Manual of NPX SIP Server
From NICEUC Company

V1.0

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Index

Chapter 1 Introduction

NPX SIP Server from NICEUC Company is a depth of customization Asterisk, which runs on the Linux platform.

Chapter 2 Preparation

Recommended Hardware：X86_64 PC

CPU：More than 1GHz single-core or multi-core

Memory：More than 1GB

Disk Space：More than 32GB

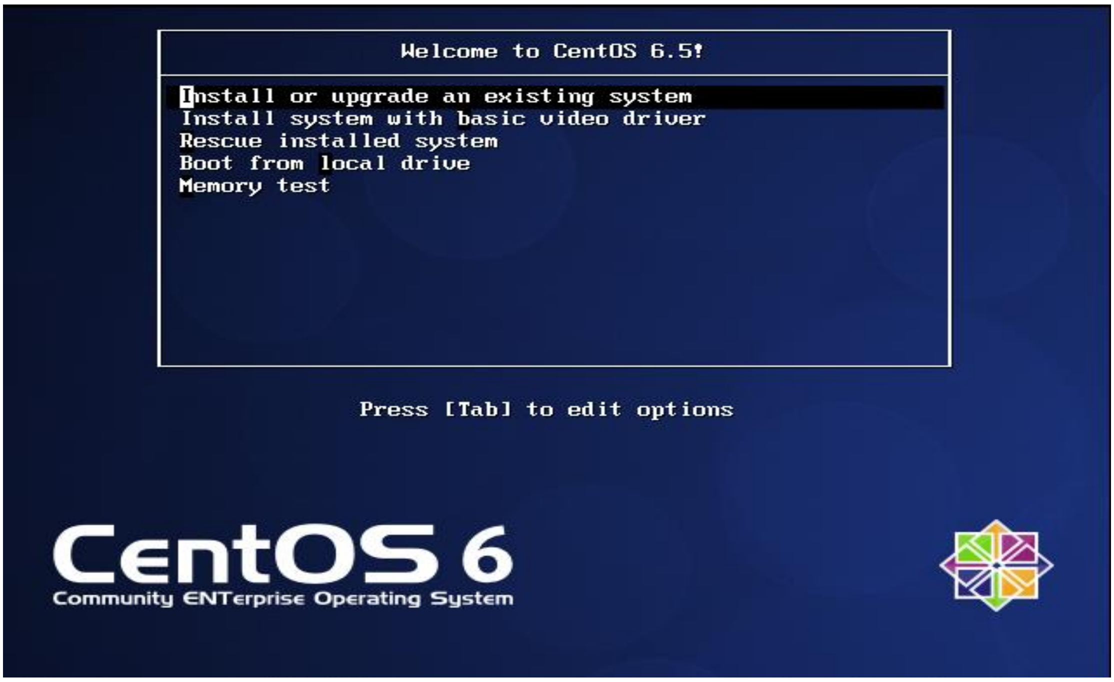
OS platform：CentOS 6.5 64Bit (2.6.32-431.el6.x86_64)

Normally we would offer Linux system installation disc or ISO image file; you can also install Linux system by yourself.

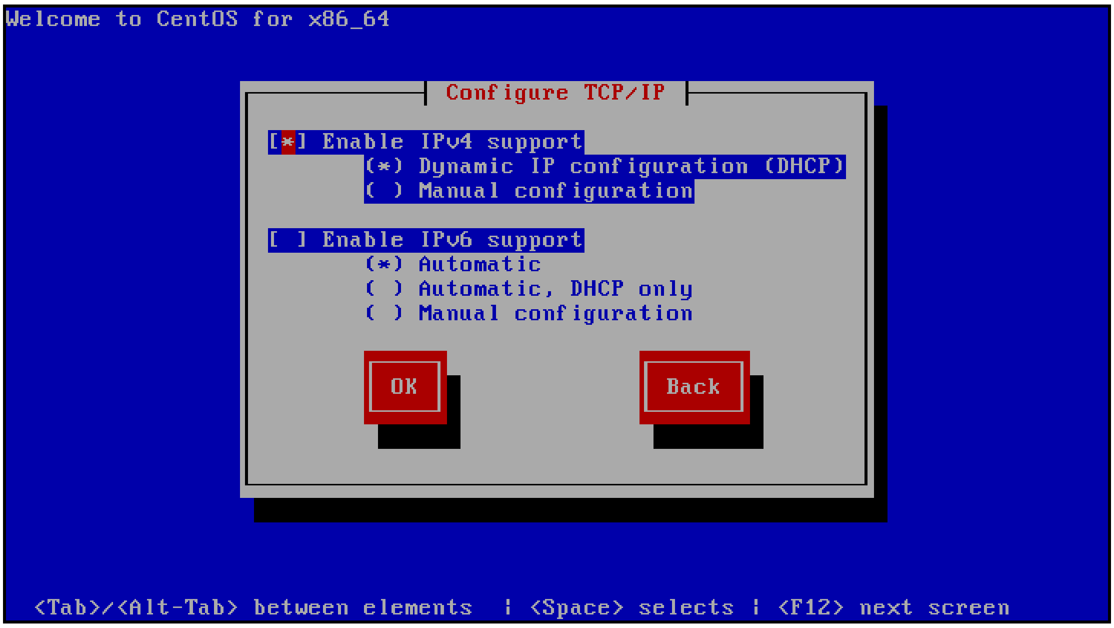
Chapter 3 Installation

3.1 Install Linux System

If there is no installation CD, please use the ISO file that we offer, mirroring burn a system installation CD. Put the prepared system installation disk into the CD driver, power on the computer, choose to boot from CD-ROM, the following screen appears after booting:



Select the “Install or upgrade an existing system “, press the Enter Key, it will be beging to do the install. A few minutes later, show the network setting:

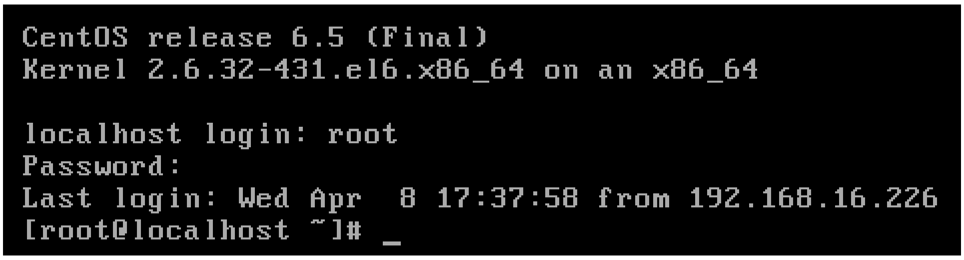


If you don’t want to set is now, you can press the F12 to next.

If you need set it now, use the Tab Key to move the selection, and Space Key to selects.

After this step, it will do the fdisk and then copy files.

After installation, you can use the “root” user to login, and the password is “niceuc”.



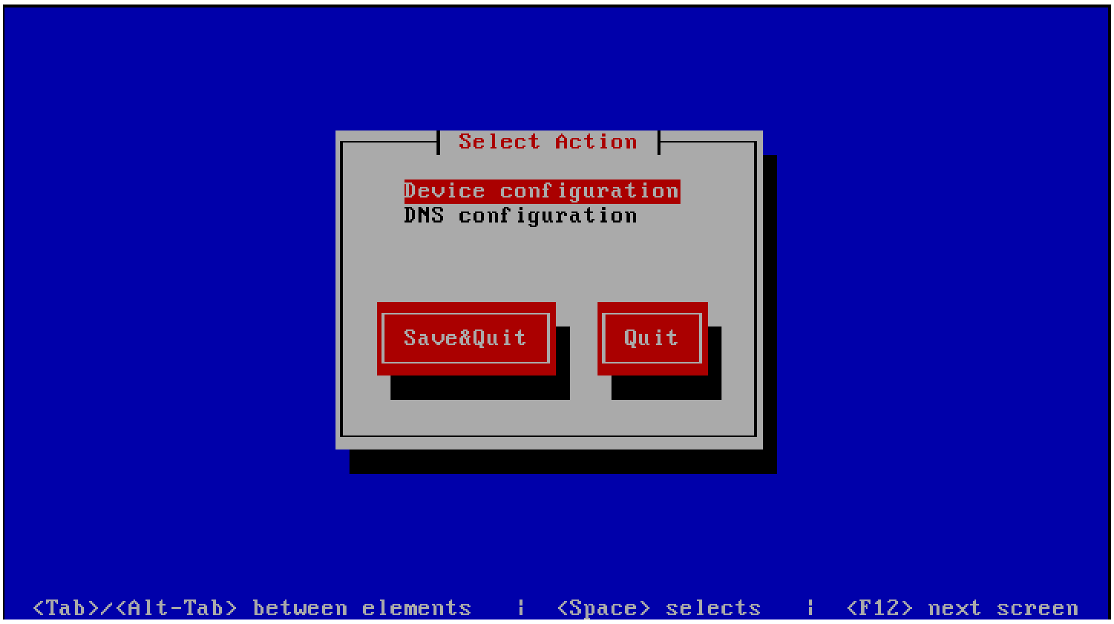
It show the “#” prompt, the system is installed successfully.

3.2Modify IP in Linux

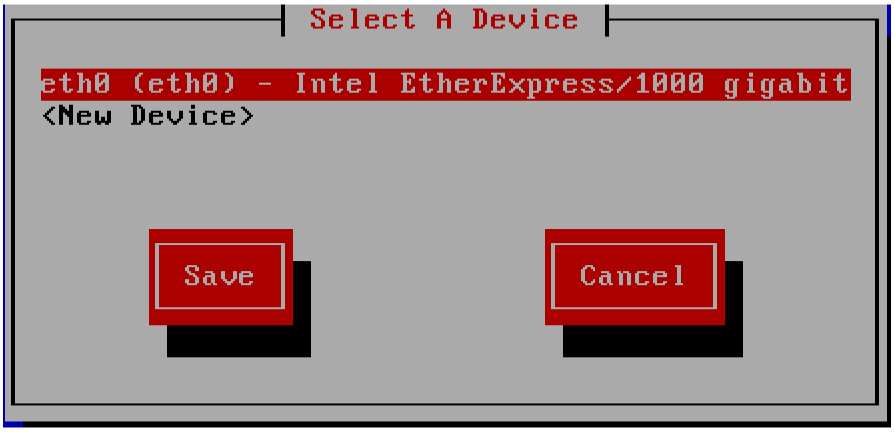
You can use the “system-config-network” command to modify the IP easily.

For different computer hardware, the network card name may be different, but the process is the same.

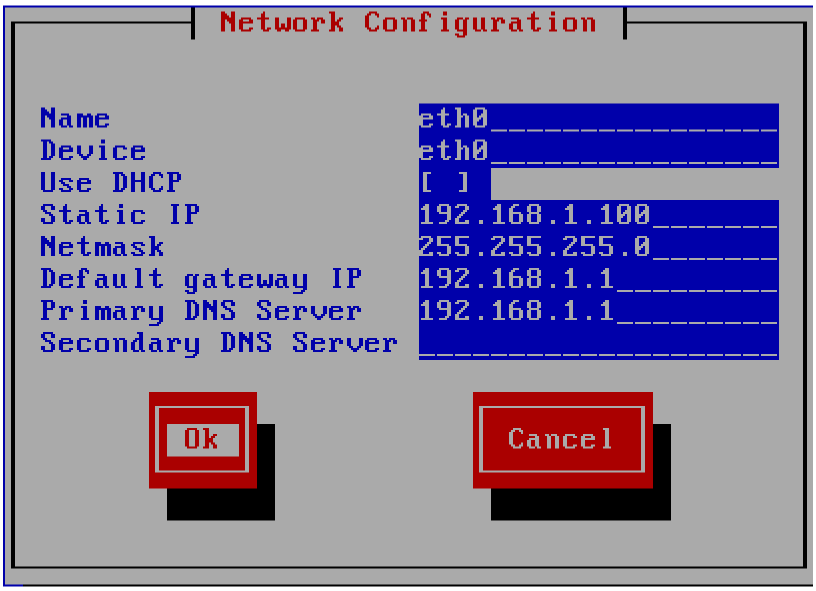
Input this command, it will show a simple UI.



First select the "Device configuration", then press the Enter Key



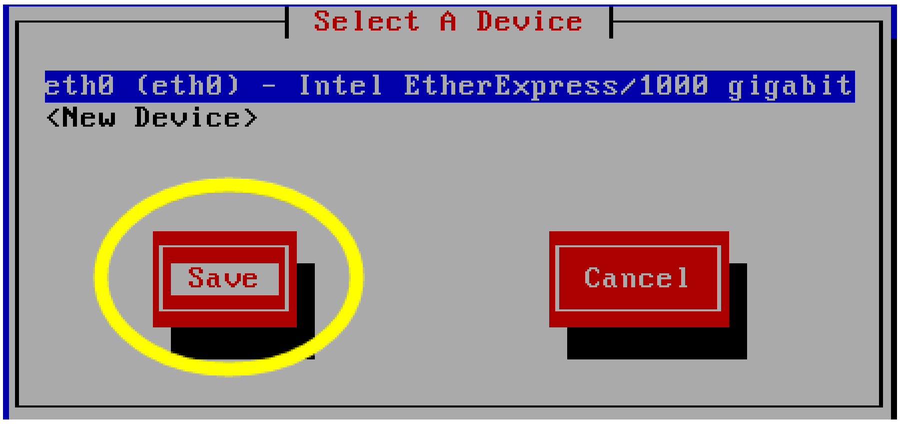
Then select the network card name, usually it is named "eth0"



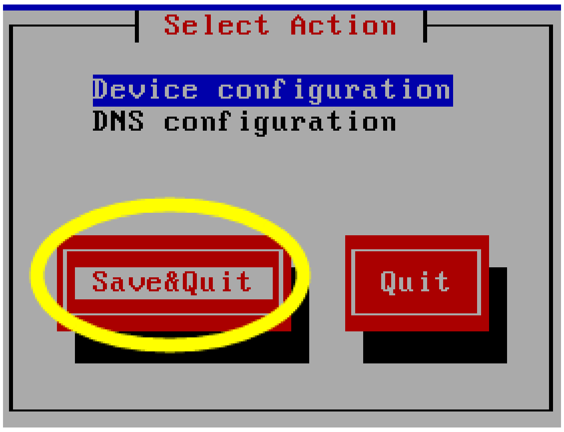
Press the Tab Key (or Up / Down / Left / Right Key) to move the cursor.

If want to use the Static IP. First need unchecked the “Use DHCP”, then fills the IP and netmask and so on.

Finally, move to “Ok” button, press Enter key to complete the setting, it will return to the previous screen

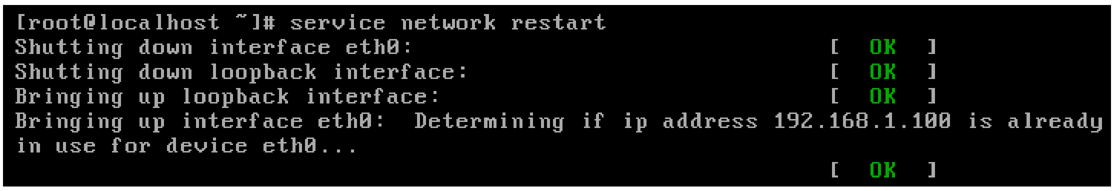


Move to “Save” button, press Enter Key to save the configuration, it will return to the previous screen



Move to “Save&Quit” button, press Enter Key to Save and Quit, complete the setting.

Usually need to reboot the system to take effect. But we can use the command to take effect instead of reboot; the command is “service network restart”.



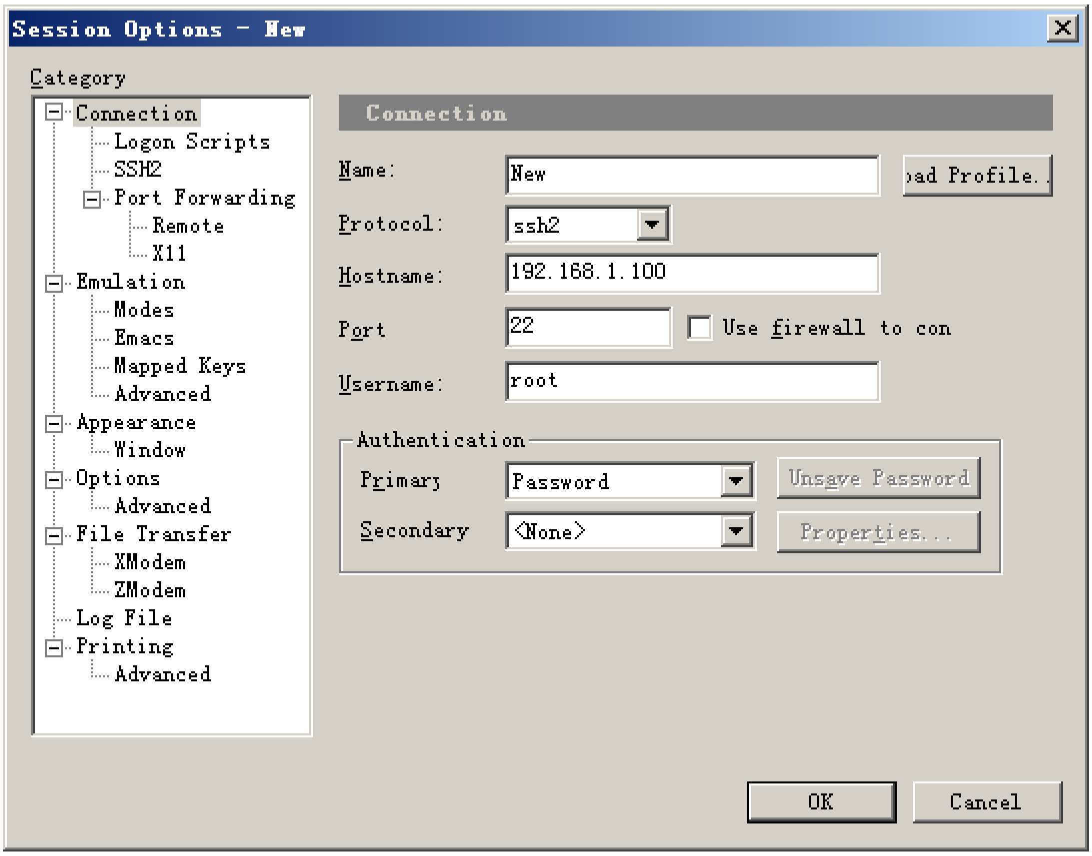
Finally, we use the “ifconfig” to check the result.

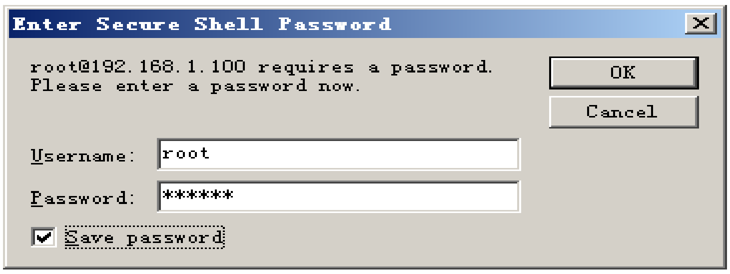
```
[root@localhost ~]# ifconfig
eth0      Link encap:Ethernet  HWaddr 00:0C:29:F9:9E:17
          inet addr:192.168.1.100  Bcast:192.168.1.255  Mask:255.255.255.0
          inet6 addr: fe80::20c:29ff:fef9:9e17/64 Scope:Link
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:2874 errors:0 dropped:0 overruns:0 frame:0
          TX packets:39 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:1000
          RX bytes:213094 (208.0 KiB)  TX bytes:4039 (3.9 KiB)
```

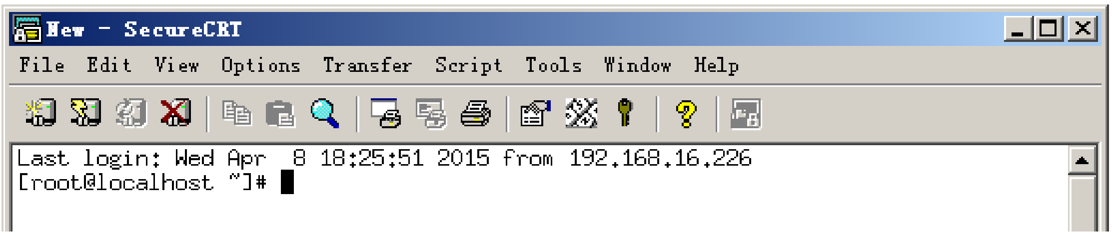
3.3Connect the SSH

SSH Server uses the default port: 22.

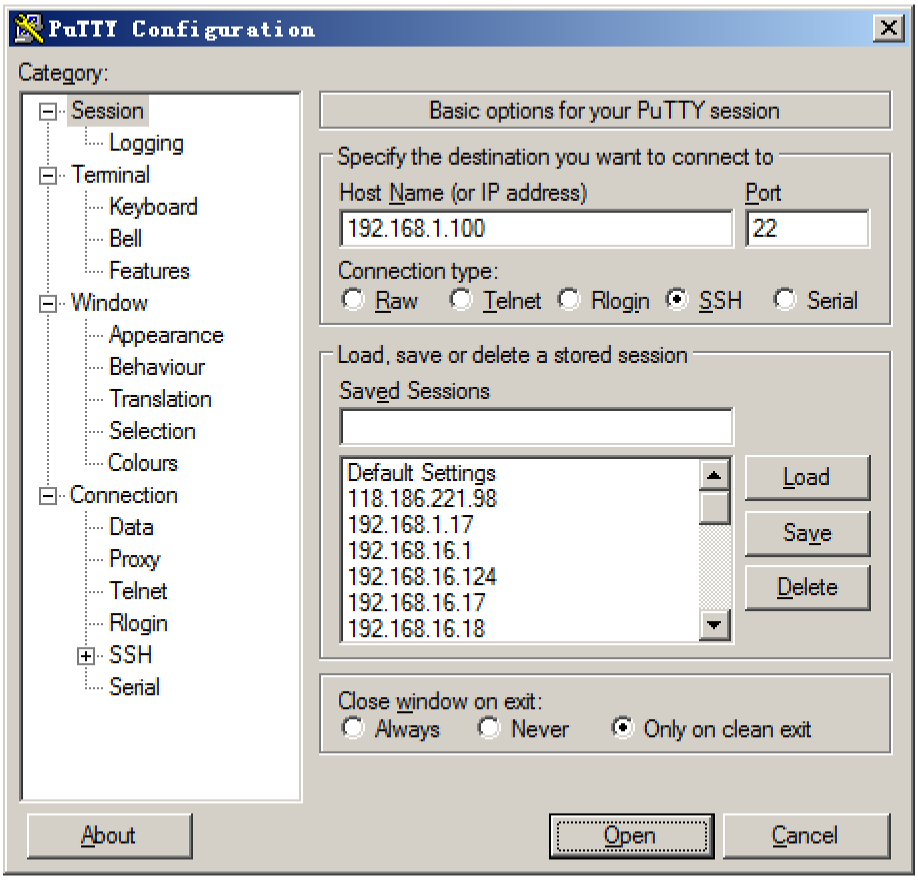
We recommend use the SecureCRT software, SecureCRT is third-party terminal software, which supports multiple protocols; it is very good but not free. Below SecureCRT 4.0 as an example, create a new connection, the new session, choose SSH2 protocol

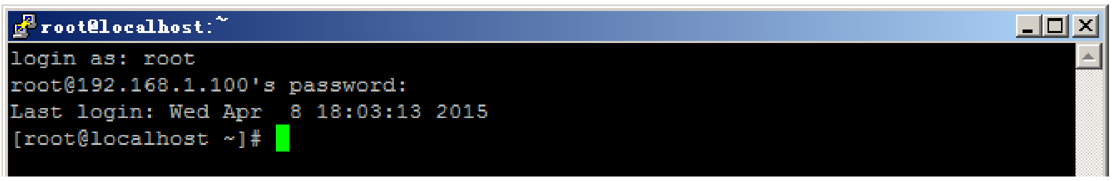






Another useful software is Putty, it is a small and simple, open source free software, It is very popular.





After connected the SSH, you can input the command like at the local machine.

3.4 Install NPX

Usually the file is named like npxinstall-20150310.tar.gz.

If use the SecureCRT, you can use the "rz" command to Trans the file on to the Linux system by Zmodem.

Or use U disk copy, or use FTP to upload.

After complete the copy of install file, do the next step:

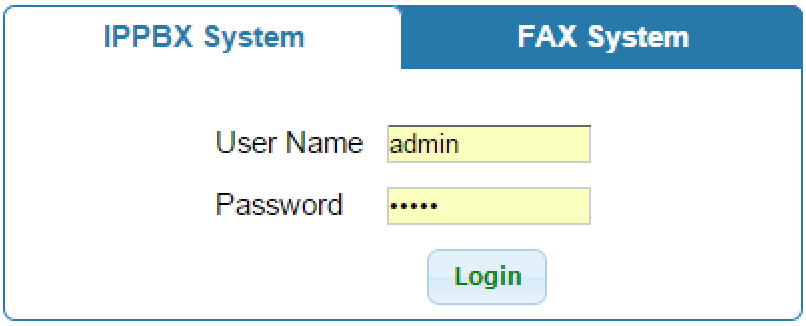
1. tar zxvf npxinstall-20150310.tar.gz
2. cd npxinstall-20150310
3. ./install.sh

Finally, it will reboot.

Chapter 4WEB System Management

4.1System License

Open the WEB brower, input the IP of Linux System at the address bar, Enter Key to connect.



User Name is "admin", Password also is "admin".

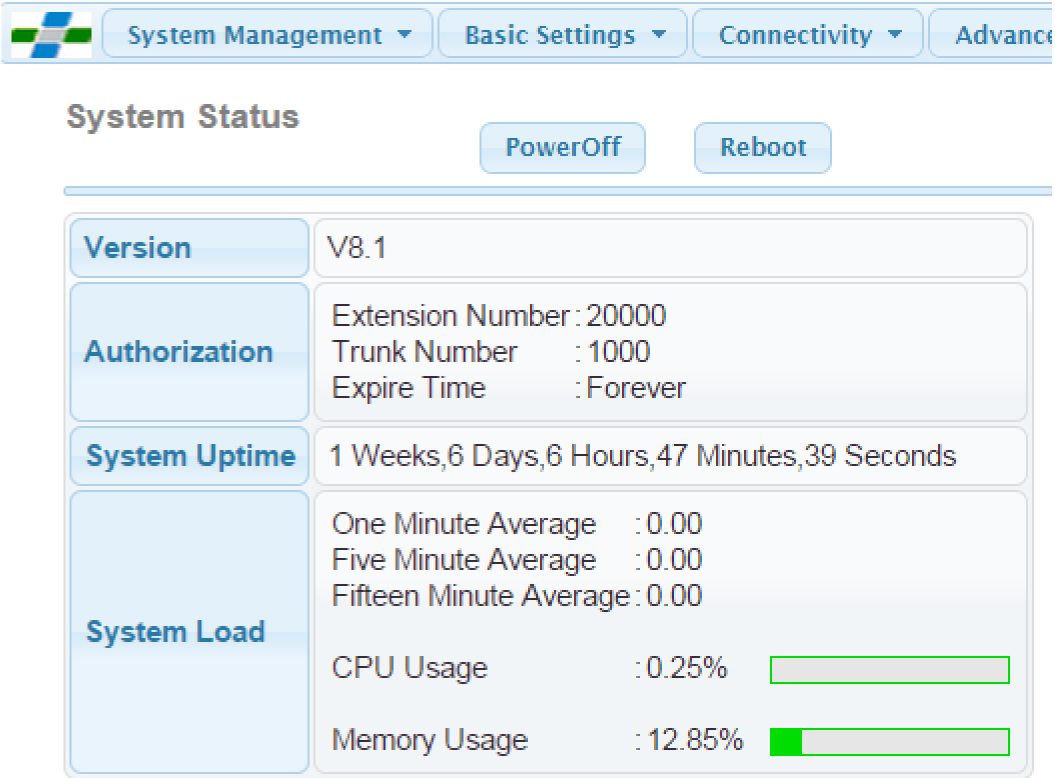
After login, click the menu "System Management ->System License" to the System License Page, Click the "Generate License ID" button, it will show the "Download License ID" link.

Click the "Download License ID" to download the license id file.

Please send the License ID file (hardinfo.bin) to us, we will sent back to an authorization file (niceuelz.bin) to you. Select the niceuelz.bin, click "Upload" button to upload it, after a successful upload, it will show a "Reboot" button, Click it, and reboot the system to make it effective.

4.2 System Status

Select the menu"System Management->System Admin" to show the system status

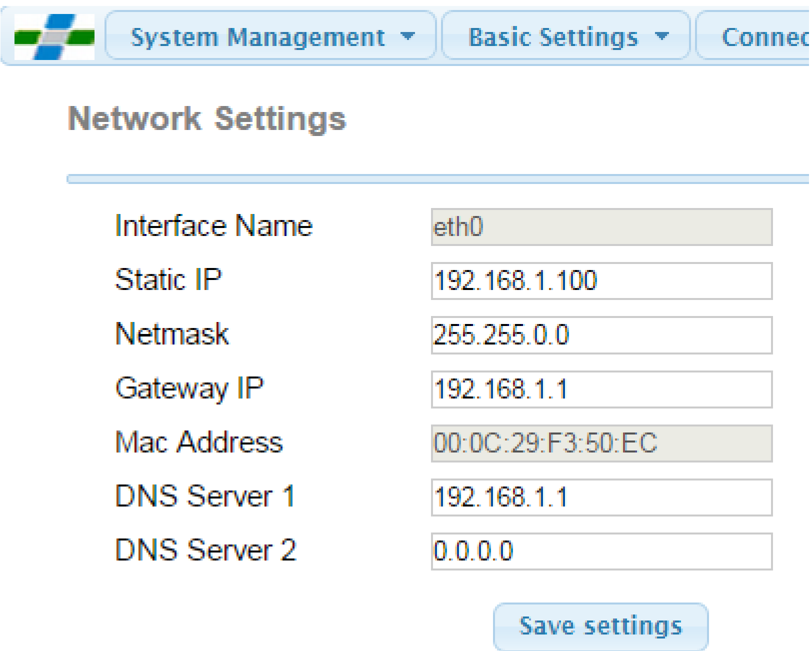


Authorization show the license content。 System Load show the CPU Usage, etc.

You can click the "PowerOff" button to do power off, or Click "Reboot" button to do reboot.

4.3 Network Settings

Modify the IP in Web is very easy. Select the menu "System Management->Network Settings" to set it.



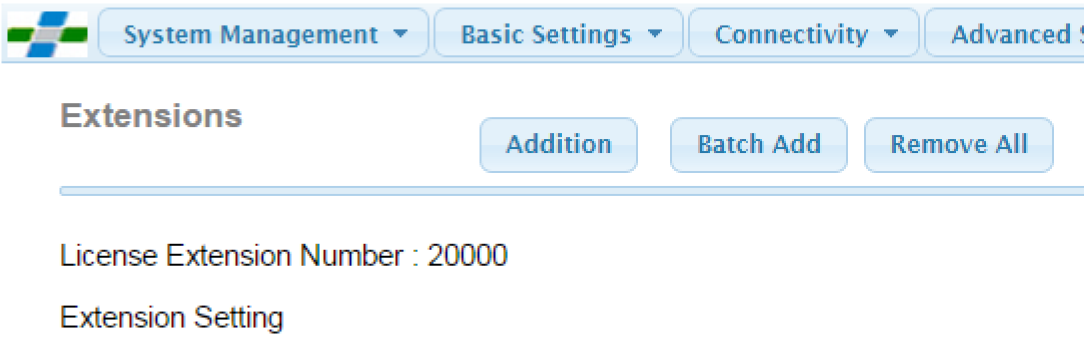
Click the “Save settings” button to save the IP, and need reboot or use command “service network restart” to take effect.

Chapter 5 Work config

Behind some of the parameters, there is an icon, Move the cursor on it, it will show the content for the detailed description of the parameter.

5.1Extensions

Select the menu “Basic Settings->Extensions” to set.



Please note the License of Extension Number; you can not add more than the license quantity.

Click the “Addition” button to add one extension, or click the “Batch Add” to add multi extensions at one time.

User Extension, the Extension phone number.You can call it from the IVR or other Extension. It is an identifier of extension. It only can use the digital.

Display Name, It will show to called side, it is the “DisplayName” field of SIP message. It can use digital or letter symbols.

RouteGroup, Ifneed call out to trunk, need config the “Outbound Routes”, Click the “Edit Route Rule” button; will pop the “Route Rule” setting dialog page. The “Route Group” is used by Extensions; one group can have many route rules. Only when the “Outbound Routes” has route rules, here can select a RouteGroup number.

Class, it is relate to the “Weights” of “Outbound Routes”. When the extension’s class greater than or equal to the “Outbound Routes”’s “Weights”, the extension can use this route rule, else can not used.So use it can restrict the call out.

Extension Options, Click on the icon after it, show or hide the more parameters.

Password, it is used for SIP register, if empty, itwill same to the extension number. In the “Batch Add”, we can use “|”as the extension number, for example, the extension number is 1234, fill in the “abc|def”, the passwordwill be submitted to “abc1234def”.

Pickup Group, it is the company group (Virtual User Group). The extensions in one same group can do the pickup for each other.

Outbound CID, When dial out to a trunk,first use this number as the CallerID, if this is empty, will use the trunk's "Outbound CallerID" as the CallerID for out call.If the trunk's "Outbound CallerID" also is empty, will use the extension number as the CallerID. In the "Batch Add", we can use "x" as the extension number, for example, the extension number is 12345, fill in the "288x", the Outbound CID will be submitted to "28812345".

DID direct, it is used for call in.When call in from trunk, will check this number, if can match it, will direct call to this extension. In the "Batch Add", we can use "x" as the extension number, for example, the extension number is 5000, fill in the "2000x", the Outbound CID will be submitted to "20005000".

DTMF Mode, set the SIP's DTMF Payload, can use "INBOUND", "SIPINFO" and "RFC2833", usually rfc2833 for most phones.

Media Direct, Re-Invite policy, if set "yes", the RTP will not pass through here, directlytransmitted by terminal each other.

Hide CallerID, if set "yes", there will not show caller id to peer side.

Show Incoming CID, if set "yes", there will not show caller ID to extension.

Permit, IP address filters, for example, 172.16.0.0/255.255.0.0

Nat, When the extension behind NAT device, need to set this "yes".

Host, usually use the "dynamic", mean it is necessary for the entity to register before PBX can call it. If set an IP, it can call peer to peer, not need register. If it is a FAX extension, should set to "127.0.0.1".

Port, when "Host" is setted to an IP, need set the peer's port. If it is a FAX extension, the port will fix to "20"+extension number, for example, the extension is "801", the port will be "20801".

Monitor, if set to "yes", will record the talk voice to a file.

Disable Dial Exten, if set "yes", there will can not dial other extension.

Ring duration max, the duration of Ring, default is 40 seconds.

Unavailable IVR, if set "yes", when the call is unavailable for extension, will into an IVR, then you can dial other extension or into Voice mail.

User Password, this is used for Phonelink.

Voicemail

Status set "Enabled" to enable the Voicemail.

Voice mail Password, the password for extension to enter into itself voice mail system.

Email Address, The email address that voicemails are sent to.

Email Attachment, if set “yes”, there will attach voice mails by email.

Play CID, Playback caller's telephone number.

Delete Voice mail, if set “yes” the message will be deleted from the voice mailbox (after emailed). Provides functionality that allows a user to receive their voice mail via email alonely, rather than having voice mail able to be retrieved from the Web interface or the Extension handset.**CAUTION**: it is necessary that there have attached voicemail by email when set “yes” otherwise your messages will be lost forever.

After the above information is complete and correct, click the “Add” button to save it. And then you can see them in the “Extension List” that is on the right side of the page.

Click on the Extension in the “Extension List”, it will show the detail of this extension’s parameters. You can change some parameters, then click the “Save” button to save it.

5.2 Trunks

Trunk is used for call out. Select menu”Connectivity” and”Trunks” to set.

Click the “Add SIP Trunk” to add it.

Trunk type, it has 3 types.

P2P, peer to peer, not need register.

Register, trunk need register to our here.

OutRegister, we need register to trunk.

Trunk Name, the name of SIP trunk, it is a globally unique, cann’t be repeated.

Outbound CallerID, When dial out to trunk,first use the extension’s “Outbound CID” as the CallerID, if it is empty, will use the trunk’s “Outbound CallerID” as the CallerID for out call. If the trunk’s “Outbound CallerID” also is empty, will use the extension number as the CallerID.

Maximum Channels, The maximum number of calls which can be received.

Convert RuleID, relate to the “Convert Rule”. Click the “Edit Convert Rule” button to modify the “Convert Rule”. Use the “Convert Rule”, you can change the prefix of number and add suffix to the number.

Host, it is used for type “P2P”. Set the peer side’s addr.

Port, set port of peer side.

Dtmfmode, set the SIP’s DTMF Payload, can use “INBOUND”, “SIPINFO” and ”RFC2833”, usually use rfc2833.

Secret, only used for “register” and “outregister” type. Set the password.

The following parameters are only used for “outregister” type.

Proxy, send outbound signaling to this proxy server, not directly to the peer side.

From Domain,when making outbound SIP INVITEs to non-peers, use your primary domain identity for “From:” headers instead of just your IP address.

From User,when making outbound SIP INVITEs to non-peers, Use this value insetad of callerid num in “From:” domain

Register String, Most VoIP providers require your system to REGISTER with theirs. Enter the registration line here. For example:

username:password@switch.voipprovider.com

Many providers will require you to provide a DID number, for example:

username:password@switch.voipprovider.com/didnumber

In order for any DID matching to work.

After the above information is complete and correct, click the “Submit” button to save it. And then you can see it in the “Trunk List” that is on the right side of the page.

Click on the Trunk in the “Trunk List”, it will show the detail of this trunk’s parameters. You can change some parameters, then click the “Submit” button to save it.

5.2.1Inbound Routes

It is used call in from trunk. Select the menu ”Connectivity ->Inbound Routes” to set it. Default, have a Route named “all”, all other call in will use it, you can not delete it.

Click the “Add Incoming Route” button to add a route.

Description, the name of the Inbound Route. The proposed naming:

DID number: 123456, you can names DID12345

CallerID number: 54321, you can names CID54321

DID Number, It is the judgment of the condition number of an incoming call, if the incoming called number and this number matches, then use this route.

CallerID Number, It is the judgment of the condition number of an incoming call, if the incoming caller number and this number matches, then use this route.

CAUTION: “**DID Number**” and”**CallerID Number**”, the two can only choose one to be used.

Set Destination, set the incoming call go to the destination.

Extension, go to an extension. (Set the extension in menu “Basic Settings -> Extensions”)

Queues, go to a queue. (Set the Queues in menu "Basic Settings -> Queues")

Timeconditions, first judge the time conditions, then go to the corresponding destination. (Set the Time Conditions in menu "Basic Settings -> Time Conditions and Time Groups")

IVR, go to an IVR. (Set the IVR in menu "Basic Settings -> IVR")

Trunks, go to a trunk. (Set the Trunks in menu "Connectivity -> Trunks")

MeetMe, in to a meeting. (Set the MeetMe in menu "Basic Settings -> MeetMe")

TerminateCall, hangup, disconnect the call.

After the above information is complete and correct, click the "Submit" button to save it. And then you can see it in the "Incoming Route List" that is on the right side of the page.

Click on the Route in the "Incoming Route List", it will show the detail of this route's parameters. You can change some parameters, then click the "Submit" button to save it.

The following is instructions for each routing destination.

[Queues]

Put the incoming calls to a wait queue, then the queue assignment to the idle extensions. Select the menu " Basic Settings ->Queues" to set it.

Click the "Add Queue" button to add it.

Queue Number, the phone number of the queue, extension can dial this number to test the queue.

Queue Name, the name of the queue, you can input some description in here.

Static Agents, the extension list to use for the queue assignment. You can click after the "Extension Quick Pick", to pick an extension in the expanded list.

Ring Strategy, allocation policy extension

ringall - ring all available channels until one answers

fewestcalls - ring the one with fewest completed calls from this queue

random - ring random interface

rrmemory - round robin with memory, remember where we left off last ring pass.(default)

linear - rings interfaces in the order specified in theStatic Agents list.

Music on Hold Class, the music class sets which music applies for this particular call queue.The only class which can override this one is if the "custom" class be seted.

Max Wait Time,This timeout represents the absolute amount of time to allow a caller to stay in the queue before the caller is removed from the queue.

Agent Timeout, This timeout specifies the amount of time to try ringing a member's phone before considering the member to be unavailable.

Join Empty, This option controls whether a caller may join a queue depending on several factors of member availability.

Join Queue Announcement,when a caller join the queue, play this announcement file. The sound file can manage in “Sounds List”.

Wrap-Up-Time,after a successful call, how long to wait before sending a potentially free member another call (default is 0, or no delay)

Max Callers,Maximum number of people waiting in the queue (0 for unlimited).

Answered Announcement,whether paly a sound file when the caller be answered.

Answer Announcement file,you can upload a sound file, and select it in here. If empty will use the default system file.

Fail Over Destination,if join the queue failed, can go to another route destination.

[Music on Hold]

It is used for the queue and meetme to play wait music. Select the menu ” Basic Settings ->Music On Hold” to set it. Click on the ”custom” or “default” button to switch the two types.

The “default” type music can not be modify. You can only modify the “custom” type music.

Voice file format requirements: there are three formats can be upload:

1. PCM, 128kbps, 8000HZ, 16bit, Mono, The file extension must be 'wav'.

2. Linear, 256kbps, 16000HZ, 16bit, Mono, The file extension must be 'sln16'.

3. Alaw, 64kbps, 8000HZ, 8bit, Mono, The file extension must be 'alaw'.

You can upload multiple sound files, they will be played loop in sequence.

[Time Conditions]

Use the time conditions, you can controle the different time periods (for example: to work and after work) using the different route destinations.Before set the time conditions, need set the time groups first.

[Time Groups]

The time period is divided into many different groups. Select the menu ” Basic Settings ->Time Groups” to set it.

Click the “Add Time Groups” button to add a time group.

Description, the name of the time group.

Time to start and **Time to finish**, set the hour and minute range.

Week Day start and **Week Day finish,** set the week range.

Month Day start and **Month Day finish**, set the day of month range.

Month start and **Month finish**, set the month range.

If you empty some field, will ignore them.

After the above information is complete and correct, click the “Submit” button to save it. And then you can see it in the “Time Groups List” that is on the right side of the page.

Click on the name of time group in the “Time Groups List”, it will show the detail of this group’s parameters. You can change some parameters, then click the “Submit” button to save it.

After finished the time groups config, then you can set the time conditions. Select the menu ” Basic Settings ->Time Conditions” to set it.

Click the “Add Time Conditions” button to add a time condition.

Time Conditions name, the description of the time condition.

Time Groups, select a time group to use it.

Destination if time matches, if time match, will go to this route destination.

Destination if time does not match, if time not match, will go to this route destination.

After the above information is complete and correct, click the “Submit” button to save it. And then you can see it in the “Time Conditions List” that is on the right side of the page.

Click on the name of time condition in the “Time Conditions List”, it will show the detail of this condition’s parameters. You can change some parameters, then click the “Submit” button to save it.

[IVR]

Config the Interactive Voice Response. Select the menu ” Basic Settings ->IVR” to set it.

Click on the “Add IVR” button to add a IVR.

IVR Name, the description of IVR.

Direct Dial, whether allow the caller to direct dial the extension number.

Announcement, select a sound file to play to the caller.

Timeout, wait the caller press key, if over this time, will play “Announcement” file again. Maximum loop 3 times.

Ext	**Destination**	**Delete**
Press key	Route destination	Click it to remove this rule

Click the ⊕ icon to add more rules.

After the above information is complete and correct, click the “Submit” button to save it. And then you can see it in the “IVR List” that is on the right side of the page.

Click on the name of IVR in the “IVR List”, it will show the detail of this IVR’s parameters. You can change some parameters, then click the “Submit” button to save it.

[Sounds List]

It is used for IVR and Queue. Select the menu ” Basic Settings -> Sounds List” to set it. Voice file format requirements: there are three formats can be upload

1. PCM, 128kbps, 8000HZ, 16bit, Mono, The file extension must be 'wav'.

2. Linear, 256kbps, 16000HZ, 16bit, Mono, The file extension must be 'sln16'.

3. Alaw, 64kbps, 8000HZ, 8bit, Mono, The file extension must be 'alaw'.

You can upload multiple sound files.

[MeetMe]

The PBX has conferencing capabilities. Select the menu ” Basic Settings ->MeetMe” to set it.

MeetMe Number,the phone number of the meeting, extension can dial this number to join the meeting.

MeetMe Name,the name of the meeting, you can input some description in here.

Join PIN,if it is not empty, will require the callers to enter a password before join this meeting.

Join MeetMe Announcement,play a sound file to the caller before join the meeting.

Talker Optimization,Turns on talker optimization. With talker optimization.

Quiet Mode,when user join or quit the meeting, do not play prompt voice.

User Count,announce user count on joining the meeting.

Music on Hold Class, play a music when the meeting has only one user.

After the above information is complete and correct, click the “Submit” button to save it. And then you can see it in the “MeetMe” list that is on the right side of the page.

Click on the name of meeting in the “MeetMe” List, it will show the detail of this meeting’s parameters. You can change some parameters, then click the “Submit” button to save it.

5.2.2Outbound Routes

It is used for extensions do the call out. Select the menu ”Connectivity ->Outbound Routes” to set it.

In the same routing group, matched according to the “Weights” order of small to large matching extension when binding authority, applied to the value of the maximum authority to bind themselves, such as: the existence of privilege is 1,2,3, if authority to bind the extension is 3, 1,2,3 are the extension takes effect.

Route Group, it is relate to the extension’s “RouteGroup”, one group can have many routes.

Weights, it is relate to the extension's "Class", when an extension's class greater than or equal to the "Weights", the extension can use this route rule, else can not used.

Regex, it is the Judging condition.

Trunks, call out used, can add multi trunks.

Priority, the order of check the Regex.

Memo, the description of the route, you can note it.

5.2.3Convert Rule

It can change the prefix of number and add suffix to the number. Select the menu "Connectivity ->Convert Rule" to set it.

ConvertGroup, it is relate to the trunk's "Convert RuleID".

Prefix, it is the Judging condition. '?' is the wildcards. If the header of the dial number match it, then will use this rule.

CutLength,removes the specified length from the front of the number.

AddPrefix,add some numbers in front of the number.

AddSuffix, add some numbers behind the numbers.

5.2.4Call Restriction

[BlackList]

Don't allowed the phone number call in.

Select the menu "Call Manage ->BlackList" to set it.

Support fuzzy matching, can use "%", such as "189%", which means that "189" numbers are matched to the beginning, "%189", which means that the number ending in "189" are matched to, Only fill "%" indicates prohibit all incoming calls.

5.3SIP Advanced Settings

Configure some global parameters of SIP in here. Select the menu "Advanced Settings ->SIP Advanced Settings" to set it.

NAT, When PBX is behind a NAT device, the "local" address (and port) that a socket is bound to has different values when seen from the inside or from the outside of the NATted network. Unfortunately this address must be communicated to the outside (e.g. in SIP and SDP messages), so need do some config.

"no" - use it in a local area network. Use rport if the remote side use it.

"yes" - always set the force_rport option and perform "comedia" RTP handling.

"force_rport" - Pretend there was an rport parameter even if there wasn't

"comedia" - Send media to the port Asterisk received it from regardless of where the SDP says to send it.

IP Configuration, Only when the softswitch behind NAT, and the terminal while another behind NAT, you need to configure this IP.

“Public IP”, if no NAT select this.

“Static IP”, if the **External IP** is a static IP, select this.External Static IP or FQDN as seen on the WAN side of the router.

“Dynamic IP”, if the External IP is a **Dynamic Host**, select this. External FQDN as seen on the WAN side of the router and updated dynamically, e.g. “mydomain.net”. And need config the **Refresh Rate** how often to lookup and refresh the External Host FQDN, in seconds.

Local networks settings in the form of ip/mask such as 192.168.1.0/255.255.255.0.For networks with more 1 lan subnets, you can add more IP separated by commas, such as 192.168.16.1/255.255.255.0,192.168.0.1/255.225.255.0

STUN Server, Address of the STUN server to query. Valid form: [(hostname | IP-address) [':' port]]. The port defaults to the standard STUN port (3478). Set to an empty value to disable STUN. Default is disabled.

MEDIA and RTP Settings

Reinvite Behavior,

“yes” - PBX by default tries to redirect the RTP media stream to go directly from the caller to the callee.

“no” - you want PBX to stay in the audio path, you may want to turn this off.

“nonat” - An additional option is to allow media path redirection (reinvite) but only when the peer where the media is being sent is known to not be behind a NAT (as the RTP core can determine it based on the apparent IP address the media arrives from).

“update” - use UPDATE for media path redirection, instead of INVITE. (yes = update + no nat)

RTP Timers, These timers are currently used for both audio and video streams. The RTP timeouts are only applied to the audio channel. The settings are settable in the global section as well as per device.

“rtptimeout”, Terminate call if this seconds of no RTP or RTCP activity on the audio channel when we're not on hold. This is to be able to hangup a call in the case of a phone disappearing from the net, like a powerloss or grandma tripping over a cable.

“rtpholdtimeout”, Terminate call if this seconds of no RTP or RTCP activity on the audio channel when we're on hold (must be > rtptimeout).

“rtpkeepalive”, Send keepalives in the RTP stream to keep NAT open (default is off - zero).

Registration Settings

Registrations, PBX can register as a SIP user agent to a out registrations, so need set the time out of it.

“registertimeout”, retry registration calls every this seconds (default)

“registerattempts”,Number of registration attempts before we give up 0 = continue forever, hammering the other server until it accepts the registration. Default is 0 tries, continue forever.

Registration Times,

“minexpiry”, Maximum allowed time of incoming registrations (seconds)

“maxexpiry”, Minimum length of registrations (default 60)

“defaultexpiry”, Default length of incoming/outgoing registration.

Advanced General Settings

Bind Address,IP address to bind UDP listen socket to (0.0.0.0 binds to all).

Bind Port,IP port bind UDP listen socket to(default is port 5060).

Bind Address2,for second network.

Bind Port2,for second network.

Audio Codecs,now allow “alaw”,”g729”,”ulaw”,”g723”.

SRV Lookup,“Enabled”, DNS SRV lookups on outbound calls.Note: Asterisk only uses the first host in SRV records. “Disabled” DNS SRV lookups disables the ability to place SIP calls based on domain names to some other SIP users on the Internet.Specifying a port in a SIP peer definition or when dialing outbound calls will supress SRV lookups for that peer or call.

DTMF MODE,set the SIP’s DTMF Payload, can use “INBOUND”, “SIPINFO” and ”RFC2833”, usually use rfc2833.

After the above information is complete and correct, don’t forget click the “Submit” button to save it.

5.4RTP Advanced Settings

You can set the RTP port range. Select the menu ”Advanced Settings ->RTP Advanced Settings” to set it.

RTP START PORT, usually begin from 20000.

RTP END PORT, usually end to 30000.

After the above information is complete and correct, don’t forget click the “Submit” button to save it.

5.5 CDRManage

PBX can send the CDR to another PC by UDP. Select the menu ”Advanced Settings ->CDR Send Settings” to set.

Send the CDR to **Address** at **Port**, **Format** now only have 0, 1 and 2.

Format 0 is same to itself CDR in mysql. The fields are:Call StartTime, Call AnswerTime, Call EndTime, Caller, Callee, Agent, Queue, Shift Number, Transfer Number, Ring Duration, Bill Duration, Process Duration, Call Flag, AnswerStatus, Record Flag.Use commas to separate them.The time format is: YYYY-MM-DD HH:mm:ss

Format 1 is a short type. The fields are:

<table>
<tr><td>Field</td><td>Head</td><td>Call Type</td><td></td><td>Extension Number</td><td></td><td>Internal Number</td><td></td><td>External Number</td><td></td><td>Answer Flag</td><td></td><td>Talk Duration</td><td>Tail</td></tr>
<tr><td>Length</td><td>1</td><td>1</td><td>1</td><td>11</td><td>1</td><td>20</td><td>1</td><td>20</td><td>1</td><td>1</td><td>1</td><td>4</td><td>2</td></tr>
<tr><td>Note</td><td>r</td><td></td><td></td><td>Align left</td><td></td><td>Align left</td><td></td><td>Align left</td><td></td><td></td><td></td><td>Align right</td><td>\r\n</td></tr>
<tr><td></td><td colspan="13">Call Type: 0-other, 1-Call In, 2-Call Out, 3-inner
Answer Flag: “+”- answer, “-“ -no answer</td></tr>
<tr><td colspan="14">e.g. : “r2 01801 01801 56985478 - 0”
It is mean, the extension 01801 call out, caller is 01801, called is 56985478, no answer, duration is 0 second.</td></tr>
</table>

Format 2 is same to Format 1, except the “External Number”. When call out, the “External Number” will be added a company group id.

PBX already have a database to store the CDR itself. Select the menu ”Call Manage ->CDR Manage” to query it.

In “Search” group, you can set the condition of the query, then click the “Query” button to do the query. For example:

	User Group	Call StartTime	Call AnswerTime	Call EndTime	caller	callee	agent	Queue	shift number	transfer number	RingSec	Bill	Duration	call flag	listen status	Monitor	
		2015-05-05 16:40:48	0000-00-00 00:00:00	2015-05-05 16:40:48	20000	20001	20001				0	0	0	inner	no answer		[Delete]
		2015-05-05 16:35:11	0000-00-00 00:00:00	2015-05-05 16:35:11	20000	20001	20001				0	0	0	inner	no answer		[Delete]
		2015-05-05 16:27:45	2015-05-05 16:27:55	2015-05-05 16:27:55	20000	222222	20000				6	0	10	callout	no answer		[Delete]

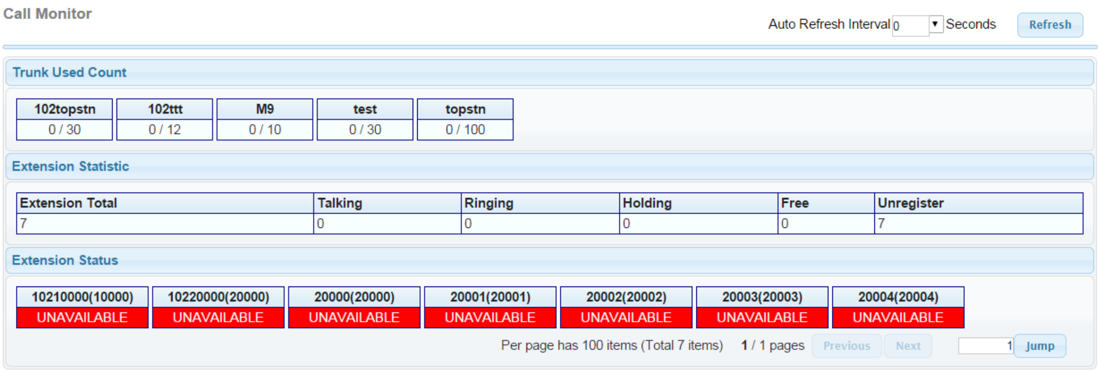
Delete selected Export Per page has 16 items (Total 3 / 271 items) 1 / 1 pages Previous Next 1 Jump

If the call have a record file, you can click the icon whitch in the “Monitor” column to play or download the record file.

You can delete them or export to a CSV file.

Chapter 6 Call Monitor

If you want to check the call status and extensions status, select the menu ”Call Manage ->Call Monitor” to view it. For example:



You can select the Refresh Interval to auto refresh it, or click the “Refresh” button to refresh it at once.

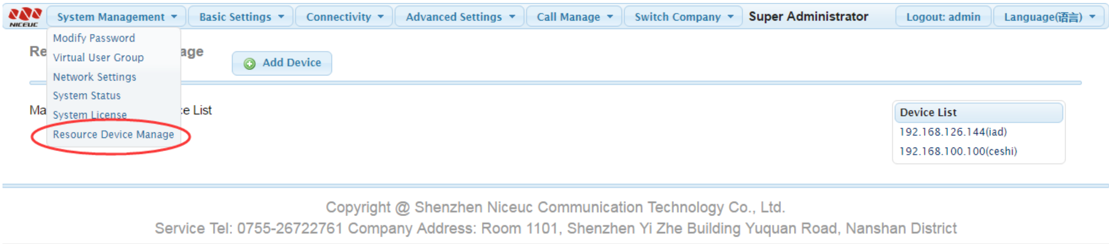
Trunk Used Count, show the used and total channel number of the trunks.

Name of Trunk
Used / Total

Extension Statistic, show the number of Total, Talking, Ringing,Holding, Free, Unregister.

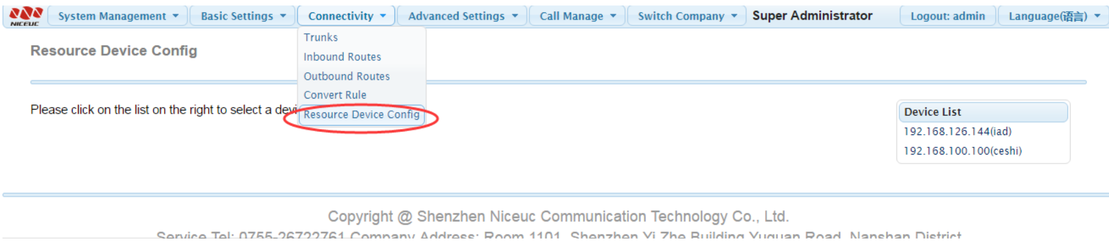
Extension Status, show the register or call status of extensions.

Chapter 7 Resource Device Manage



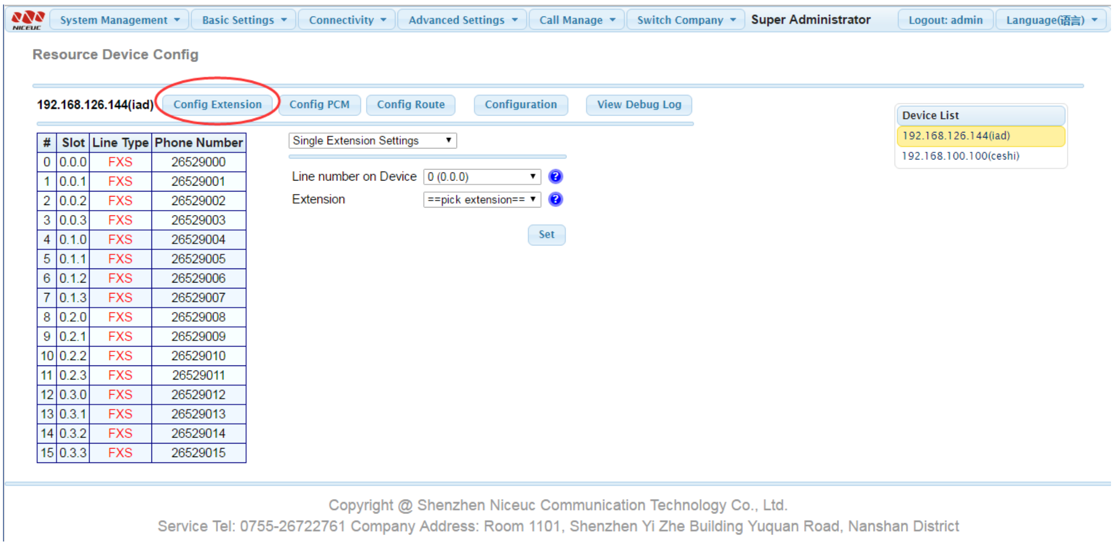
Manage the Devices.

Chapter 8 Resource Device Manage

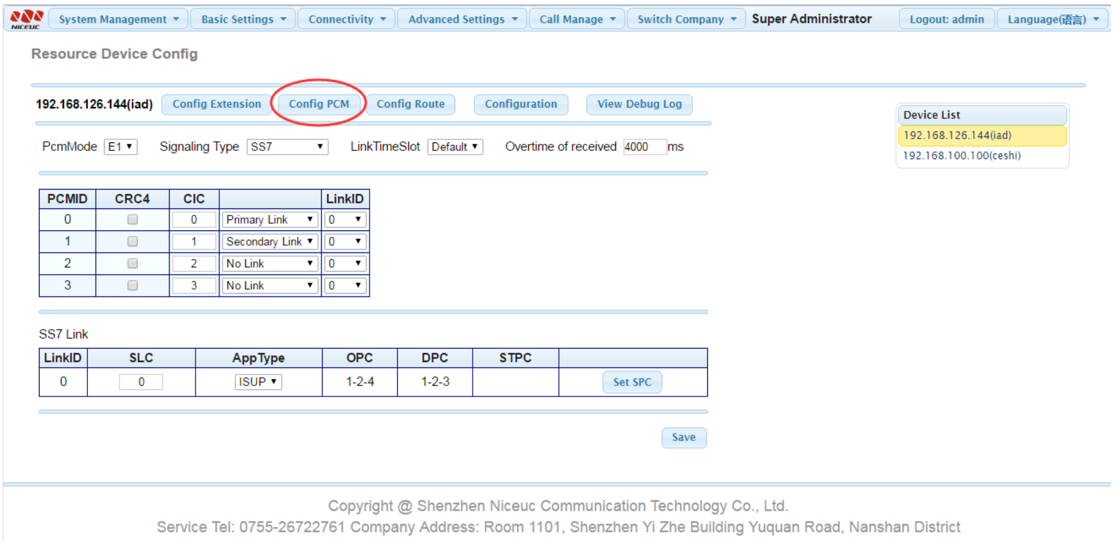


Config the Device

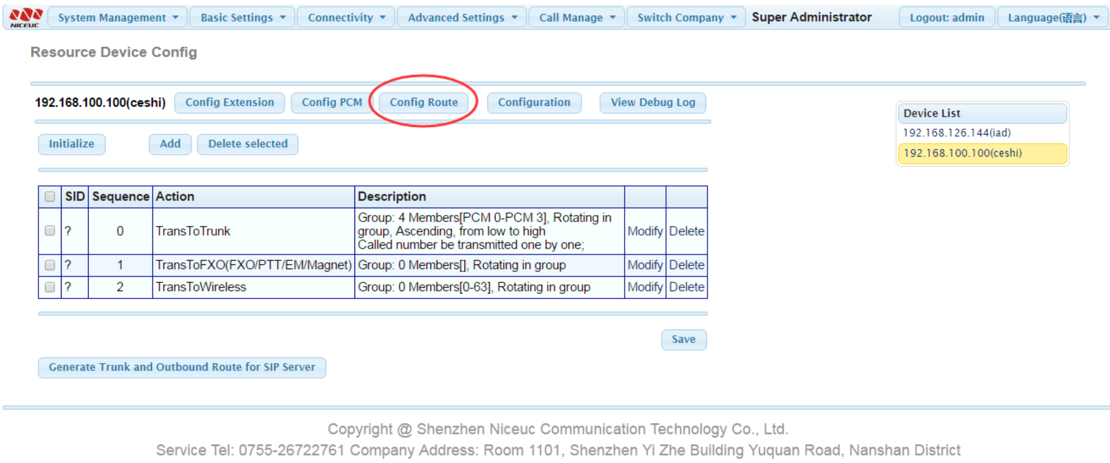
8.1 Config Extension



8.2 Config PCM



8.3 Config Route



8.4 Configuration Manage