

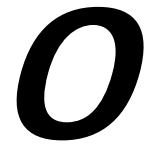


User Manual

(English Version)

Yeastar Technology Co., Ltd.

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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 SiSkyE1 and help improve its operation. Thank you for purchasing SiSkyE1!

SiSkyE1 —Business Class Skype Server

Skype is the largest global VoIP provider and it is world-famous for its high-performance of voice quality, allowing people everywhere to make unlimited voice calls for free. Skype makes use of innovative P2P technology to encrypt the calls and messages, keeping them unavailable by third party if without invitation. The safety, reliability and security of Skype win high favor of businessmen.

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 SiSkyE1 that saves the outstanding features of Skype flexibly and meets the enterprises' requirements of multiple trunks, sharable and usable at anytime. Attaching SiSkyE1 to the PBXs that with T1 or E1 digital interface, free call service will be available; enterprises will communicate and collaborate with clients and partners easily, efficiently and economically.

Who Suit SiSkyE1?

SiSkyE1 is the newest business class PBX-to-Skype gateway. It is specially aims at large company that utilizing PBX with T1 or E1 digital interface. SiSkyE1 is power to handle up 30 Skype concurrent calls at high performance.

Bringing Skype into enterprise existing PBX which has a T1 or E1 digital interface, SiSkyE1 allows all extensions to make and receive calls of Skype and PSTN over the same office phone in the comfortable way that they have got used to.

Why Choose SiSkyE1?

SiSkyE1 builds up the inter-connected channel for branch offices to make free calls and make Skype unlimited call service sharable among all extensions that obviously reduced call costs. SiSkyE1 is the best solution that upgrades your PBX to Skype PBX.

1. Skype is the largest and most popular free VoIP network in the world, which is

much more ensured and convenient in services.

2. Simply install Yeostar's YE110 hardware card on a server and connect it with PBX's T1/E1 interface, so as to get 30 Skype trunks on PBX.

3. SiSkyE1 saves all the important features of Skype:

- Take advantage of SkypeIn/SkypeOut/WebCall services
- Crystal clear voice & Strong ability to NAT traversal
- Cryptographic technique for secure communication
- Make Skype call to landlines and mobile phones at local charge

4. It allows enterprises to run multiple units of SiSkyE1 together.

5. SiSkyE1 owns the excellent voice quality because of the professional voice processing techniques developed by Yeostar people themselves.

6. The use values of SiSkyE1 breathed new life into the traditional telephone system.

SiSkyE1 Functions

- Supports multiple 30 Skype/SkypeOut lines
- Supports cascade connection of multiple SiSkyE1 servers to unlimited extend Skype trunks
- Provides unique optimal designed frame to realize global branch offices connections
- Load balancing: system will equally distribute call rate among trunks.
- Allow user to forward Private Skype ID call to his/her extension
- Allow user to take use of the phonebook to reach contacts when he is out of office
- Multi-User Mode allows users to create and manage their own contacts
- Remote control by VNC web administrations
- Call Log for all inbound and outbound calls
- Designate an extension to ring for Skype incoming calls.
- Notifies Admin and display prompts for not sufficient SkypeOut credits
- Find idle port to dial/transfer Skype calls when line is busy
- User can customize the dial plan to match his habit
- Supports Speed-dial and IVR function
- Caller ID allows users to be prepared before answering a call
- Provides noise reduction, echo cancellation and lost packets compensation techniques
- supports main countries and districts tone cadence

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:

CPU	Intel Xeon E5410 * 2
Mainboard	MCH Intel 5000V ICH Intel 6321ESB
Memory	ECC 2G * 2
Disk	SATA 80G 7200rpm
Power supply	AC Input: 100~240V, 50~60Hz, 3~6A, AC Output: 600W(max.)
Network	RJ45 (100M Ethernet)
OS	Windows XP / 2003
Application	SiSkyE1; Skype3.6
Internet Connection	Dedicated Broadband downlink \geq 2Mbps, uplink \geq 2Mbps

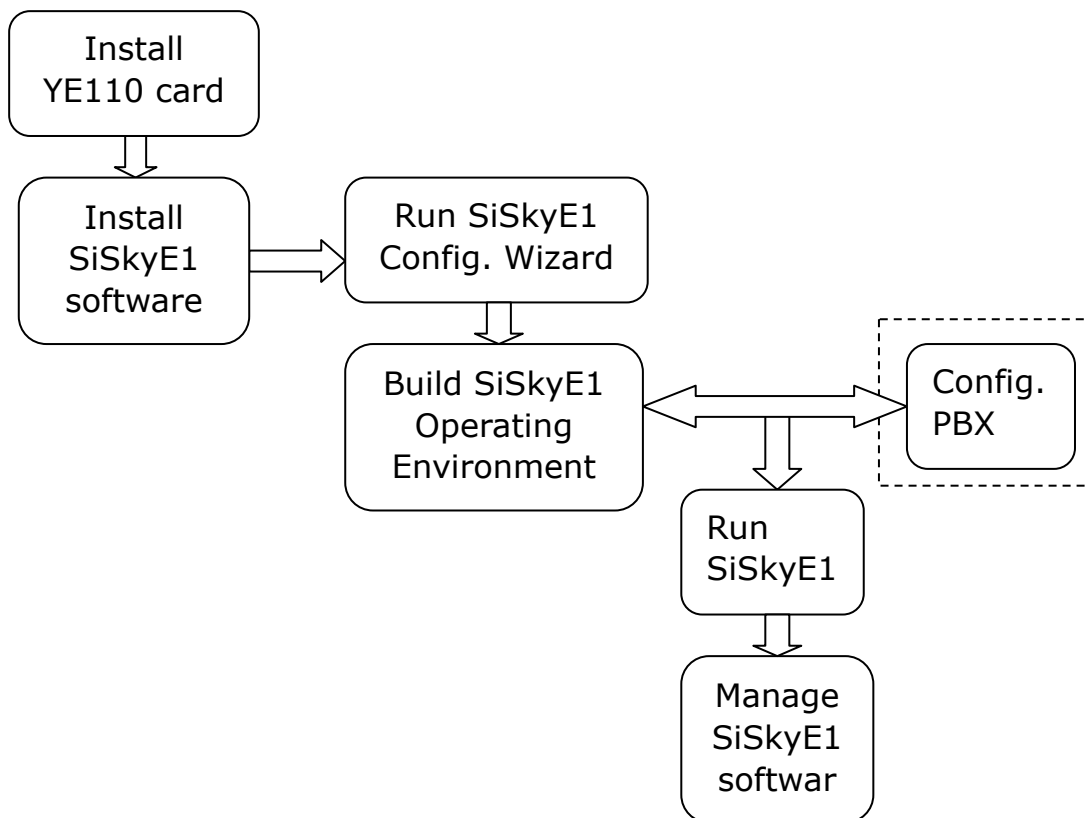
2. Install OS Windows XP Professional SP2 or Windows Server 2003 SP2.
3. Don't use the clone way to install the operating system of SiSkyE1 Server PC.
4. Make sure SiSkyE1 server PC designated for SiSkyE1 has a clean system with only anti-virus installed.
5. Make sure SiSkyE1 Server PC designated for SiSkyE1 has no Skype applications installed.
6. Make sure SiSkyE1 Server PC designated for SiSkyE1 has no domain controller installed and the disable the password complexity.
7. Make sure SiSkyE1 Server has a dedicated broadband access to ensure voice quality.
8. Make sure to login on SiSkyE1 Server with an account that has administrator privileges.
9. Please don't login SiSkyE1 server by 'Remote Desktop Connection', but use VNC in case you need remote login SiSkyE1 server.

Installation Procedure

3

Below is SiSkyE1 installation and configuration flow diagram that gives you an overview of all the steps required in installing SiSkyE1.

SiSkyE1 Installation Flow (Dotted line is option)



Installing YE110 Digital Card 4

In order to successfully install the YE110 digital card into your PC, please follow these steps carefully. Otherwise, your hardware maybe will not function properly and could become damaged.

1. YE110 digital interface card installation steps:

1. Power down your computer and open the chassis.
2. Check the jumper setting to ensure it matches your requirement. The default setting is E1 mode. If taking out the jumper, the card will be T1 mode.
3. Insert the card into a PCI slot.
4. Fix the card and close the chassis.

2. Conneting T1/E1 cable

YE110 provide digital trunk interface for connecting PBX's T1/E1 port.

Plug the RJ45 end of T1/E1 cable into YE110 card and connect the other end with coaxial cable or UTP (Unshielded Twisted Pair). If it's coaxial cable, please refer the connection on following picture.

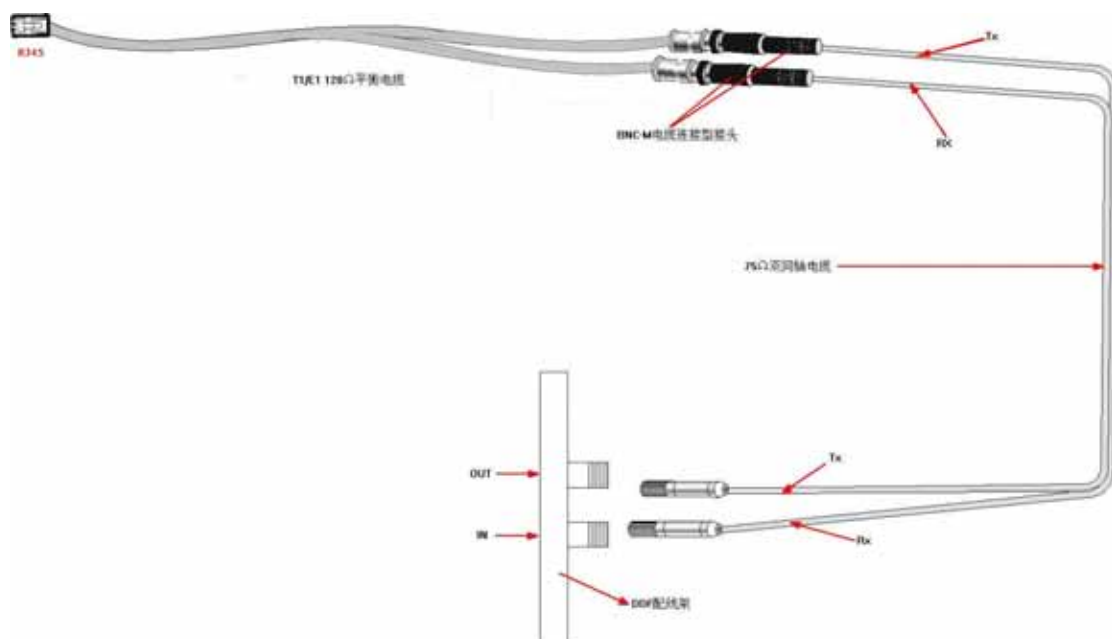


Figure 4-1 T1/E1 Cable Connection

3. Power on the PC

When Windows starts, a 'Find a New Hardware' screen will appear to remind you that you should to install SiSkyE1 driver. Please click 'Cancel' because the driver will be installed automatically on the following section **Running SiSkyE1 Config. Wizard.**

That's all for installation hardware card YE110.

How to confirm the card is installed successfully?

When continue to section 6 'Run SiSkyE1 Config. Wizard', you will see the lighted LED on YE110 card. If the wizard reads all the configurations of card correctly, it indicates the success of installation. Or else, please check if the PCI slot is in good contact. If you are unable to clear of trouble, contact your reseller.

Installing SiSkyE1 Software

5

This section shows how to install SiSkyE1 software on the PC.

Note: Before installing SiSkyE1 software, you should first uninstall Skype software on your computer if you already have Skype software installed.

1. Download the software from web <http://www.yeostar.cn/> and double click it to run the software.
2. A **Welcome to the SiSkyE1 Installation** screen will come up. Click **Next** to continue.



Figure 5-1

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

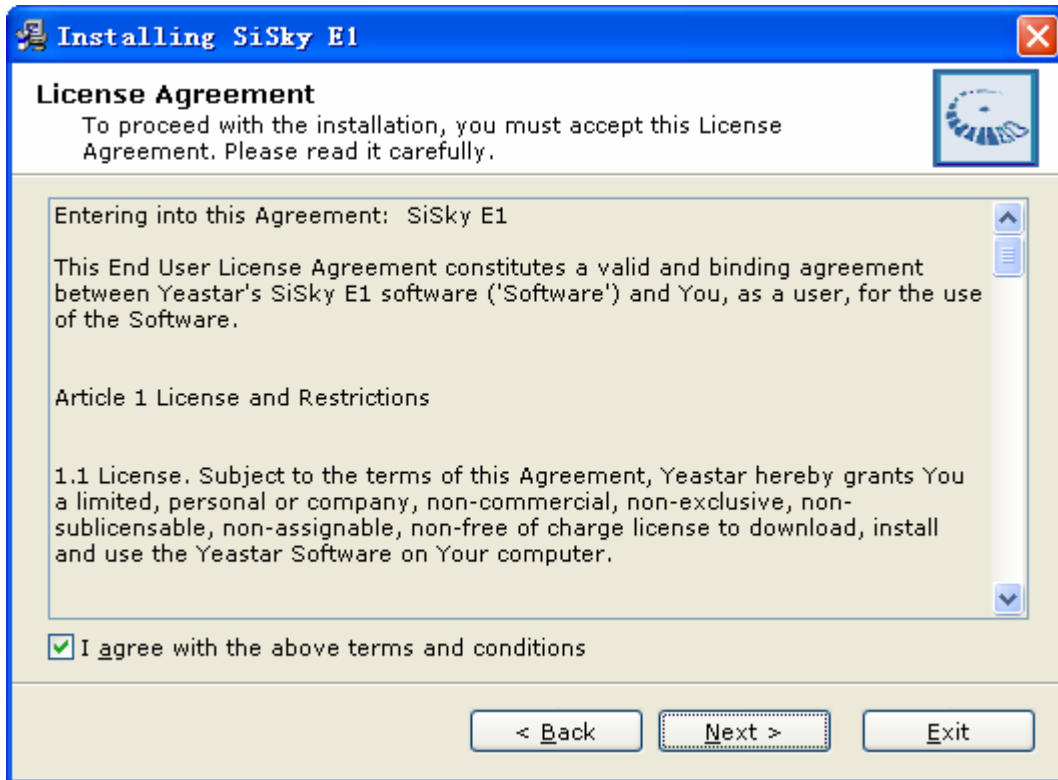


Figure 5-2

- An info window will appear to remind you that SiSkyE1 needs the dedicated digital interface hardware card YE110. You are not allowed to use the SiSkyE1 software if without the card. Click **Next** if the card is ready in PC.

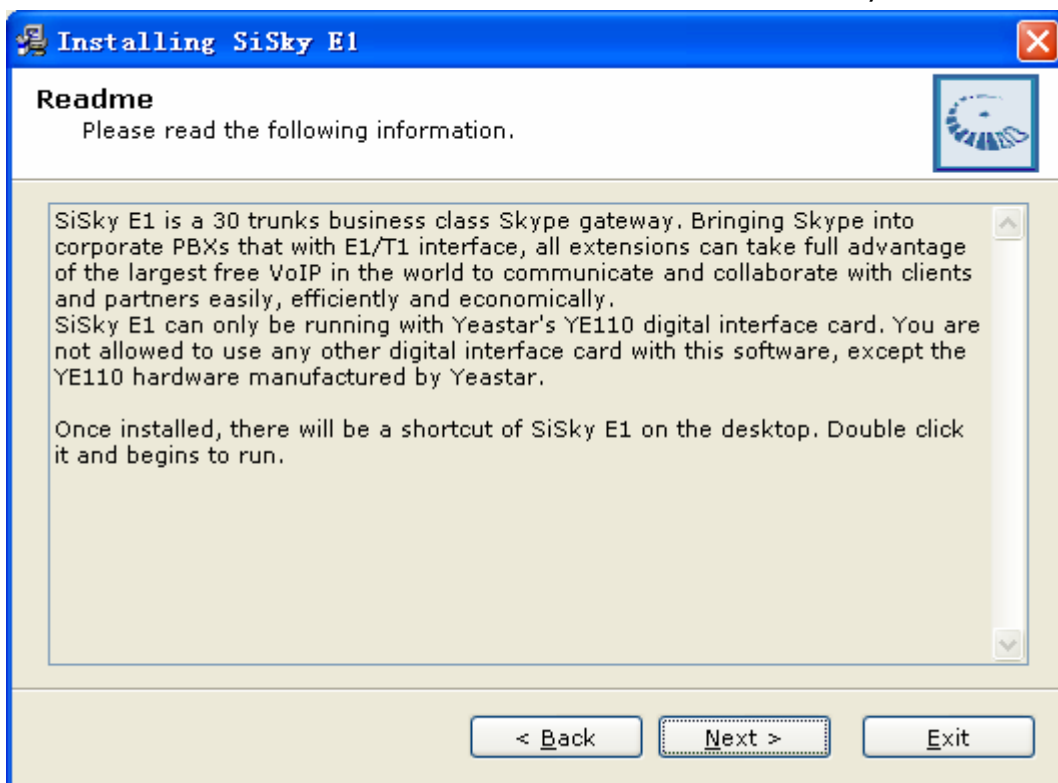


Figure 5-3

5. The Destination folder window will offer you the option where you would like SiSkyE1 to be stored on your computer. Click **Next** to continue.

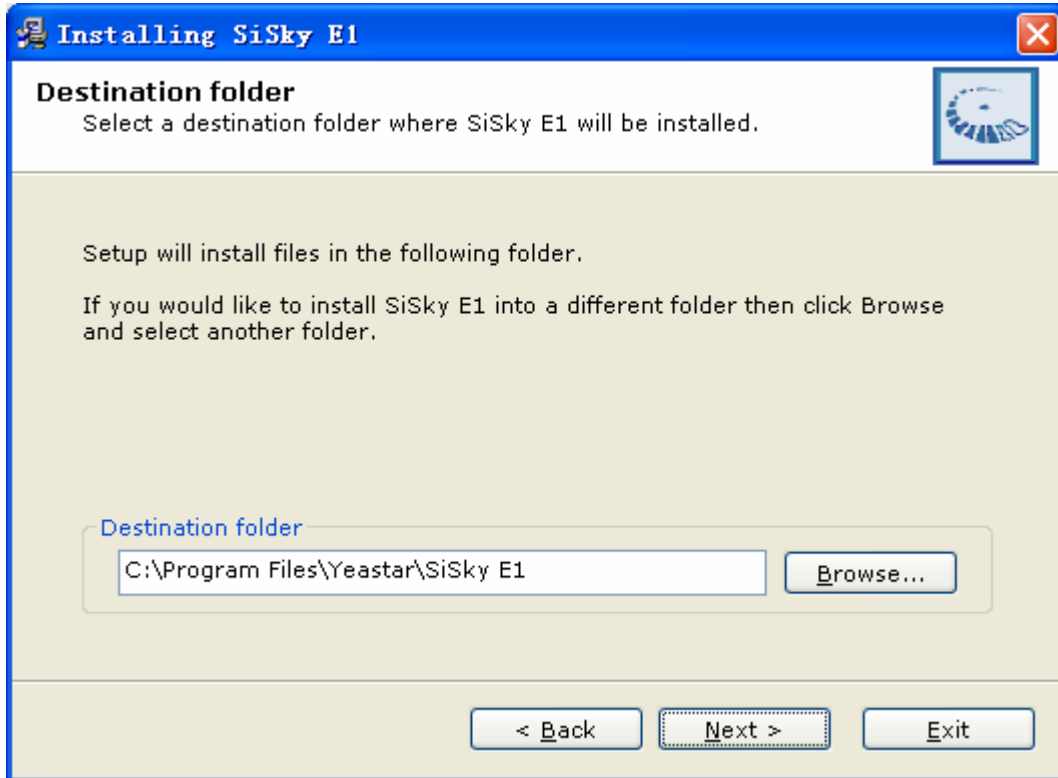


Figure 5-4

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

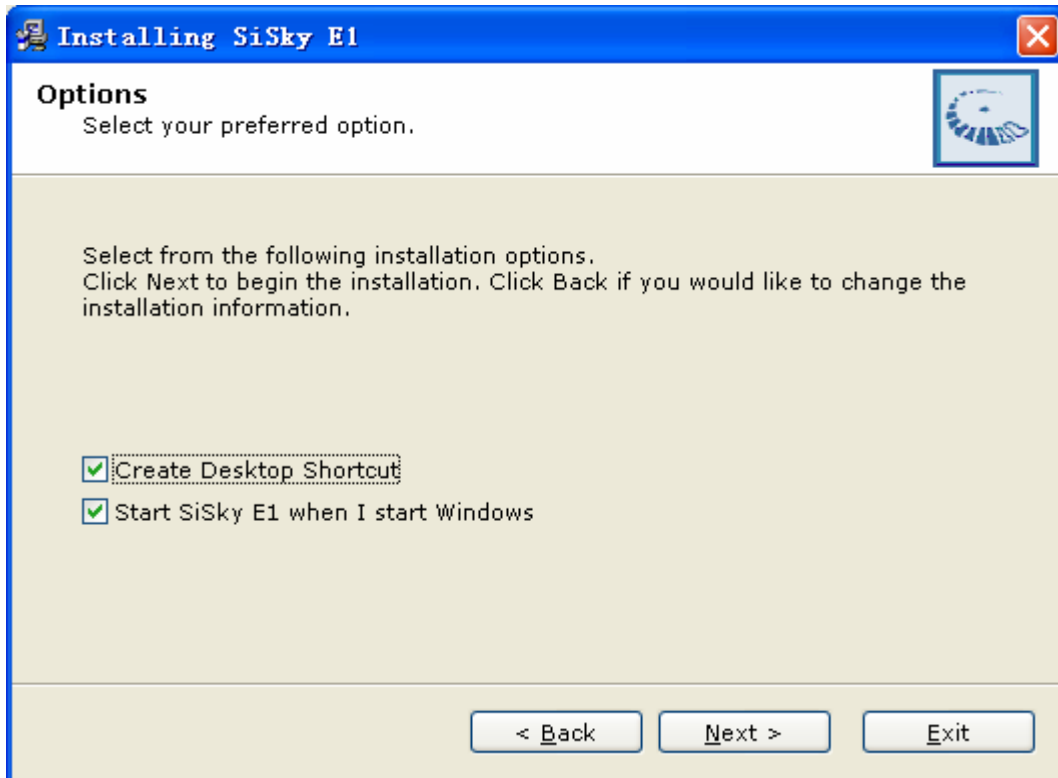


Figure 5-5

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

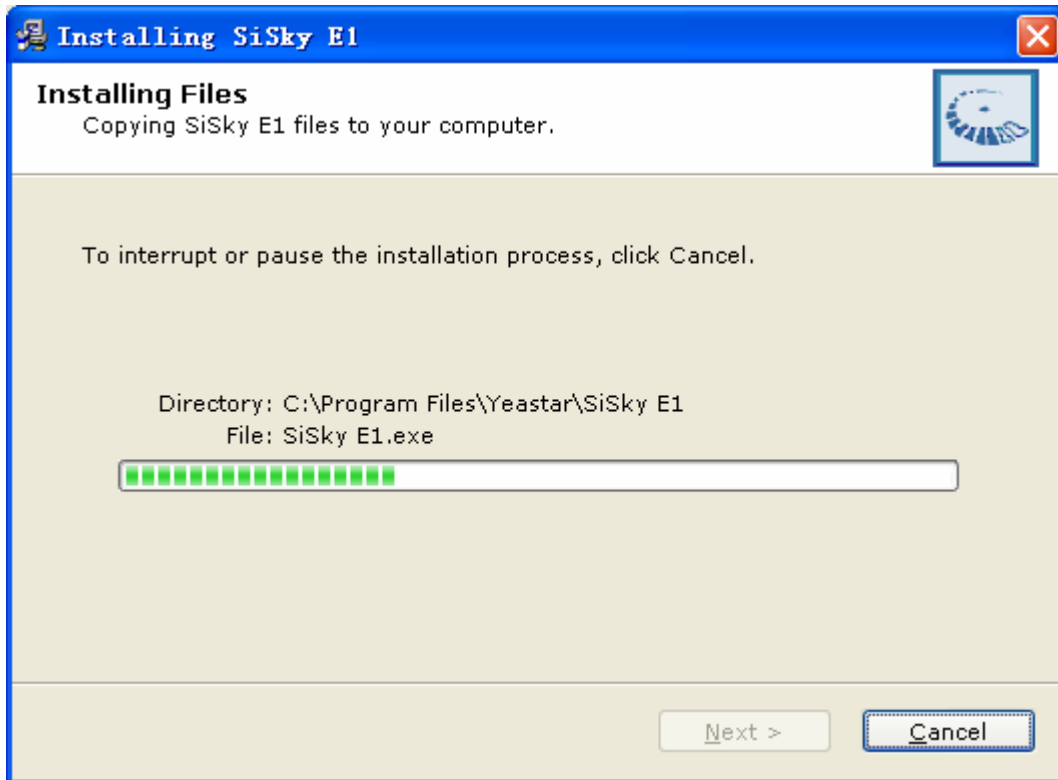


Figure 5-6

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

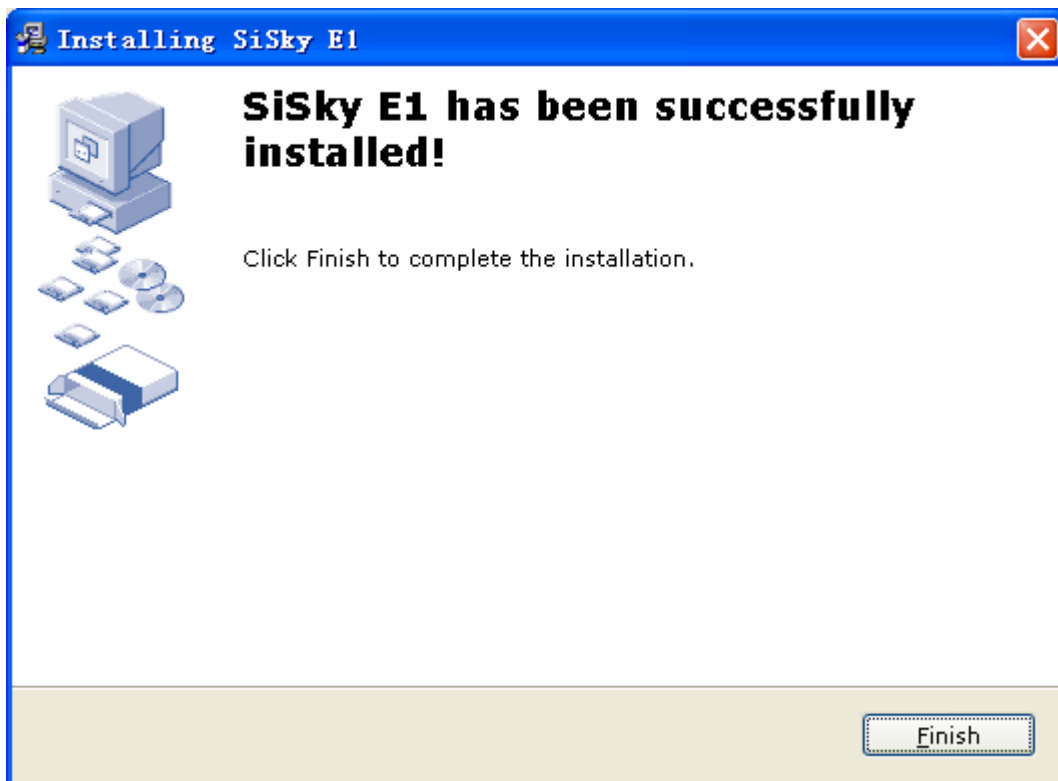


Figure 5-7

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

11. Launch SiSkyE1 to enter the next chapter.

Running SiSkyE1 Config Wizard **6**

For the first time to run SiSkyE1, a Config Notice screen will pop-up and click **Yes** to launch the Wizard.

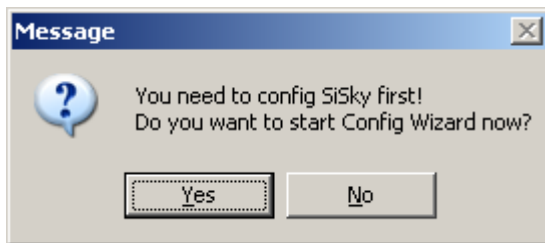


Figure 6-1

Or you can click the **Config** on SiSkyE1 to launch the wizard. If the **Config** button is invalid, please click **Stop** to stop SiSkyE1 first.

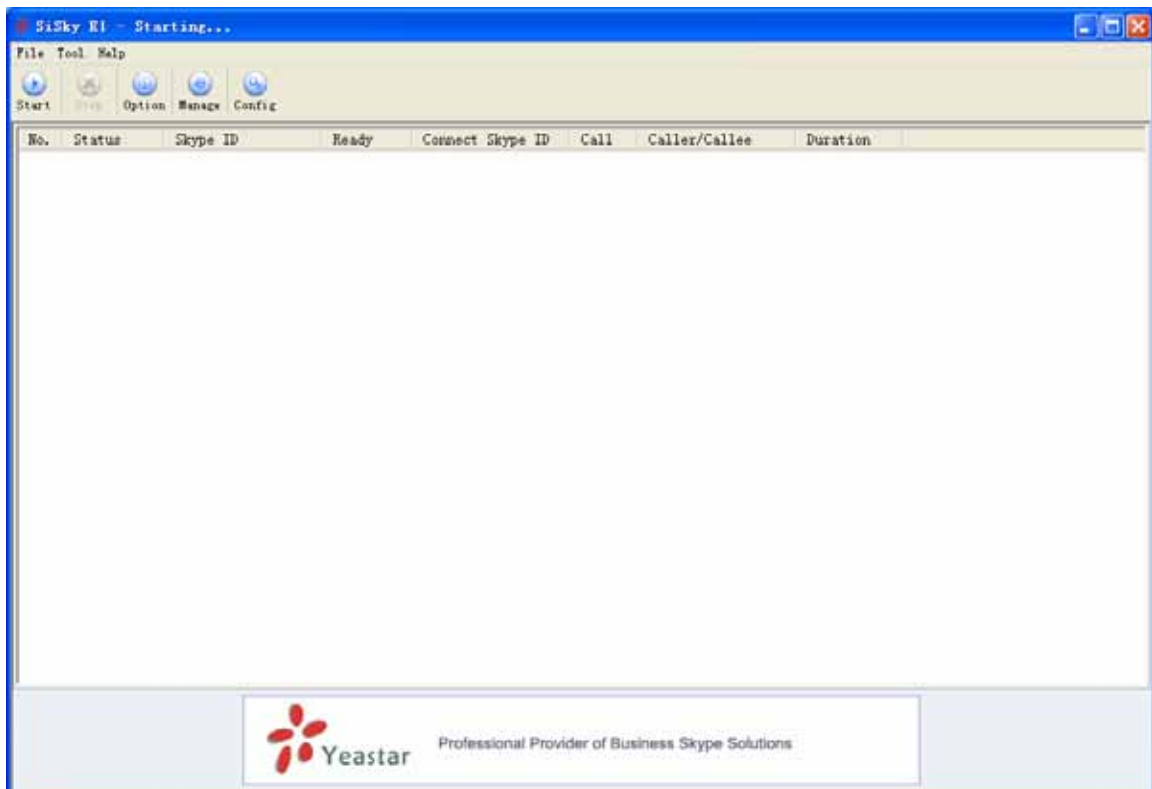


Figure 6-2

6.1 Launch the Config Wizard

Click **Next**

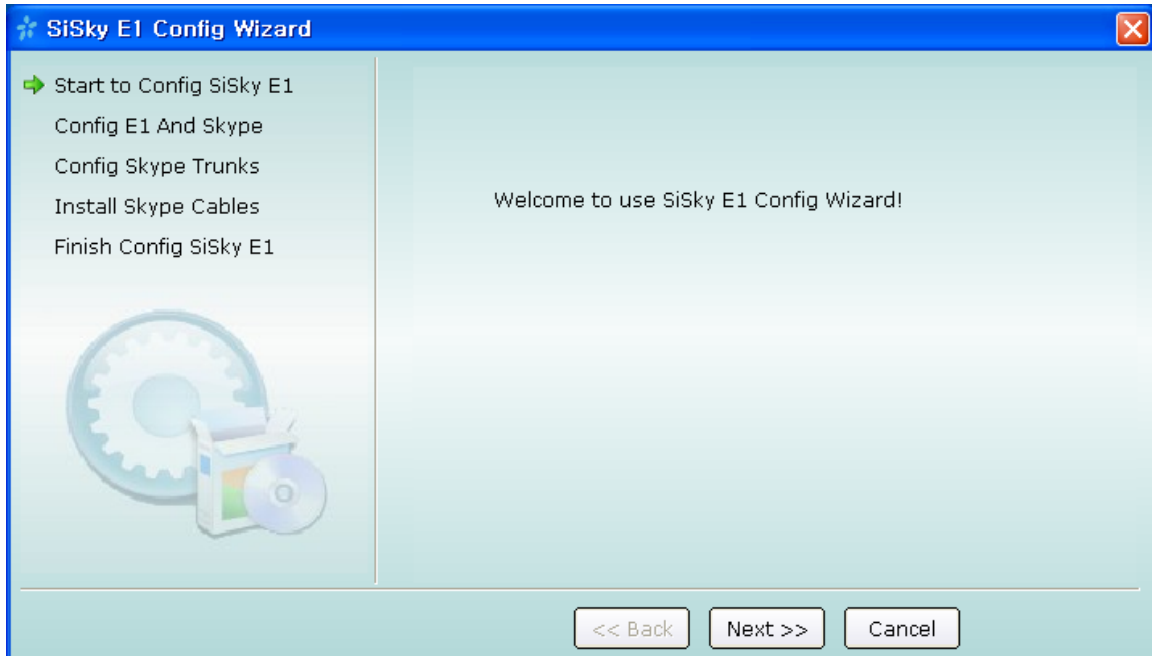


Figure 6-3

6.2 Install E1 Driver

Click 'Install E1 Driver' and select 'Skype trunks' that you want to run.

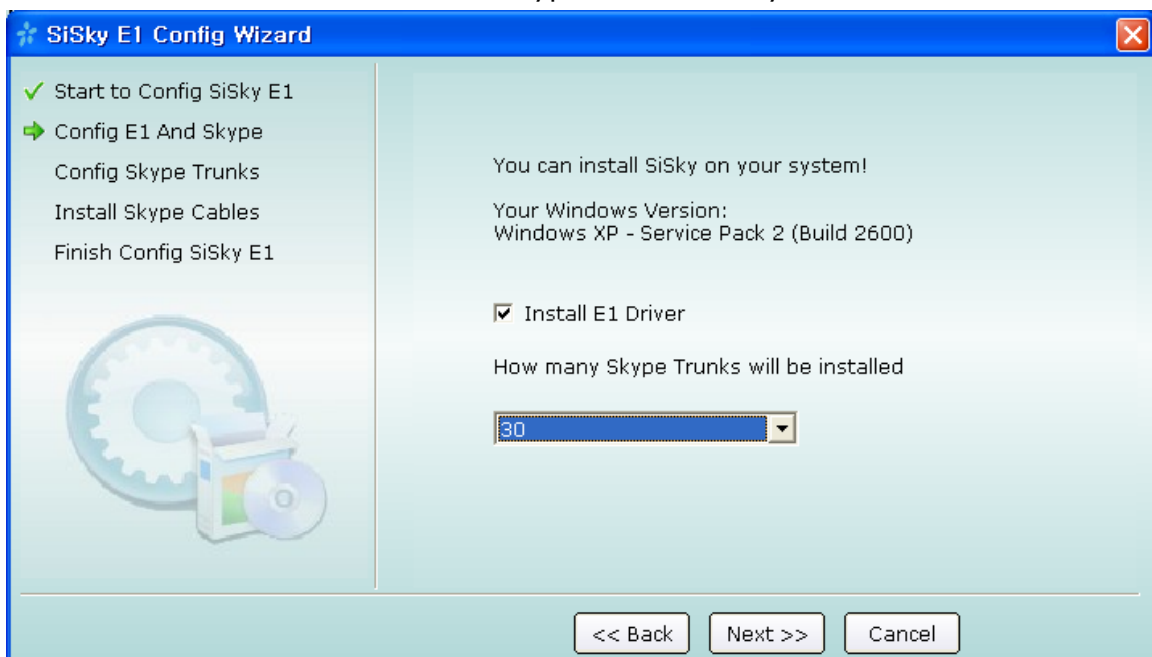


Figure 6-4

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



Figure 6-5

6.3 Configure Skype for Each Port

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

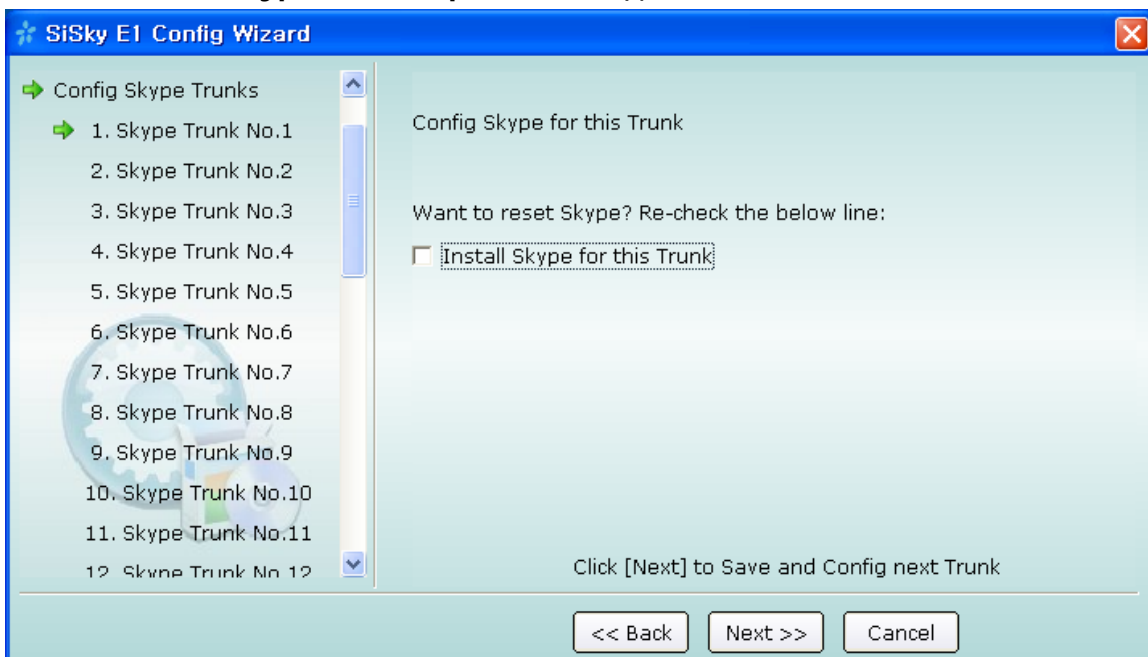


Figure 6-6

A **Skype?-Creat Account** will appear. Create a new account (see Figure 6-6) or Cancel it and log in by using an existing Skype account.

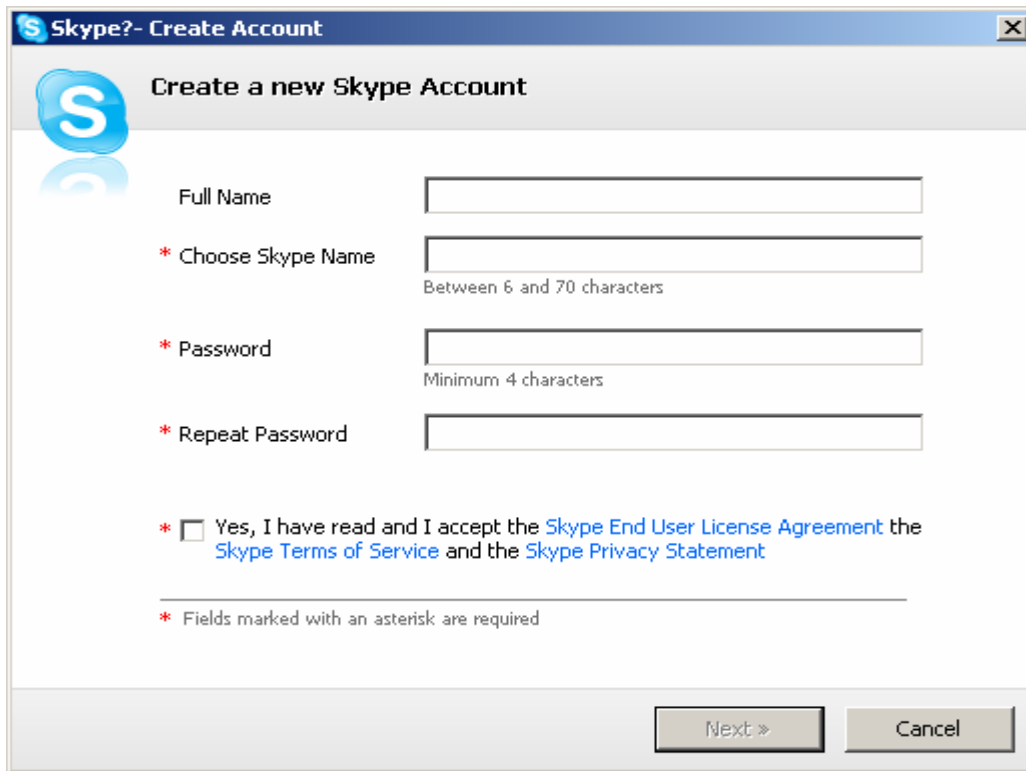


Figure 6-7

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



Figure 6-8

When SiSkyE1 get the Skype Name automatically, the configuration of Port 1 is finished. Click Next to configure other ports by the same way.

Note: Please follow step 6.3 to configure the Skype accounts for remaining ports. There will be a green tick before every port after configured all ports. See Figure 6-8.

6.4 Install Skype Cables

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

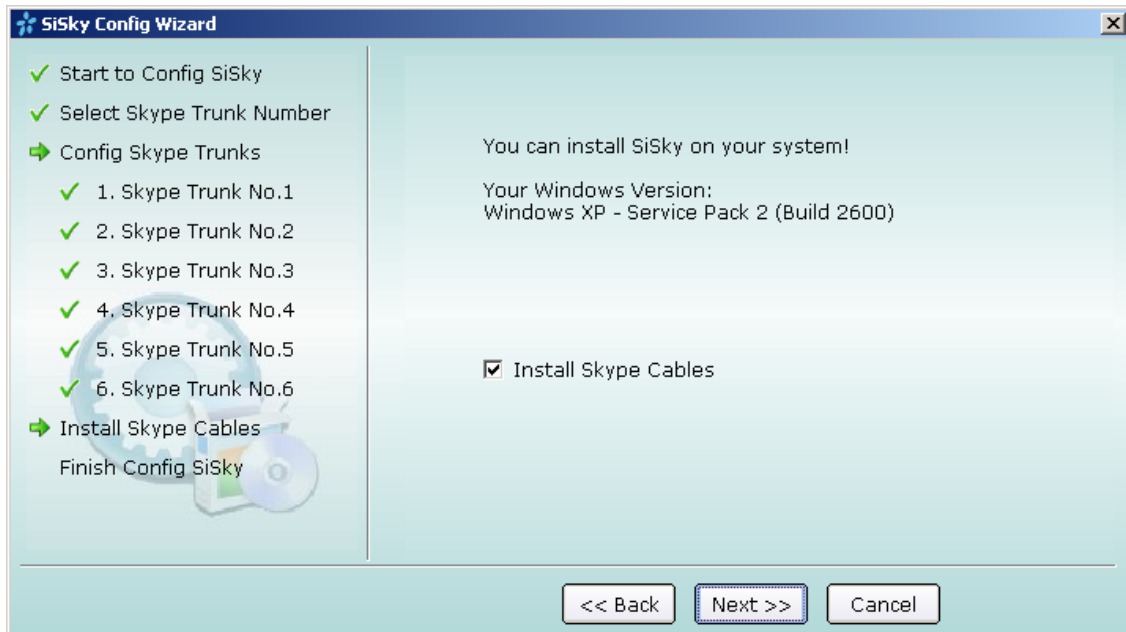


Figure 6-9

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



Figure 6-10

6.5 Finish Config Wizard

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

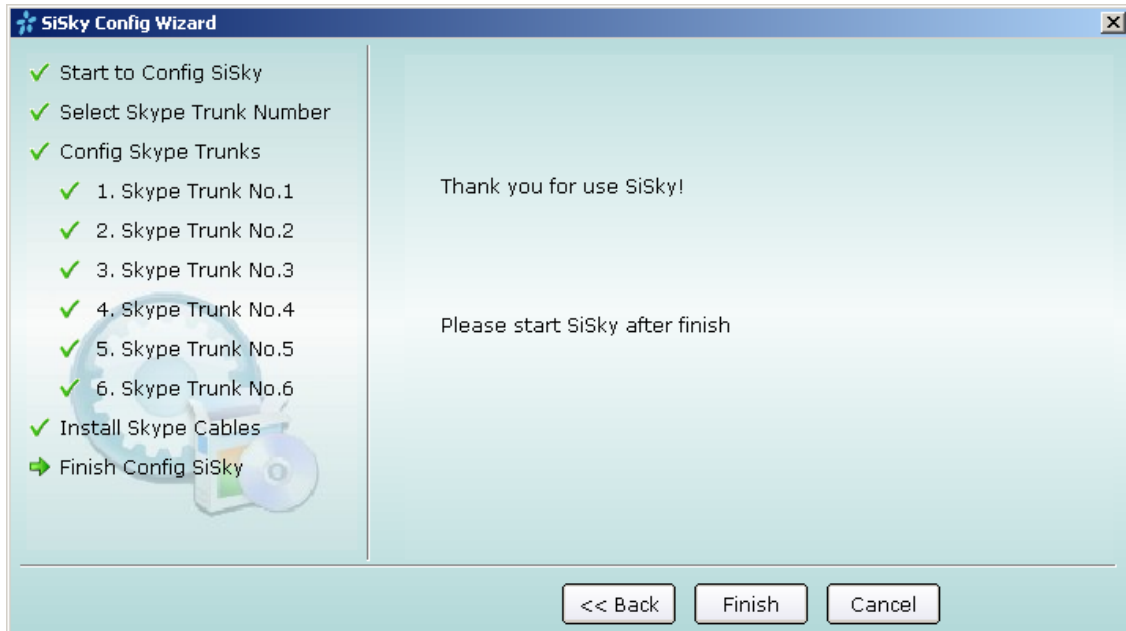


Figure 6-11

SiSkyE1 system configuration is complete!

Running SiSkyE1 Software

7

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

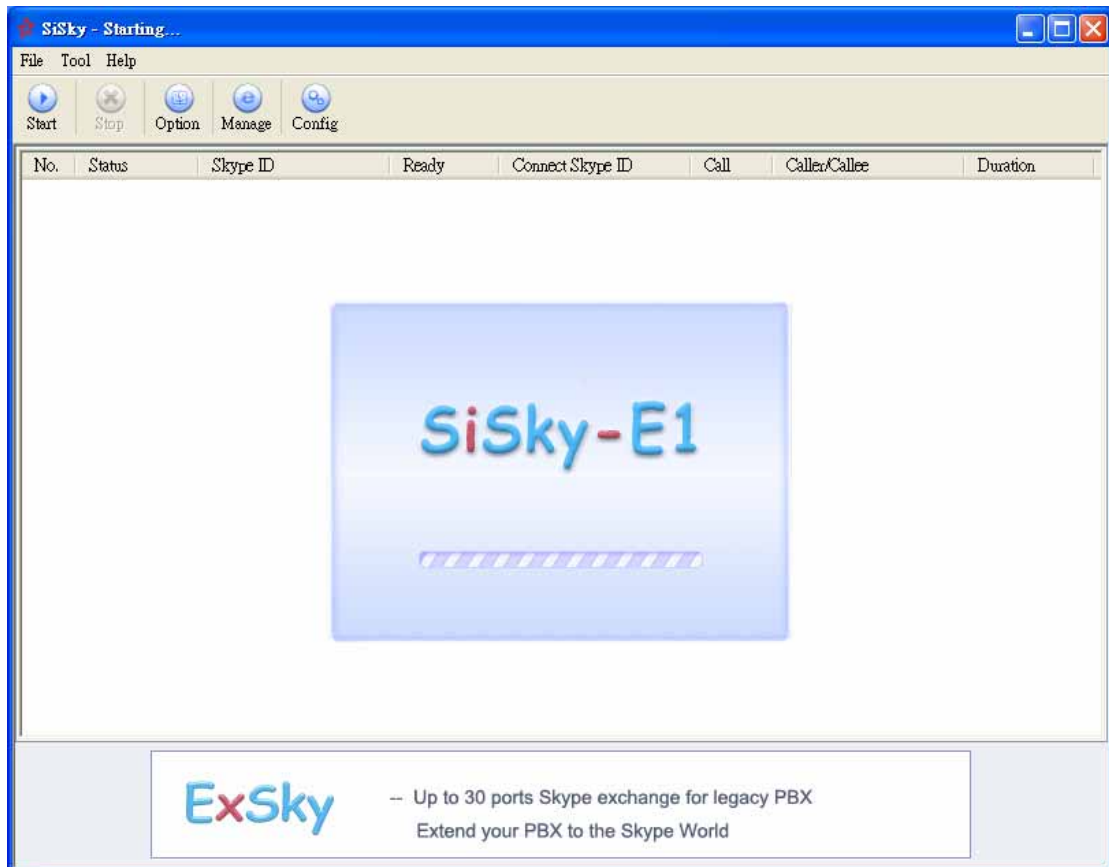


Figure 7-1

Step 2: Check if the status of all ports and Skype are absolutely normal

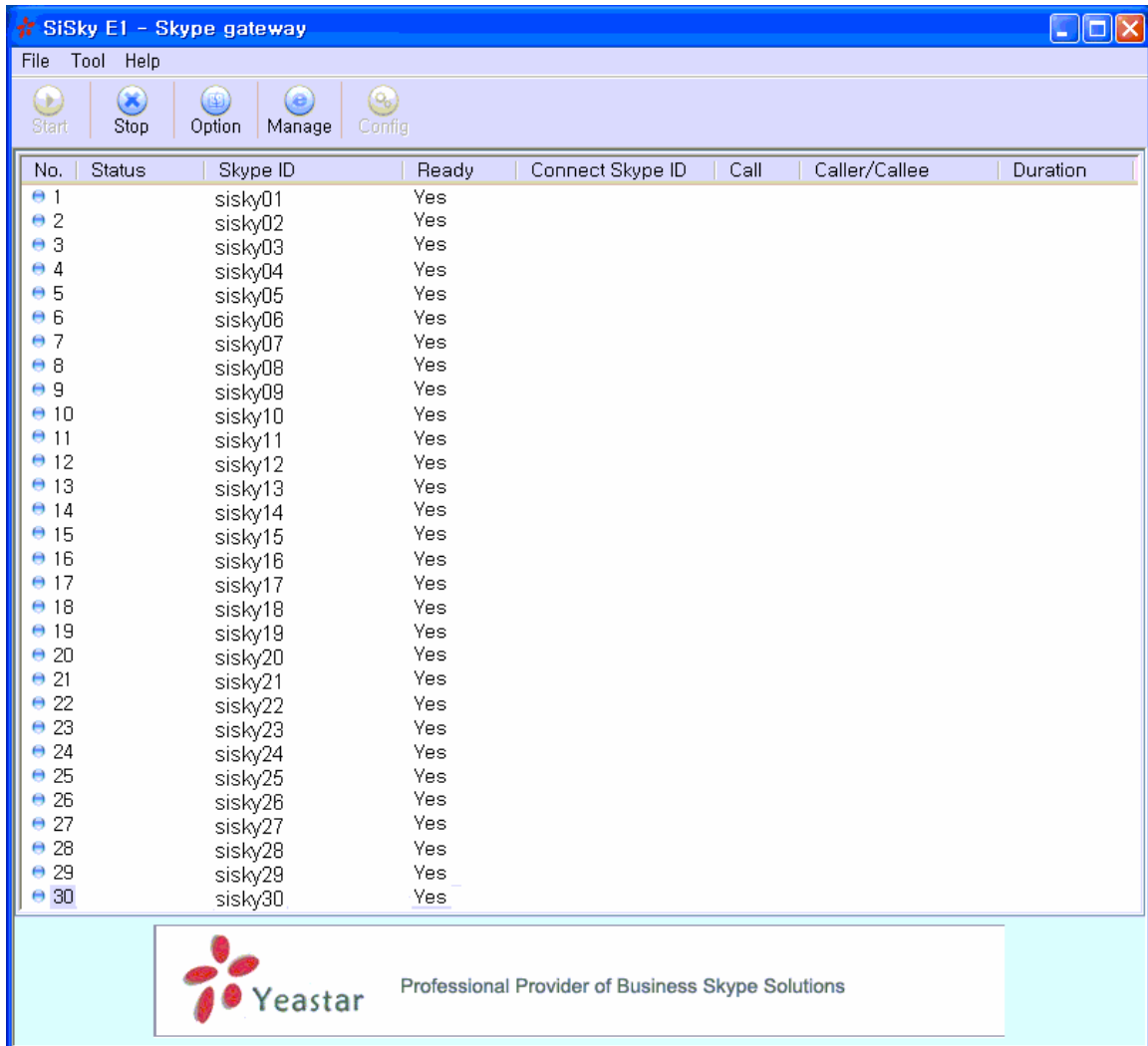


Figure 7-2

Description of all SiSkyE1 items:

No:

- 1) ●: indicates failed initialization. The port is not available or Skype is not working properly.
- 2) ●: indicates succeeded initialization. The port line is idle.
- 3) ●: indicates succeeded initialization. The port is in service.

Line Status:

- 1) N/A: The connection of E1 port is not available.
- 2) Idle: The port is idle
- 3) Busy: The port is in service

Skype ID :

Skype ID corresponds to the port number

Ready:

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

Put through SkypeID:

The SkypeID is busy in a call

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

Using SiSkyE1

8

Make three examples for the below typical applications

- Connections between branch offices
- Call to regular phone or mobile or Skype ID through SkypeOut
- Skype Incoming Calls through SiSkyE1

All the below demonstration are found on SiSkyE1 connected to PBX' s T1/E1 interface.

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

Note: This section just introduce you some basic applications of SiSkyE1. If you need more information, such as setting password for port, enable PIN number under multi-user mode, please referrer to SiSkyE1 Adadvanced Application.

8.1 Dial Rule

1) Open SiSkyE1 Homepage

If you are operating in SiSkyE1 Server, please click the Manage button or you can open the IE browser and enter in: <http://127.0.0.1:8080>

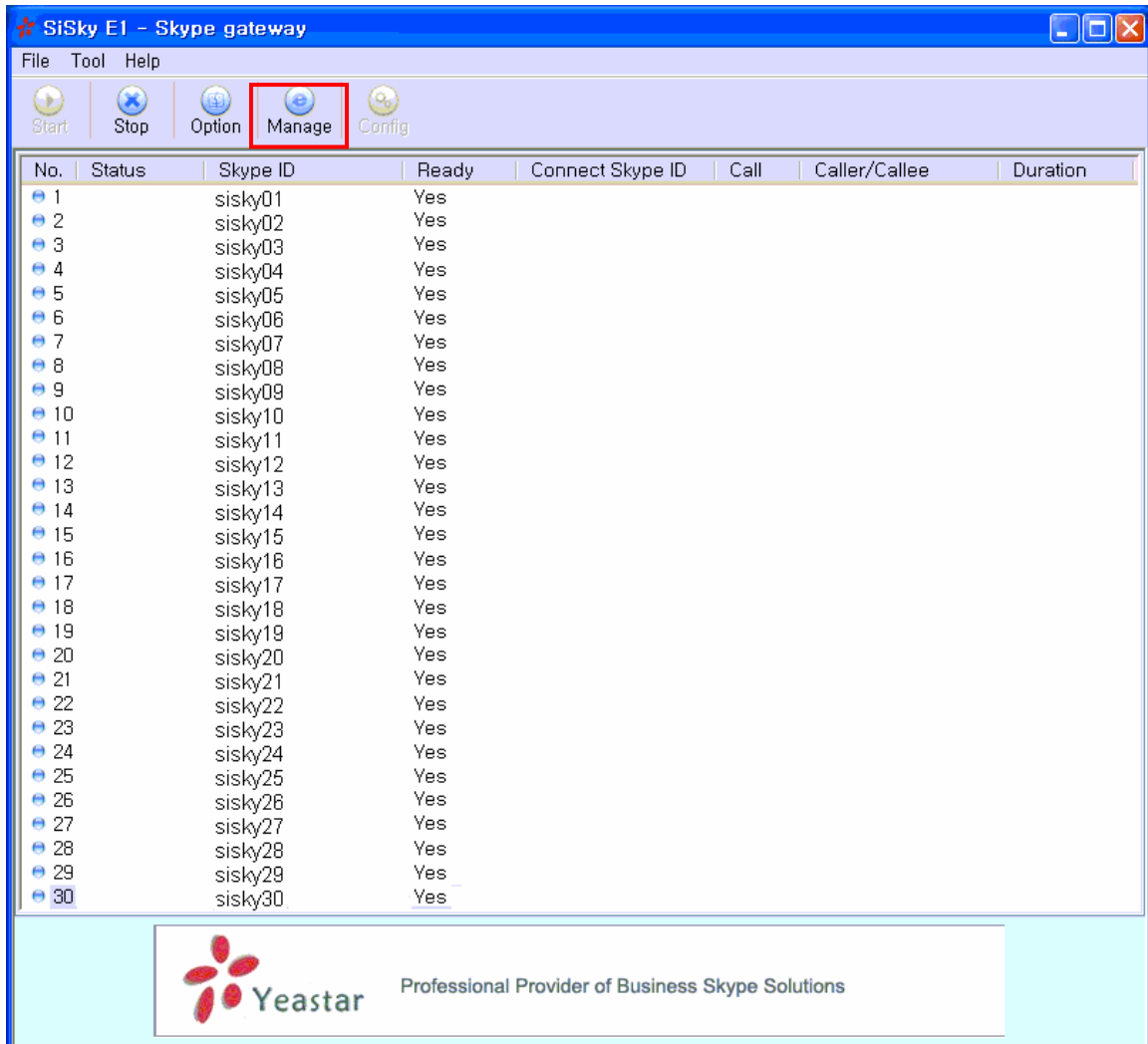


Figure 8-1

If you are not operating in SiSkyE1 Server, please open the 8080 port of SiSkyE1 Server through IE browser. For example, if the IP address of SiSkyE1 Server is 192.168.5.66, then you can type in <http://192.168.5.66:8080>

2) Administrator Logging in

Open the SiSkyE1 user login page. Input your username and password to gain access. Use the default administrator username '[admin](#)' and the default password '[password](#)' to login.

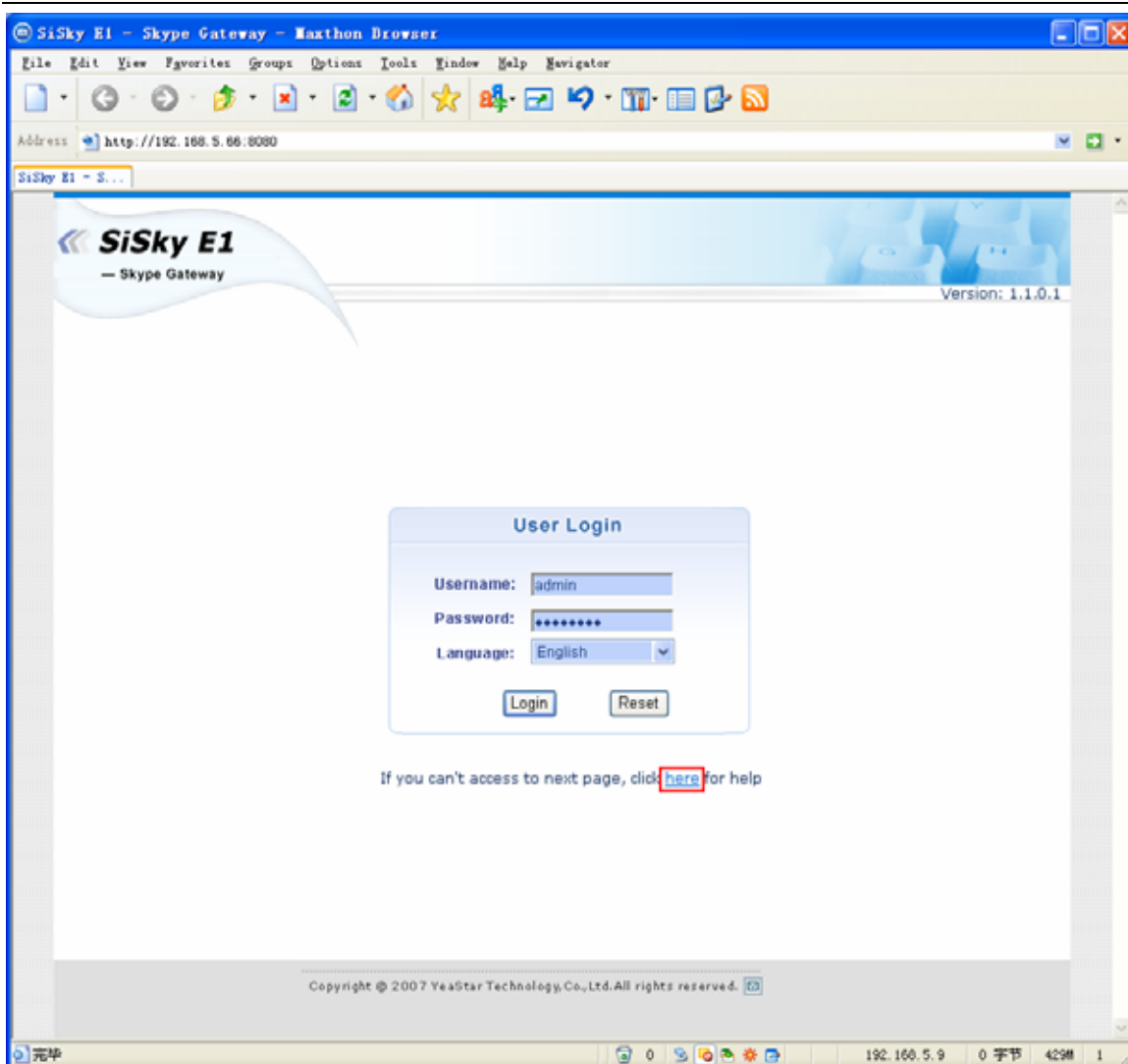


Figure 8-2

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.

3) Set Dialing Rule

Making calls through Skype, one have to conform to the Skype dialing scheme as well as calling through PSTN. Maybe you had already got used to PSTN 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.

For example, a PSTN user in the mainland of China wants to make an international call IDDD (international direct distance dialing), which format is **00+country code+area code+telephone number**; make a 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**.

An International call: for example, a user in China calls to Canada

1) in traditional PSTN format: the country code is 1, the area code is 416, phone number is "12345678", then he would dial: **00** +1+ 416 + 12345678;

2) in SkypeOut format: the country code is 1, the area code is 416, phone number is "12345678", then he would dial: **00** +1+ 416 + 12345678;

In this example, the PSTN and SkypeOur format for an international call are the same.

A Domestic Distance call: a Chinese user in Paking calls to Shenzhen

3) in traditional PSTN format: the area code is "755", phone number is "12345678", then he would dial: **0**+755+12345678;

4) in SkypeOut format: the area code is "755", phone number is "7571234", then he would dial: **00+86** (country code)+755+12345678;

In this example, the PSTN and SkypeOur format for an domestic distance call are different.

In order to not change the PSTN dialing format, this user can take advantage of the Dial Rule to place 0086 with 0, as shown on Figure 7-3

When the number begins by 00, SiSkyE1 will identify it as international call requirement. When the dialing call begins by 0, SiSkyE1 will identify the 0 as domestic call requirement and transfer it to 0086 automatically to conform SkypeOut format.

Why we substitute 00 for 00? Say it simply, to prevent 00 matches with 0. Because both the international and domestic calls are begin by 0, for fear of replacing **00**141612345678 with **0086**0141612345678, substitution rules priority are listed basipetally. Dialing number will search the suitable rule from the beginning.

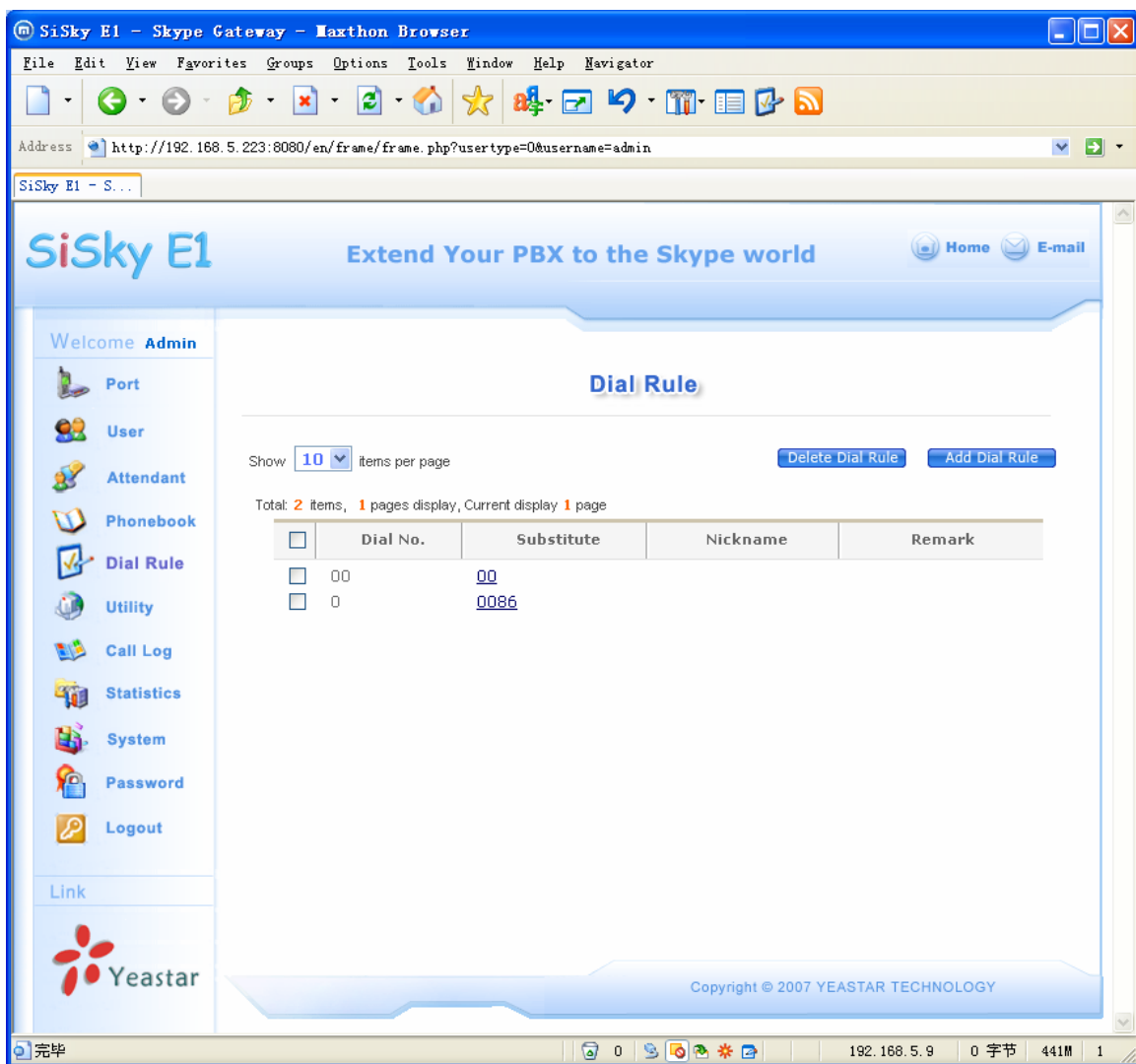
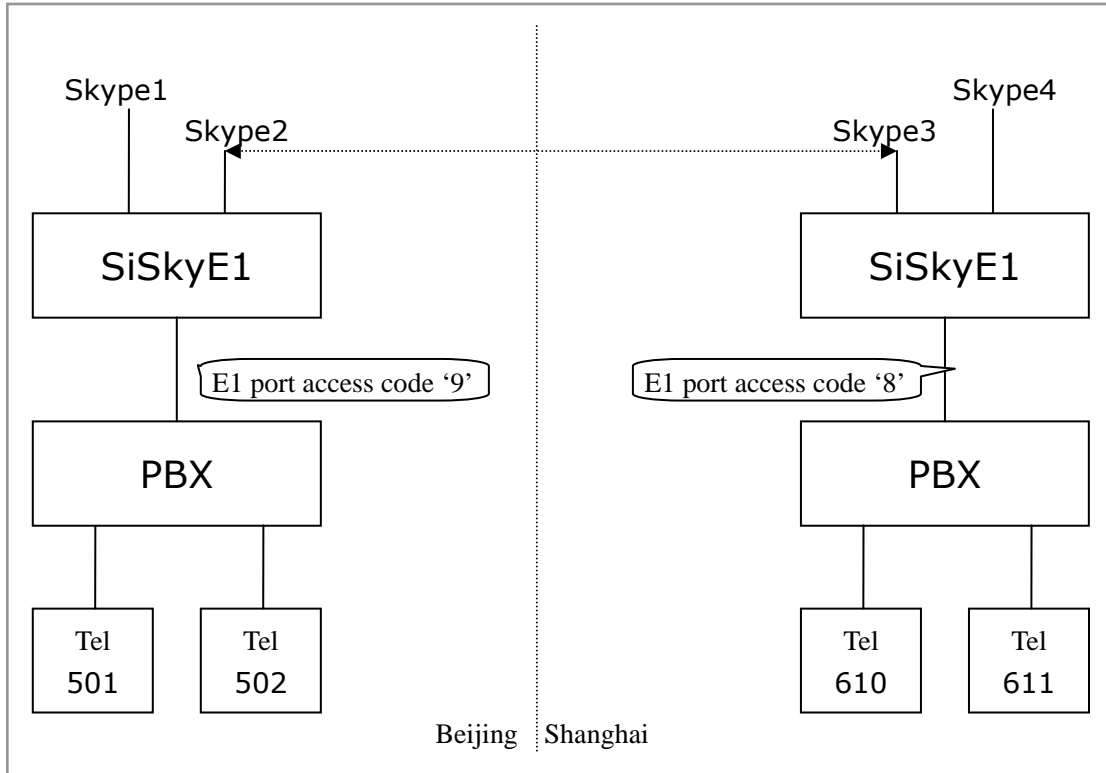


Figure 7-3

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

8.2 Application of Branch Offices Connection

Through E1 port, SiSkyE1 can realize free communication between branch offices. For example, both Beijing and Shanghai offices are using SiSkyE1 Servers



On the above demonstrated environment:

On the **Public Phonebook** of Beijing, user sets a Speed-dial number 111 for Skype 3;

On the Public Phonebook of Shanghai, user sets a Speed-dial number 444 for Skype 2;

8.2.1 Beijing Calling Shanghai

Make a call from Beijing Extension 501 to Shanghai Extension 610

Through Skype1: pick up -> dial 9111# (hearing SiSkyE1's music on hold and PBX's Auto Attendant) -> dial 610

Through Skype2: pick up -> dial 9111# hearing SiSkyE1's music on hold and PBX's Auto Attendant) -> dial 610

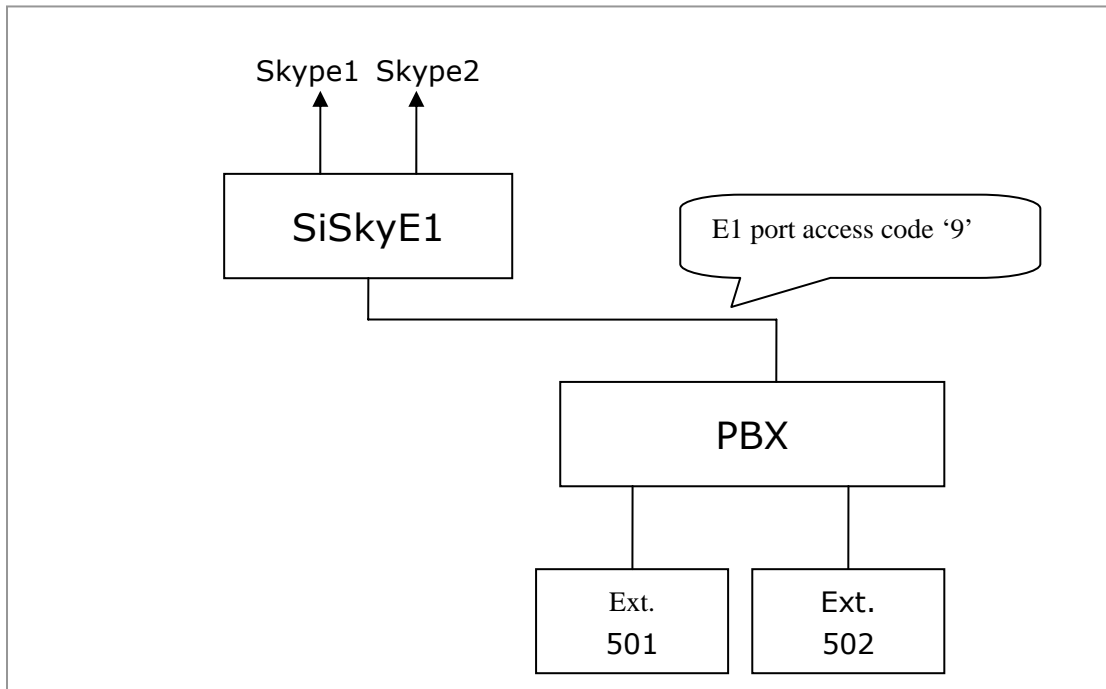
8.2.2 Shanghai Calling Beijing

Make a call from Shanghai Extension 610 to Beijing Extension 501

Through Skype3: pick up -> dial 8444# (hearing SiSkyE1's music on hold and PBX's Auto Attendant) -> 501

Through Skype4: pick up -> dial 8444# (hearing SiSkyE1's music on hold and PBX's Auto Attendant) -> 501

8.3 Call to regular phone or mobile or Skype ID through SkypeOut



On the above demonstrated environment:

8.3.1 Call to regular phone or mobile phone

1) Call to regular phone

If extension 501 wants call the PSTN number 0755-12345678 :

Pick up-> dial 90755-12345678#

2) Call to mobile phone

If extension 501 wants call the mobile number 15960123456 :

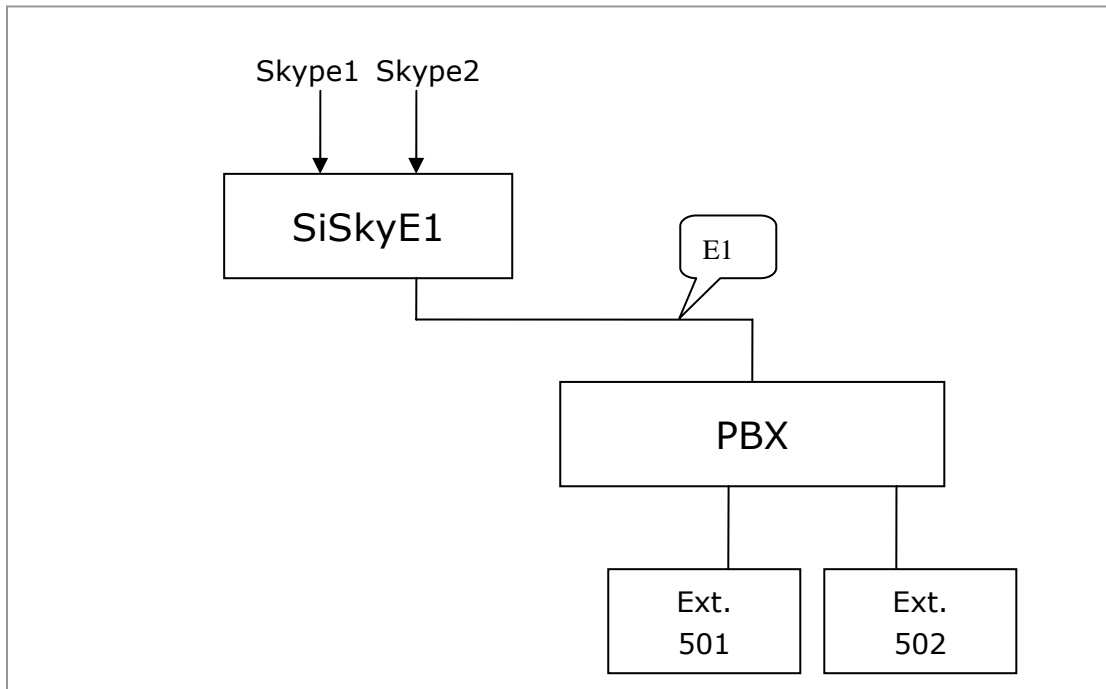
Pick up -> dial 9015960123456#

8.3.2 Call to Skype ID

If the Extension user 501 wants to dial Skype ID 'echo123', and the speed-dial number of this ID on Public Phonebook is '123':

Pick up -> dial 9123#

8.4 Skype Incoming Calls Through SiSkyE1



When there's an incoming call to Skype1 or Skype2, caller will first hear the music on hold and then hear an auto attendant from SiSkyE1. Caller will continue to dial the extension number. Except the music on hold, all are same as calls from PBX's PSTN line.

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

SiSkyE1 General Settings

9

Click the 'Option' tab in SiSkyE1 interface to set the operating environment.

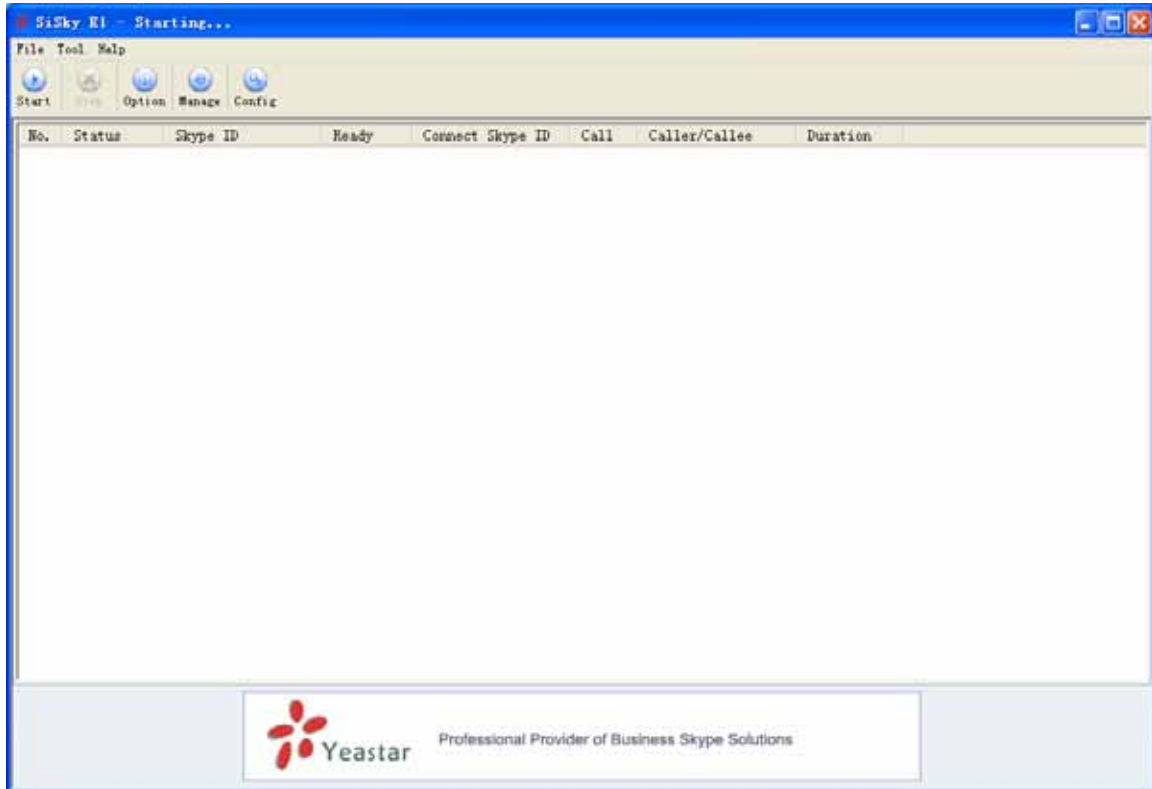


Figure 9-1

9.1 General Setting

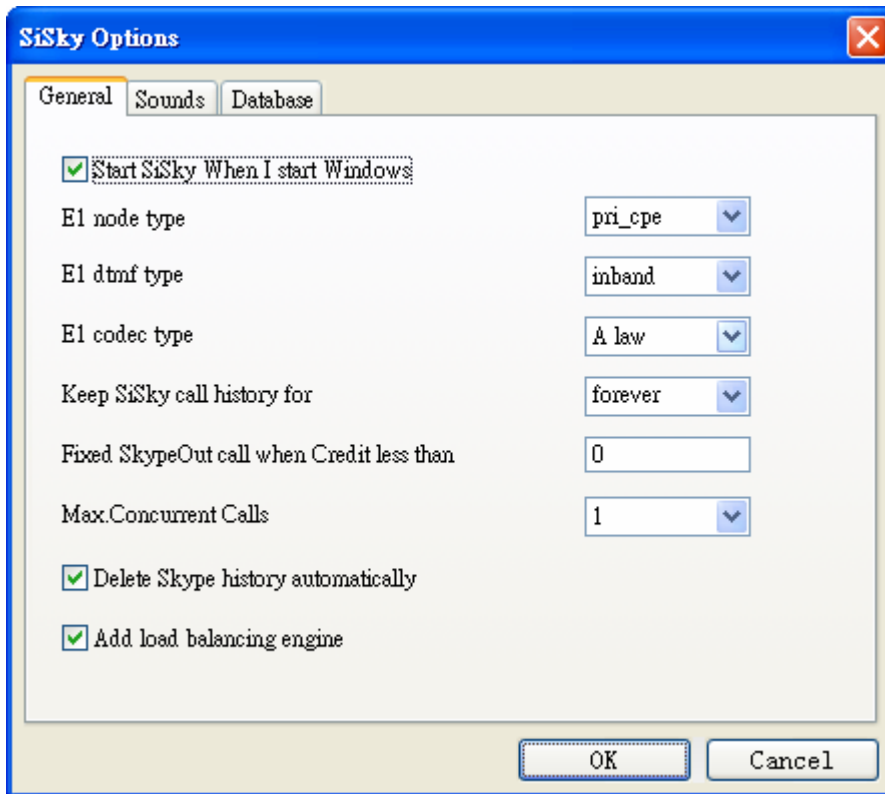


Figure 9-2

1. Start SiSky When I start Windows
2. E1 node type
E1 node type: pri_net or pri_cpe
3. E1 dtmf type
E1 dtmf type: inband or pri_info
4. E1 codec type
E1 codec type: A_LAW or U_LAW
5. Keep SiSky call history for
You can choose 'no history', 'one month', 'three months', 'six months' or 'forever'. You can delete call history manually as well through web administration.
6. Fixed SkypeOut call when Credit less than
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.
7. Max.Concurrent Calls
8. Delete Skype history automatically
Skype history soon accumulate as time goes on, which will need more memory and affect the system performance. If enable this option, system will clean all Skype history automatically when starting SiSkyE1.

9. Add load balancing engine

After enabled this function, SiSkyE1 will balance the flow rate among Skype IDs of all port.

For example, a user configured 8 trunks on SiSkyE1, 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.

9.2 Sounds Setting

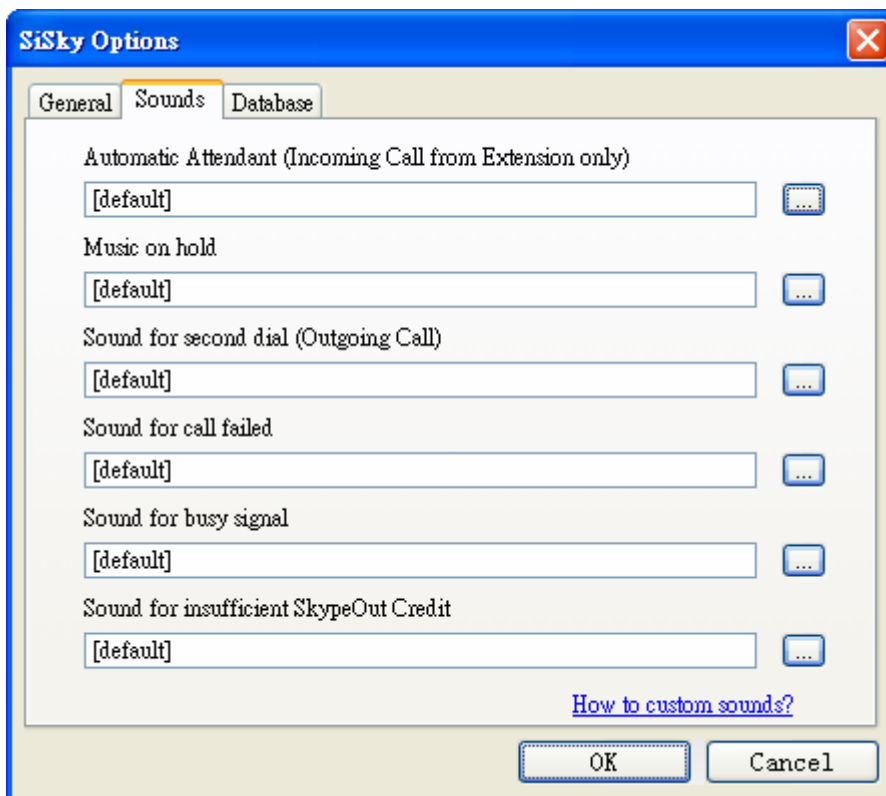


Figure 9-3

1. Automatic Attendant

When there is a Skype incoming call, SiSkyE1 will display the auto attendant and guide the caller to the dedicated extensions.

SiSkyE1 has default auto attendant. If you want to customize it, please make reference to [Appendix A](#).

2. Music on Hold

SiSkyE1 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).

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. SiSkyE1 has default sound – Dial Tone. If you want to customize it, please make reference to [Appendix A](#).

4. Sound for call failed

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

5. Sound for busy signal

Caller will hear this prompt sound when callee is busy.

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 *chapter8.1.4*

9.3 Database Setting

- 1) If you have no need of cascade connecting multiple SiSkyE1 Servers, please don't enable **Connect to Primary Server**.
- 2) Backup and Restore the data in this PC.

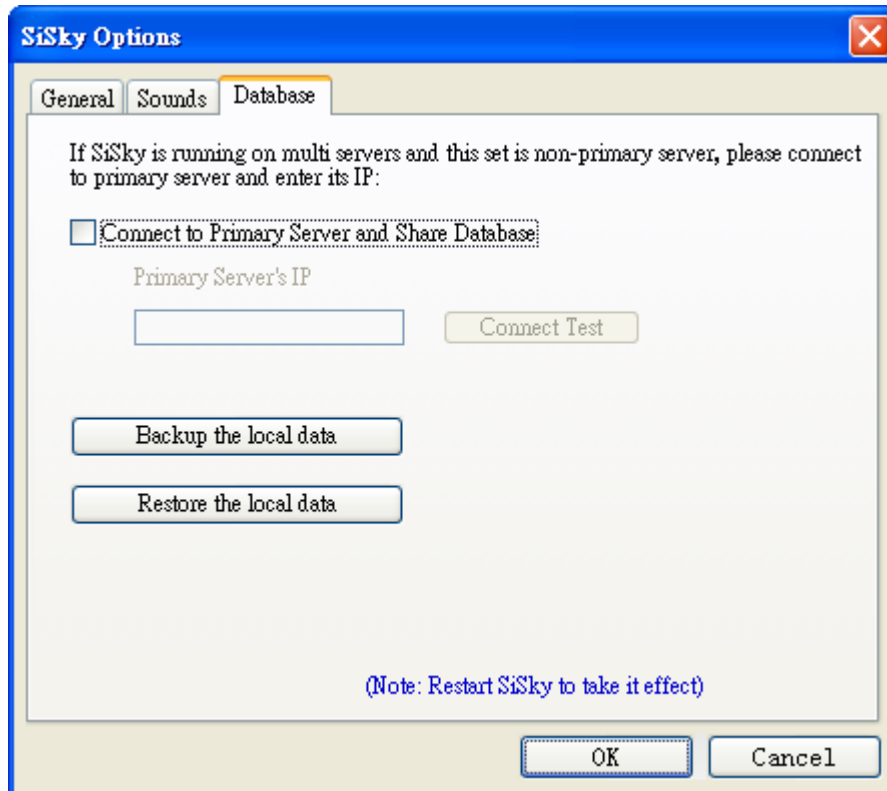


Figure 9-4

SiSkyE1 Advanced Functions **10**

10.1 Open SiSkyE1 Homepage

If you are operating in SiSkyE1 Server, please click the **Manage** button or you can open the IE browser and enter in <http://127.0.0.1:8080>

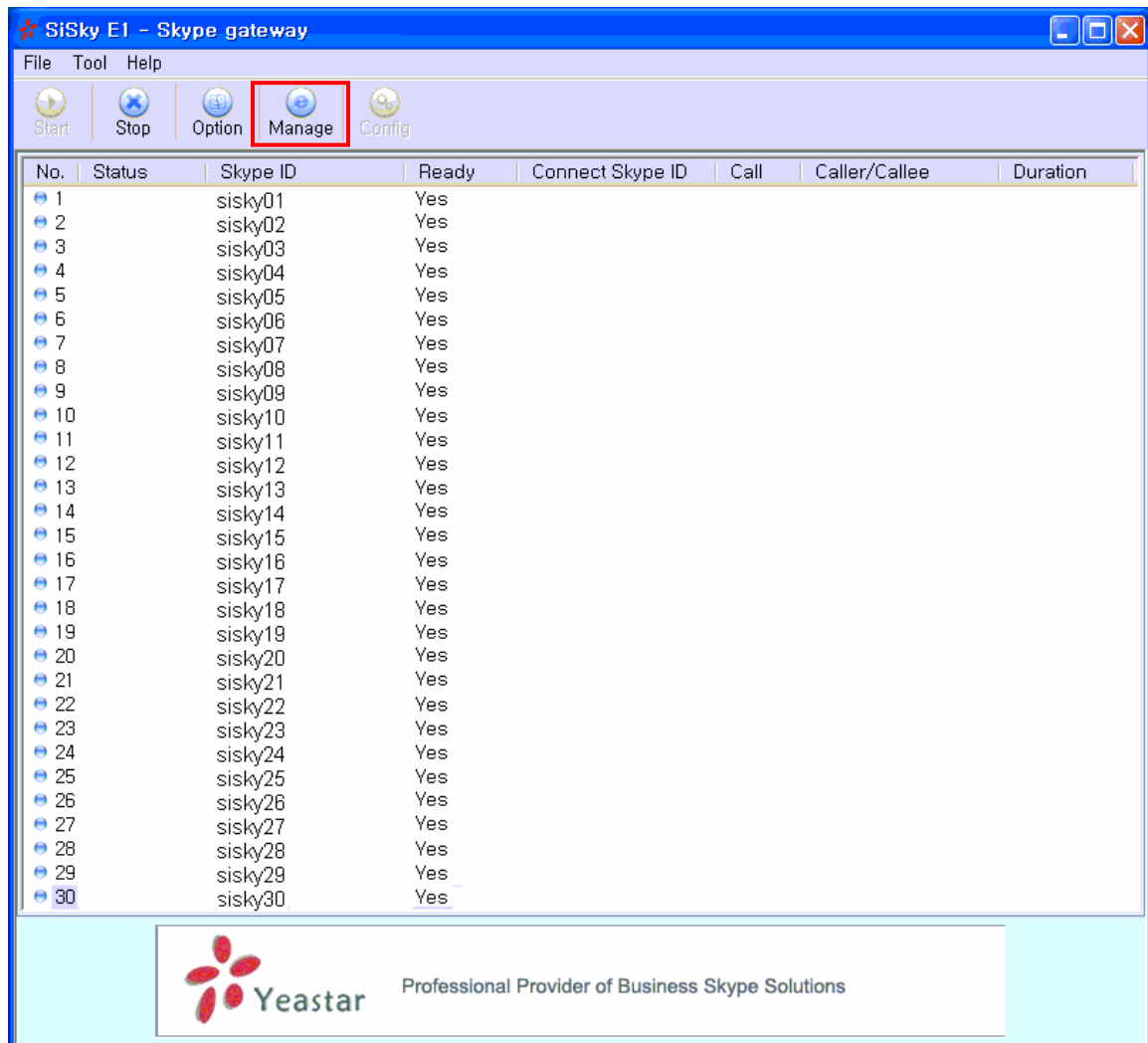


Figure 10-1

If you are not operating in SiSkyE1 Server, please open the 8080 port of SiSkyE1 Server through IE browser. For example, if the IP address of SiSkyE1 Server is 192.168.5.66, then you can type <http://192.168.5.66:8080> in the IE address bar.

10.2 Administrator Logging in

Open the SiSkyE1 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.

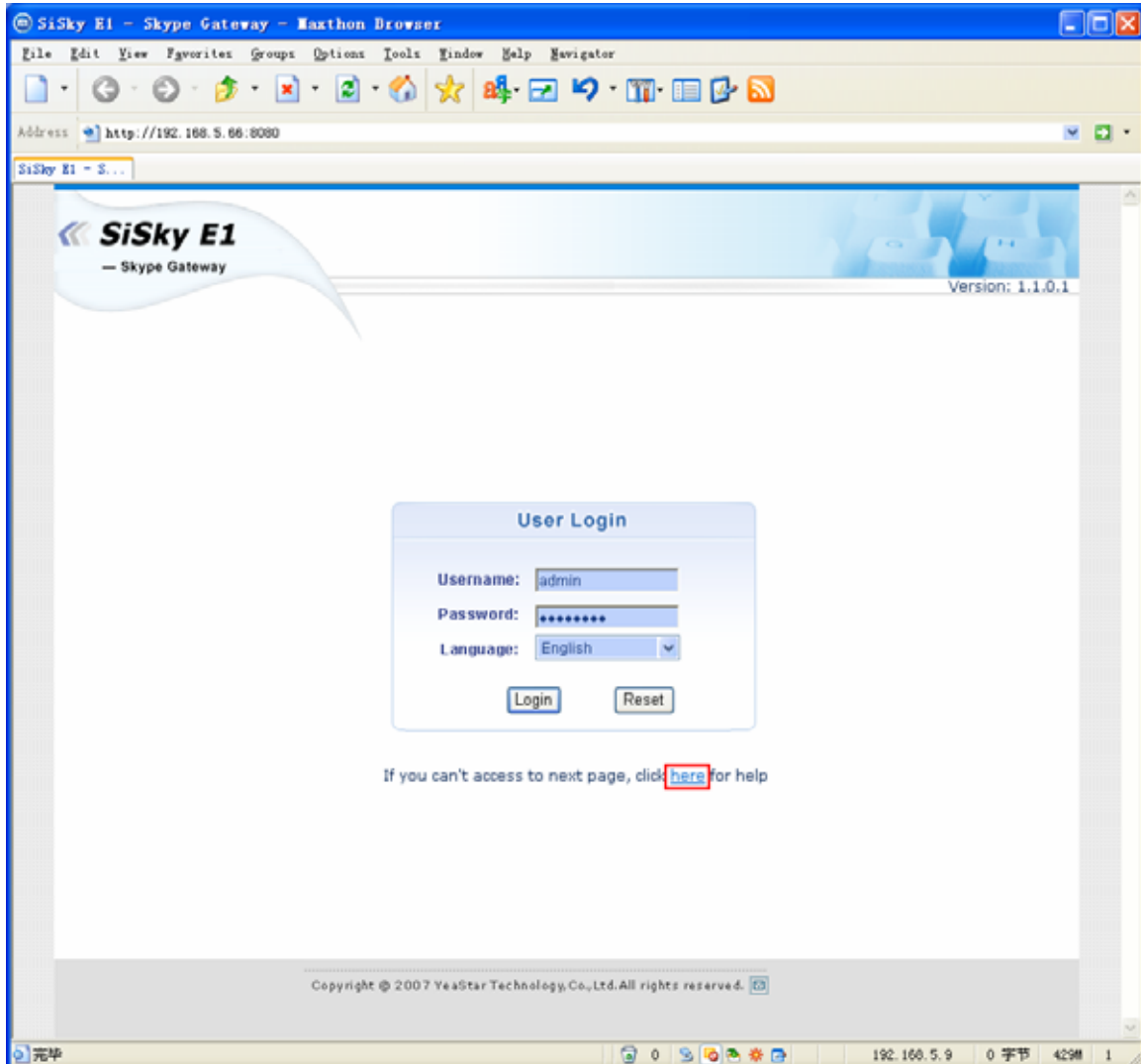


Figure 10-1

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.

10.3 Port Status and Management

10.3.1 Timely display the status of every port

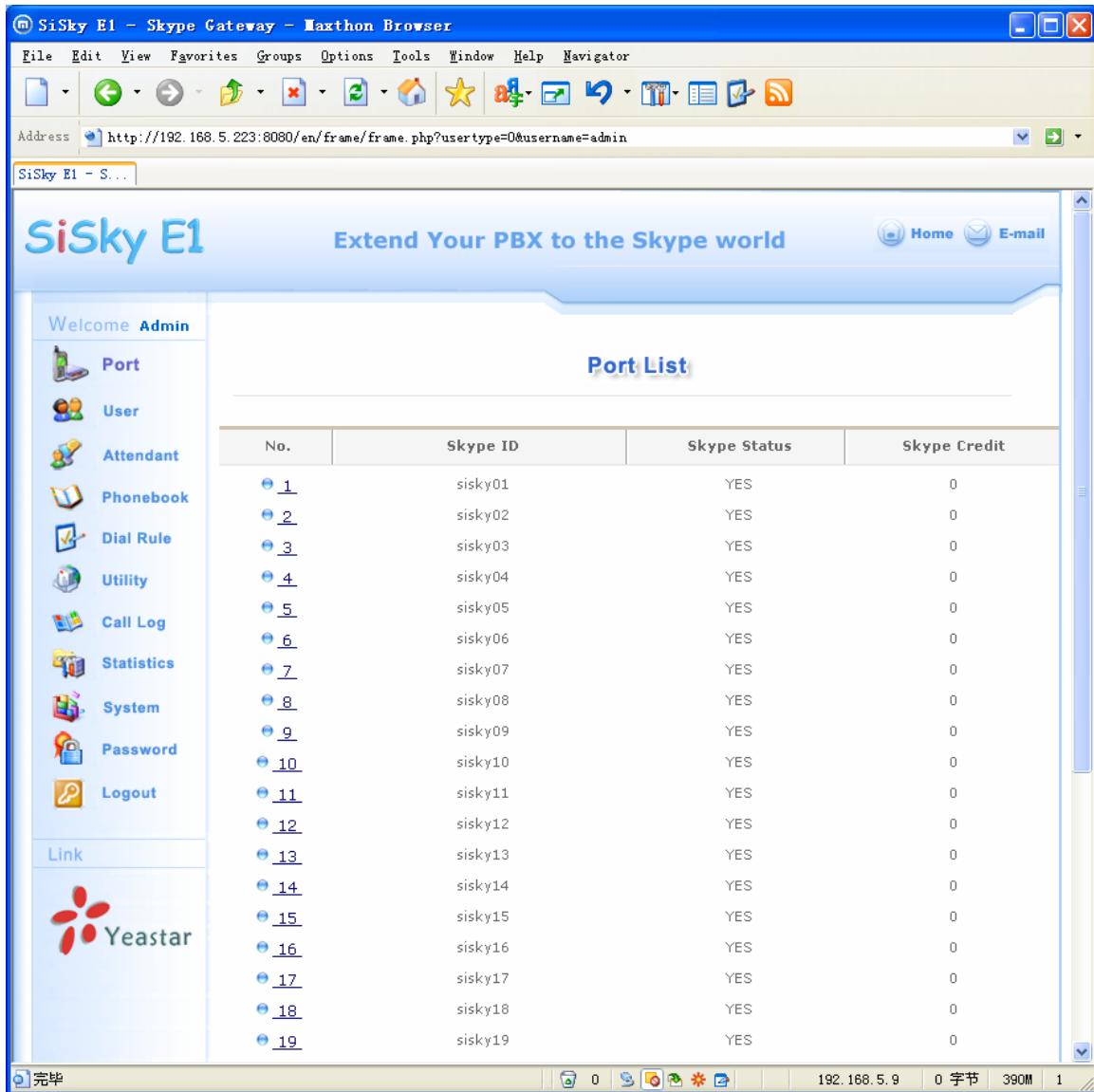


Figure 10-2

10.3.2 Managing Port's Settings

1. Skype Setting

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

Enable Skype: Whether or not to enable this port's Skype ID.

2. Others

Direct In: Enter in an extension number here. Incoming Skype calls to this port through SiSkyE1 will ring the extension phone directly and the automatic attendant will be unavailable on this port.

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.

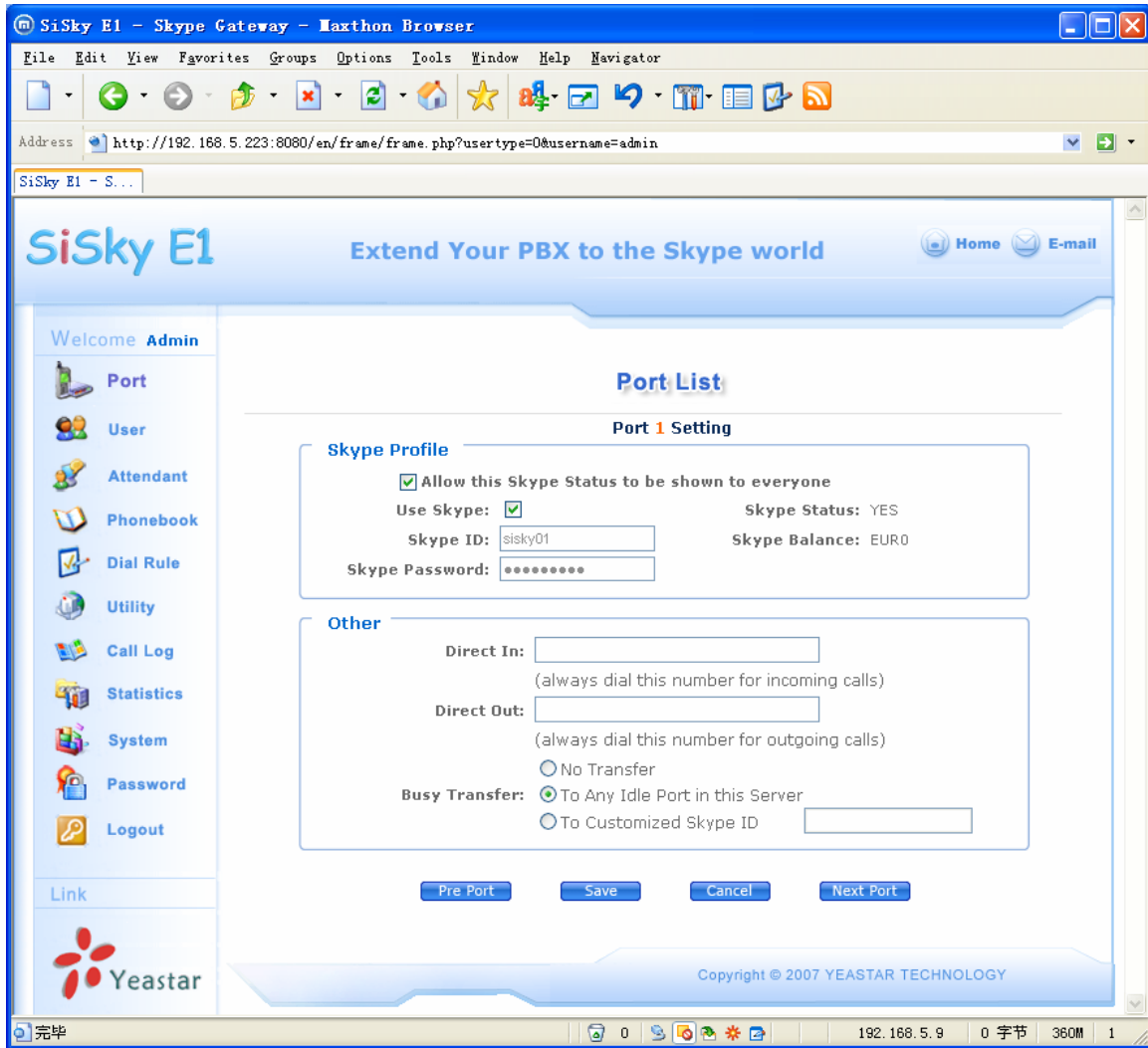


Figure 10-3

10.4 User Management

The User (PIN) List will be unavailable until you enable the multi-user mode. When running SiSkyE1 software under multi-user mode, every user is possible to access his/her own private phonebook after logging in WEB. (See details on [chapter 10.13](#)---Logging In As Standard Users). User needs to enter his/her own PIN number when making calls.

10.4.1 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.

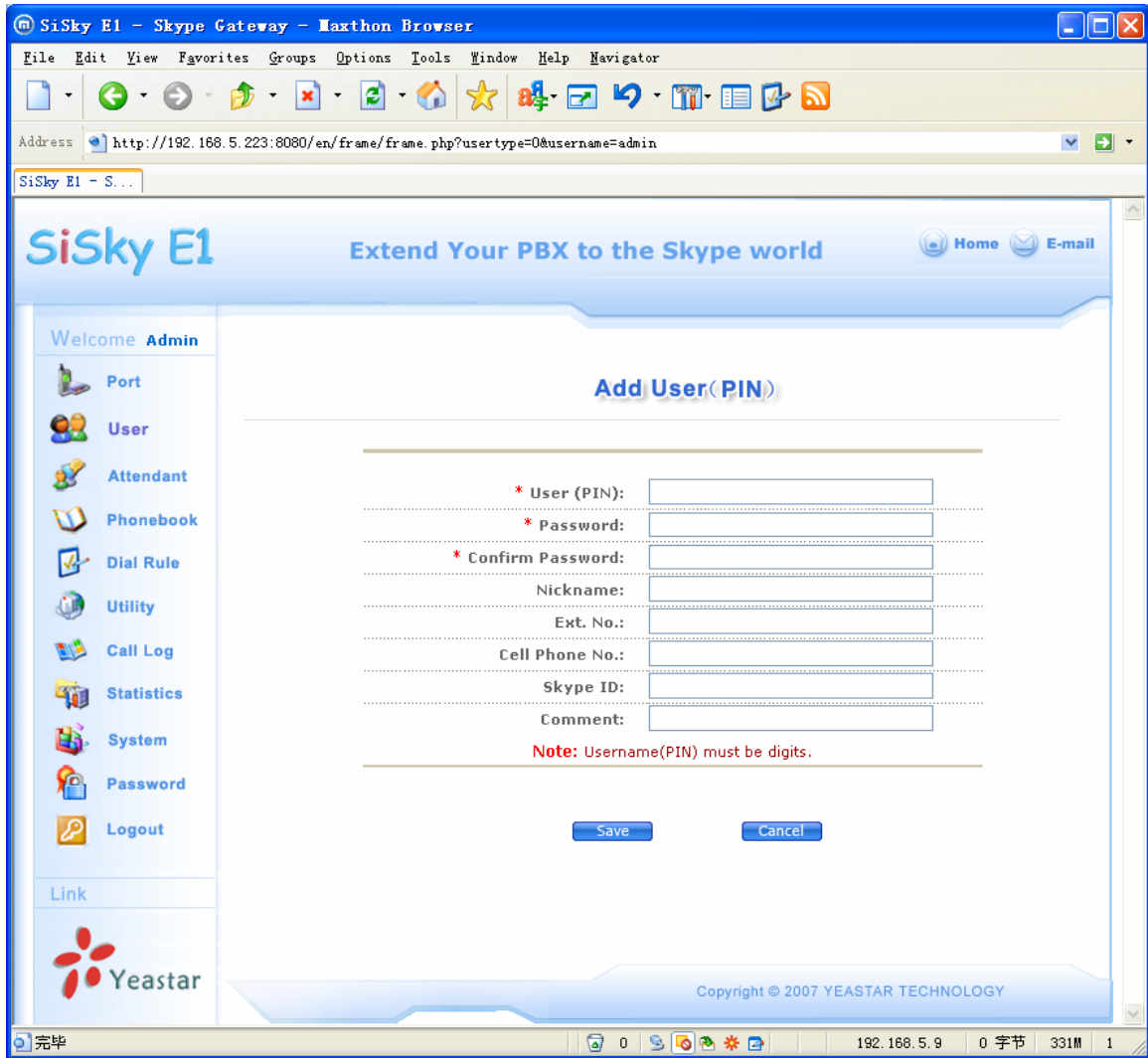


Figure 10-4

10.4.2 Deleting User

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

10.5 Managing Automatic Attendant

When there's incoming call from Skype port, SiSkyE1 will play the auto attendant first and transfer call to destination extension according to the caller's the second dialing (DTMF). This part will introduce the rules of transferring.

10.5.1 Transferring List

10.5.1.1 Adding Transferring Entry

Click **Add Item** to enter Figure 10-5. 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.

10.5.1.2 Deleting Transferring Entry

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

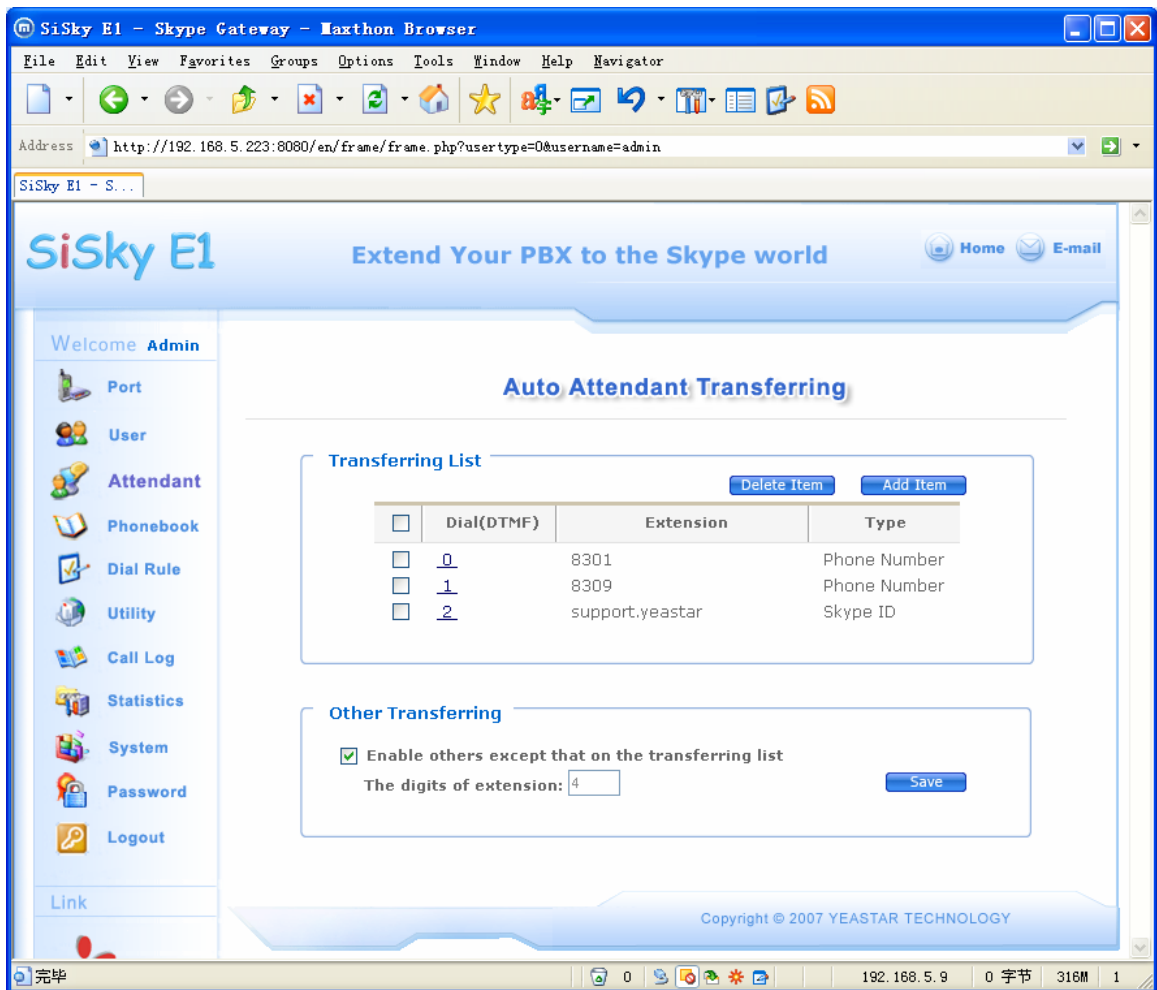


Figure 10-5

10.5.2 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 Figure10-5, input the digits of extension to transfer in time when caller finish dialing.

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 XX 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 Figure10-5.

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.

10.6 Phonebook

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

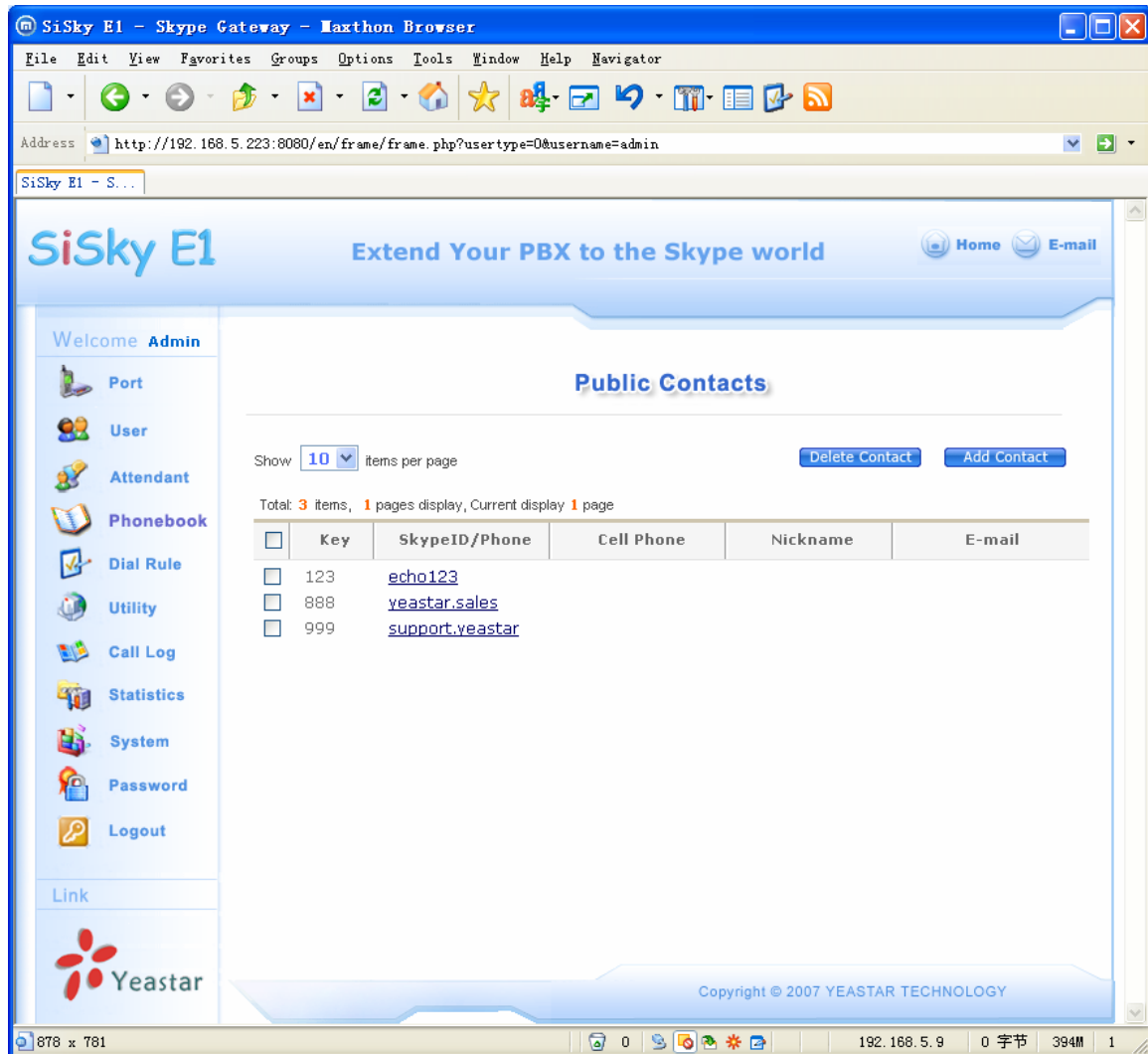


Figure 10-6

10.6.1 Adding Public Contact

Click the **Add Contact** button 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.

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**.

10.6.2 Deleting Public Contact

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

10.7 Dial Rule

Making calls through Skype, one have to conform to the Skype dialing scheme as well as calling through PSTN. Maybe you had already got used to PSTN 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.

For example, a PSTN user in the mainland of China wants to make an international call IDDD (international direct distance dialing), which format is **00+country code+area code+telephone number**; make a 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**.

An International call: for example, a user in China calls to Canada

1) in traditional PSTN format: the country code is 1, the area code is 416, phone number is "12345678", then he would dial: **00 +1+ 416 + 12345678**;

2) in SkypeOut format: the country code is 1, the area code is 416, phone number is "12345678", then he would dial: **00 +1+ 416 + 12345678**;

In this example, the PSTN and SkypeOur format for an international call are the same.

A Domestic Distance call: a Chinese user in Paking calls to Shenzhen

3) in traditional PSTN format: the area code is "755", phone number is "12345678", then he would dial: **0+755+12345678**;

4) in SkypeOut format: the area code is "755", phone number is "7571234", then he would dial: **00+86 (country code)+755+12345678**;

In this example, the PSTN and SkypeOur format for an domestic distance call are different.

In order to not change the PSTN dialing format, this user can take advantage of the Dial Rule to place 0086 with 0, as shown on Figure 7-3

When the number begins by 00, SiSkyE1 will identify it as international call requirement. When the dialing call begins by 0, SiSkyE1 will identify the 0 as domestic call requirement and transfer it to 0086 automatically to conform SkypeOut format.

Why we substitute 00 for 00? Say it simply, to prevent 00 matches with 0. Because both the international and domestic calls are begin by 0, for fear of replacing **00141612345678** with **00860141612345678**, substitution rules priority are listed basipetally. Dialing number will search the suitable rule from the beginning.

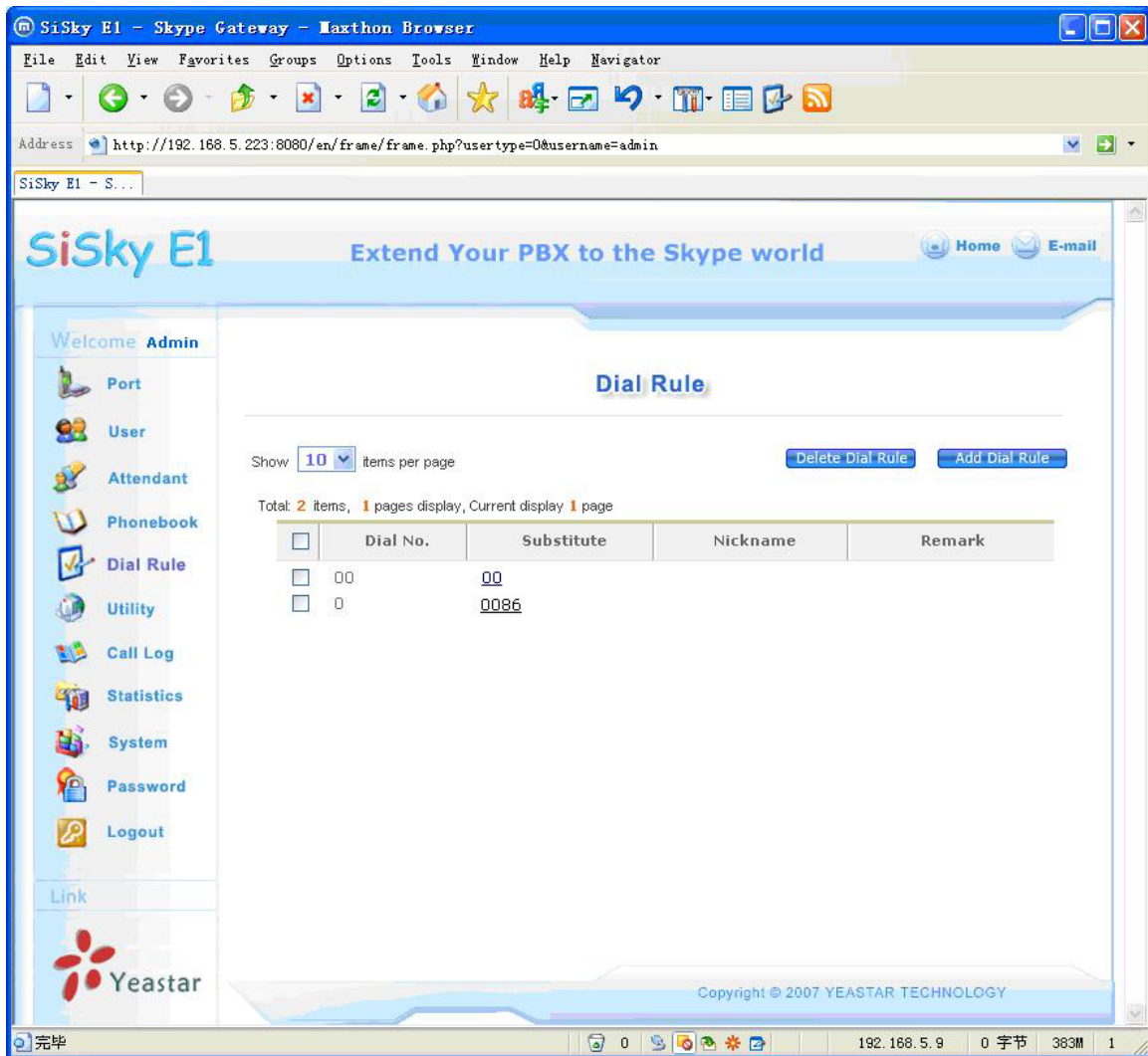


Figure 10-7

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

10.8 Backing up & Restoring

10.8.1 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 Figure10-8. Select the type of application you want to backup, then click **BACK UP** button 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, and then click **RESTORE**. The data is restored to the database.

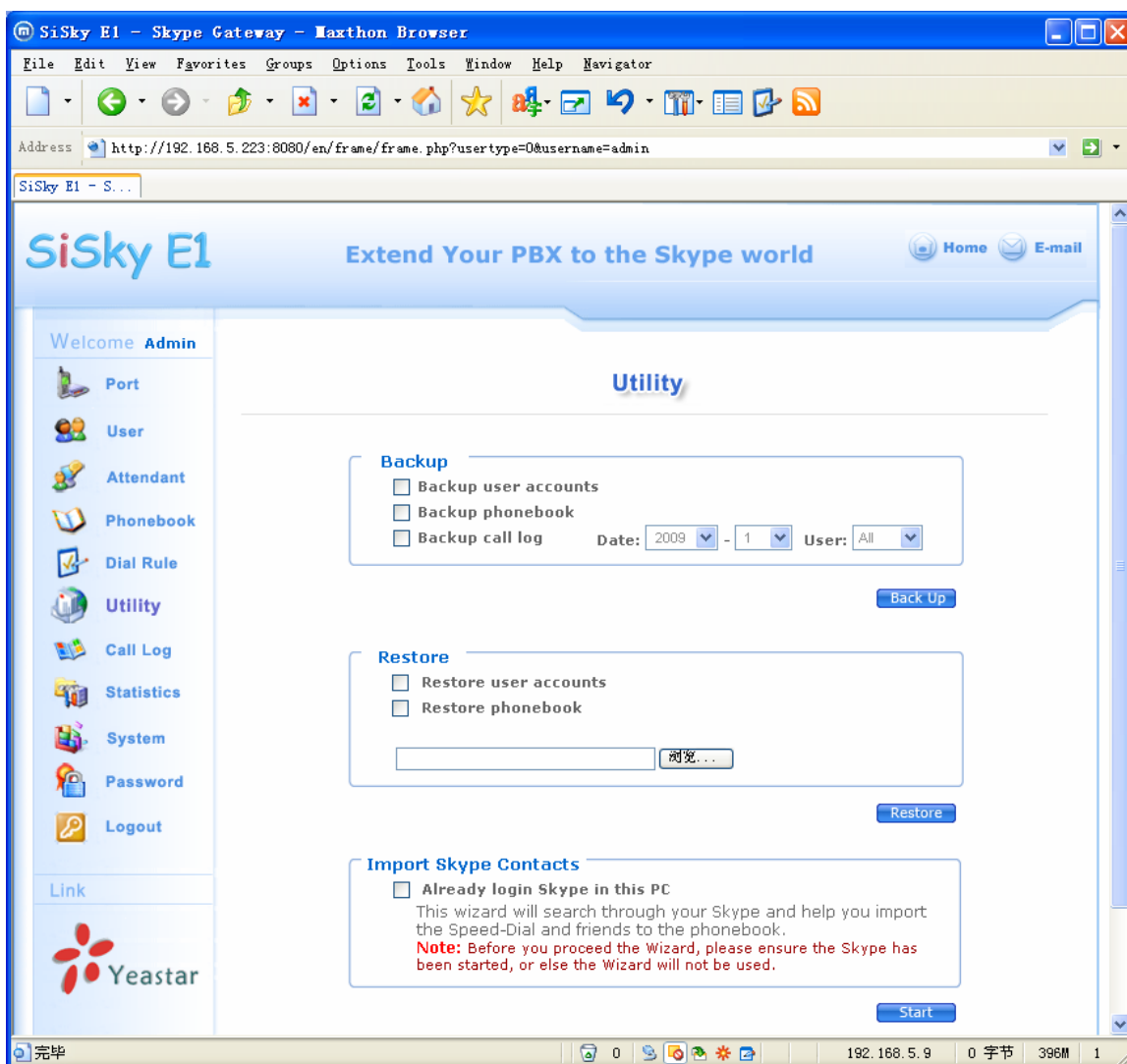


Figure 10-8

10.8.2 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.

Step 1: Get Skype Contacts

Click **Get Contacts** as shown on Figure 10-9

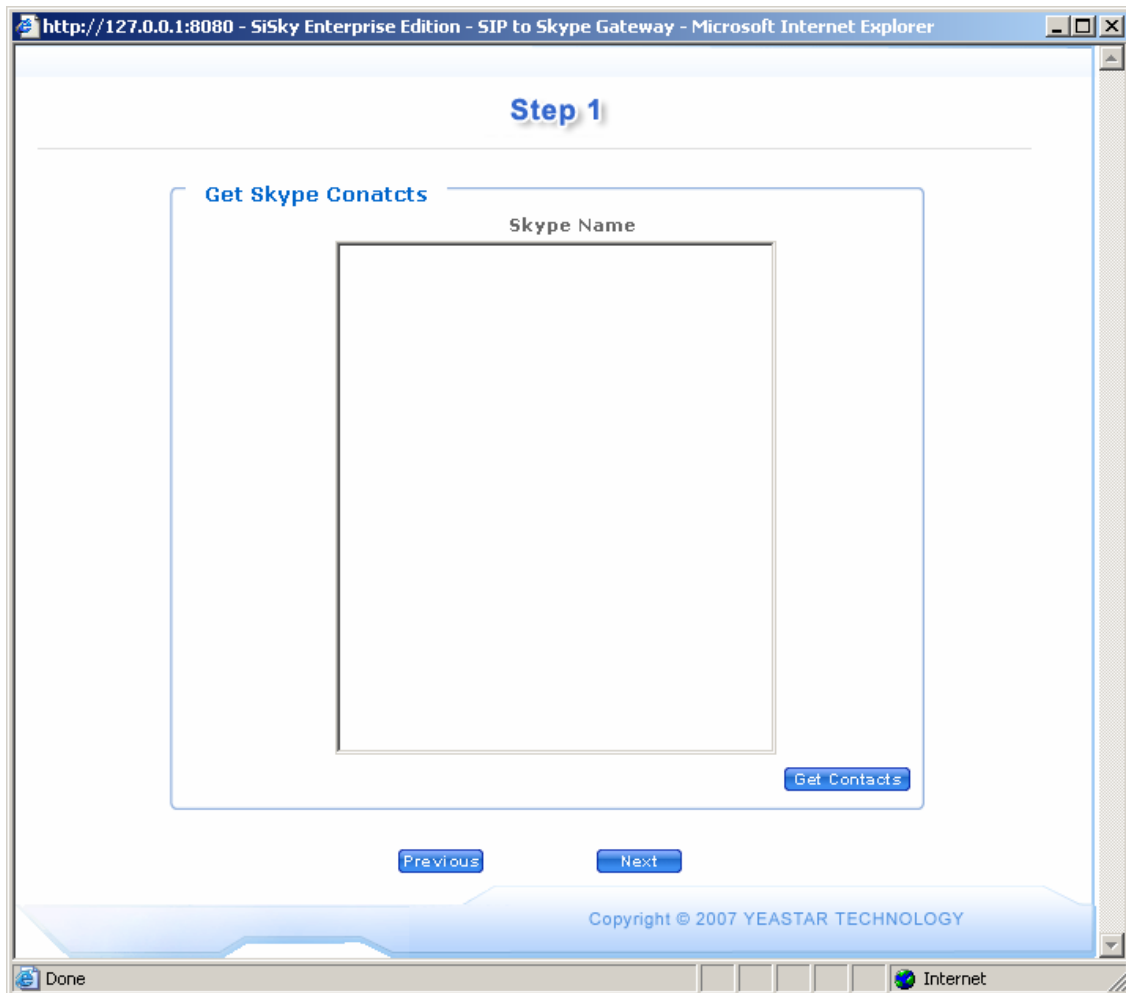


Figure 10-9

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 10-10

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



Figure 10-11

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.

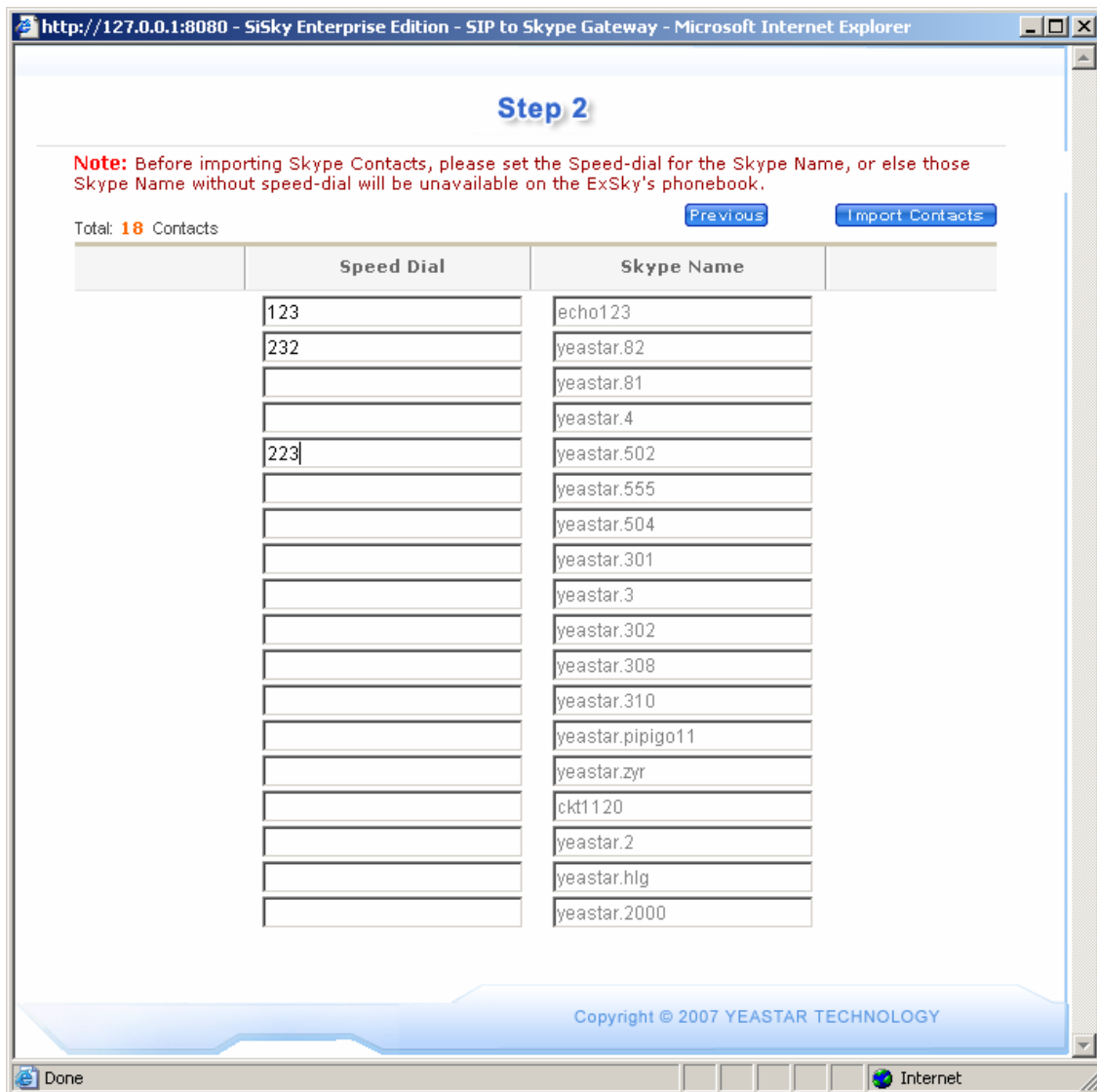


Figure 10-12

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

When finished, a message window will appear as Figure 10-13. Click the **Finish** button to close.

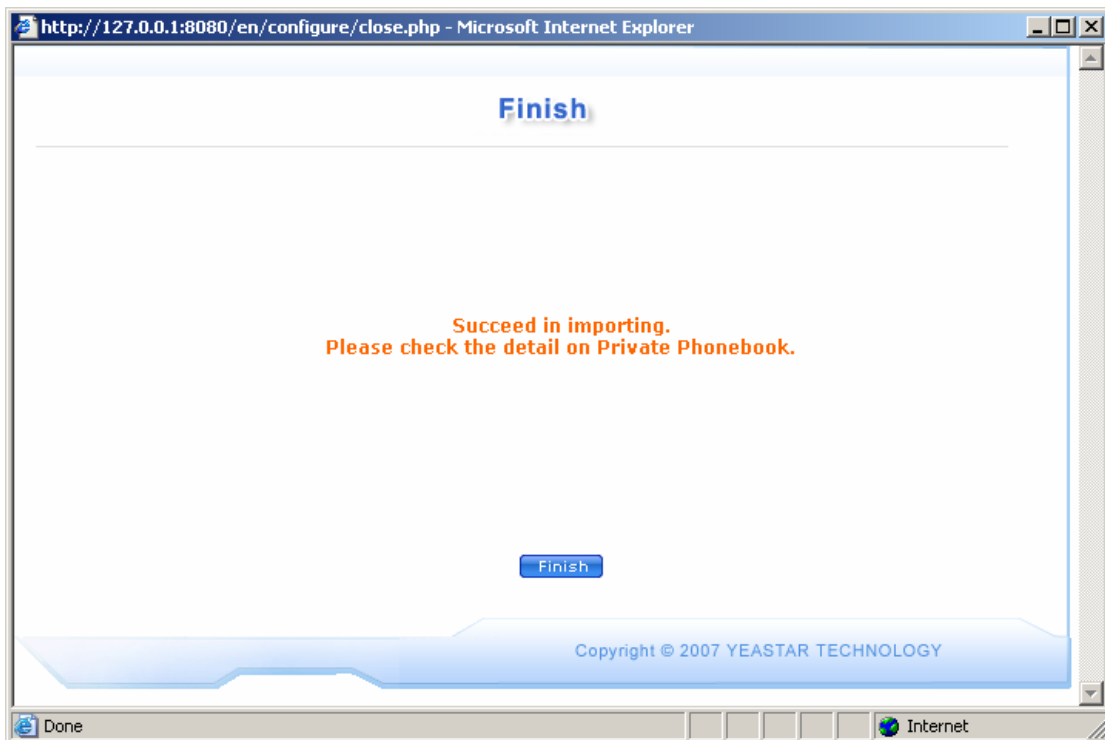


Figure 10-13

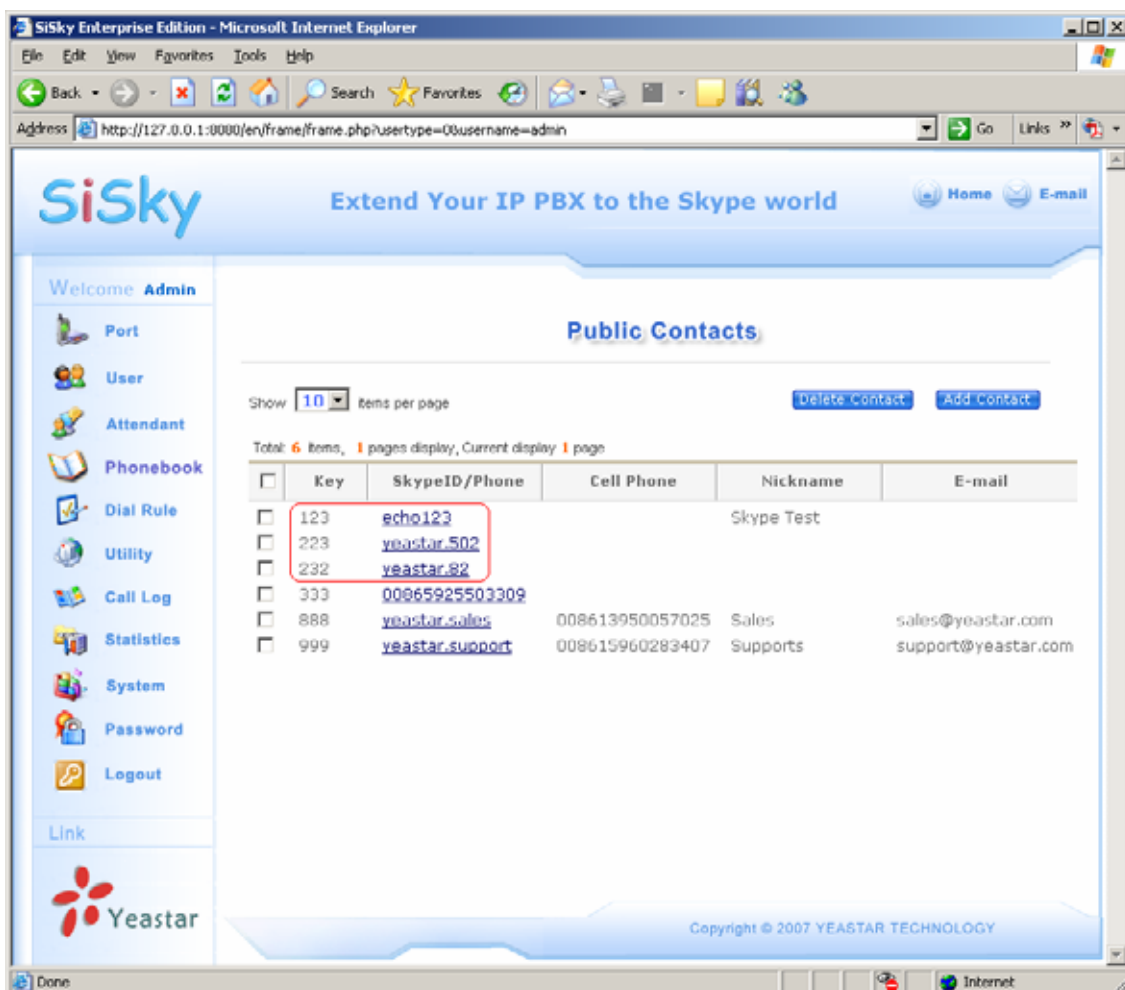
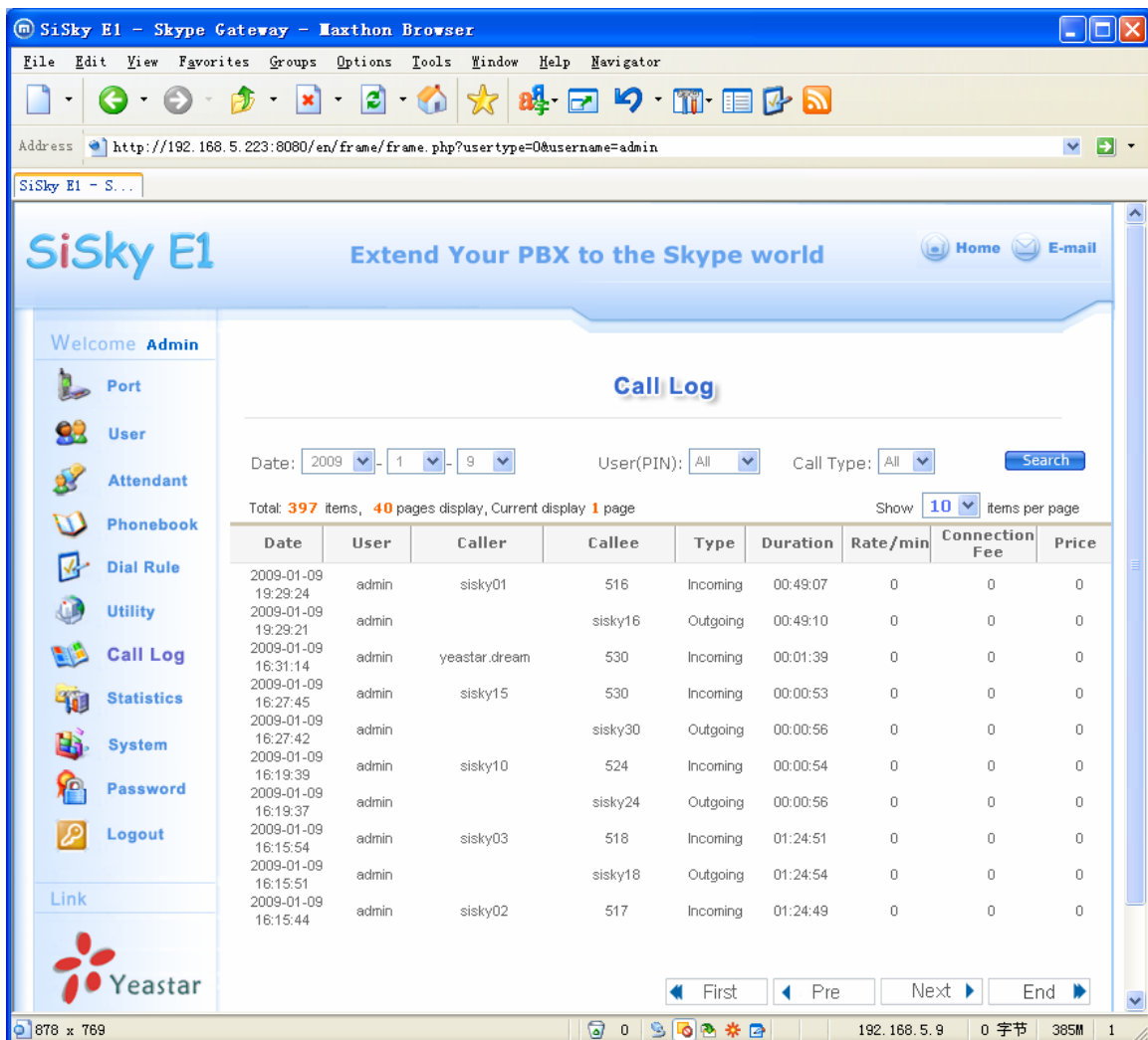


Figure 10-14

10.9 Viewing Call Log

Administrators can view all users' call log. Call Log captures all call details: calling time, caller, callee number, and 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.



Date	User	Caller	Callee	Type	Duration	Rate/min	Connection Fee	Price
2009-01-09 19:29:24	admin	sisky01	516	Incoming	00:49:07	0	0	0
2009-01-09 19:29:21	admin		sisky16	Outgoing	00:49:10	0	0	0
2009-01-09 16:31:14	admin	yeastar.dream	530	Incoming	00:01:39	0	0	0
2009-01-09 16:27:45	admin		sisky15	Incoming	00:00:53	0	0	0
2009-01-09 16:27:42	admin		sisky30	Outgoing	00:00:56	0	0	0
2009-01-09 16:19:39	admin		sisky10	Incoming	00:00:54	0	0	0
2009-01-09 16:19:37	admin		sisky24	Outgoing	00:00:56	0	0	0
2009-01-09 16:15:54	admin		sisky03	Incoming	01:24:51	0	0	0
2009-01-09 16:15:51	admin		sisky18	Outgoing	01:24:54	0	0	0
2009-01-09 16:15:44	admin		sisky02	Incoming	01:24:49	0	0	0

Figure 10-15

10.10 Viewing Statistics

After you are done with the configuration procedure, SiSkyE1'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.

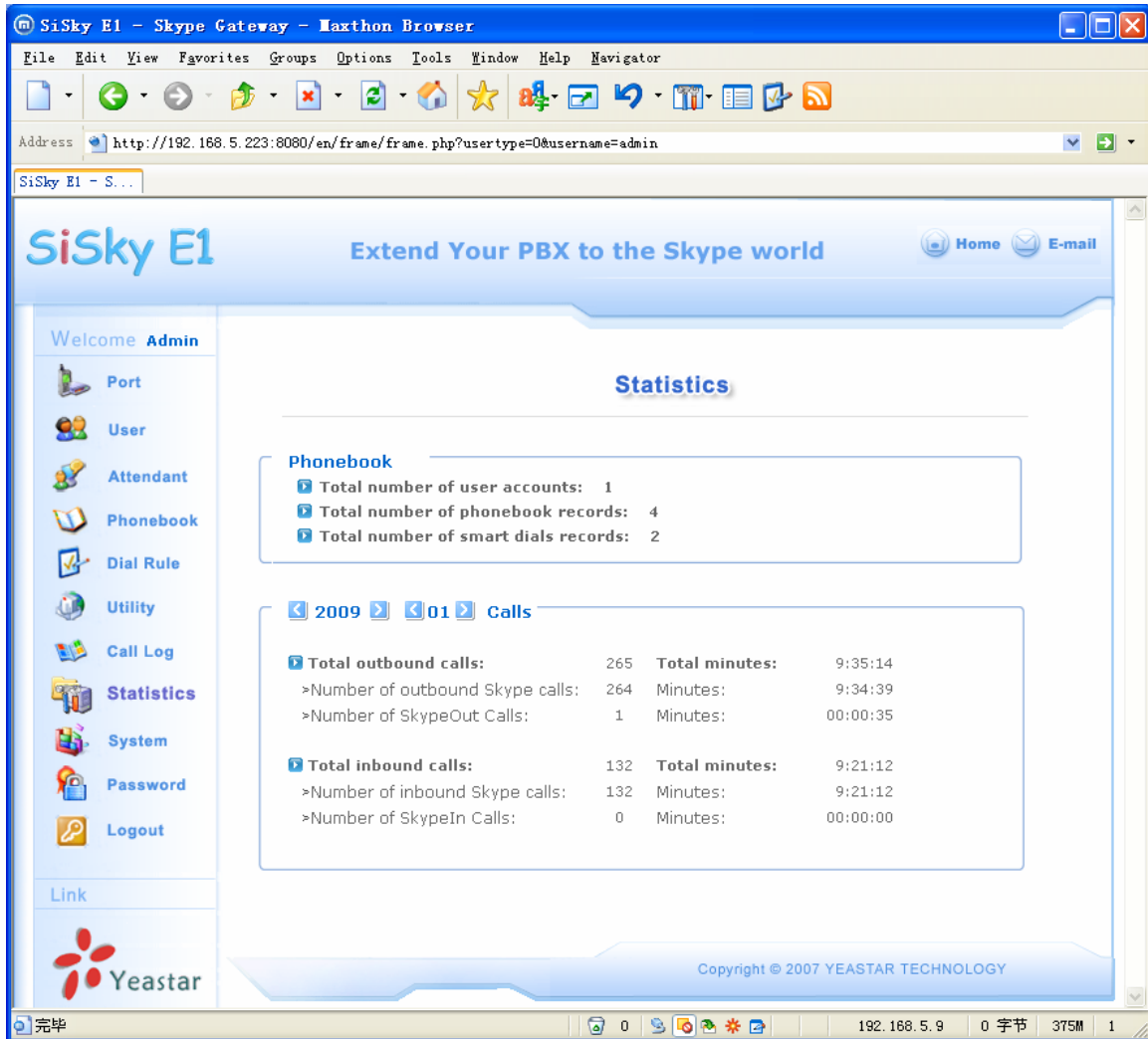


Figure 10-16

10.11 System

Global and Alert Setting, please refer to [9.2 General Setting](#).

Sounds Setting, please refer to [9.3 Sounds Setting](#).

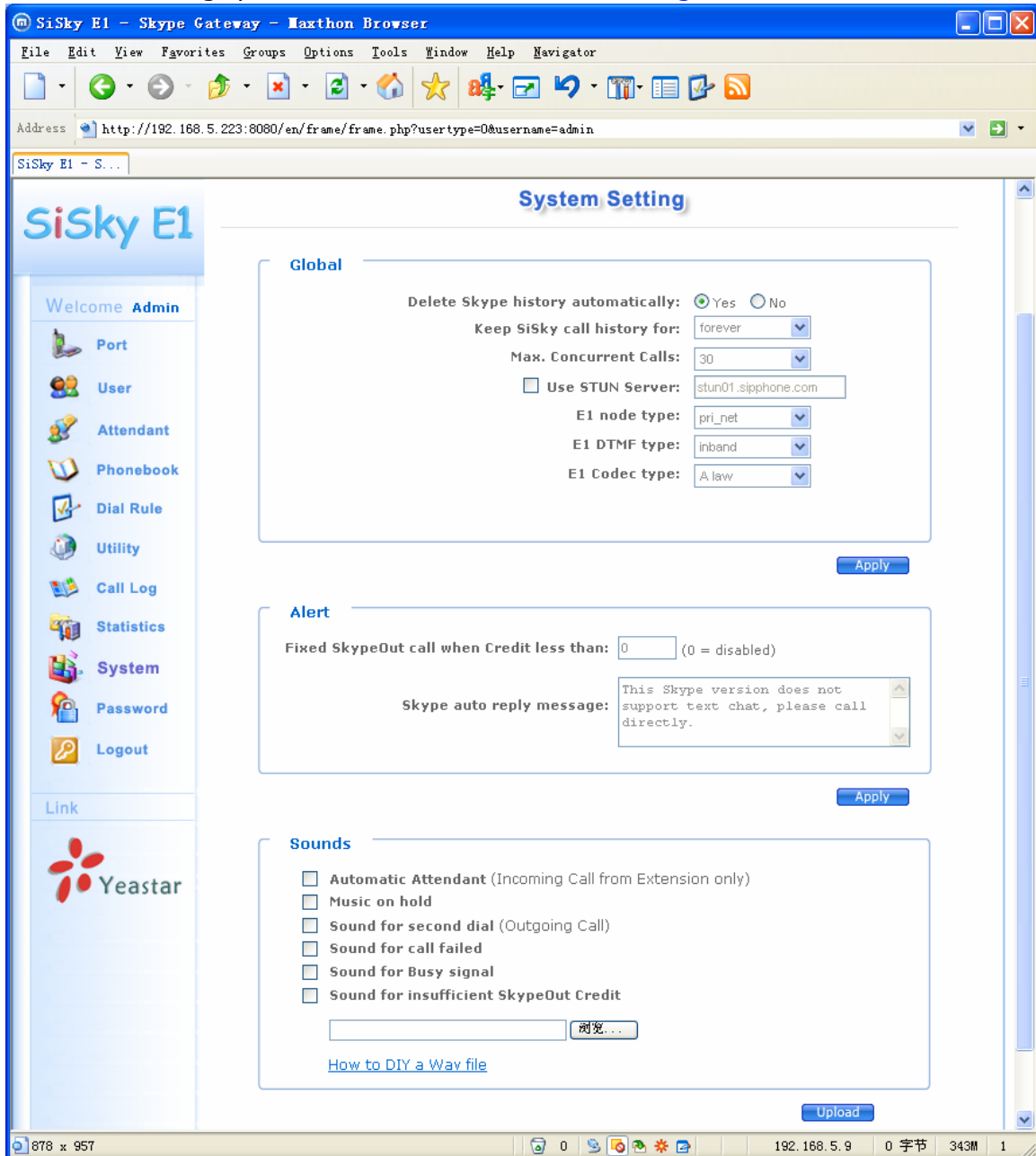


Figure 10-17

10.12 Password

You can change the password here.

10.13 Logging In As Standard Users

10.13.1 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 Figure10-18.

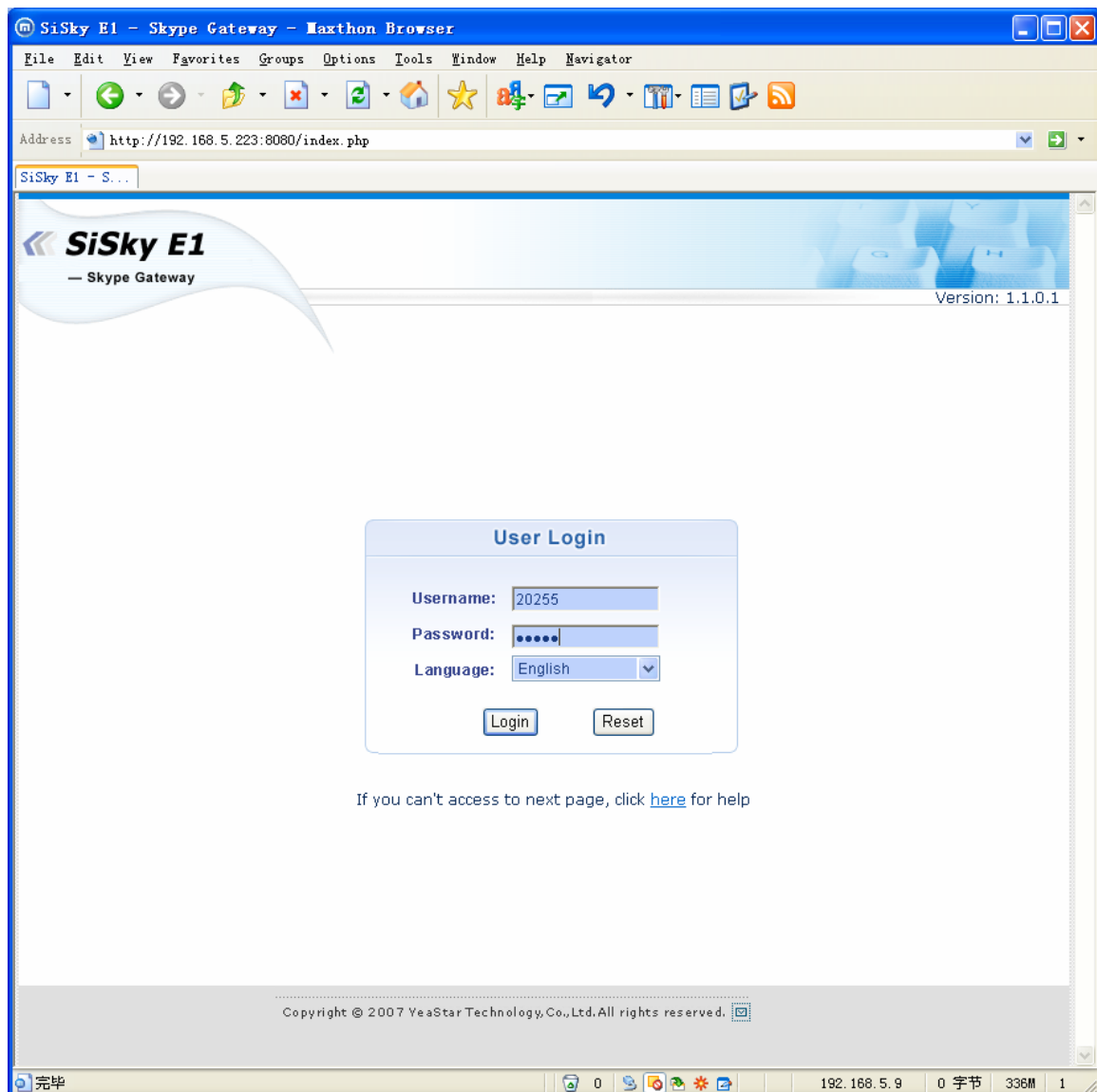


Figure 10-18

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.

10.13.2 Phonebook

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

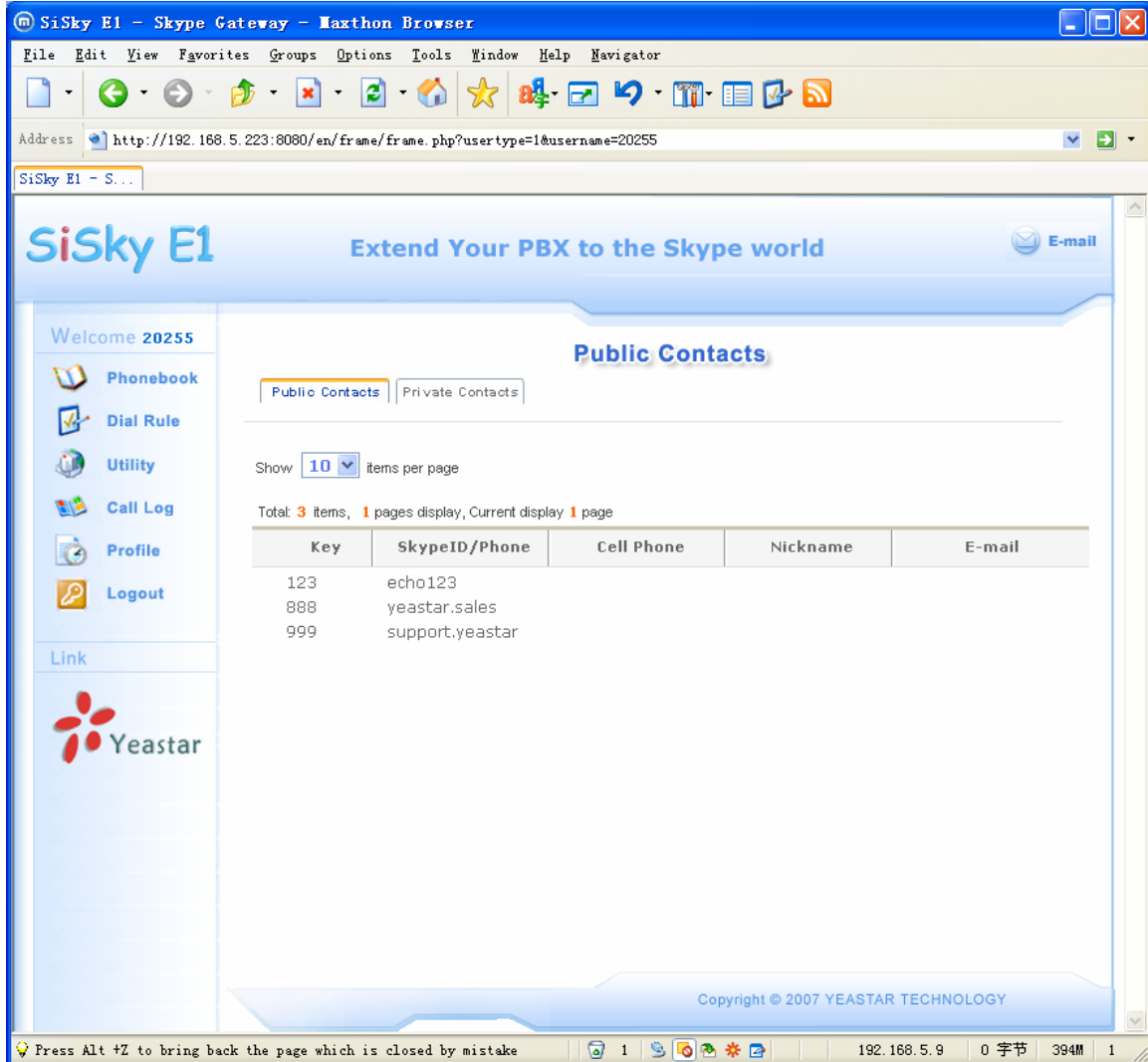


Figure 10-19

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

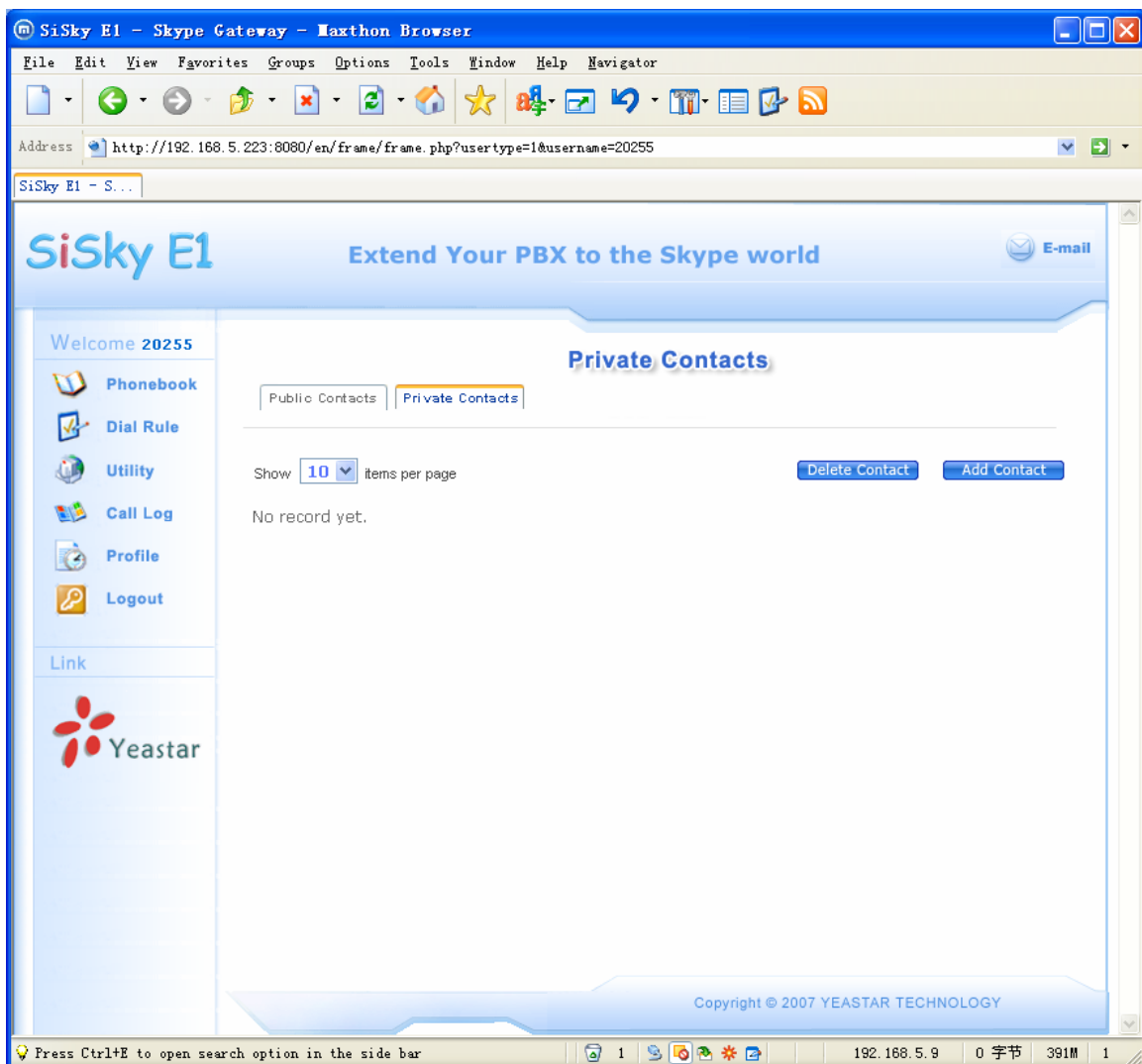


Figure 10-20

10.13.3 Dial Rule

Standard user has no right to modify, view only.

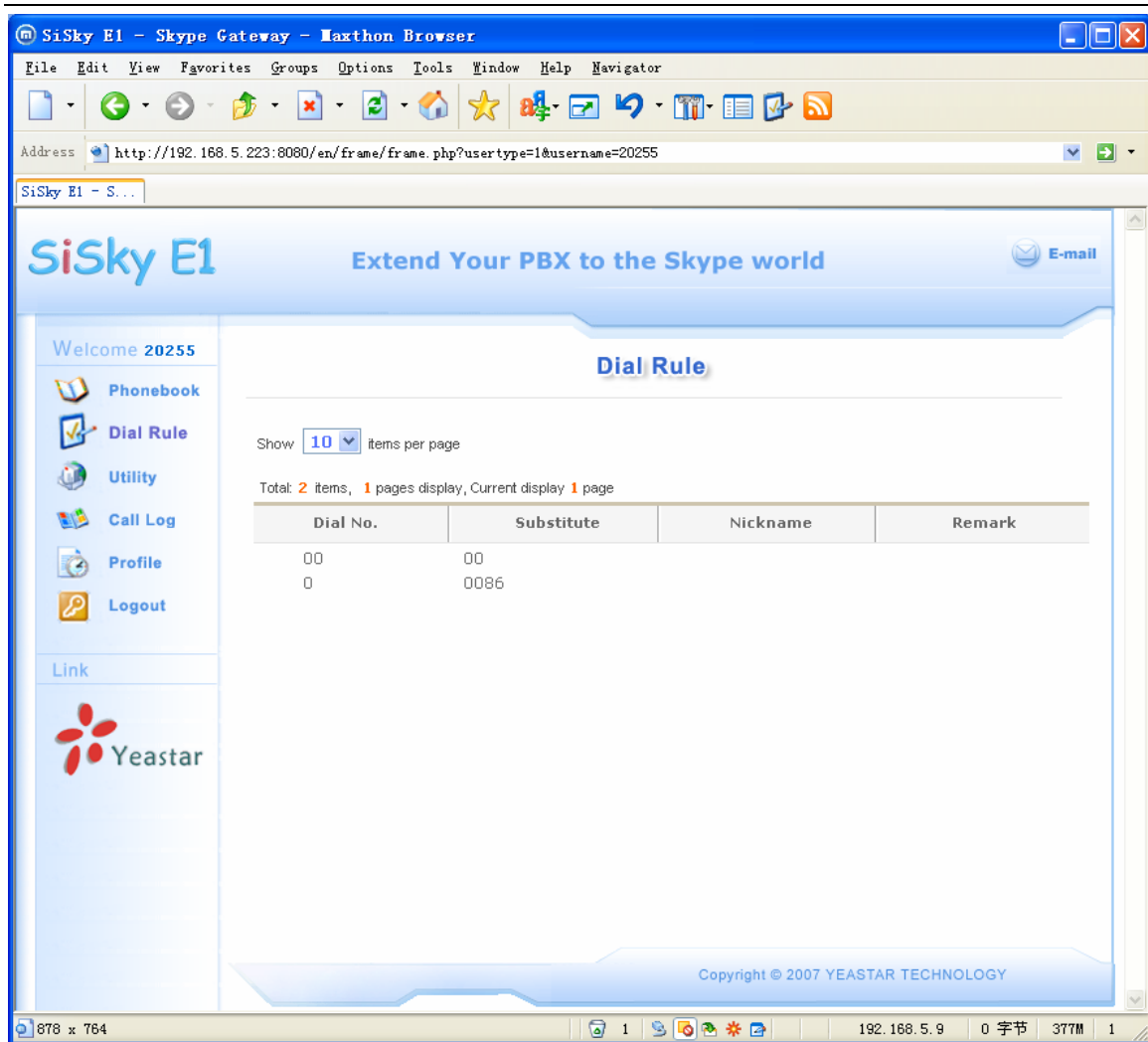


Figure 10-21

10.13.4 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 10.8.2](#). Logging in as standard user, the contacts will import to private phonebook rather than public phonebook.

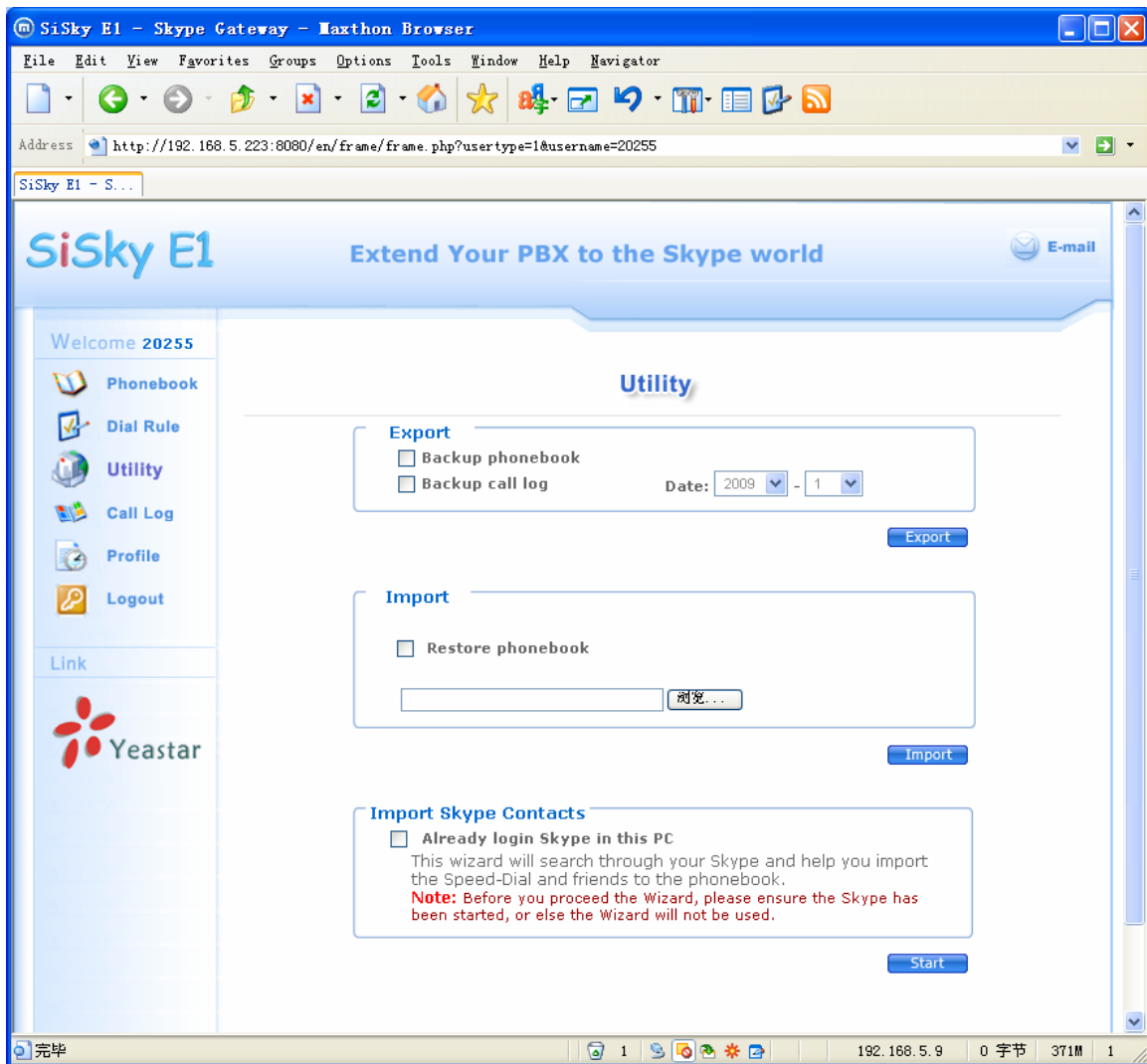


Figure 10-22

10.13.5 Viewing Private Call Log

10.13.6 Profile

You can change the login password here.

SiSkyE1 Advanced Application **11**

The main differences between SiSkyE1 Advanced Application and former chapter 8 Using SiSkyE1 is following:

Using under multi-user mode and Web800 application are only available on Advanced Application.

Make three examples for the below typical applications

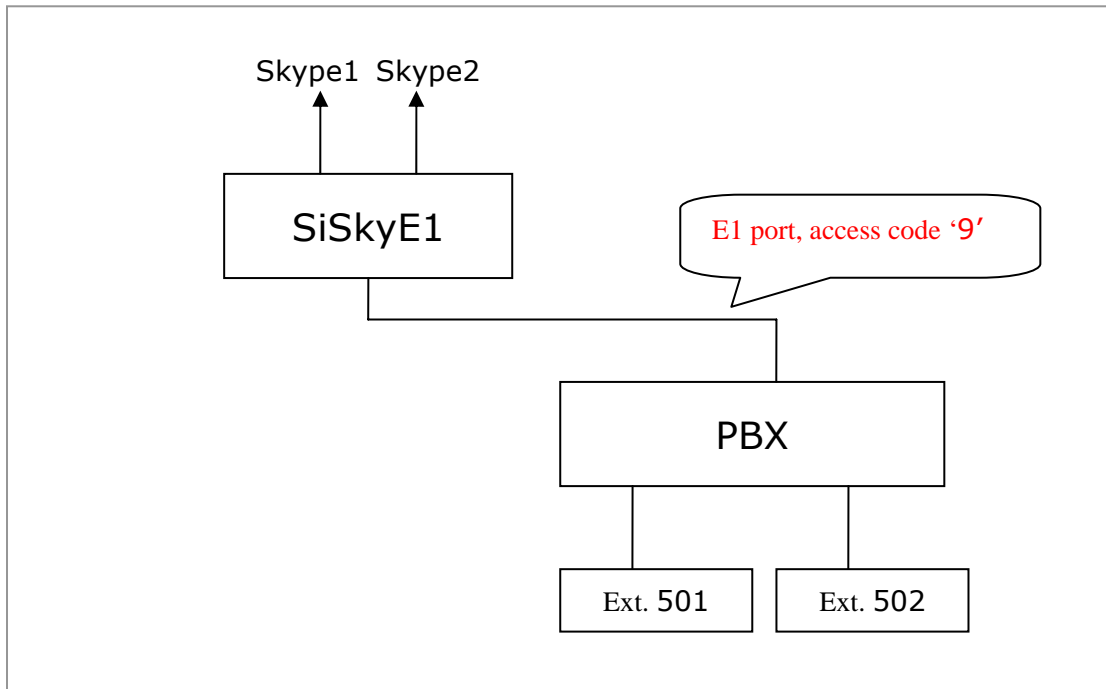
- Connections between branch offices
- Dialing SkypeOut to regular phone/mobile or Calling Skype ID
- Website Click-to-Call (SkypeMe) and receiving SkypeIn calls

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

11.1 Application of Branch Offices Connection

Through E1 port, SiSkyE1 can realize free communication between branch offices. For example, both Beijing and Shanghai offices are using SiSkyE1 Servers

11.2 Dialing SkypeOut or Calling Skype ID



On the above demonstrated environment:

11.2.1 Dialing SkypeOut Call to regular phone or mobile

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

Under Non-multiuser Mode:

Pick up→9 001312567234#

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

ick up→9 20255 001312567234#

11.2.2 Dialing Skype ID Call

If the user of Ext.Tel.501 wants to dial Skype ID 'echo123' :

Under Non-multiuser Mode:

The speed-dial number of ID 'echo123' in **Public Phonebook** is 123:

501 Pick up→9 123#

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

(1) The speed-dial number of ID 'echo123' in **Private Phonebook** is 123:

501 Pick up→ (hearing sound for second dial) 920255123#

(2) The speed-dial number of ID 'echo123' in **Public Phonebook** is 333:

501 Pick up→ (hearing sound for second dial) 9333#

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

Using SiSkyE1, you can make a Company Skype ID through two ways: apply an ID and public it on website directly as shown on figure11-1 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.

11.3.1 Showing Company Skype ID as SkypeMe on Website

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

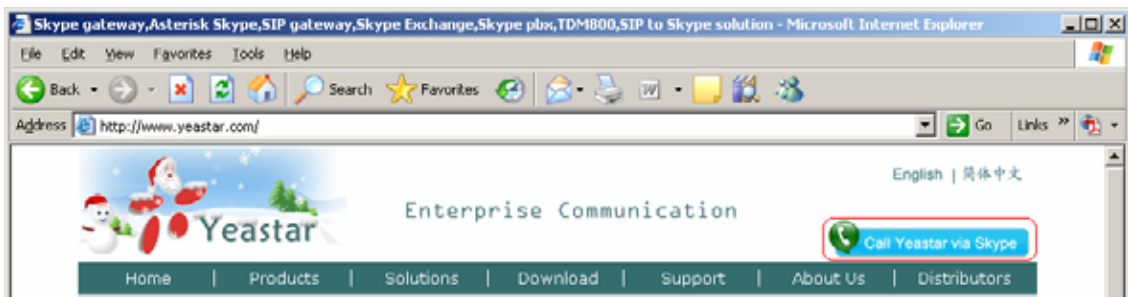


Figure 11-1

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

- Log in Skype on Company Skype ID (Skype1)
- Click 'Tools' → 'Options' on menu, as Figure 11-2
- Click on checkbox 'Allow my status to be shown on web' as Figure 11-3
- Save settings

Step 2: Public SkypeMe button on the website

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

```
<a href="skype: Skype1?call"></a>
```

Here *Skype1* stands for your Company Skype ID.

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

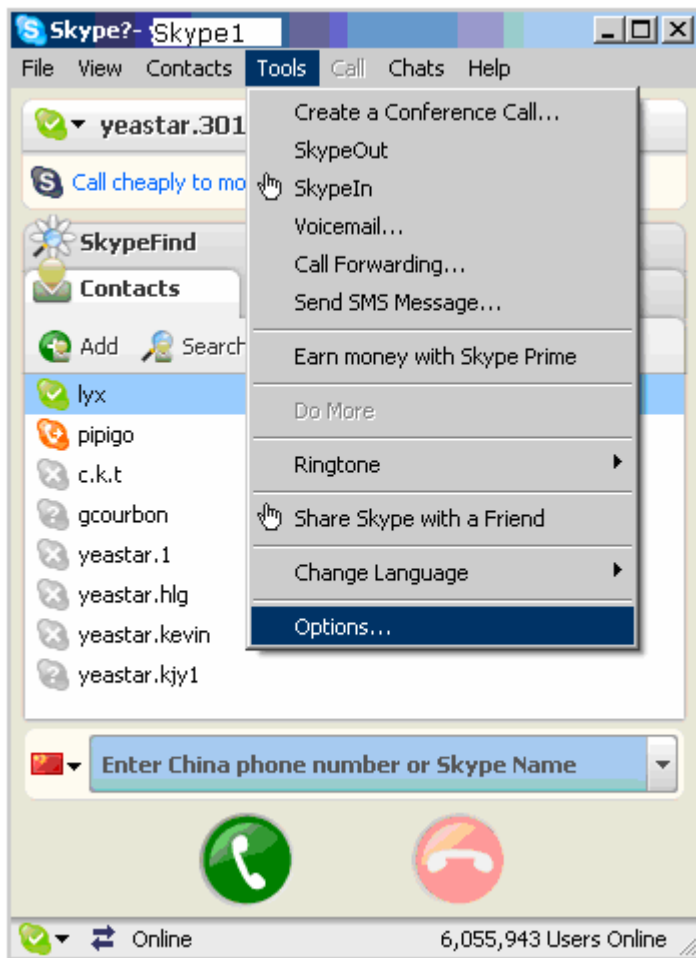


Figure 11-2

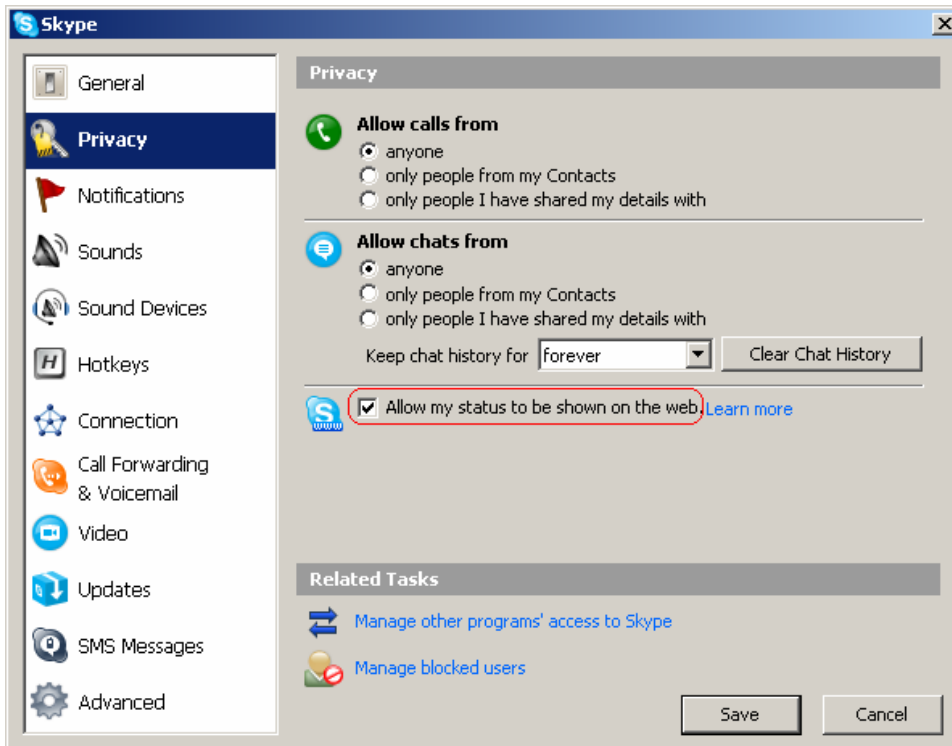
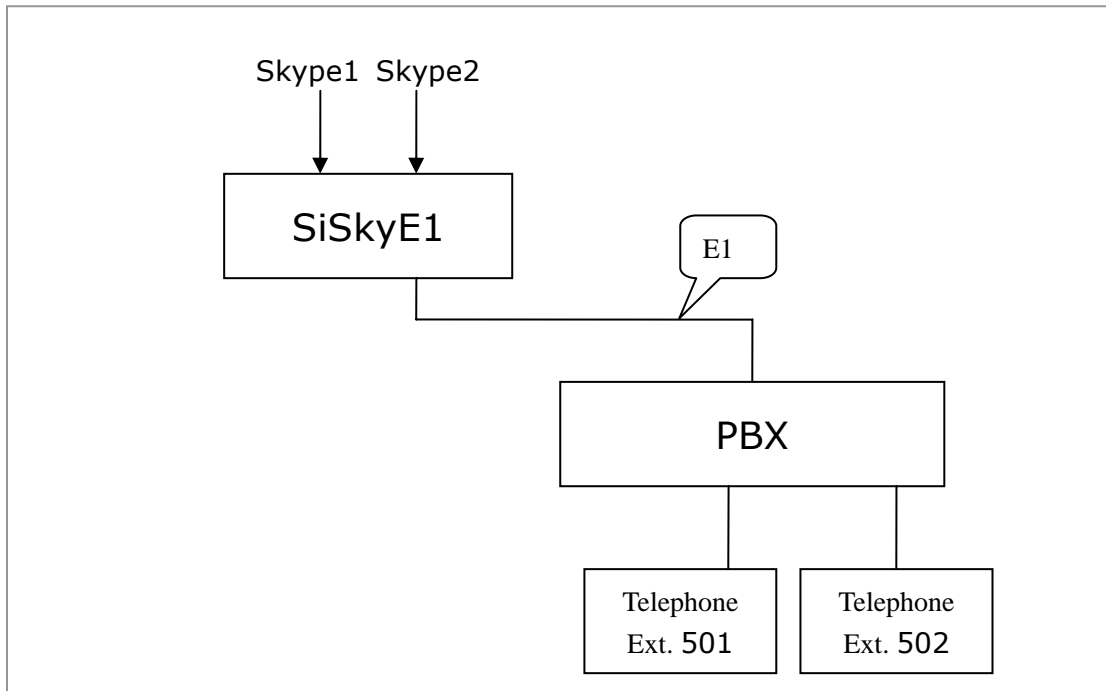


Figure 11-3

11.3.2 Skype Incoming Call



When there's an incoming call to Skype1 or Skype2, caller will first hear the music on hold and then hear an auto attendant from SiSkyE1. Caller will continue to dial the extension number. Except the music on hold, all are same as calls from PBX's PSTN line.

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

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)

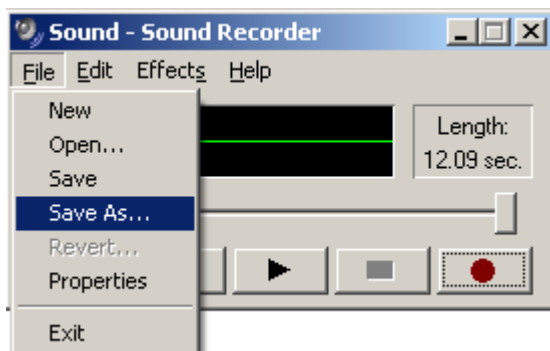
Step:

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

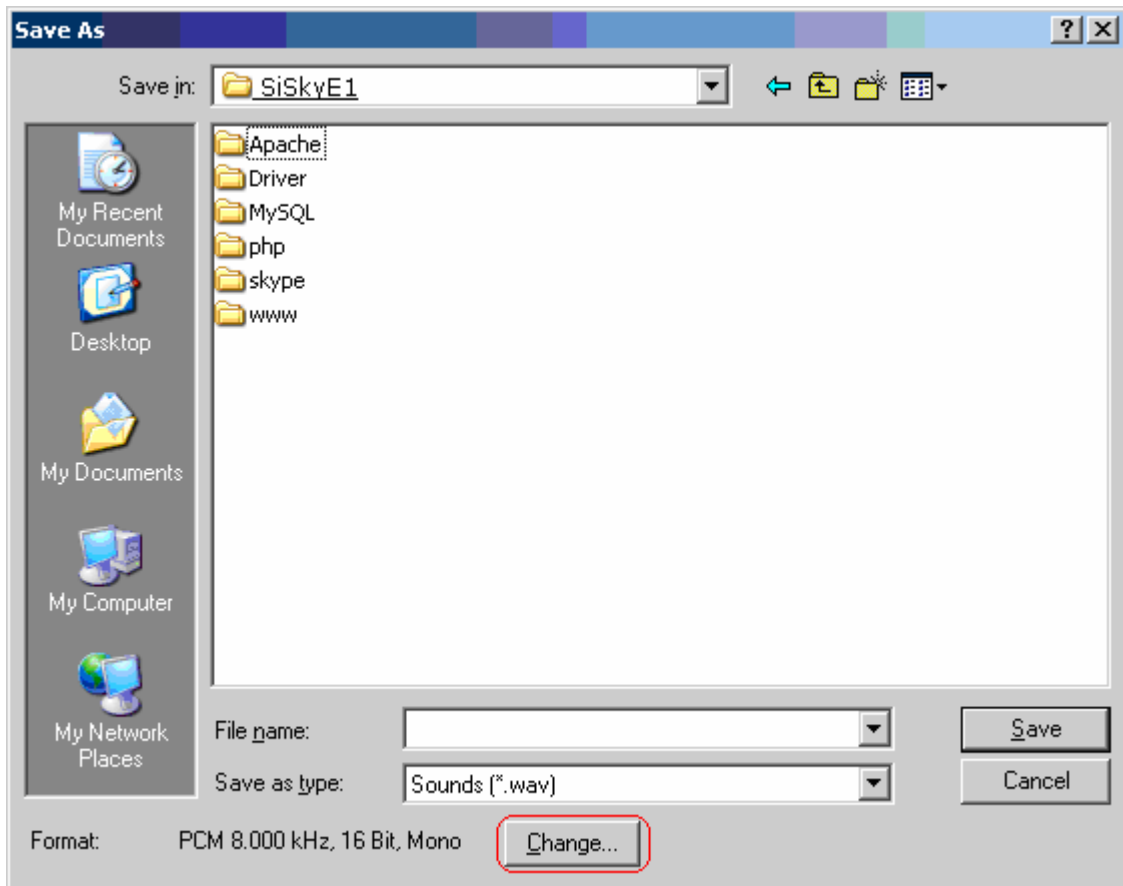
1. Record the prompt message file



2. Save it As.....



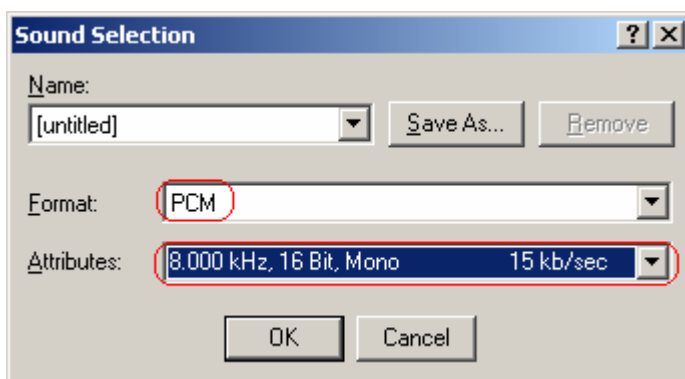
3. Change the format, click the button 'Change'



4. Choose an entry on 'Sound Selection'

Format: PCM

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



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

Appendix B Glossary

Busy tone: Low-pitched tone interrupted sixty times per minute. This tone indicates that the called station is off-hook or busy.

Dial tone: A constant high-pitched tone indicating that the system is ready to process your call. It's essential that users listen to this tone before dialing phone numbers.

DTMF (Dual Tone Multi-Frequency) : DTMF is used for most call set ups between the telephone set and telephone switch.

Off-hook: The handset is removed from the cradle or you are in conversation.

On-hook: The handset is in the cradle.

PBX (Private Branch Exchange) : It is a private telephone network used within an enterprise. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX.

PSTN (Public Switched Telephone Network) : It is the world's collection of interconnected Voice-oriented public telephone networks.

VoIP (Voice over Internet Protocol): A category of hardware and software that enables people to use the Internet as the transmission medium for telephone calls by sending voice data in packets using IP rather than by traditional circuit transmission of the PSTN.

Appendix C Country/Region List

SiSkyE1 supports the following countries' and region's telecommunication parameters.

ARGENTINA	LEBANON
AUSTRALIA	LUXEMBOURG
AUSTRIA	MACAO
BAHRAIN	MALAYSIA
BELGIUM	MALTA
BRAZIL	MEXICO
BULGARIA	MOROCCO
CANADA	NETHERLANDS
CHILE	NEWZEALAND
CHINA	NIGERIA
COLUMBIA	NORWAY
CROATIA	OMAN
CYPRUS	PAKISTAN
CZECH	PERU
DENMARK	PHILIPPINES
ECUADOR	POLAND
EGYPT	PORTUGAL
ELSALVADOR	ROMANIA
FINLAND	RUSSIA
FRANCE	SAUDIARABIA
GERMANY	SINGAPORE
GREECE	SLOVAKIA
GUAM	SLOVENIA
HONGKONG	SOUTHAFRICA
HUNGARY	SOUTHKOREA
ICELAND	SPAIN
INDIA	SWEDEN
INDONESIA	SWITZERLAND
IRELAND	SYRIA
ISRAEL	TAIWAN
ITALY	THAILAND
JAPAN	UAE
JORDAN	UK
KAZAKHSTAN	USA
KUWAIT	YEMEN
LATVIA	

Appendix D 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 SiSkyE1 work with SkypeIn?

A: SiSkyE1 can work with SkypeIn, especially SkypeIn can support concurrent multiple incoming calls after using SiSkyE1.

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.



6. Q: How can I set my Skype status always 'Online'?

A: You need to do nothing for it. SiSkyE1 will set it for you automatically and Skype of all trunks will be kept 'Online' status always.

B. SiSkyE1 Issues

1. Q: How can I do if I need more than 30 Skype trunks?

A: There's no limitness of SiSkyE1 trunks. You can install SiSkyE1 in more than one computer and link them together; trunks will be sharable among computers.

2. Q: Why the LEDs on card are still not lighted yet after I installed YE110 driver through running SiSkyE1 Installation Wizard?

A: Please check if the hardware YE110 card stays in good contact with PCI slot.

3. Q: How can I call my Skype contacts from the extension?

A: Before you start using SiSkyE1, login to the Web Management console to configure your phonebook and add your Skype contacts and speed dial key.

4. 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.

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