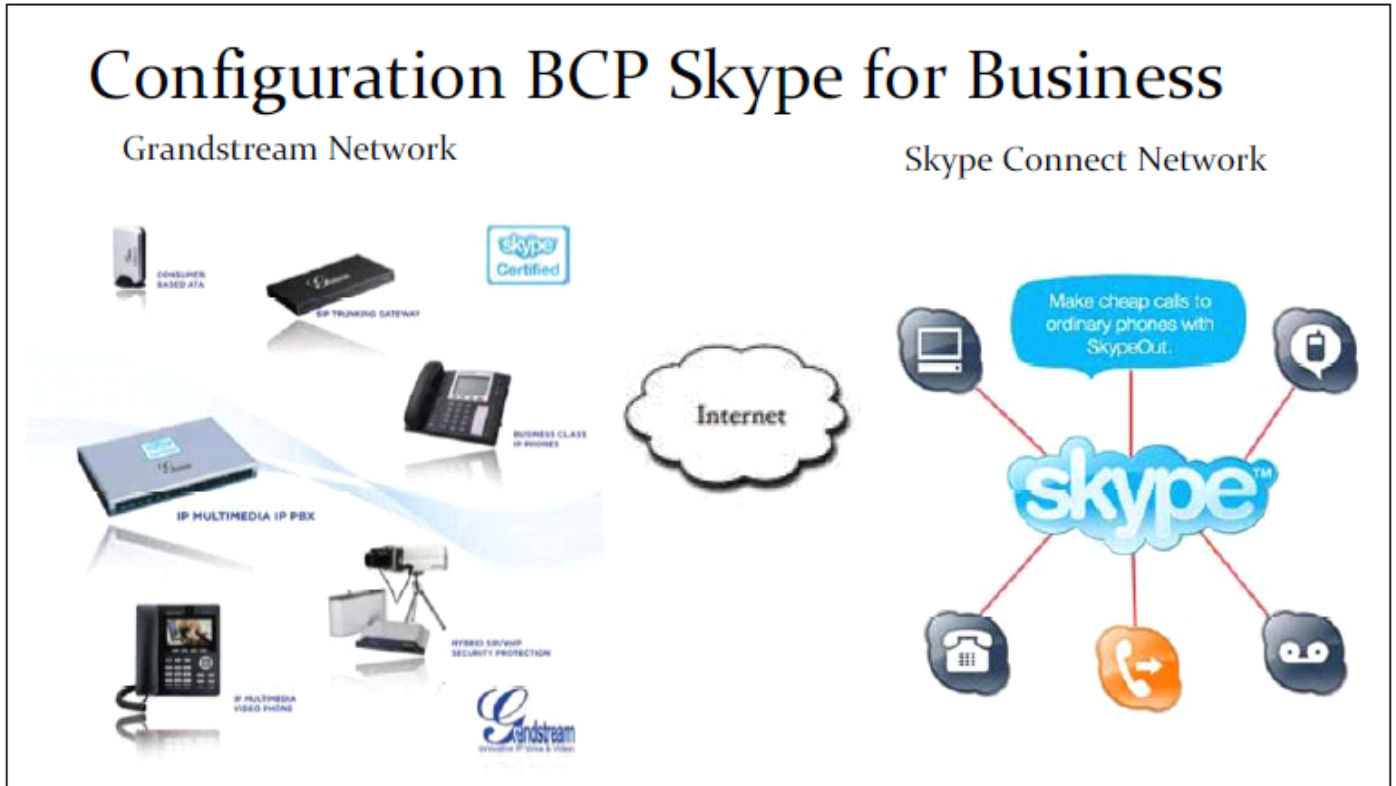
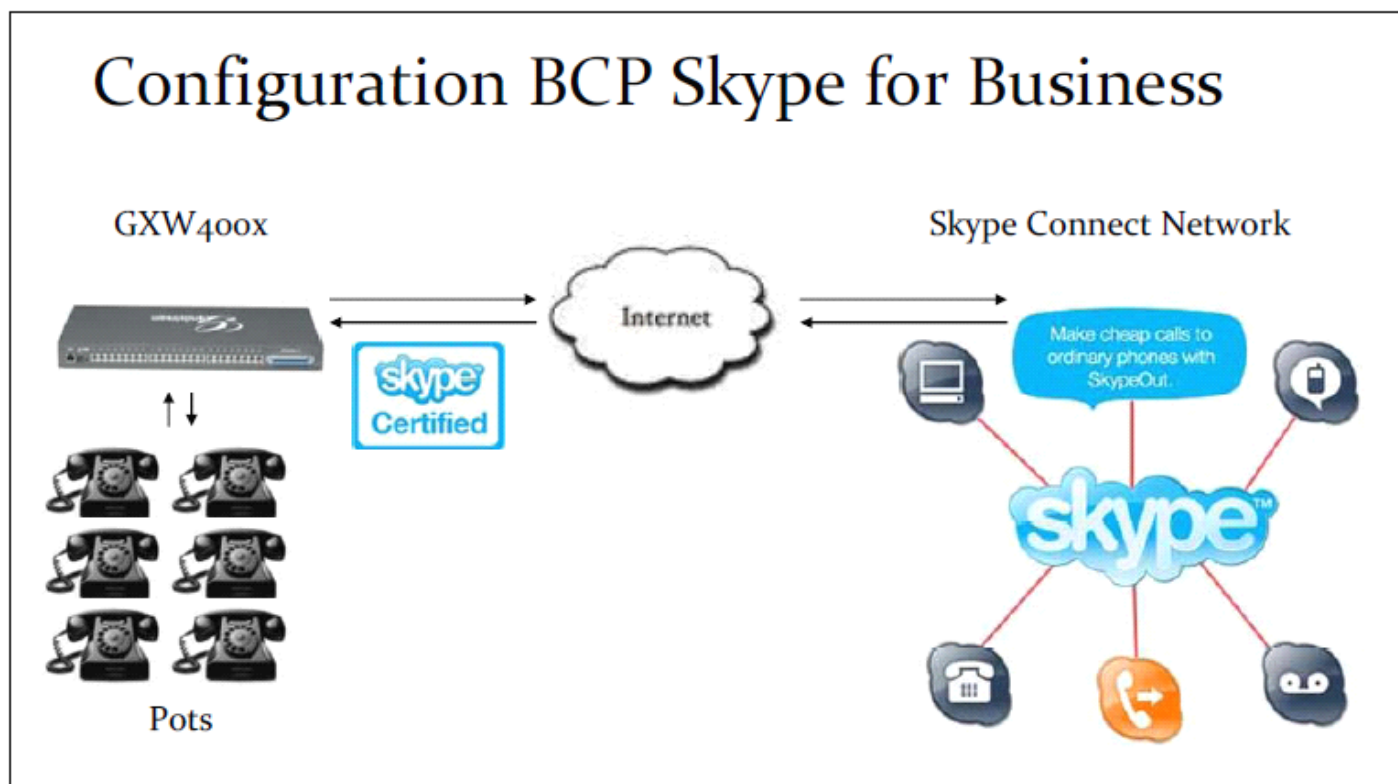


# 潮流网络 CPE 设备与 Skype 实现 SIP 对接的方式

首先非常感谢您对如何运用潮流网络 SIP 设备，对接 Skype's SIP 服务器的兴趣。本文档主要详细说明了潮流网络 GXW40XX 系列 模拟网关以及 GXE502x 系列 IPPBX 与 Skype 实现 SIP 对接的详细配置。



## GXW400x 与 Skype 的 SIP 对接



### SIP 对接配置方式

1. 点击 GXW400x 的 Profilex 选项（图 6-1）
2. 在主 SIP 服务器选项填入 Skype SIP 服务器地址，如下图所示。
3. 在 FXS 口页面，填入 SIP 用户 ID，鉴权 ID，密码和用户名等信息（图 6-2）

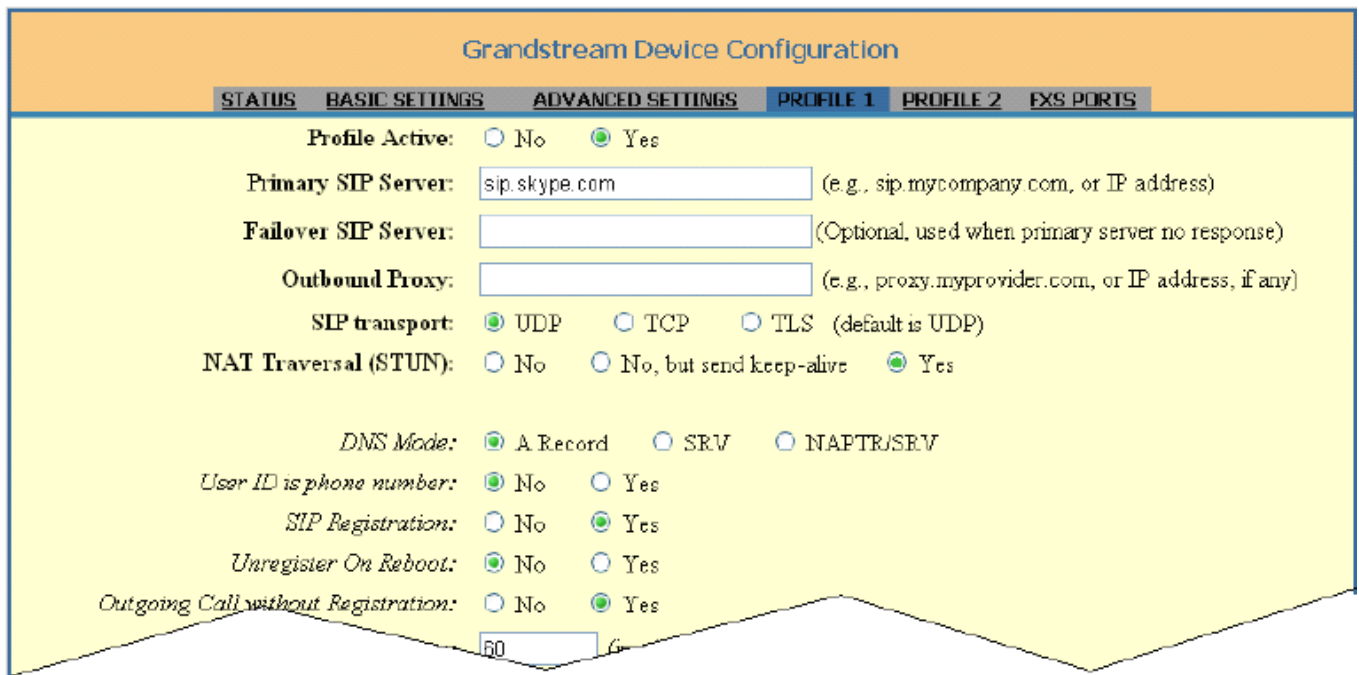
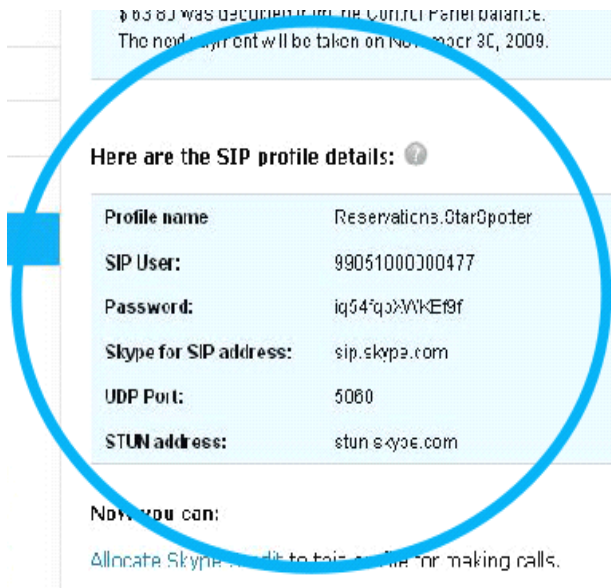


Figure 6-1

注意：以下图片说明了如何在潮流网络 GXW400x 设备上正确填写 SIP Skype BCP 信息。



- SIP User → 帐号和鉴权 ID
- Password → 密码
- Skype for SIP Address → SIP 服务器 URL
- UDP Port → 在 GXE 上，默认的 UDP 端口是 5060。如果您使用的 UDP 端口不是 5060，则要在 **SIP 服务器 URL** 后增加这个 UDP 端口。例如：sip.skype.com:6060
- STUN address → 点击系统配置>系统设置>页面右上角的高级配置>STUN 服务器（这项不在 **SIP 中继** 配置页面）

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS PROFILE 1 PROFILE 2 FXS PORTS

User Settings

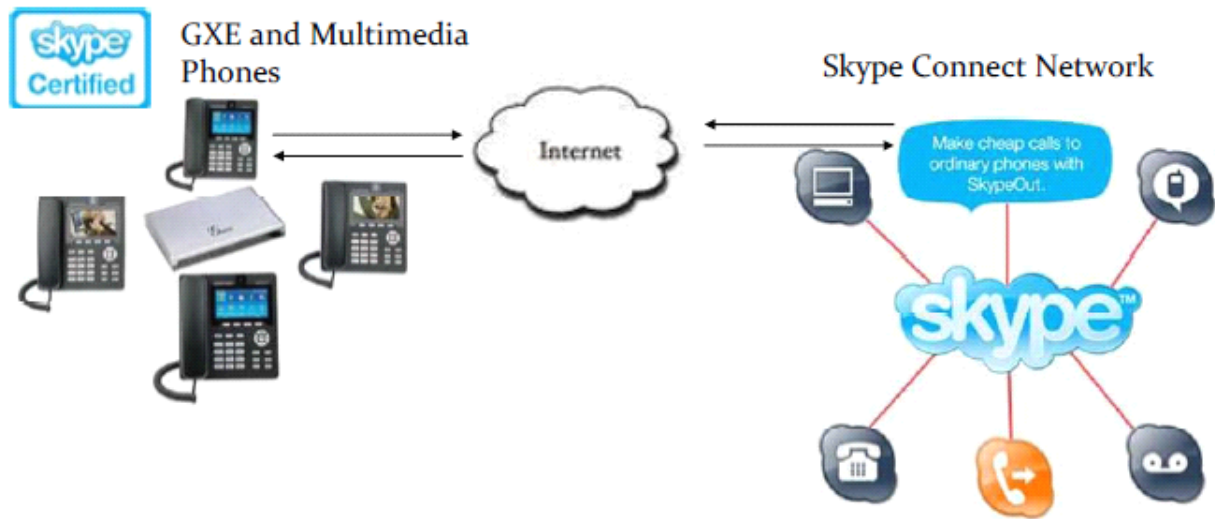
Port	SIP User ID	Authenticate ID	Password	Name	Profile ID	Hunting Group
1	9905xxxxxxxx	9905xxxxxxxx	.....	John Doe	Profile 1	None
2					Profile 1	None
3					Profile 1	None
4					Profile 1	None

Figure 6-2

Profile 帐号必须和 Skype SIP 服务器上的 Profile 号码一致。

## GXE502x 与 Skype 的 SIP 对接

# Configuration BCP Skype for Business



## 在 GXE 的 SIP Trunk 上注册 Skype

1. 进入 GXE 配置页面，点击 **PSTN 中继/电话端口**----> **SIP 中继**----> **增加 SIP 中继**，将显示如下图所示的页面。
2. 在该页面填入 Skype 提供的 SIP profile details 的信息。至于 **中继名称**和**帐号名称**，您可以自定义。下图如何在潮流网络 GXE502x 服务器上正确填写 SIP Skype BCP 信息：

**GXE5024 IPPBX Administration**

→ 增加 SIP中继

Here are the SIP profile details:

Profile name	Reservations.StarQpoter
SIP User:	99051000300477
Password:	iq54fq3xWKE9f
Skype for SIP address:	sip.skype.com
UDP Port:	5060
STUN address:	stun.skype.com

Now you can:  
Allocate Skype... to this... for making calls.

中继名称	
启用标志	<input checked="" type="radio"/> 激活 <input type="radio"/> 关闭
SIP 服务器 URL	
代理服务器 URL	
帐号名称	
帐号	
鉴权ID	
密码	
最大呼叫数	8
拨出前缀	

提交

图 1

Skype BPC 与 GXE502x 配置页面

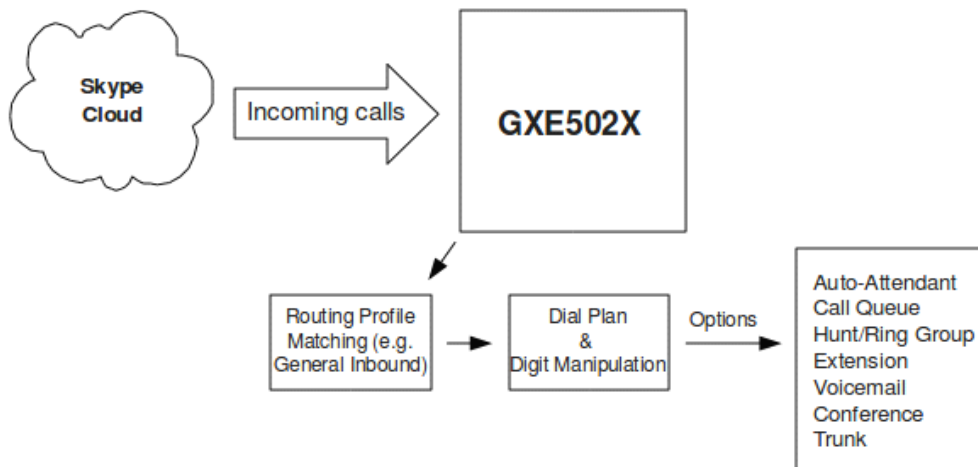
- SIP User → 帐号和鉴权 ID
- Password → 密码
- Skype for SIP Address → SIP 服务器 URL

- UDP Port → 在 GXE 上，默认的 UDP 端口是 5060.如果您使用的 UDP 端口不是 5060，则要在 **SIP 服务器 URL** 后增加这个 UDP 端口。例如：sip.skype.com:6060
  - STUN addresss → 点击系统配置>系统设置>页面右上角的高级配置>STUN 服务器（这项不在 **SIP 中继** 配置页面）
3. 在 Skype 的 BCP 页面，您可以通过下拉选项配置“Max Concurrent Calls Allowed”值，以限制该 SIP 中继允许的最大呼叫数量。当然，您也可以自定义该值。
4. 如果设置了**拨出前缀**，那么 GXE502x 会自动在“General Outbound”拨号脚本增加一条相应的拨号规则。用户可以通过拨打该拨出前缀后，就可以经由与 Skype 对接的 SIP 中继出局。您可以修改此拨出前缀。
5. 点击 **SIP 中继** 页面右上角的**高级配置**，在呼叫路由脚本的已选项列表框中已经默认选择了“General Inbound”。这表示所有经由该 SIP 中继入局的号码都将使用“General Inbound”路由脚本。我们建议您使用此默认值，除非您确定要使用另一个路由脚本。P.S 在已选项列表框中只能选择一项。

## GXE502x 上的呼叫路由

入局呼叫：GXE 上的**呼叫路径**允许管理员通过配置决定入局呼叫的下一跳或设置影响入局呼叫到达下一跳条件。例如，管理员可以设置所有经由与 Skype 对接的 SIP 中继入局后送到自动话务台或群组号码。同时，他可以根据入局号码的 CID，DID 号码，入局日期，星期等增加限制条件。

出局呼叫：GXE 上的**呼叫路径**允许管理员通过配置**拨号规则**和**号码变换**来决定用户拨哪个前缀可以出局，且以什么号码出局。这包括了设置经由与 Skype 对接的 SIP 中继的出局前缀，出局号码变换，授权



等。

更多的详细信息请参考《GXE 502x 用户手册》

[http://grandstream.com.cn/documents/GXE502X\\_UserManual.pdf](http://grandstream.com.cn/documents/GXE502X_UserManual.pdf)



# Skype for SIP

## 快速应用指南



《快速配置手册》介绍了申请GXE502x和Skype Beta 版本的对接帐号的步骤。SIP服务器和Skype Beta 版本是通过Skype BCP进行对接的。Skype官方网站上的《Skype for SIP Business Control Panel User Guide》介绍了BCP的详细使用方法，《Skype for SIP IT Guide》介绍了对接SIP服务器和Skype的配置方式和使用方法。

更多信息，请访问 [skype.com/business/support](http://skype.com/business/support)

在开始之前，您需要：



Icon Key:

Information

Action

Note

### 1 Sign into or register for the Business Control Panel

The Business Control Panel (BCP) lets you manage your company's Skype products and Skype Credit online

When registering for a BCP, please bear in mind this BCP account will be used to administer Skype products and Skype Credit throughout your business. We therefore recommend you create a new Skype account using your company name to register for a BCP. If you do not have Skype, please download the latest version at [skype.com/download](http://skype.com/download) and create a new Skype account.

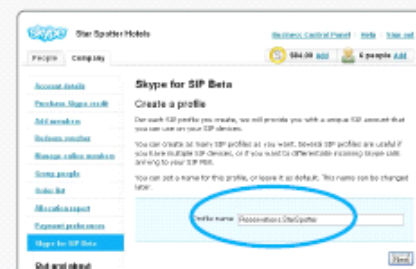
### 2 Activate Skype for SIP and create a SIP Profile

You need to be signed into the BCP in order to access the Skype for SIP Beta settings.

#### 2.1 On the Company tab, click the **Skype for SIP Beta** link.



#### 2.2 Create a new SIP Profile and give it a friendly name so it's easier to remember.







- **2.3** Enter the number of channels you require and complete the purchase instructions on screen (this is the amount of concurrent calls you would like to use with your Skype for SIP Beta product and that these channels are charged on a monthly basis).
- **2.4** After completing the channel purchase your SIP Profile will be created with your login credentials displayed on screen.



- **2.5** Allocate the amount of Skype Credit you need to your SIP Profile by clicking on the **Allocate Skype Credit** section.



- 💡 Please refer to the SIP Profile managing credit section in the Skype for SIP Business Control Panel User Guide for more details on how to set up auto-recharging. This will generate a set of SIP Profile credentials (username, password etc) for you to configure your Skype for SIP certified PBX. We recommend you write them down or print them as you'll need them later.

- 📘 If you do not want to make outbound calls with Skype for SIP Beta please proceed to step 5.



### 3 Configure your Skype for SIP certified PBX for Outbound Calls

- Enter your SIP Profile credentials created in the previous step and configure your Skype for SIP certified PBX as per your manufacturer's instructions.



### 4 Make a test Outbound Call

- Call +1760-660-4690 to test audio quality and connectivity. It's Skype's echo test online number.
- 📘 If you want to receive inbound calls with Skype for SIP Beta proceed to step 6, if not, you have now successfully set up your Skype for SIP profile to make outbound calls.

### 5 Configure your Skype for SIP certified PBX for Inbound Calling

- Refer to your manufacturer's manual for instructions for setting up Skype for SIP Beta.

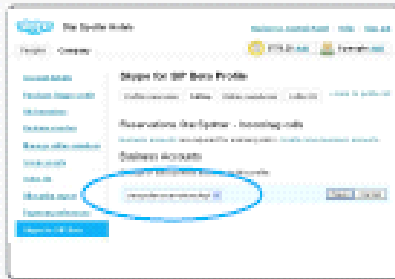


### 6 Set up a Skype Business Account to test Inbound Calls from people with Skype

- **6.1** Create a Business Account in the Business Control Panel by selecting **Add members** in the Company options. Refer to the Business Control Panel User Guide if you require help.



- ➔ **6.2** Assign the newly created Business Account In the Calling section to the SIP Profile you created during step 2.

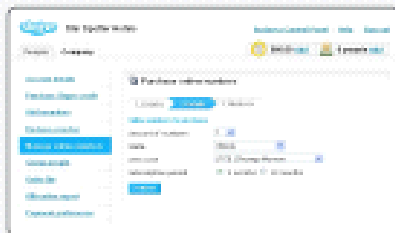


## 7 Make a test Inbound Call from Skype

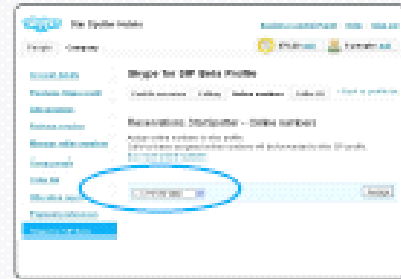
- ➔ Call the Business Account's Skype Name you created In step 6 from Skype.

## 8 Assign an Online Number to receive calls from landlines and mobile phones

- ➔ **8.1** Buy an Online Number In the BCP.

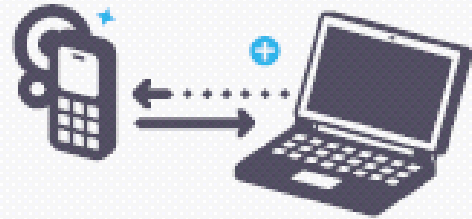


- ➔ **8.2** Associate that Online Number to the SIP Profile you have created.



## 9 Make a test Inbound Call from a landline or mobile phone

- ➔ Call the Online Number from a landline or mobile phone.



- i** You have now successfully set up Skype for SIP Beta for use with your Skype for SIP certified PBX.

For more assistance with setting up and using Skype for SIP Beta please see the resources available at; [skype.com/business/support](http://skype.com/business/support).

- !** Access to a broadband Internet connection is required. Skype is not a replacement for traditional telephone service and cannot be used for emergency calling. Skype for SIP is meant to complement existing traditional telephone services used with a corporate PBX, not as a standalone solution. Skype for SIP users need to ensure all calls to emergency services are terminated through traditional telephone services.



Important Information about your SIP enabled PBX configuration Skype for SIP open beta needs to authenticate your requests against the generated SIP User name and password that was created when you set up your SIP profile in the BCP. In your SIP enabled PBX's configuration ensure you have the following:

- SIP registrations and authentication using your SIP user name and password provided when you set up your SIP profile
- ★ If you are making outgoing calls please ensure your SIP user name is in the From field in the SIP message (ie: 990500000231@sip.skype.com) otherwise Skype will reject your call
- Also ensure you have turned on INVITE authentication if it is a feature as Skype will request authentication even for a SIP INVITE
- ★ If you are making outgoing calls to landline and mobile numbers make sure they are sent to Skype as full international format only (eg country code and landline/mobile numbers even if you are making a local call)
- If you are using Skype for SIP to receive calls, we will send the SIP user name in the Request URI (R-URI), but will not request authentication as you are a trusted user.



**Warning** It is very common that a SIP enabled PBX with an incorrect configuration may operate successfully within the business's telephone system, yet will suffer problematic start up and usage if the credentials are not correct and the SIP user name is not the From header for outgoing calls.

