

Skype for SIP configuration

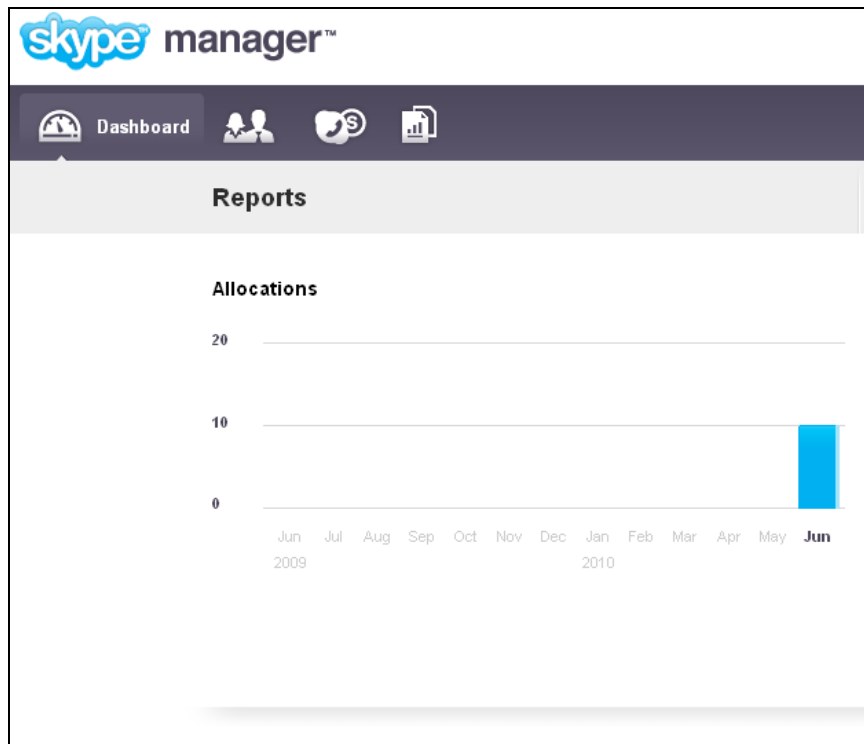
Skype now supports a business class SIP line connection. This document will guide you through the steps required to configure Skype for SIP integration. In order for you to start the Skype configuration, you will need to make a Skype account.

A Business Manager account will need to be created in order to utilize Skype for SIP. Once your account is created, Skype requires that you download their software application and run it at least one time. Once downloaded you will be able to create Skype accounts.

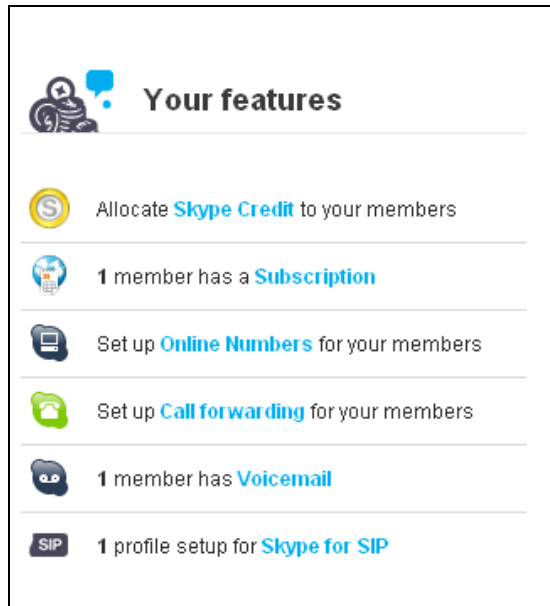
To accomplish this, you will need to go to the Skype website at www.skype.com.

▶ **STEPS:**

- 1 Login to the Skype Manager dashboard using your username and password that you created when you setup the Skype account.
- 2 You will be presented with the following page.



- 3 You will find features for your account on this page.



- 4 Select Skype for SIP option.
- 5 Setup a Profile. Once the Profile is created you can assign a channel. You will need a channel for every concurrent call you intend to carry. So if you would like 4 consecutive calls to be carried by the SIP trunk then you will need to purchase 4 channels.
- 6 Setup your outbound and inbound. If you would like an inbound number you must either port a number you already own, or you can purchase a new one. There is a fee for new numbers. If you do not require inbound then no number will need to be purchased. For outbound you must bind the out bound to your account that you initially created. This will result in outbound long distance fees that may be associated when dialing.
- 7 Configure the IPitomy's PBX system for the Skype interface. Authentication may be done either through a User Name and Password combination, or through IP address binding. It is recommended that you use a Password and Username combination.
- 8 Set a SIP trunk up in your IPitomy system. Use the following diagram for an example of what parameters need to be configured.
- 9 In the diagram below, no caller ID is configured. However, you can populate that field if required. Your call Limit should match the number of channels you purchased.

SIP Provider		
Name:	<input type="text" value="99051000111727"/>	
User Type:	friend ▼	
DTMF Mode:	rfc2833 ▼	
Host:	<input type="text" value="sip.skype.com"/>	
Port:	<input checked="" type="radio"/> Default <input type="radio"/> Custom <input type="text"/>	
Register:	<input type="radio"/> Yes <input type="radio"/> No <input checked="" type="radio"/> Custom <input type="text" value="register => 99051000111"/>	
Authentication:	<input type="radio"/> Yes <input checked="" type="radio"/> No <input type="radio"/> Custom <input type="text"/>	
Auth User:	<input checked="" type="radio"/> Default <input type="radio"/> Custom <input type="text"/>	
From User:	<input type="radio"/> Default <input checked="" type="radio"/> Custom <input type="text" value="99051000111727"/>	
From Domain:	<input type="radio"/> Default <input checked="" type="radio"/> Custom <input type="text" value="sip.skype.com"/>	
Realm:	<input checked="" type="radio"/> Default <input type="radio"/> Custom <input type="text"/>	
Outbound Proxy:	<input checked="" type="radio"/> Disabled <input type="radio"/> Enabled <input type="text"/>	
Username:	<input type="text" value="99051000111727"/>	
Secret:	<input type="text" value="uixTKuKX7KPVJY"/>	
Inbound Caller ID:	<input type="text"/>	
Outbound Caller ID Name:	<input type="text"/>	
Outbound Caller ID Number:	<input type="text" value="99051000000080"/>	
Call Limit:	<input type="text" value="1"/>	
Qualify:	<input type="text" value="30000"/>	
Default Destination:	<input type="text" value="Menus"/> ▼ <input type="text" value="Menu: Main Menu"/> ▼ ...	
Dial Prefix:	<input type="text" value="+"/>	
Area Code:	<input type="text"/>	
Generate Ringing on outbound calls:	<input type="checkbox"/>	
Allow Outbound Caller to transfer:	<input checked="" type="checkbox"/>	
Allow Call Recording:	<input checked="" type="checkbox"/>	
Ext CID Override:	<input checked="" type="checkbox"/>	
Restrict CID Override:	<input type="checkbox"/>	
Can Reinvite:	<input checked="" type="radio"/> Yes <input type="radio"/> No <input type="radio"/> N/A	
Insecure:	Invite ▼	
Allow Codecs:	ITU G.711 ulaw 64Kbps, (US) <input checked="" type="checkbox"/> ITU G.711 alaw 64Kbps <input checked="" type="checkbox"/> ITU G.723.1 (5.3/6.3 Kbps, 30ms frame size) <input type="checkbox"/> ITU G.728 - 16/24/32/40 Kbps <input type="checkbox"/> GSM - 13 Kbps (full rate), 20ms frame size <input type="checkbox"/> iLBC - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame <input type="checkbox"/>	

10 Once the parameters are set, Save and Apply Changes.

11 Add your SIP trunk to your outbound routes, and additionally set your inbound answer points in call routing.

NOTE: You must populate the Custom Domain and User field with the appropriate information from your Skype Account. Additionally to call out you will need to add a “+” to the outbound prefix.