Instruction of the SiSky Enterprise Edition Software

KEYWORD: SiSky, BizSkype, SIP, Skype, SIP to Skype, Skype to SIP, Trunks, Business, Asterisk, Trixbox, IPPBX,

Installation

1. Download and Install SiSky software

2. Configure Asterisk for SiSky

According to <Asterisk Configuration for SiSky>,
Download < Asterisk Configuration for SiSky >from:
http://www.yeastar.com/download/AsteriskConfig.pdf

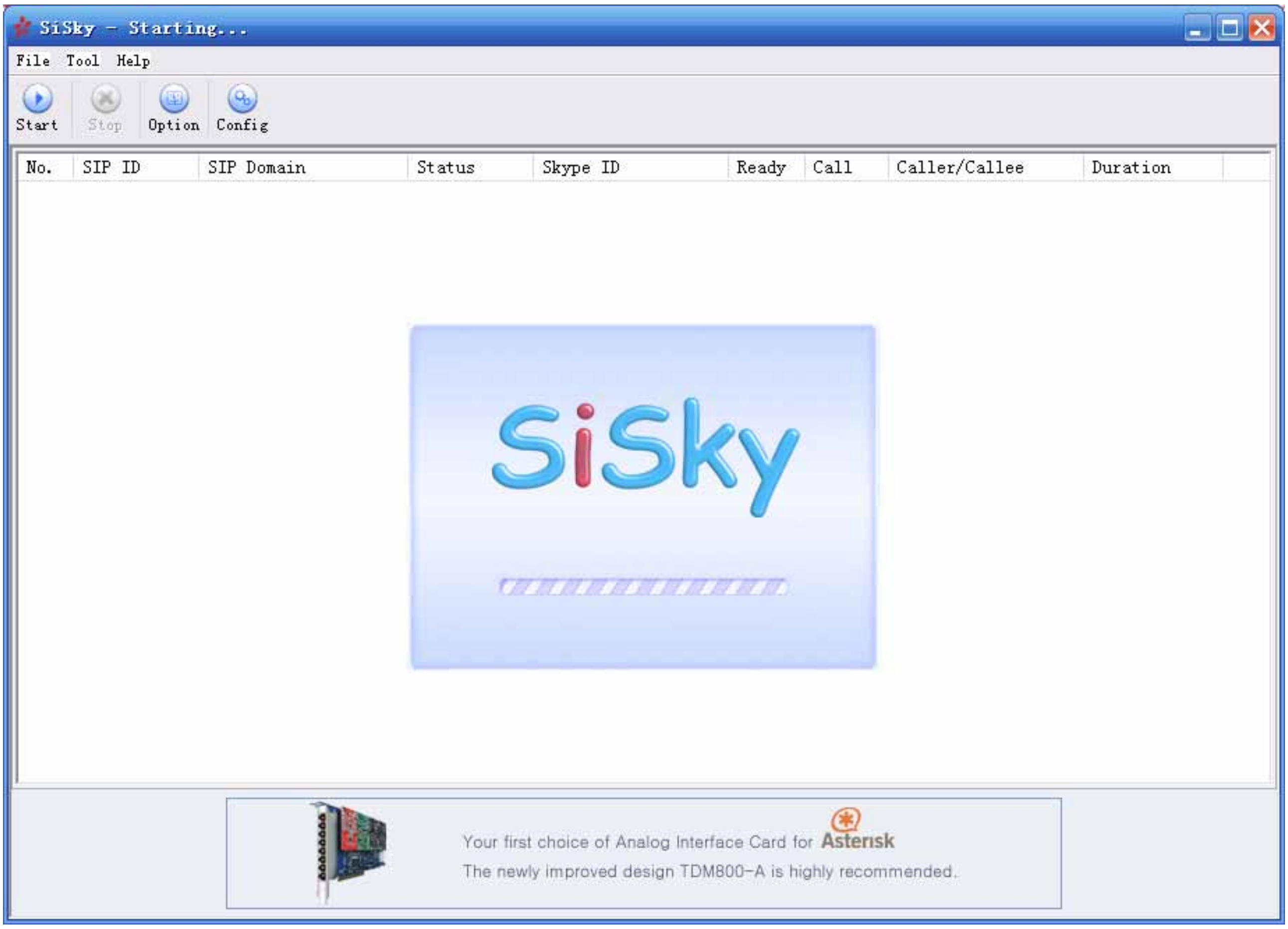
3. Configure SiSky

According to <SiSky Config Wizard> and <SiSky's SIP Endpoints Configuration>
Download <SiSky Config Wizard>from:
http://www.yeastar.com/download/SiSkyEEConfigWizard.pdf
Download <SiSky's SIP Endpoints Configuration>from:
http://www.yeastar.com/download/SiSkyEESIPConfig.pdf

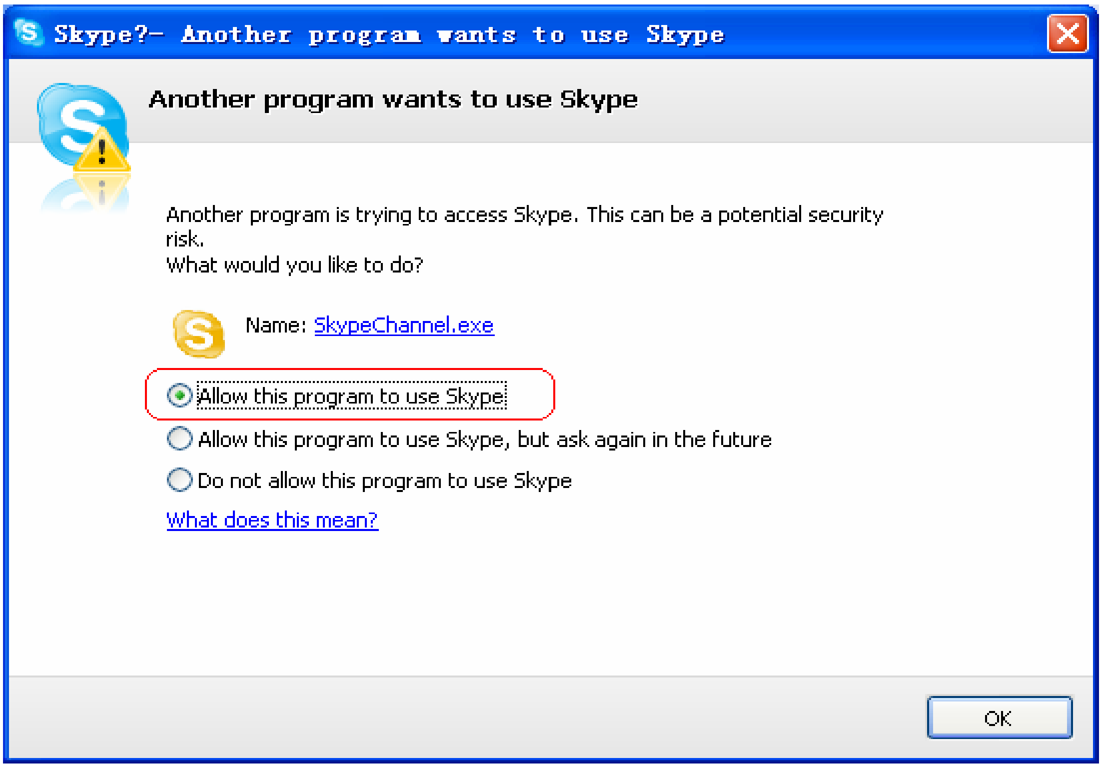
Please confirm you have been successfully finished the above steps:

Set Up

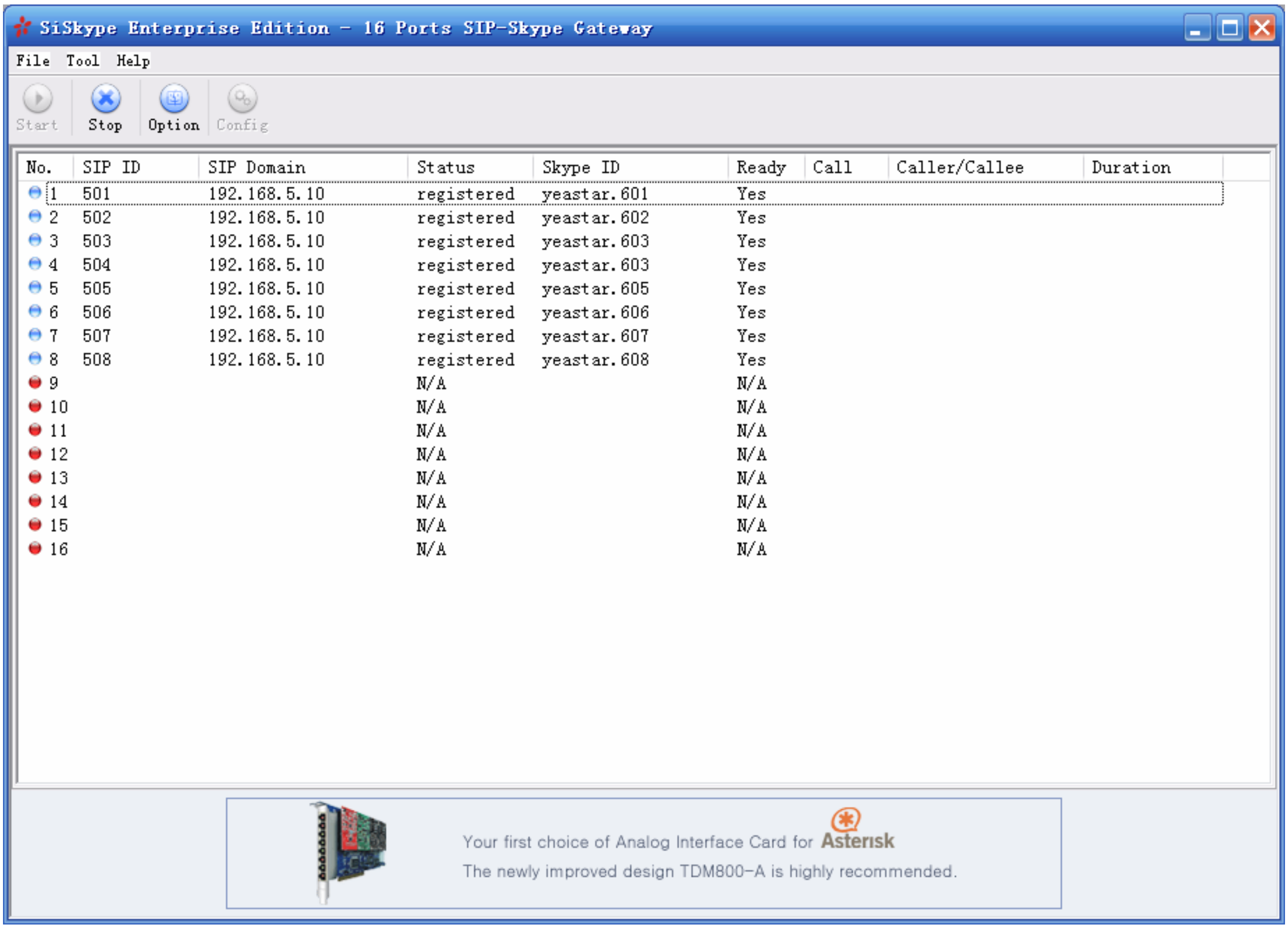
Step 1. Double-click the shortcut on desk to run the software, please wait for few seconds to Sign in all your Skype ID.



Step 2. Allow SkypeChannel.exe program to use Skype for every Skype ID



Step 3. Check if the status of SIP ports are 'registered' and Skype are ready.



Description of all items:

No. :

1) : Red light means failed initialization, port unavailable or Skype account unavailable
2) : Blue light means succeeded initialization, port corresponded line is idle
3) : Green light means succeeded initialization, port is in service

SIP ID:

SIP ID corresponds to the port number.

SIP Domain:

SIP Domain corresponds to the SIP ID.

SIP Status:

1) N/A: This SIP Port is not enabled
2) Registering: Registering or failed to register
3) Registered: Register have successfully finished

Skype ID:

Skype ID corresponds to the port number

Skype Ready:

1) N/A: this port is not configured Skype yet
2) Yes: This Skype is ready to use
3) No: This Skype is unready to use

Call:

1) Call in
2) Call out

Caller / Callee:

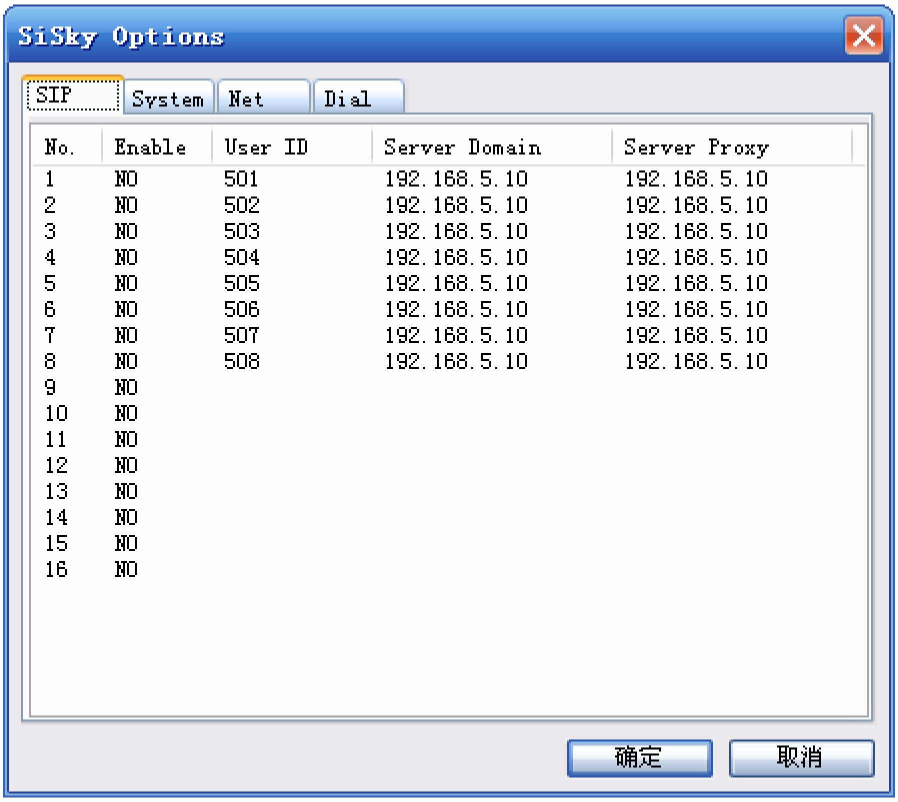
The telephone number or Skype ID of the other side

Duration:

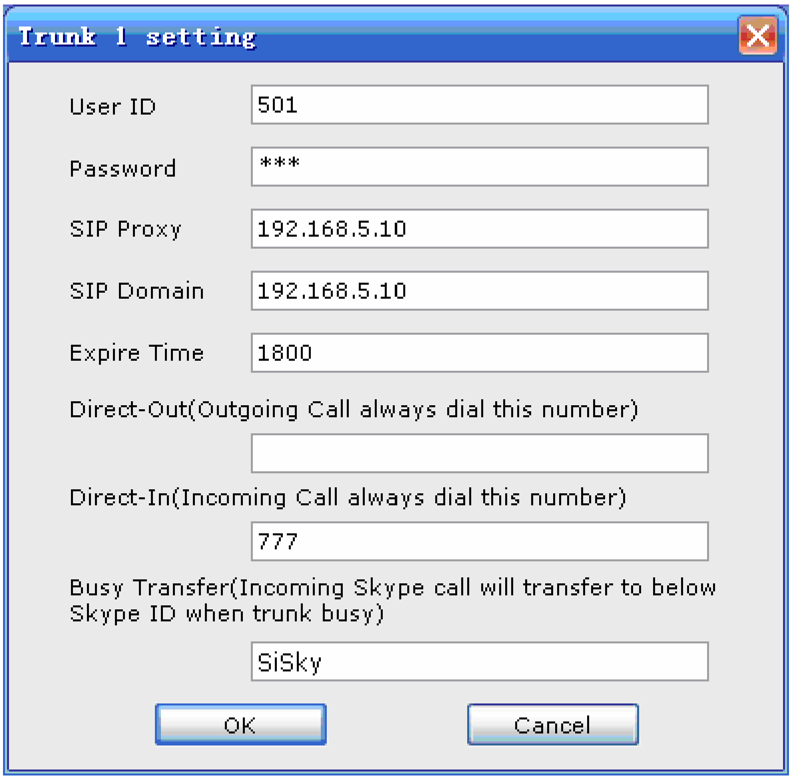
The duration of speaking

Step 4. Set SiSky Options

1）For ‘SIP’



The SIP port setting is description in <SiSky’s SIP Endpoints Configuration>



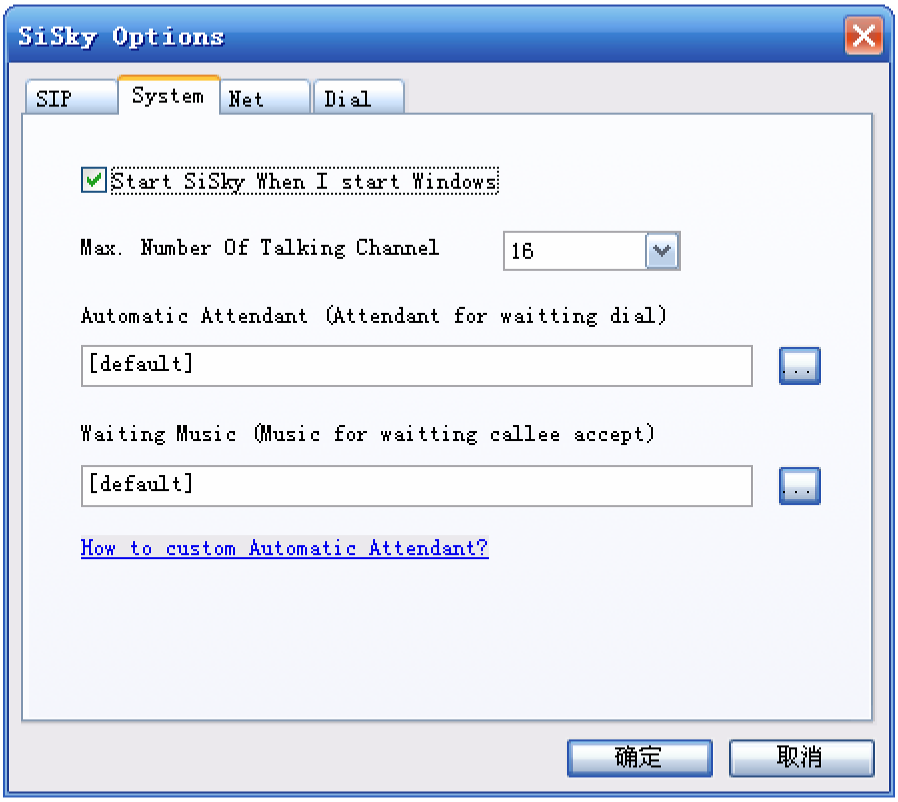
- Direct-Out: If you set up this direct number, every time when this port is off-hook, SiSky

will help you directly dialing this number, and through this port can only dial this fixed number. Generally this setting is used to connect with branch offices.
Note: You can input multi-item (Skype ID) and set off them by a semi-colon (e.g. no1;no2;no3). It means a call unsuccessful to first item, it will be passed to next item automatically.

- Direct-In: If you set up this direct number, every time when this port is off-hook, SiSky will help you directly dialing this number, and through this port can only dial this fixed number.

- Busy Transfer: When you set up this function, even if the port is busy, SiSky will transfer calls to those idle Skype IDs.
  Note: You can input multi-item (Skype ID) and set off them by a semi-colon (e.g. no1;no2;no3), It means a call unsuccessful to first item, it will be passed to next item automatically.
  You can use "SiSky" instead of all trunk's Skype ID. For Example: As Picture 3, "SiSky" equals to "yeastar.601; yeastar.602; ...... yeastar.608".

2) For ‘System’



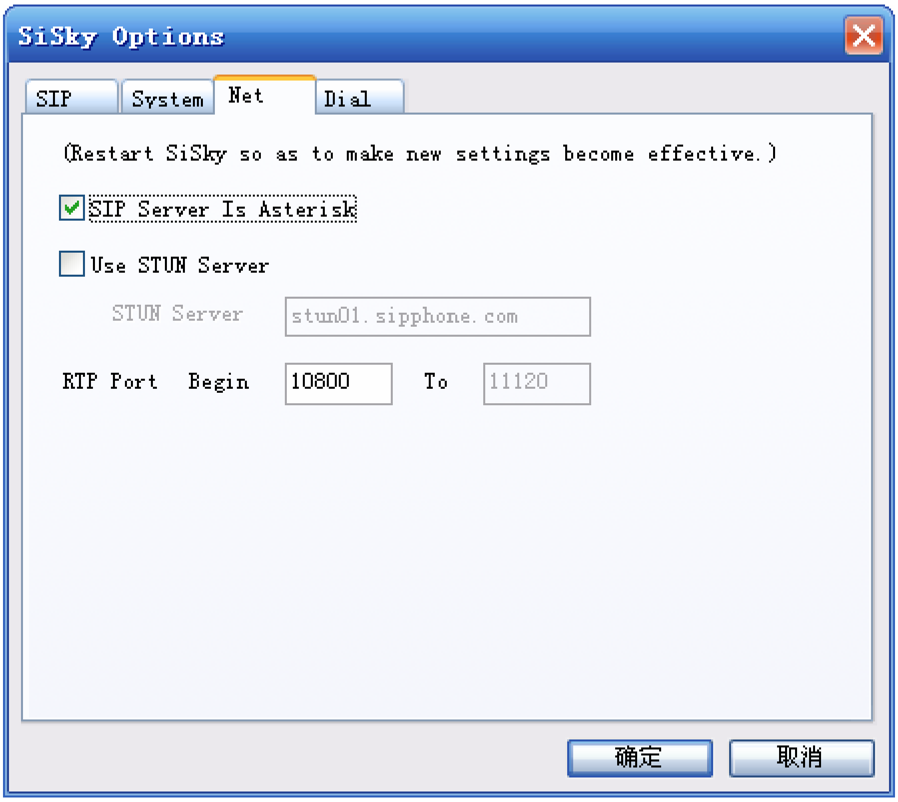
- Select Max. Number of Talking Channel. It depends on your PC performance. For your information:

Simultaneous Talking Ports		3	5~6	8~9	15
PC Requirement	CPU	Celeron	P4 2.66G	P4 2.8G	Intel Core Duo

		2.8G		Dual Core	1.86G
	Memory	512M	1G	1G	1G

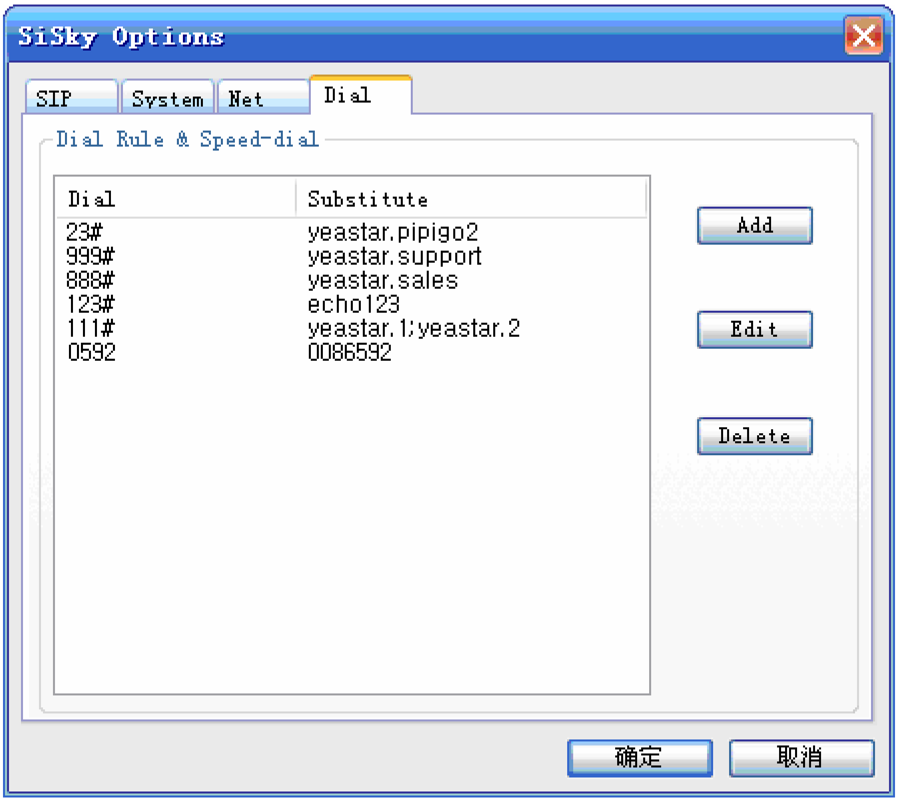
- Automatic Attendant and waiting music are set as default
  You can select a new WAVE file (8000Hz,16Bit,mono) for your automatic attendant or waiting music.
- Check ‘Start SiSky When I start Windows’

3) For ‘Net’



- If your SIP server based on Asterisk, Check it; if not, uncheck it.
- Use STUN Server for SIP Endpoints.
- RTP Port for SIP Endpoints. Please don’t change it generally.

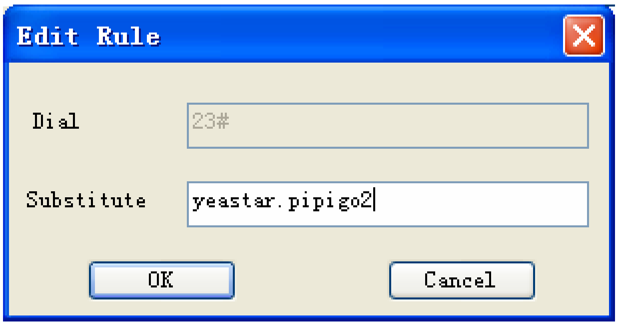
4）For ‘Dial’



You can define Dial Rule or Speed-dial here for all ports. The priority of Dial Rule is in Descend of dial number

5) For 'Edit Rule'

- 'Dial' is the prefix of dialing telephone number and 'Substitute' is the replacement of telephone number.



For example, set up the Dial Rule as following:

No.	Dial	Substitute
1	00186	0086
2	0592	0086592
3	23#	yeastar.pipigo2
4	1#	00865921234567
…	…	…

- Keep the dialing habit as same as PSTN

For example, the IDDD (international direct distance dialing) for a PSTN user in Korea is 001+country code+ area code+ telephone number, like 001+ 86+592+5500000. While the VoIP provider (Skype) requires the way as 00 + country code+ area code+ telephone number, like 00+86+592+5500000. Follow up the above way to keep the dialing habit as usual.

- Speed Calling, like No. 4. user dial 1#, Skype will call out 00865921234567

Usage of SiSky

(1) Outgoing Call to Skype ID

a. You are using another SIP extension (IP Phone or ATA), want to dial 'yeastar.pipigo2' through Skype trunk 1:
Offhook -> Dial 50123 or Offhook -> Dial 501 -> Dial 23#

b. You are using another SIP extension (IP Phone or ATA), want to dial 'yeastar.pipigo2' through idle Skype trunk:
Offhook -> Dial 50023 or Offhook -> Dial 500 -> Dial 23#

Note: If you using softphone (X-Lite, SJ-Phone, 3CX phone, and so on), want to dial 'yeastar.pipigo2' through Skype trunk 1:
Offhook->Dial 501yeastar.pipigo2 or Offhook->Dial 50123 or Offhook->Dial 501->Dial 23#

(2) Outgoing Call to SkypeOut number

a. You are using another SIP extension (IP Phone or ATA), want to dial '+865925503309' through Skype trunk 1:
Offhook -> Dial 50100865925503309 or Offhook -> Dial 501 -> Dial 00865925503309#

b. You are using another SIP extension (IP Phone or ATA), want to dial '+865925503309' through Skype trunk 1:
Offhook -> Dial 50100865925503309 or Offhook -> Dial 501 -> Dial 00865925503309#

(3) Incoming Call

You want to dial extension '666' from Skype trunk 1:
Call Skype ID 'yeastar.801' -> Dial 666.

- **WebCall**

You can create a Skype ID as the WebCall usage, and release this ID on the website as 'SkypeMe', and set up this Skype ID as one of SiSky trunk. Then your website visitor can click 'SkypeMe' button to make call to your company (IPPBX).
If you want to receive multi-incoming calls from this Skype ID, please setup 'Busy Transfer'.

(FINISH)