

User Manual

(Version: 2.6.0.1)

Yeastar Technology Co., Ltd



Table of Contents

0

	34
9.6 Phonebook	
9.5.2 Enable others except that on the transferring list	
9.5.1 IIdlistering List	
9.5 Managing Automatic Attendant	
9.4.2 Deleting User	
9.4.1 Adding Users	
9.4 User Management	
9.3.2 Managing Port's Settings, see Figure 49	
9.3.1 Timely display the status of every port	
9.3 Port Status and Management	
9.2 Administrator Logging in	
9.1 Open SiSky Homepage	
9. Managing SiSky Software	43
8.4 How to Create SIP Extension for SiSky without freePBX	
8.3 How to Create SIP Extension for SiSky with freePBX	
8.2 How to Create SIP Trunk for SiSky without freePBX	
8.1 How to Create SIP Trunk for SiSky with freePBX	
8. Configure Asterisk for SiSky	32
7.6 Audio Setting	
7.5 Database Setting	
7.4 Net Setting	
7.3 Sounds Setting	
7.2 General Setting	
7.1 Port Settings	
7. Building SiSky Operating Environment	23
6. Running SiSky Software	20
5.5 Finish Config. Wizard	
5.4 Install Skype Cables	
5.3 Configure Skype for Each Port	
5.2 Select Skype Trunk Number	
5.1 Launch the Config Wizard	
Running SiSky Config.Wizard 5	14
5. Installing SiSky Software	
3. Installation Procedure	
2. Before You Proceed	
1 3 Minimum System Requirements	5
1.1 Fullculous	4
1. 1 Eurotions	
1 Introduction	1



9.6.1 Adding Public Contact	54
9.6.2 Deleting Public Contact	54
9.7 Dial Rule	57
9.8 Backing up & Restoring	59
9.8.1 Database Backup and Restore	59
9.8.2 Importing Skype Contacts	60
9.9 Viewing Call Log	65
9.10 Viewing Statistics	66
9.11 System	67
9.11.1 Balancing engine	67
9.11.2 Skype Rules Settings	67
9.11.3 Other Settings	67
9.12 Password	68
9.13 Logging In As Standard Users	69
9.13.1 Logging In	69
9.13.2 Phonebook	69
9.13.3 Dial Rule	71
9.13.4 Backing up & Restoring	72
9.13.5 Viewing Private Call Log	73
9.13.6 Profile	73
10. Using SiSky	74
10.1 Application of Branch Offices Connection	74
10.2 Dialing SkypeOut or Calling Skype ID	76
10.2.1 Usages of Working as PBX's Trunk	76
10.2.2 Usages of working as PBX's Extension	77
10.3 Usages of Website Click-to Call (SkypeMe) & SkypeIn	79
10.3.1 Showing Company Skype ID as SkypeMe on Website	79
10.3.2 Skype Incoming Calls through SiSky	
Uninstalling SiSky Software 11	82
Appendix A Customizing WAV Format (Auto Attendant)	83
Appendix B FAQ	85



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 ca	ills	3	5~6	8~9	15	23	30
РС	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
Requirement	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 ca	alls	3	5~6	8~9	15	23	30
PC	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
Requirement	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.

Please don't login SiSky server by "Remote Desktop Connection", but use
 VNC in case you need remote login SiSky server.



Installation Procedure



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

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

- 1. Download Install software from website <u>http://www.yeastar.com</u>. 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.

월 Installing SiSk	y Enterprise Edition	×
K a	Welcome to the SiSky Enterprise Edition Installation!	
~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~	This setup program will install SiSky Enterprise Edition on your computer. Click Exit if you do not want to install this application. Click Next to continue the installation.	
	WARNING: This program is protected by international copyright law and treaties.	
	Unauthorized reproduction or distribution of this program, or any portion of it, may result in severe civil and criminal penalties and will be prosecuted to the maximum extent of the law.	
	<u>Next &gt;</u> <u>E</u> xit	

<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.



😼 Installing SiSky Enterprise Edition 🛛 🔀				
License Agreement To proceed with the installation, you must accept this License Agreement. Please read it carefully.				
Entering into this Agreement: SiSky EE This End User License Agreement constitutes a valid and binding agreement between Yeastar's SiSky software ('Software') and You, as a user, for the use of the Software.				
Article 1 License and Restrictions				
1.1 License. Subject to the terms of this Agreement, Yeastar hereby grants You a limited, personal or company, non-commercial, non-exclusive, non- sublicensable, non-assignable, non-free of charge license to download, install and use the Yeastar Software on Your computer.				
I agree with the above terms and conditions				
< <u>B</u> ack <u>N</u> ext > <u>E</u> xit				

<Figure 2>

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

📲 Installing SiSky Enterprise Edition 🛛 🔀
Readme Please read the following information.
SiSky Enterprise Edition is a software that generates Max. 30 ports Skype trunk for Asterisk. After add Skype trunks for your Asterisk, you can take full advantage of the largest free VoIP in the world to communicate and collaborate with clients and partners easily, efficiently and economically.
SiSky is a shareware, not need hardware support, you can try it freely. When you want to get license, click menu [Help] -> [Register Now]
Once installed, there will be a shortcut of SiSky EE on the desktop. Double click it and begins to run.
< <u>B</u> ack <u>Next &gt; E</u> xit

- <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.



🚰 Installing SiSky Enterprise Edition 🛛 🔀
Destination folder Select a destination folder where SiSky Enterprise Edition will be installed.
Setup will install files in the following folder.
If you would like to install SiSky Enterprise Edition into a different folder then click Browse and select another folder.
Destination folder
C:\Program Files\yeastar\sisky <u>B</u> rowse
< <u>B</u> ack <u>Next</u> <u>E</u> xit

#### <Figure 4>

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

🚝 Installing SiSky Enterprise Edition	×
Options Select your preferred option.	ļ
Select from the following installation options. Click Next to begin the installation. Click Back if you would like to change the installation information.	
<ul> <li>Launch SiSky Enterprise Edition After Installation</li> <li>Create Desktop Shortcut</li> </ul>	
< <u>B</u> ack <u>Exit</u>	

<Figure 5>

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



🛃 Installing SiSky Enterprise Edition 🛛 🔍 🗙
Installing Files Copying SiSky Enterprise Edition files to your computer.
To interrupt or pause the installation process, click Cancel.
Directory: C:\Program Files\Yeastar\SiSky File: SiSky.exe
<u>N</u> ext > <u>Cancel</u>

<Figure 6>

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

覺 Installing SiSky	/ Enterprise Edition	×
E A	SiSky Enterprise Edition has been successfully installed!	
4400 C	Click Finish to complete the installation.	
<b>N</b>		
	Einish	



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.

Message			×
?	You need to cor Do you want to	ıfig SiSky first! start Config Wi	zard now?
	<u>Y</u> es	No	

< 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

👬 SiSky Config Wizard	×
✓ Start to Config SiSky	
💠 Select Skype Trunk Number	
Config Skype Trunks	You can install SiSky on your system!
Install Skype Cables	Your Windows Version:
Finish Config SiSky	WINDOWS XP - Service Pack 2 (Build 2000)
	How many Skype Trunks will be installed
	<< Back Next >> Cancel

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.

S Skype?	- Create Account		×
S	Create a new Skype	e Account	
ă	Full Name * Choose Skype Name		
	* Password	Between 6 and 70 characters Minimum 4 characters	
	* Repeat Password		
	* 🔲 Yes, I have read an Skype Terms of Ser	d I accept the Skype End User License Agreement the vice and the Skype Privacy Statement	
	<ul> <li>Fields marked with an aste</li> </ul>	erisk are required	
		Next » Cancel	

<Figure 14>

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



🔕 Sk	уре						
<u>F</u> ile	<u>A</u> ccount	⊆all	⊂ <u>h</u> ats	⊻iew	<u>T</u> ools	H <u>e</u> lp	
	V	Vel	com	e to	Sk	ype	
	Skype	Name	:				
	yeasta 1 Do	r n't hav	e a Sky	oe Nam	e?		•
	Passw	ord					
	*****						
	Forgot	your pa	assword	?			
	🔽 Sig	ın me ir	when s	5kype s	tarts		
	🗖 Sta	art Sky	pe wher	) the co	mputer	starts	
						Sign i	n
<b>3</b> -	Not Con	necteo					1.

#### <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** 

Hardwa	re Installation
	The software you are installing for this hardware: Yeastar Skype Cable (Multiple Channels) (WDM) has not passed Windows Logo testing to verify its compatibility with Windows XP. ( <u>Tell me why this testing is important</u> .) <b>Continuing your installation of this software may impair</b> or destabilize the correct operation of your system either immediately or in the future. Microsoft strongly recommends that you stop this installation now and contact the hardware vendor for software that has passed Windows Logo testing.
	Continue Anyway STOP Installation

<Figure 17>

## 5.5 Finish Config. Wizard

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



👬 SiSky Config Wizard		×
<ul> <li>Start to Config SiSky</li> <li>Select Skype Trunk Number</li> <li>Config Skype Trunks         <ul> <li>1. Skype Trunk No.1</li> <li>2. Skype Trunk No.2</li> <li>3. Skype Trunk No.3</li> <li>4. Skype Trunk No.4</li> <li>5. Skype Trunk No.5</li> <li>6. Skype Trunk No.6</li> </ul> </li> <li>Install Skype Cables</li> <li>Finish Config SiSky</li> </ul>	Thank you for use SiSky! Please start SiSky after finish	
	<< Back Finish Cancel	

<Figure 18>

SiSky system configuration is complete!



# 6

**Running SiSky Software** 

**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.



S Skype?-	- Another program wants to use Skype	×
S	Another program wants to use Skype	
	Another program is trying to access Skype. This can be a potential security risk. What would you like to do?	
	S Name: <u>SkypeChannel.exe</u> Publisher: Yeastar Technology Co.,Ltd.	
	O Allow this program to use Skype	
	$\odot$ Allow this program to use Skype, but ask again in the future	
	$\odot$ Do not allow this program to use Skype	
	What does this mean?	
	OK	

<Figure 20>

<b>File</b> To	<b>y Enterprise Edi</b> ool Help	tion - SIP-Skyj	pe Gateway						_D×
) Start	Stop Optio	on Manage	(%) Config						
No. • 1 • 2 • 3 • 5 • 6 • 7 • 8 • 9 • 10	SIP Type Extension Extension Extension Extension Extension Extension Extension Extension	SIPID	Status	Skype ID yeastar.601 yeastar.602 yeastar.603 yeastar.604 yeastar.605 yeastar.606 yeastar.607 yeastar.608 yeastar.801 yeastar.802	Ready Yes Yes Yes Yes Yes Yes Yes Yes Yes	Call	Caller/Callee	Duration	
			T	Your first choice of Ar The newly improved d	nalog Interface esign TDM800	Card for <b>A</b> -A is highly	sterisk recommended.		

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

🐈 SiSk	y Enterprise Editi	ion - SIP-Skyp	e Gateway						
File To	ol Help								
) Start	Stop Option	Manage o	(Sonfiq						
No. 1 2 3 4 5 6 7 8 9 10	SIP Type Extension Extension Extension Extension Extension Extension Extension Extension	SIPID	Status Disable Show Skype Hang Up	Skype ID yeastar.601 yeastar.602 astar.603 astar.604 astar.606 astar.606 astar.607 astar.608 astar.801 yeastar.802	Ready Yes Yes Yes Yes Yes Yes Yes Yes Yes	Call	Caller/Callee	Duration	
				Your first choice of An The newly improved d	alog Interface esign TDM800-	Card for A	sterisk recommended.		

< Figure 22>

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

•		
🐈 Port 1 Setting		×
SIP Setting		
• Works as Asteri	sk'/IPPBX's Extension	
User ID	501	
Password	•••	
SIP Proxy	192.168.5.10 : 5060	
SIP Domain	192.168.5.10 Asterisk/IPPBX Server IP	
Expire Time	1800	
O Works as Asteri	sk'/IPPBX's Trunk	
User ID		ОК
Password		Cancel

< Figure 23>

#### SIP Setting:

. Yeastar

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.

🐈 SiSk	y Enterprise Edit	ion - SIP-Skyp	e Gateway						_ 🗆 🗵
File To	ool Help								
$\bigcirc$	8		<u></u>						
Start		Manage   0	Lonfiq						1
No.	SIP Type	SIPID	Status	Skype ID	Ready	Call	Caller/Callee	Duration	
🗎 1	Extension			yeastar.601	Yes				
0	Extension			yeastar.602	Yes				
	Extension			yeastar.603	Yes				
4	Extension			yeastar.604	res				
	Extension			yeastar.605	Yes				
II 🎽 7	Extension			veastar 607	Yee				
II 🍒 8	Extension			veastar 608	Yes				
9	Extension			veastar 801	Yes				
0	Extension			veastar.802	Yes				
				,					
11									
11									
11									
11									
<u> </u>									
			1			(	*		
				Your first choice of Ana	alog Interface	Card for A	sterisk		
				The newly improved de	sign TDM800-	A is highly	recommended.		
			1						
		L							

< Figure 24>

SiSky Options	X
General Sounds Net Database Audio	
✓ Start SiSky When I start Windows	
Delete Skype history automatically	
Keep SiSky call history for	forever 👻
Fixed SkypeOut call when Credit less than	0
Max.Concurrent Calls	5 👻
Answer SIP incoming call immediately(Nee	d Restart SiSky)
Add load balancing engine	
Max.limiting calling time per day (min.)	0
Max. different called numbers per day	0
Access IP Address:	
	(/
Note: Multiple If address separated by ";	"(e.g.1F1;1F2;1F3)
	确定 取消

<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.



 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

SiSky Options	×
General Sounds Net Database Audio	
Automatic Attendant (Incoming Call from Extension only)	
[default]	
Music on hold	
[default]	
Sound for second dial (Outgoing Call)	
[default]	
Sound for call failed	
[default]	
Sound for busy signal	
[default]	
Sound for insufficient SkypeOut Credit	
[default]	
Cancel	

<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 <u>Appendix A</u>.

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 <u>Appendix A</u>.

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

SiSky Options		×
General Sounds Net	Database Audio	
(Restart SiSky so as to r	nake new settings become effective.)	
🔽 Work SiSky with Ast	erisk	
🗖 Use STUN Server	stun01.sipphone.com	
Local Port	5060	
RTP Port Begin	10800 To 11000	
	OK Cance	

< 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

SiSky Options 🔀
General Sounds Net Database Audio
If SiSky is running on multi servers and this set is non-primary server, please connect to primary server and enter its IP:
Connect to Primary Server(Restart ExSky to take it effect)
Primary Server's IP
Connect Test
Backup the local data
Restore the local data
OK Cancel

< 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

SiSky Options	×
General Sounds Net Database Audio	
Codec preferences	
Codec Name	
G.711 U-Law Codec	Up
GSM 6.10 Codec	
	Down
	Default
I want to use the below Codec and got its licence already	
□ 6.729 Coded	
G.723.1 Codec	
	)K Cancel

< 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

8.2 How to Create SIP Trunk for SiSky without freePBX

8.3 How to Create SIP Extension for SiSky with freePBX

8.4 How to Create SIP Extension for SiSky without freePBX

### 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

Outgoing Settings Trunk Name: sip_trunk_901 PEER Details: canreinvite=no context=from-trunk fromdomain=192.168.5.8 fromuser=901 host=192.168.5.8 insecure=very secret=901 type=friend username=901			
Trunk Name: sip_trunk_901 PEER Details: canreinvite=no context=from-trunk fromdomain=192.168.5.8 fromuser=901 host=192.168.5.8 insecure=very secret=901 type=friend username=901	Outgoing Settings	2	
PEER Details: canreinvite=no context=from-trunk fromdomain=192.168.5.8 fromuser=901 host=192.168.5.8 insecure=very secret=901 type=friend username=901	Trunk Name:	sip_trunk_901	
canreinvite=no context=from-trunk fromdomain=192.168.5.8 fromuser=901 host=192.168.5.8 insecure=very secret=901 type=friend username=901	PEER Details:		
context=from-trunk fromdomain=192.168.5.8 fromuser=901 host=192.168.5.8 insecure=very secret=901 type=friend username=901	canreinvite=no	_	^
fromuser=901 host=192.168.5.8 insecure=very secret=901 type=friend username=901	context=from-trup fromdomain=192.16	1k 58.5.8	
host=192.168.5.8 insecure=very secret=901 type=friend username=901	fromuser=901		
Insecure=very secret=901 type=friend username=901	host=192.168.5.8		
type=friend username=901	insecure=very secret=901		
username=901	type=friend		
×	username=901		
			*

#### <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. *fromuse*r 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:

Registration	
Register String:	
901:901@192.168.5.8/9	01
Submit Changes	

<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

Dial Patterns		
	7].	
	~	
	Clean & Remove duplicates	
Dial patterns wizards:	(pick one) 😽	
Trunk Sequence		
	SIP/sip_trunk_901 🔽	
Submit Changes		

<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

Edit Incoming Route		
DID Number: Caller ID Number: <b>OR</b>	901	
Zaptel Channel:		

<Figure 34>

Core: 102 <102>
O IVR: 123 💌
C Recordings: 12345 🖌
Conferences: asterisk <800>
🔿 Ring Groups: 🛛 <600> 🔽
Custom App
Submit

<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.





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 <mark>=</mark> from-internal)
canreinvite=no
callerid=device <109>
disallow=all
allow=ulaw
allow=alaw
allow=g729
allow=g723.1

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

<Figure 39>

```
[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
```



freePBX 2.2.1 on 192.168.5.10	Setup Tools Reports Panel Recordings
Basic	
Administrators	Add SIP Extension
Extensions	
General Settings	
Outbound Routes	Add Extension
Trunks	
Inbound Call Control	User Extension (501)
Inbound Routes	Display Name 501
	5
	Extension Options
	Direct DID
	DID Alert Info
	Outbound CID
	Emergency CID
	Device Options
	This device uses sin technology
	secret (501)
	dtmfmodo fo ²⁹³³
	Fax Handling
	Fax Extension freePBX default

<Figure 43>

Apply Configuration Changes



freePBX 2.2.1 on 192.168.5	.to Setup Tools Reports Panel Recordings
Apply Configuration Cl	nanges
Basic	
Administrators	Add an Extension
Extensions	
General Settings	Please select your Device below then click Submit
Outbound Routes	Device
Trunks	
Inbound Call Control	
Inbound Routes	Device Generic SIP Device 🚩
	Submit freePBX 2.2.1 licensed under GPL :: UI Design @2006 Fischer Design, licensed under Creative Commons

<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



# 9

## Managing SiSky Software

## 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

<b># SiSk</b> File To	SiSky Enterprise Edition - SIP-Skype Gateway							<u> </u>	
) Start	Image: Start     I								
No.	SIP Type	SIP ID	Status	Skype ID	Ready	Call	Caller/Callee	Duration	
0	Extension	501	Registered	yeastar.601	Yes				
0 2	Extension	502	Registered	yeastar.602	Yes				
0 3	Extension	503	Registered	yeastar.603	Yes				
0 4	Extension	504	Registered	yeastar.604	Yes				
05	Extension	505	Registered	yeastar.605	Yes				
0 0	Extension	506	Registered	yeastar.606	Yes				
	Extension	507	Registered	yeastar.607	Yes				
0	Extension	508	Registered	yeastar.608	Yes				
	Trunk	901	Connected	yeastar.oui	Yes				
	Trunk	902	Connected	yeasiar.ouz	res			J	
Ľ									
		-9				1	<b>2</b>		
				Your first choice of Ana	alog Interface (	Card for As	sterisk		
				The newly improved do	sign TDM800-	A is highly	recommended		
				The newly improved de	sign i DivioUU-	A IS HIGHLY	recommended.		
			ur in the second s						

#### < 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 <a href="http://192.168.0.101:8080">http://192.168.0.101:8080</a> 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.

🚰 SiSky Enterprise Edition - SIP to Skype Gateway - Microsoft Internet Explorer	
Elle Edit View Favorites Tools Help	<b>A</b>
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Address 🕘 http://127.0.0.1:8080/	🔽 🔁 Go 🛛 Links 🌺 👻
Sip to Skype Gateway         Image:	Version: 2.0.0.1
Copyright © 2007YeaStar Technology, Co., Ltd. All rights reserved. 🔯	
	V Constant
	Turenec //

<Figure 47>

If the next page is abnormal after you login, that mainly because of the safety setting of IE browser. Click <u>here</u> 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

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Address 🗃 http://127.0.0.1:8080/en/frame/frame.php?usertype=0&username=admin 🔽 💽 Go 🛛 Links » 📆 🗸								
Extend Your IP PBX to the Skype world								
SISKy Extend four IP Pox to the skype world								
					<u> </u>			
Weld	come Admin							
2	Port				Port	List		
<u></u>	User							
R	Attendant	No.	SIP Mode	SIP Account	SIP Status	Skype ID	Skype Status	Skype Credit
m	Phonobook	<u>⊖ 1</u>	Extension	501	Registered	yeastar.601	YES	0
<u> </u>	Phonebook	. <u>9</u> _2	Extension	502	Registered	yeastar.602	YES	0
1	Dial Rule	● <u>3</u>	Extension	503	Registered	yeastar.603	YES	0
١	Utility	<u>● 4</u>	Extension	504	Registered	yeastar.604	YES	0
	Call Log	⊖ <u>  5  </u>	Extension	505	Registered	yeastar.605	YES	0
	Call Log	⊖ <u>6</u>	Extension	506	Registered	yeastar.606	YES	0
-	Statistics	<u>● 7</u>	Extension	507	Registered	yeastar.607	YES	0
<b>1</b>	System	⊖ <u>8</u>	Extension	508	Registered	yeastar.608	YES	0
6	Descoursed	<u>9</u>	Trunk	901	Waiting	yeastar.801	YES	0
-	Password	⊖ <u>10</u>	Trunk	902	Connected	yeastar.802	YES	0
$\square$	Logout							
Link								
1	Yeastar							
	Custan					Copyright © 2007 YE/	ASTAR TECHNOLO	GY
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ど Done	🖹 Done							

<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 Enternrise Edition - Mic	icrosoft Internet Explorer								
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Sisky	CICIN Extend Your TP PBX to the Skype world								
JUNY									
Welcome Admin									
Port	Port List								
Ser User	Port 1 Setting								
Attendant	Work as Asterisk'/IPPBX's Extension     O Work as Asterisk'/IPPBX's Trunk								
M Phonebook	SIP Account: 501 SIP Account:								
	Password: Password:								
Dial Rule	SIP Proxy: 192.168.5.10 : 5060								
Utility	SIP Domain: 192.168.5.10								
Call Log									
Statistics	Skype								
NS Sustan	☐ Allow this Skype Status to be shown to everyone								
System	Use Skype: Yes Skype Status: YES Skype ID: veastar.601 Skype Balance: EURO								
Password									
Logout	Other								
Link	(always dial this number for incoming calls)								
LINK	Direct Out:								
	(always dial this number for outgoing calls)								
Yeastar	C No Transfer								
	Save								
	Copyright © 2007 YEASTAR TECHNOLOGY								
A Done	🔰 👘 Internet	1.							

<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 <u>chapter 9.13</u>---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.

🖉 SiSky Enterprise Edition - M	1icrosoft Internet Explorer		
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Address 🗃 http://127.0.0.1:808	80/en/frame/frame.php?usertype=0&username=admin		🔽 🄁 Go 🛛 Links » 🐑 🔹
SiSky	Extend Your IP PBX to t	he Skype world	🕞 Home 🍚 E-mail
Welcome Admin			
Port	Add	User(PIN)	
States User			
Attendant	* User (RIN)		
M Phonebook	* Password:		
Diel Bule	* Confirm Password:		
Dial Rule	Nickname:		
Utility	Ext. No.:		
Call Log	Cell Phone No.:		
Statistics	Skype ID:		
195° Suntan	Comment:		
System	Note: Usernan	ne(PIN) must be digits.	
Password	Save	Reset	
Logout			
Link			
/ Yeastar		Copyright © 2007 YEAST	AR TECHNOLOGY
			<b></b>
E Done			🔏 🚺 🔮 Internet 🥢

<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 Editior	- Microsoft Internet Exp	lorer					<u>- 🗆 ×</u>		
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorite	es <u>T</u> ools <u>H</u> elp						- <b>-</b>		
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Address 🕘 http://127.0.0.1	:8080/en/frame/frame.php?i	usertype=0&username=-	admin		•	🔁 Go 🛛 Links	» 🔁 🗸		
							<u> </u>		
SiSky Extend Your IP PBX to the Skype world									
Walcome Admin									
Welcome Admin									
Port			User(PIN	I) List					
🤮 User	User Mode								
💰 Attendant	⊙ Enable	Multi-user Mode	C Disable Multi-	-user Mode		Save			
D Phonebook									
Dial Rule	Show 10 Titer	ns per page			Delete User	Add User			
🕡 Utility	Total: 1 items, 1 pa	ages display, Current dis	olay <b>1</b> page						
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Password									
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<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 <u>chapter 7.3.1</u>) 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.



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<Figure 52>



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SiSky	Extend Your IP PBX	to the Skype world	Home E-mail
Welcome Admin			
Port	Add	Transferring Item	
Stendant			
Bharshock	* Dial(DTMF):	1	
Dial Rule	* Extension:	301	
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Utility		Skype ID	
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Logout			
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Done			V Internet

<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 <u>chapter 9.8.2</u> 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.



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Si	Extend Your IP PBX to the Skype world									
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	Port				Public Conta	ICIS				
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8	Attendant	onon [								
<b>V</b>	Phonebook		Key	SkyneID / Phone	Cell Phone	Nickname	F-mail			
1	Dial Rule		123	echo123	con r none	Skype Test	2			
	Utility		333	00865925503309						
	Call Log		888 999	<u>veastar.sales</u> veastar.support	008613950057025 008615960283407	Sales Supports	sales@yeastar.com support@yeastar.com			
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<Figure 55>



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Sisky	Extend Your IP PBX to the Skype work	d 🕞 Home 🍚 E-mail
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8 Attendant	* Speed-Dial Key:	
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<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.

- 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;
- 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.



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<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



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<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.

<mark>§</mark> Skype™f	or Internet Explorer (BETA)	×
S	A web page is attempting to use Skype contact management	
	A web page at 127.0.0.1 wants to use Skype contact management to create and manage contacts and contact groups.	
	C Allow this site to use Skype contact management	
	$\odot$ Allow this site to use Skype contact management, but ask again in the future	
	$\bigcirc$ Do not allow this site to use Skype contact management	
	Cancel	



#### <Figure 61>

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

🖉 http://127.0.0.1:8080 - SiSky Enterprise Ed	tion - SIP to Skype Gateway - Micros	soft Internet Explorer	
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Uone			11.

<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.



#### SiSkyEE User Manual

ttp://127.0.0.1:8080 - SiSk	y Enterprise Edition -	- SIP to Skype Gateway - Microsoft Interr	net Explorer	_ []
		Step 2		
<ul> <li>Note: Before importing Skype Name without sp</li> </ul>	) Skype Contacts, pl ieed-dial will be unav	ease set the Speed-dial for the Skype Na vailable on the ExSky's phonebook.	ame, or else those	
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Total: 10 Contacts	Encod Dial	Skupa Nama		
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2	32	yeastar.82		
		yeastar.81		
		yeastar.4		
2	23	yeastar.502		
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<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.



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<Figure 64>

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<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:

- 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.

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#### <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 <u>7.2 General Setting</u>. **Sounds** Setting, please refer to <u>7.3 Sounds Setting</u>.



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Welcome Admin	System Setting	
See User	Global	
<ul> <li>Attendant</li> <li>Phonebook</li> <li>Dial Rule</li> </ul>	Delete Skype history automatically: Ores ONO Keep SiSky call history for: formed form	
Utility Call Log	Max. limiting calling time per day(min): Max. different called numbers per day: Access IP Address: Note:Multiple IP address separated by ";", e.g.IP1;IP2;IP3.	3
System	Apply	
Password	Alert         Fixed SkypeOut call when Credit less than:       12       (0 = disabled)	
Link	Skype auto reply message:	
• Yeastar	Apply	
	Sounds Automatic Attendant (Incoming Call from Extension only) Music on hold Sound for second dial (Outgoing Call) Sound for call failed Sound for Busy signal Sound for insufficient SkypeOut Credit	
完成		🔍 100% 🔹

<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.

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SIP to Skype Gateway	
	Version: 2.0
User Login	
Password: • Language: English • Login Reset	
If you can't access to next page, click(here)for help	
Copyright © 2007 YeaStar Technology, Co., Ltd. All rights reserved.	

<Figure 69>

If the next page is abnormal after you login, that mainly because of the safety setting of IE browser. Click <u>here</u> 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.



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Profile	Key	SkypeID/Phone	Cell Phone	Nickname	E-mail			
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• Yeastar			Сор	yright © 2007 YEASTAF	RTECHNOLOGY			

<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



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<Figure 71>

#### 9.13.3 Dial Rule

Standard user has no right to modify, view only.



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Welc	Welcome 20255 Dial Rule							
$\square$	Phonebook							
<u>-</u>	Dial Rule	Show <b>10</b> items per p	age					
	Utility	Total: 2 items, 1 pages dis	play, Current display 1 page					
<b>1</b>	Call Log	Dial No.	Substitute	Nickname	Remark			
Ò	Profile	001 0	00 0082	IDD DDD				
	Logout							
Link								
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	reastar							
	Copyright © 2007 YEASTAR TECHNOLOGY							
ど Done					👋 🔯 Internet 💋			

<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 <u>chapter 9.8.2</u>. Logging in as standard user, the contacts will import to private phonebook rather than public phonebook.


🖉 SiSky Enterprise Edition - M	icrosoft Internet Explorer	
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	Copyright © 2007 YEASTAR	START
A Done		

<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 receiing 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:

<u>Through Skype 1:</u>

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

Or

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

Through Skype 2:

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

### 10.1.2 Shanghai Calling Beijing

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

<u>Through Skype 3:</u>

Pick up  $\rightarrow$  850 (hearing SiSky's Auto Attendant) $\rightarrow$  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) :

<u>Through Skype1:</u>

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 Phoneboo**k 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 $\rightarrow$ 85 20255(hearing sound for second dial)  $\rightarrow$ 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 $\rightarrow$ 85 20255(hearing sound for second dial)  $\rightarrow$ 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  $\rightarrow$  505 (hearing sound for second dial)  $\rightarrow$ 001312567234#

Or

Pick up→ 505 001312567234#

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

Through Skype 1:

Pick up  $\rightarrow$  505 (hearing sound for second dial)  $\rightarrow$  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 Phoneboo**k is 999:

<u>Through Skype 1:</u>

Pick up  $\rightarrow$  505 (hearing sound for second dial)  $\rightarrow$  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 $\rightarrow$ 505 (hearing sound for second dial)  $\rightarrow$ 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  $\rightarrow$ 505 (hearing sound for second dial)  $\rightarrow$ 20255 999#

Or

Pick up →505 20255 999#

Or

Pick up  $\rightarrow$  505 (hearing sound for second dial)  $\rightarrow$  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.

🗿 Skype gateway,Asterisk Skype,SIP gateway,Skype Exchange,Skype pbx,TDM800,SIP to Skype solution - Mic	rosoft Internet Explorer	<u>_   ×</u>
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Home   Products   Solutions   Download   Support   Ab	oout Us   Distributors	

<Figure 74>

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

- Log in Skype on Company Skype ID (yeastar.301)
- Click 'Tools'  $\rightarrow$  '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: <a href="skype: Skype1?call"><img src="http://mystatus.skype.com/smallclassic/Skype1" style="border: none;" width="114" height="20" alt="My status" /></a>

Here *Skype1* stands for your Company Skype ID.

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





### <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



Save As								<u>? ×</u>
Save jn:	🗀 ExSky				•	(= 🖻 🖻	* 🎟 🕇	
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My Computer								
My Network	File <u>n</u> ame:						•	<u>S</u> ave
Places	Save as <u>t</u> ype:	Sounds	(*.wav)			1	•	Cancel
Format: P0	CM 8.000 kHz, 16 B	it, Mono	<u>C</u> hange	a				

<Figure 79>

4. Choose an entry on 'Sound Selection'

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

Sound Sele	ection
<u>N</u> ame:	
[untitled]	Save As <u>R</u> emove
<u>F</u> ormat:	(PCM)
<u>A</u> ttributes:	(8.000 kHz, 16 Bit, Mono 15 kb/sec 💌
	OK Cancel

<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.