through the PBX.

   An example of the changes needed is given below. **`1234`** is the conference room and **`6789`** is the PIN to enter the conference.

   ```
   [default]
   ...
   exten => 1234,1,Meetme,1234,6789  ; dial 1234, conf room 1234,
   pin 6789
   ```


3 BROADSOFT

Conference calls can be made using functionality provided by the Citel Gateway, without the need for additional configuration. Making conference calls locally in this way also reduces the network traffic. To make local conference calls, you need to leave the **Conference URI** (see *Figure 4* for the location of the **Conference URI**) blank and map a button on the telephone to the conference feature.

Please see the documentation provided with your BroadSoft softswitch for full configuration instructions. The information in this chapter is provided to assist you in enabling three-way conference call facilities and shared call appearances when using the BroadSoft softswitch.

Conferencing

The BroadSoft softswitch allows you to specify a **User ID** for each user/extension, which is used when configuring the Citel Gateway.

Check your BroadSoft media server is running Release 11 (or later) and that its SIP interface is enabled.

BroadSoft Configuration

In this example, extension 6001 has been allocated the **User ID** of test6001.

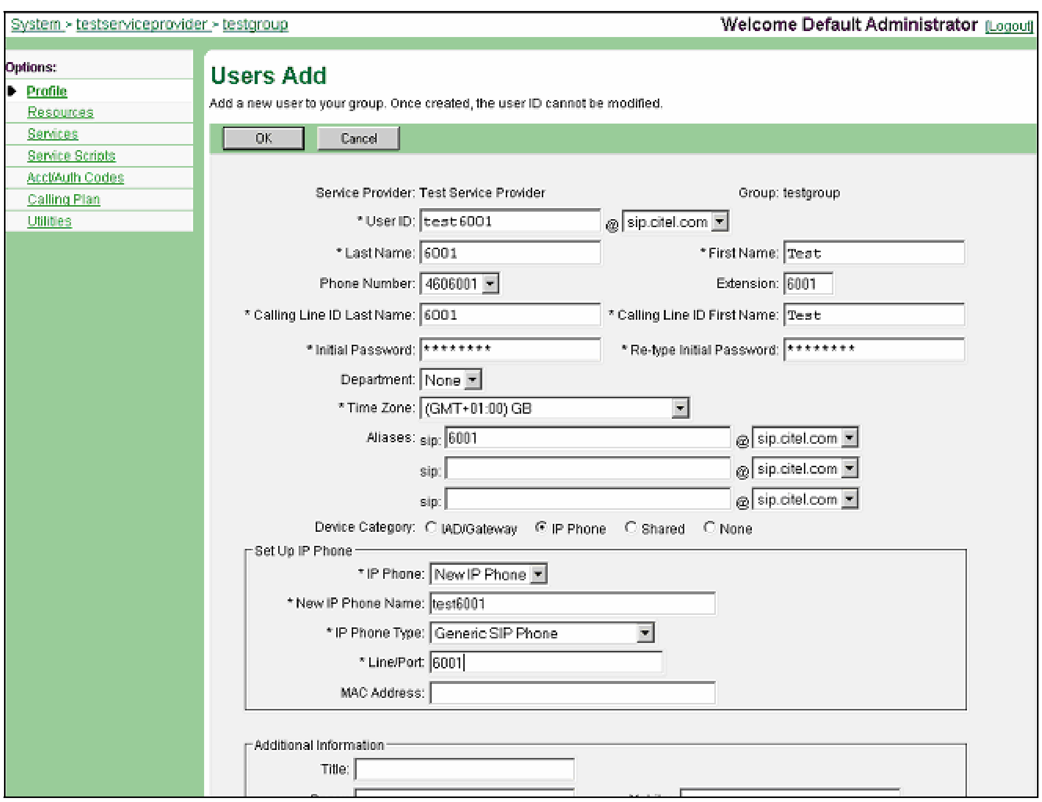


Figure 3: Allocating the User ID of extension 6001

Citel Gateway Configuration

The extension must be allocated to a phone on the Citel Gateway. In the example in *Figure 4*, the phone is connected to Port 1, Line 1.

The **Conference URI** is constructed as follows:

sip:conf=#@<BroadSoft Conference Server IP address>;x-route=direct

For example:

sip:conf=#@192.168.5.245;x-route=direct

Alternatively, you could use the fully qualified domain name of the machine instead of the IP address.

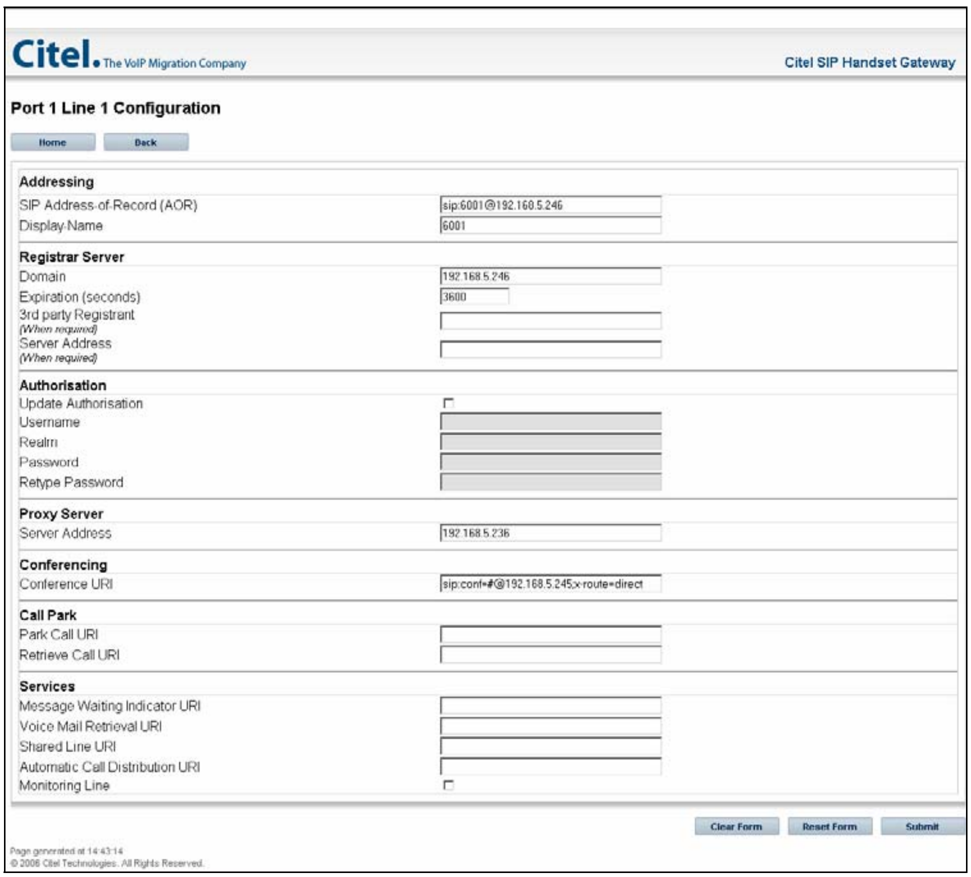


Figure 4: Entering information as part of the line configuration

The line (in this example, Line 1) must also appear on at least two telephone handset button mappings, as shown in *Figure 5*

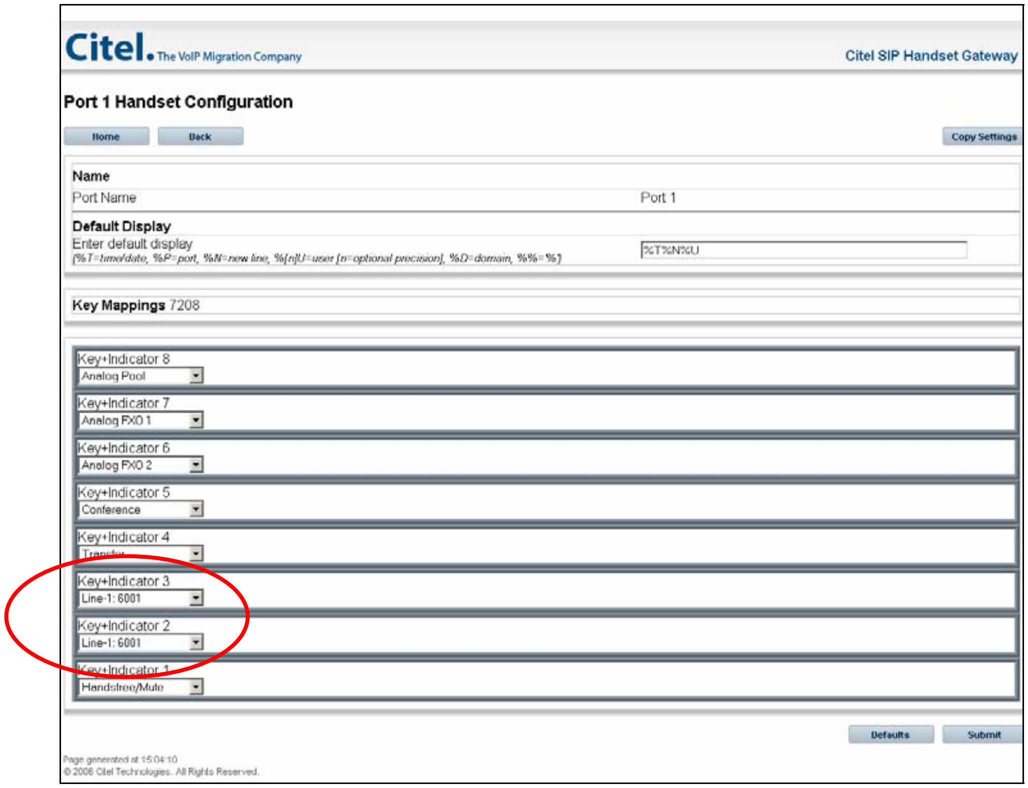


Figure 5: Allocating the line to two button mappings

Shared Call Appearances

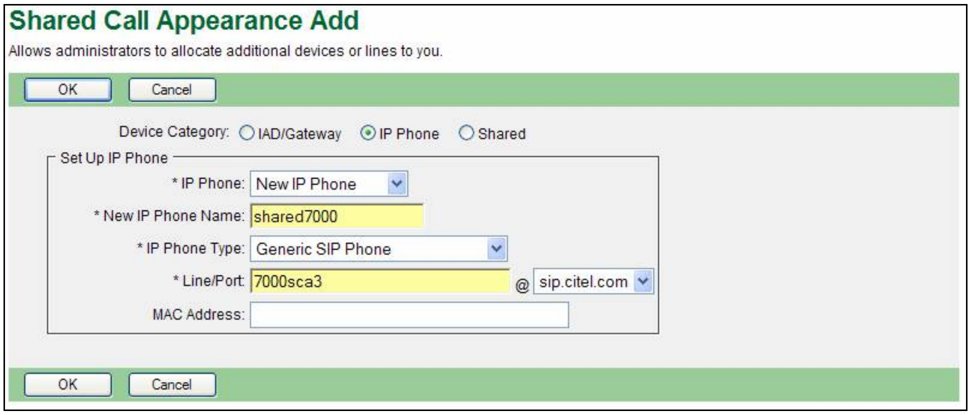
The BroadSoft softswitch uses shared call appearances (SCA) to allow more than one phone to ring when a particular extension is called.

For example, you have three phones (phoneA, phoneB and phoneC) that are all in the same SCA (Shared Call Appearances) group. A call is made to phoneC.

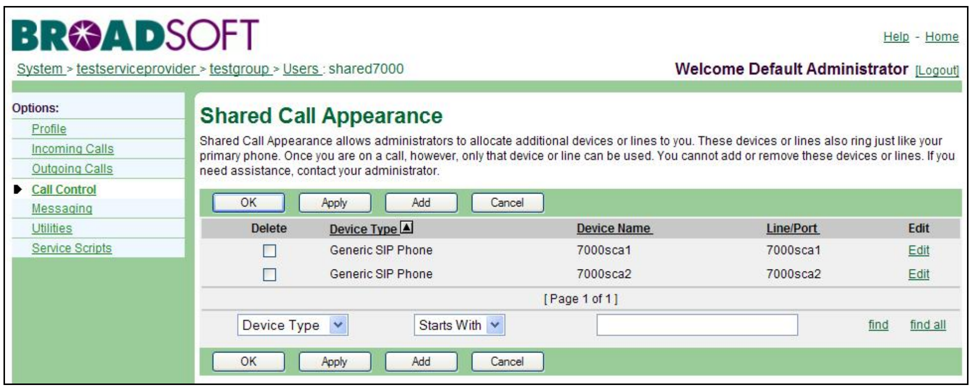
- All three phones ring.
- When one of the phones answers the call, the other phones show the call in progress for that line.
- If the call is put on hold, this will be shown on all three phones. The call can be picked up by any phone in the group.

BroadSoft Configuration

1. Define the users for each phone in the group.
2. Go to **Call Control – Shared Call Appearance** for the phone you are configuring.
3. Create the call appearances you plan to use, by adding a new entry for each phone. Use a unique line/port id for each new call appearance – for example, add a suffix or prefix such as sca1, sca2 and sca3.



4. The lines that you have configured for the other phones in the group are displayed.



Citel Gateway Configuration

1. For each phone, assign one of the available lines to one of the SIP AORs for the **Shared Call Appearance** line/port that you specified in step 3 of the BroadSoft configuration instructions.

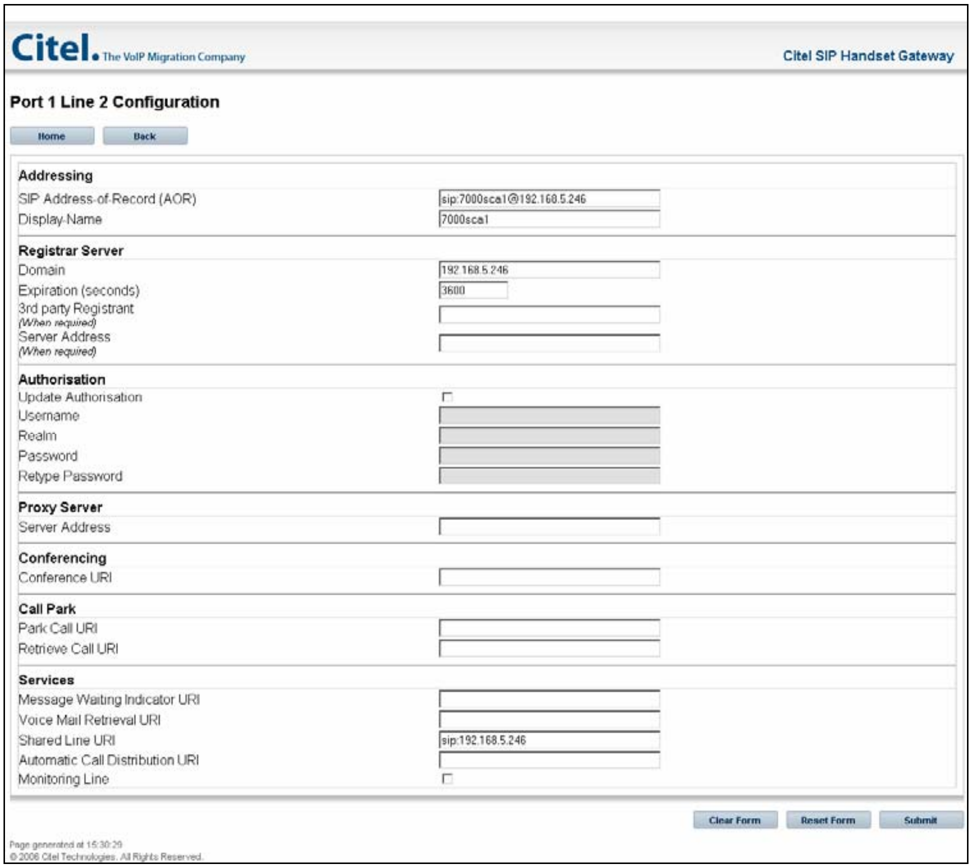
   Use a different line/port id for each phone. For example, sca1 on phoneA, sca2 on phoneB, sca3 on phoneC.

2. In the **Services** section of the line configuration, type the **Shared Line URI**.

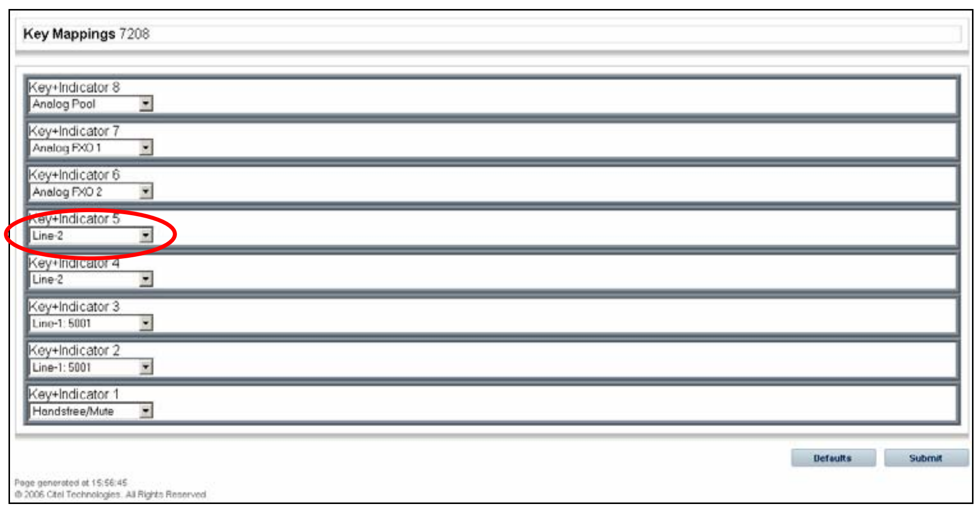
   The **Shared Line URI** in this example is the BroadSoft Server IP address.

   For example:

   `sip:192.168.5.246`



3. For each phone, ensure the Line you set up (in step 2) appears on at least one phone button with an indicator.



4 SYLANTRO

Conference calls can be made using functionality provided by the Citel Gateway, without the need for additional configuration. Making conference calls locally in this way also reduces the network traffic. To make local conference calls, you need to leave the **Conference URI** (see *Figure 7* for the location of the **Conference URI**) blank and map a button on the telephone to the conference feature.

Please see the documentation provided with your Sylantro softswitch for full configuration instructions. The information in this chapter is provided to assist you in enabling three-way conference call facilities and bridges line appearances when using the Sylantro softswitch.

Conferencing

The Sylantro softswitch enables you to specify a **SIP User ID** for each telephone user, which is used when configuring the Citel Gateway.

Sylantro Configuration

In this example, extension 7001 has been allocated the **User ID** of 2066927001.

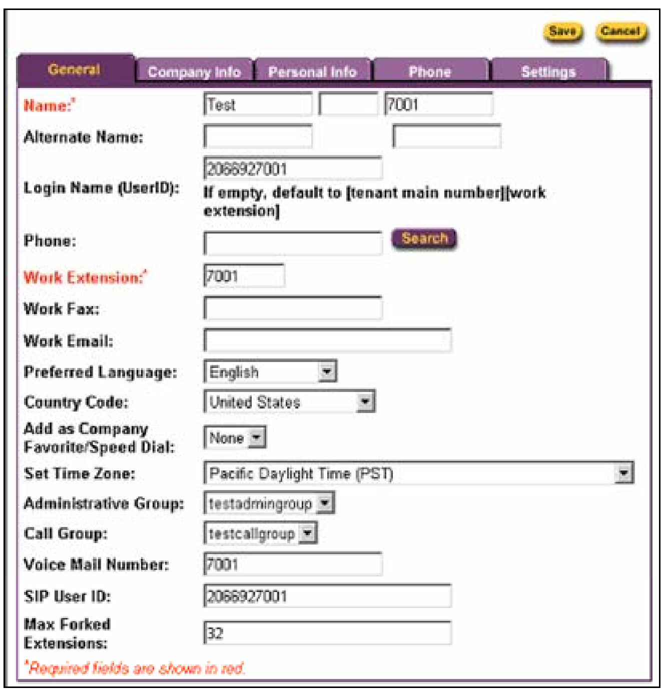


Figure 6: Allocating a SIP User ID to extension 7001

Citel Gateway Configuration

The SIP address needs to be entered in to the Citel line configuration page, as shown in *Figure 7*.

The **Conference URI** is constructed as follows:

`sip:conf=#@<BroadSoft Conference Server IP address>;x-route=direct`

For example:

`sip:conf=#@192.168.5.245;x-route=direct`

Alternatively, you could use the fully qualified domain name of the machine instead of the IP address.

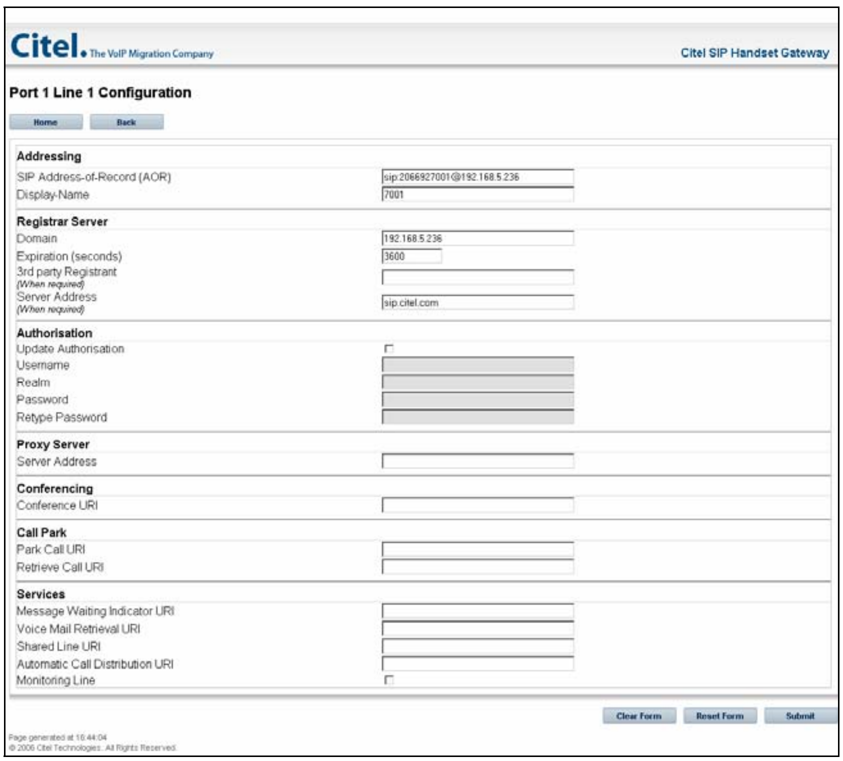


Figure 7: Information for conferencing entered into the Citel configuration page

The line (in this example, Line 1) must also appear on at least two telephone handset button mappings, as shown in *Figure 8*.

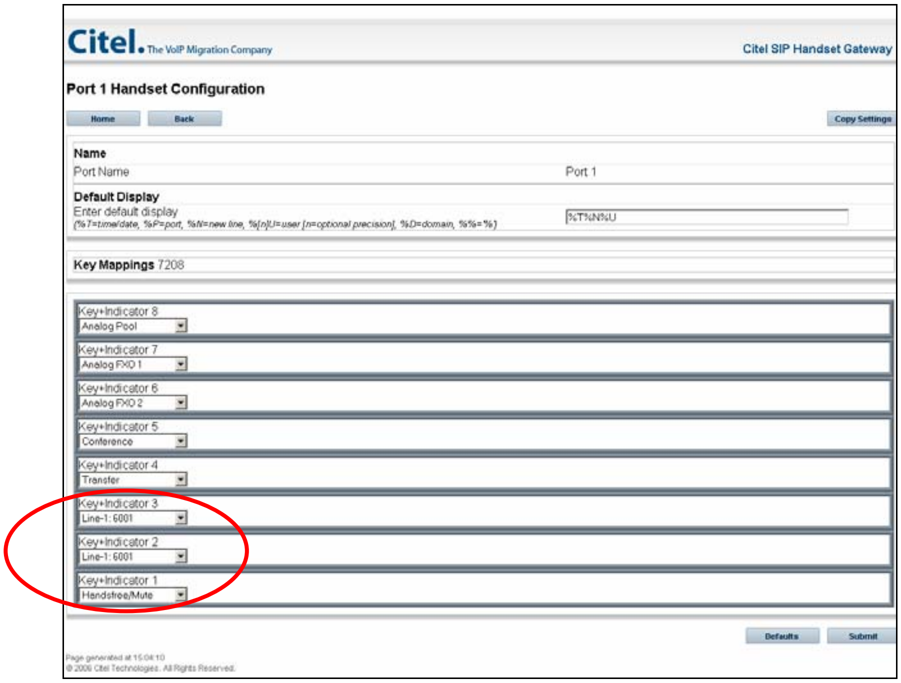


Figure 8: Allocating the line to two button mappings

Bridged Line Appearances

The Sylantro softswitch uses bridged line appearances (BLA) to enable more than one phone to ring when an extension is called.

For example, you have three phones: PhoneA (extension 3000), PhoneB (extension 3001) and PhoneC (extension 3002). PhoneB and PhoneC both have a bridged line to PhoneA.

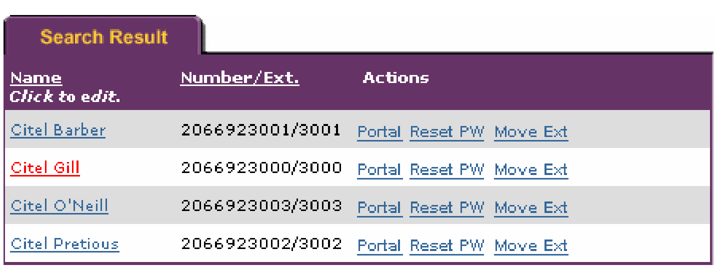
When a call is made to PhoneA:

- All three phones ring.
- When one of the phones answers the call, the other phones show the call in progress for that line.
- If the call is put on hold, this will be shown on all three phones. The call can be picked up by any phone in the group.

Sylantro Configuration

1. Use the **Management Portal** to enable **SIP Forking** in the Class of Service template for the phone to be bridged.

   a) Click on the name of the phone to which you are creating the bridge. This would be PhoneA in our example.



b) Click the **Settings** tab.

c) Select the **SIP Forking** option.

d) **Save** your changes.

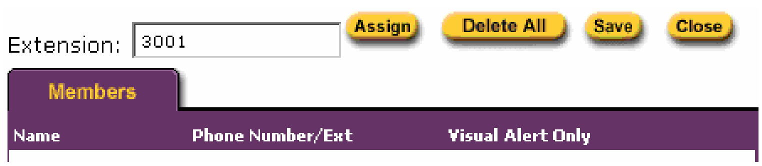
2. Assign all the phones who are to have their extensions bridged to PhoneA.

a) Click on the name of the phone to which you are creating the bridge. This would be PhoneA (extension 3000) in our example.

b) Click the **Phone** tab.

c) Click **Bridged Line Authorization**.

d) In the **Extension** box, enter the extension number of the phone you want to bridge to PhoneA.



e) Click **Assign**.

f) Repeat these last two steps for each phone which is to be bridged. In our example, this is PhoneB and PhoneC.

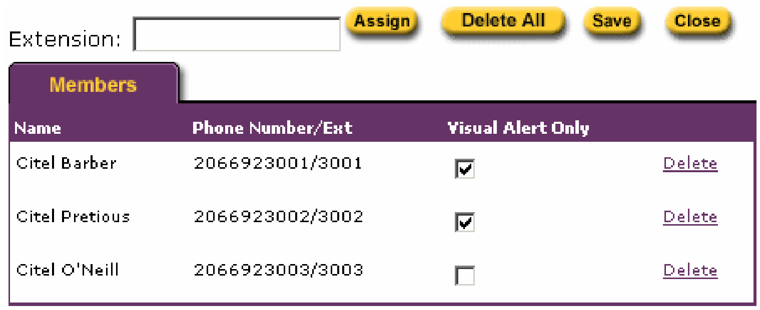
3. If you do not want to have all the phones ringing when PhoneA is dialed, set **Visual Alert only** when setting the **Bridged Line Authorization** for PhoneA in the Management Portal.

a) Click the name of the phone to which you are creating the bridge.

b) Click the **Phone** tab.

c) Click **Bridged Line Authorization**.

d) Select the **Visual Alert Only** box for each phone you are including in the bridged group, but which you don't want to hear ringing.



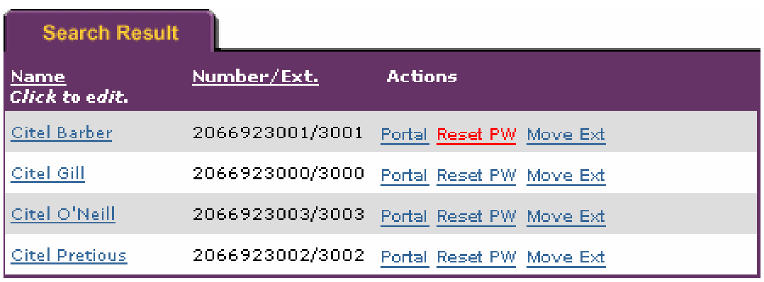
e) Click on **Save** to store the settings.

Alternatively you can select the **Monitoring Line** option in the **Line Configuration** screen in the Citel Gateway, for each phone you are including in the bridged group, but which you don't want to hear ringing.

---

4. Use the **Management Portal** to set up line authorization for the bridged line (PhoneA). The other phones should not need this to be done as their line 1 authorization should already be set correctly.

a) Click on **Reset PW** for the phone to be authorized.



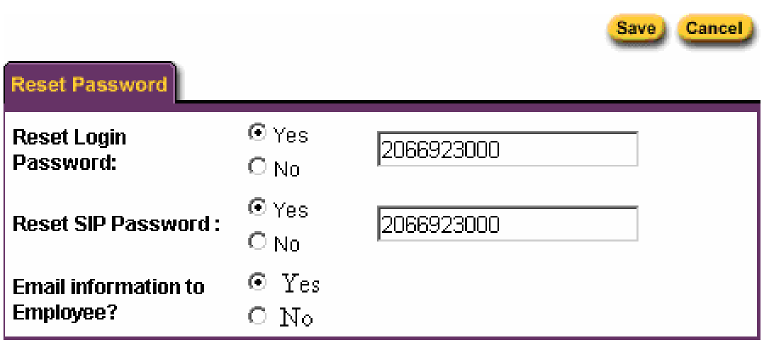
b) Select the button to reset the **Login Password** and enter the new password.

You may find it useful to use the phone's number as the password.

c) Select the button to reset the **SIP Password** and enter the new password.

You may find it useful to use the phone's number as the password.

d) Select **Save** to store the changes.



Citel Gateway Configuration

For each phone (PhoneB and PhoneC in our example) that will be able to answer the bridged extension, you must set one of its available lines to the SIP AOR of the bridged extension. The example in *Figure 9* shows that ext. 3001 has line 2 set to ext. 3000.

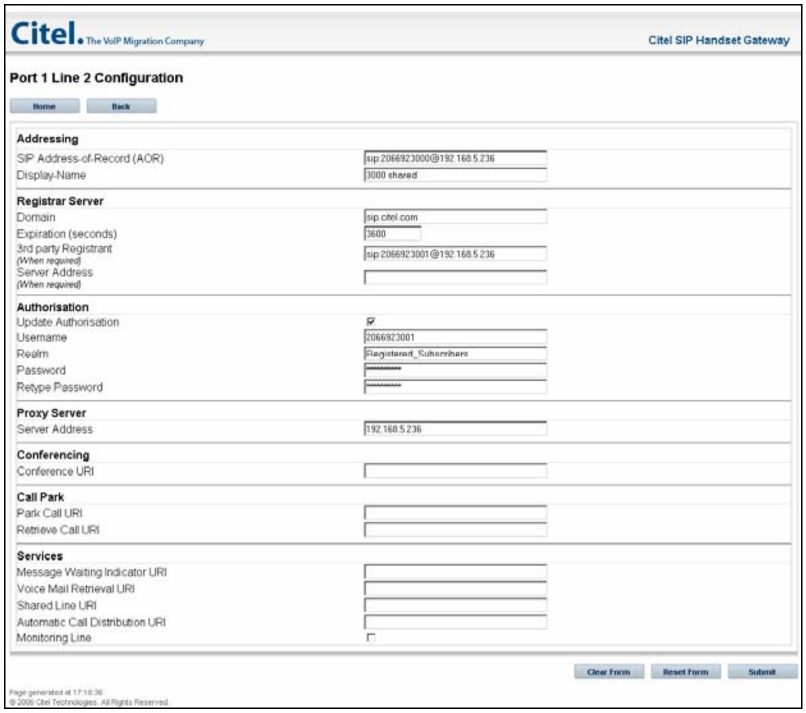


Figure 9: Setting the line configuration for a phone bridging to another phone

1. For each of the phones in the bridging group:

    a) Update the **Registrar Server** so the domain is your Sylantro Server domain. For the phones to be bridged (PhoneB and PhoneC), enter the phone's own line 1 SIP AOR in the **3rd party Registrant** field.

    b) Update the **Authorization**.

    Set the **Username** to the phone's line 1 username.

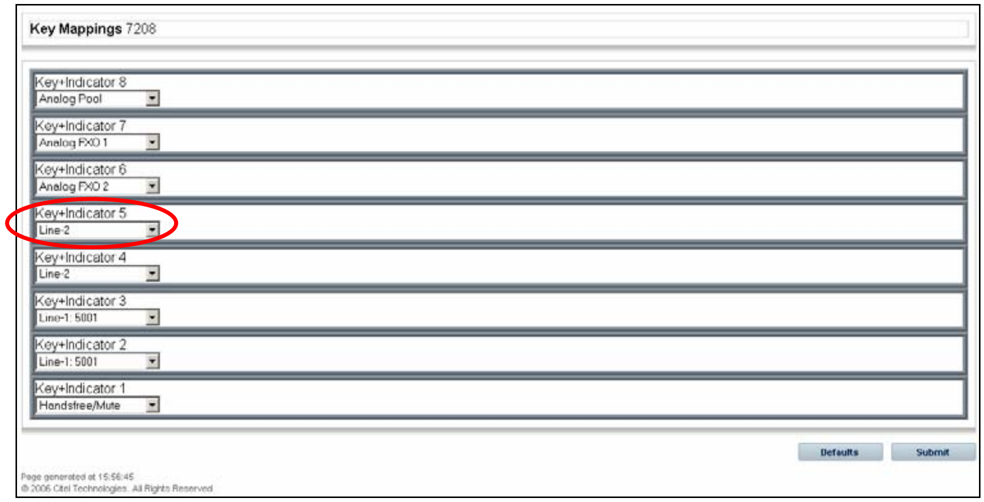
    Enter the **Realm**.

    Type the phone's line 1 **Password**, which must be the same as the one used in the Sylantro SIP password authorization for line 1.

    c) Click on **Submit** to save the settings.

    d) Click on **Reset** to apply the saved settings.

2. For each phone, go to **Handset** and ensure the **Line** you set up (in step 2) appears on at least one phone button with an indicator.


Key Mappings 7208
Key+Indicator 8
Analog Pool
Key+Indicator 7
Analog FXO 1
Key+Indicator 6
Analog FXO 2
Key+Indicator 5
Line-2
Key+Indicator 4
Line-2
Key+Indicator 3
Line-1: 5001
Key+Indicator 2
Line-1: 5001
Key+Indicator 1
Handsfree/Mute
Defaults
Submit
Page generated at 15:56:45
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