SiSky

User Manual

(Version: 2.6.0.1)

Yeastar Technology Co., Ltd

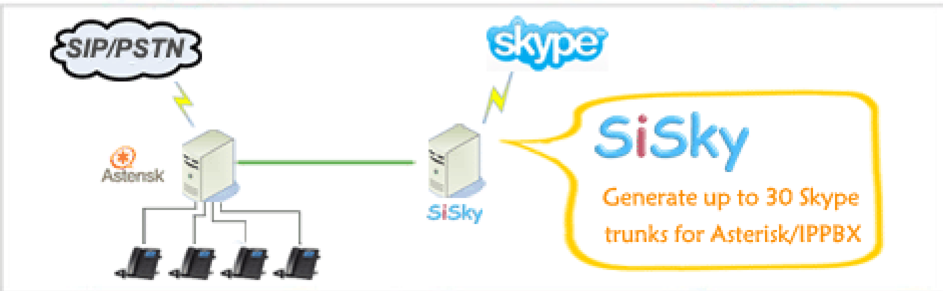
Table of Contents 0

Introduction 1

You may skip section 1- *Introduction* and directly move to section 2- *Before you Proceed*. However, we recommend reading through the introduction to familiarize yourself with the features and functions of SiSky and help improve its operation. Thank you for using SiSky!

SiSky — Up to 30 Skype Trunks Gateway

Nowadays, Skype is very popular and you may found many customers are Skype users. Let your customers who are used to Skype to contact with you quickly and conveniently is becoming the main job of your Asterisk/IPPBX system. SiSky is the best solution for you to connect PBX to the Skype world. However, Skype is mostly limited to personal usage. In order to bring Skype Internet calls to an office environment, Yeastar has succeed in developing the solution SiSky that saves the outstanding features of Skype flexibly and meets the enterprises' requiremens of multiple trunks, sharable and usable at anytime.
Attaching SiSky to the Asterisk/IPPBX, free call service will be available; enterprises will communicate and collaborate with clients and partners easily, efficiently and economically.



1.1 Functions

Website Click-to-Call: Receive calls from website and SkypeIn number with multi-trunk.
Interoffice Trunking: Builds enterprise branch communications network through Skype with optimal design.
Skype Trunking:
• Skype Incoming: Receive Skype calls from customers who are using Skype.
• Outgoing to Skype: Make free calls to numerous Skype users by

office phone.

• SkypeOut: Provide trunks of making landline/mobile calls.

Remote Extension: Use Skype as Fixed Remote Extension.

• SiSky include a SIP Server, it can work as Asterisk'/IPPBX's SIP Trunk or SIP Extension.

• Remote monitoring and managing by Web.

• Independent Phonebook Utility includes public & private phonebook.

• Optional Multi-User mode feature allows every user to create his own private contacts.

• Play the auto attendant for Skype incoming call, and forward it to extension or Skype

• Call Log & Call Statistics.

• Database sharable among cascade connection of SiSky Servers.

• Backup and restore functions of database.

1.2 Features

● Supports 30 Skype trunks (concurrent calls) on one computer

● Supports cascade connection of multiple computers to unlimited extend Skype trunks

● Builds enterprise branch communications network through Skype with optimal design

● Company Skype ID, add the effective voice trunk to Internet

● Automatically finds idle trunk to transfer Skype Incoming calls.

● Automatically finds idle trunk to make Skype Outgoing calls.

● Delivers Skype functionality into enterprise extension system

● Enable customized Speed-dial or PSTN matchable dialing plan

● Sets a dedicated phone extension to ring for Skype incoming call

● Multi-User Mode allows users to create and manage their own contacts

● 'Utility' allows every user to export his personal Skype contacts into his private phonebook

● Noise reduction, echo cancellation and compensation for losing packet techniques ensure the excellent voice quality

● Remote monitoring by Web

● Utilize web for remote administration SiSky

● Independent Phonebook Utility includes web administration of public phonebook

● Excellent the Call Log function including call type, rate per minute and total price

● Add backup and restore functions of database

1.3 Minimum System Requirements

● Hardware:

PC：Idle PCI Slot available

Concurrent calls		3	5~6	8~9	15	23	30
PC Requirement	CPU	Celeron 2.8G	P4 2.66G	P4 2.8G Dual Core	Intel Core Duo 1.86G	Intel Core 2 Quad 2.4G	Intel Xeon 2 CPUs
	Memory	512M	1G	1G	2G	2G	3G

● Operating System：Windows XP Professional + SP2 or
Windows Server 2003 + SP2

● Internet Connection：Different numbers of Skype concurrent calls require different bandwidth; each port average occupies 3Kb/s to 16 Kb/s.

Before You Proceed 2

Note the following precautions before you install the software.

1. Choose a computer with suitable configuration to run as the server:

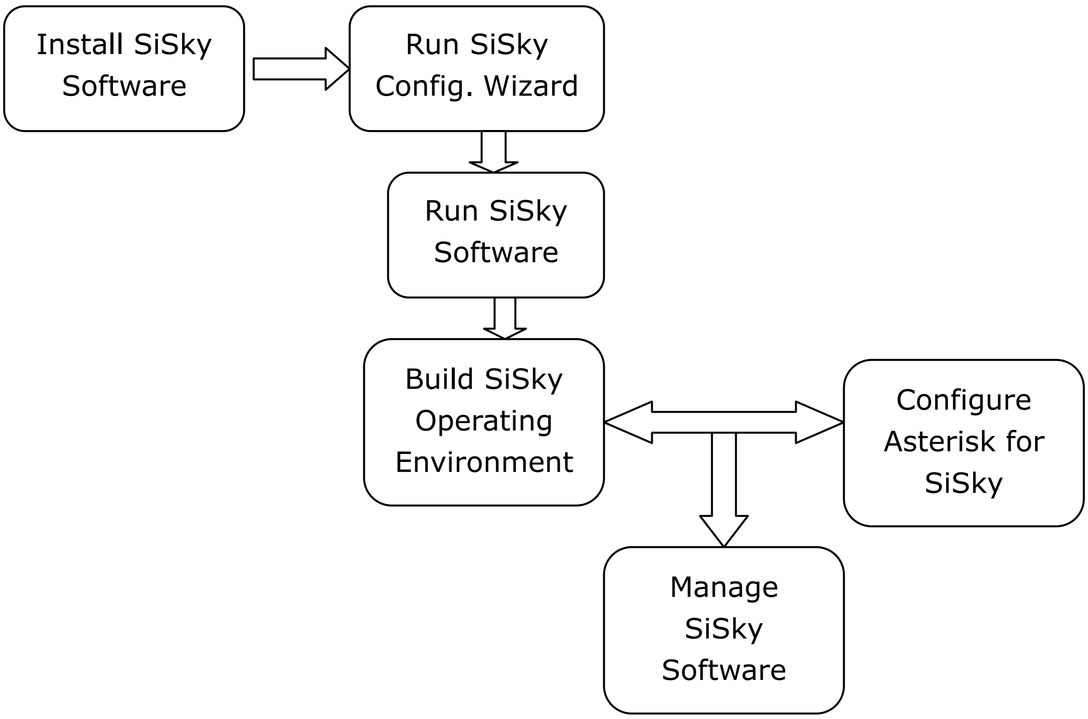
Concurrent calls		3	5~6	8~9	15	23	30
PC Requirement	CPU	Celeron 2.8G	P4 2.66G	P4 2.8G Dual Core	Intel Core Duo 1.86G	Intel Core 2 Quad 2.4G	Intel Xeon 2 CPUs
	Memory	512M	1G	1G	2G	2G	3G

2. Install OS Windows XP Professional SP2 or Windows Server 2003 SP2.
3. Don't use the clone way to install the operating system of SiSky Server PC.
4. Make sure SiSky Server PC designated for SiSky has a clean system with only anti-virus installed.
5. Make sure SiSky Server PC designated for SiSky has no Skype applications installed.
6. Make sure SiSky Server PC has no SIP Server or SIP Softphone installed.
7. Make sure SiSky Server PC designated for SiSky has no domain controller installed and the disable the password complexity.
8. Make sure SiSky Server has a dedicated broadband access to ensure voice quality.
9. Make sure to login on SiSky Server with an account that has administrator privileges.
10. Please don't login SiSky server by "Remote Desktop Connection", but use VNC in case you need remote login SiSky server.

Installation Procedure 3

Below is SiSky installaton and configuration flow diagram that gives you an overview of all the steps required in installing SiSky.

SiSky Installation Flow



Installing SiSky Software 4

This section shows how to install SiSky software on the PC.

1. Download Install software from website http://www.yeastar.com. And Double-Click to Start Installation Process.

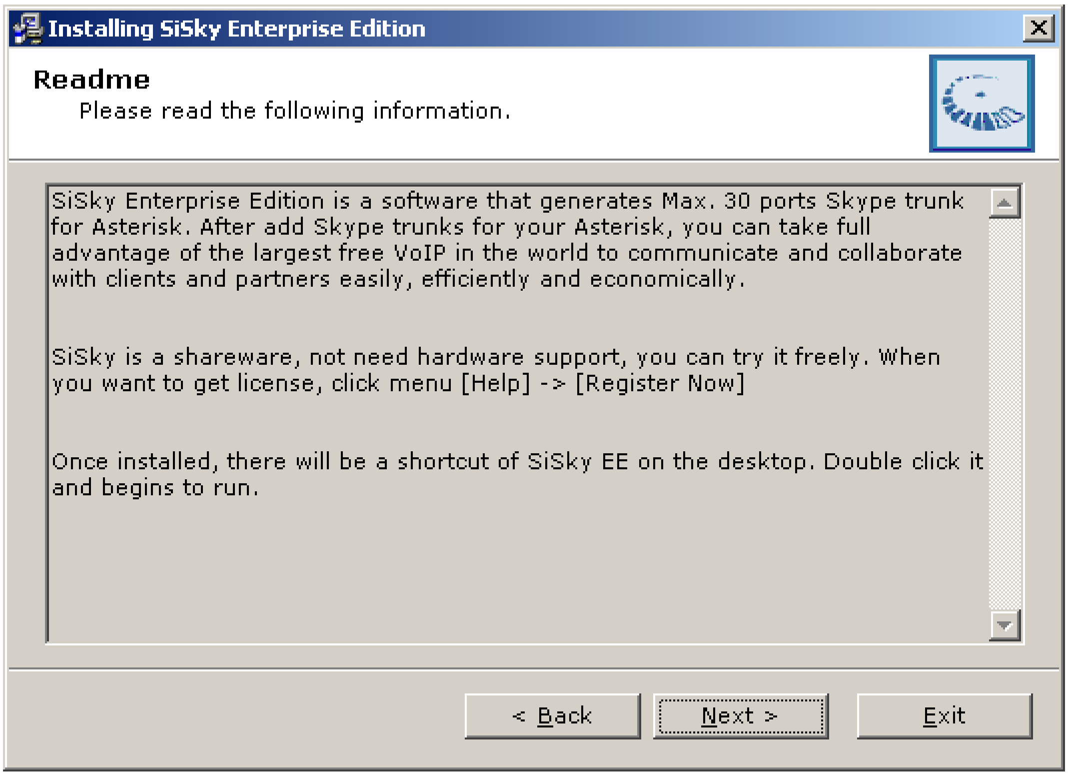
2. A **Welcome to the SiSky Installation** screen will come up. Click **Next** to continue. See Figure 1.

<Figure 1>

3. Read through the software **License Agreement**, **select I agree with the above terms and conditions**, and then click **Next** to continue. See Figure 2.

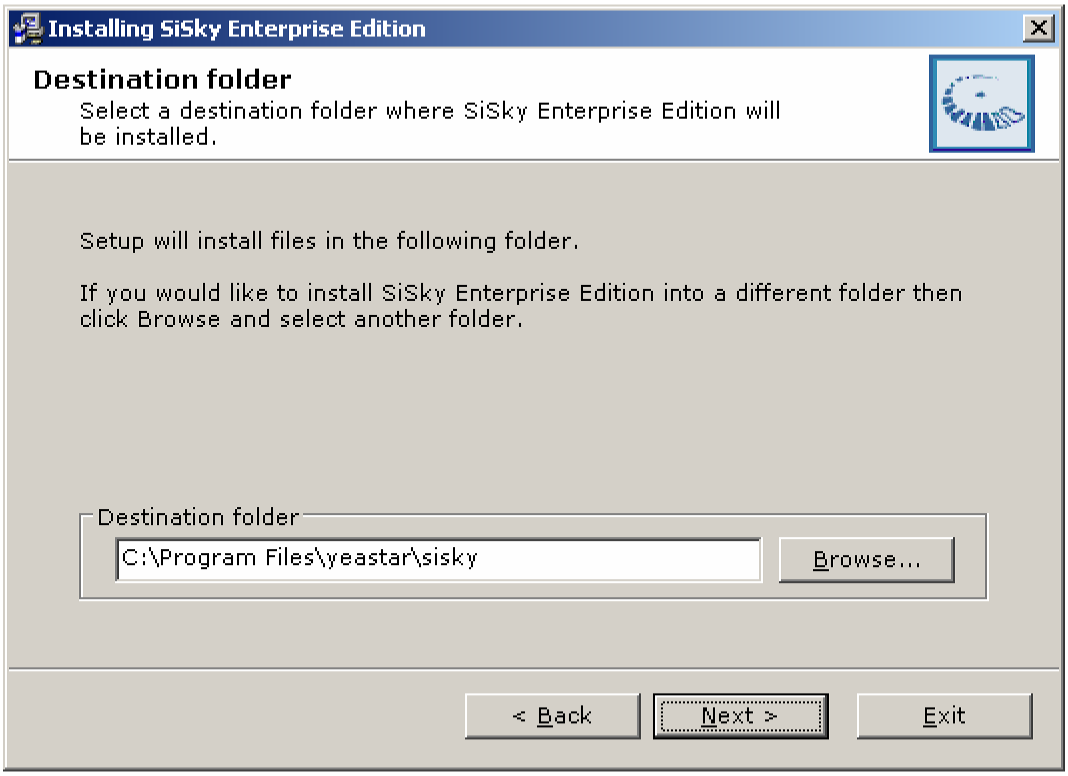
<Figure 2>

4. An Readme info window will appear. Click **Next** to continue.



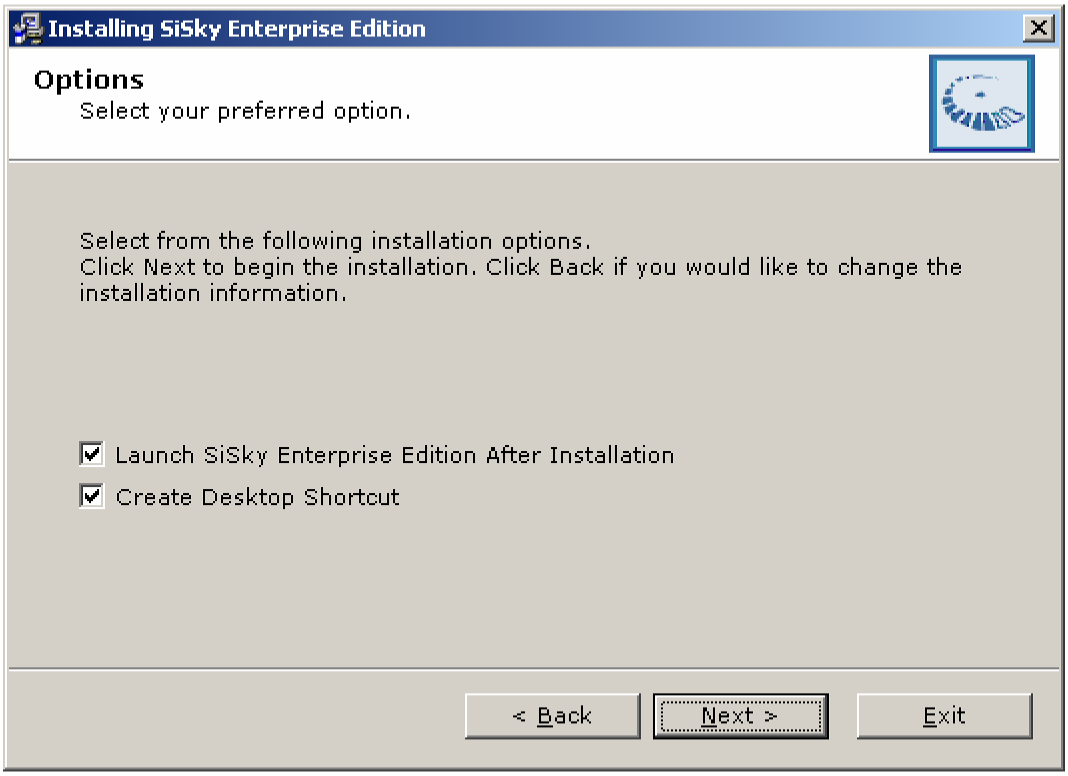
<Figure 3>

5. The **Destination folder** window will offer you the option where you would like SiSky to be stored on your computer. Click **Next** to continue.



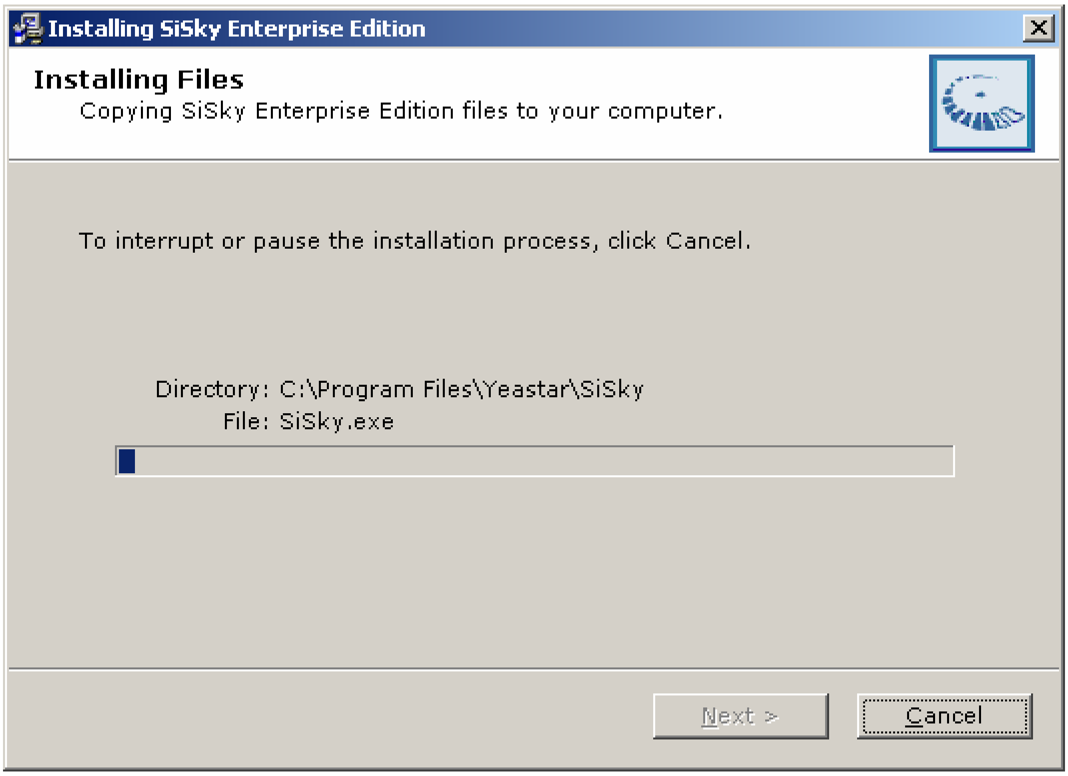
<Figure 4>

6. Enable the options by your own demands, and then click **Next**.



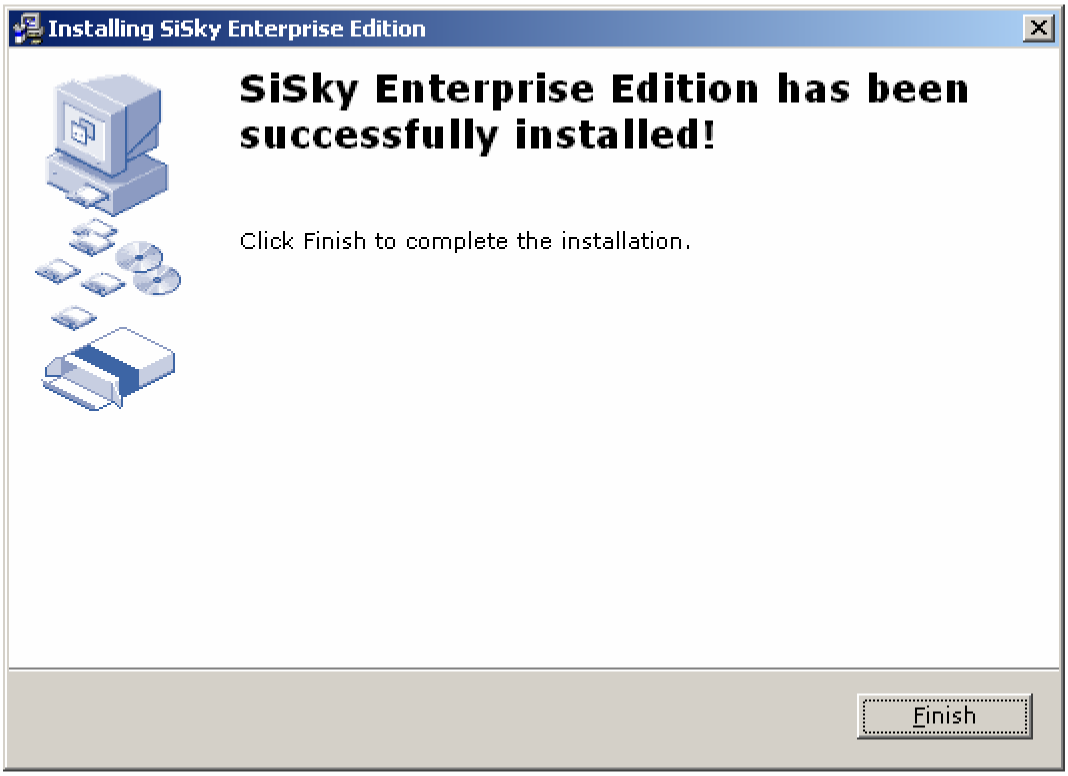
<Figure 5>

7. Enter into the **Installing Files**, system begin to configuration, which will last for a while.



<Figure 6>

9. When the installation is complete, a screen will pop-up to notify you that the software is installed successfully. Click **Finish**.



<Figure 7>

10. The final screen reminds you to restart computer in order to complete the installation. You would better to restart now.

< Figure 8>

11. Launch SiSky to enter the next chapter.

Note: Before installing SiSky software, you should first **uninstall** Skype software on your computer if you already have Skype software installed.

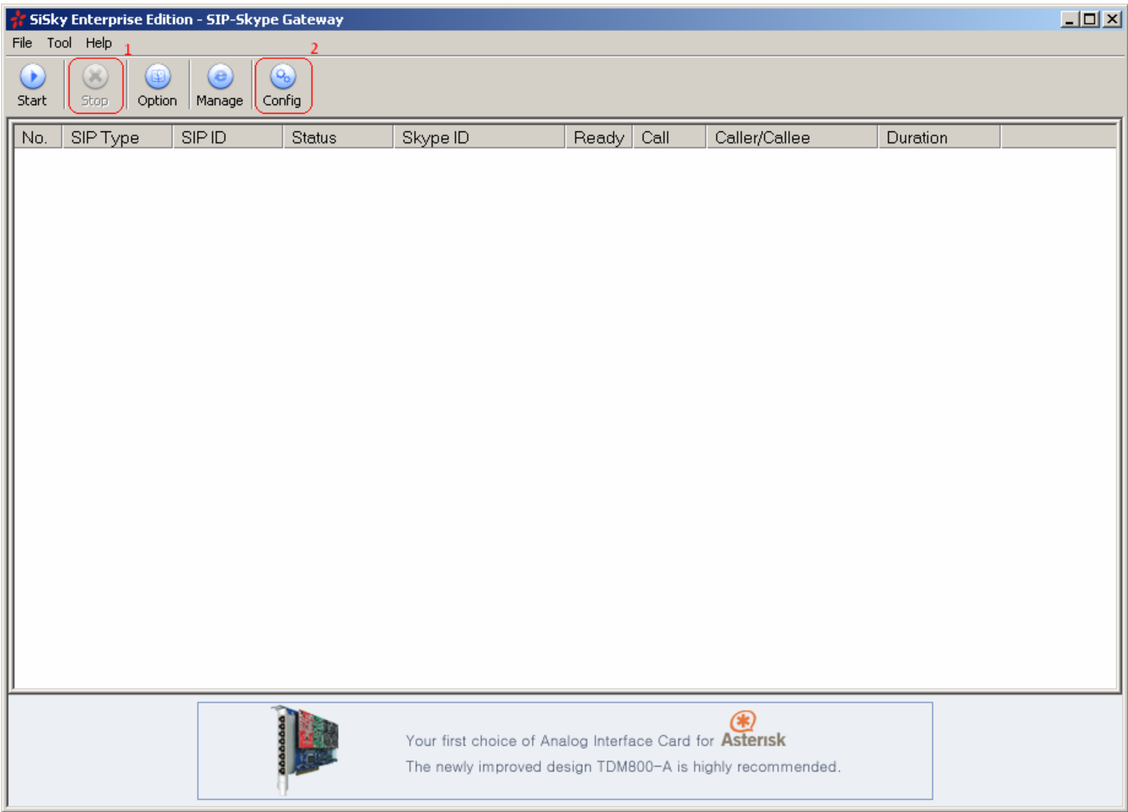
Running SiSky Config.Wizard 5

Configure SiSky through Wizard.

On initial use, a **Message** screen will pop-up and click 'Yes' to launch the Wizard.

< Figure 9>

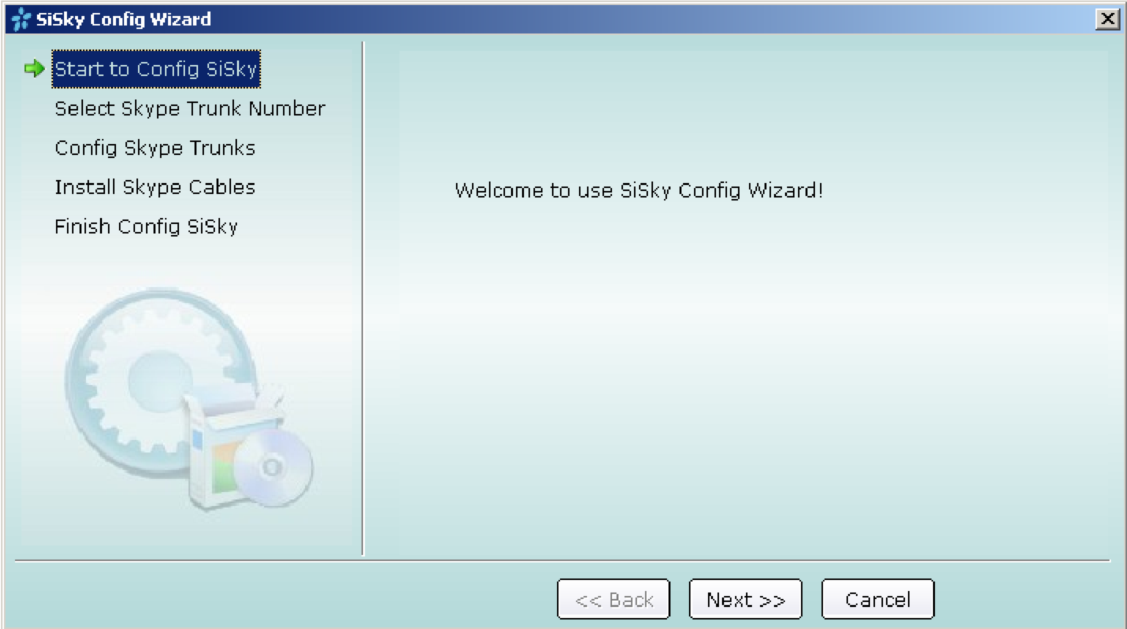
Or you can click the **Config** on SiSky to launch the Wizard. (If the **Config** button is invalid, please click **Stop** to stop SiSky first)



<Figure 10>

5.1 Launch the Config Wizard

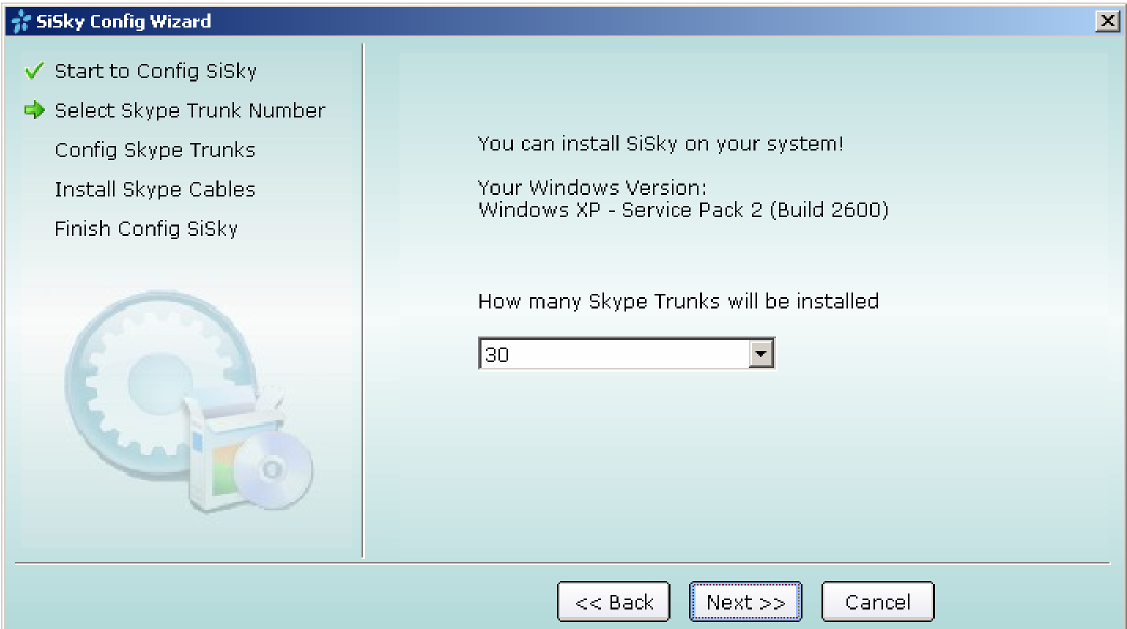
Click **Next**



<Figure 11>

5.2 Select Skype Trunk Number

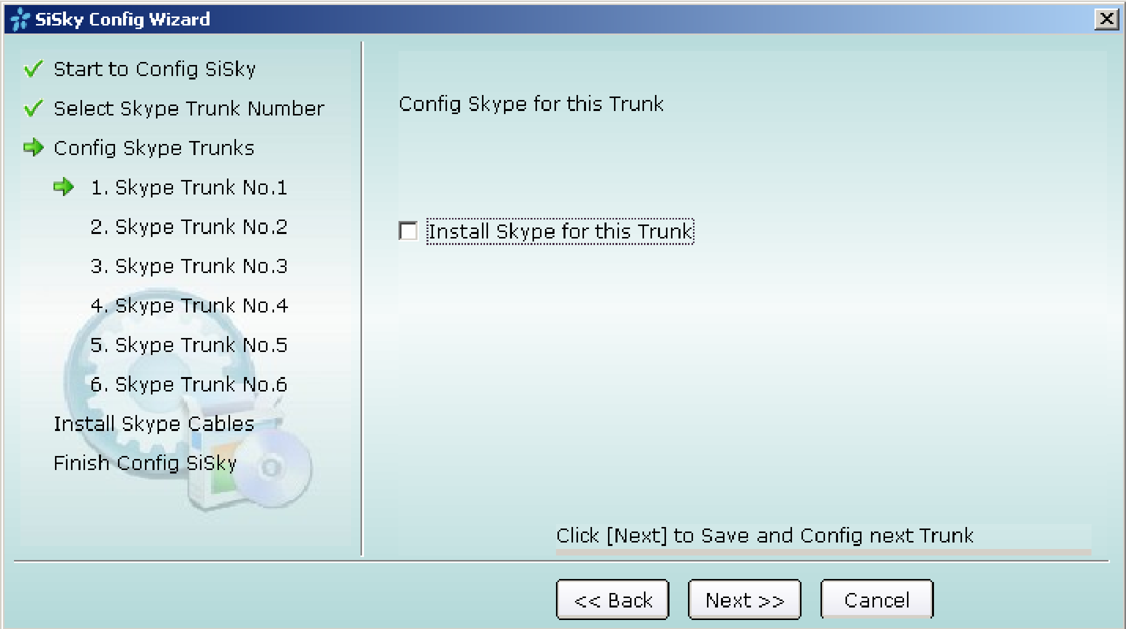
Select the **Skype Trunk Number** and then click **Next**



<Figure 12>

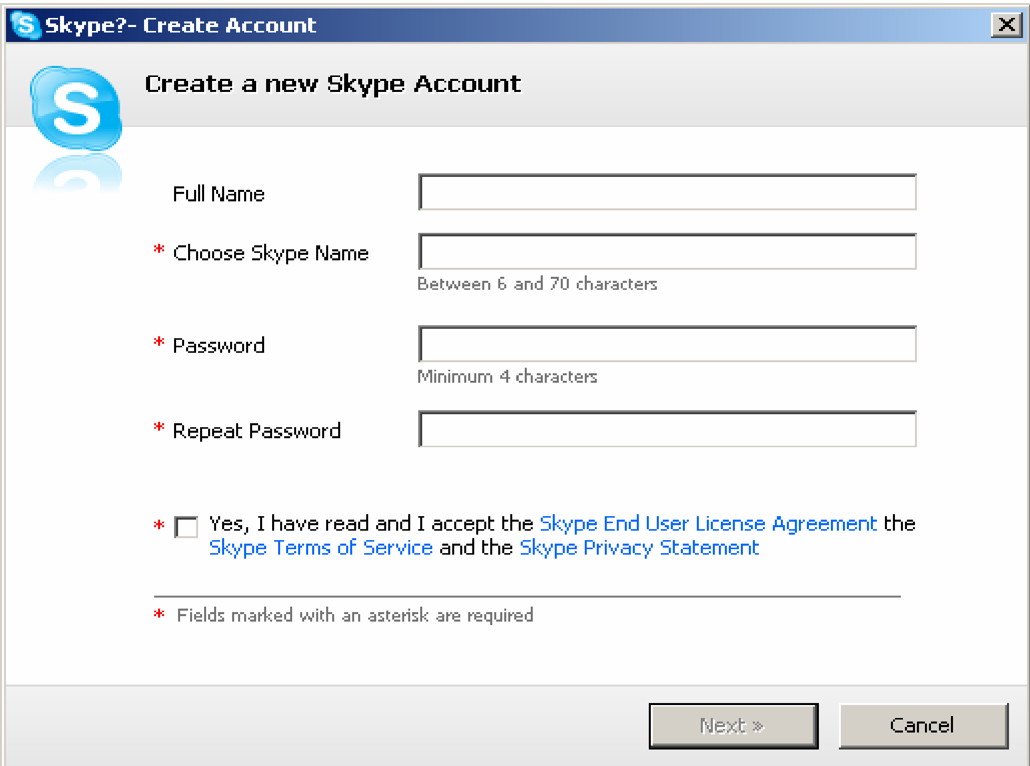
5.3 Configure Skype for Each Port

Enable **Install Skype for this port** and Skype will launch.



< Fgiure 13>

A **Skype?—Creat Account** will appear. Create a new account (see Figure 14) or Cancel it and log in by using an existing Skype account.



<Figure 14>

Enable the **Sign me in when Skype Starts** and wait for Skype to log you in.

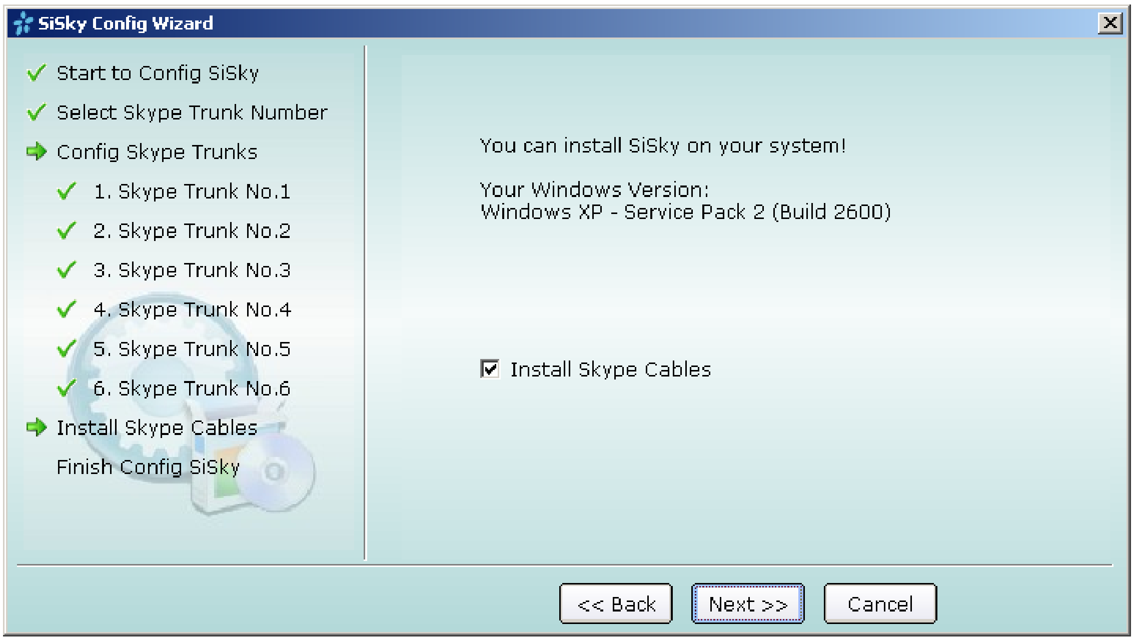
<Figure 15>

When SiSky get the Skype Name automatically, the configuration of Port 1 is finished. Click **Next** to configure other ports by the same way.

Note: After these steps are complete, repeat step 6.3 to configure the remaining ports and their Skype accounts. When the remaining ports are congifured, there will be a green tick before every port. See Figure 16

5.4 Install Skype Cables

Enable **Install Skype Cables** and click **Next** to continue.



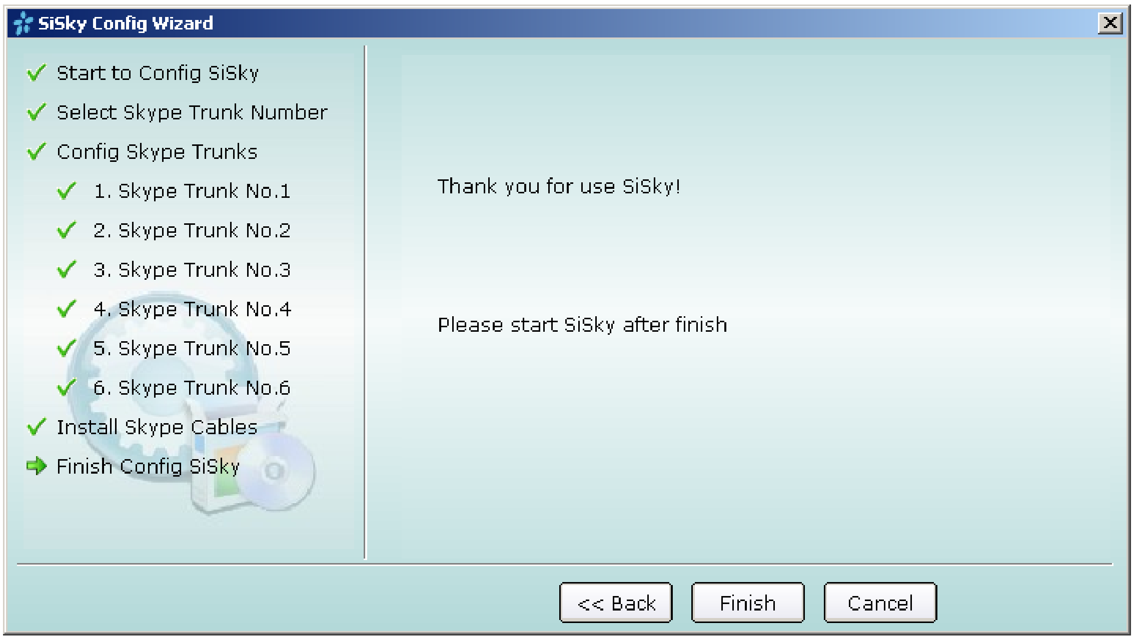
<Figure 16>

The following window maybe will appear during the system installation. Click **Continue Anyway**

<Figure 17>

5.5 Finish Config. Wizard

Select the country you are living, and then click 'Finish'.

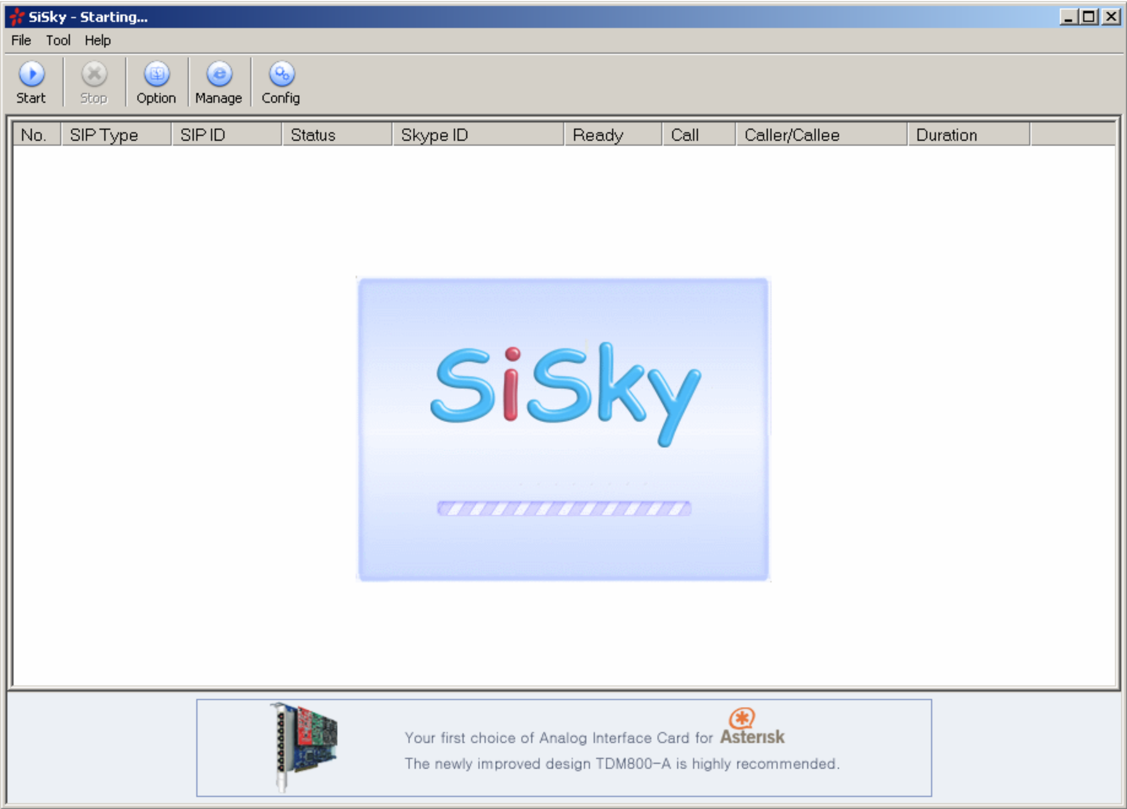


<Figure 18>

SiSky system configuration is complete!

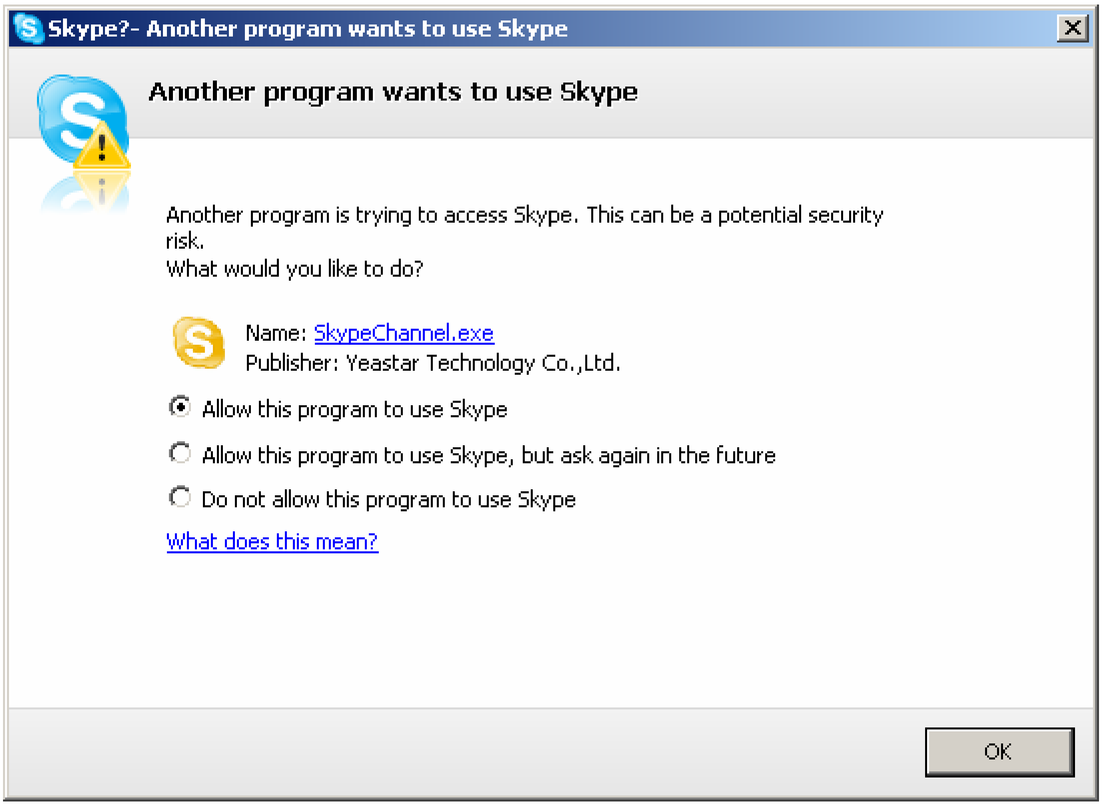
Running SiSky Software 6

Step 1 Double-click the shortcut on desk to run the software



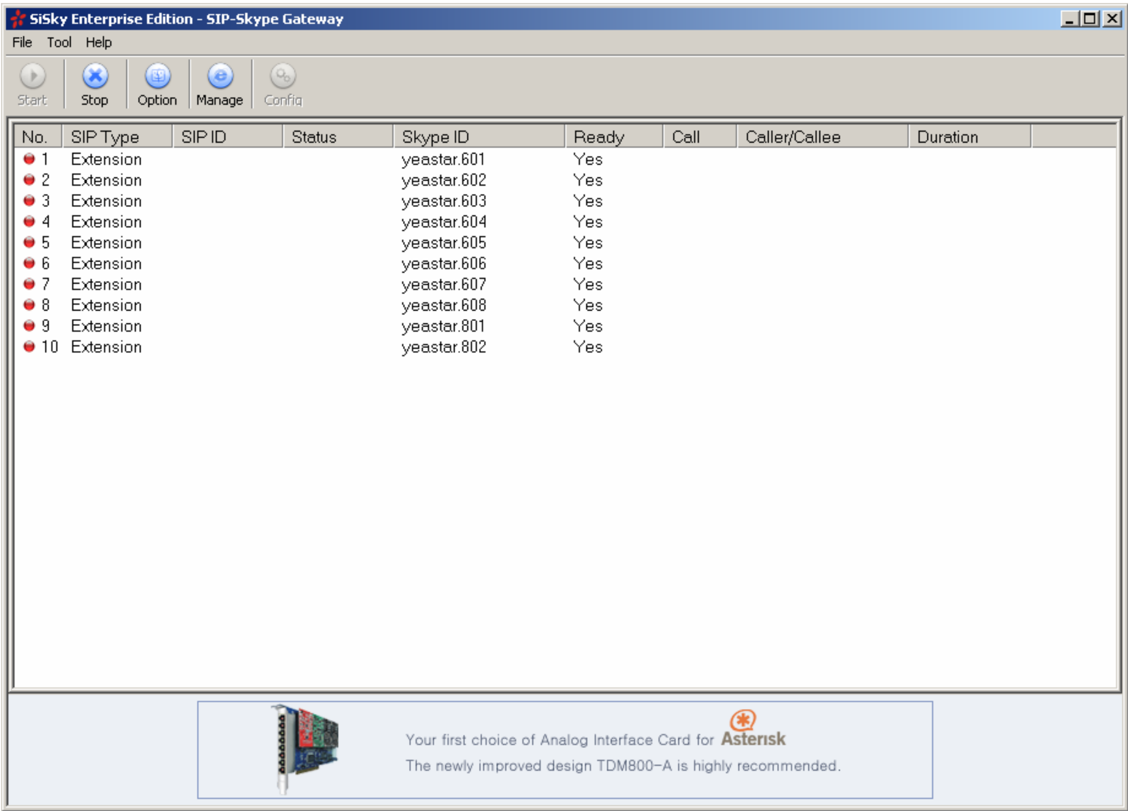
<Figure 19>

Step 2 Wait for all Skype IDs to log in. **Another program wants to use Skype** screen will come up. Click on the circle next to the first option, Allow this program to use Skype, and click **OK** to save.



<Figure 20>

Step 3 Check if the status of all ports and Skype are absolutely normal



<Figure 21>

Description of all SiSky items

No. :

1) : indicates failed initialization. The port's SIP or Skype is not working properly.
2) : indicates succeeded initialization. The port line is idle.
3) : indicates succeeded initialization. The port is in service.

SIP Type:

1) Extension: This port work as Asterisk'/IPPBX's SIP Extension
2) Trunk: This port work as Asterisk'/IPPBX's SIP Trunk

SIP ID:

SIP Trunk ID or SIP Extension ID for this port.

Status:

1) (NULL): Unsetup.
2) Registering: This port works as Asterisk'/IPPBX's SIP Extension. It is trying to register in Asterisk/IPPBX.
3) Registered: This port works as Asterisk'/IPPBX's SIP Extension. It has registered in Asterisk/IPPBX.
4) Waiting: This port works as Asterisk'/IPPBX's SIP Trunk. It is waiting for the register from Asterisk/IPPBX.
5) Connected: This port works as Asterisk'/IPPBX's SIP Trunk. It has accepted the register from Asterisk/IPPBX.

Skype ID:

Skype ID corresponds to the port number.

Ready:

1) N/A: This port has no configured Skype yet
2) Yes: The Skype is ready to use
3) No: The Skype is unready, maybe is logging in or offline

Call:

1) In：Call in
2) Out：Call out

Caller / Callee:

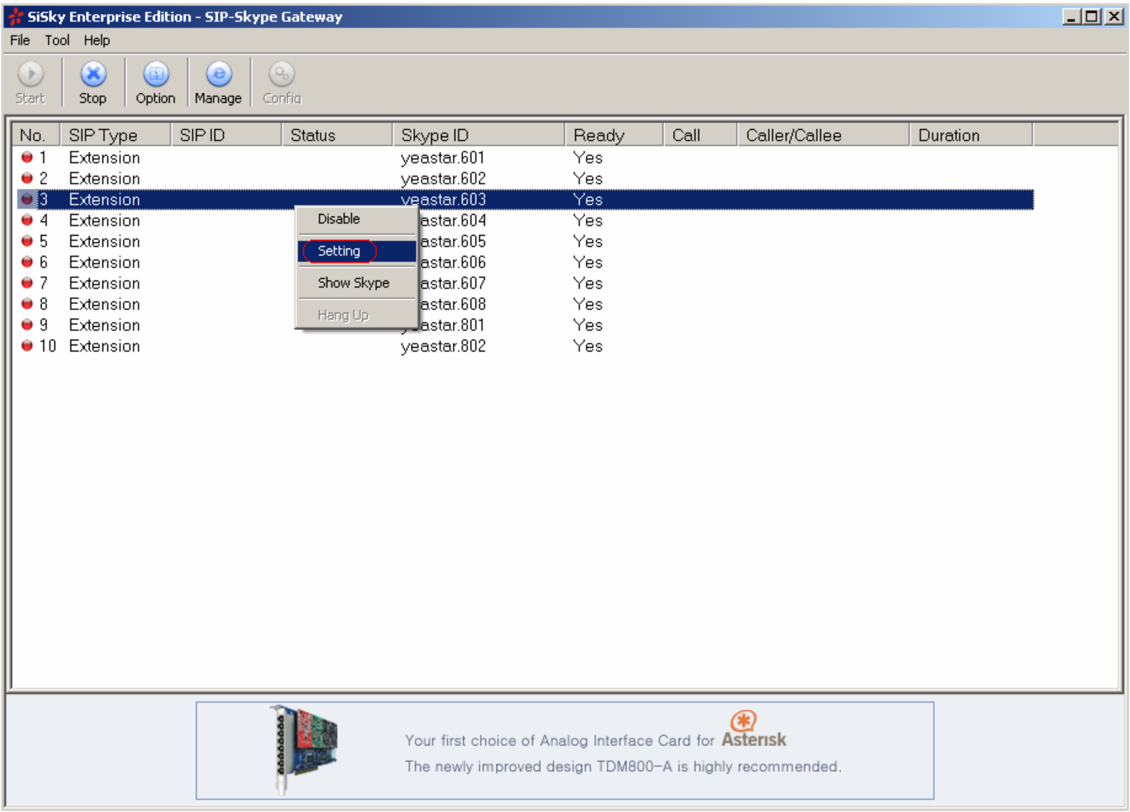
The telephone number or Skype ID of the other side

Duration:

The duration of speaking

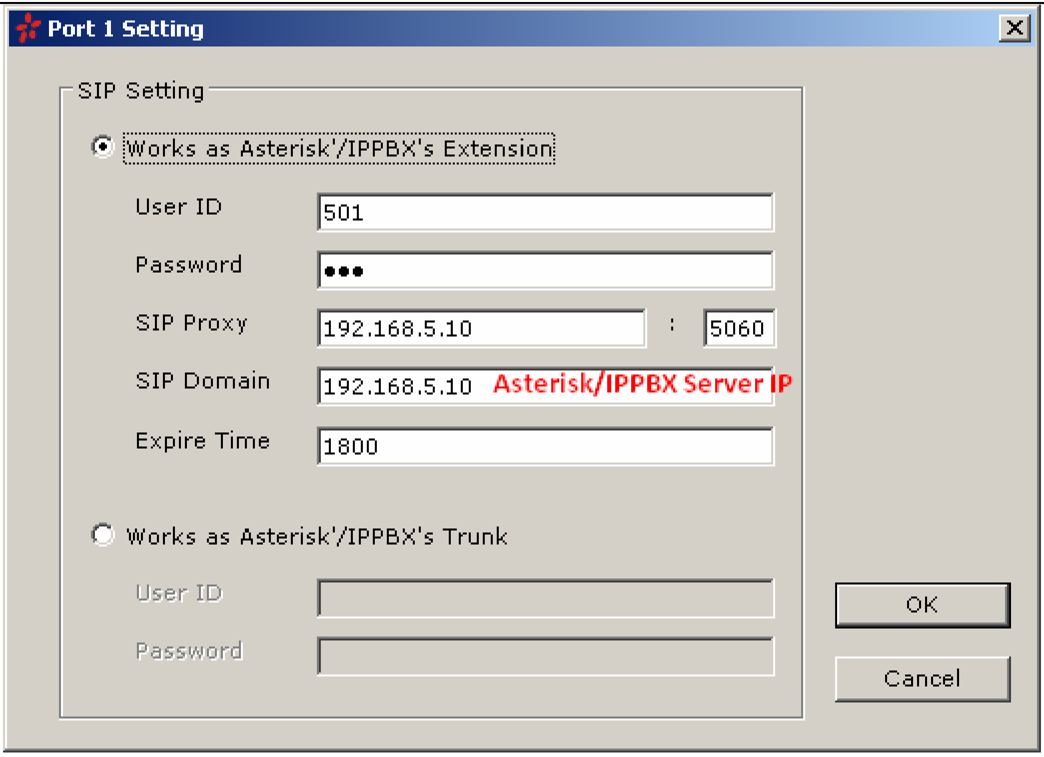
7 Building SiSky Operating Environment

7.1 Port Settings



< Figure 22>

Double click on the port or Click the 'Setting' on the popup menu to make configuration for this port.



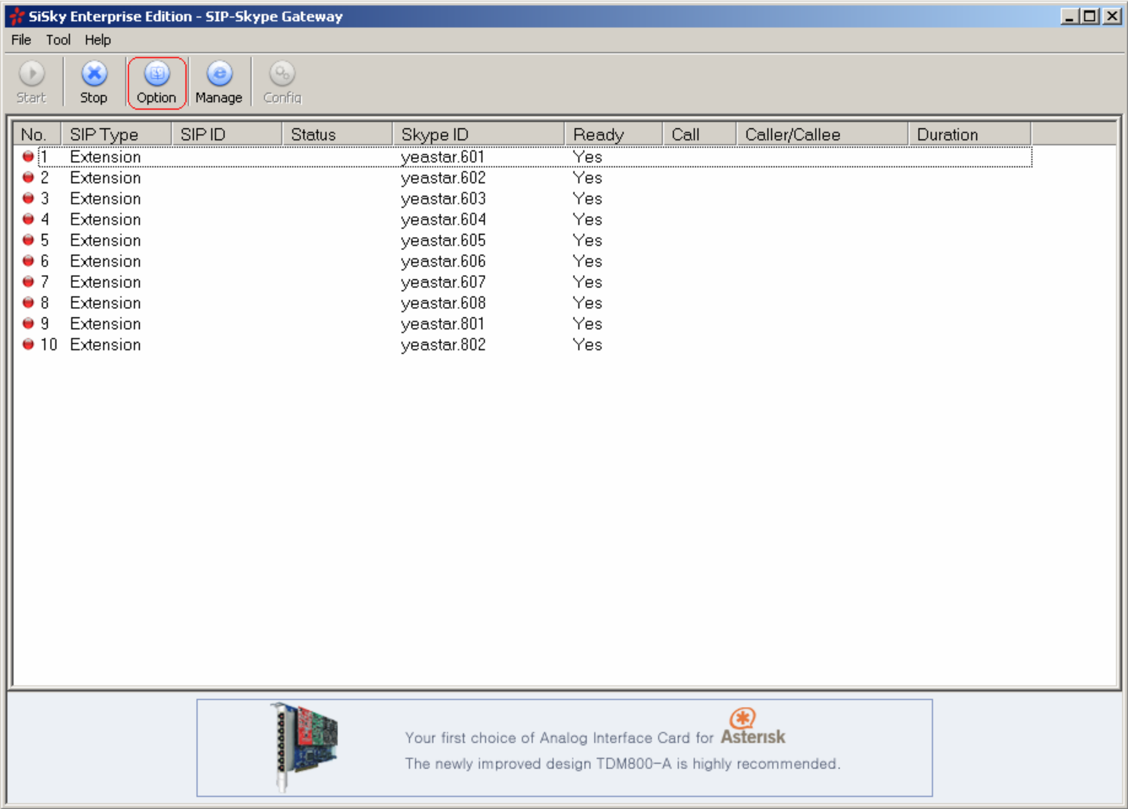
< Figure 23>

SIP Setting:

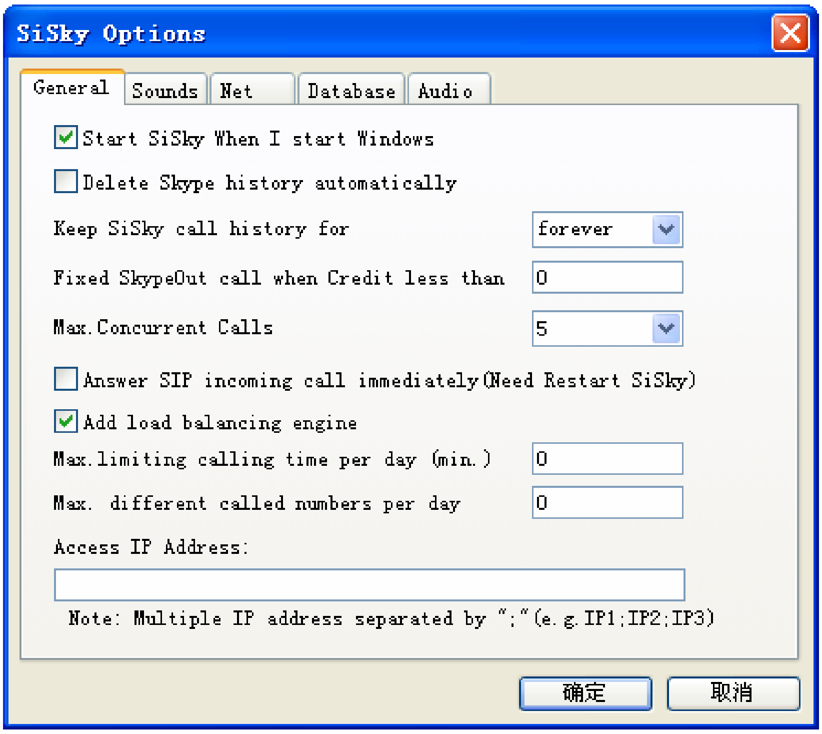
Firstly choose the working mode for this port. If enable *Work as Asterisk'/IPPBX's Extension*, please enter the SIP Extension information and configure the corresponding SIP Extension on Asterisk/IPPBX. If enable *Work as Asterisk'/IPPBX's Trunk*, please configure the SIP Trunk information and configure the SIP Trunk on Asterisk/IPPBX.

7.2 General Setting

Click the **Option** tab in SiSky interface to set the other operating environment.



< Figure 24>



<Figure 25>

1 Start SiSky when I start Windows

2 Delete Skype history automatically

Skype history soon accumulate as time goes on, which will need more memeory and affect the system performance. If enable this option, system will clean all Skype history automatically when starting SiSky.

3 Keep SiSky call histroy

4 Fixed SkypeOut call when Credit less than the appointed amount

If this port works as SkypeOut Trunk, Skype ID on this port must have Credit. When credit less than an appointed amount, caller will hear a voice prompt.

Please refer to *chapter 7.3.6*

5 Max.Current Calls

For example, you configured 8 ports on SiSky, and your SiSky Server is Celeron CPU, from the *chapter 1.4* we can know that Celeron CPU generally supports 3 Skype concurrent calls, so when 3 of the 8 Skype trunks are in calls, the forth call through skype will influence the communication quality. In this situation, you should set the Max.Currnet Calls as 3 in order to ensure the excellent voice quality and the forth Skype call will be rejected directly.

6 Answer SIP incoming call immediately

When SIP call reaches SiSky, whether or not respond immediately.
Don't enable: SiSky will not respond the SIP call unless Skype answers. (SIP client will start the time calculation when Skype answers)
Enable: SiSky will answer the calls from SIP immediately. (Start the time calculation when SIP client connects Skype.)

7 Add load balancing engine

After enabled this function, SiSky will balance the flow rate among Skype IDs of all ports.
For example, a user configured 8 trunks on SiSky, totally 8 Skype IDs. When a SIP user calls to the first trunk on SiSky, it will be out from the Skype ID with the shortest speaking time.

8 Skype Rules Settings

Note: This setting aims to the Skype Unlimited World service specially, if you are using other Skype Unlimited Call services, it's not necessary to configure the following settings.
Because of the fair usage policy that calls to phones and mobiles and Skype To Go* calls are included in users' subscription subject to a fair usage limit of 10,000 minutes per user per month, with a maximum of 6 hours per day. Also, no more than 50 different numbers in total can be called per day, here two options are available on SiSky for users to set up in order to apply with Skype rules.

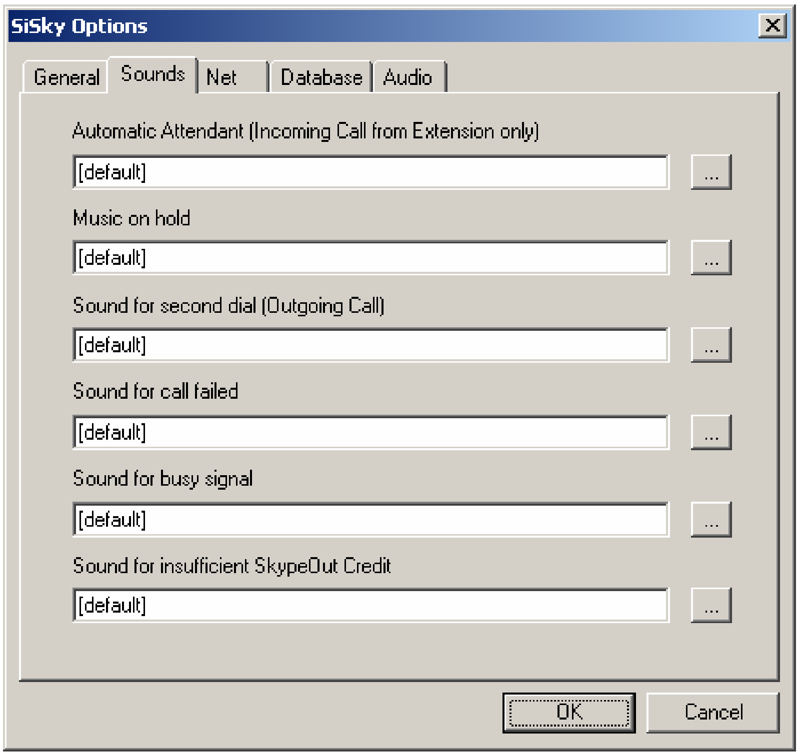
1) Maximum Skype call time in total per day (minutes)
Default value is 0 that stands for unlimited. The suggested value is equal to or less than 300 minutes.

2) Maximum different numbers in total per day
Default value is 0 that stands for unlimited. The suggested value is equal to or no more than 50.

9 Access IP Address

This is setting only work with P2P module.
Note: Multiple IP address separated by ";" (e.g. 192.168.5.7; 192.168.5.8)

7.3 Sounds Setting



<Figure 26>

7.3.1 Automatic Attendant

When there is a Skype incoming call from Extension port, SiSky will display the auto attendant and guide the caller to the dedicated extensions.
SiSky has default auto attendant. If you want to customize it, please make reference to **Appendix A**.

7.3.2 Music on Hold

When there is a Skype incoming call from Trunk port, SiSky will ring the trunk normally and display the music on hold until the trunk picked up.

SiSky has default music. If you want to customize it, please use the tool (Cool Edit) to transfer the music format into required WAV format (PCM, 8000Hz, 16Bit, Mono).

7.3.3 Sound for Second Dial

If you make an Outgoing call from Extension port, after you only dialing Extension number to reach this port in first time, you will hear this sound that guide you to continue dialing number. SiSky has default sound – Dial Tone. If you want to customize it, please make reference to **Appendix A**.

7.3.4 Sound for call failed

Caller will hear this prompt sound when incoming or outgoing call is failed.

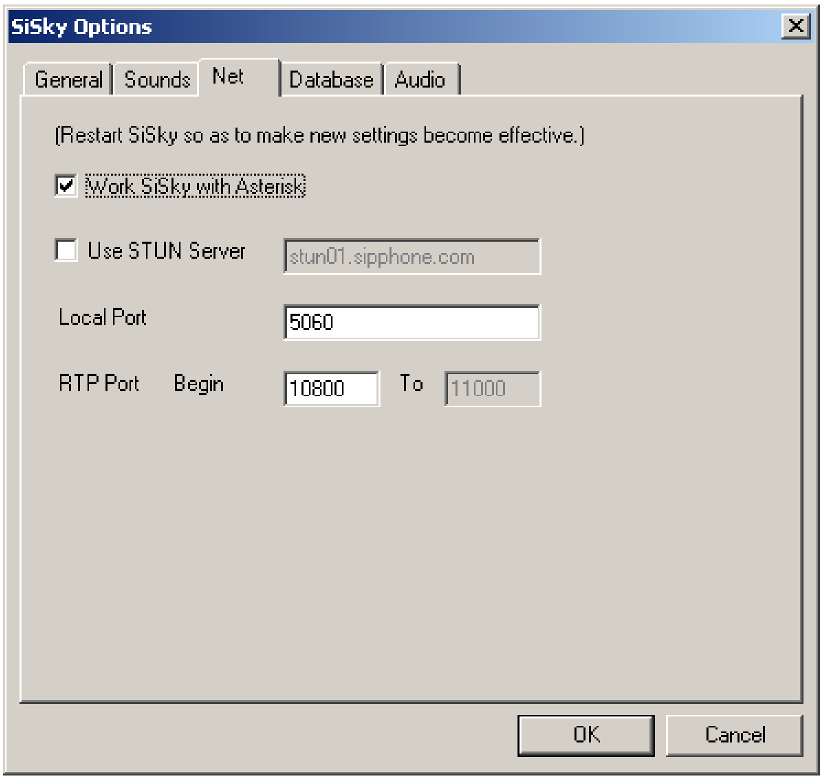
7.3.5 Sound for busy signal

Caller will hear this prompt sound when callee is busy.

7.3.6 Sound for insufficient SkypeOut Credit

Caller will hear this prompt voice when making SkypeOut Call but the Credit less than an appointed amount. Please refer to *chapter 7.2.3*

7.4 Net Setting

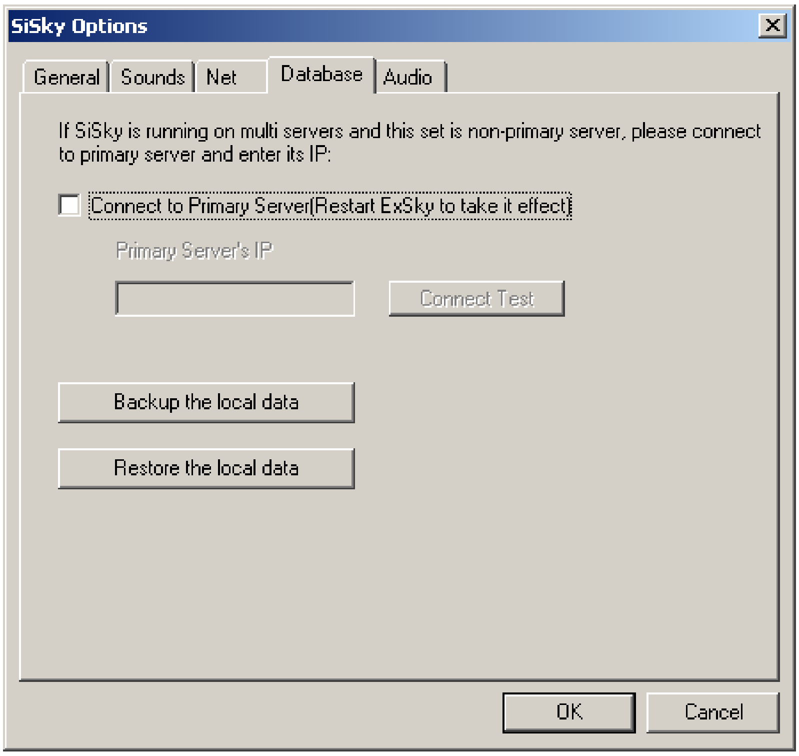


< Figure 27 >

If the IPPBX that SiSky works with is based on Asterisk, enable it; if not, disenable it.
Use STUN Server for SIP Extensions.

Local Port and RTP Port are for SIP. Please don’t change it generally.

7.5 Database Setting

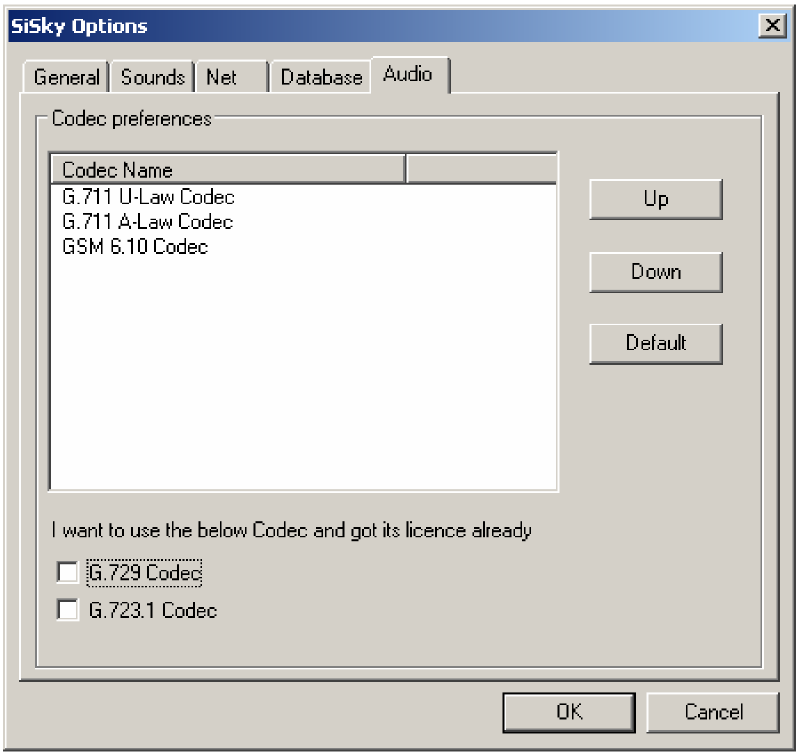


< Figure 28>

1) If you have no need of cascade connecting multiple SiSky Servers, please don't enable **Connect to Primary Server**.

2) Backup and Restore the data in this PC.

7.6 Audio Setting



< Figure 29>

For the compatibility with your IP PBX, we choose those kinds of codec and make the priority as in Figure 29. If you are using Asterisk based IP PBX, the default Codec can meet the requirements well. If your IP PBX insists on G.729 or G.723.1, please make sure you have got the license before using it.

Configure Asterisk for SiSky 8

If your IPPBX is basd on Asterisk, please refer to this chapter to configure Asterisk for SiSky.

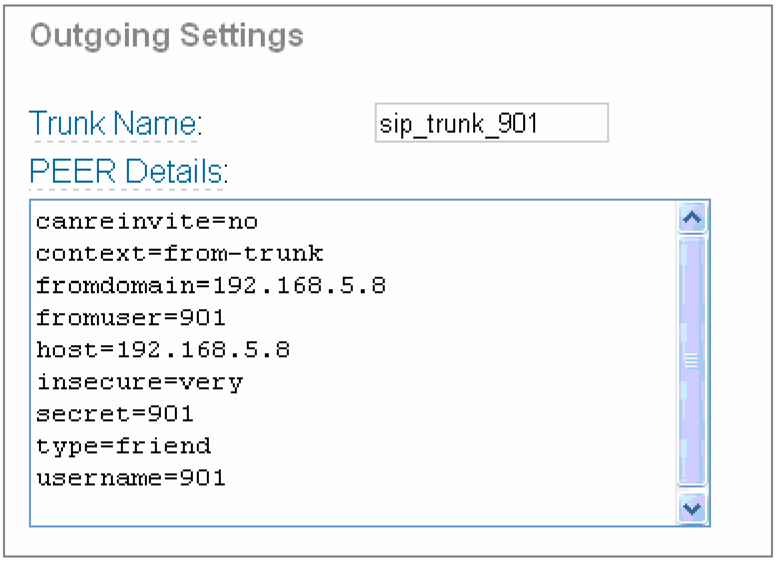
If your Asterisk is freePBX installed, please refer to *chapter 8.1* and *8.3*; or *chapter 8.2* and *8.4*.

8.1 How to Create SIP Trunk for SiSky with freePBX (Asterisk+freePBX or Trixbox)

8.1.1 Add SIP Trunk

Open freePBX —> Setup —> Trunks —> Add SIP Trunk

8.1.1.a Make Outgoing Settings as shown on Figure 30



<Figure 30>

Note: 1. *Trunk Name*: You can write willfully except empty it.

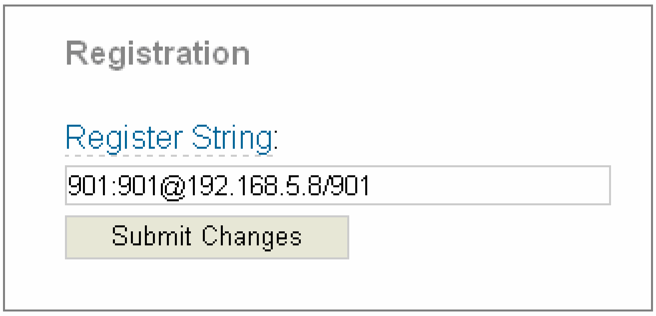
2. *fromdomian* and *host*: You must write the IP address (or domain) of SiSky Server

3. *fromuser* and *username*: They must be sip id.

4. *secret*: It's the password of sip id

5. Please keep the rest entries no change.

8.1.1.b Registration configuration as:
<sip id>:<secret>@<sip sever ip>/<sip id>
(in which ***sip sever ip*** can be domain name) as shown on Figure 31:



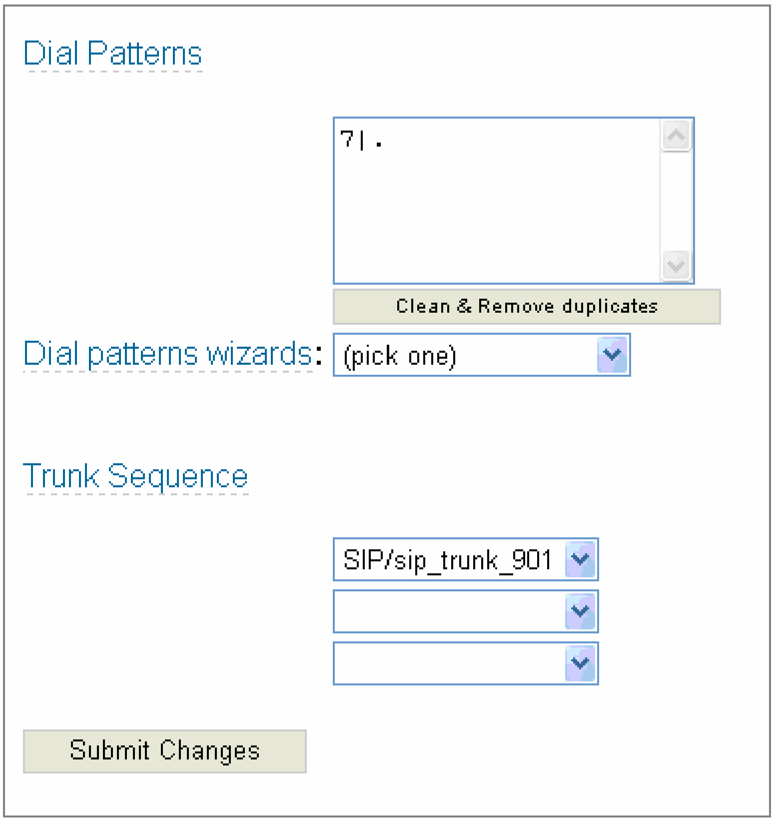
<Figure 31>

On the above example, SIP ID=901, Password=901, SiSky Server IP = 192.168.5.8

8.1.1.c Click 'Submit Changes', and then click 'Apply Configuration Changes' tab.

8.1.2 Configure Outbound Routes

Click Outbound Routes-- Add Route



<Figure 32>

Note: 1. the prefix is 7 digits; you can change it as your own will.
2. If you set multiple sip trunks on Trunk Sequence, and asterisk can't find the idle trunk when you dialing out, you need to modify the *extension.conf* file.

Open extensions.conf, please delete the two lines on [macro-dialout-trunk]:
exten => s-BUSY,1,NoOp(Dial failed due to trunk reporting BUSY –giving up) and

exten => s-BUSY,2,Busy(20) See Figure 33

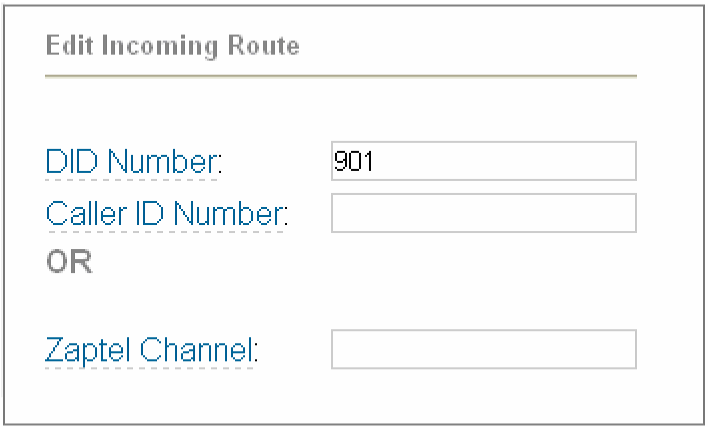
```
exten => s,n(chanfull),Noop(max channels used up)

;exten => s-BUSY,1,NoOp(Dial failed due to trunk reporting BUSY - giving up)
;exten => s-BUSY,2,Busy(20)
```

< Figure 33 >

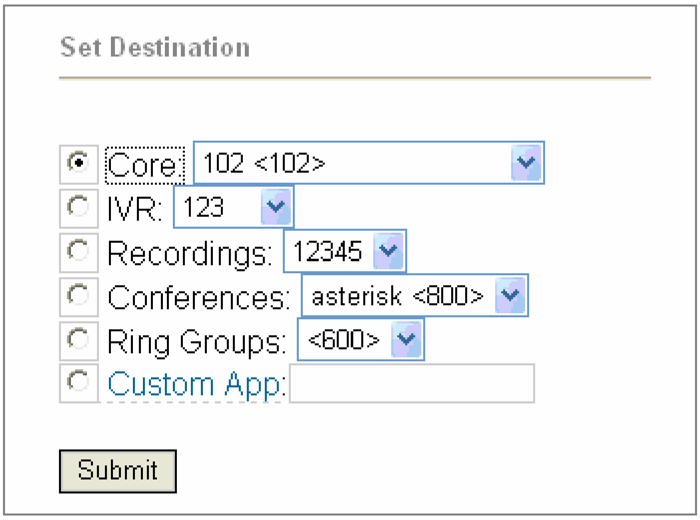
Click 'Submit Changes' and then click the tab 'Apply Configuration Change' on upper of the page

8.1.3 Configure Inbound Routes

Click Inbound Routes—Add Incoming Route



<Figure 34>



<Figure 35>

Note: 1. DID Number: Enter the sip id that you wrote on sip trunk.

2. Set Destination: Choose the routing destination. Here we set it to route 102 extension, you can choose IVR as well if you want.

Click 'Submit Changes' and then click the tab 'Apply Configuration Change' on upper of the page

You have finished the configuration of SIP Trunk 901. Please configure other SIP

Trunk by the same way. The SIP ID and Password of SIP Trunk should be as same as them on SiSky.

8.2 How to Create SIP Trunk for SiSky without freePBX

Note:

a. If you have multiple SIP ID, then you can create multiple SIP TRUNK. Here for example, we use two SIP IDs (901 and 902) to create two SIP Trunks.

b. You should replace SIP ID 901 and 902 with your own SIP ID. And change the IP address 192.168.5.35 on file to your own SiSky Sever address.

8.2.1 Add SIP Trunk

Enter into Asterisk Configuration Catalog, and open *sip.conf* to configure sip trunk

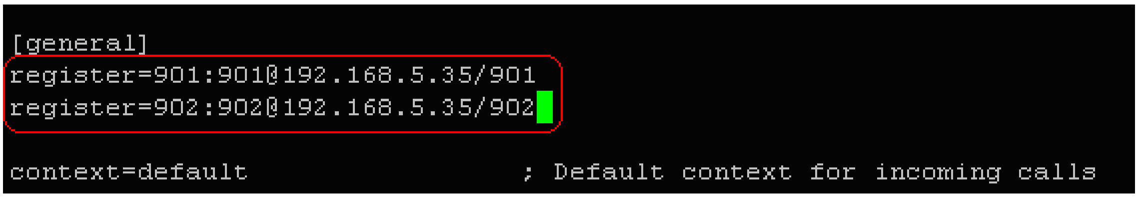
[root@linux ~]# cd /etc/asterisk

[root@linux asterisk]# vi sip.conf

8.2.1.a On [general] you should register your own SIP ID.

register=<SIP ID>**:**<SECRET>**@**<SIP SEVER IP> **/** <SIP ID>

On Figure 36 we register two SIP IDs.



<Figure 36>

You need to replace 901,902 and 192.168.5.35 with your own SIP IDs and SiSky Server IP.

8.2.1.b On file tail part, SIP ID configuration:

```
[sip_trunk_SIP ID]
username=SIP ID
type=friend
secret=SIP ID SECRET
nat=yes
insecure=very
host=SEVER IP ADRESS
fromuser=SIP ID
fromdomain= SEVER IP ADRESS
dtmfmode=rfc2833
context=from-trunk
canreinvite=no
qualify=yes
disallow=all
allow=ulaw
allow=alaw
```

allow=gsm
allow=g729
allow=ilbc

You could add coding according to your own requires. The format is: allow=coding.
On our example, the SIP IDs are 901 and 902, host IP is 192.168.5.35(SiSky Server IP), so we fill in as the Figure 37:

```
[sip_trunk_901]
username=901
type=friend
secret=901
nat=yes
insecure=very
host=192.168.5.35
fromuser=901
fromdomain=192.168.5.35
host=192.168.5.35
dtmfmode=rfc2833
context=from-trunk
canreinvite=yes
qualify=yes
disallow=all
allow=gsm
allow=ulaw
allow=alaw
allow=g729
allow=ilbc

[sip_trunk_902]
username=902
type=friend
secret=902
nat=yes
insecure=very
host=192.168.5.35
fromuser=902
fromdomain=192.168.5.35
host=192.168.5.35
dtmfmode=rfc2833
context=from-trunk
canreinvite=yes
qualify=yes
disallow=all
allow=gsm
allow=ulaw
allow=alaw
allow=g729
allow=ilbc
```

<Figure 37>

8.2.1.c Check whether your sip trunk is registered successfully.
If asterisk unstarted, please start it and executive command on asterisk console:
sip show registry
If asterisk started, please restart it and executive command on asterisk console:
sip show registry
You will see the output similar as Figure 38

```
linux*CLI> sip show registry
Host                    Username      Refresh State           Reg.Time
192.168.5.35:5060       902               105 Registered      Fri, 14 Dec 2007 17:23:11
192.168.5.35:5060       901               105 Registered      Fri, 14 Dec 2007 17:23:11
linux*CLI>
```

<Figure 38>

If the state is not registered, you have to check the information on Figure 37.

8.2.2. Configure Outbound Routes

It needs extension phone for dialing out. Here we use SIP extension, you can use zap, iax2 as well.

```
[109]
type=friend
secret=109
record_out=Adhoc
record_in=Adhoc
qualify=yes
port=5060
nat=yes
mailbox=109@device
host=dynamic
dtmfmode=rfc2833
dial=SIP/109
context=from-internal
canreinvite=no
callerid=device <109>
disallow=all
allow=ulaw
allow=alaw
allow=g729
allow=g723.1
```

<Figure 39>

Note:

a. The value of context is *from-internal*, so on extensions.conf file, it must have the dial rule of this context.

b. Here we set that if dial the numbers begin by 7 should use this two sip trunks. You can replace the prefix 7 according to your own requires.

On extensions.conf file, we make configuration as Figure 40

```
[from-internal]
exten => _7.,1,Dial(SIP/${EXTEN:1}@sip_trunk_901,30,r)
exten => _7.,n,Dial(SIP/${EXTEN:1}@sip_trunk_902,30,r)
exten => _7.,n,Hangup
~
-- INSERT --
```

<Figure 40>

Note:

a. Change the from-internal into the context value when you configuring extension

b. Replace 7 with the prefix you want to dial

c. Replace sip_trunk_901 and sip_trunk_902 to sip_trunk_SIP ID you want

d. The ringing time is 30 seconds, you can modify it if you want

e. If you want to add sip trunk, please make it above **exten => _7.,n,Hangup()**

For example, we want to add sip_trunk_903, as Figue 41

```
[from-internal]
exten => _7.,1,Dial(SIP/${EXTEN:1}@sip_trunk_901,30,r)
exten => _7.,n,Dial(SIP/${EXTEN:1}@sip_trunk_902,30,r)
exten => _7.,n,Dial(SIP/${EXTEN:1}@sip_trunk_903,30,r)
exten => _7.,n,Hangup
-- INSERT --
```

<Figure 41>

8.2.3 Configure Inbound Routes

Here you can configure the detination of inbound routes. For example, we congifure the calls from trunk 901 to 5001 extension, and calls from trunk 902 to 5002 extension. Please configure them on extensions.conf file:

```
[from-trunk]
exten => 901,1,Dial(SIP/5001)
exten => 901,n,Hangup
exten => 902,1,Dial(SIP/5002)
exten => 902,n,Hangup
~
```

<Figure 42>

Note: You should change the 901 and 902 to the SIP ID when configuring SIP Trunk and replace the routes destination SIP/5001 and SIP/5002 as your own demands.

8.2.4 Restart Asterisk to finish configuration

8.3 How to Create SIP Extension for SiSky with freePBX (Asterisk+freePBX or Trixbox)

Following instructions are based on the example of 16 channels, Create 16 SIP account from 501 to 516 for SiSky.

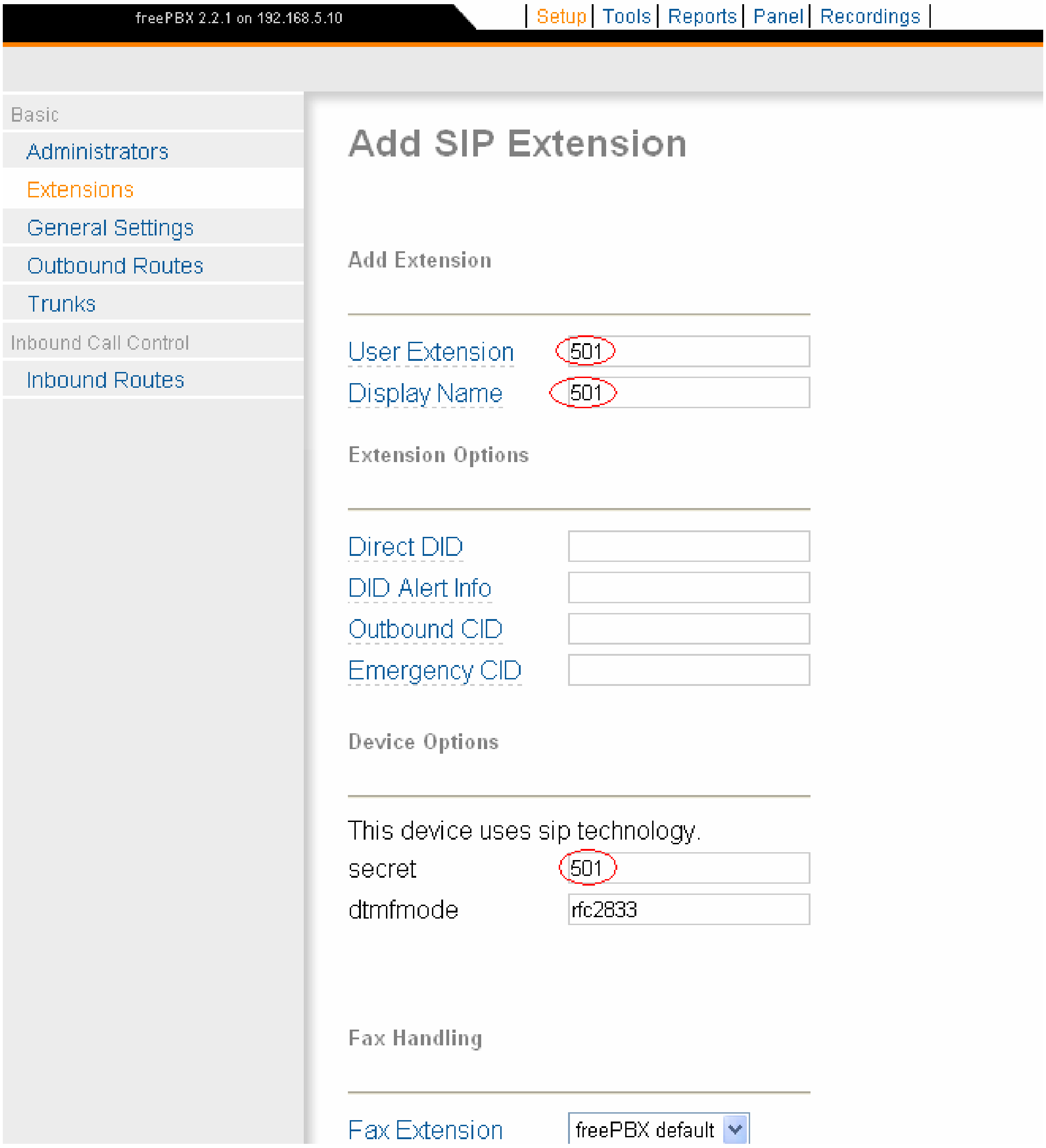
8.3.1 In extensions_custom.conf, add below lines into the [from-internal- custom] context:

exten => _50.,1,Dial(SIP/${EXTEN:0}@501)
exten => _501.,1,Dial(SIP/${EXTEN:0}@501)
exten => _502.,1,Dial(SIP/${EXTEN:0}@502)
exten => _503.,1,Dial(SIP/${EXTEN:0}@503)
exten => _504.,1,Dial(SIP/${EXTEN:0}@504)
exten => _505.,1,Dial(SIP/${EXTEN:0}@505)
exten => _506.,1,Dial(SIP/${EXTEN:0}@506)
exten => _507.,1,Dial(SIP/${EXTEN:0}@507)
exten => _508.,1,Dial(SIP/${EXTEN:0}@508)
exten => _509.,1,Dial(SIP/${EXTEN:0}@509)
exten => _510.,1,Dial(SIP/${EXTEN:0}@510)
exten => _511.,1,Dial(SIP/${EXTEN:0}@511)
exten => _512.,1,Dial(SIP/${EXTEN:0}@512)
exten => _513.,1,Dial(SIP/${EXTEN:0}@513)
exten => _514.,1,Dial(SIP/${EXTEN:0}@514)
exten => _515.,1,Dial(SIP/${EXTEN:0}@515)
exten => _516.,1,Dial(SIP/${EXTEN:0}@516)

Notes:
exten => _50.,1,Dial (SIP/${EXTEN:0}@501)
---that means if you dial **500**+Phone Number, SiSky will automatically find idle Skype trunk from 1 to 16 to dial out.
exten => _501.,1,Dial(SIP/${EXTEN:0}@501)
---that means dial out through Skype trunk 1.

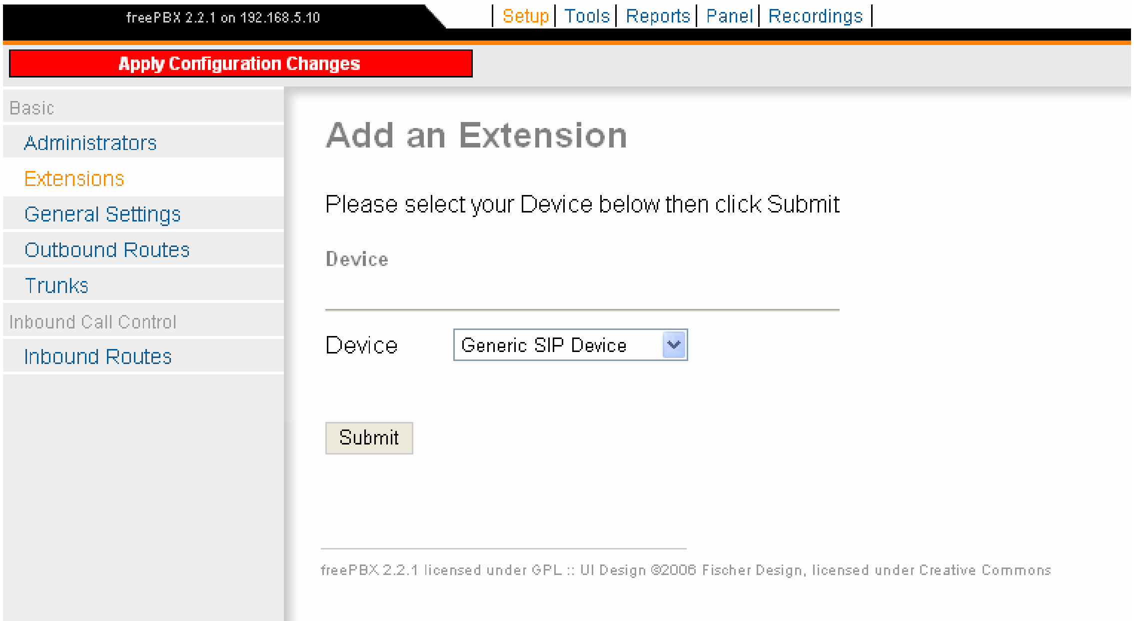
8.3.2 Set up SIP Extension in freePBX.

Set extension 501 as below:
"User Extension":501
"Display Name":501
"Secret":501
"dtmfmode":rfc2833



<Figure 43>

Apply Configuration Changes



<Figure 44>

Set other extensions as same steps as setting extension 501.

8.3.3 Apply Configuration Changes or restart Asterisk.

8.4 How to Create SIP Extension for SiSky without freePBX

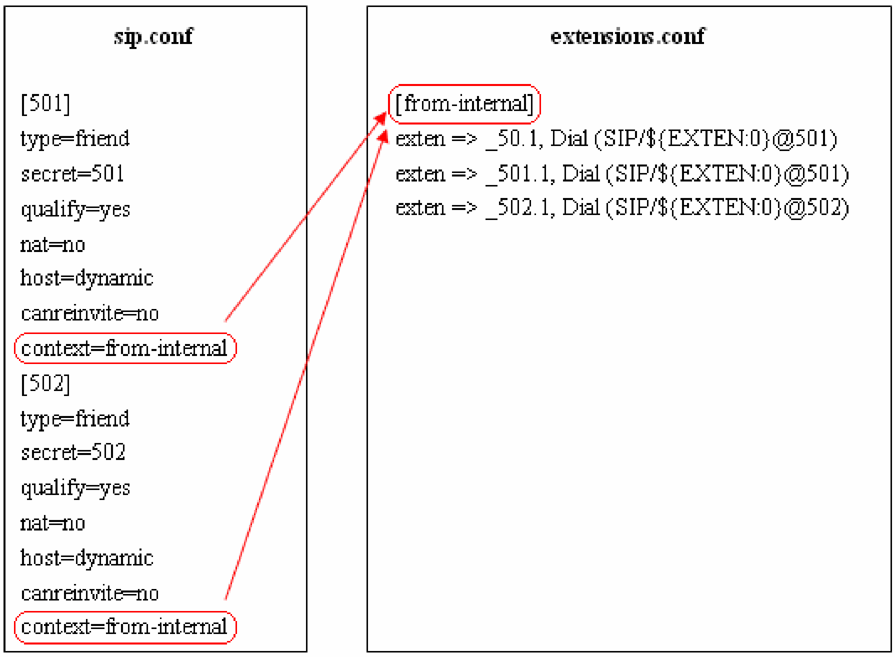
Following instructions are based on the example of 2 channels, Create SIP account from 501 and 502 for SiSky.

8.4.1 Define SIP Extensions in file sip.conf

for example:
[501]
type=friend
secret=501
qualify=yes
nat=no
host=dynamic
canreinvite=no
context=from-internal
Set other extensions as same as setting extension 501.

8.4.2 Add below lines into file extensions.conf:

[from-internal]
exten => _50.,1, Dial (SIP/${EXTEN:0}@501)
exten => _501.,1, Dial (SIP/${EXTEN:0}@501)
exten => _502.,1, Dial (SIP/${EXTEN:0}@502)



<Figure 45>

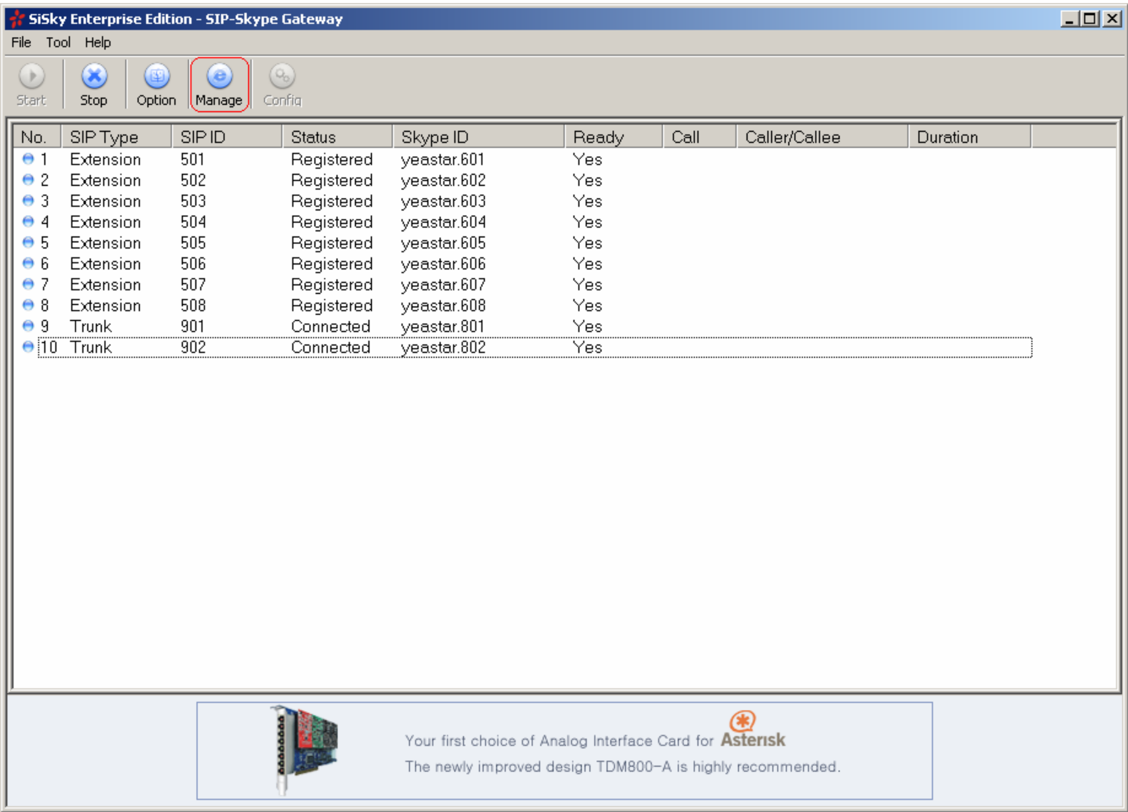
Notes: if you dial **500**+Phone Number, SiSky will automatically find idle Skype trunk from 1 to 16 to dial out.

8.4.3 Restart Asterisk

Managing SiSky Software 9

9.1 Open SiSky Homepage

If you are operating in SiSky Server, please click the **Manage** button or you can open the IE browser and enter in: http://127.0.0.1:8080

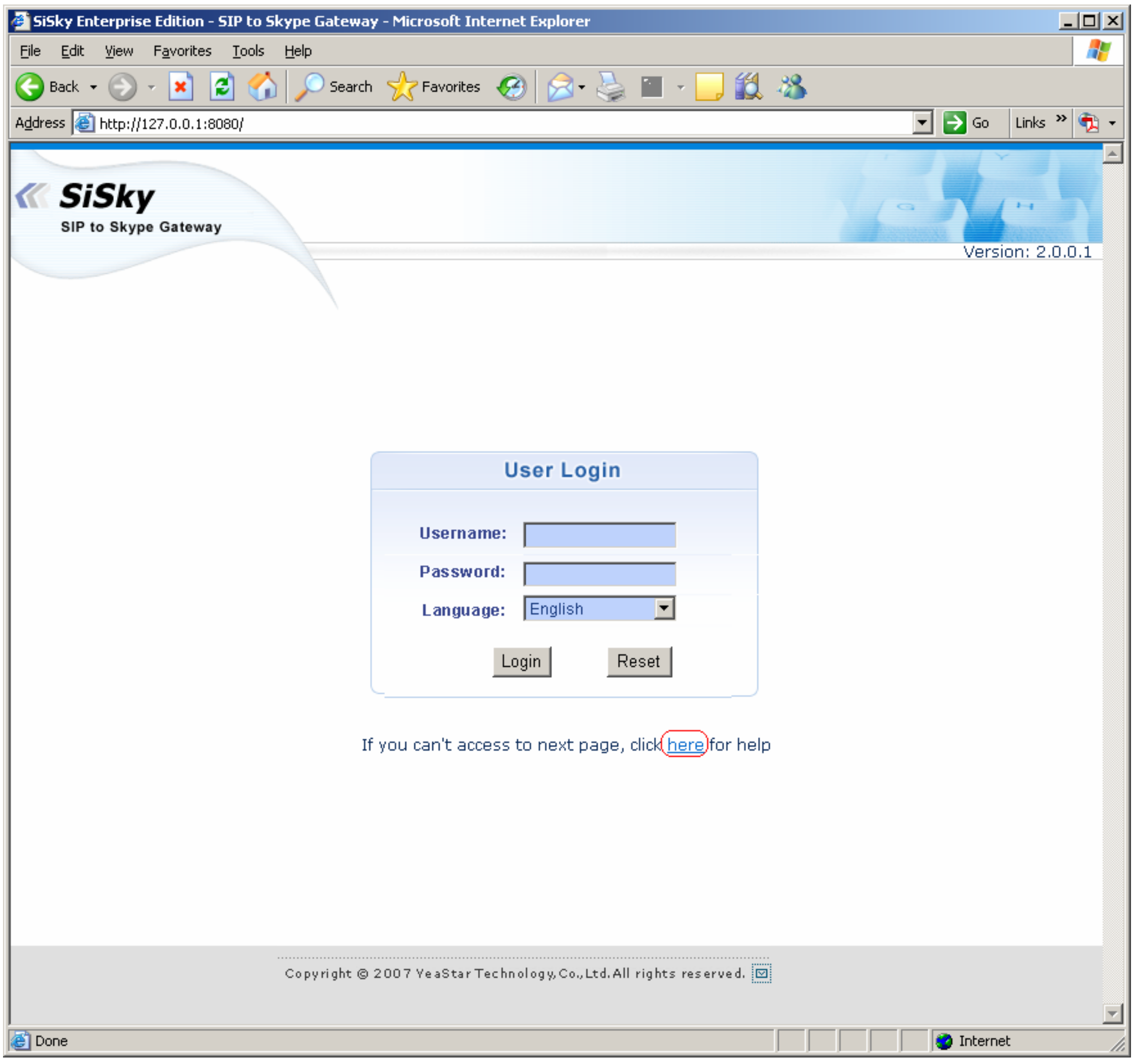


< Figure 46>

If you are not operating in SiSky Server, please open the 8080 port of SiSky Server through IE browser. For example, if the IP address of SiSky Server is 192.168.0.101, then you can type in http://192.168.0.101:8080 in the IE address bar.

9.2 Administrator Logging in

Open the SiSky user login page. Input your username and password to gain access. Use the default administrator username 'admin' and the default password 'password' to login. System has three languages for choose, English, Simplified Chinese and Traditional Chinese.

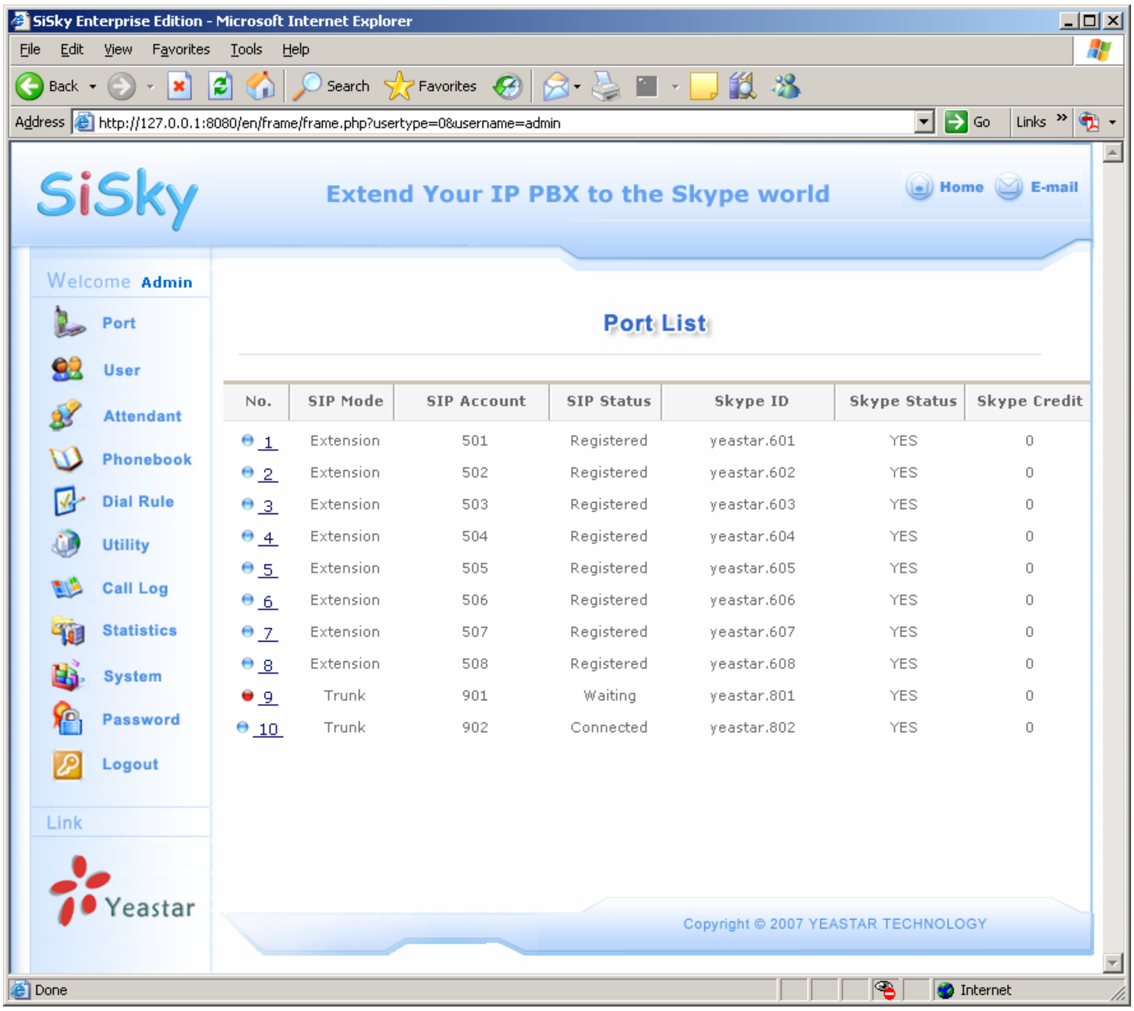


<Figure 47>

If the next page is abnormal after you login, that mainly because of the safety setting of IE browser. Click **here** link on this page and follow the guides to restore the normal page.

9.3 Port Status and Management

9.3.1 Timely display the status of every port



<Figure 48>

9.3.2 Managing Port's Settings, see Figure 49

● **SIP Setting:**

Firstly choose the working mode for this port. If enable *Work as Asterisk'/IPPBX's Extension*, please enter the SIP Extension information and configure the corresponding SIP Extension on Asterisk/IPPBX. If enable *Work as Asterisk'/IPPBX's Trunk*, please configure the SIP Trunk information and configure the SIP Trunk on Asterisk/IPPBX.

● **Skype Setting:**

Allow this Skype status to be shown to everyone: Allow all Skype users to see the Skype ID status of this port.

● Direct In： Enter in an extension number here. Incoming Skype calls to this port through SiSky will ring the extension phone directly and the automatic attendant will be unavailable on this port. If Direct-In number is empty and this port work as Asterisk'/IPPBX's Extension, incoming call from Skype will reach to Automatic Attendant of SiSky.

Note: In Asterisk, dial 7777 means to simulate an incoming call from trunk.

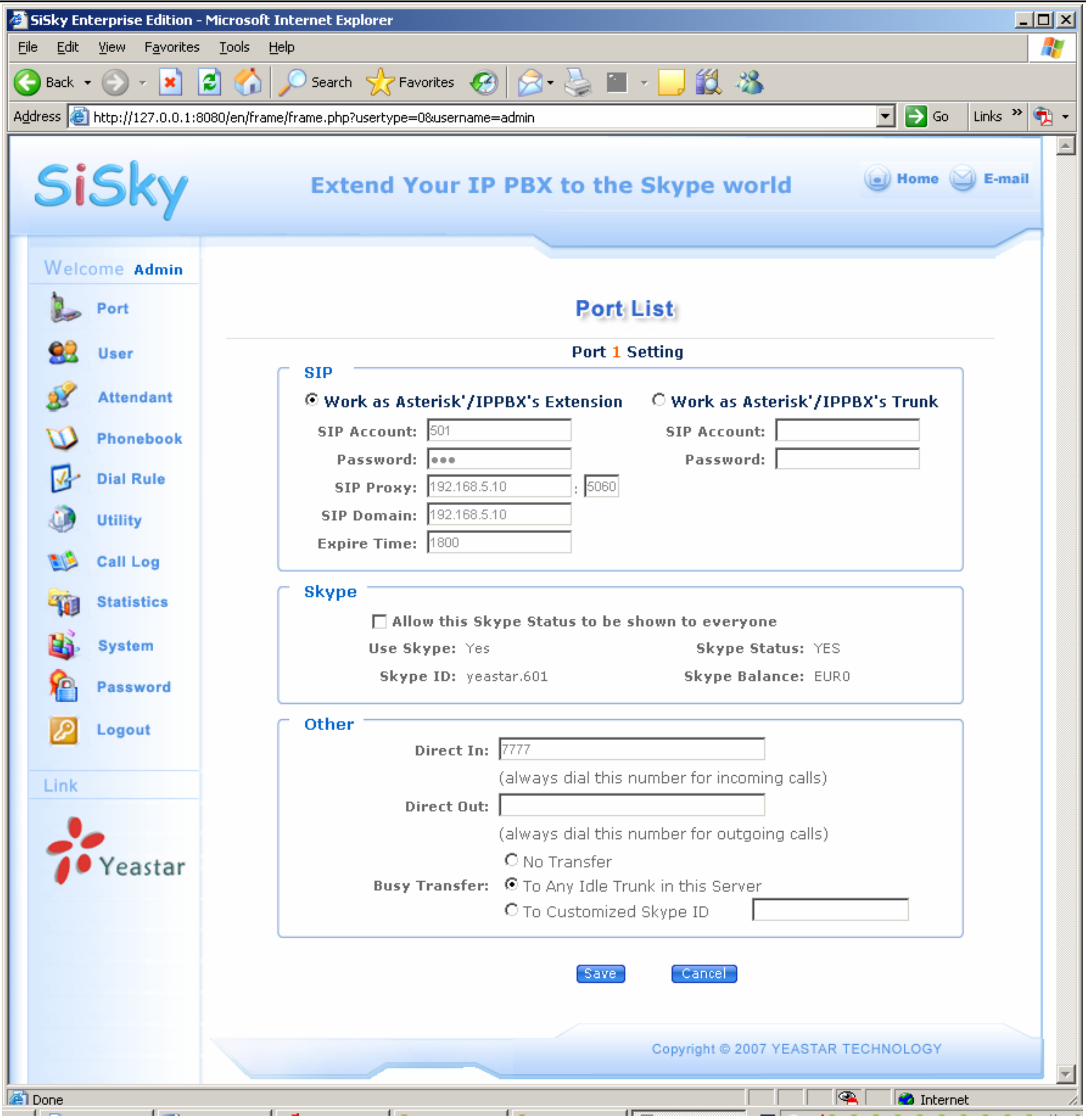
● Direct Out: All the outgoing calls through this port will to this phone number directly. Generally it is for branch connection convenience and set it by Skype ID. You can enter multiple Skype IDs and separate them by a semi-colon (e.g. no1;no2;no3). When the first ID is unreachable, a transfer will be attempted to the next ID automatically.

● Busy Transfer: It helps you to deal with other incoming calls when the port is busy.

1) No Transfer: New incoming call will be hangup

2) To Any Idle Trunk in this Server: New incoming call will be transferred to any idle trunk

3) To Customized Skype ID: You can enter Skype ID of this Server or of other Servers. Multiple Skype IDs are acceptable, separated them by a semi-colon (e.g. no1;no2;no3). When the first ID is unreachable, a transfer will be attempted to the next ID automatically.


SiSky Enterprise Edition - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites
Address http://127.0.0.1:8080/en/frame/frame.php?usertype=0&username=admin
Go Links
SiSky
Extend Your IP PBX to the Skype world
Home E-mail
Welcome Admin
Port
User
Attendant
Phonebook
Dial Rule
Utility
Call Log
Statistics
System
Password
Logout
Link
Yeastar
Port List
Port 1 Setting
SIP
Work as Asterisk'/IPPBX's Extension
Work as Asterisk'/IPPBX's Trunk
SIP Account: 501
Password: •••
SIP Proxy: 192.168.5.10 : 5060
SIP Domain: 192.168.5.10
Expire Time: 1800
SIP Account:
Password:
Skype
Allow this Skype Status to be shown to everyone
Use Skype: Yes
Skype Status: YES
Skype ID: yeastar.601
Skype Balance: EUR0
Other
Direct In: 7777
(always dial this number for incoming calls)
Direct Out:
(always dial this number for outgoing calls)
No Transfer
Busy Transfer: To Any Idle Trunk in this Server
To Customized Skype ID
Save
Cancel
Copyright © 2007 YEASTAR TECHNOLOGY
Done
Internet

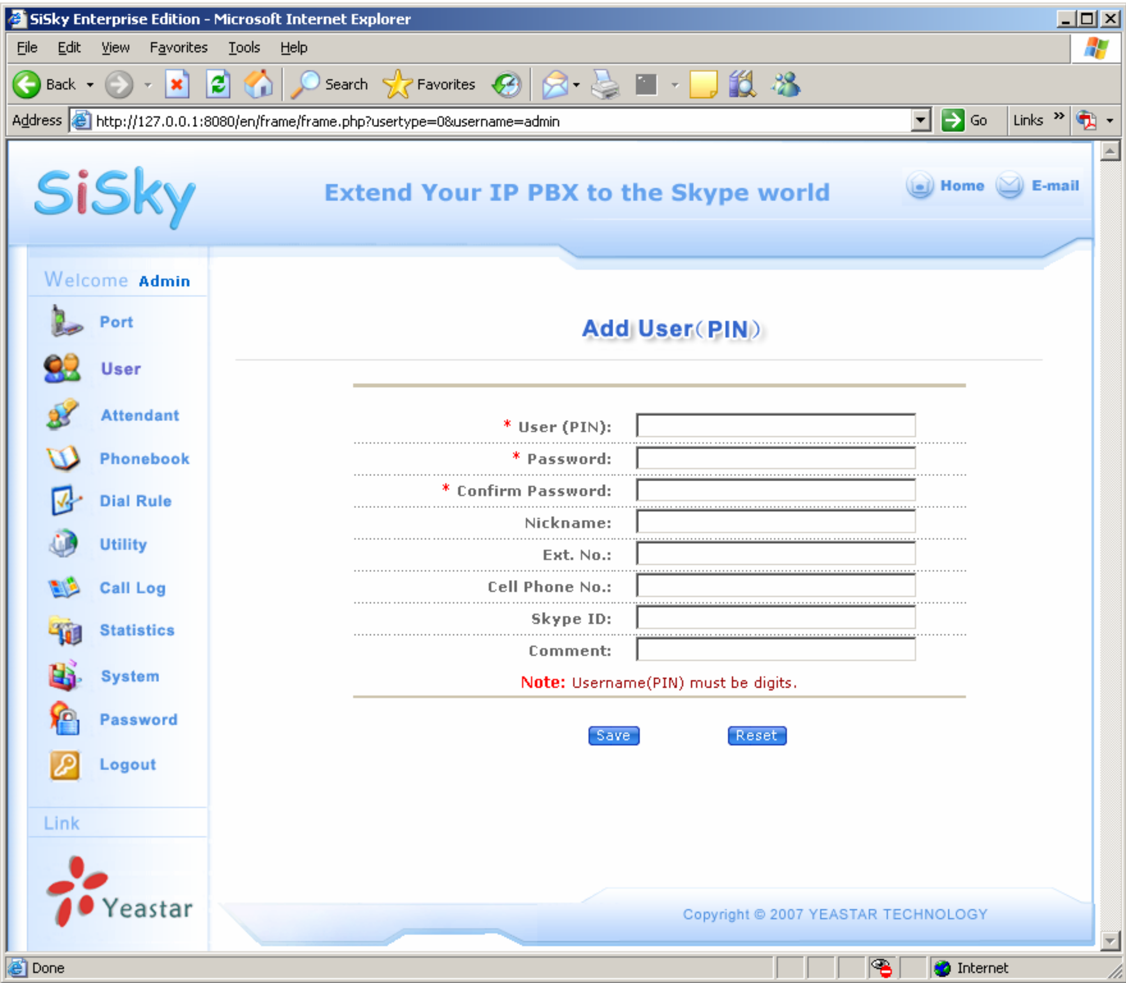
<Figure 49>

9.4 User Management

The User (PIN) List will be unavailable until you enable the multi-user mode. When running SiSky software under multi-user mode, every user is possible to access his/her own private phonebook after logging in WEB. (See details on chapter 9.13---Logging In As Standard Users). User needs to enter his/her own PIN number when making calls.

9.4.1 Adding Users

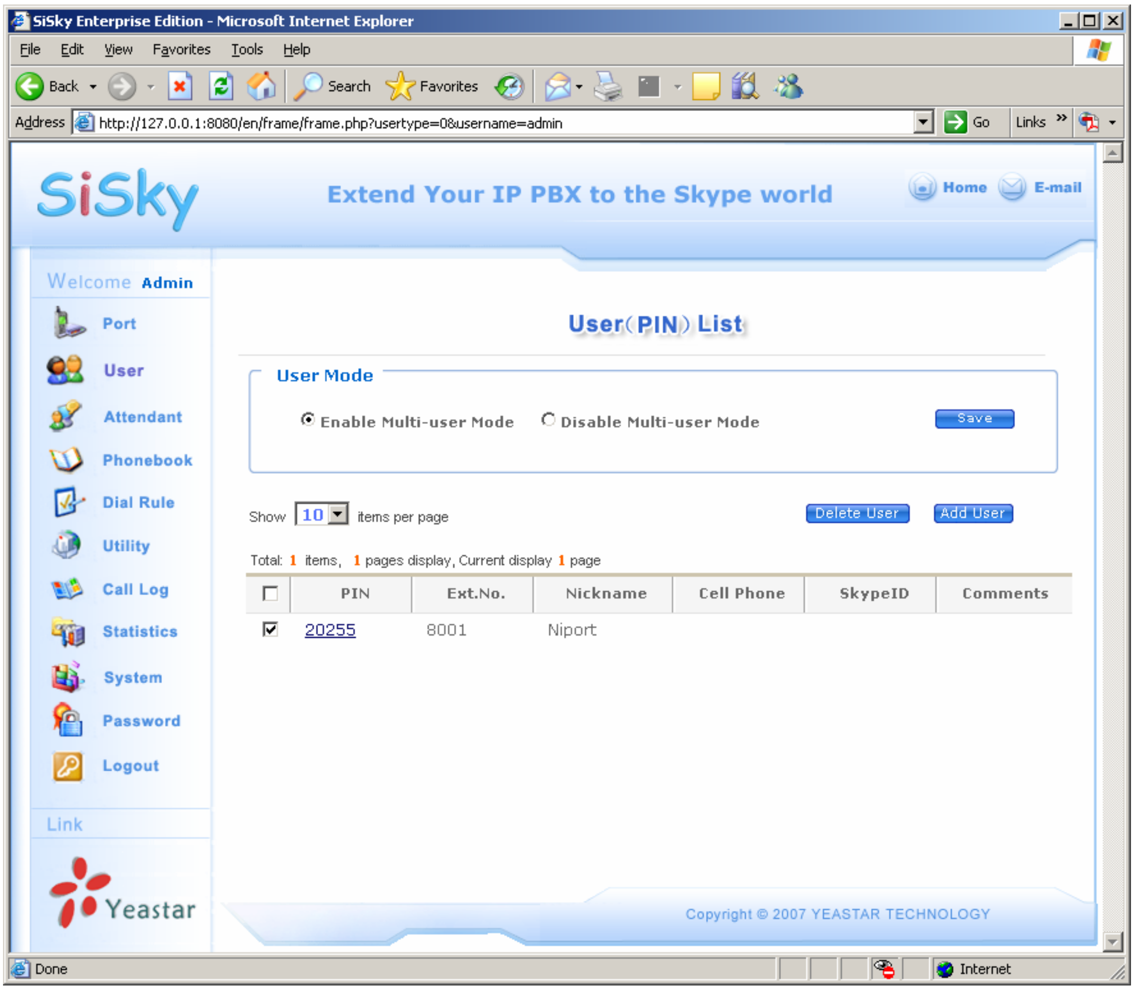
Click **Users** in the left panel. The required fields are PIN and password. User should use the assigned unique PIN and password to login WEB and manage his own private phonebook. And the PIN will be identified when making calls.



<Figure 50>

9.4.2 Deleting User

Select the checkbox next to the user (PIN) you want to delete, click the 'Delete User' button to delete the user.


SiSky Enterprise Edition - Microsoft Internet Explorer
http://127.0.0.1:8080/en/frame/frame.php?usertype=0&username=admin
Extend Your IP PBX to the Skype world
Home
E-mail
Welcome Admin
Port
User
Attendant
Phonebook
Dial Rule
Utility
Call Log
Statistics
System
Password
Logout
Link
User(PIN) List
User Mode
Enable Multi-user Mode
Disable Multi-user Mode
Save
Show 10 items per page
Delete User
Add User
Total: 1 items, 1 pages display, Current display 1 page
PIN
Ext.No.
Nickname
Cell Phone
SkypeID
Comments
20255
8001
Niport
Copyright © 2007 YEASTAR TECHNOLOGY

<Figure 51>

9.5 Managing Automatic Attendant

Auto Attendant is only effect to Extension port.
When there's incoming call from Extension port, SiSky will play the auto attendant first (configured as chapter 7.3.1) and transfer call to destination extension according to the caller's the second dialing (DTMF). This part will introduce the rules of transferring.

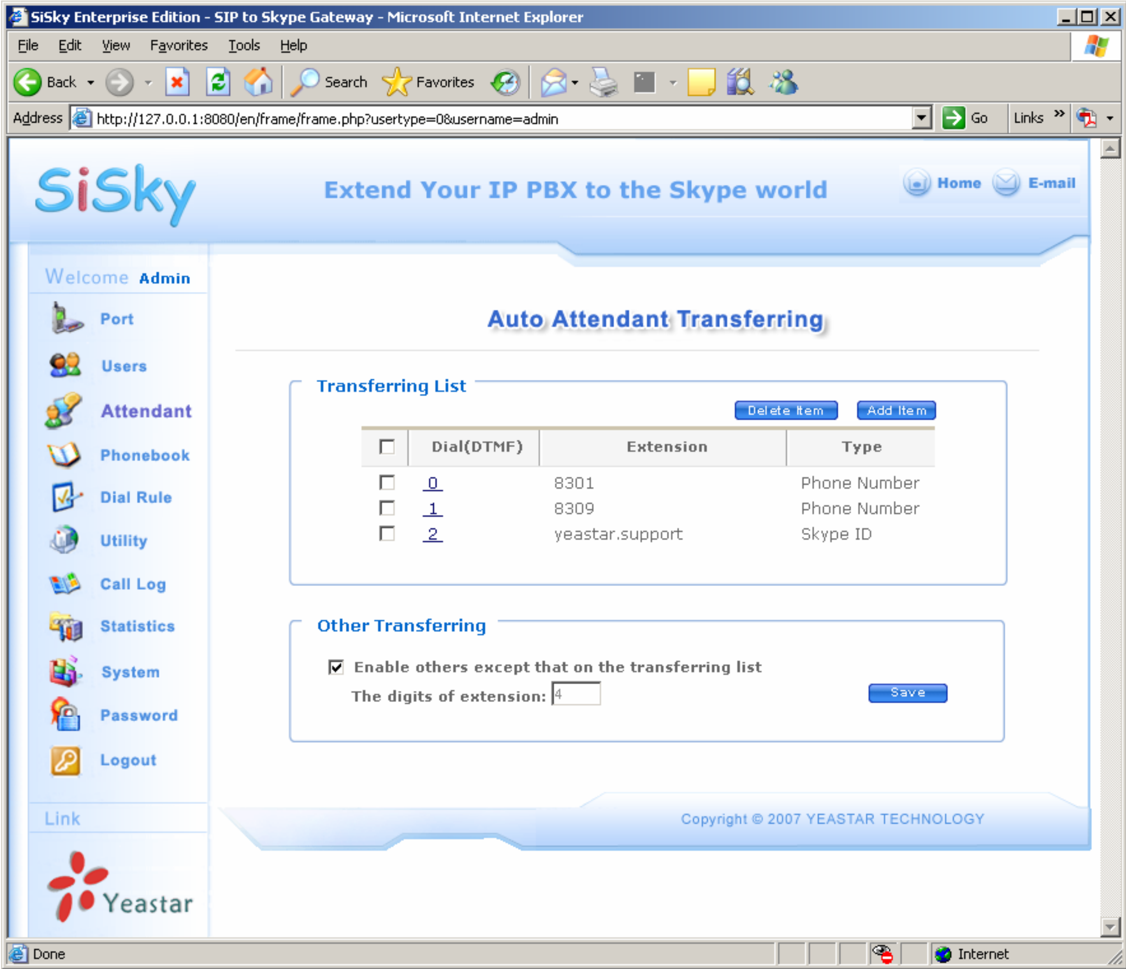
9.5.1 Transferring List

9.5.1.1 Adding Transferring Entry

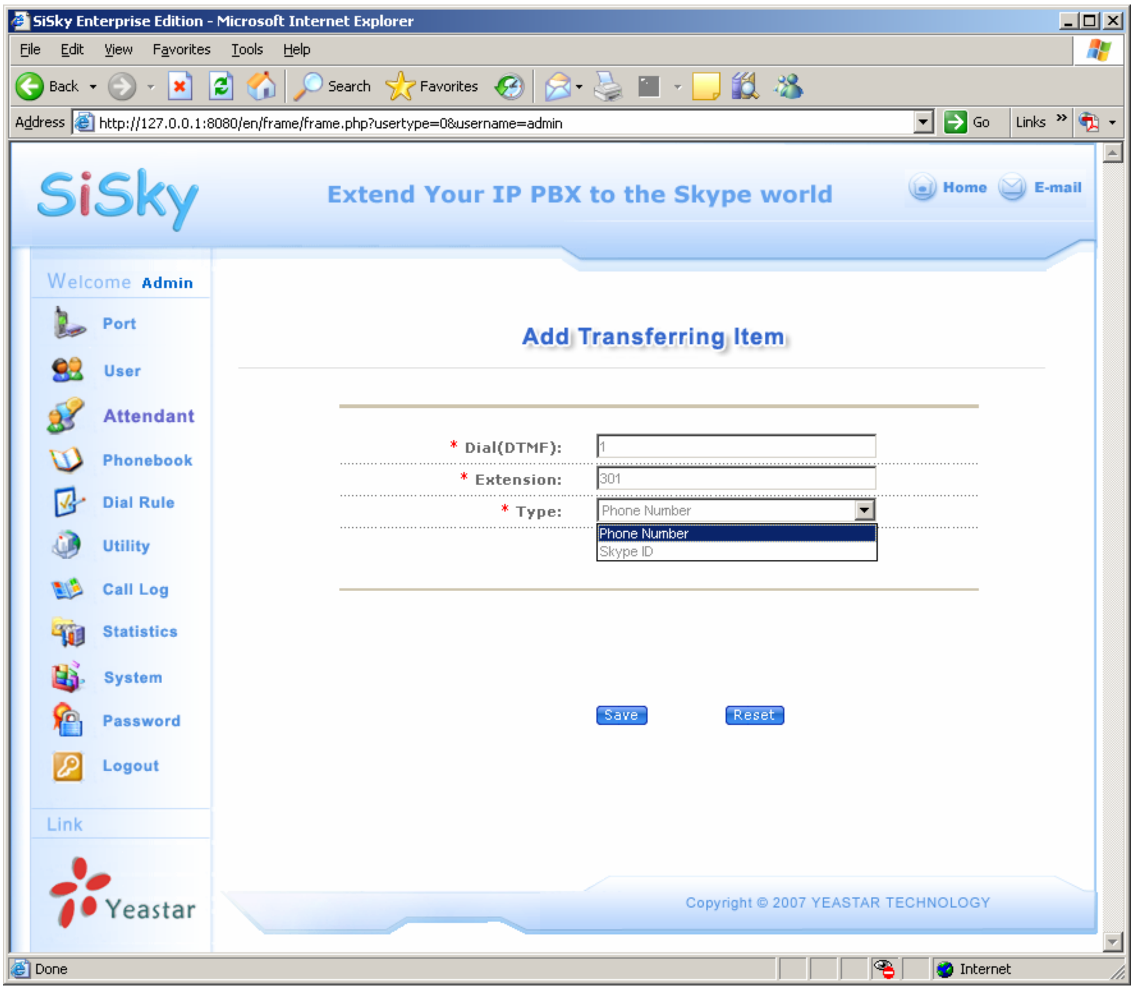
Click **Add Item** to enter Figure 53. Type the destination number (DTMF), corresponding extension number and the extension's type either phone number or Skype ID. For example, on the Auto Attendant, dial 1 to find sales, dial 2 to find support and dial 0 to operator. I set 1 on DTMF and its extension is 8309 and the extension's type is Phone Number.
If the type is Skype ID, the Skype account on user's PC will work as extension.

9.5.1.2 Deleting Transferring Entry

Select the checkbox next to the DTMF you want to delete, click the Delete Item button to delete the entry.


SiSky Enterprise Edition - SIP to Skype Gateway - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites
Address http://127.0.0.1:8080/en/frame/frame.php?usertype=0&username=admin
Go Links
SiSky
Extend Your IP PBX to the Skype world
Home E-mail
Welcome Admin
Port
Users
Attendant
Phonebook
Dial Rule
Utility
Call Log
Statistics
System
Password
Logout
Link
Yeastar
Auto Attendant Transferring
Transferring List
Delete Item
Add Item
Dial(DTMF)
Extension
Type
0
8301
Phone Number
1
8309
Phone Number
2
yeastar.support
Skype ID
Other Transferring
Enable others except that on the transferring list
The digits of extension: 4
Save
Copyright © 2007 YEASTAR TECHNOLOGY
Done
Internet

<Figure 52>



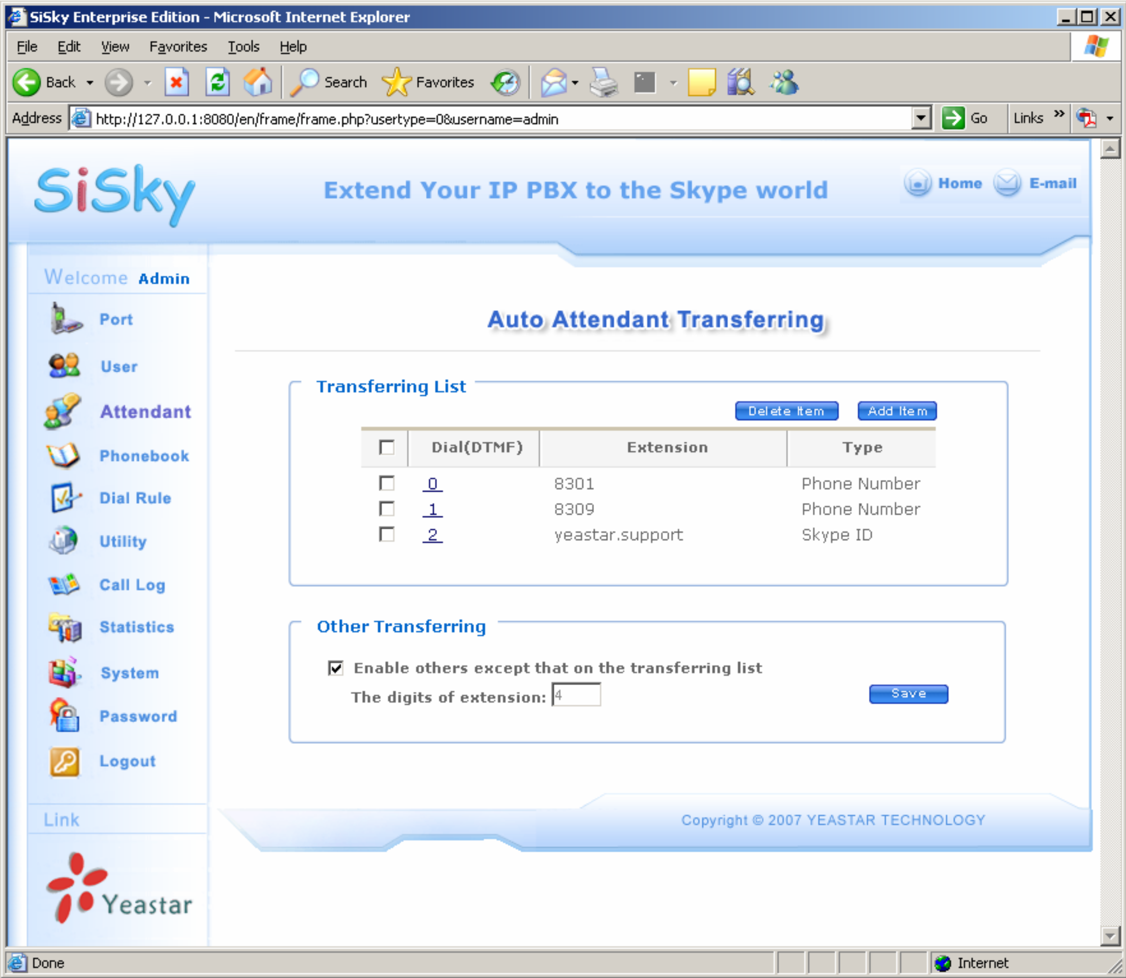
<Figure 53>

9.5.2 Enable others except that on the transferring list

If enable other transferrings, when the caller dials a number (DTMF) that can match none of the entries on Transferring List, he must enter the full extension number to transfer. As Figure 52, input the digits of extension to transfer in time when caller finish dialing.

9.5.3 Example

The auto attendant plays: Welcome to Yeastar Company. For product information, please press 1; For technical support, please press 2; For help press 0 or dial extension number directly.
In Yeastar company, 8301 is the extension of operator; 8309 of sales; technical support wants to answer calls by Skype and his ID is *yeastar.support*.
See Figure 54. **Enable** others except that on the transferring list, and input **4** for 'The digits of extension', then if caller dial the extension number 8306 directly after hearing auto attendant, the call will be reachable to 8306 Extension.



<Figure 54>

9.6 Phonebook

Click Phonebook in the left panel to check all the public contacts, Figure 55. As an administrator, you can manage (adding or deleting) the public contacts.

9.6.1 Adding Public Contact

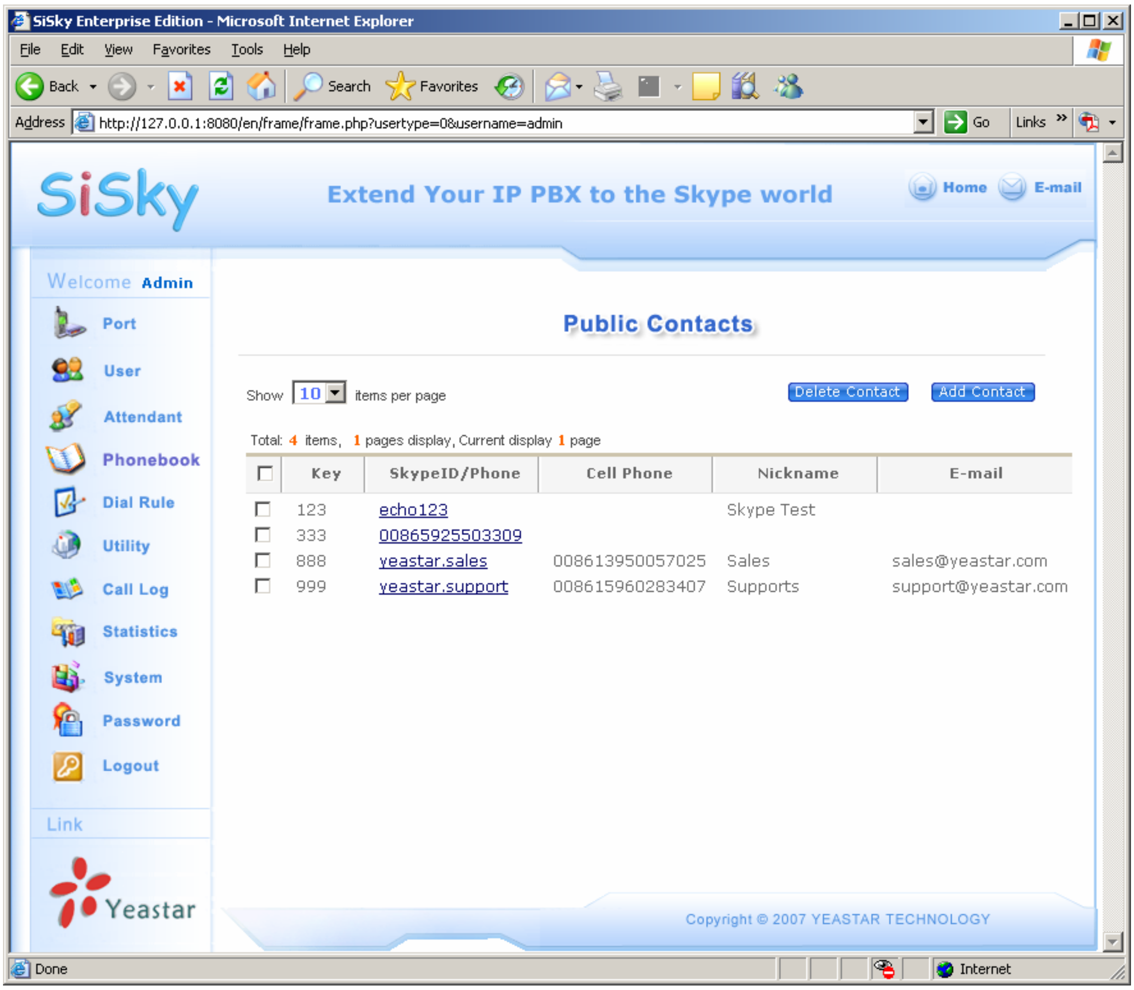
Click the **Add Contact** button (as shown Figure 55) to add a new public contact.
Please input the required information, such as Speed-Dial Key and Skype ID/Phone. The other information is optional. Click **Save** to save the settings.

Please refer chapter 9.8.2 to Import Skype Contacts to public Phonebook.

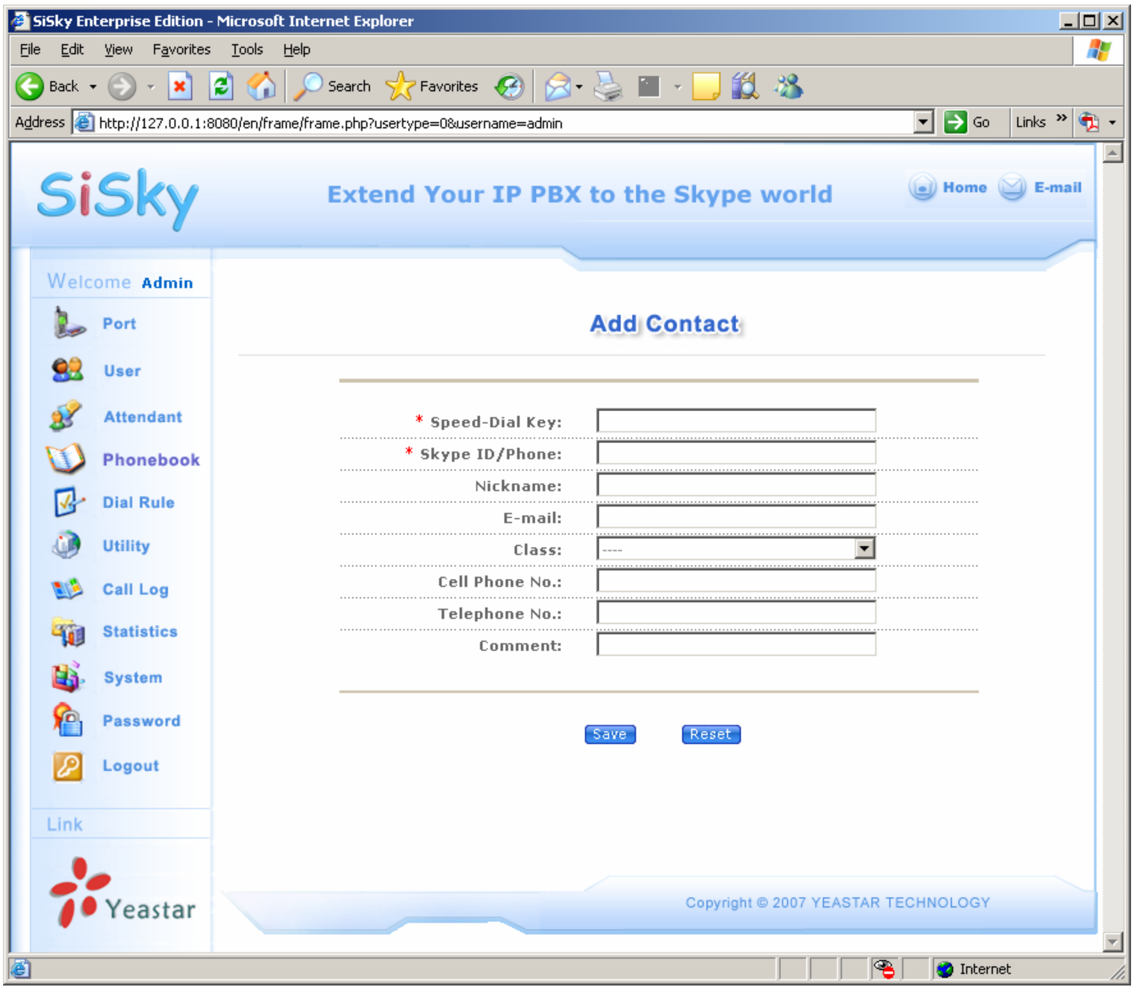
Note: If you want to add a regular phone number, input the number into Skype ID/Phone field in the following SkypeOut format: **00+country code+ area code+local phone number** or **00+country code+mobile phone number**.

9.6.2 Deleting Public Contact

Select the checkbox next to the contact you want to delete, click the 'Delete Contact' button to delete the entry.


SiSky Enterprise Edition - Microsoft Internet Explorer
http://127.0.0.1:8080/en/frame/frame.php?usertype=0&username=admin
Extend Your IP PBX to the Skype world
Home
E-mail
Welcome Admin
Port
User
Attendant
Phonebook
Dial Rule
Utility
Call Log
Statistics
System
Password
Logout
Link
Public Contacts
Show 10 items per page
Delete Contact
Add Contact
Total: 4 items, 1 pages display, Current display 1 page
Key
SkypeID/Phone
Cell Phone
Nickname
E-mail
123 echo123 Skype Test
333 00865925503309
888 yeastar.sales 008613950057025 Sales sales@yeastar.com
999 yeastar.support 008615960283407 Supports support@yeastar.com
Copyright © 2007 YEASTAR TECHNOLOGY
Done
Internet

<Figure 55>


SiSky Enterprise Edition - Microsoft Internet Explorer
http://127.0.0.1:8080/en/frame/frame.php?usertype=0&username=admin
SiSky
Extend Your IP PBX to the Skype world
Home
E-mail
Welcome Admin
Port
User
Attendant
Phonebook
Dial Rule
Utility
Call Log
Statistics
System
Password
Logout
Link
Add Contact
* Speed-Dial Key:
* Skype ID/Phone:
Nickname:
E-mail:
Class:
Cell Phone No.:
Telephone No.:
Comment:
Save
Reset
Copyright © 2007 YEASTAR TECHNOLOGY

<Figure 56>

9.7 Dial Rule

Making calls through Skype, one have to conform to the Skype dialing scheme as well as calling through SIP. Maybe you had already got used to SIP scheme and not accustomed to Skype rules. Therefore, **Dial Rule** settings will assist you to make Skype calls based on traditional PSTN calling habits. Keep the dialing habit as same as PSTN.

Before making a SkypeOut call, make sure you have purchased SkypeOut credit for the Skype account.

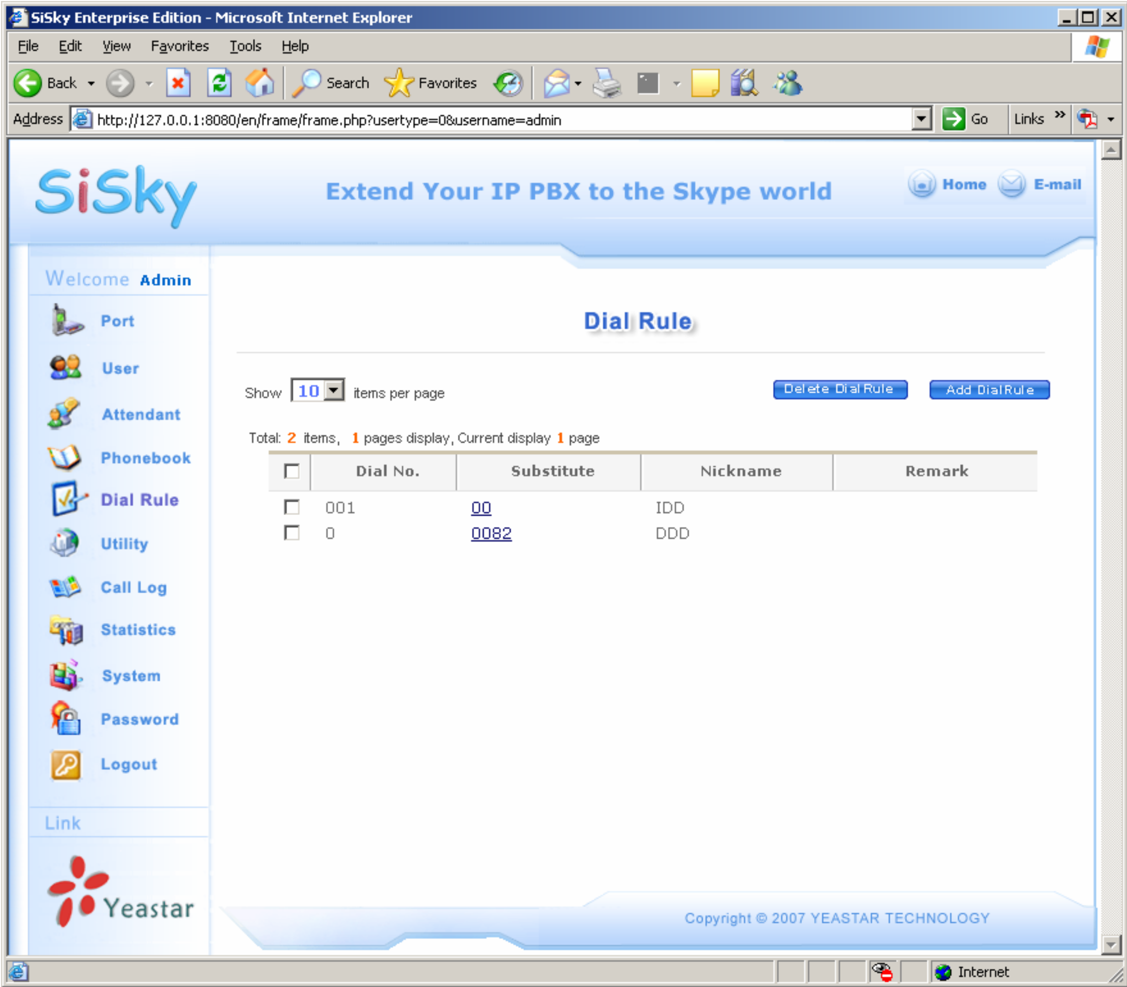
For example, a SIP user **in Korea** wants to make an international call IDDD (international direct distance dialing),which format is **001+country code+ area code+ telephone number**; make an domestic call DDD (domestic distance dialing), which format is **0+ area code+telephone number.** While the SkypeOut format no matter of domestical or international calls is **00+ country code+ area code+telephone number,** therefore you can setup the dialing rules as Figure 57.

1. A Korea user calls to China in **SIP format**: the area code is "592", phone number is "5503309", then he would dial: 001 +86+ 592 + 5503309;
2. A Korea user calls to China in **SkypeOut format**: the area code is "592", phone number is "5503309", then he would dial: 00+86+592+5503309

In order to not change the dialing habit, user in Korea can take advantage of the Dial Rule to substitute **00** for **001.** When he dialing call begin by 001, SiSky will identify the 001 as international call requirement and transfer it to 00 automatically to conform SkypeOut format.

3. A Korea user in Pusan calls to Seoul in traditional **SIP format**: the area code is "2", phone number is "7571234", then he would dial: 0+2+7571234;
4. A Korea user in Pusan calls to Seoul in traditional **SkypeOut format**: the area code is "2", phone number is "7571234", then he would dial: 00+82(country code)+2+7571234;

In order to not change the dialing habit, user in Korea can take advantage of the Dial Rule to substitute **0082** for **0.** When he dialing call begin by 0, SiSky will identify the 0 as domestic call requirement and transfer it to 0082 automatically to conform SkypeOut format.



<Figure 57>

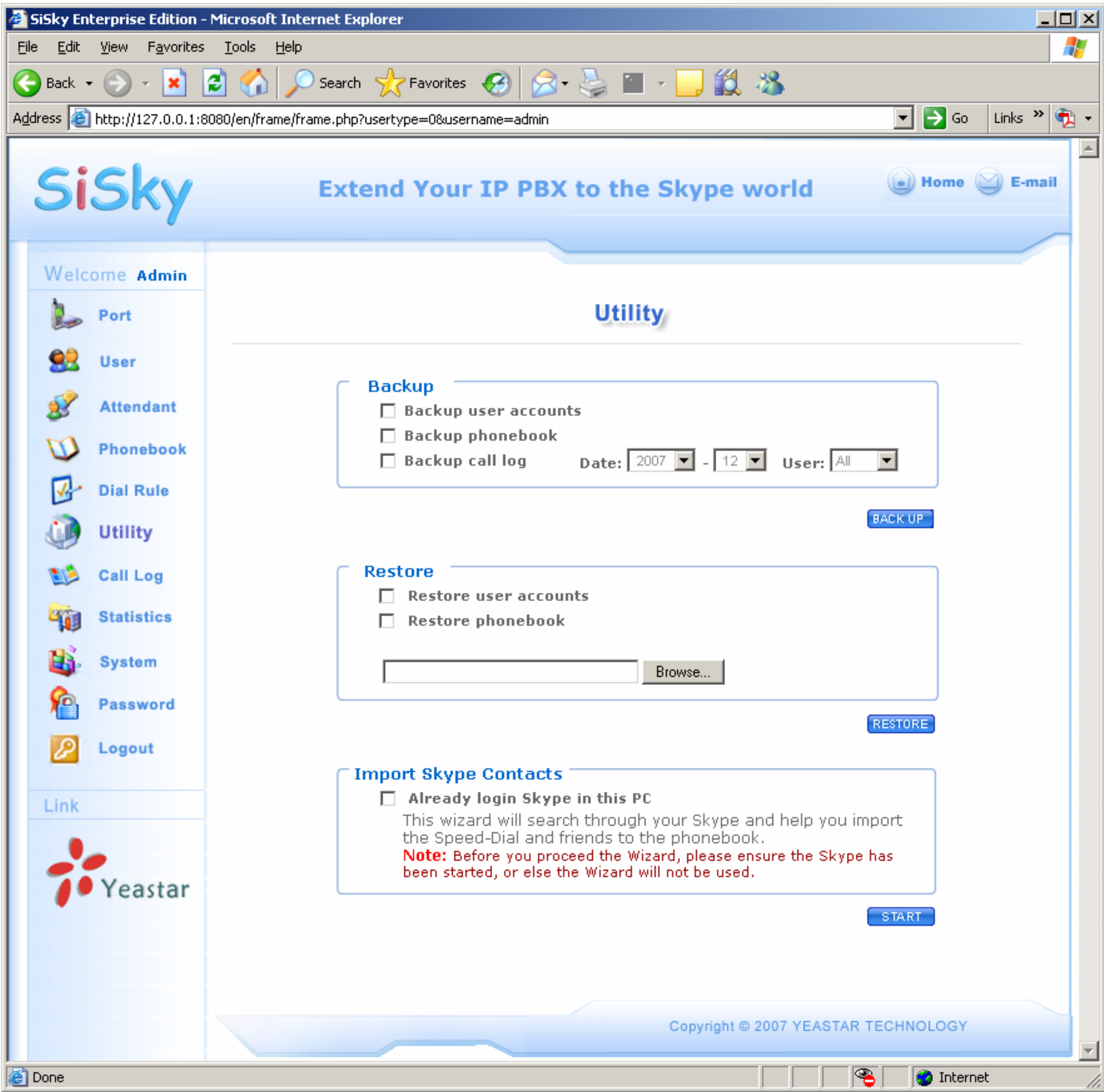
Although the Dial Rule Configuration seems a little complicated, you will find its powerful functions if you can well understand its usages.

9.8 Backing up & Restoring

9.8.1 Database Backup and Restore

As an administrator, you can backup and restore user accounts, phonebook, and call log by choosing the **Utility** in the left panel as shown on Figure 58. Select the type of application you want to backup, then click **BACK UP** butoon and choose a destination to save the file, in which call log can be respectively backed up yearly/monthly or user class.

The **Restore** option will restore your data to the existing database. Select the type of data you want to restore and click **Browse** to choose the location of the backup file to load, then click **RESTORE**. The data is restored to the database.

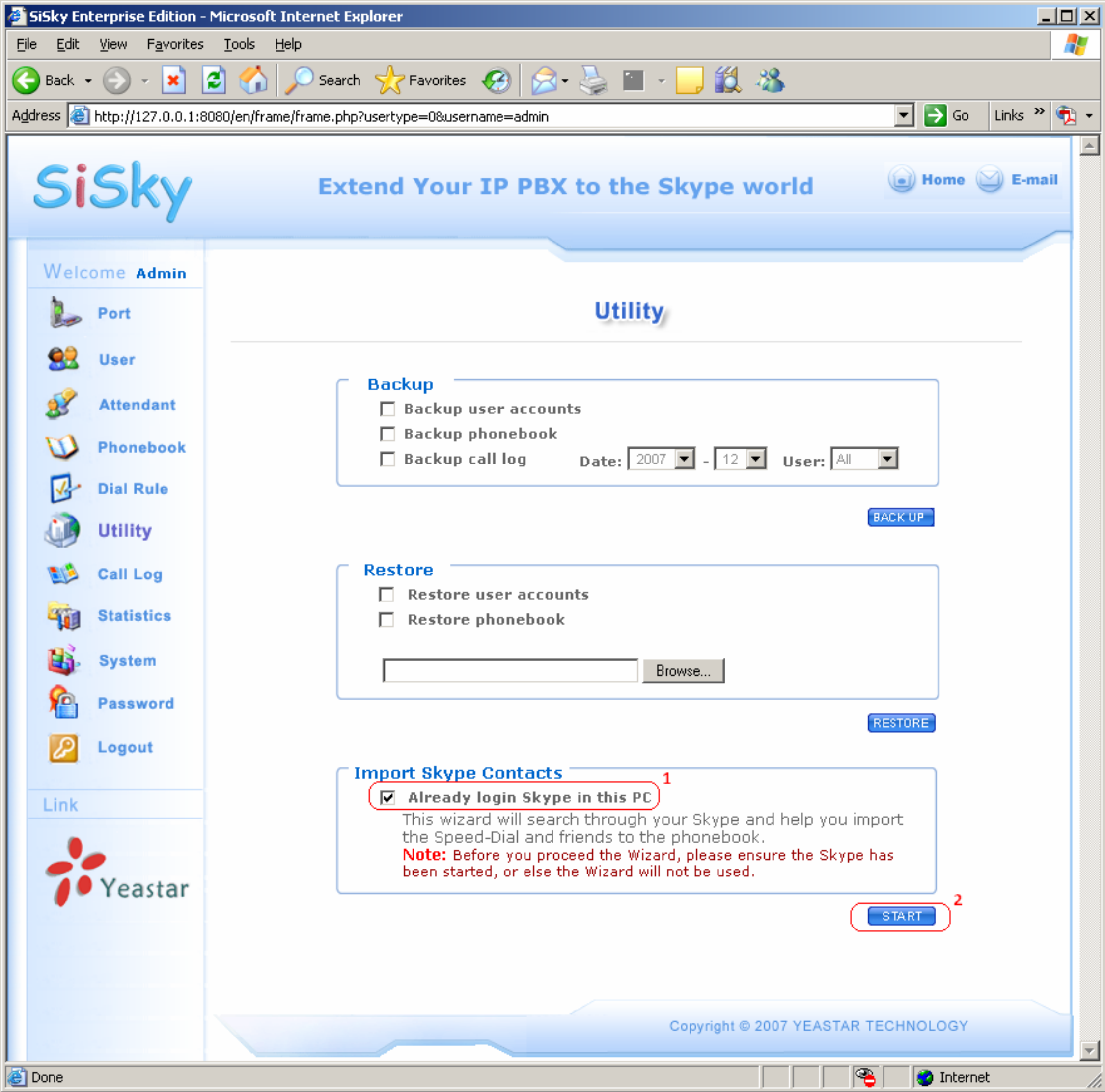


<Figure 58>

9.8.2 Importing Skype Contacts

The function can be used to import all Skype contacts in this Server to phonebook. As an administrator, it will import to public contacts.

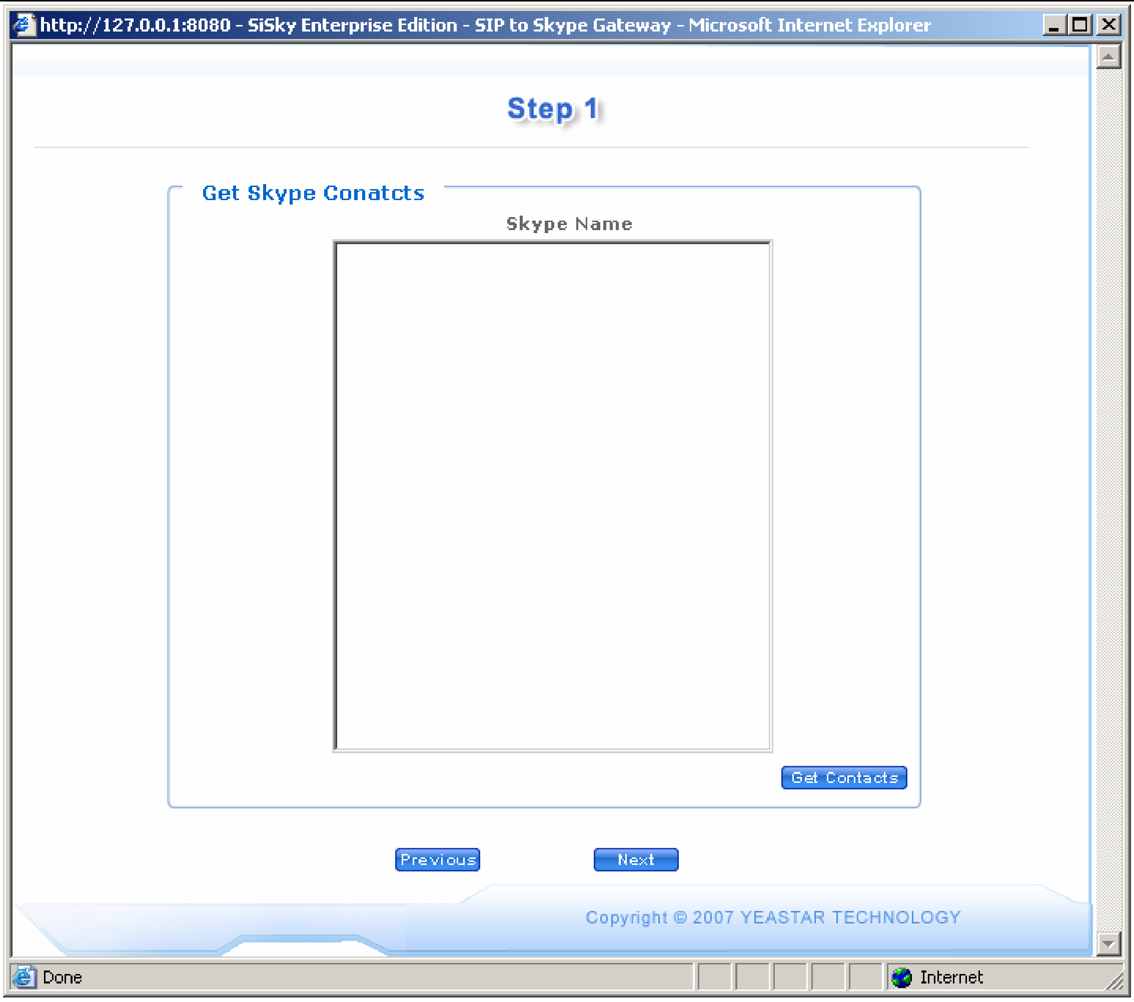
Make sure the Skype in this PC had already log you in, enable **Already login Skype in this PC**, then click **Start** to import contacts. The wizard will search through your Skype and import the Speed-Dial contacts and friends to the phonebook. See Figure 59



<Figure 59>

Step 1: Get Skype Contacts

Click **Get Contacts** as shown on Figure 60



<Figure 60>

On initial use, the window **A web page is attempting to use Skype contact management** will appear. Click either the first or the second options to allow this site to use Skype contact management. Then click **OK** to save the settings.

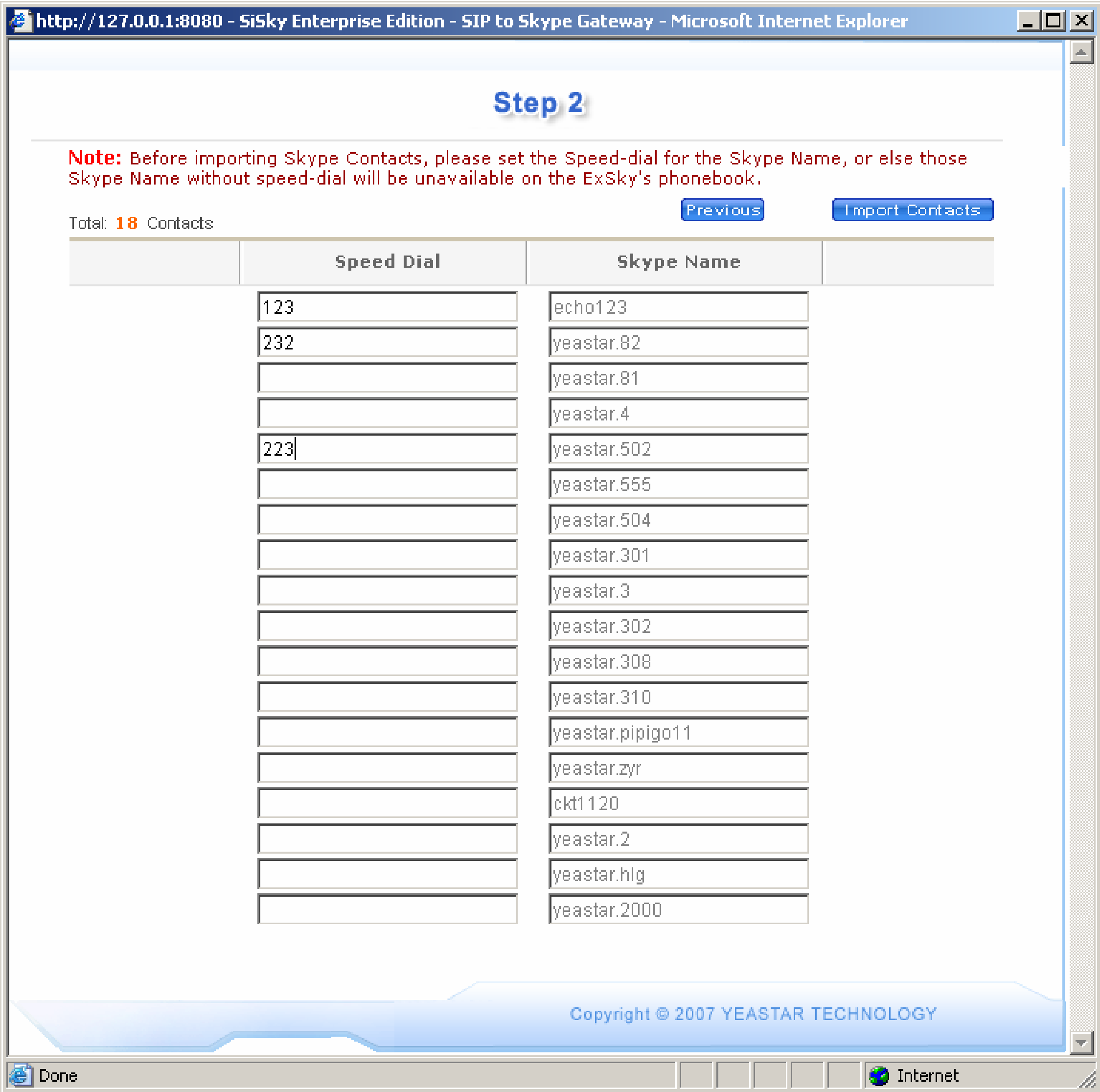
<Figure 61>

Your Skype contacts will show up in the column Get Skype Contacts, click **Next** to next step.

<Figure 62>

Step 2: Setting Speed-Dial number for contacts

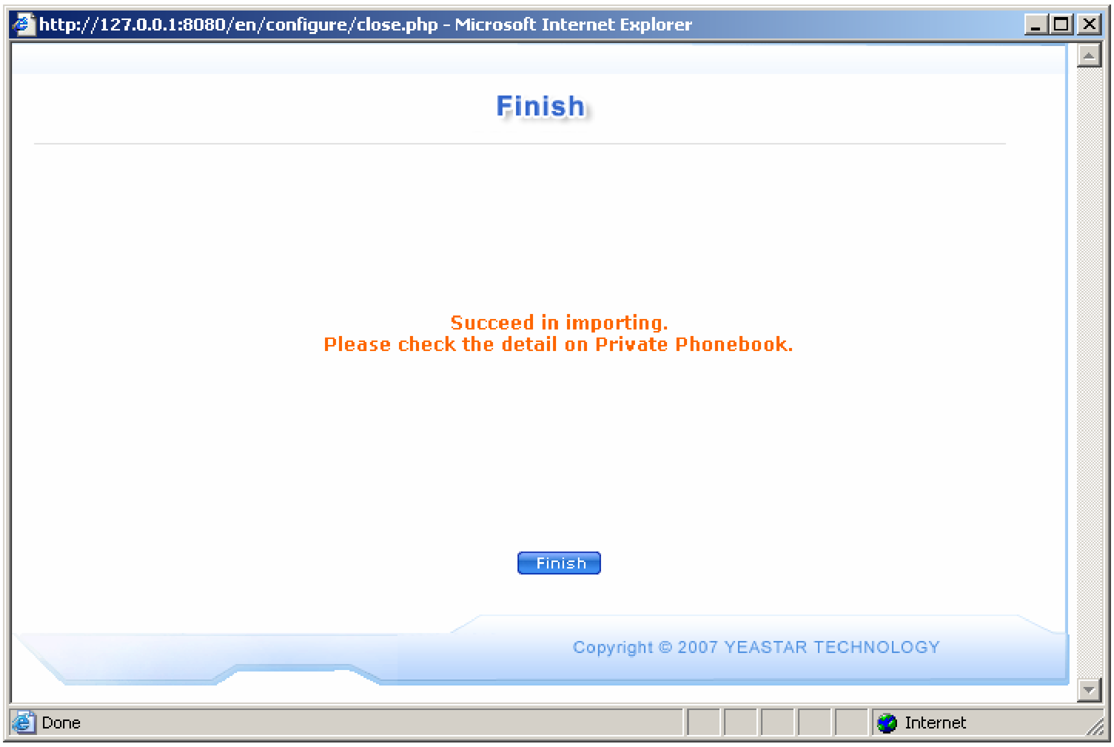
Set Speed-Dial number for those Skype Names who you want to import to phonebook. Empty it for those Names you don't want to import. The Skype Name without speed-dial number will be unavailable on the SiSky's phonebook. Click **Import Contacts** button.



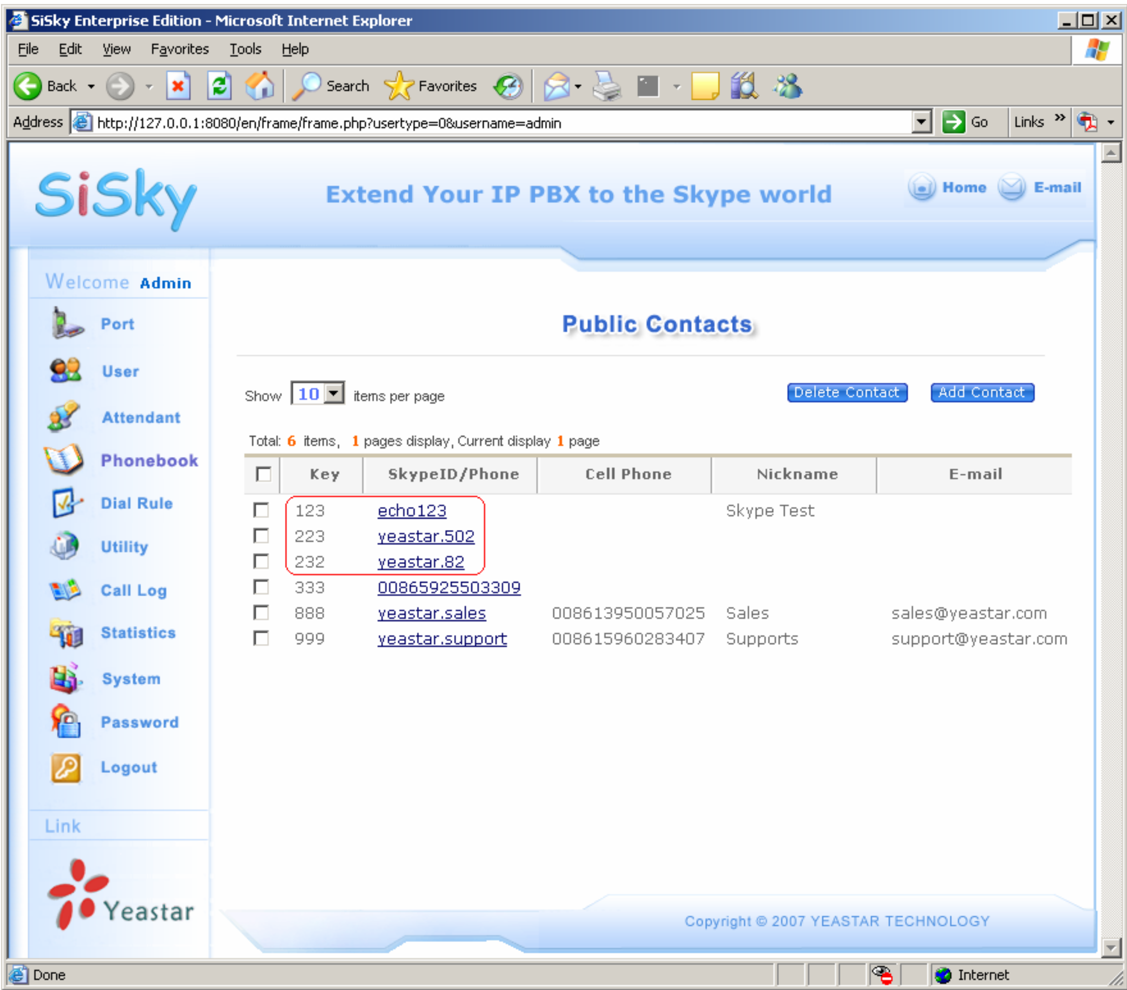
<Figure 63>

Step 3: Finishing and finding the newly imported name on **Public Contacts** as Figure 63.

When finished, a message window will appear as Figure 64. Click the **Finish** button to close.



<Figure 64>

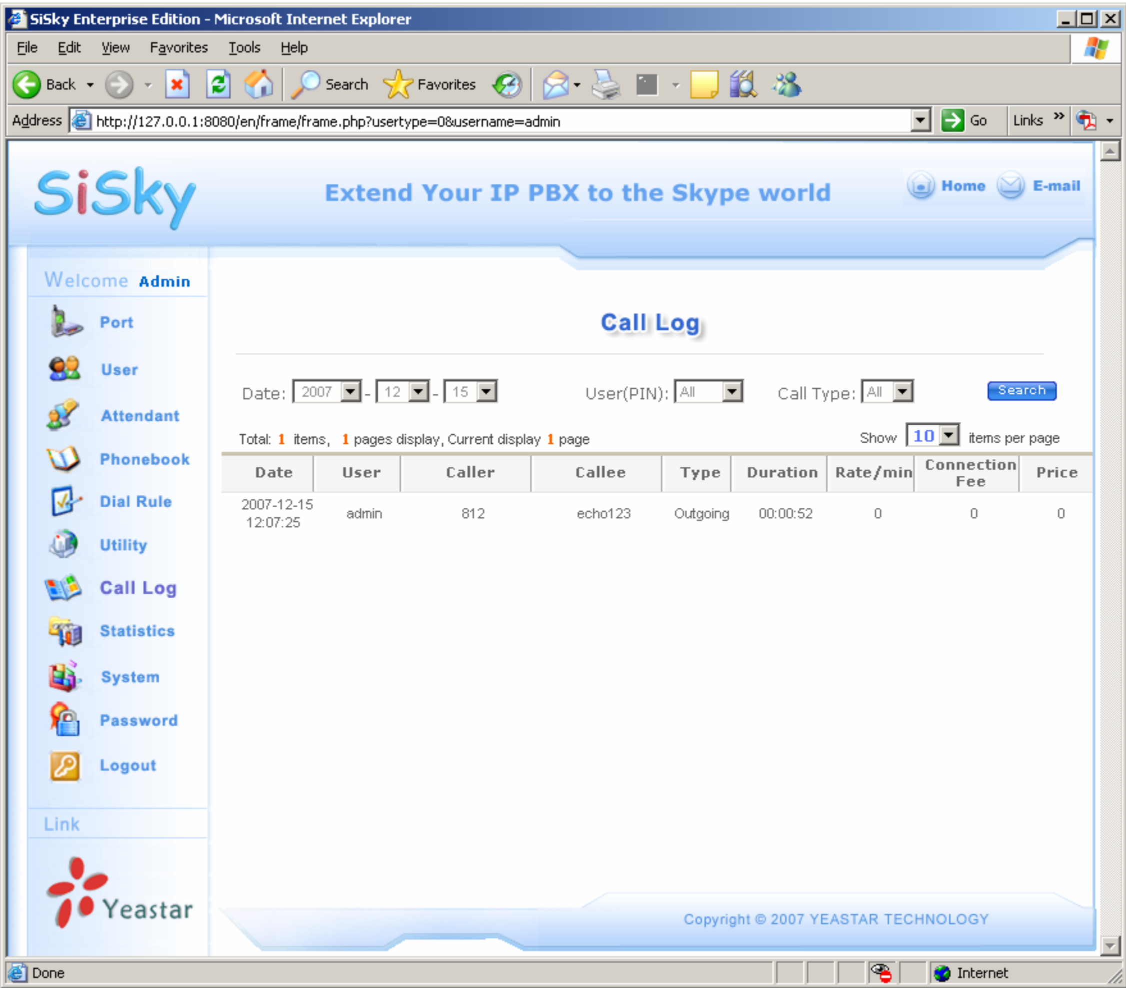


<Figure 65>

9.9 Viewing Call Log

Administrators can view all users' call log. Call Log captures all call details: calling time, caller, callee number, nick name, call type, call duration, rate per minute and total price. The SkypeOut calling rate is not dependant on from where the call is made, only to where it is made. For your convenience, to check on rates, the calling rates are obtained from Skype. However, keep in mind the followings:

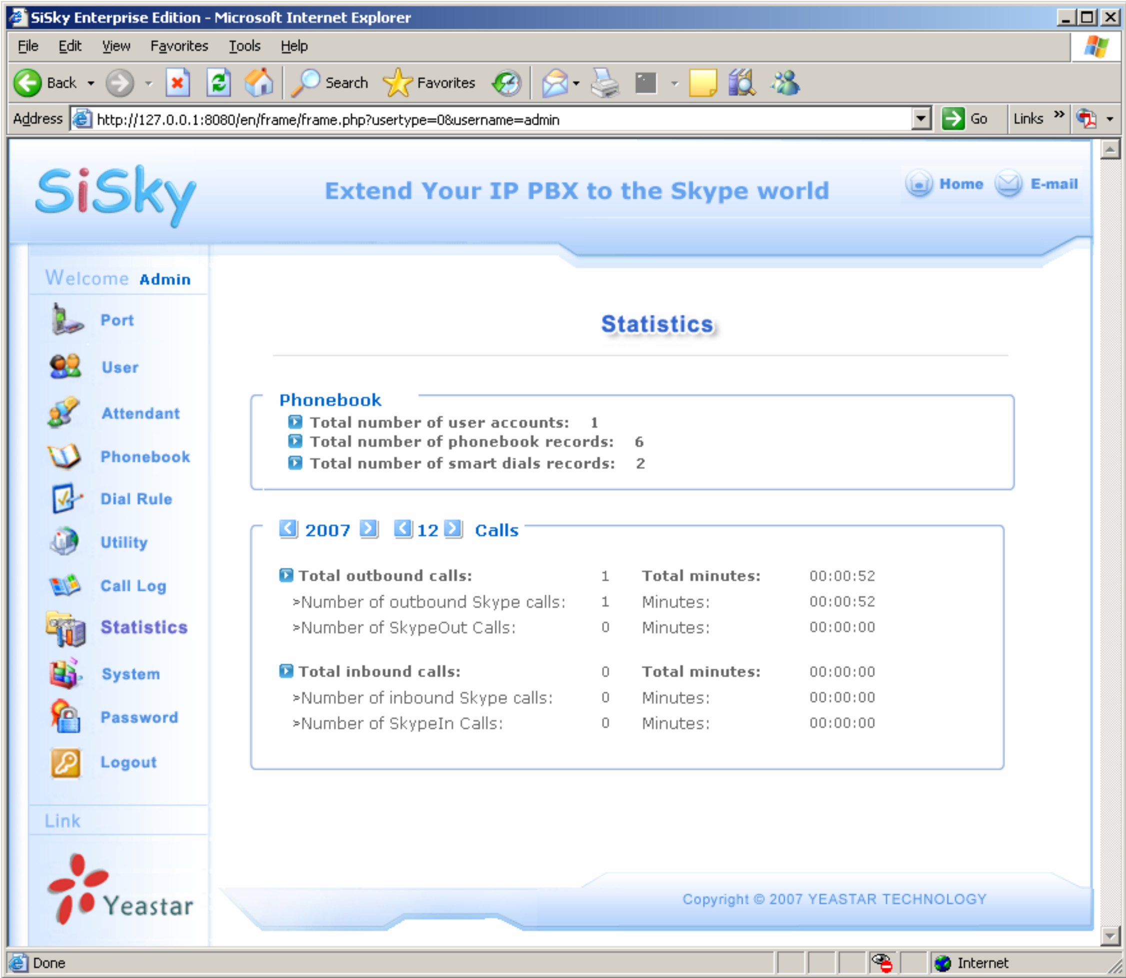
1. The SkypeOut rates and charges are in the same currency setting as when SkypeOut credits are purchased. If you change the currency setting, the rates and charges for the future calls will be changed to the new currency setting, but those before adjustment on the call log will show the prvious currency setting.
2. SkypeOut credit is on minute.
3. The call duration of Call Log is generated from the Skype client unit.
4. The SkypeOut duration record from the client unit might be 3-5 seconds different from the Skype server. There might be one-minute charge difference. Use the charge from the Skype server for the final charges.



<Figure 66>

9.10 Viewing Statistics

After you are done with the configuration procedure, SiSky's statistics screen will show you all the data in your phonebook, number of user accounts, number of calls and each line's usage rate.



<Figure 67>

9.11 System

9.11.1 Balancing engine

After enabled this function, SiSky will balance the flow rate among Skype IDs of all ports.

For example, a user configured 8 trunks on SiSky, totally 8 Skype IDs. When a SIP user calls to the first trunk on SiSky, it will be out from the Skype ID with the shortest speaking time.

9.11.2 Skype Rules Settings

Note: This setting aims to the Skype Unlimited World service specially, if you are using other Skype Unlimited Call services, it's not necessary to configure the following settings.

Because of the fair usage policy that calls to phones and mobiles and Skype To Go* calls are included in users' subscription subject to a fair usage limit of 10,000 minutes per user per month, with a maximum of 6 hours per day. Also, no more than 50 different numbers in total can be called per day, here two options are available on SiSky for users to set up in order to apply with Skype rules.

1) Maximum Skype call time in total per day (minutes)

Default value is 0 that stands for unlimited. The suggested value is equal to or less than 300 minutes.

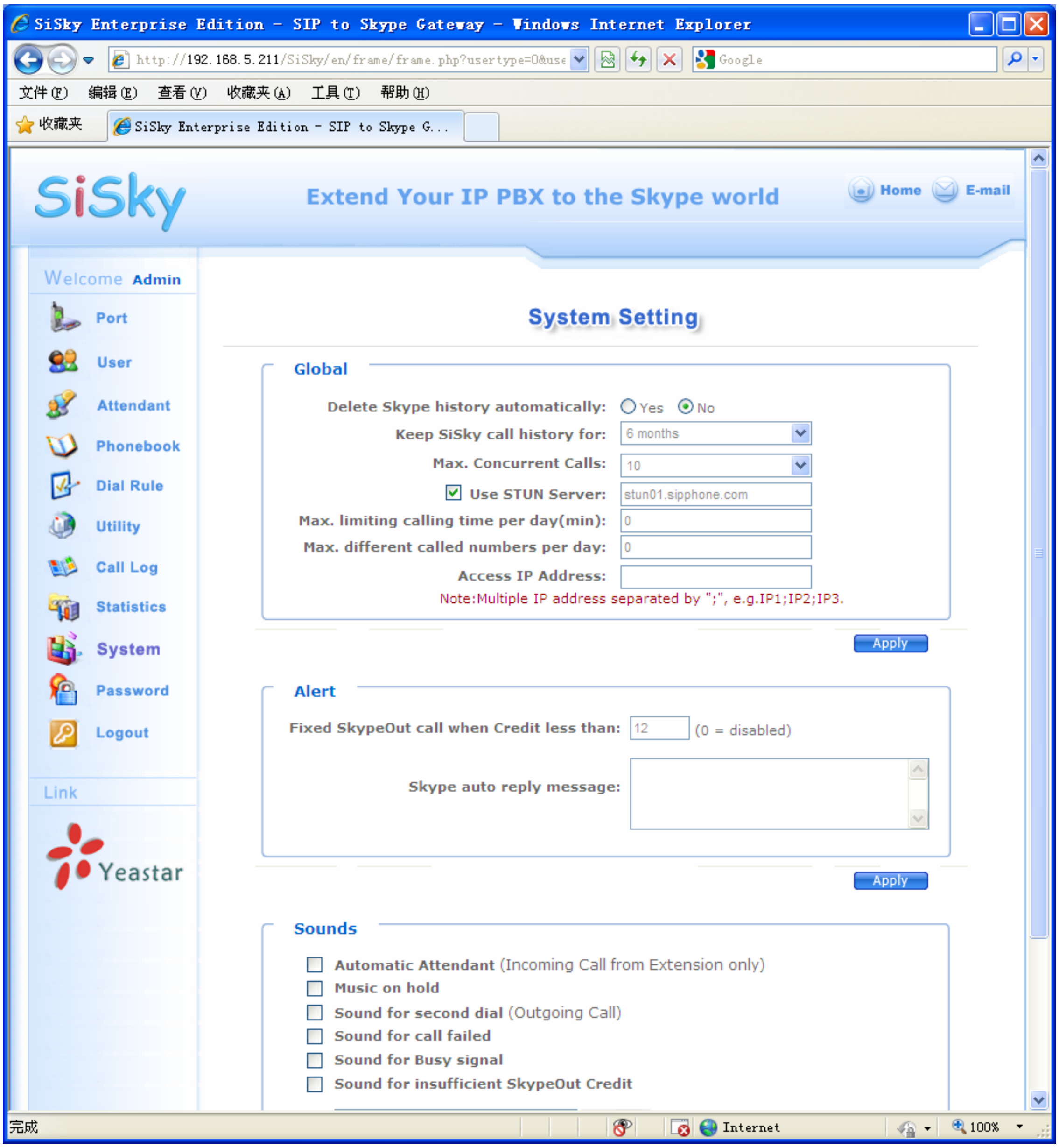
2) Maximum different numbers in total per day

Default value is 0 that stands for unlimited. The suggested value is equal to or no more than 50.

9.11.3 Other Settings

Global and **Alert** Setting, please refer to 7.2 General Setting.

Sounds Setting, please refer to 7.3 Sounds Setting.



<Figure 68>

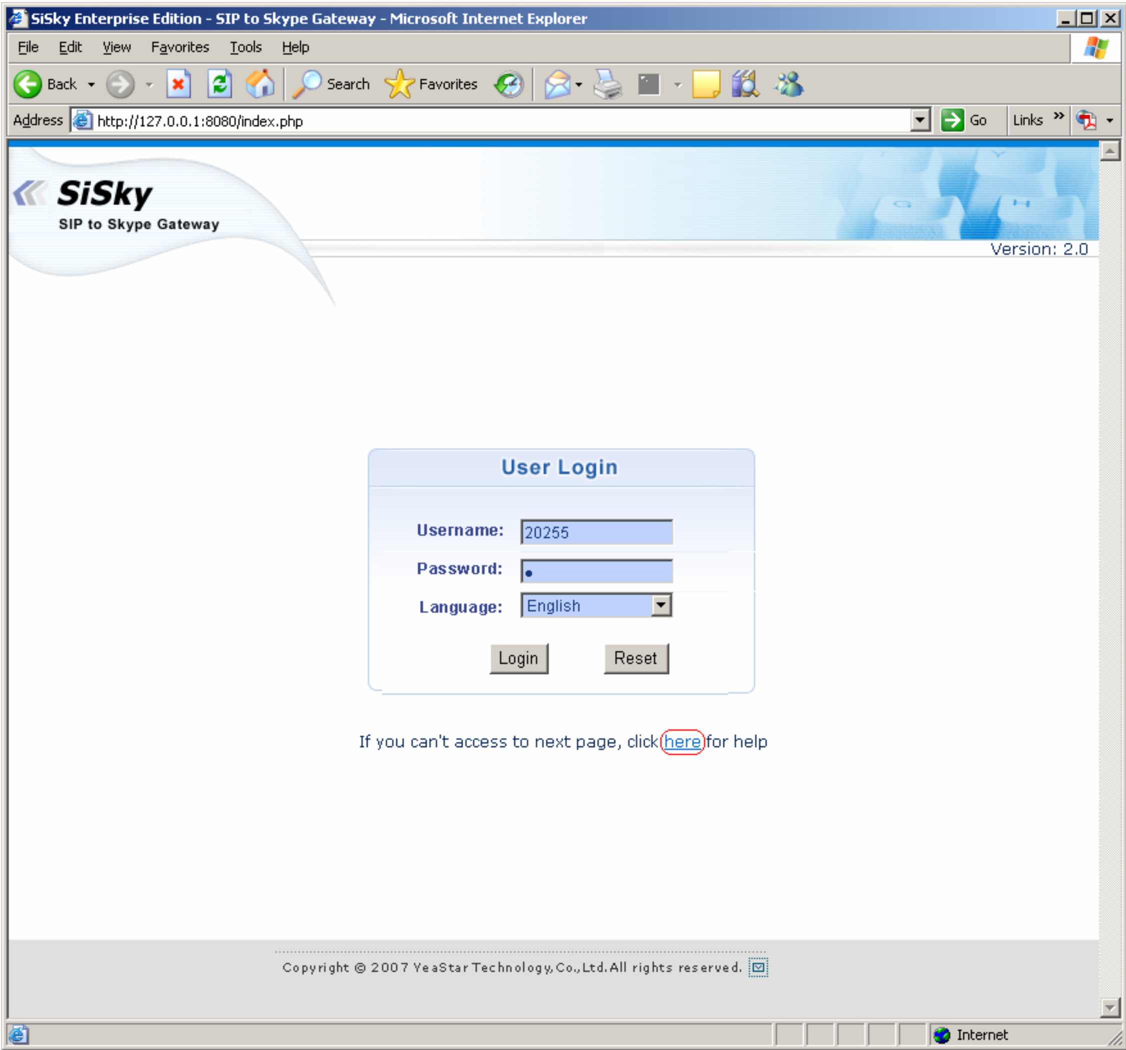
9.12 Password

You can change the password here.

9.13 Logging In As Standard Users

9.13.1 Logging In

Once the administrator enables the multi-user module and adds a user, the user can ask the administrator for his/her own PIN (Username) and password to log in to WEB interface, as below Figure 69.

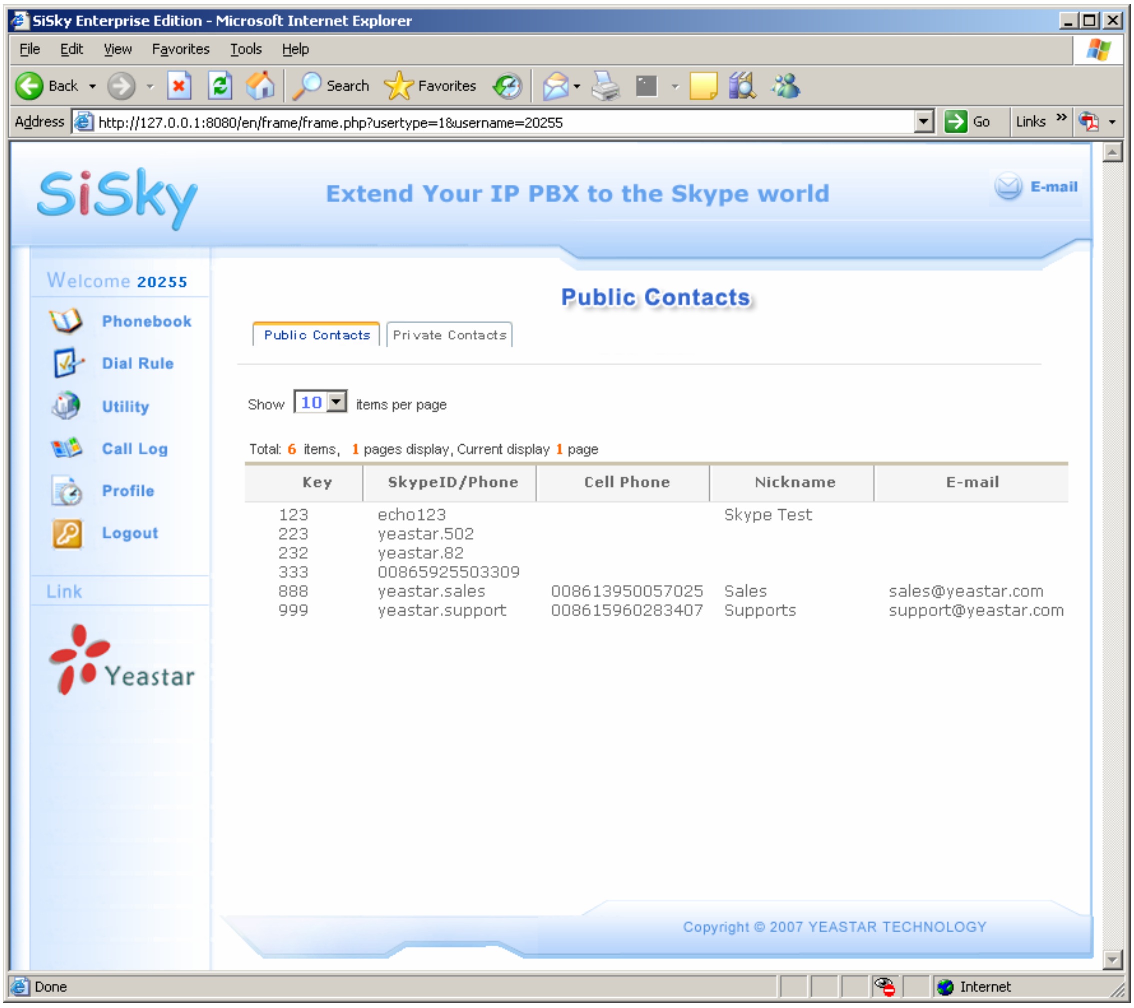


<Figure 69>

If the next page is abnormal after you login, that mainly because of the safety setting of IE browser. Click **here** on this page follow the guides to restore the normal page.

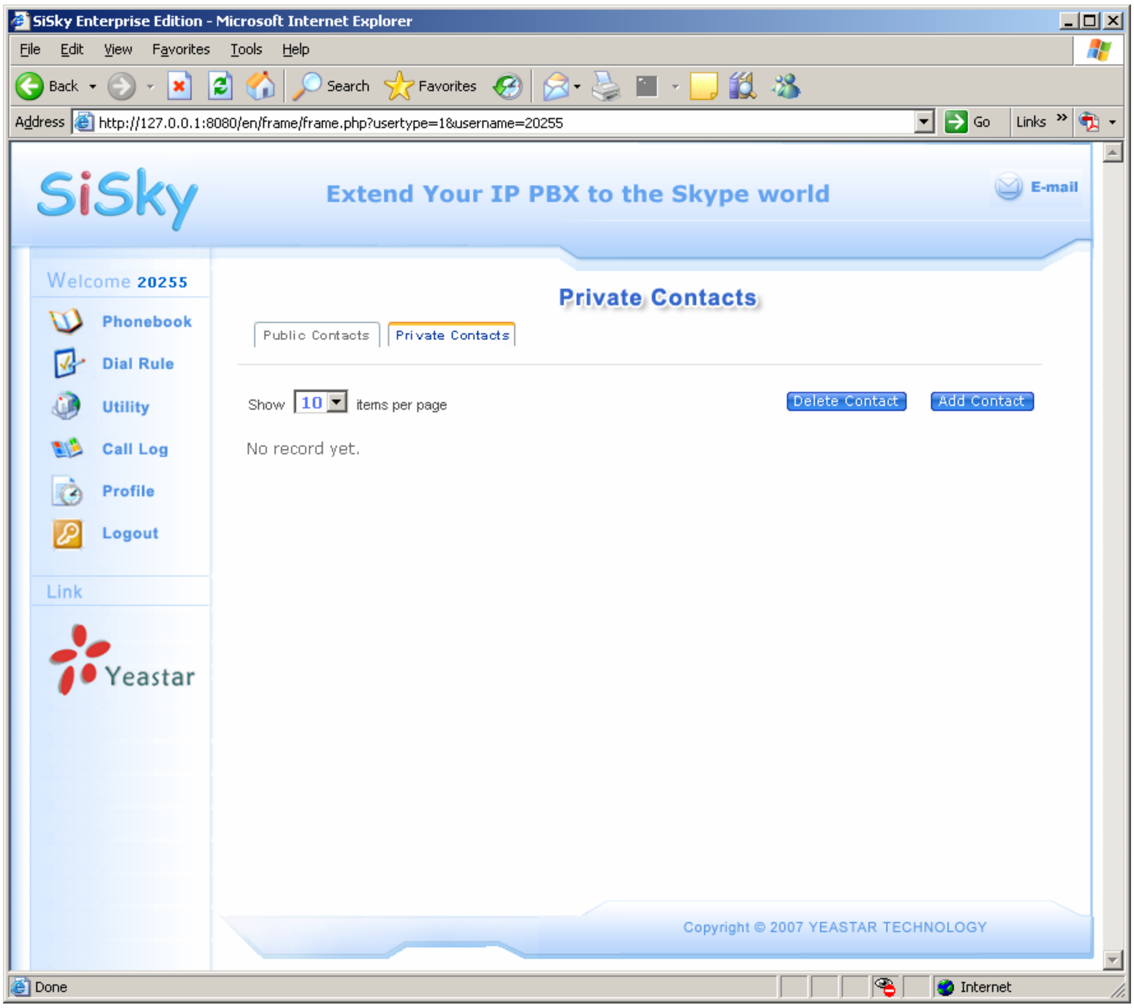
9.13.2 Phonebook

Click **Phonebook** to view the public contacts list. A standard user can only view the information but cannot modify or delete entries. See Figure 70.



<Figure 70>

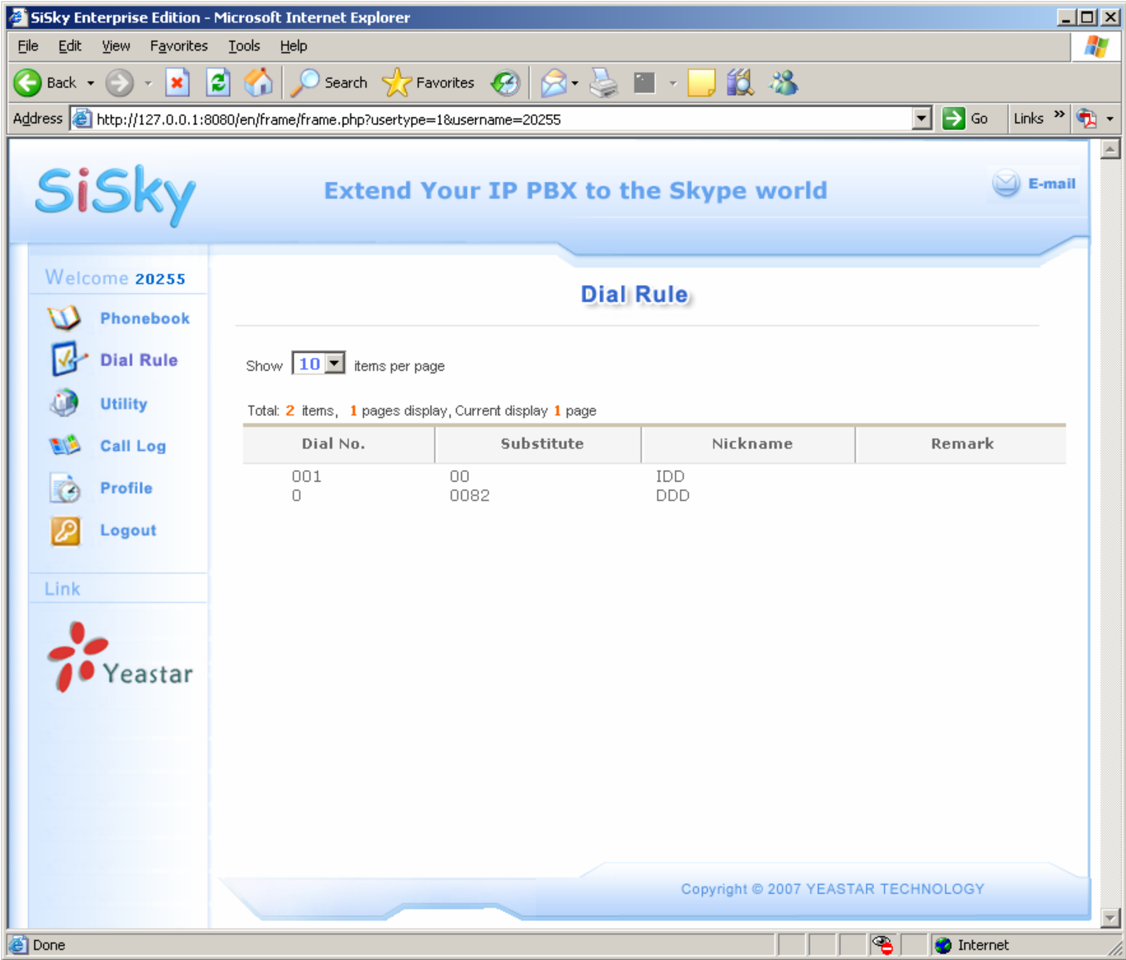
If standard user wants to view his own private contacts list, click the **Private Contacts** to add or delete contacts. See Figure 71



<Figure 71>

9.13.3 Dial Rule

Standard user has no right to modify, view only.



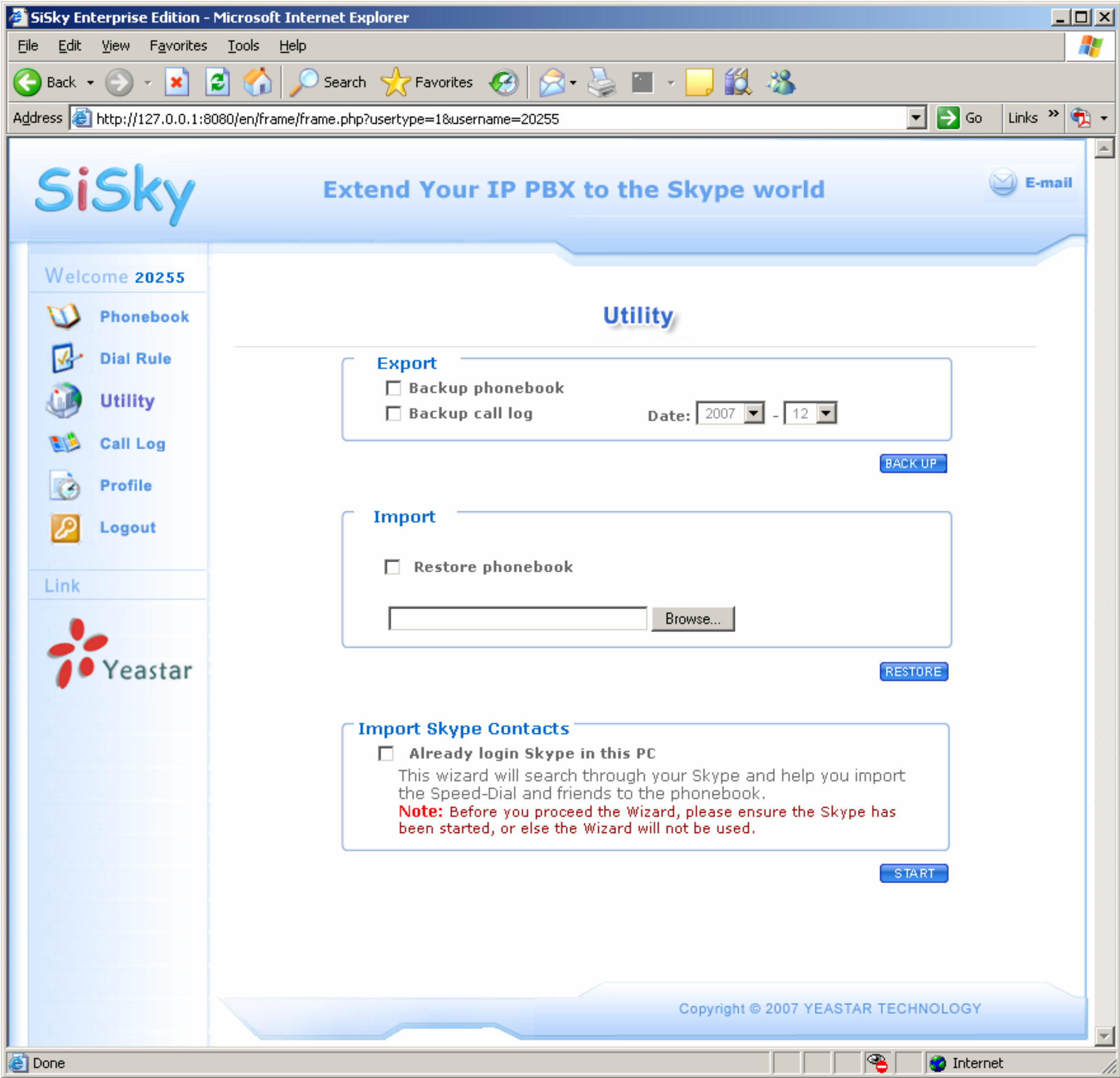
<Figure 72>

9.13.4 Backing up & Restoring

Standard user can export private phonebook and call log yearly/monthly, and he has right to restore the phonebook from backed up database.

Import Skype Contacts

Import the Skype contacts in this PC to the private phonebook, same operating procedure as **chapter 9.8.2**. Logging in as standard user, the contacts will import to private phonebook rather than public phonebook.



<Figure 73>

9.13.5 Viewing Private Call Log

9.13.6 Profile

You can change the login password here.

Using SiSky 10

Make three examples for the below typical applications

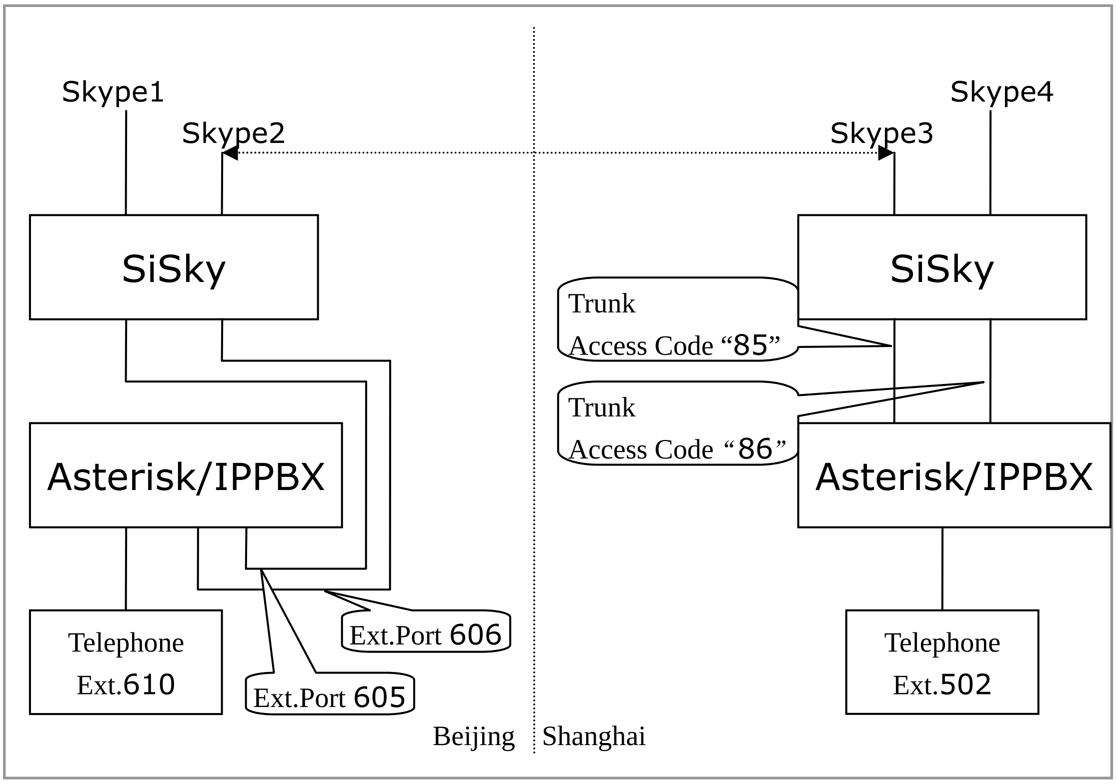
- Connections between branch offices
- Dialing SkypeOut or Calling Skype ID
- Website Click-to-Call (SkypeMe) and receving SkypeIn calls

All the below demonstration are found on SiSky connected to IPPBX system.

All the telephone numbers are overed by "#" in order to quick up the callout, which is not necessary.

10.1 Application of Branch Offices Connection

Through different configuration, SiSky port can work as Asterisk'/IPPBX's Trunk or Extension. For example, in Beijing branch SiSky port work as Extension, and in Shanghai branch SiSky port work as Trunk, as program 1.



<Program 1>

On the above demonstrated environment:

If a user sets the Skype 3 as the 'Direct Out' number for the port of Skype 2;

and also sets the Skype 2 as the 'Direct Out' number for the port of Skype 3. (chapter **9.3.2 Managing Port's Settings**)

On the **Public Phonebook of Shanghai**, user sets a Speed-dial number 111 for Skype 1; on the **Public Phonebook of Beijing**, user sets a Speed-dial number 444 for Skype 4.

10.1.1 Beijing Calling Shanghai

Make a call from Beijing Extension telephone 610 to Shanghai Extension telephone 502:

Through Skype 1:

Pick up→ 605 (hearing sound for second dial) → 444# (hearing the SiSky's music on hold and PBX's Auto Attendant) → 502

Or

pick up→ 605 444# (hearing the SiSky's music on hold and PBX's Auto Attendant) → 502

Through Skype 2:

Pick up→ 606 (hearing SiSky's music on hold and PBX's Auto Attendant)→ 502

10.1.2 Shanghai Calling Beijing

Make a call from Shanghai Extension telephon 502 to Beijing Extension telephone 610:

Through Skype 3:

Pick up→ 85**0** (hearing SiSky's Auto Attendant)→ 610

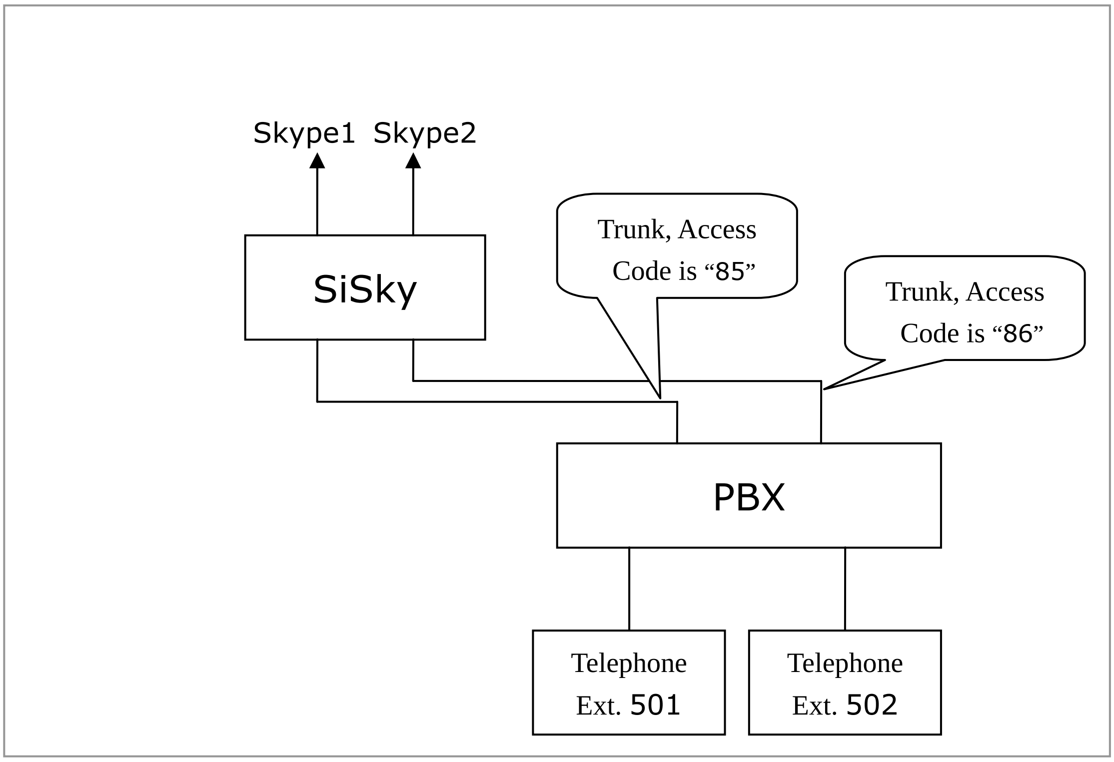
Through Skype 4:

Pick up→ 86 111# (hearing SiSky's Auto Attendant)→ 610

10.2 Dialing SkypeOut or Calling Skype ID

Through different configuration, SiSky port can work as Asterisk'/IPPBX's Trunk or Extension. The usages are little different.

10.2.1 Usages of Working as PBX's Trunk



<Program 2>

On the above demonstrated environment:

10.2.1.1 Dialing SkypeOut Call

If the user of Ext.501 wants to make call 001312567234 through SkypeOut:

Under Non-multiuser Mode:

Through Skype1:

Pick up→85 001312567234#

Under Multi-user Mode（User's PIN is 20255）:

Through Skype1:

Pick up→85 20255(hearing sound for second dial)→ 001312567234#

Or

Pick up→85 20255 001312567234#

10.2.1.2 Dialing Skype ID Call

If the user of Ext.Tel.501 wants to dial Skype ID yeastar.support:

Under Non-multiuser Mode:

The speed-dial number of ID 'yeastar.support' in **Public Phonebook** is 999:

Through Skype 1:

Pick up→85 999#

Under Multi-user Mode（User's PIN is 20255）:

1. The speed-dial number of ID 'yeastar.support' in **Private Phonebook** is 999:

Through Skype 1:

Pick up→85 20255(hearing sound for second dial) →999#

Or

Pick up→85 20255 999#

2. The speed-dial number of ID 'yeastar.support' in **Public Phonebook** is 999:

Through Skype 1:

Pick up→85 20255(hearing sound for second dial) →999#

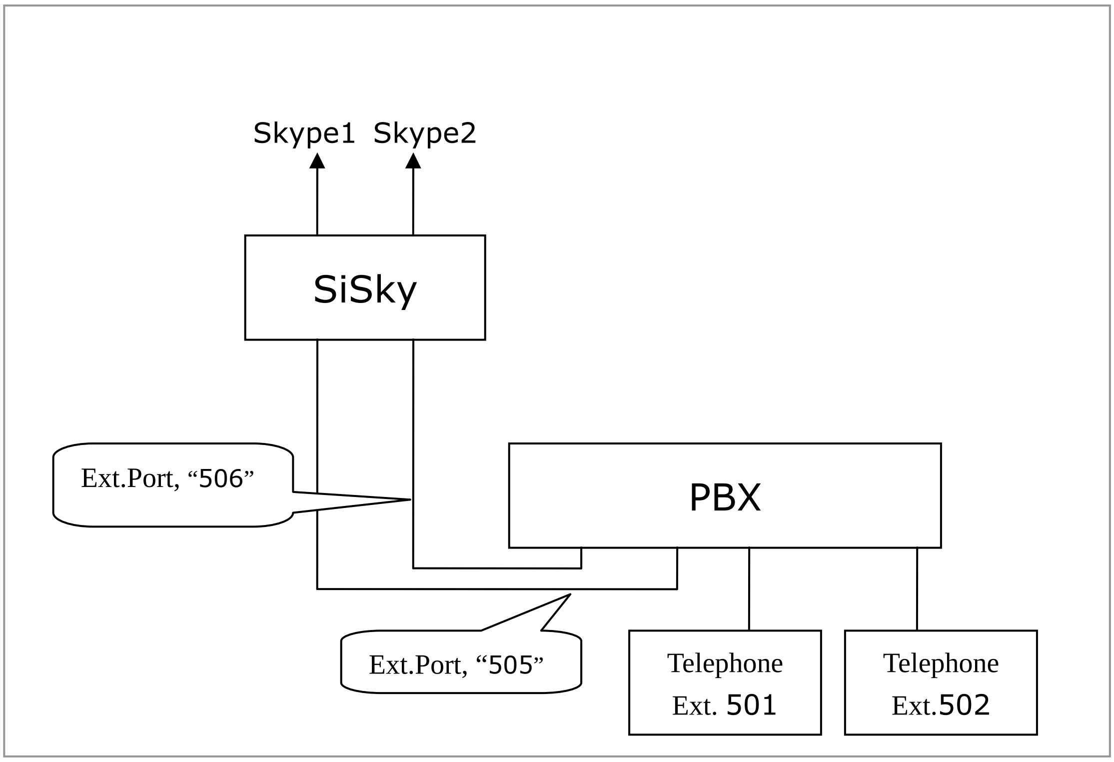
Or

Pick up→85 20255 999#

Or

Pick up→85 999#

10.2.2 Usages of working as PBX's Extension



<Program 3>

On the above demonstrated environment:

10.2.2.1 Dialing SkypeOut Call

If the user of Ext.tele.501 wants to make call 001312567234 through SkypeOut:

Under Non-multiuser Mode:

Through Skype 1:

Pick up→ 505（hearing sound for second dial）→001312567234#

Or

Pick up→ 505 001312567234#

Under Multi-user Mode（User's PIN is 20255）:

Through Skype 1:

Pick up→ 505（hearing sound for second dial）→20255 001312567234#

Or

Pick up→ 505 20255 001312567234#

10.2.2.2 Dialing Skype ID Call

If the user of Ext.Tel.501 wants to dial Skype ID yeastar.support:

Under Non-multiuser Mode:

The speed-dial number of ID 'yeastar.support' in **Public Phonebook** is 999:

Through Skype 1:

Pick up→ 505（hearing sound for second dial）→999#

Or

Pick up→ 505 999#

Under Multi-user Mode（User's PIN is 20255）:

1. The speed-dial number of ID 'yeastar.support' in **Private Phonebook** is 999:

Through Skype 1:

Pick up→505（hearing sound for second dial）→20255 999#

Or

Pick up→505 20255 999#

2. The speed-dial number of ID 'yeastar.support' in **Public Phonebook** is 999:

Through Skype 1:

Pick up →505 (hearing sound for second dial）→20255 999#

Or

Pick up →505 20255 999#

Or

Pick up →505 (hearing sound for second dial）→999#

Or

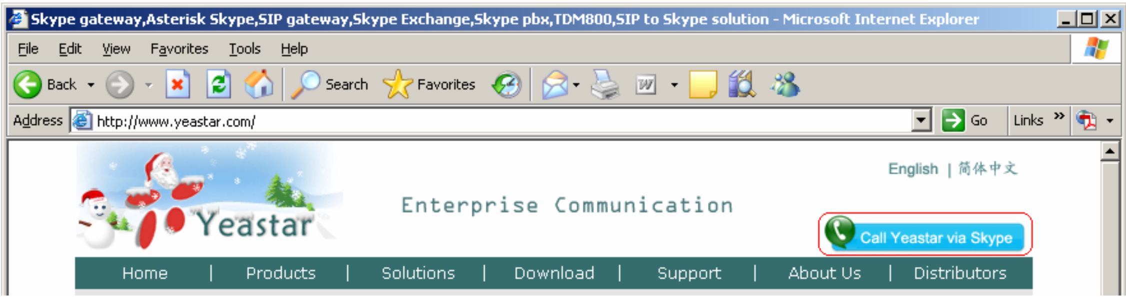
Pick up →505 999#

10.3 Usages of Website Click-to Call (SkypeMe) & SkypeIn

Using SiSky, you can make a Company Skype ID through two ways: apply an ID and public it on website directly as shown on figure 74, or apply an SkypeIn number as Company Skype ID, and then clients can make call to company through web directly and multiple concurrent calls are acceptable.

10.3.1 Showing Company Skype ID as SkypeMe on Website

It allows visitors to call the company directly through SkypeMe button.



<Figure 74>

Step 1: Allow online status to be shown on the website

- Log in Skype on Company Skype ID (yeastar.301)
- Click 'Tools' → 'Options' on menu, as Figure 75
- Click on checkbox 'Allow my status to be shown on web' as Figure 76
- Save settings

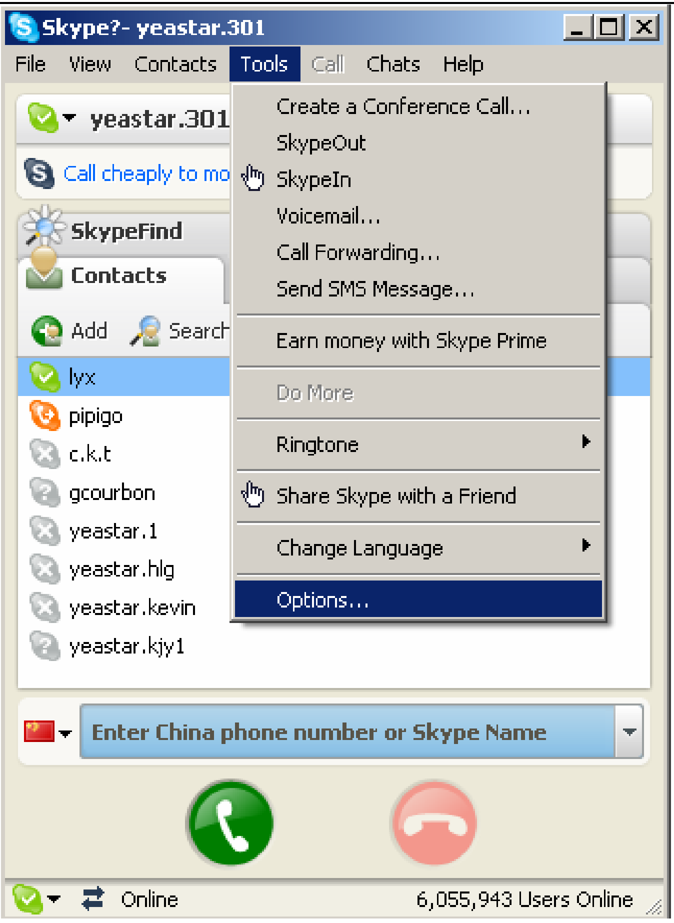
Step 2: Public SkypeMe button on the website

Please add the following html code on the relevant position of the web:

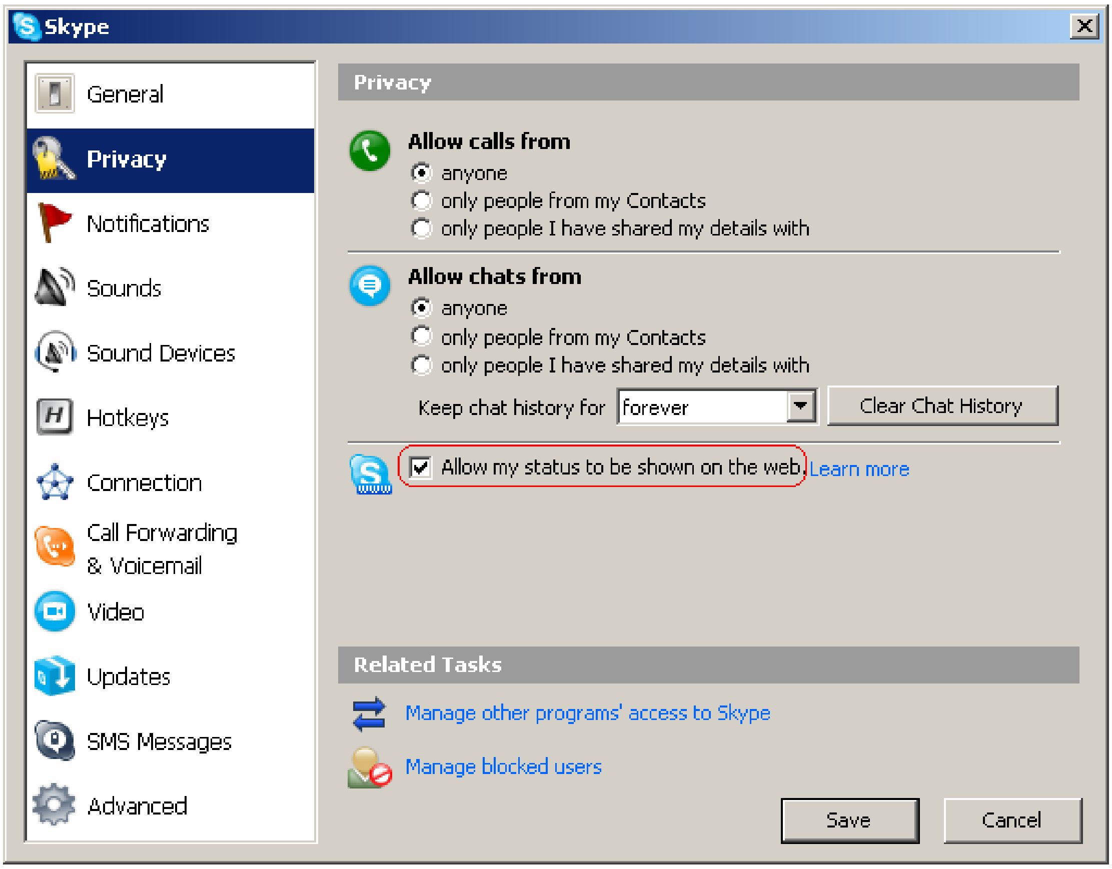
**

Here *Skype1* stands for your Company Skype ID.

Company Skype ID is complete. You are now ready to receive calls from website.



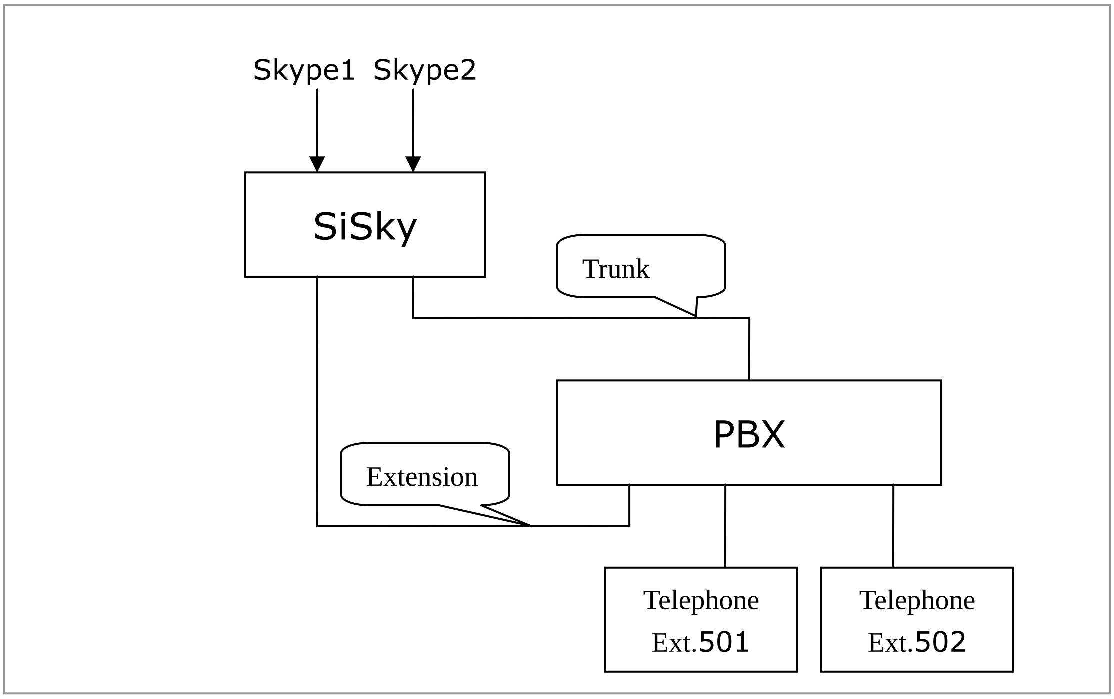
<Figure 75>



<Figure 76>

10.3.2 Skype Incoming Calls through SiSky

Through different configuration, SiSky port can work as Asterisk'/IPPBX's Trunk or Extension. The usages are little different.



<Program 4>

On the program 4, Skype 1 work as PBX's Extension, and Skype 2 work as PBX's Trunk.

When there's an incoming call to Skype 1, if there isn't Direct-In number for this port, caller will first hear the SiSky's auto attendant and then dial the extension number, auto attendant will distribute the call under rule on transferring list.

When there's an incoming call to Skype 2, caller will first hear the music on hold and then hear an auto attendant from PBX if PBX's trunk has auto attendant. Caller will continue to dial the extension number. Except the music on hold, all are same as calls from PBX's other trunk.

The call will ring the extension phone and callee just needs to pick up to answer it.

Uninstalling SiSky Software 11

Uninstalling SiSky Software will delete all your data, including phonebook, call log and recorded voice prompt recordings on this PC. Before you uninstall SiSky, make sure you have backed up the necessary data.

Method 1:

1. Go to the Window's **Start** menu and open SiSky Uninstall, click **Yes** on the pop-up confirmation window 'Do you want to uninstall SiSky'.
2. The next prompt window will guide you to **Close** the current running SiSky program first. Choose '**Stop**' to abandon uninstallation; 'Cancel' to continue the uninstallation program.
3. The Uninstallation process will delete the SiSky files from your computer. When the uninstallation process is complete, the **Uninstall Finished** screen will pop up. Click **Finish** button.

Method 2:

1. Go to the Window's Start menu and open the **Control Panel**.
2. Go to the Add or Remove Programs, which will open a new window with a list of software programs installed on the computer. Scroll down to **SiSky** and click **Change/Remove** button.

Appendix A Customizing WAV Format（Auto Attendant）

Here, users can customize a WAV file through Windows accessories **recorder**, or through other software to finish the format, like CoolEdit.

(WAV Format: 8,000 kHz,16 Bit,mono)

Step:

Start menu→ **Program**→ **Attachment** → **Entertainment**→ **Recorder**

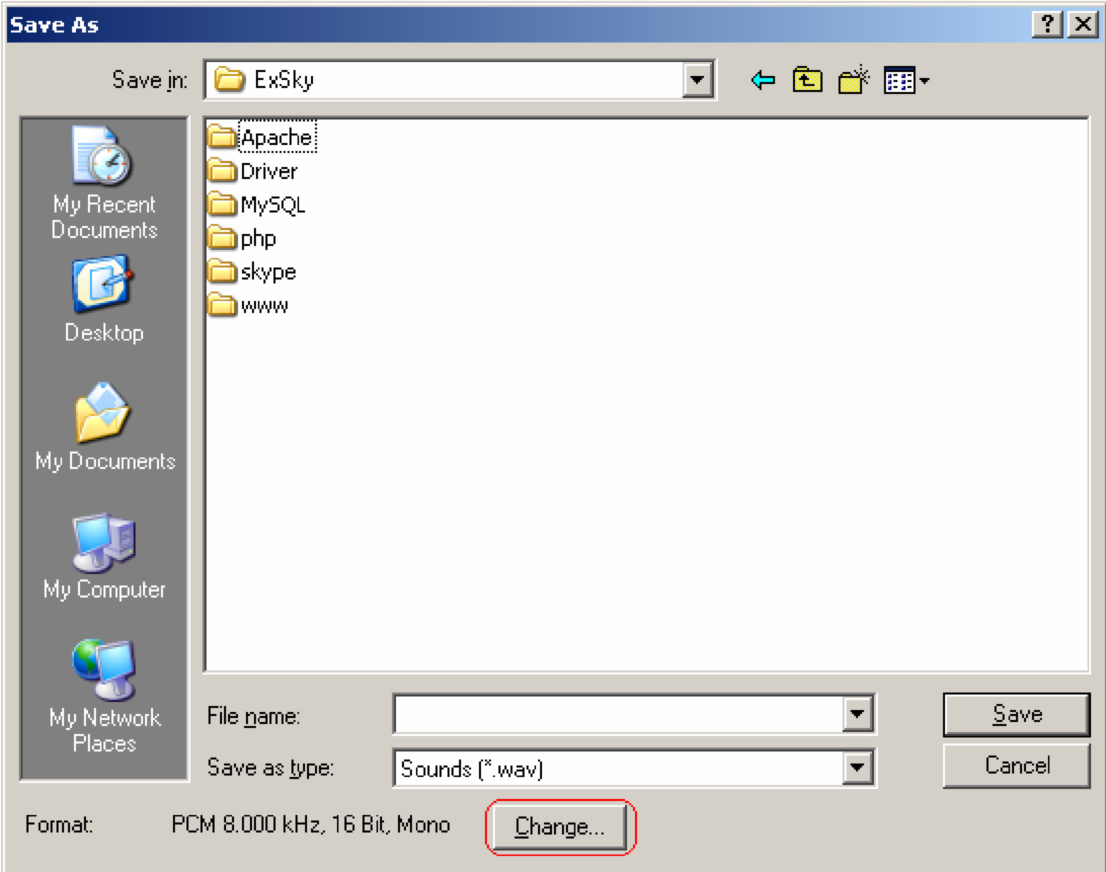
1. Record the prompt message file

<Figure 77>

2. Save it As

<Figure 78>

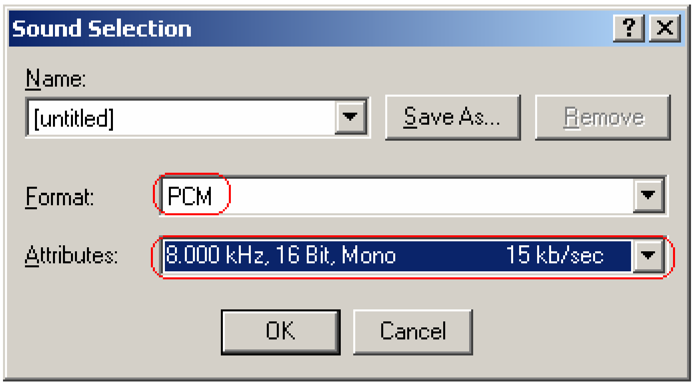
3. Change the format, click the button as Figure 79



<Figure 79>

4. Choose an entry on 'Sound Selection'

Format: PCM
Attributes: 8,000 kHz,16 Bit, Mono 15KB/sec



<Figure 80>

5. Click 'OK' to save the settings and voice files.

Appendix B FAQ

A. Skype Issues

1. **Q: What is the bandwidth for each Skype Call?**
   A: The bandwidth of one Skype Call depends on Internet access bandwidth and PC's CPU performances of the both sides of the conversation. According to statistics provided by the official website of Skype, each Skype Call requires an average bandwidth of 3Kb/s-16Kb/s, please refer to http://www.skype.com/help/faq/technical.html

2. **Q: What is SkypeOut**?
   A: SkypeOut allows you to place calls to regular telephones (landlines or mobiles) all over the world for a low fee.

3. **Q: What is SkypeIn?**
   A: SkypeIn allows you to own a regular phone number. Thus your contacts can call your Skype account on a regular phone.

4. **Q: Can SiSky work with SkypeIn?**
   A: SiSky can work with SkypeIn, especially SkypeIn in some areas can support concurrent multiple incoming calls.

5. **Q: What is SkypeMe?**
   A: Please refer to Figure 81. You can make a SkypeMe button on the web or email and allow visitors to view the online status so as to click the button to call the relevant Skype account.

<Figure 81>

6. **Q: How can I set my Skype status always 'Online'?**
   A: You need to do nothing for it. SiSky will set it for you automatically and Skype of all trunks will be kept 'Online' status always.

B. SiSky Issues

1. **Q: How many Skype trunks can be installed in one PC?**
   A: You can install 30 Skype trunks and let them all in calls at the same time, But you should make sure that the PC meets the corresponding requirement.

2. **Q: How can I do if I need more than 30 Skype trunks?**
   A: There's no limitness of SiSky trunks. You can install SiSky in more than one computer and link them together; trunks will be sharable among computers.

3. **Q: Why the Skype can't be started automatically when I running SiSky Installation Wizard?**
   A: Main reasons maybe are:
   1. this PC is Domain Controller.
   2.this PC has minimum password length requirement (no zero) .
   3. Services in the **Manage and Applications**, the **Secondary Logon** service haven't been started.

4. **Q: What is the function of the one DOS program that launches after completing the installation of SiSky software?**
   A: The DOS program is used for Web/SQL services. DO not close it.

5. **Q: Why can't I call my Skype contacts from the extension?**
   A: Before you start using SiSky, login to the Web Management console to configure your phonebook and add your Skype contacts and speed dial key.

6. **Q: Why can't I make SkypeOut calls?**
   A: Main reasons are: 1. have not purchased SkypeOut credit for the relevant trunk's Skype ID; 2. dialed telephone numbers in error SkypeOut format, the right format is: 00+country code+ area code+ telephone number or mobile phone number; 3. the administrator has enabled the Port Password for this trunk.

If you encounter any other problems about our products, please contact us or view in the following way: http://www.yeastar.com/support/faqsisky.asp

Thank you for using SiSky. We value your comments and concerns.