SP4 *SIP based audio system*

User Manual

Directory

1. Brief Introduction

The SIP based audio system SP4 utilizes the built-in intercom and paging capability already inherent in most modern IP PBX systems and enhances this to improve end user experience by providing a dedicated high performance digital amplifier on which to broadcast announcements or play background music.

They are ideal solutions to deploy in all types of environments including schools, offices, airports, and other public spaces. For additional functionality, an IP video camera can be connected to SP4 Ethernet interface and the resulting high definition images are transmitted to monitors located in assigned rooms.

Aside from announcements, remote access to the web interface provides remote control and configuration of your SP4 if the unit is located in other rooms or floors. Additionally there are 4 freely programmable pins that can be used for controlling the peripheral devices around SP4.

2. Delivery Contents

- Base Unit
- Pre-assembled Brackets for Wall-mounting
- Pre-assembled phoenix connectors
- Power Supply

3. Safety Precautions

Note: To avoid any device damage and bodily injury caused by improper use, please observe the following rules.

- Please use only the power adapter provided by LucidPhone. Other power supplies may damage or even destroy the device, and as such will not be covered by the product warranty.
- Ensure that the supply voltage matches the specifications indicated on the rear panel of SP4.
- The device is for INDOOR use ONLY! and NOT FOR OUTDOOR USE!
- To avoid an electric accident, DO NOT open or remove the cover of SP4 when it is powered on.
- DO NOT install the device in rooms with high humidity (such as bathroom, laundry room, damp basement, kitchen). Keep the device far away from the water or other liquid!
- Before cleaning the device, shut off the power supply. DO NOT clean it with a wet cloth and NEVER use any other liquid cleaning method. The most suitable method is to use an anti-static cloth.
- DO NOT install the device in surroundings at risk of explosions (such as gas station, paint store, etc.). Also DO NOT use the device when you smell any fumes with the potential to explode.
- DO NOT use the device in thunderstorms in case of any electric shocks by lightning.
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4. Device Installation

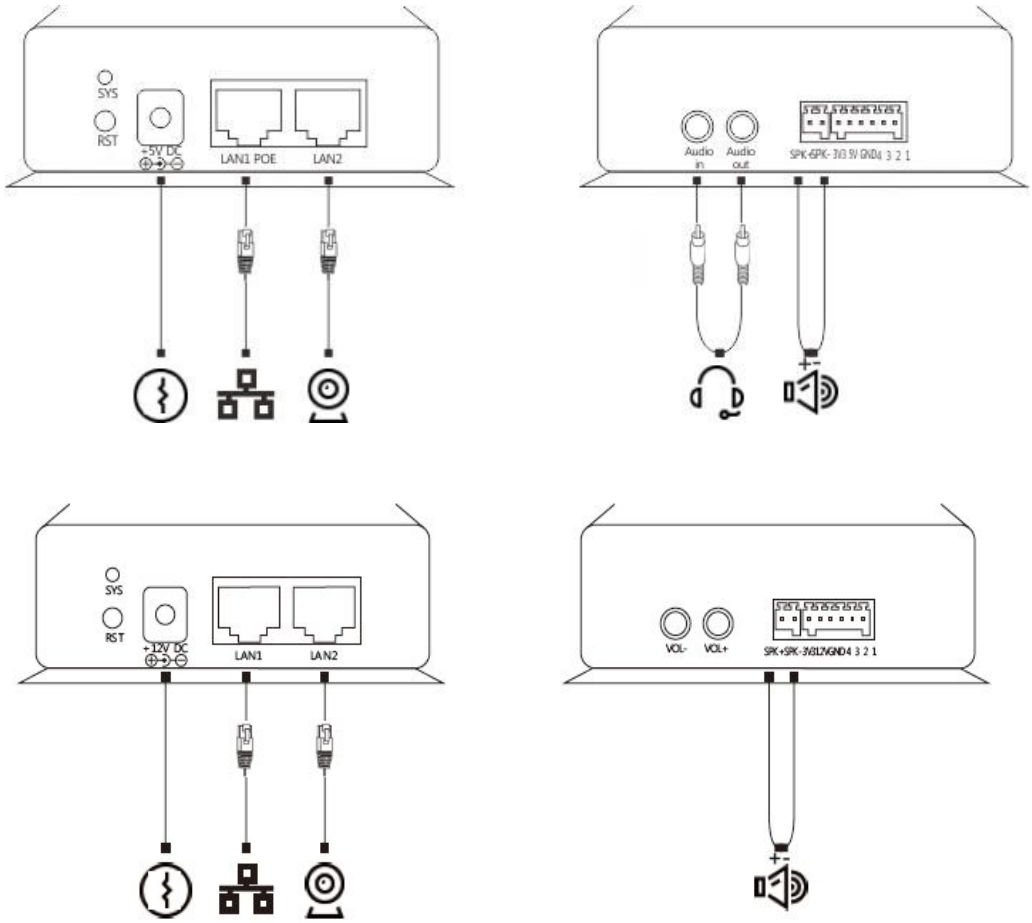
4.1 Device Information

Item		SP4
Telephony Protocol		SIP 2.0(RFC-3261)
CPU		MIPS+DSP
Keypad Expansion		Support 4×6, total of 24 keys
LCD Expansion		Support SPI mode (Custom)
Voice	Microphone	1 onboard connector
	Speaker Output	3.5W
	Handset	3.5mm standard input/ output interface
	Speaker Impedance	4Ω
	DSP	Support echo cancellation and noise suppression
Voice Flood	Supported protocol	RTP
	Codec	G.711,G.722,G.723,G.726,G.729, etc..
Other Interface	Input Power	5V/1A DC
	PoE	PoE 802.3af (Class 0 - 12.95W)
	LAN	2x10/100BASE-T Auto-MDIX, RJ45
	RS232	Optional
	GPO	4 programmable output pins
Environment	Operation Temperature	0℃ to 40℃
	Storage Temperature	-40℃ to 70℃
	Humidity	10% to 95%,No Dew

4.2 Physical Connection

- SP4 supports PoE. If the switch server supports PoE, please plug the Ethernet cable to the RJ45 interface “LAN 1 POE” of SP4 . The two Ethernet interfaces of SP4 support bridge mode only; LAN2 is used to connect other devices for network expanding. If your IP PBX server doesn’t support PoE, please use the 5V/1A power supply provided by LucidPhone to power for SP4.

- The interface of “SPK+” and “SPK-” on SP4 are used to connect the loudspeaker. If you are using headset, please plug the microphone of headset to “Audio in” of SP4 and plug the speaker of headset to “Audio out”.



iSpeaker C20 Connection Diagram:

5. Configuration

5.1 Web Login

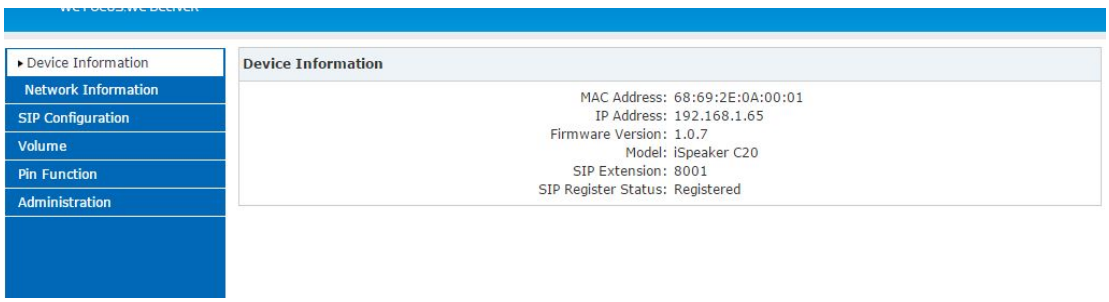
Step1. Plug the Ethernet cable to LAN 1 or LAN 2 of SP4.
Notice: To avoid network loops, the LAN 1 and LAN 2 cannot be connected to the same switch.

Step2. Open web browser and input http://192.168.119.110/, which is default IP address of SP4.
Notice: Please use the IE browser(Ver8.0 or higher), Firefox or Chrome.

Step3. Input default username (admin) and password(admin).

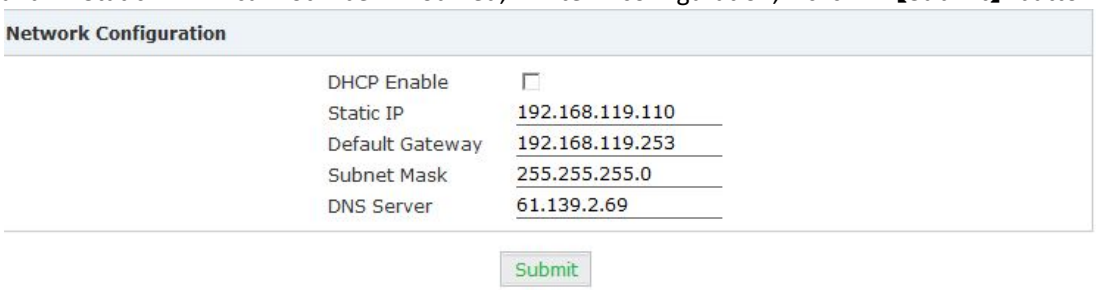
Default URL Address: http://192.168.119.110/
Default Username：admin
Default Password：admin

Step4. Go to the home page after successful authentication.
(If you visit http://192.168.119.110 again without closing the active browser, authentication is not needed because the browser already recorded login information.)

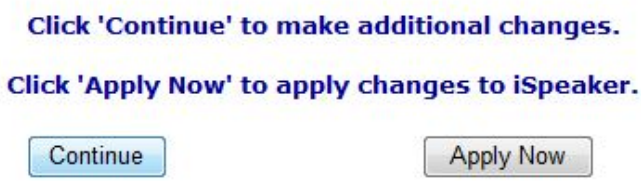


5.2 Network Configuration

Step1. Click【Network Configuration】to show the following display. Set static IP address for SP4 or dynamic IP address by enabling DHCP. When DHCP enabled, it will get the dynamic IP address and "Static IP" cannot be modified; After configuration, click 【Submit】button.



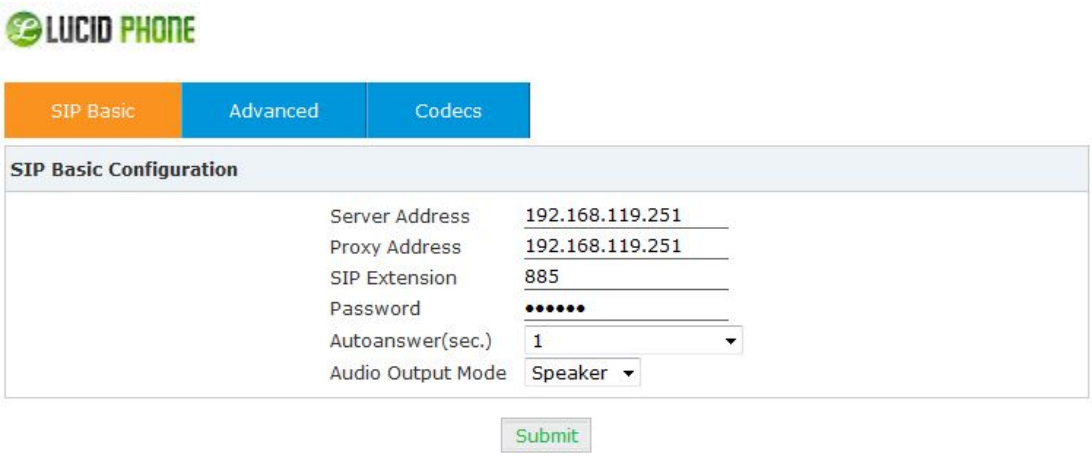
Step2. After configuration, click 【Submit】 to display the following window; you have to choose 【Continue】to make additional changes, or【Apply Now】to apply changes to device.



5. SIP Registration

6.1 SIP Basic Configuration

Click【SIP Configuration】to go to【SIP Basic Configuration】, including Server Address, SIP Extension, Autoanswer, Audio Output Mode(Defaulted Speaker), etc.. After configuration, please click "Submit".

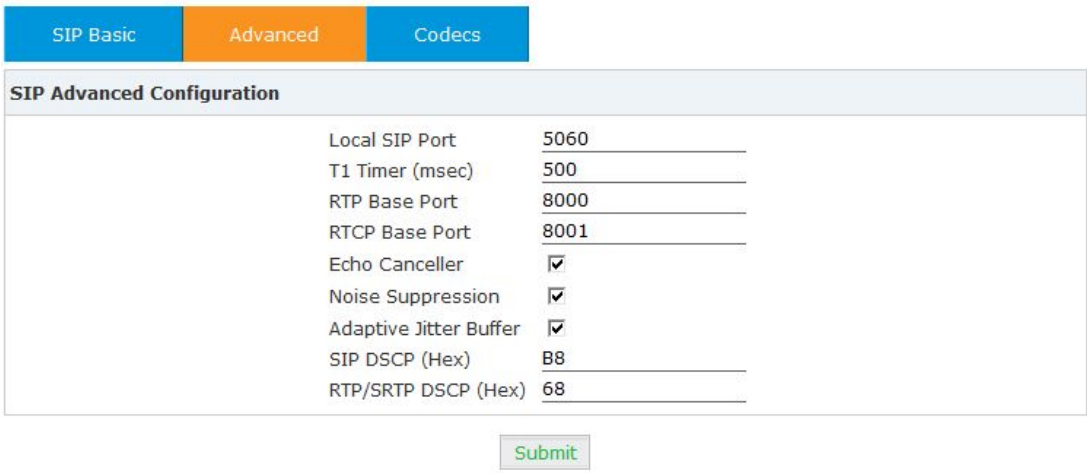


SIP Basic Configuration

Item	Explanation
Server Address	IP Address of IP PBX
Proxy Address	Proxy address of SIP proxy. Normally the proxy server and IP PBX is the same one, so the IP address is same.
SIP Extension	Registered SIP extension number provided by IP PBX
Password	Password of registered extension provided by IP PBX
Autoanswer	The time of Auto answer (by second). Default is 1; if set as 0, there is no auto answer.
Audio Output Mode	Output mode of audio (Speaker or Handset). Default is Speaker.

6.2 SIP Advanced Configuration

Click【Advanced】to go to SIP Advanced Configuration. You can change the configuration based on your requirement, such as Local SIP Port(Default is 5060) and RTP/RTCP DSCP (Default is 8000/8001). After modification, please click "Submit".



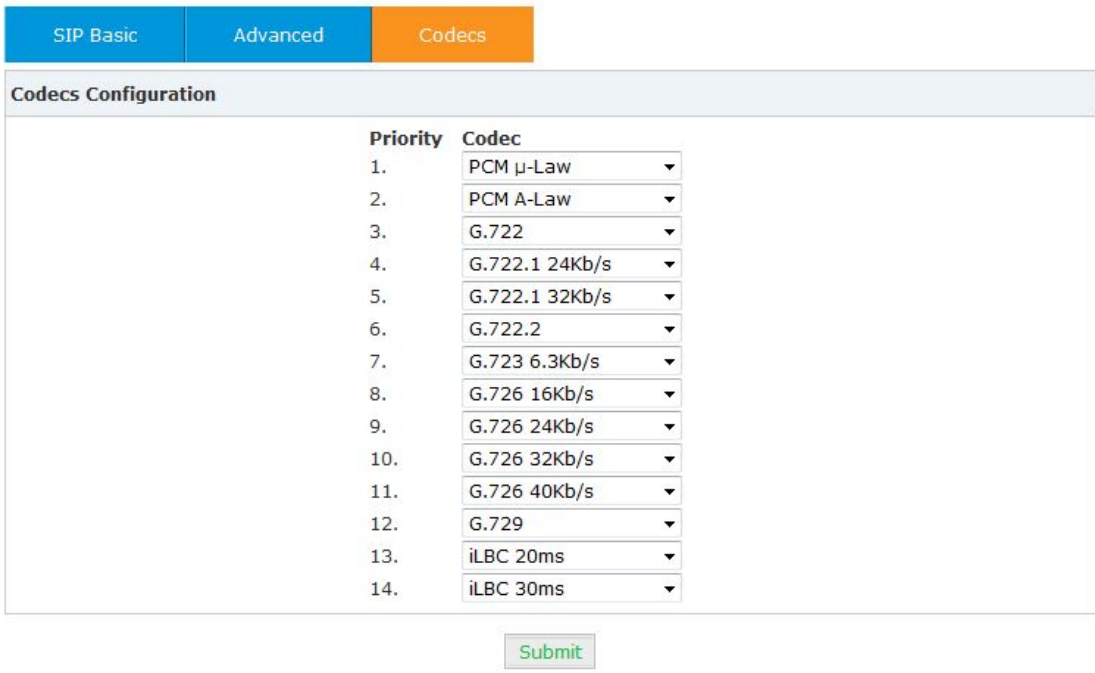
SIP Advanced Configuration

Item	Explanation
Local SIP Port	Local SIP Port. Default is 5060.
T1 Timer	Timer for sending SIP message (by second). Default is 500 ms.
RTP Base Port	RTP base port of voice data. Default is 8000.

RTCP Base Port	RTCP base port for voice data. Default is 8001
Echo Canceller	Set echo cancellation. Default is enabled.
Noise Suppression	Set noise suppression. Default is enabled.
Adaptive Jitter Buffer	Set adaptive jitter buffer when receive/send SIP voice. Default is enabled.
SIP DSCP（Hex）	Set the hex value of SIP DSCP. Default is B8. DSCP（Differentiated Services Code Point）
RTP DSCP(Hex)	Set hex value of RTP DSC. Default is 68.

6.3 SIP Voice Codec

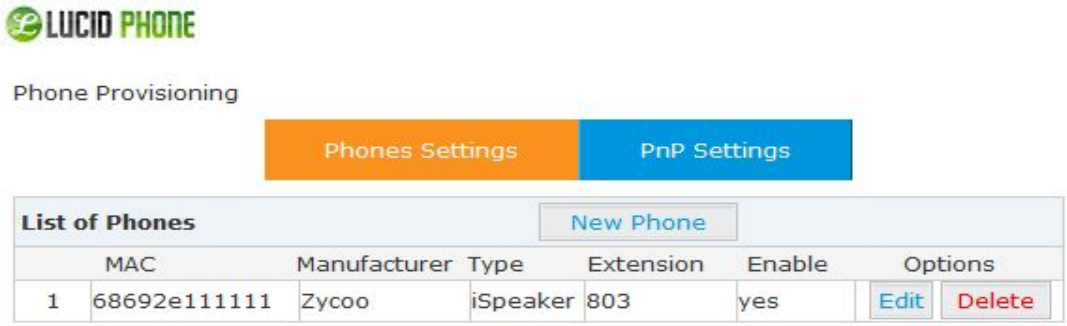
Click 【Codecs】 to configure the priority order of SIP codecs. After configuration, click 【Submit】.



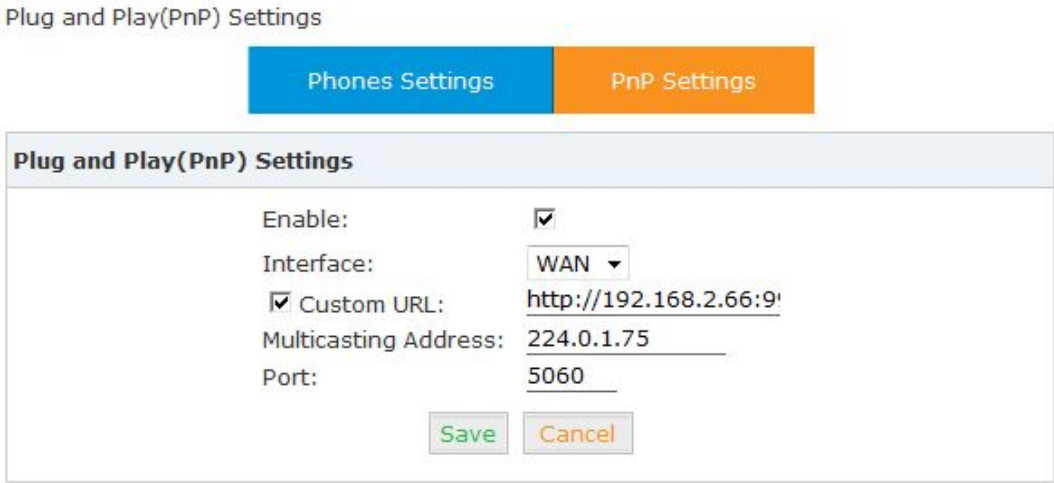
6.4 SIP Auto Provision (Hidden by default)

SIP Auto Provision is very useful function for company to deploy the telephony terminals such as IP Phone and sip speaker quickly.

Step1. Open the IP PBX GUI, click【Phone Provisioning】to make settings of phone provisioning.
Notice: please make sure that your IP PBX support auto provision of SP4; if not, please download the patch and update your PBX.

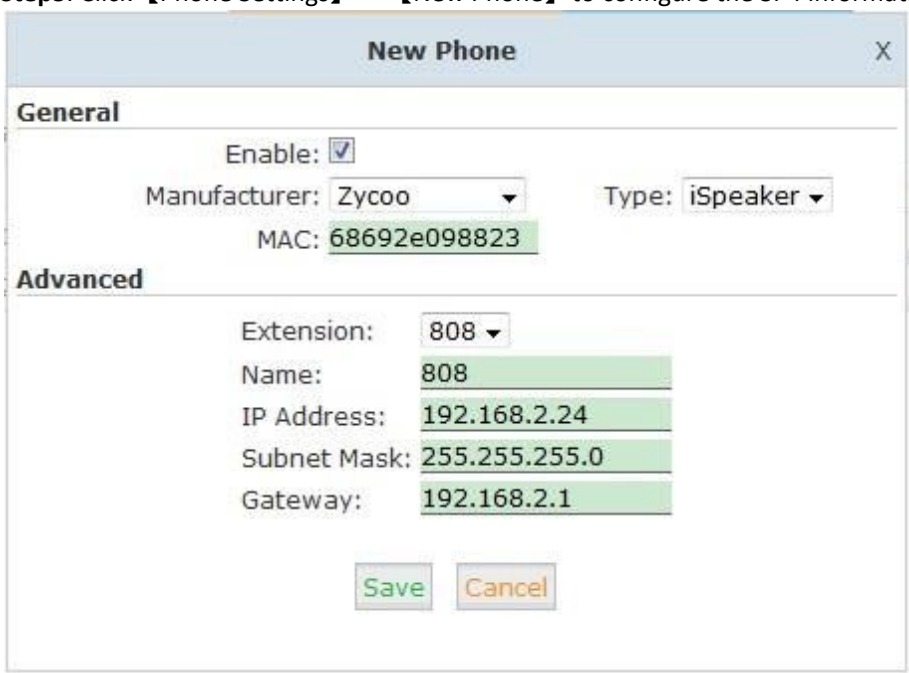


Step2. Click【PnP Settings】to configure, input URL, use the default multicasting address: 224.0.1.75 and default port 5060.



If custom URL is required, the IP PBX server IP address will be displayed here automatically once tick this option (E.g. : http://192.168.2.66:9999/phones).

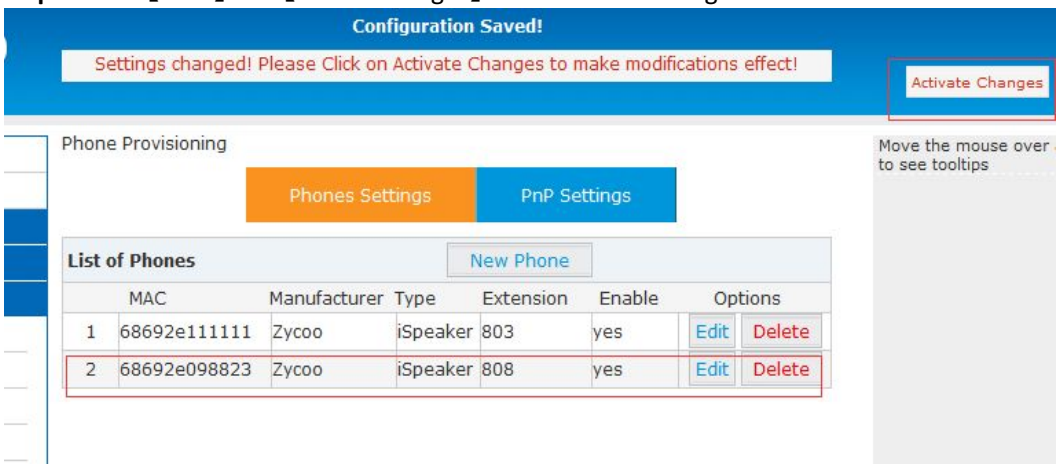
Step3. Click【Phone Settings】→【New Phone】to configure the SP4 information.



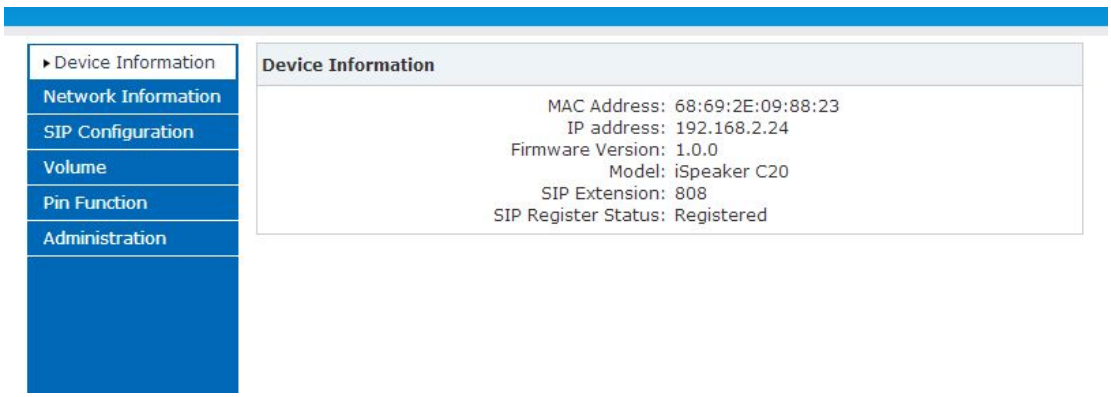
Item	Explanation

Enable	Enable to make the phone provisioning effective
Manufacturer	Select the factory of SP4:
Type	Select device type: SP4
MAC	MAC of SP4 device
Extension	Select extension number of SP4
Speaker Name	Define the name for this SP4.
Speaker IP	Define the fixed IP of SP4
Subnet Mask	Define the subnet mask of SP4
Gateway	Define the default gateway of SP4

Step4. Click【Save】→【Active Changes】to make the settings effective.



Step5. Reboot SP4, and visit the device with the previous defined IP in Phone Provisioning, e.g.: http://192.168.2.24, input default username and password "admin/admin", then you will see the device information which shows the SP4 is registered.



6. Volume Settings

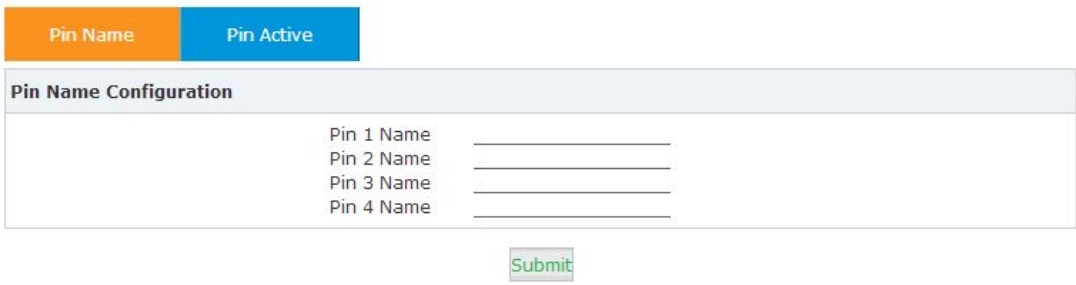
Output volume is for speaker and "Audio out"; Input Volume is for "Audio in".
Click【Volume】to set the output volume and input volume.

After configuration, please click【Submit】.

7. PIN Settings

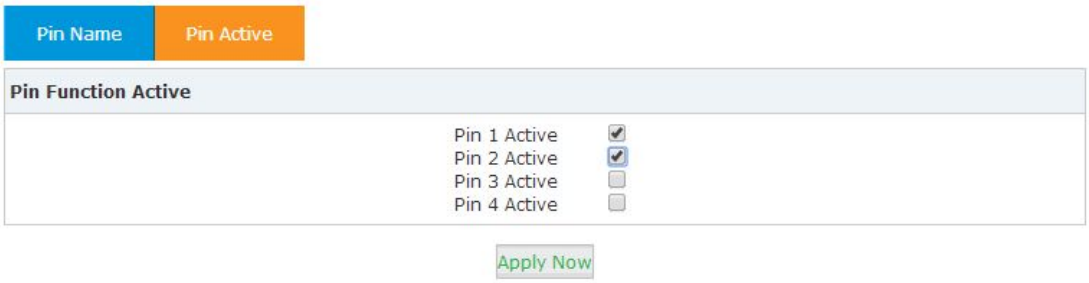
8.1 Pin Configuration

Pin is used to connect some related terminals, such as alarm, door-lock. Pin Function is used to define the function of pins.
Click【Pin Function】to go to Pin Name Configuration and define the name from Pin 1 to Pin 4.
Notice: Space is not allowed in Pin Name; please use "_" to replace space.
After configuration, please click【Submit】.



8.2 Pin Active

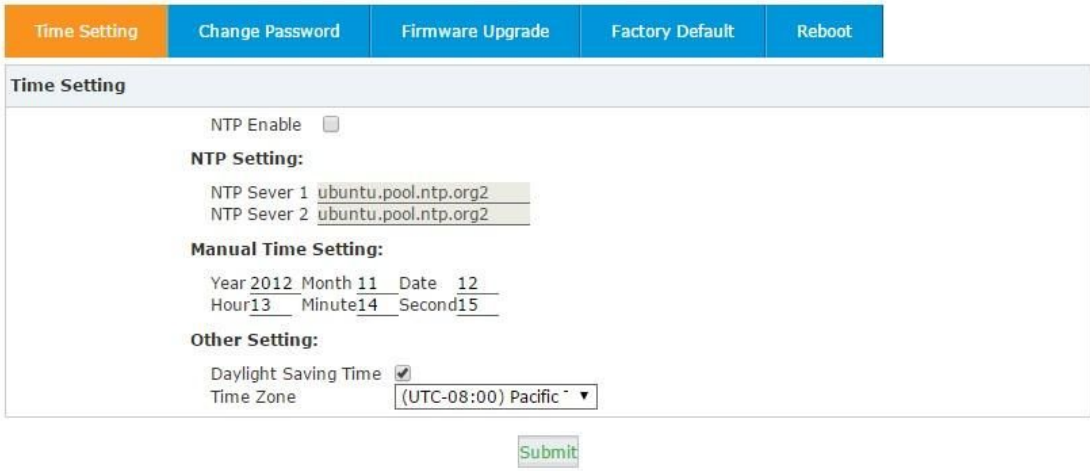
Once enabled "Pin Active", the corresponding Pin will output high voltage; otherwise, the corresponding Pin will output low voltage.
Click【Pin Active】to apply the function of Pin1 to Pin 4.



8. System Management

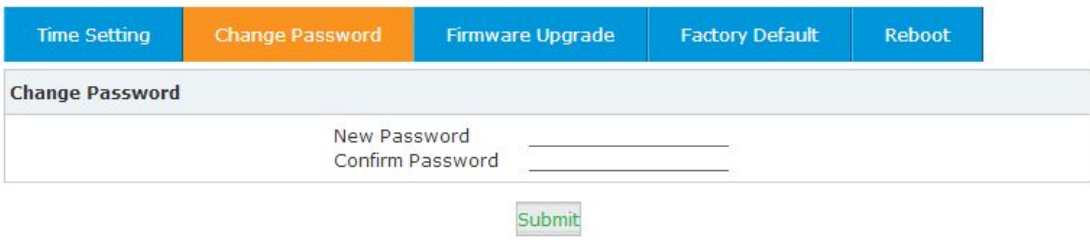
9.1 Time Setting

Click【Administration】→【Time Setting】to configure the time. If NTP is enabled, the system will configure the system time by NTP mode (NTP is enabled by default). If NTP is not enabled, time should be set manually in “Manual Time Setting”.
“Daylight Saving Time” is available to choose (Not enabled by default). Also you can choose the time zone, default is “(UTC+08:00)Beijing”.



9.2 Change Login Password of Web

Click【Administration】→【Change Password】, input new password and confirm, then submit to make it effective.

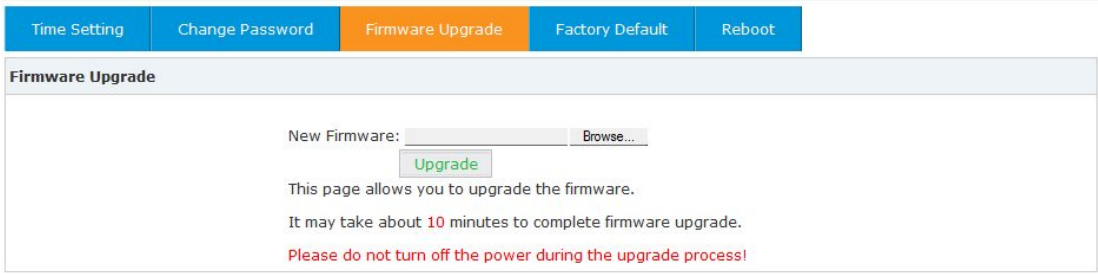


9.3 Firmware Upgrade

Click【Administration】→【Firmware Upgrade】to upgrade the firmware. Download the firmware from LucidPhone official website; to make sure successful upgrade please modify the

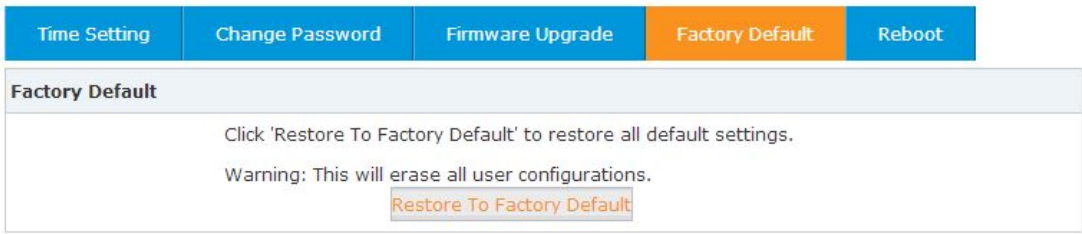
firmware name as“SP4_”.

Notice: Please DO NOT power off or reboot the device in upgrade; the system will reboot automatically after firmware upgrade succeeded.



9.4 Restore to Factory Default

Click 【Administration】→【Factory Default】, and click the button “Restore to Factory Default”. The system will be reset after 3~5 seconds, all the settings will be cleared and reset to factory default.



9.5 Reboot

Click【Administration】→【Reboot】and click the button “Reboot”, the system will reboot after 3~5 seconds.