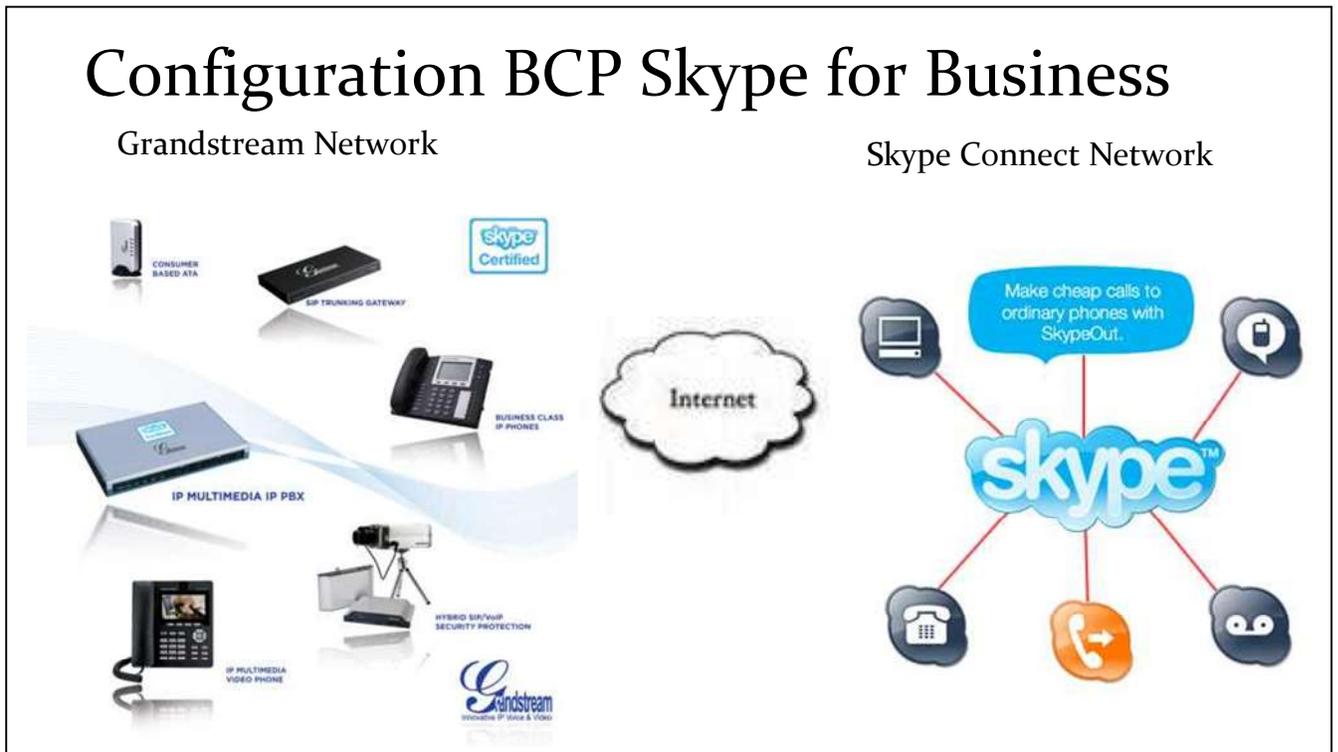


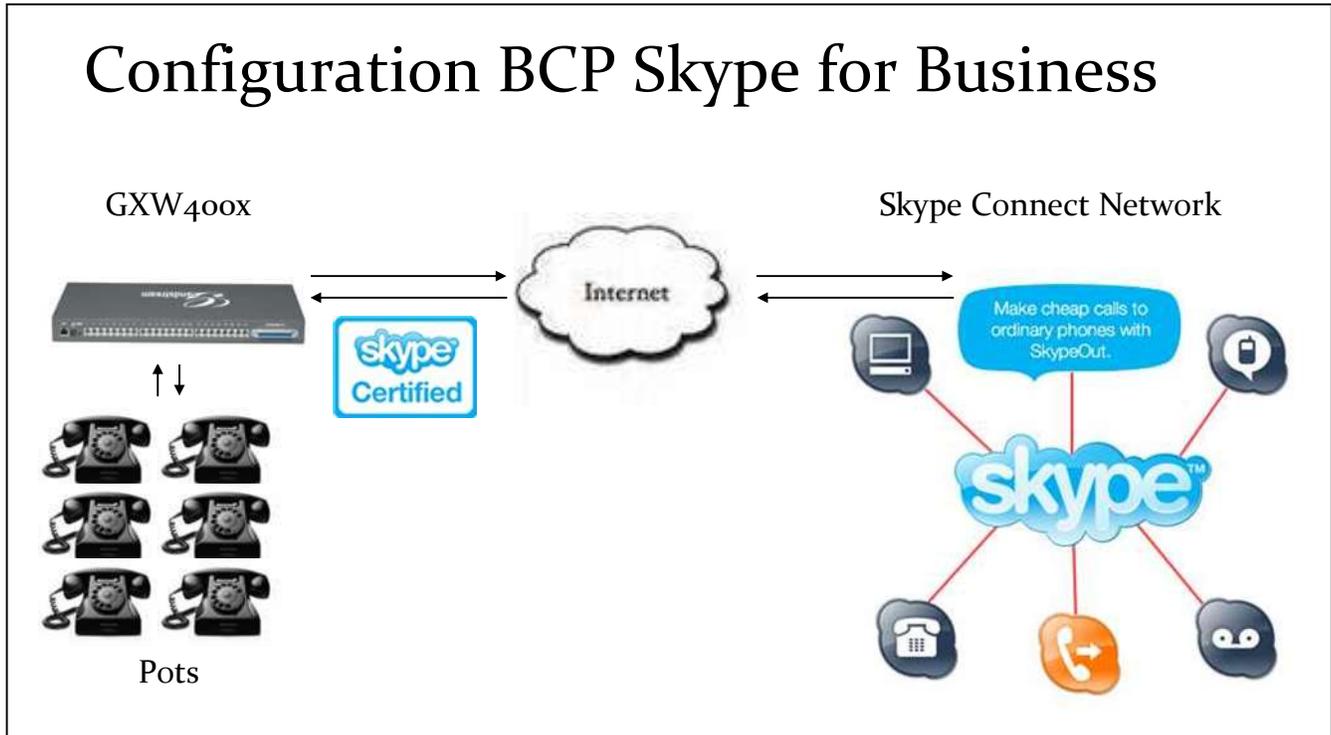
Configuring Skype For Business using Grandstream CPE Devices

Thank you for interest in configuring Grandstream SIP devices for Skype's SIP Trunking Service. This document describes the basic configuration need to start your Skype for SIP Service using the GXW400x and GXE502x.



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.

This page was deprecated in the Control Panel interface. The next migration will be taken on November 30, 2009.

Here are the SIP profile details: ?

Profile name	Reservations.StarOpoter
SIP User:	99051000300477
Password:	iq54fq3WKE9f
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

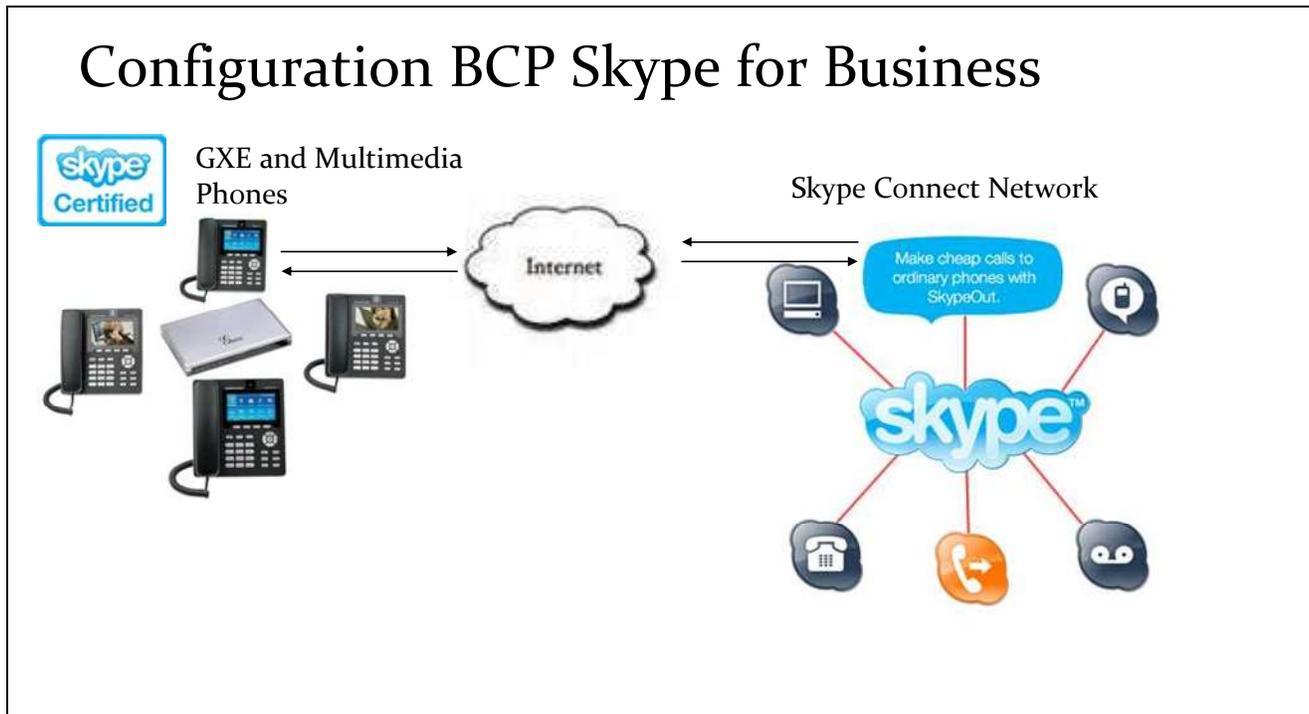
User Settings

Port	SIP User ID	Authenticate ID	Password	Name	Profile ID	Hunting Group
1	<input type="text" value="9905xxxxxxxxxx"/>	<input type="text" value="9905xxxxxxxxxx"/>	<input type="password" value="*****"/>	<input type="text" value="John Doe"/>	Profile 1 <input type="button" value="v"/>	None <input type="button" value="v"/>
2	<input type="text"/>	<input type="text"/>	<input type="password"/>	<input type="text"/>	Profile 1 <input type="button" value="v"/>	None <input type="button" value="v"/>
3	<input type="text"/>	<input type="text"/>	<input type="password"/>	<input type="text"/>	Profile 1 <input type="button" value="v"/>	None <input type="button" value="v"/>
4	<input type="text"/>	<input type="text"/>	<input type="password"/>	<input type="text"/>	Profile 1 <input type="button" value="v"/>	None <input type="button" value="v"/>

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



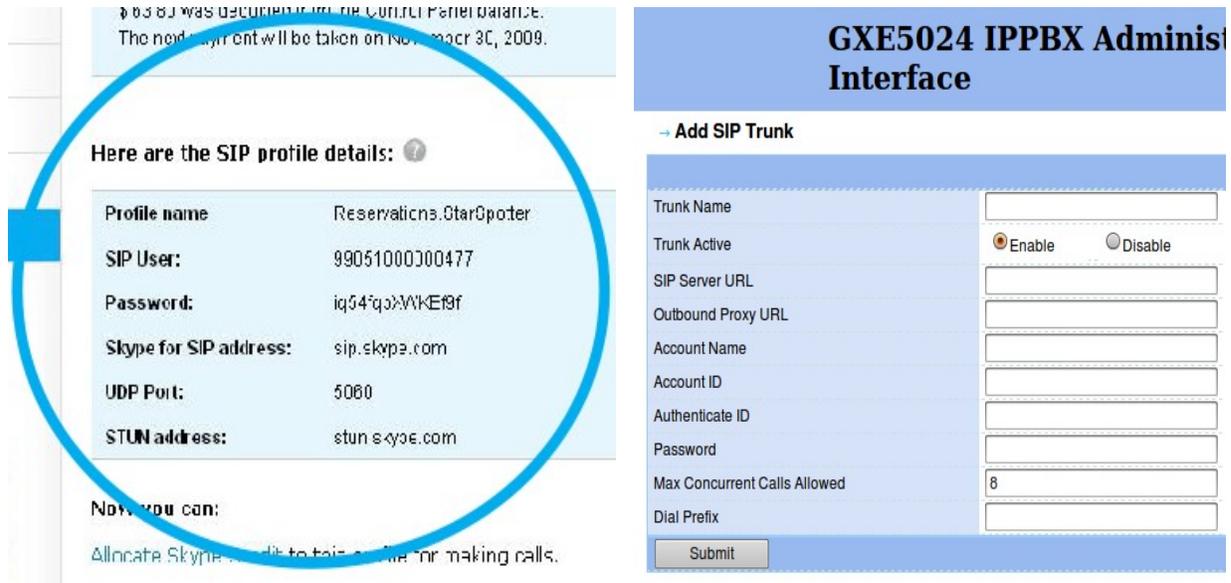
Configuring Skype for SIP Trunk

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 addresss → * (System Configuration > System Settings (Advanced) > STUN Server).

* Outside the SIP Trunk Configuration page.



The screenshot shows the GXE5024 IPPBX Administration Interface. On the left, a table displays SIP profile details for a profile named 'Reservations.StarSpoter'. A blue circle highlights this table. On the right, the 'Add SIP Trunk' form is visible, featuring fields for Trunk Name, Trunk Active (radio buttons for Enable/Disable), SIP Server URL, Outbound Proxy URL, Account Name, Account ID, Authenticate ID, Password, Max Concurrent Calls Allowed (set to 8), and Dial Prefix. A 'Submit' button is at the bottom of the form.

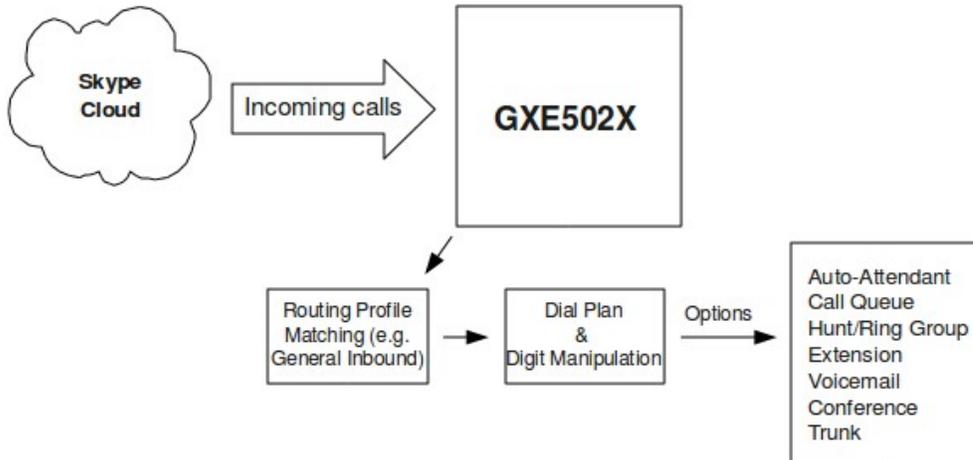
Here are the SIP profile details:	
Profile name	Reservations.StarSpoter
SIP User:	9905100000477
Password:	iq54fq>WKEf9f
Skype for SIP address:	sip.skype.com
UDP Port:	5060
STUN address:	stun.skype.com

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

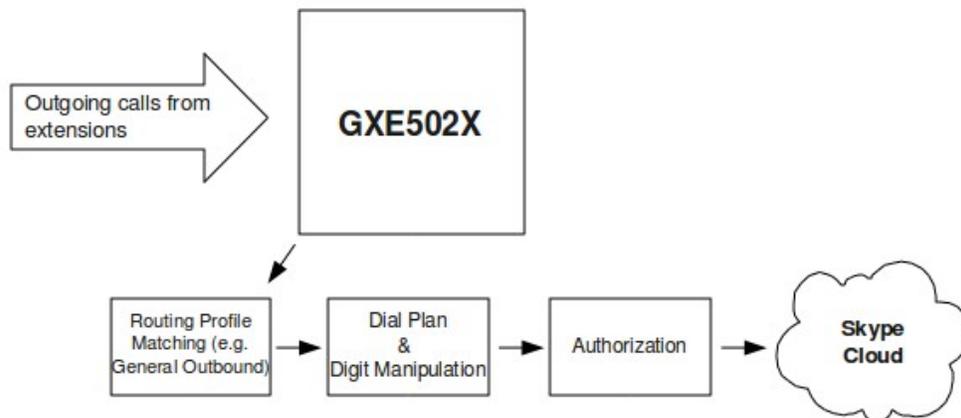
3. To set a limit on the number of calls allowed through this trunk, please set the “Max Concurrent Calls Allowed” field to the limit allowed by Skype provider, or a lower number if you wish.
4. If desired set the “Dial Prefix” to a number users will use to make outbound calls with this trunk. This rule 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 allows 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