

Configuring Skype for Business using Grandstream CPE Devices

Thank you for your interest in configuring Grandstream's SIP devices for Skype's SIP Trunking Service. This document describes the basic configuration needed to begin using Skype for SIP Service with Grandstream's GXW400x Analog Gateway Series and GXE502x Series IP PBX.

Configuration BCP Skype for Business

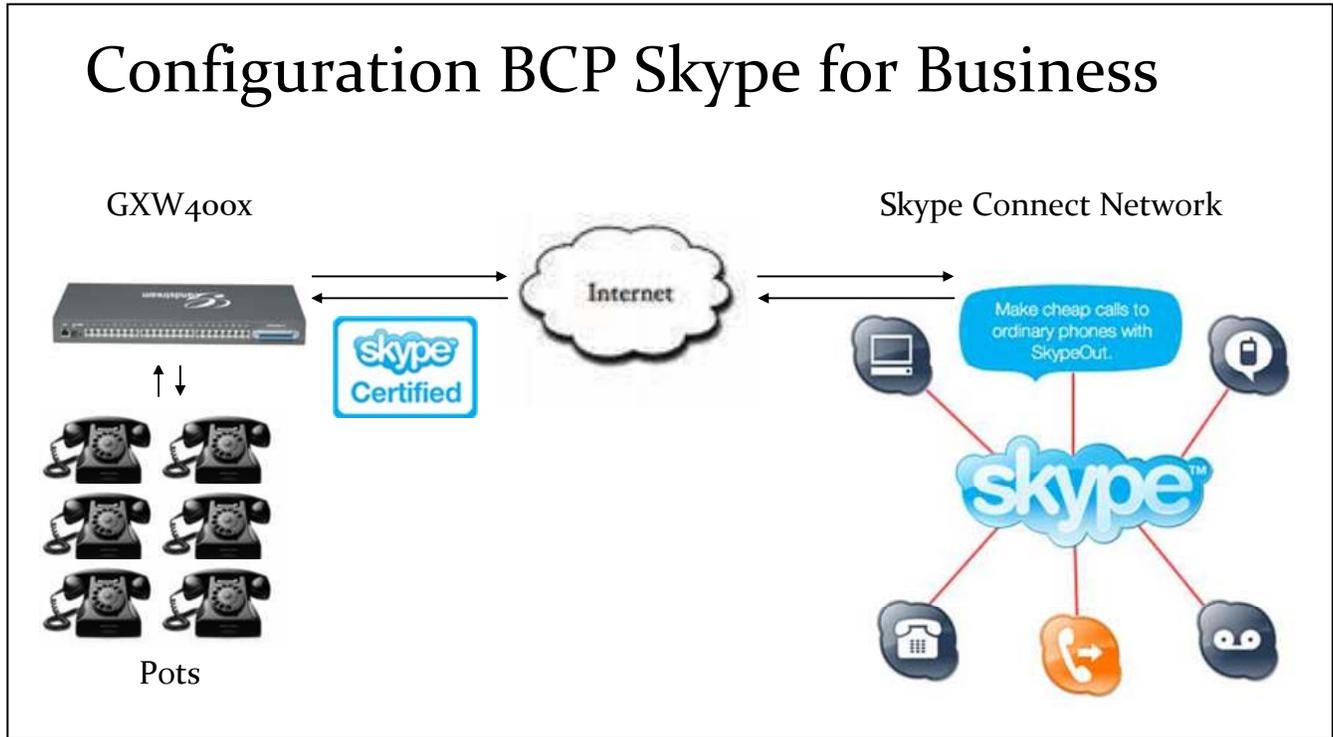
Grandstream Network

Skype Connect Network



Skype for SIP with the GXW400x

Configuration BCP Skype for Business



Configuring Skype for SIP Trunk

1. Click on ProfileX in the configuration page (Figure 6-1).
2. Fill in the Primary Sip Server with the credentials provided by Skype as shown below.
3. Fill in the Sip User ID, Authenticate ID, Password and Name in the FXS Ports page (Figure 6-2)

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS **PROFILE 1** PROFILE 2 FXS PORTS

Profile Active: No Yes

Primary SIP Server: (e.g., sip.mycompany.com, or IP address)

Failover SIP Server: (Optional, used when primary server no response)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

SIP transport: UDP TCP TLS (default is UDP)

NAT Traversal (STUN): No No, but send keep-alive Yes

DNS Mode: A Record SRV NAPTR/SRV

User ID is phone number: No Yes

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Figure 6-1

Note: Below we provide a match of what it is obtained from Skype's BCP and the ProfileX and FXS Port pages in the GXW40xx.

Here are the SIP profile details:

Profile name	Reservations.StarOpoter
SIP User:	99051000300477
Password:	iq54tq3WKE9f
Skype for SIP address:	sip.skype.com
UDP Port:	5060
STUN address:	stun.skype.com

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

- SIP User → Sip User ID and Authenticate ID
- Password → Password
- Skype for SIP Address → Primary Sip Server
- UDP Port → The GXW assumes port 5060 as the default value. If other port number is given, append it to the Primary Sip Server URL using a colon and the port number (e.g. Port 6060 sip.skype.com:6060).
- STUN address → Set Nat Traversal (STUN) to YES and enter the address in Advance Settings page > Stun Server

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS PROFILE 1 PROFILE 2 **FXS PORTS**

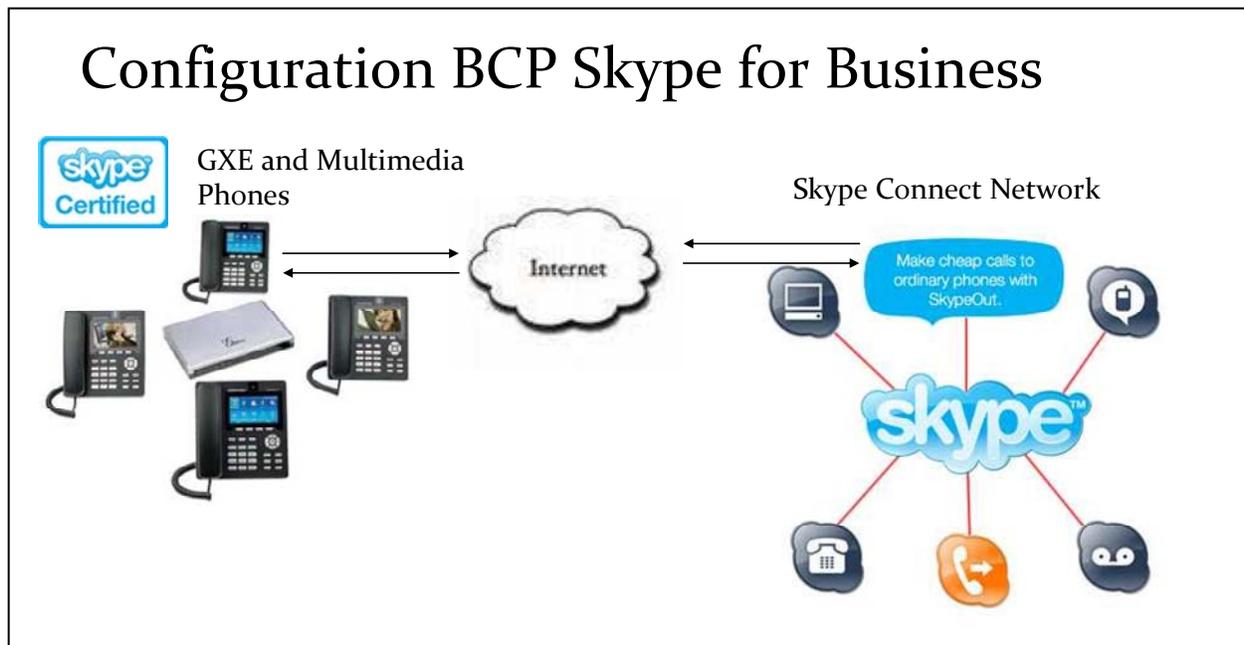
User Settings

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

Figure 6-2

The Profile ID number needs to correspond with the Profile number where you entered the Skype Sip server.

Skype for SIP with the GXE502X



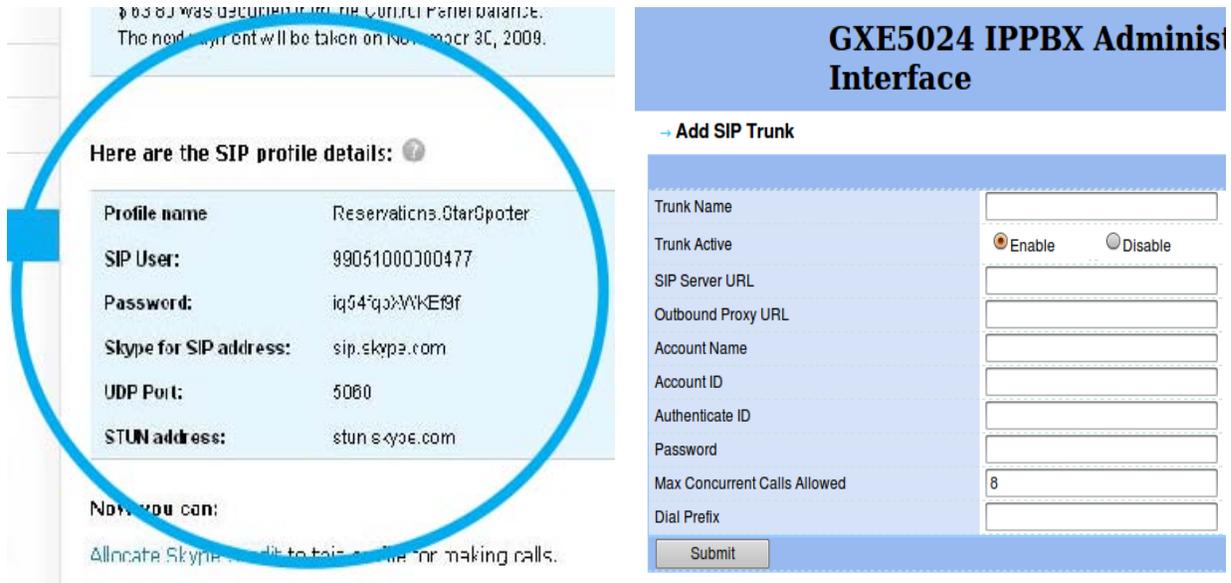
Configuring Skype for SIP Trunk

1. Click on Trunk/Phone Lines on the left menu bar and then click on "SIP Trunk" to load the SIP Trunk configuration page (Figure 7-1).

2. Fill in the SIP Trunk with the credentials provided by Skype as shown below. For the fields "Trunk Name" and "Account Name" you can use an arbitrary name. Below we provide a match of what it is obtained from Skype's BCP and the SIP Trunk Configuration fields in the GXE502X.

- SIP User → Account ID and Authenticate ID
- Password → Password
- Skype for SIP Address → SIP Server URL
- UDP Port → The GXE assumes port 5060 as the default value. If other port number is given append it to the SIP Server URL using a colon and the port number (e.g. Port 6060 sip.skype.com:6060).
- STUN address → * (System Configuration > System Settings (Advanced) > STUN Server).

* Outside the SIP Trunk Configuration page.

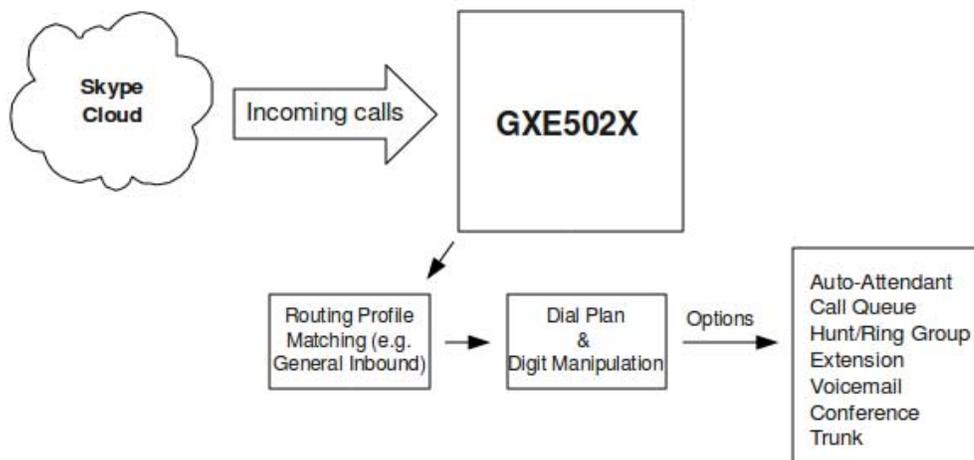


Skype BCP and GXE5024 Web Interfaces (Figure 7-1)

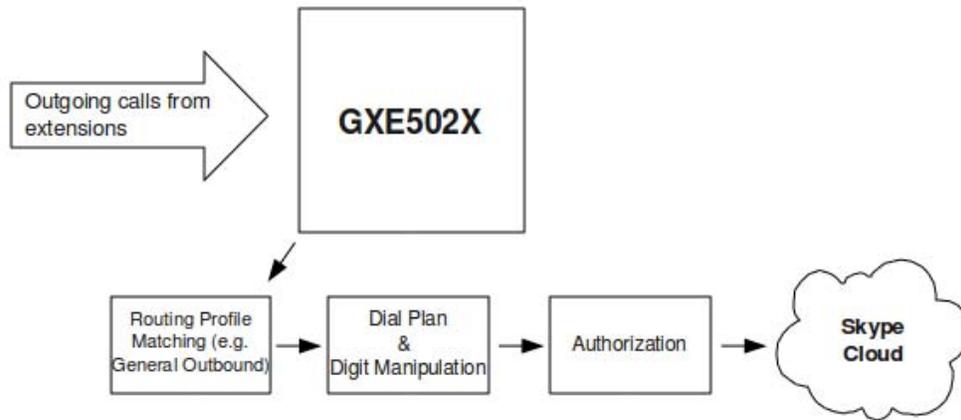
3. To set a limit on the number of calls allowed through this trunk, please set the “Max Concurrent Calls Allowed” field to the limit allowed by Skype provider, or a lower number if you wish.
4. If desired set the “Dial Prefix” to a number users will use to make outbound calls with this trunk. This rule will be added automatically to the “General Outbound” Call Routing Profile and can be modified later.
5. “General Inbound” is chosen as the default Call Routing Profile and we recommend you to keep it that way unless you are sure about applying these modifications. The Call Routing Profile assigned to a trunk will determine what will happen when a call comes into the trunk. The Selected List should only include one profile.

Call Routing in GXE502X

Inbound Calls. GXE Call Routing capabilities allow the administrator to decide where the call will be routed and/or any additional conditions that may impact the routing of the call. For example the organization may want all of the calls to be answered by an Auto-Attendant or a Ring a Group. At the same time specific conditions can be added depending on the Caller Party CLI Number (Caller Line Identification), DID and any specific date, weekday or time of the day.



Outbound Calls. GXE Call Routing capabilities allows the administrator to create dial plan and digit manipulation rules to delimit the phone numbers users will be able to dial and how users will be dialing out. This may include using a prefix to access the SIP for Skype Trunk, Digit Manipulation for Local Digit Dialing, Permissions and so forth.



For more information please refer to the GXE User Manual:http://www.grandstream.com/support/gxe_series/gxe502x/documents/gxe502x_user_manual_english.pdf



Skype for SIP Quick Start Guide



This quick start guide outlines the main steps to set up the open beta version of Skype for SIP. Skype for SIP Beta is set up via Skype's Business Control Panel (BCP). For detailed information on how to use Skype's BCP, please refer to the Skype for SIP Business Control Panel User Guide and Skype for SIP IT Guide about implementing and using Skype for SIP.

Further support resources can be found at skype.com/business/support

Before you get started you will need



An internet connection with a public IP address



An online Skype for SIP certified PBX



The ability to configure your SIP-enabled PBX



An accepted payment method



A Skype Account

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 and create a new Skype account.

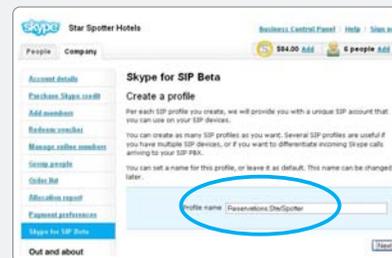
2 Activate Skype for SIP and create a SIP Profile

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.

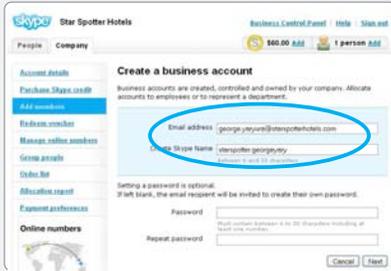
5 Configure your Skype for SIP certified PBX for Inbound Calling

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

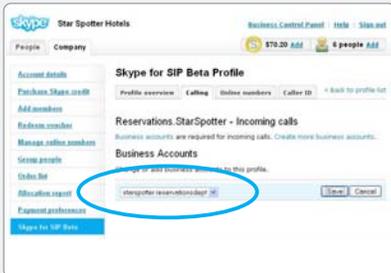


6 Set up a Skype Business Account to test Inbound Calls from people with Skype

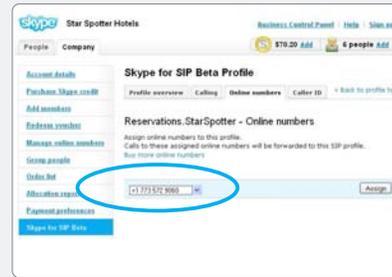
➤ **6.1** Create a Business Account in the Business Control Panel by selecting **Add members** in the Company options. Refer to the Business Control Panel User Guide if you require help.



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



- ➔ **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.

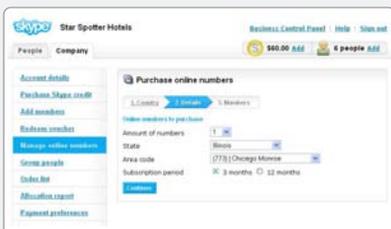


7 Make a test Inbound Call from Skype

- ➔ 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.



- 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.

- 💬** 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.

