New Rock Technologies, Inc.

OM Series IP Telephony System

Administrator Manual

OM4
OM12
OM80
OM200

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Manual Description

This manual is applicable to New Rock's OM series office IP telephony system (referred to as the "device") with firmware release version 74. It describes how to use Web utility to manage the device, including parameter configuration and maintenance operation such as firmware upgrade, capturing log files and etc. Notice that some frequent used parameters can be configured on a phone, as described in OM's **User Manual**.

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1 Start

1.1 Logon to the Device

Step1 Connect your PC to the device

Using a CAT5 cable to connect the device to the local network where the PC is connected, or connect the device directly to the PC.

Step2 Obtain the IP address

Plug an analog phone to one of the Phone ports on the device; pick up the phone and listen to the announcement after dialing ##.

Step3 Login to the Web utility

Enter the device IP address in the browser address bar on your PC, and select English on the logon interface. After entering the password, click **Login**.

Figure 1-1 Login interface

Table 1-1 Login parameters

Item	Description
Language	Select a language.

Item	Description
Role	The Web utility provides two authority levels, administrator and operator. • An administrator is allowed to make changes to any configuration and login passwords. • An operator is allowed to navigate configuration pages and make limited changes to configurations. • The device allows multiple users to login, in which case the administrator has the authority of changing the configuration while the operator is limited to read the configuration only. In any situation, there is only one user allowed to make changes to configurations.
Password	***admin*** by default.

Note

- Before logon to the Web utility, make sure the PC and the device are on the same subnet.
- Please change the default password after your first time logon to the Web utility and keep it secure. See Chapter ***6.2 Changing the administrator password***.

1.2 Directory of Web Pages

If you are unfamiliar with the organization of the Web utility, the directory links on the top of the interface can help you to navigate the Web pages based on the tasks you would like to perform.

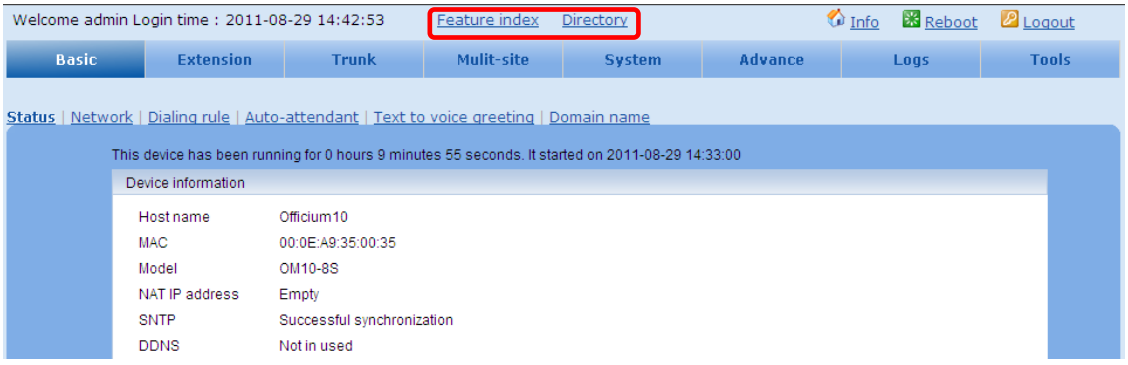
Feature Index

The index of features and functions directs you to the intended pages.

Directory

The sitemap of the Web utility is presented.

Figure 1-2 Directory interface



1.3 Network Settings

The networks settings may need to be changed based on the installation conditions.

Go to the **Basic** > **Network** page for network settings. See Fig. 1-3.

Figure 1-3 Network setting interface

Table 1-2 Network parameters

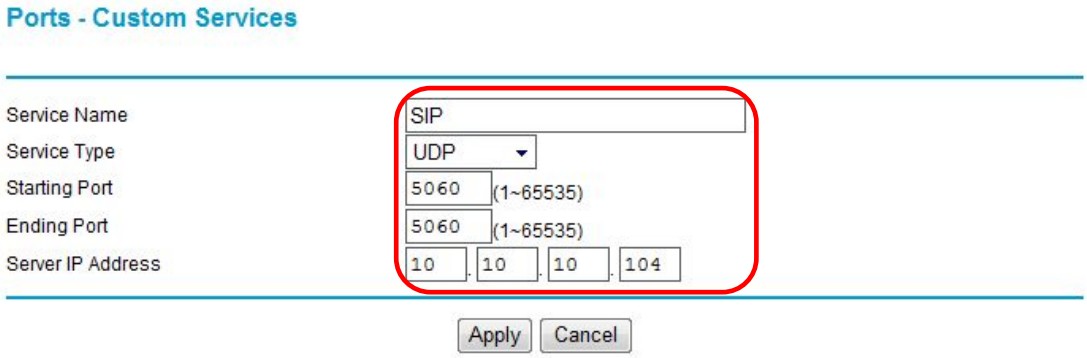
Item	Description
Static	To assign a static IP address. This is the default setting.
DHCP	Use the dynamic host configuration protocol (DHCP) to allocate IP addresses.
PPPoE	Select PPPoE when ADSL modem is connected to the device, and enter username and password obtained from ISP.

1.4 Port Forwarding

Port forwarding is used to permit external hosts to communicate with the device installed behind a router in a private local area network. Such communication is required in remote device management, SIP terminal registration to the device and the SIP trunk between the device and an external device.

The port forwarding is configured on the router, and the example of the configuration on a router is illustrated in Fig. 1-4.

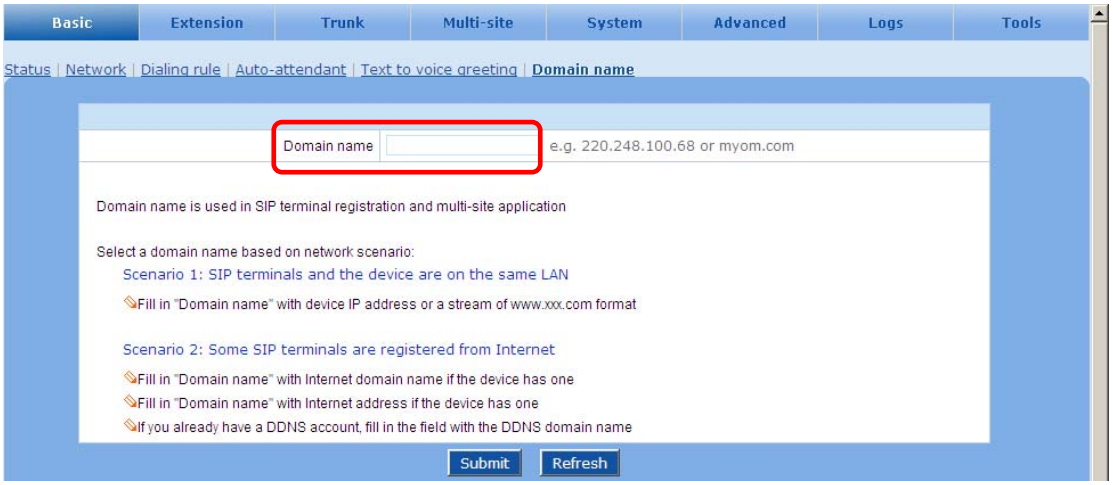
Figure 1-4 Port forwarding interface of router



The WAN port IP address of the router is static

Go to the **Basic** > **Domain name** page and enter WAN port IP address of the router. See Fig. 1-5.

Figure 1-5 Domain name setting interface



The WAN port of the router does not have a fixed IP address

Go to the **Basic > Network** page to enable STUN.

Figure 1-6 Network setting interface

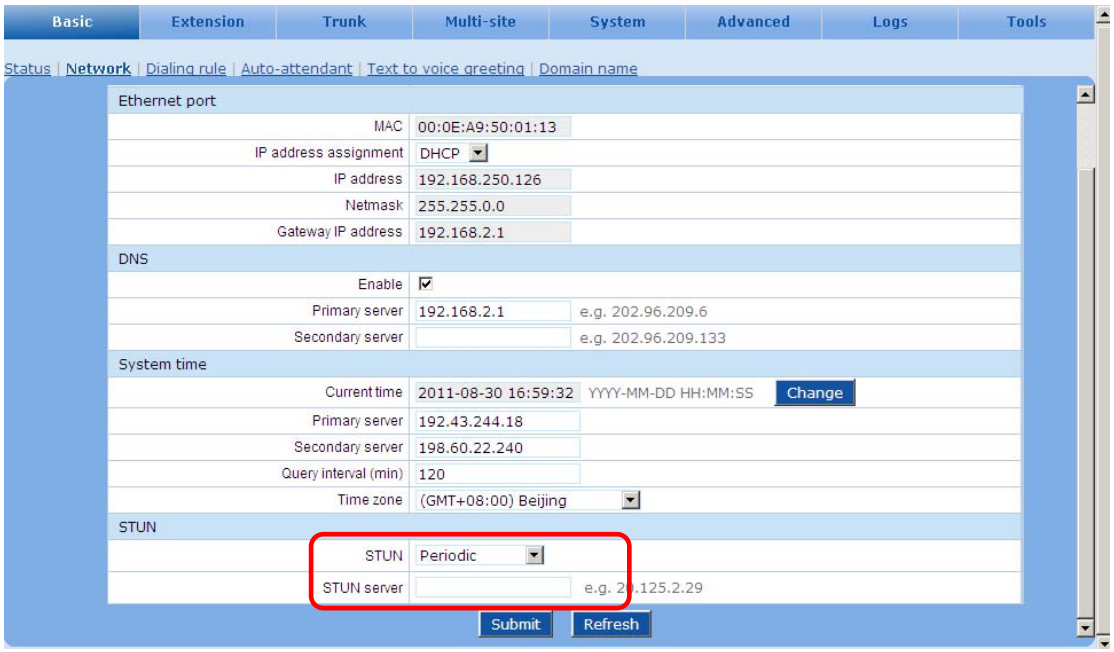


Table 1-3 Network parameters

Item	Description
STUN	Select a trigger for launching STUN inquiry: • Startup: make inquiry for the SIP signaling when the App activated • Periodical: periodically make inquiry • Periodic & RTP: periodically make inquiry for both SIP signaling and RTP stream
STUN Server	Enter the IP address of STUN server, such as *20.125.2.29*.

2 Auto Attendant

2.1 Operators/Receptionists

In conjunction with the auto attendant function, the device supports manual call transfer to extensions. Up to five operator phones can be connected to the device. By factory default, the first Phone port is reserved to the operator with extension number 200.

Click **Basic** > **Auto Attendant** to add more operators or to change auto attendant related settings. See Figure 2-1.

Figure 2-1 Auto-Attendant setting interface

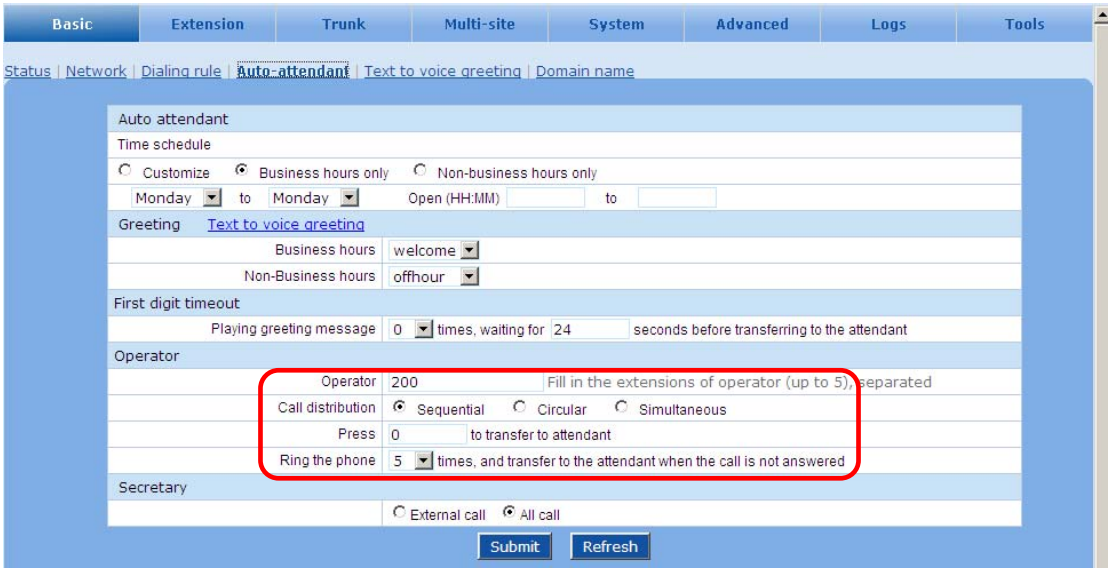


Table 2-1 Auto-attendant Parameters

Item	Description
Operator	Enter up to five operators' extension numbers, separated with ",".
Call Distribution	Select a call distribution scheme below when there are more than one operator: • Sequential: terminate the incoming call to the first available extension on the operator list starting from the first one; • Circular: terminate the incoming call to the first available extension on the operator list in fixed order beginning from the last one left off; • Simultaneous: terminate the incoming call to all available extensions on the operator list simultaneously and the first one to pick up is connected.
Call to an Operator	This dial prefix is used to call an operator. The default prefix is 0.
Ring the phone	When an incoming call is not answered after the set ring times, the device provides busy tone to the caller. The default ring time is 5.

2.2 Greetings

The device provides a greeting voice message to the caller when a call comes in, and the content of the greeting may vary depending on the time.

2.2.1 Factory Default Greetings

There are two default greetings, one for business hours and one for off-business hours, as shown in Table 2-2.

Table 2-2 Default greeting files

Type	File name	Content
business hours	welcome	Thank you for calling. If you know your party's extension, please dial it now. Or, to transfer to an operator, press zero.
off-business hours	Off-hour	Thank you for calling. Our office is closed. If you know the extension, please dial it now.

2.2.2 Changing the Greetings

Three ways are provided to change the greetings:

- Synthesizing the greetings based on the text content.
- Recording the greetings via a phone connecting to the device.
- Uploading the prepared greeting files to the device.

Synthesizing greetings

This is a simple way to customize the greetings in Chinese or English with high voice quality. The synthesizing service is provided by the speech synthesis engine publically accessible on the Internet, offered by New Rock Technologies, Inc., and to perform the synthesis the device is required to connect to the Internet.

Step1 Enable the DNS service

Go to the **Basic > Network** page to enable/configure DNS server. See Figure 2-2.

Figure 2-2 DNS service interface

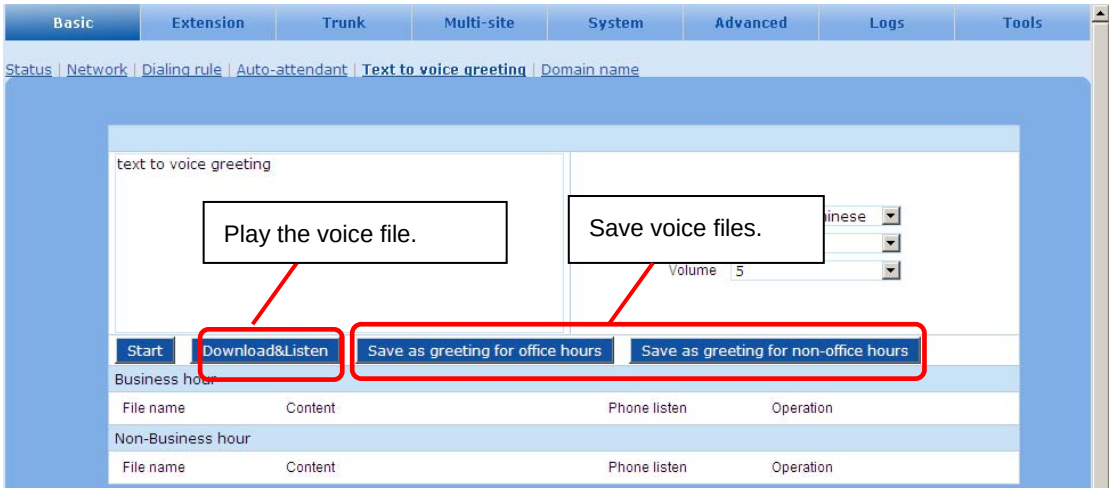
Step2 Synthesizing the greeting file from text

Go to the **Basic** > **Text to voice greeting** page. Input the content of greeting in either English or Chinese and click **Start**.

Figure 2-3 Interface to enter the text of greeting

Step3 Play and save the file

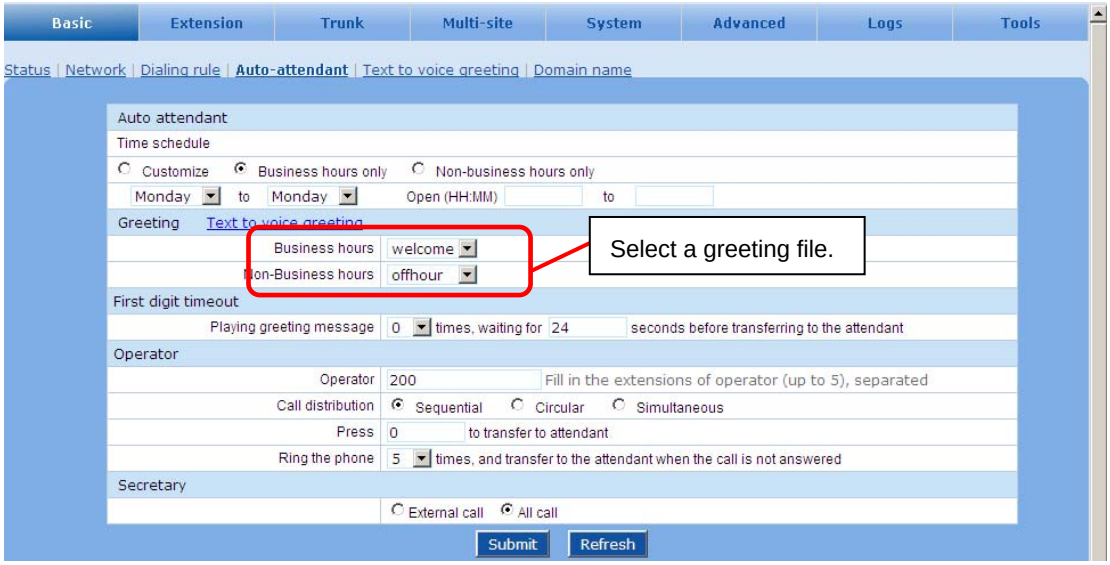
Figure 2-4 Interface to save the greeting files



Step4 Replacing the existing greeting file

Go to the **Basic** > **Auto-attendant** page to select the greeting file. See Figure 2-5.

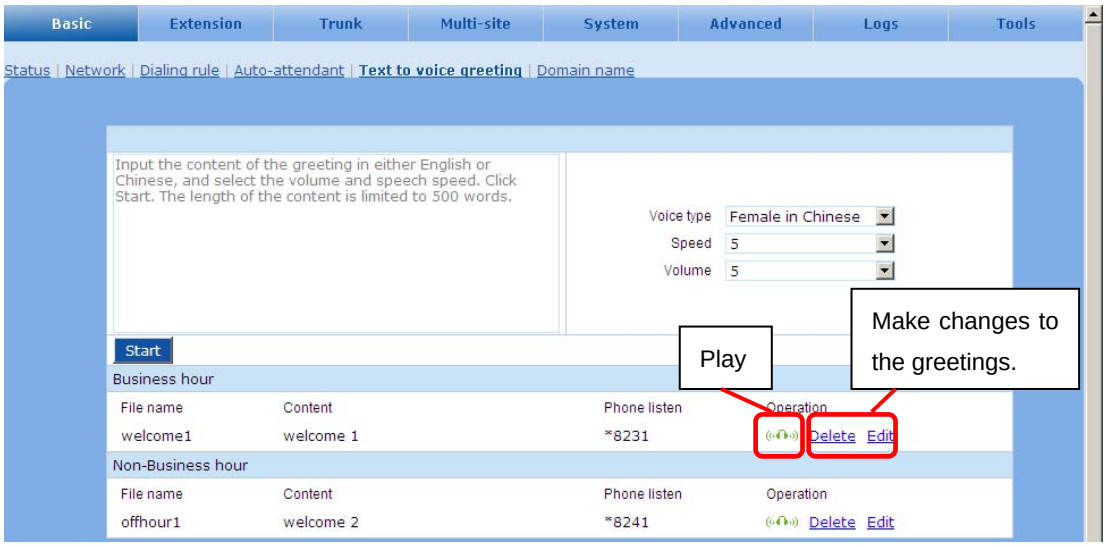
Figure 2-5 Interface to select greeting files



Step5 Managing the greeting files

Go to the **Basic > Text to voice greeting** page for file management.

Figure 2-6 Interface to managing the greeting files



Recording the greeting file via a phone

The greeting file can be recorded directly on a phone by using the operations listed in Table 2-3. To ensure the high quality, it is suggested to make the recording in a quiet room.

Table 2-3 Recording the greeting file on a phone

Item	Operation
Recording	Pick up any phone connecting to the device and press *81 to start the recording after the prompt, and hang up the phone to finish the recording.
Listen	Press *8200 to listen to the voice recording

Item	Operation
Save	• Press *8301 and hang up the phone to replace the welcome file, or • Press *8302 and hang up the phone to replace the off-hour file.
Play the latest greeting file	• Press *8201 to listen to welcome greetings; • Press *8202 to listen to off-hour greetings.
Recovery	Press *8300 to recover replaced voice greeting file.

Note

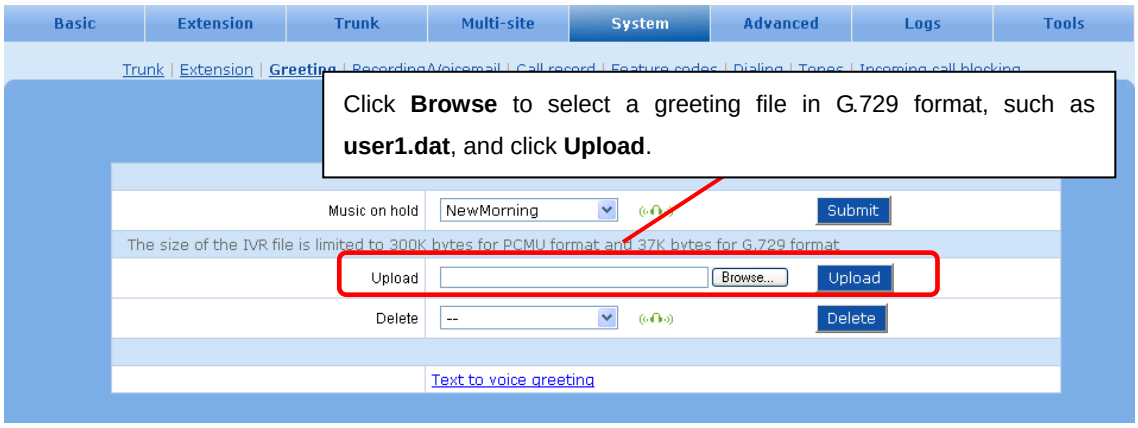
Never restart your device during recording.

Uploading a prepared greeting file to the device

Step1 Uploading the file

Go to the **System** > **Greeting** page. See Figure 2-7.

Figure 2-7 Greeting setting interface



Note

- For G.711 voice file, the name should be in the format of userxxx.pcm, and for G.729 voice file should be in the format of userxxx.dat, where the xxx is a string of characters.
- An audio tool such as 'Cool Edit' can be used to transform the voice file into other formats, like G.711 or G.729.
- Restart the device after the uploading to take effect.

Step2 Replacing the default file with the new one

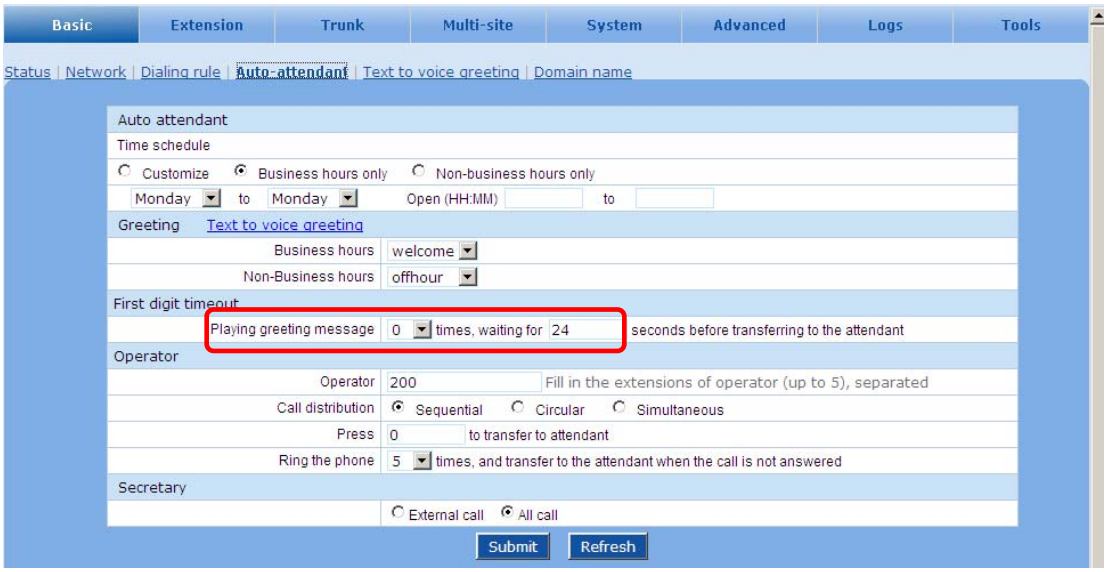
Go to the **Basic** > **Auto-attendant** page to select the greeting file. See Figure 2-5.

2.3 Greeting Repeat

The device plays a greeting to callers of incoming call and followed by a voice menu for callers to select the terminating party. While it waits for callers to enter the selection or enter an extension number it replays the greeting several times. By default, the device replays the greeting three times before it directs the calls to an operator for assistance. The greeting repeat can be set as shown in Fig. 2-8.

Go to the **Basic > Auto-attendant** page to set repeating times. See Figure 2-8.

Figure 2-8 Auto-attendant interface

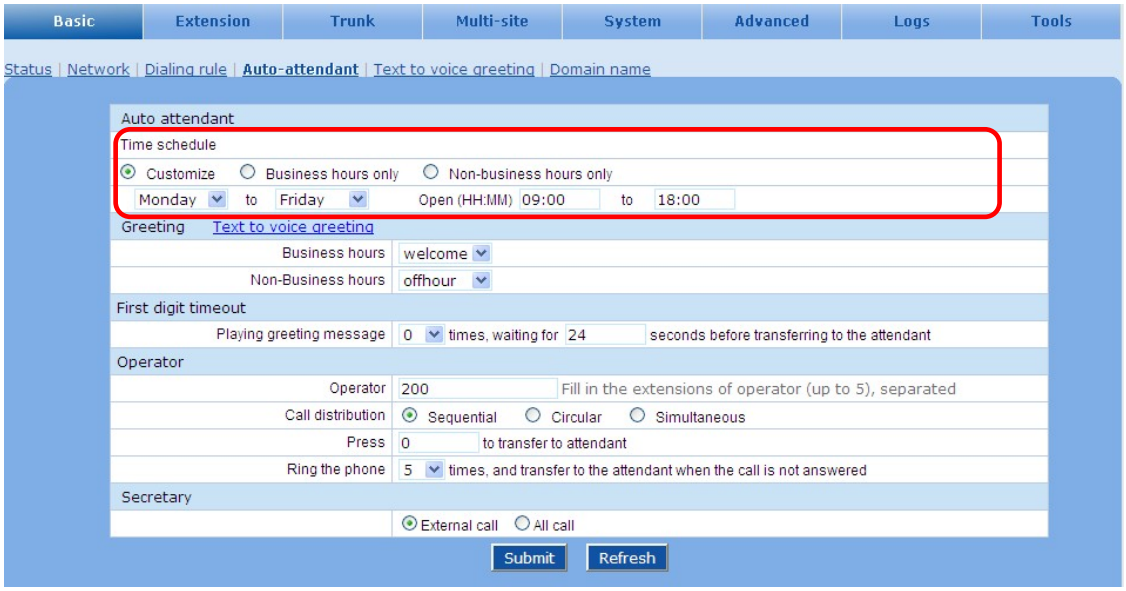


2.4 Playing Greetings Depending on Business Hours

The business and off-business hours and their related greetings can be set.

Go to the **Basic > Auto-attendant** page to define the work schedule.

Figure 2-9 Time schedule setting interface

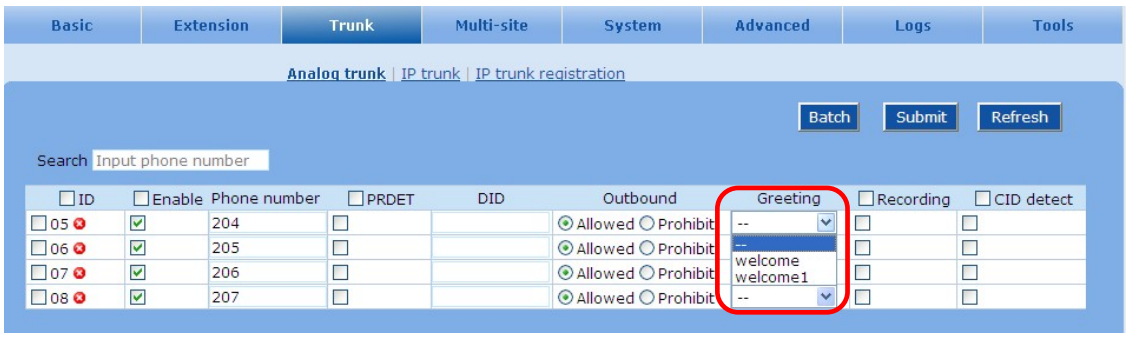


2.5 Assigning a Designated Greeting for a Trunk

In addition to the greeting of the auto-attendant, the device can associate a dedicate greeting to a specific trunk. When an incoming call arrives at the trunk, instead of the greeting of auto attendant the designated greeting is played to the caller.

Go to the **Trunk > Analog trunk** page to select a greeting from the drop-down list. See Figure 2-10.

Figure 2-10 Greeting setting interface



3 Settings of Extensions

3.1 Making Outbound Calls

The device provides two dialing schemes to meet users' dial behavior:

- Making outbound call with prefix: when making an outbound call the user dials an extra digit to select the trunk followed by the destination number;
- Making outbound call without prefix: when making an outbound call the user dials directly the destination number.

The first scheme is the factory default, and the default prefixes are shown in Fig. 3-1. The dial scheme and related settings can be changed.

Go to the **Basic** > **Dialing rule** page. See Figure 3-1.

Figure 3-1 Outbound calls setting interface

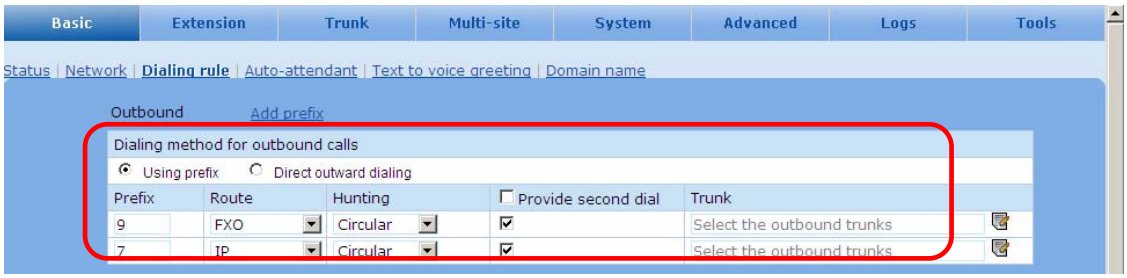


Table 3-1 Dial scheme parameters

Item	Description
Prefix	Choose a Prefix for a specific route;
Route	The device can provides different routes of outbound call for user to select • FXO: the Line port is selected to make the call to PSTN • IP: the SIP trunk is selected to make the call to PSTN • Route table: the route defined by routing table is used to make the call to PSTN
Hunting	If there is more than one analog trunk (Line), the line selection is made according to the following schemes. • Sequential: the first available line on the list is selected starting from the first one • Circular: the first available line on the list is selected beginning from the one left off last time

Note

When direct outward dialing is enabled, press * before dialing another extension number.

3.2 Hunt Group

A hunt group is a set of extensions which are organized to process specific calls. When the dial prefix of the hunt group is dialed by the user the device terminates the call to an available extension in the hunt group. Users can dial the prefix, such as 1 for sales and 2 for support, in both voice menu of auto attendant and direct dialing.

Go to the **Basic** > **Dialing rule** page for hunt group setting.

Figure 3-2 Hunt group setting interface

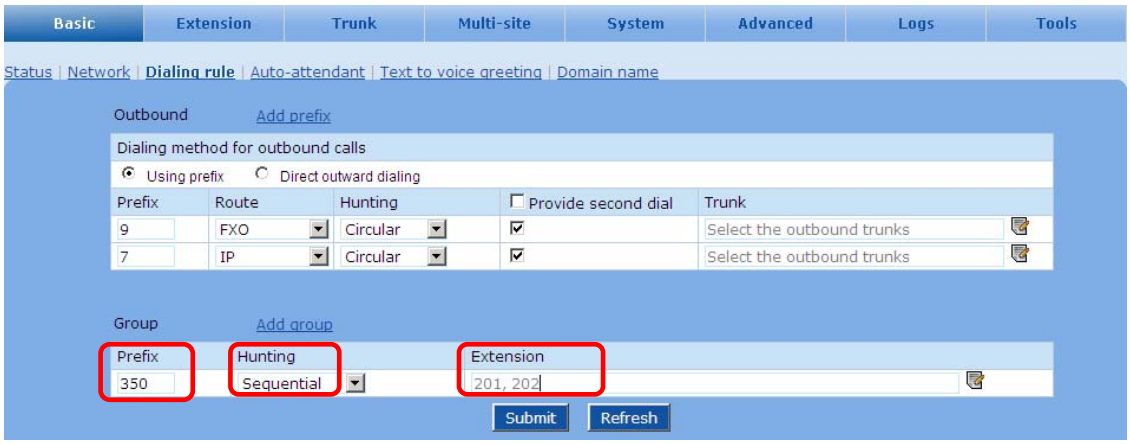


Table 3-2 Hunt group parameters

Item	Description
Prefix	Choose the prefix of a hunt group, like 3.
Hunting	Select a hunting scheme of the group: • Sequential: terminate the incoming call to the first available extension in the group starting from the first one; • Circular: terminate the incoming call to the first available extension in the group in fixed order beginning from the last one left off; • Simultaneous: terminate the incoming call to all available extensions in the group simultaneously and the first one to pick up is connected.
Extension	Enter the extensions in the group, separated by a comma.

3.3 Outbound Call Barring

Each extension has an assigned privilege of making outbound call. When a user makes a call beyond its

restriction, the device rejects the call with a voice announcement. The definitions of restriction level are listed below.

Go to the **Extension** > **Analog ext.** page to define extension's restriction level. See Figure 3-3.

Figure 3-3 Outbound call barring setting interface

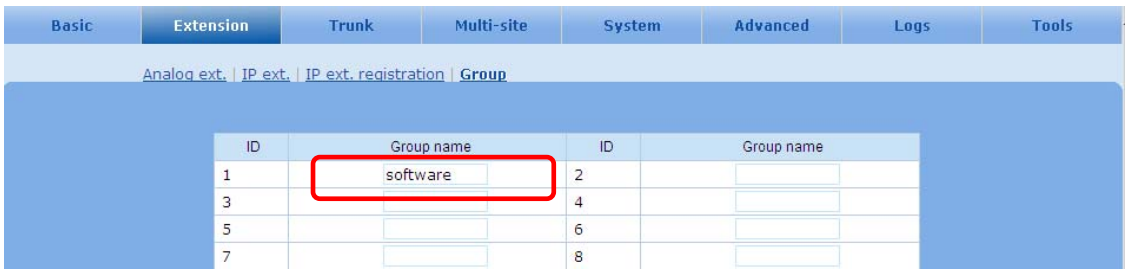
3.4 Group Call Pick up

The users are allowed to pick up a ringing phone of others in the same group by dialing *56. The call pick up group can be set as follows.

Step1 Create the name of a group

Go to the **Extension** > **Group** page to name the group.

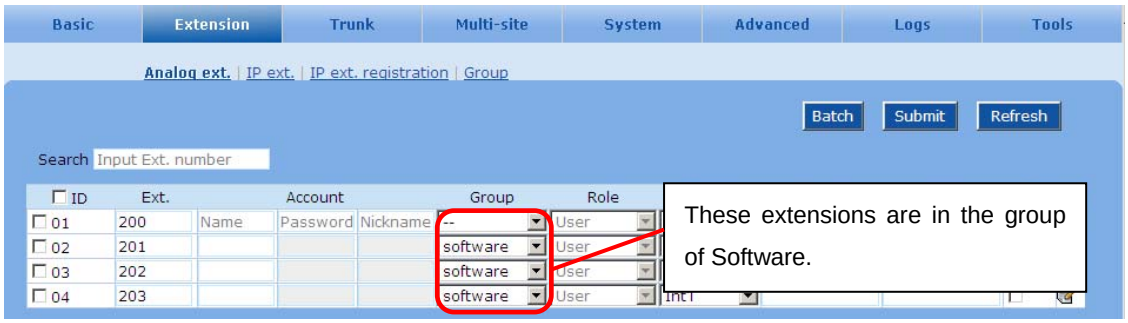
Figure 3-4 Group setting interface



Step2 Assign a group to extensions

Go to the **Extension** > **Analog ext.** page to assign a group to extensions.

Figure 3-5 Group assigning interface



3.5 Fax

If you have fax machines to connect to the device, please follow the steps below.

Connecting a fax machine to the device

The fax machine is connected to a Phone port on the device.

Making a fax call

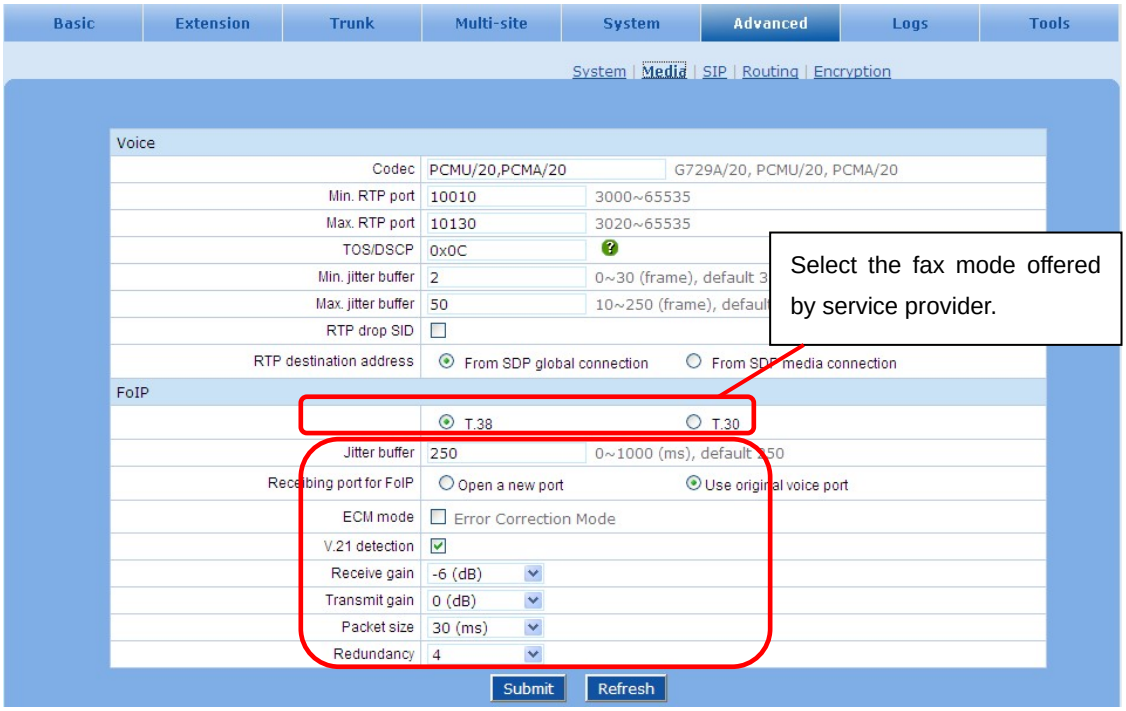
Pick up the phone on the fax machine and enter the receiving party's number or enter the prefix and followed by the receiving party's number, according to the dial scheme.

Fax related settings

To avoid the auto attendant, it is suggested to configure the extension as DID. That is, to bundle the line designated to the fax with the extension which connects to the fax machine.

Go to the **Advanced > Media** page to enable FoIP via IP trunk. See Figure 3-6.

Figure 3-6 FAX setting interface



3.6 Soft Attendant Console

Soft attendant console provides the status of extensions and trunks, and the operators use it to transfer a call by just making a click on the console. The soft attendant console can be assigned to the extensions as an Operator.

Go to the **Extension > Analog ext.** page to assign the role of the extensions.

Figure 3-7 Soft attendant console setting interface

Note

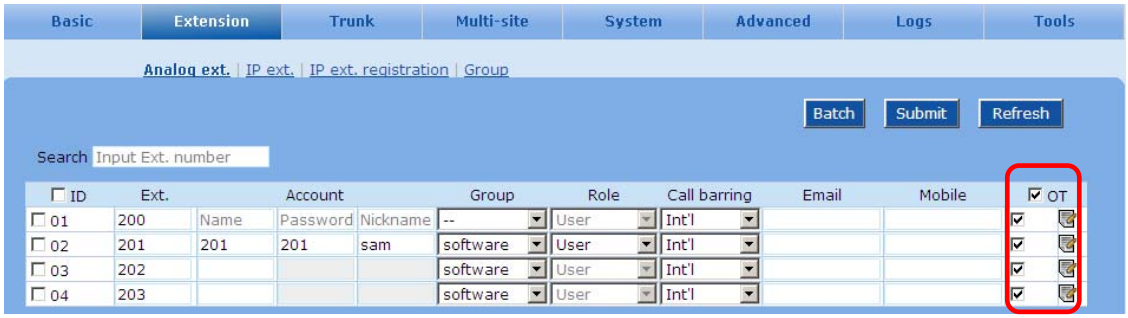
The soft attendant console is a Windows-based software provided by New Rock.

3.7 Transferring an Incoming Call to an External Party

An incoming call is allowed to be transferred to an external party.

Go to the **Extension** > **Analog ext.** page to select **OT**. See Figure 3-8.

Figure 3-8 Outbound transfer setting interface



Note

During an outbound transfer, two lines are used.

3.8 Other Features

The device provides dozens of features for extension users, such as call forwarding, speed dialing, color ring back tone, call forking, recording and etc. Some of the features are enabled as factory default and others need to be enabled by the administrator as shown in below.

Go to the **Extension** > **Analog ext. /IP ext.** page and click the icon to set up other features of extension.

Figure 3-9 Extension's feature setting interface

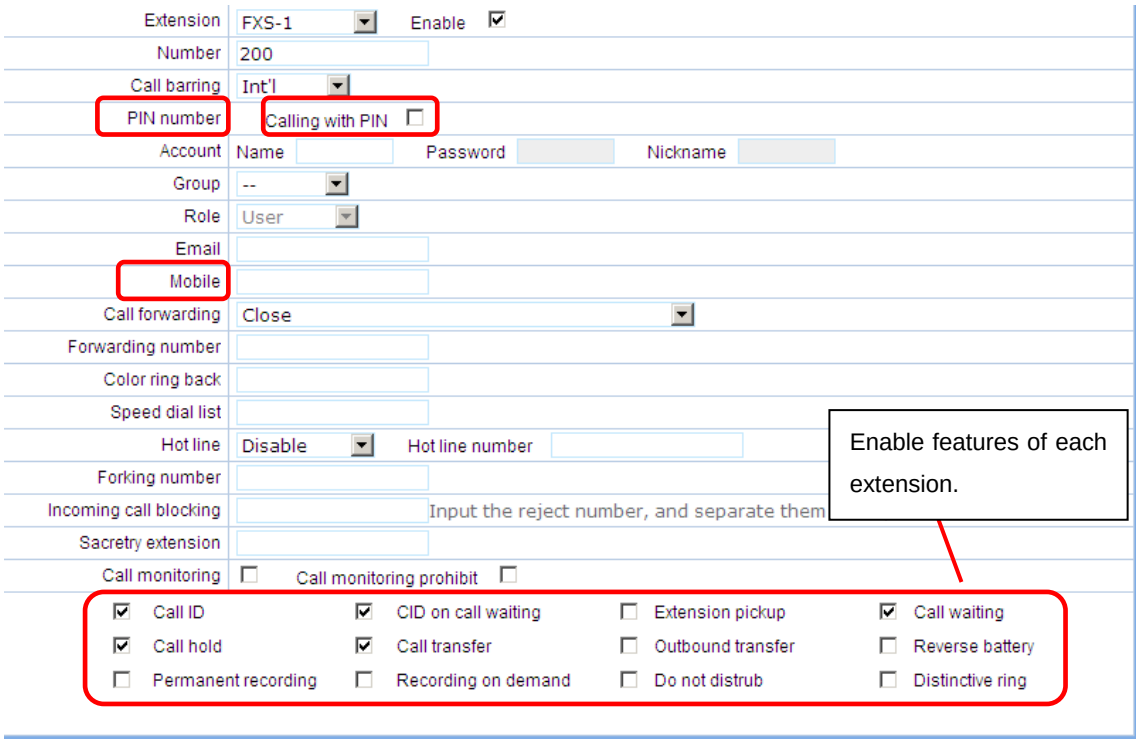


Table 3-3 Extension feature related parameters

Item	Description
PIN number	Using *33 & *99 to allow users to access the device with an external phone.
Calling with PIN	Select it to enable the PIN on the extension. When the user needs to make outbound calls, the PIN should be entered before the calling number.
Mobile	Instead of PIN number a mobile phone number of the user can be used for auto authentication of *33 and *99 for external access.
Call forwarding	Select one of the settings of call forwarding feature: • Disable • Forwarding all calls to another phone • Forwarding unanswered calls to another phone • Forwarding all calls to voicemail • Forwarding unanswered calls to voicemail
Forwarding number	Enter the destination number of forwarding calls. Depending on the outbound dial scheme a prefix may be required, separated from the phone number with comma.
Forking number	The forking function allows the device to terminate a call simultaneously to the extension and another terminal. Enter the phone number of the terminal to enable the feature.

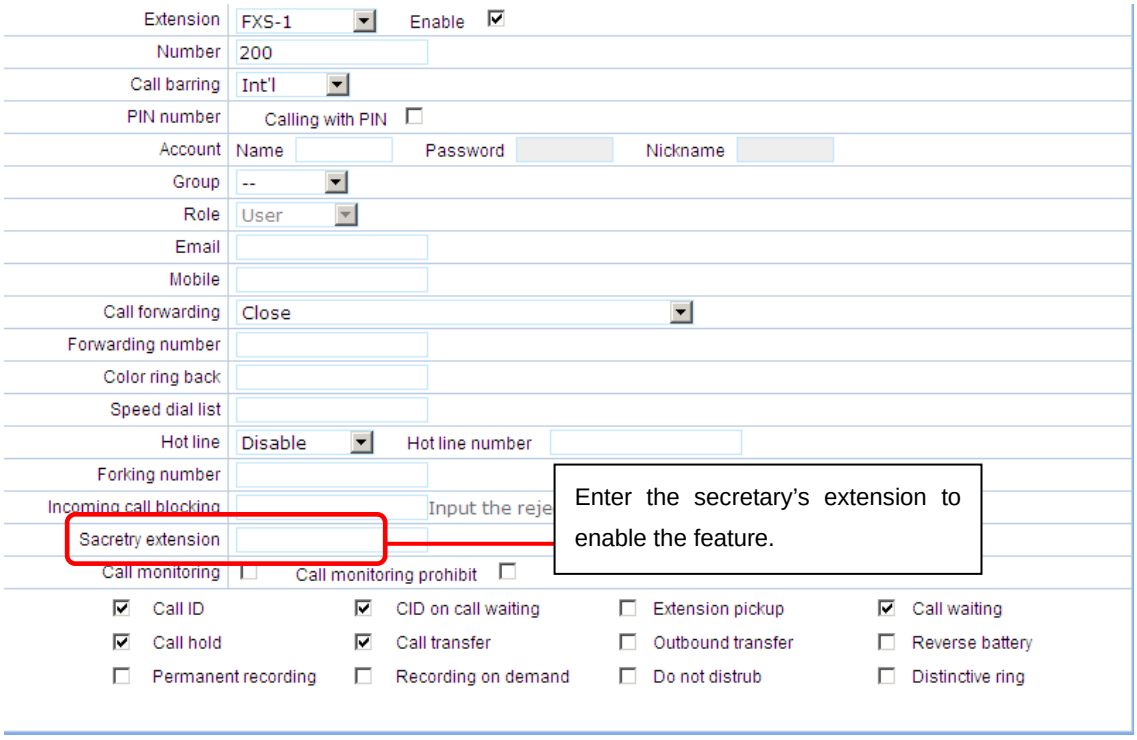
3.9 Secretary Assistant

A secretary's extension can be bundled with his or her manager's extension so that a call to the manager will be redirected to the secretary and further transferred to the manager's extension by the secretary.

Step1 Input secretary's extension number

Enter into the manager's extension feature page, and input secretary's extension number.

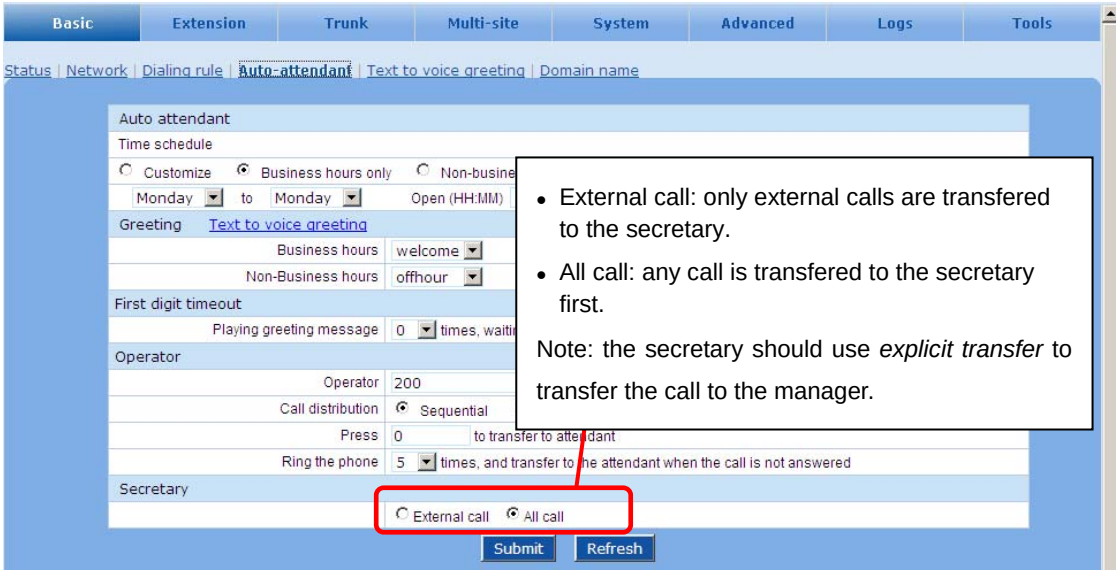
Figure 3-10 Extesion feature interface



Step2 Select the mode of secretary assistant

Go to the **Basic > Auto-attendant** page and select secretary's pickup mode.

Figure 3-11 Auto attendant configuration interface



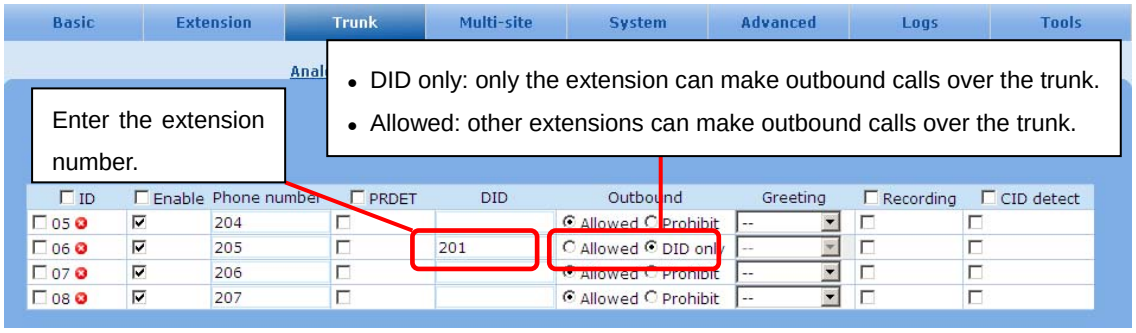
4 Trunk

4.1 Direct Inward Dialing (DID)

An extension can be bundled with a trunk line which is associated with a public phone number, and the device forwards the incoming call to the phone number to the corresponding extension.

Go to the **Trunk** > **Analog trunk** page. See Figure 4-1.

Figure 4-1 DID interface

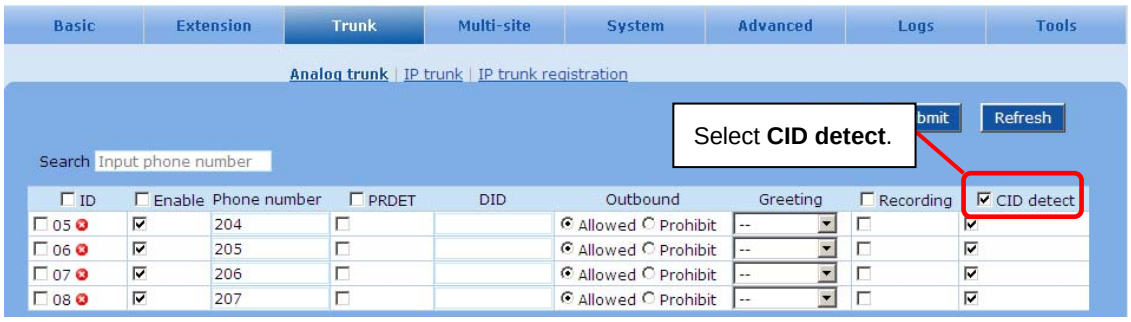


4.2 Caller ID Detection on Line Port

Step1 Enable the caller ID detection feature.

Go to the **Trunk** > **Analog trunk** page to enable caller ID detection.

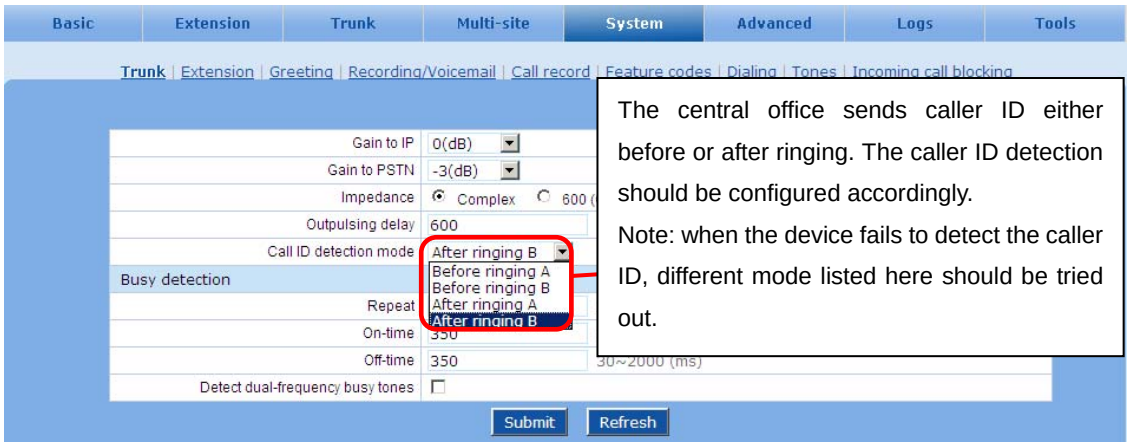
Figure 4-2 CID detect setting interface 1



Step2 Select caller ID standard.

Go to the **System** > **Trunk** page to configure Caller ID detection mode.

Figure 4-3 CID detect setting interface 2

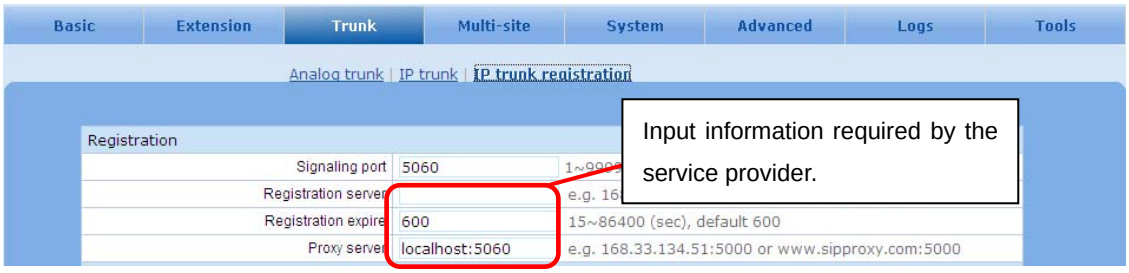


4.3 SIP Trunk Registration to an ITSP

Step1 Configure ISTP server related settings

Go to the **Trunk > IP trunk registration** page. See Figure 4-4.

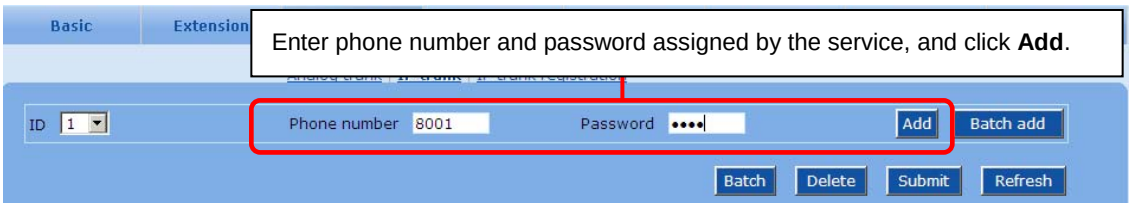
Figure 4-4 IP trunk registration setting interface



Step2 Enter username and password

Go to the **Trunk > IP trunk** page. See Figure 4-5.

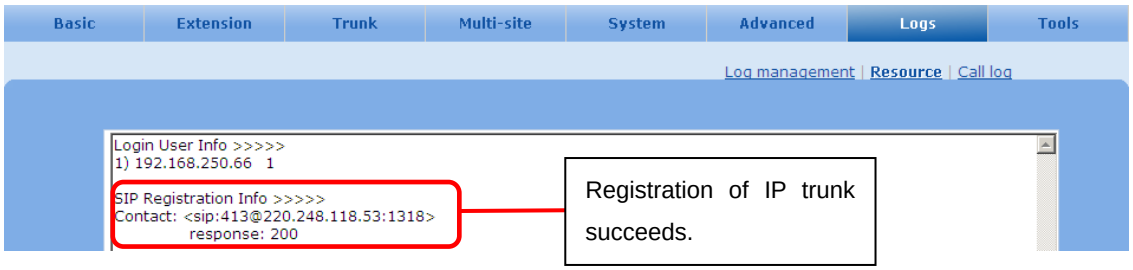
Figure 4-5 IP trunk setting interface



Step3 Verify the registration status

After configuration, go to the **Logs > Resource** page to check the status of registration.

Figure 4-6 Registration status interface



5 Multi-site Telephony Network

Devices installed in one or multiple sites can be interconnected for the following purposes:

- Expanding the capacity of the facility;
- Making direct and free inter-site calling and making outbound call to PSTN via trunks on other devices.

5.1 Multi-site Network with Simple Dialing Scheme

In this simple form of multi-site network, inter-site calls are made by dialing the called party's extension without extra prefix digits. The dialing plan of this scheme is simple and straightforward, and therefore, it is suitable to applications in which multiple devices are stacked to expand the port capacity , and it is also suitable to form a private network of headquarter and its branch offices. The multi-site network with simple form requires a carefully designed numbering plan for each device to avoid potential number conflicts.

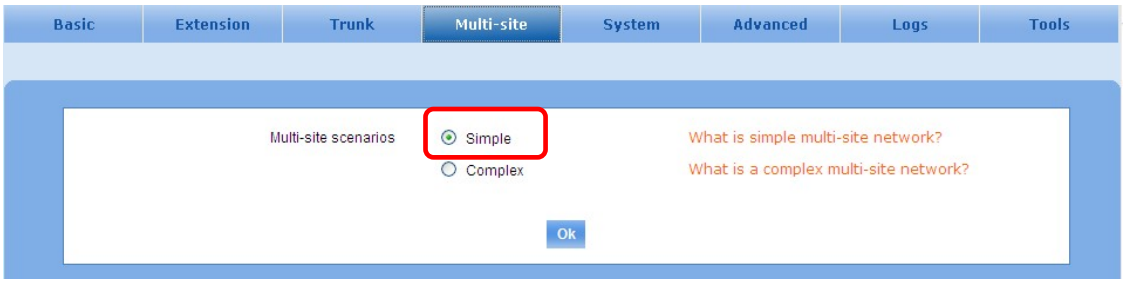
The configuration of multi-site network with simple form includes the configurations of the device on the managing site and the devices on regular sites. The device list of the multi-site network is maintained by the device on the managing site, which distributes the device list to the devices on the regular sites, illustrated as below.

The configuration on the managing site

Step1 Select multi-site network form

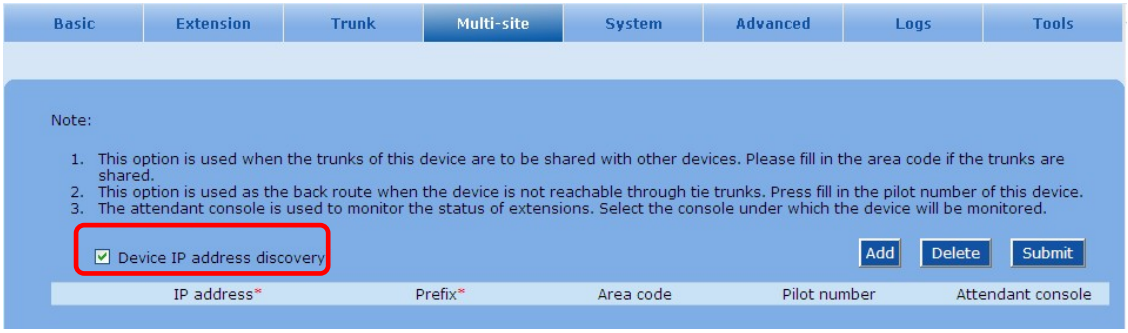
Go to the **Multi-site** page and select **Simple** as the scenario. See Figure 5-1.

Figure 5-1 Multi-site scenarios



Step2 Enable auto discovery of devices

Figure 5-2 Auto discovery of devices



Step3 Add the managing site to the device list

Click **Add** to enter information of the managing site.

Figure 5-3 Device list setting interface

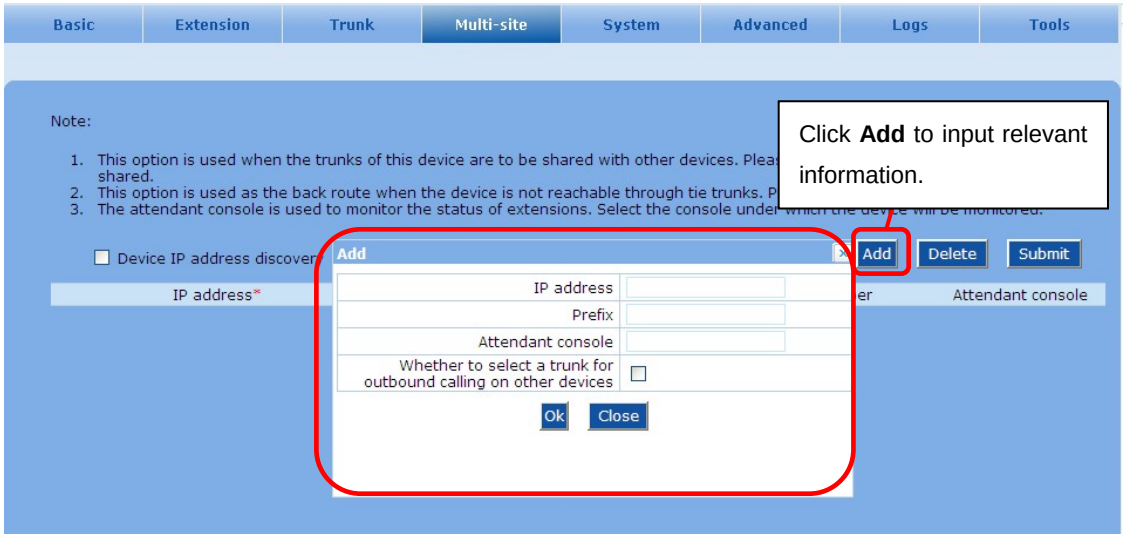


Table 5-1 Multi-site network parameters

Item	Description
IP address	IP address and port number of a device. Note: the address of the managing site must be set on the top of the list.
Prefix	The expression of the extension numbers of the device. For example, 2/3 expresses 3-digit number started with 2. Note: The numbering plans of devices involved in the network must be carefully examined to avoid potential number conflicts. A number conflict is a situation in which a prefix of an extension number on one device is identical to the prefix of an extension number on another device. E.g.: suppose the expressions of device A and B are 2/3 and 21/3, respectively. When the user of device A dials 210, an extension in local, the call could be routed to device B.
Whether to select a trunk for outbound calling on other devices	Allow the trunks on the device to be used for outbound calling originated from other devices.
Area code	Fill in the area codes of the device. A call to the region covered by the area codes could be routed via the trunk of the devoce.

Item	Description
Pilot number	Enter the pilot number of this device. When the IP connection is disconnected, the inter-site calling will be routed via PSTN.

Step4 Add other devices to the list

Click **Add** to input information of other devices in the network.

Step5 Broadcast the device list

The managing site, which is on the top of the list, will send the latest device list to other devices in the network.

Regular Site

Step1 Select multi-site network form

Go to the **Multi-site** page and select **Simple**. See Figure 5-1.

Step2 Enable the auto discovery

Step3 Add managing device to the list

Click **Add** to input relevant information of managing site. See Figure 5-3.

Step4 Add this device to the list

Click **Add** to input information of the device.

Step5 Get the latest device list from managing site

The managing site, which is on the top of the list, will send the latest device list to other devices in the network.

5.2 Multi-site Network with Complex Dialing Scheme

In this scheme, special dialing digits are added at the front and the end of the destination number to avoid potential dialing number conflict. With this form, a large scaled multi-site telephony network can be built up, in which the numbering plan of each device can be managed independently.

5.2.1 Configuration of Managing Device

The managing device is one of the devices in the multi-site telephone network, which is responsible to

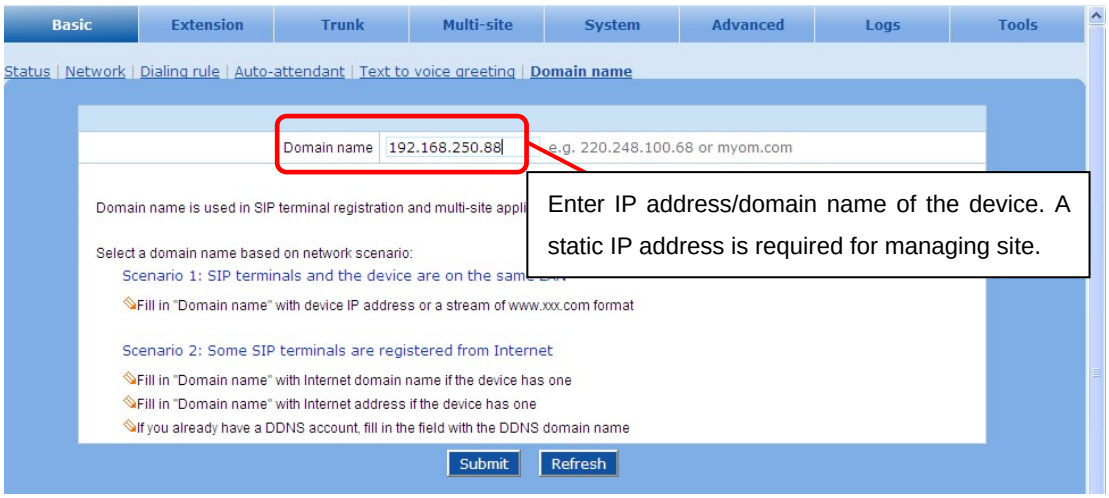
- Authenticate the devices in the network
- Maintain and broadcast the latest address information of devices
- Make change and broadcast the site prefix information
- Make change and broadcast the outbound trunk sharing information
- Response to the inquiry of the above information from other devices in the network

The procedure of configuring the managing device is illustrated below.

Step1 Input the domain name of the managing device

Go to the **Basic > Domain name** page. Enter IP address of managing site and click **Submit**.

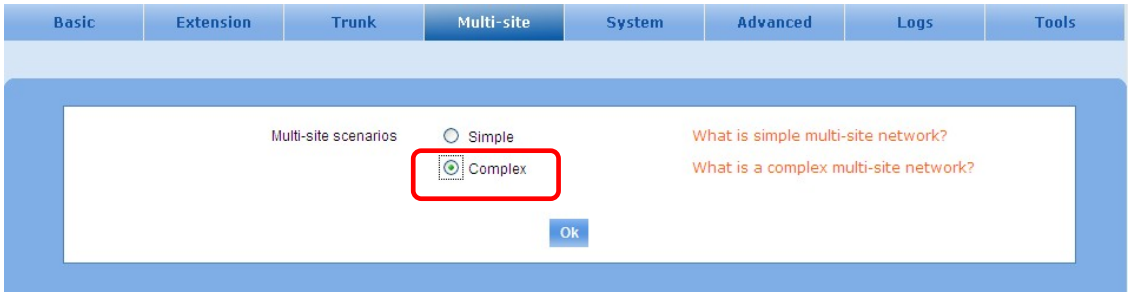
Figure 5-4 Domain name interface



Step2 Select multi-site form

On **Multi-site** interface, select **Complex** for Multi-site scenarios, and click **OK**. See Figure 5-5.

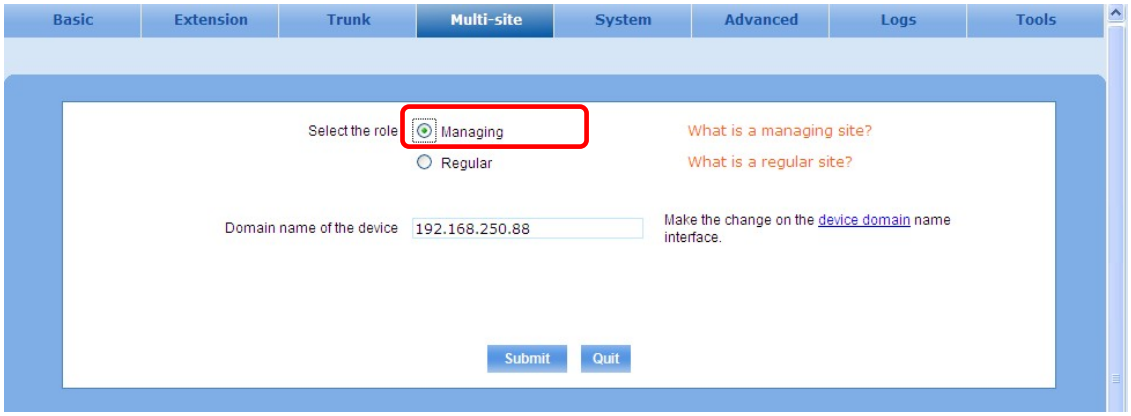
Figure 5-5 Multi-site scenarios setting interface



Step3 Select the role of a managing site

Select **Managing** as the role and click **Submit**.

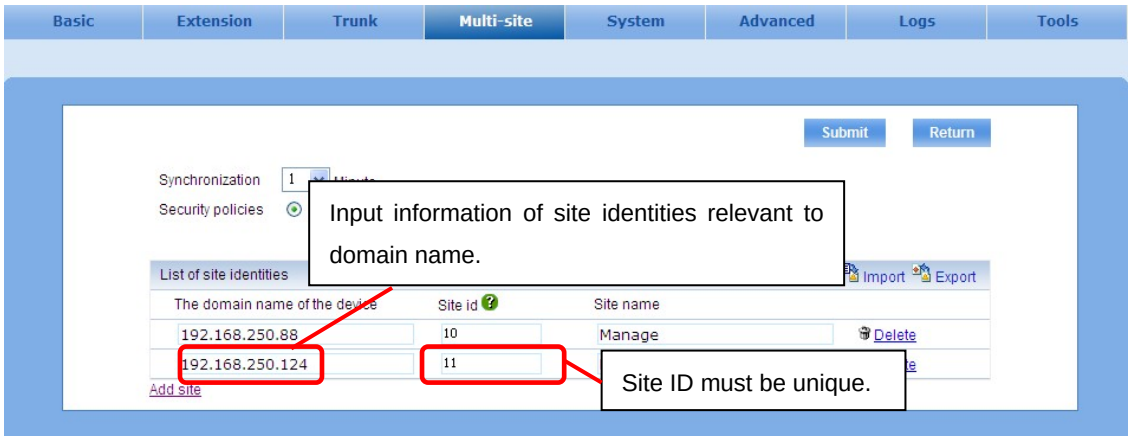
Figure 5-6 Role selecting interface



Step4 Create the device list

Enter into the **Add and update sites** page. See Figure 5-7.

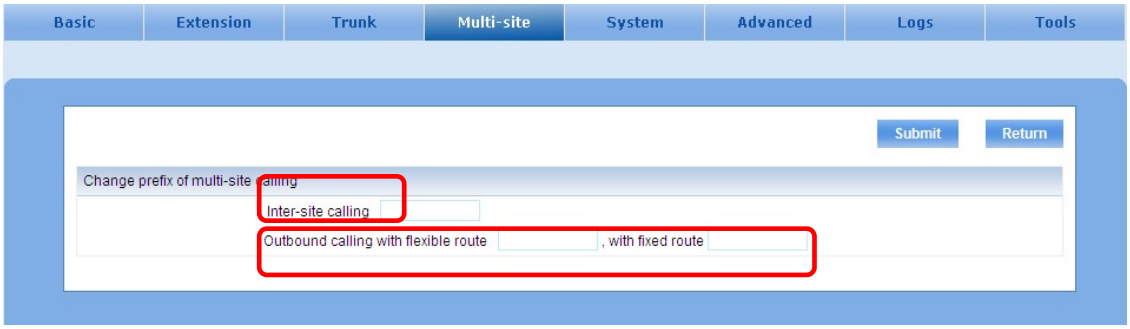
Figure 5-7 Device list setting interface



Step5 Create or update the dialing prefix

Enter into the **Change prefix of multi-site calling** interface to create or change the dialing prefix. See Figure 5-8.

Figure 5-8 Prefix setting interface



Note

The domain name and the port number will be used in configuring the device list on managing site.

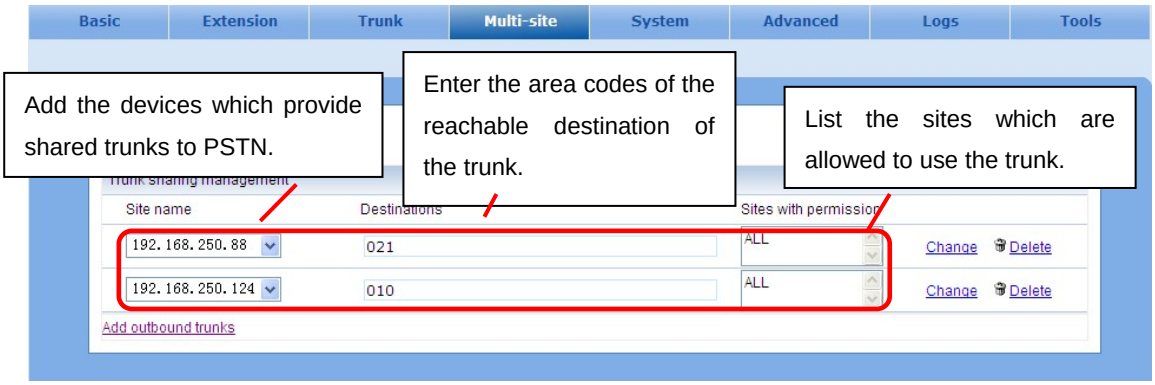
Dialing prefix of inter-site calling is dialed before the extension number of the remote site. For example, **#1-11-200#**, in which **1** is the prefix, **11** is the site id of the called party, and **200** is the extension of the called party.

Dialing prefix of outbound calling is dialed ahead of the destination number. For example by dialing **#1-11-8005551212#** the call to **8005551212** will be routed to PSTN through a trunk on site **11**, and by dialing **#2-8005551212#** the call to **8005551212** will be routed to PSTN through a trunk selected by the managing site.

Step6 Configuration of trunk sharing

Go to **Trunk sharing management** interface.

Figure 5-9 Trunk sharing interface



Note

Unless the site ID is dialed for the outbound call to PSTN, the device will search the allowed outbound trunk from the top of the list.

5.2.2 Regular Site

Step1 Enter the device domain name

On the **Basic > Domain name** interface, enter domain name of regular sites and click **Submit** seen in Figure 5-10.

Figure 5-10 Domain name setting interface

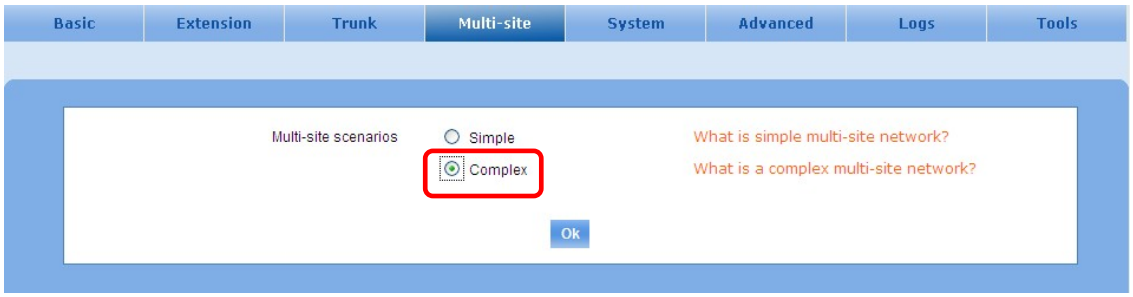
Note

The domain name and the port number will be used in configuring the device list on managing site.

Step2 Select scenarios of the multi-site application

On **Multi-site** interface, select **Complex** and click **OK**.

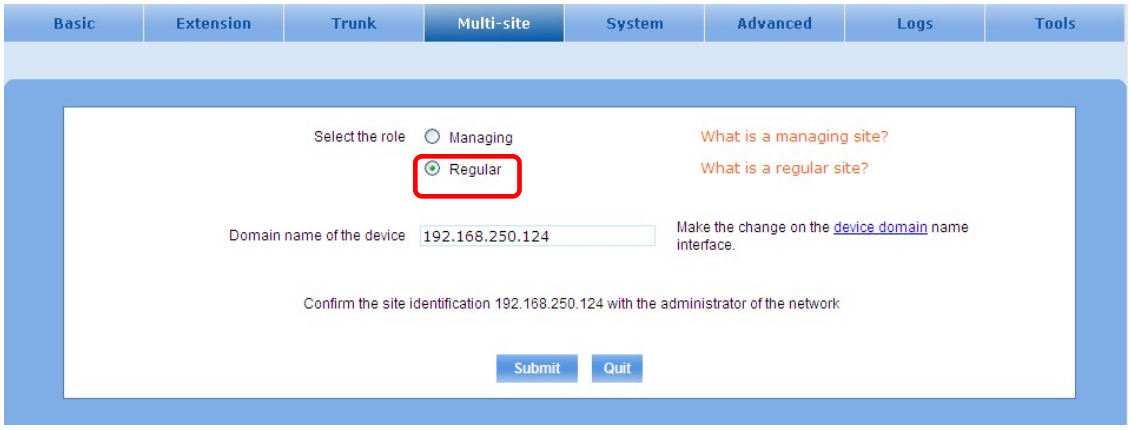
Figure 5-11 Interface of multi-site scenarios



Step3 Select the role

Select **Regular** as the role and submit. See Figure 5-12.

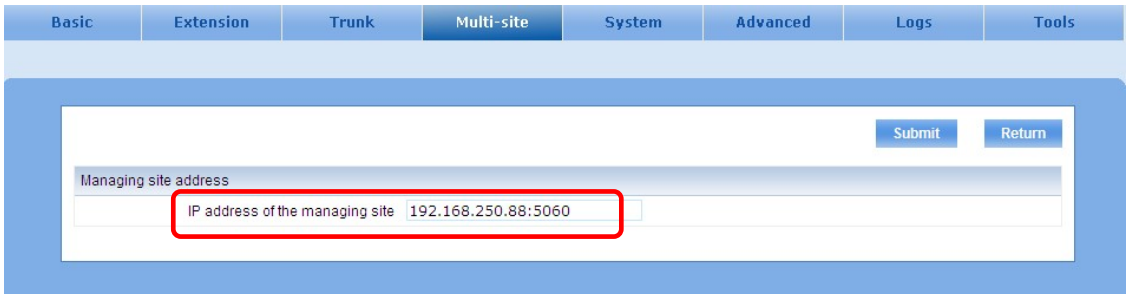
Figure 5-12 Interface of site role



Step4 Enter the IP address of managing site

On the **Managing site address** interface, enter IP address of the managing site.

Figure 5-13 Managing site address interface



When configuration succeeds, the icon turns green.

Figure 5-14 Multi-site networking status interface

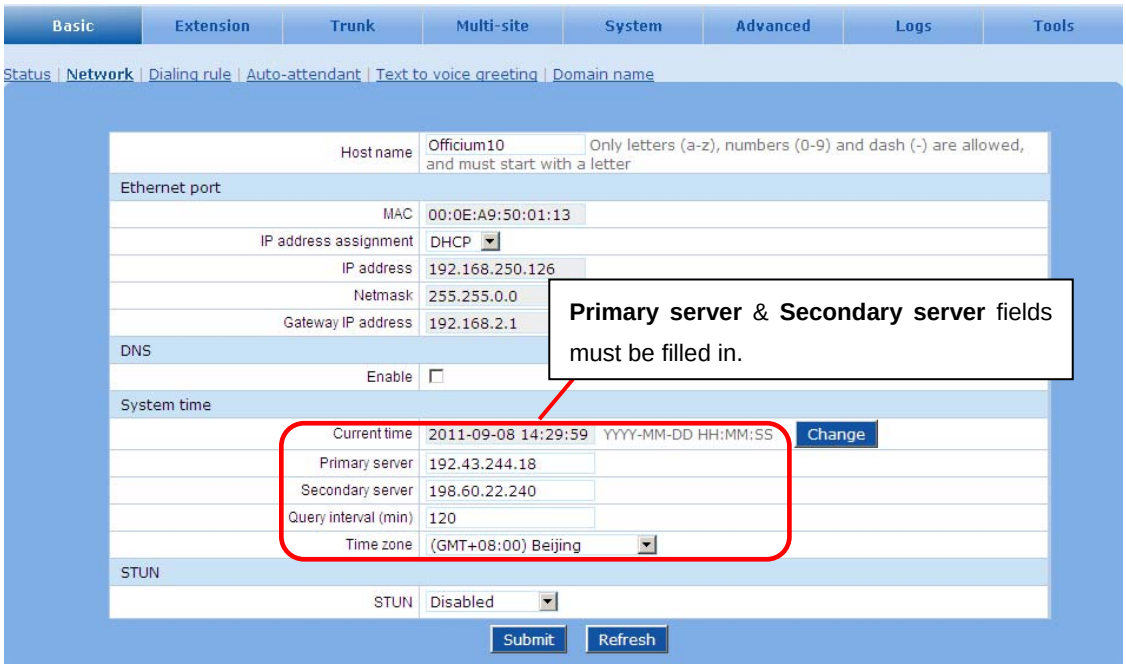
6 Device Settings

6.1 Time

The device obtains its time from a time server in the network, and in case the time service is not available, the timer of the device can be set manually through the Web utility.

Go to the **Basic > System time** page.

Figure 6-1 System time interface



Note

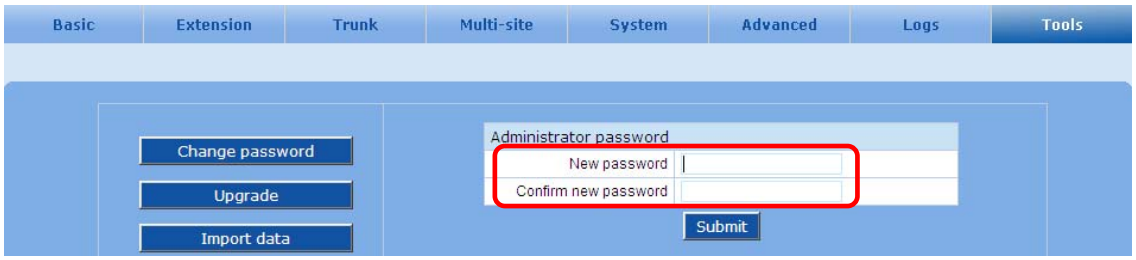
If the system fails to synchronize to the time server, click **Change** to manually set the time.

6.2 Changing the Administrator Password

It is strongly recommended to change your password after you login the Web utility the first time.

On the **Tools** interface, click **Change password** to modify administrator password.

Figure 6-2 Interface of changing password



6.3 SIP Client Account

A SIP based voice terminal can be connected to the device, including softphone and SIP phone. After registering to the device, the SIP terminals can communicate to other terminals connected to the device, such as terminals on FXS ports, SIP phones, and remote terminals through analog trunks, tie trunks and SIP trunks.

A SIP account needs to be set up on the device for each SIP terminals. The device will reject any visit from a SIP terminal which fails in authentication.

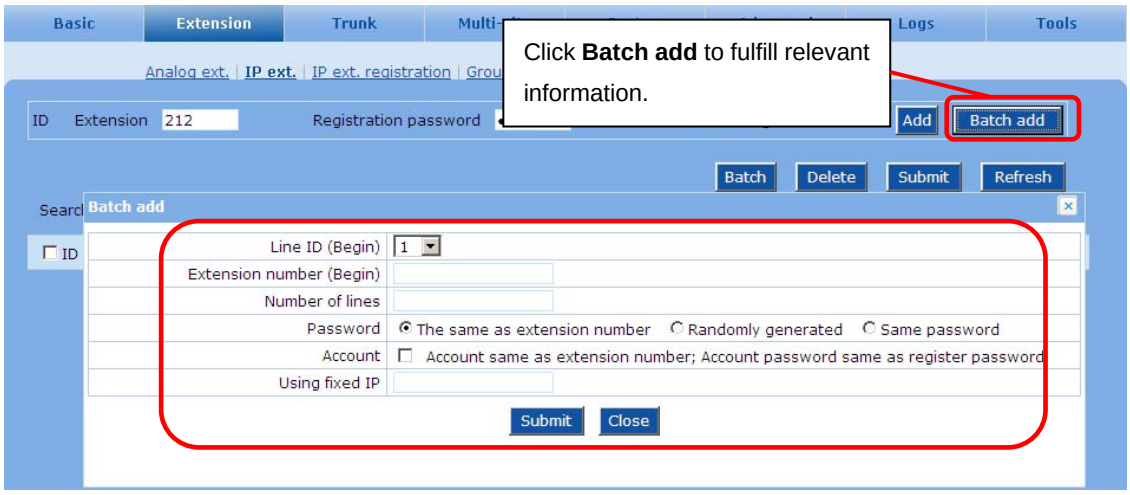
On the **Extension > IP ext.** interface, click **Add** to register. See Figure 6-3.

Figure 6-3 IP extension interface

Adding IP extension in bulk

Go to the **Extension > IP ext.** page, and click **Batch add** to input relevant information.

Figure 6-4 Adding IP extension in bulk interface



6.4 Call Recording and Voicemail

A file server is required in order to use call recording and voicemail functions. In addition, a mail server is required if the messages of the voicemail are forwarded to the receipts through email.

Step1 Configure the file server and mail server addresses

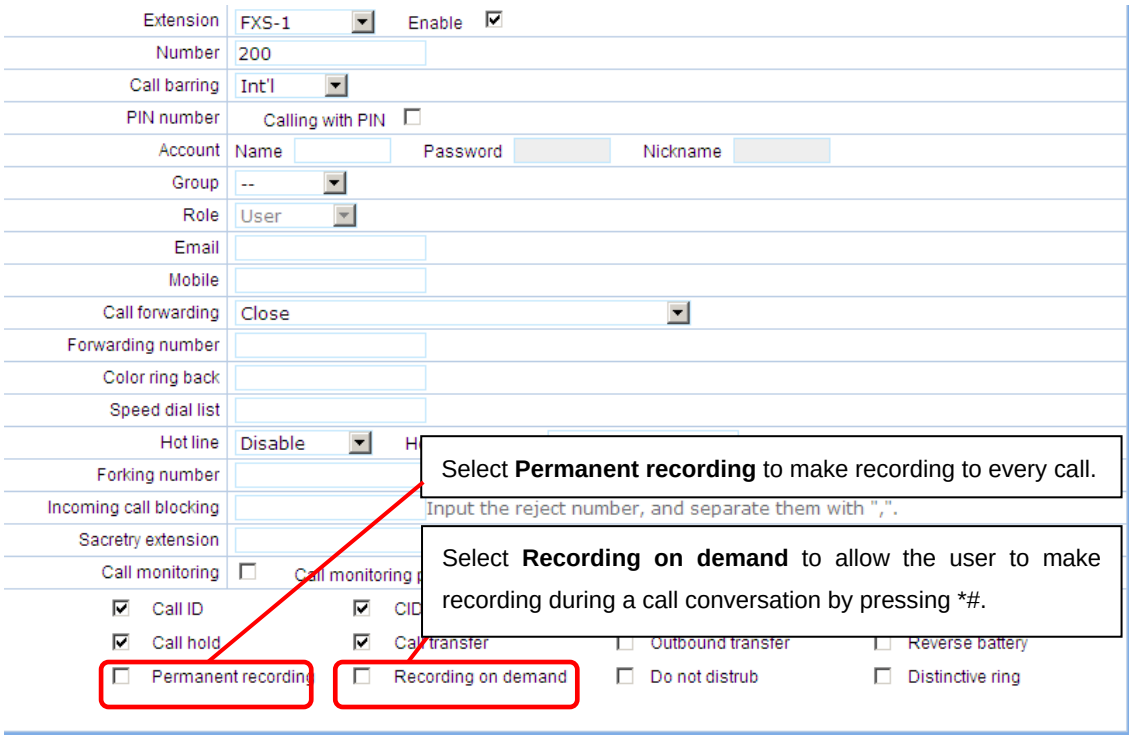
On the **System > Recording/Voicemail** interface, enter relevant information of recording/mail server.

Figure 6-5 Recording/Voicemail interface

Step2 Enable the recording function of an extension

On the **Extension > Analog/IP ext.** interface, click the icon to enter into configuration page of extension and enable recording.

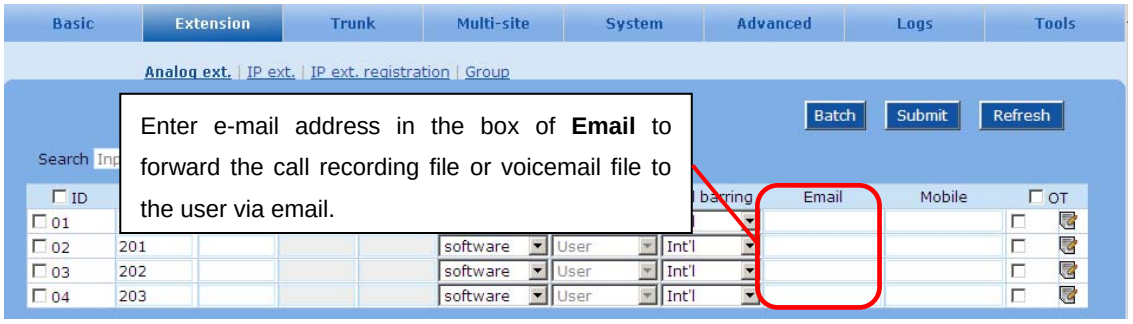
Figure 6-6 Interface of extension features



Step3 Fill in mail server address

Go to the **Extension > Analog ext.** interface, and fill in mail server address.

Figure 6-7 Interface of mail server settings



Note

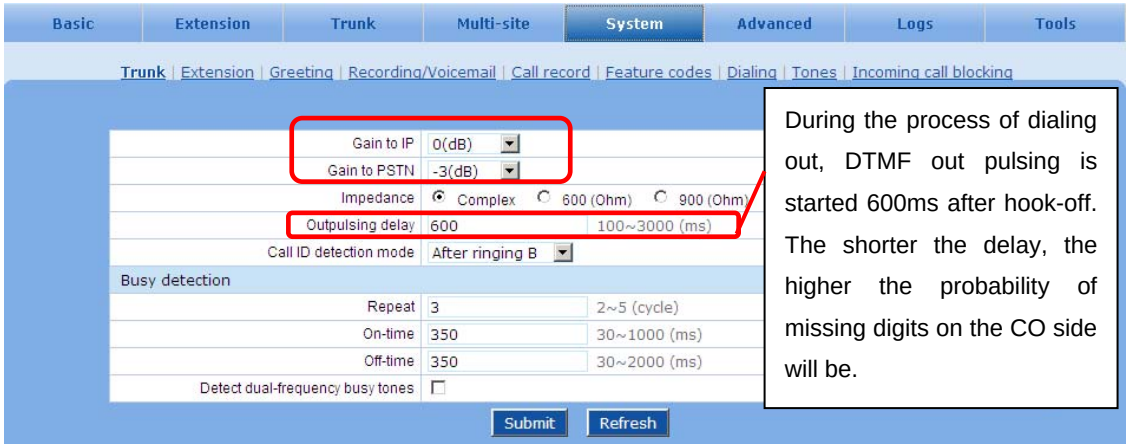
Users can listen and manage the voice files through the application running on the recording server.

6.5 Analog Trunk Port Settings

The parameters relate to the volume, the caller ID detection, and busy tone detection of a trunk. There is no need to make changes to the default values, unless there is an issue with one of the functions.

Go to the **System > Trunk** page to configure related parameters of trunk.

Figure 6-8 Trunk setting interface

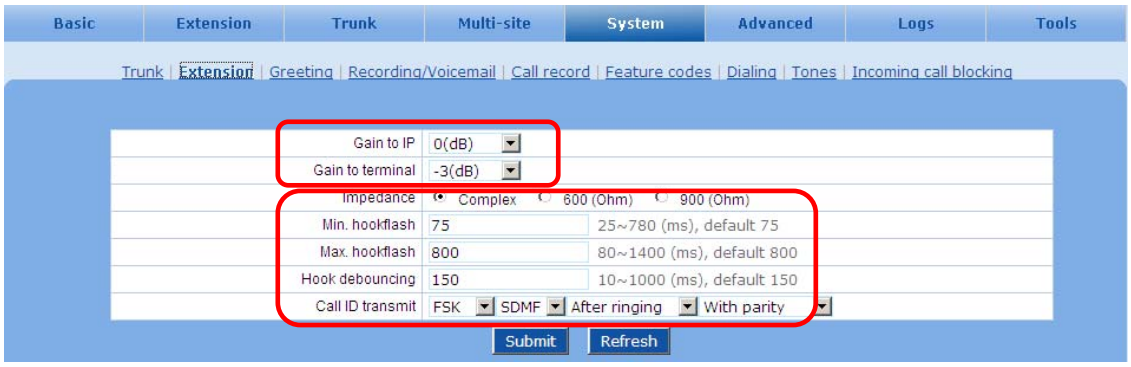


6.6 Analog Phone Port Settings

The parameters are provided to have better interconnection with analog phones from different vendors. There is no need to make changes to the default values, unless there are issues with voice volume, hook-flash and caller ID sent to the phone.

Go to the **System > Extension** page to configure related parameters of an analog phone port.

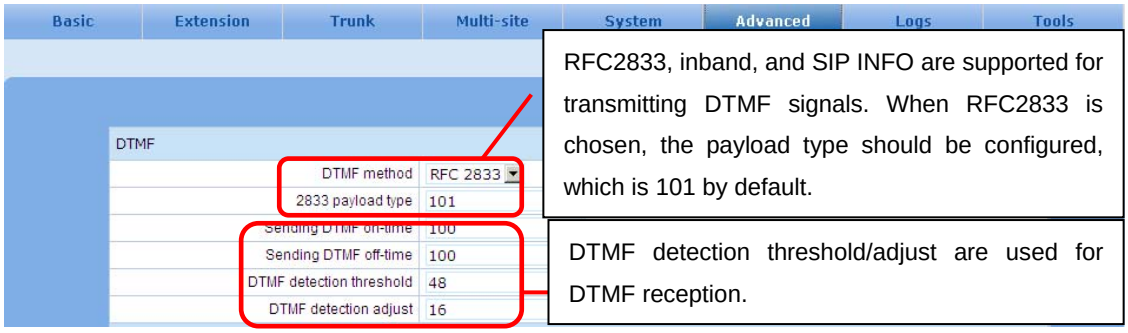
Figure 6-9 Analog phone port setting interface



6.7 DTMF

Go to the **Advanced > System** page to configure DTMF parameters.

Figure 6-10 DTMF setting interface



6.8 Voice Media Settings

Go to the **Advanced > Media** page to configure media related parameters.

Figure 6-11 Media setting interface

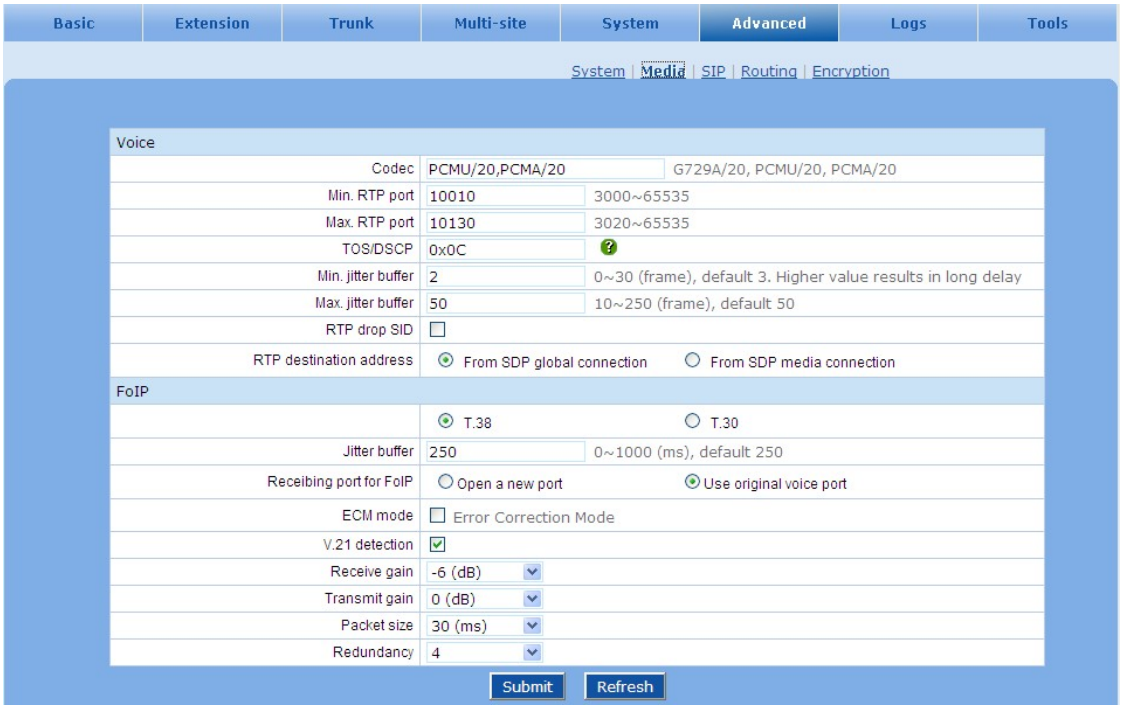


Table 6-1 Description of media parameters

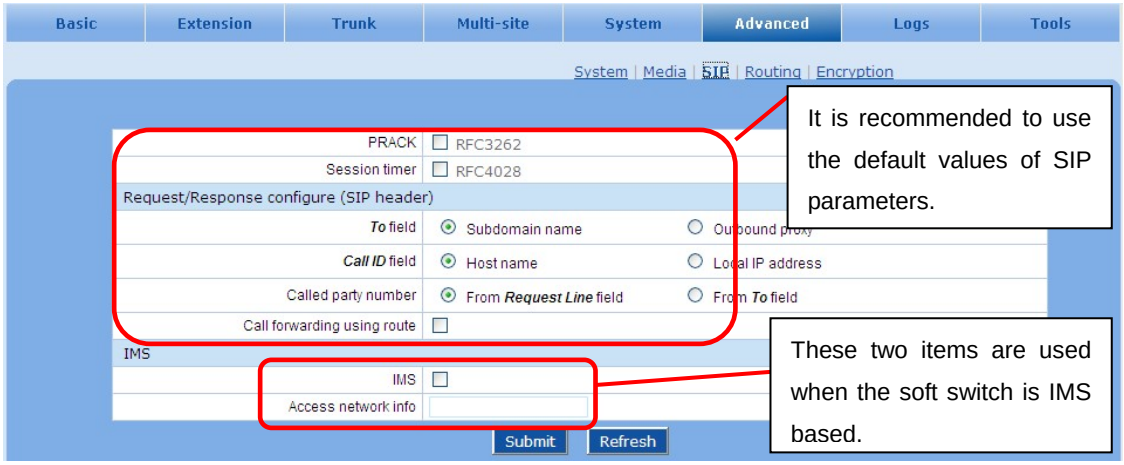
Item	Description
Codec	Valid values are G729A/20, PCMU/20 and PCMA/20, and multiple codecs can be entered, separating with a comma.
Min. RTP port	The lower bound of RTP transmission and receiving port. The valid range is 10010-10500.
Max. RTP port	The upper bound of RTP transmission and receiving port. This value must be greater than or equal to “2× number of lines＋min. RPT port”.
TOS/DSCP	This parameter specifies the priorities of media stream.

Item	Description
Min. jitter buffer	Used to reduce the impact of network jitter.
Max. jitter buffer	
RTP drop SID	If it is selected the received RTP SID voice packets will be discarded. By default, SID voice packets will not be dropped.
RTP destination address	This parameter specifies how to obtain the IP address of the receiving side for RTP packets. By default, the IP address is obtained “From SDP global connection”. • From SDP global connection: obtain the IP address from SDP global connection; • From SDP media connection: obtain the IP address from SDP media description.

6.9 SIP Related Advanced Settings

Go to the **Advanced** > **SIP** page to configure advanced SIP parameters.

Figure 6-12 SIP related advanced setting interface

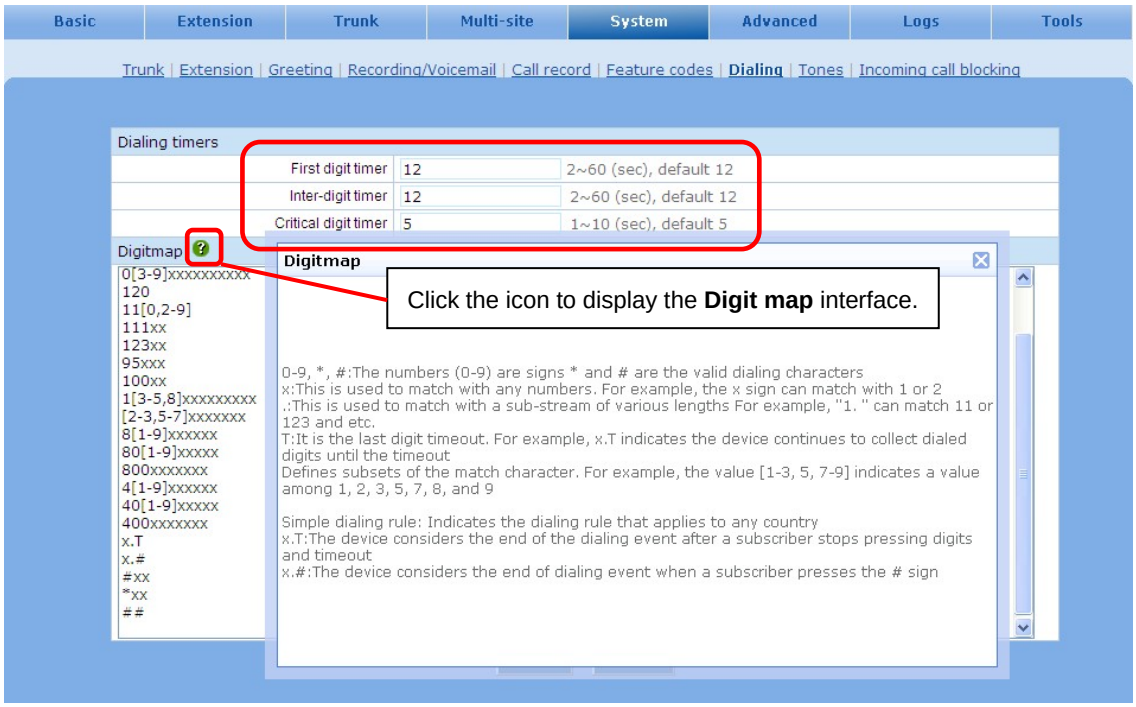


6.10 Dial Number Detection and Digit Map

A digit map consists of a set of digit strings, called rules, which describe the dialing plan of the device. During the process of collecting DTMF digits, the device matches the receiving digit string with the rules in the digit map. The receiving process is completed when a match situation is encountered. A well defined digit map helps to speed up the call setup.

Go to the **System** > **Dialing** page seen in Figure 6-13.

Figure 6-13 Dialing interface



A digit map in the device is composed with up to sixty rules each with up to 32 digits, and the total length of a digit map is limited to 2280 digits. The rules in the factory default digit map covers the dialing patterns of feature keys, such as *30, and others. The syntax of digit map is illustrated as below.

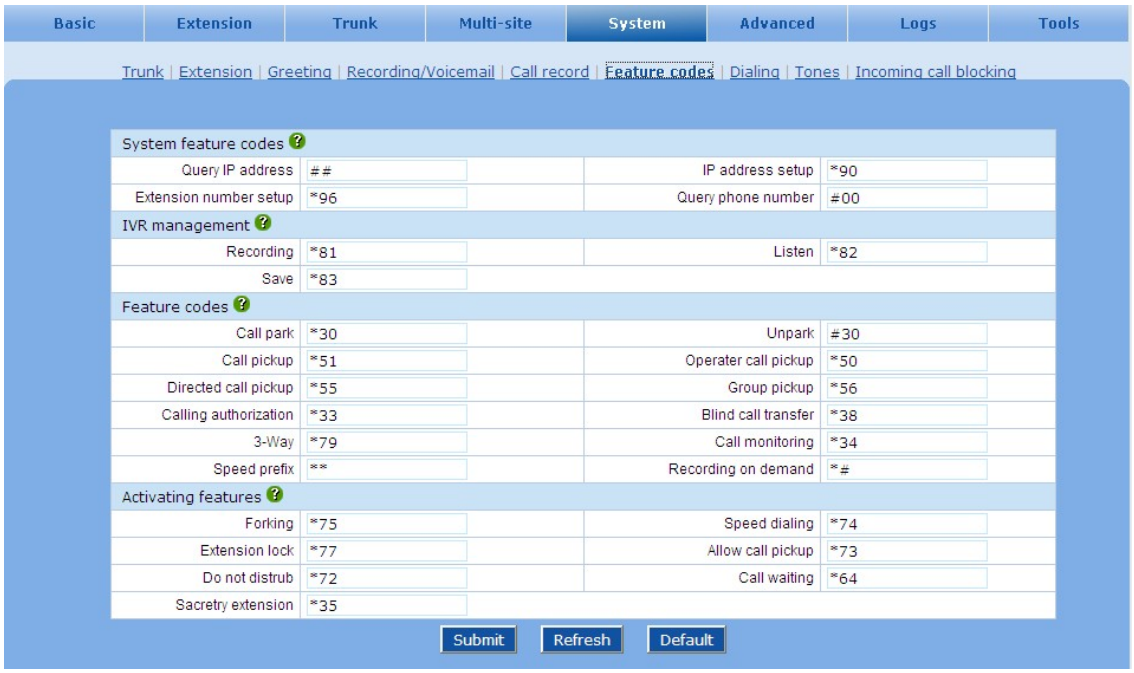
Table 6-2 Description of Digit map

Character	Description
0-9, *, #	Match to specific a DTMF signal.
x	Match to any 1-digit DTMF signal. E.g.: x can be matched to 1 or 2.
.	Match to any string of DTMF signals.
T	End of collecting DTMF digits after timeout of waiting for next digit
[ ]	Match to a set of DTMF digits. For example,[1-3，5，7-9] means the set of 1, 2, 3, 5, 7, 8 and 9.
x.T	Match to a string a DTMF string with any length, and the ending is triggered by the timeout of waiting for the next digit
x.#	Match to a DTMF string with any length ended with #
[2-8]xxxxxxx	Match to an 8-digit string starting with of any number between 2 and 8
02xxxxxxxxx	Match to an 11-digit string starting with 02
013xxxxxxxxx	Match to a 12-digit string starting with 013
13xxxxxxxxx	Match to an 11-digit string starting with 13
11x	Match to a 3-digit string starting with 11
9xxxx	Match to a 5-digit string starting with 9

6.11 Feature Codes

Go to the **System > Feature codes** page seen in Figure 6-14.

Figure 6-14 Feature codes interface



Note

Number conflicts must be avoided while making changes to feature codes.

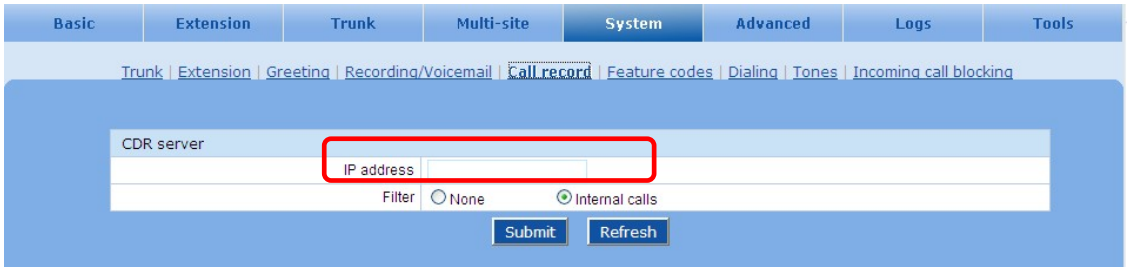
6.12 Call Detailed Record

The device is capable to output a detailed record for each call to an external storage server. The information of a detailed call record includes, among many, the calling party number, called party number, the starting and the ending timestamps of a call.

The detailed call records are output to a storage server after the completion of calls, and they can be read, searched, saved and deleted through off-line software.

Go to the **System > Call record** page seen in Figure 6-15.

Figure 6-15 CDR server setting interface

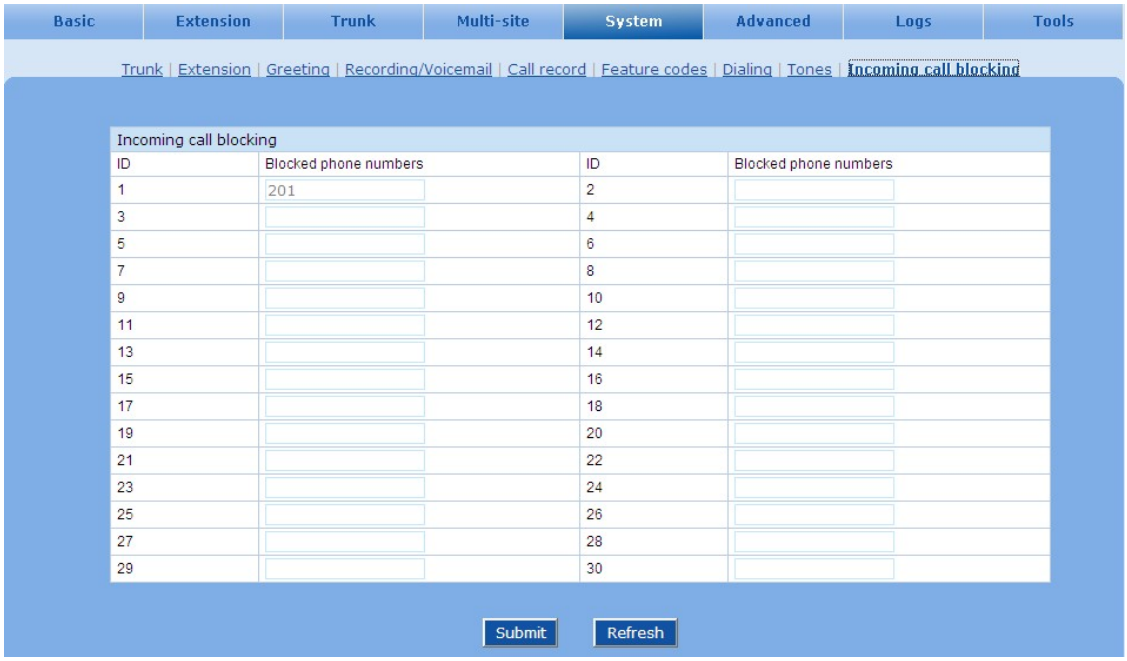


6.13 Selective Inbound Call Blocking

A black list of phone numbers can be set on the device to block un-wanted incoming calls.

Go to the **System > Incoming call blocking** page seen in Figure 6-16.

Figure 6-16 Incoming call blocking interface



Note

- Up to thirty numbers can be listed.
- The device provides the busy tone if the caller is on the blocking list.
- Caller ID detection feature should be enabled for this feature to take effect.

6.14 Outbound Call Screening

During the processing of an outbound call, the device screens the received dial number against two screening tables, one is white list and the other is black list.

White list

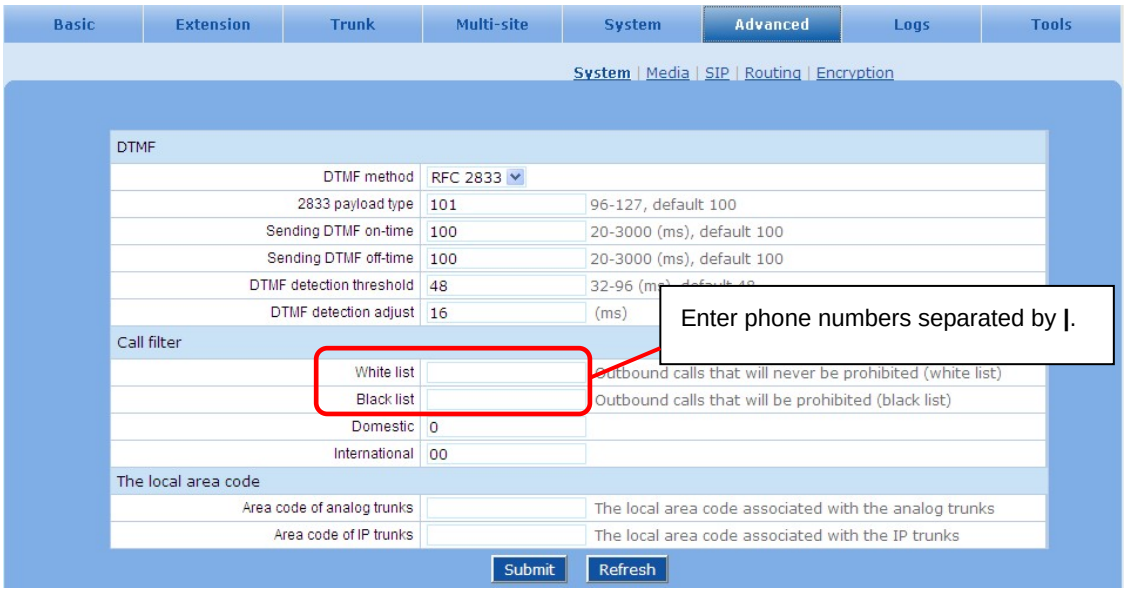
An outbound call with the dial number on the list is allowed to make regardless of the outbound restriction of the extension.

Black list

An outbound call with the dial number on the list is blocked.

Go to the **Advanced > System** page seen in Figure 6-17.

Figure 6-17 Interface of call screening



7 Maintenance

7.1 Firmware Upgrading

Step1 Select the **Tools** > **Upgrade** page, seen in Figure 7-1.

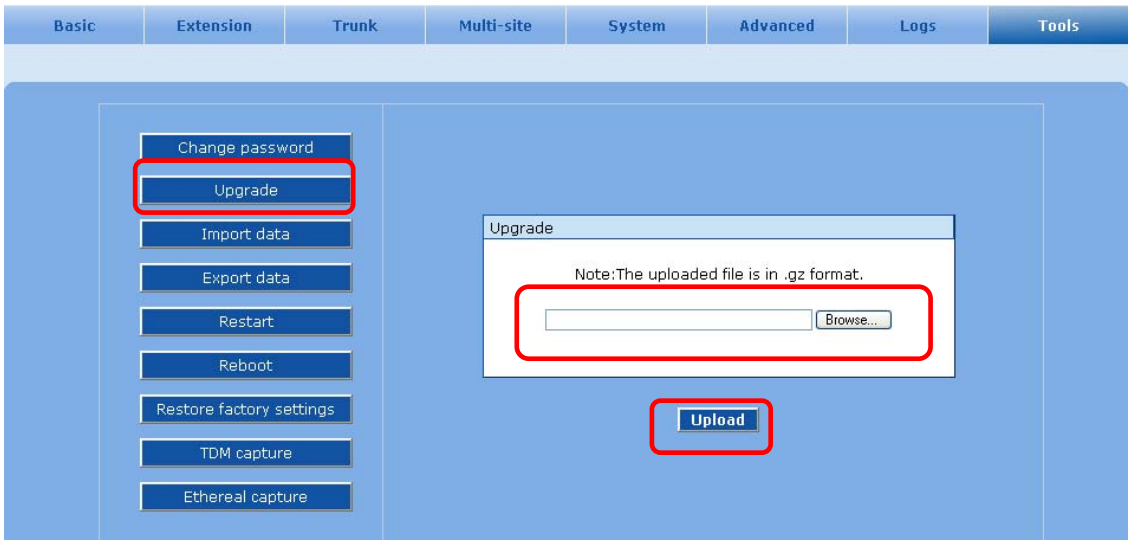
Step2 Click **Browse** to select the upgrade package.

Step3 Click **Upload** to upload the package to the system.

Note

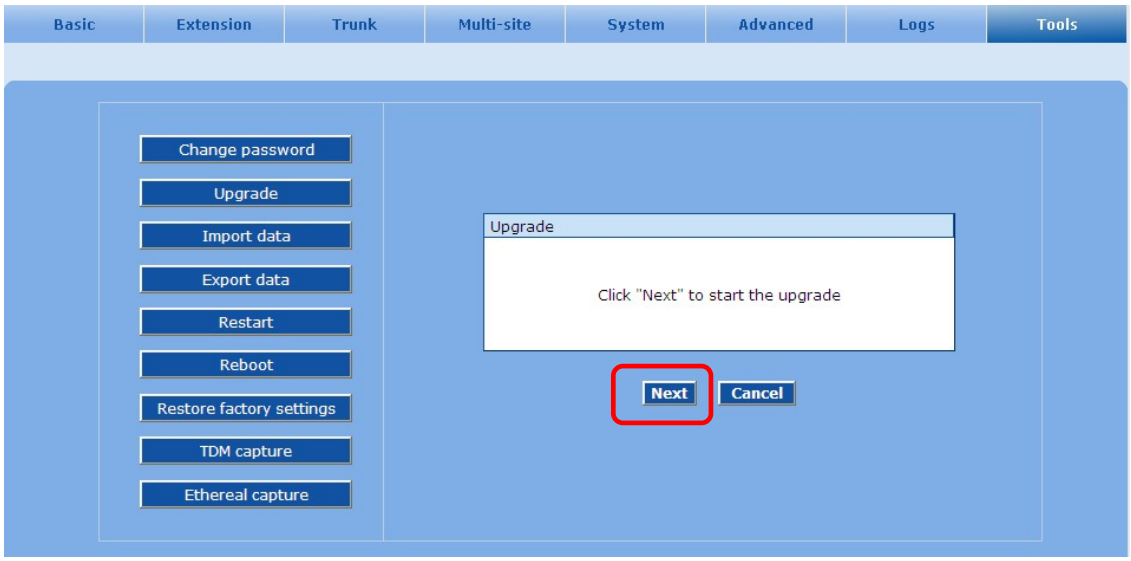
Please contact the supplier to obtain the latest firmware release.

Figure 7-1 Firmware upgrading interface



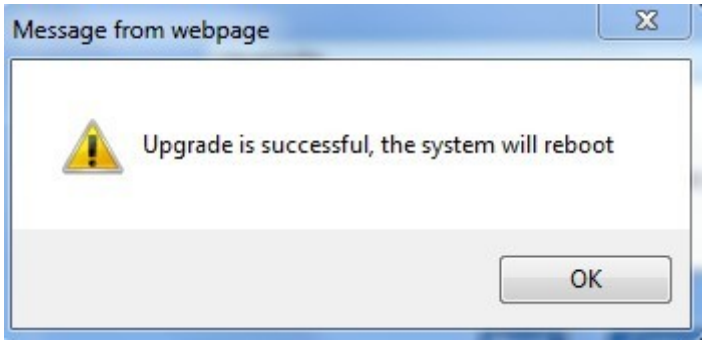
Step4 After uploading, click **Next** to upgrade the system.

Figure 7-2 Interface of upgrade firmware



Step5 When the prompt ***Upgrade is successful, the system will reboot*** is shown up, click **OK** to finish the process of upgrade.

Figure 7-3 Firmware upgrading interface



7.2 System Reboot

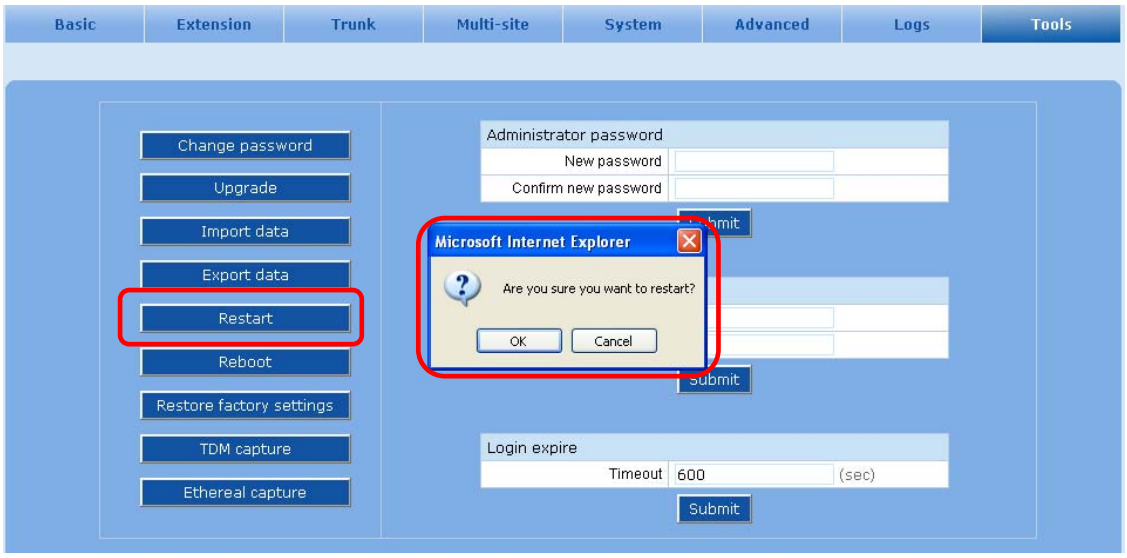
Step1 Go to the **Tools > Restart** page, seen in Figure 7-4.

Step2 Click **OK** to restart the system.

Note

It takes around two minutes to restart the system.

Figure 7-4 System restarting interface



7.3 Export Configuration Files

Step1 Go to the **Tools** > **Export data** page seen in Figure 7-5.

Step2 Click **Download** to download and save files to the destination you select.

Figure 7-5 Exporting configuration files interface

Note

By default, downloaded file will be named after **t1.tar.gz**, then **t2.tar.gz** and so on. Recounting begins after restart.

7.4 Import Configuration Files

Step1 Go to the **Tools > Import data** page seen in Figure 7-6.

Step2 Click **Browse** to select the packet, like t2.tar.gz.

Step3 Click **Upload** to import data, referring ***7.1 Firmware Upgrading***.

Figure 7-6 Importing configuration files interface

8 Appendix：Registering a SIP Terminal to OM

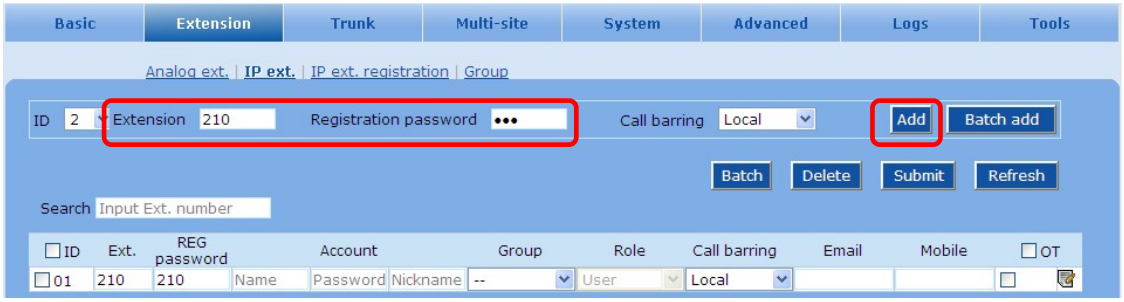
8.1 Softphone

CounterPath's SIP-based softphone X-Lite 3.0 is used to illustrate the configuration steps.

Step1 Create an account for the softphone

Go to the **Extension > IP ext.** page and enter the extension number and registration password.

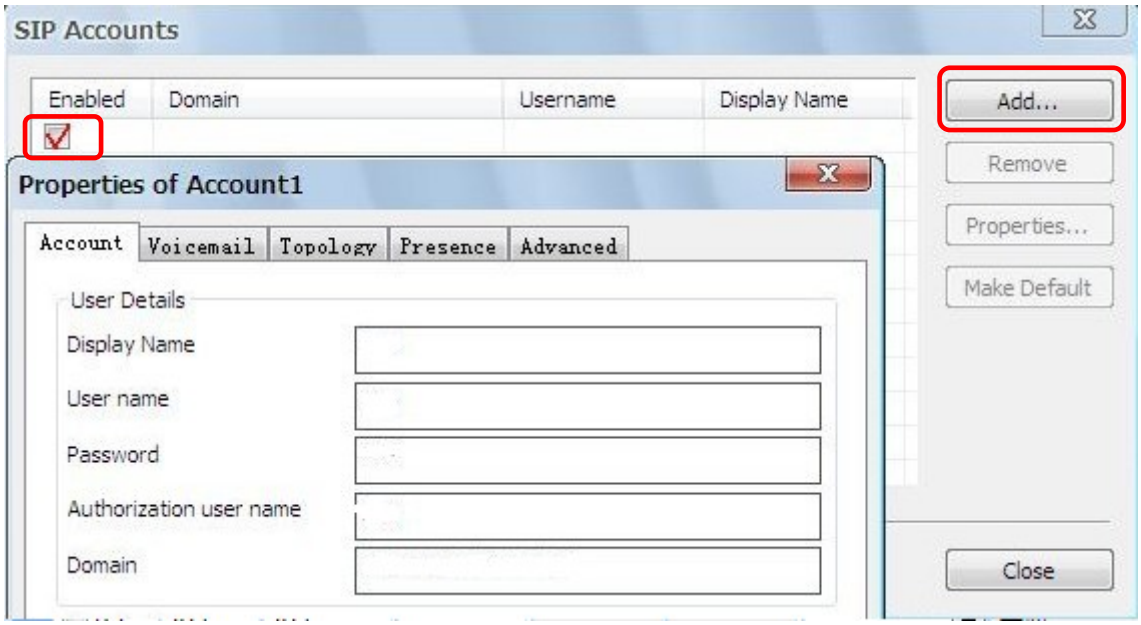
Figure 8-1 Account creating interface



Step2 Logon to the Web configuration page of X-Lite

Enter into the SIP configuration page. Click the button **Add…** popping up an interface of **Properties of Account**.

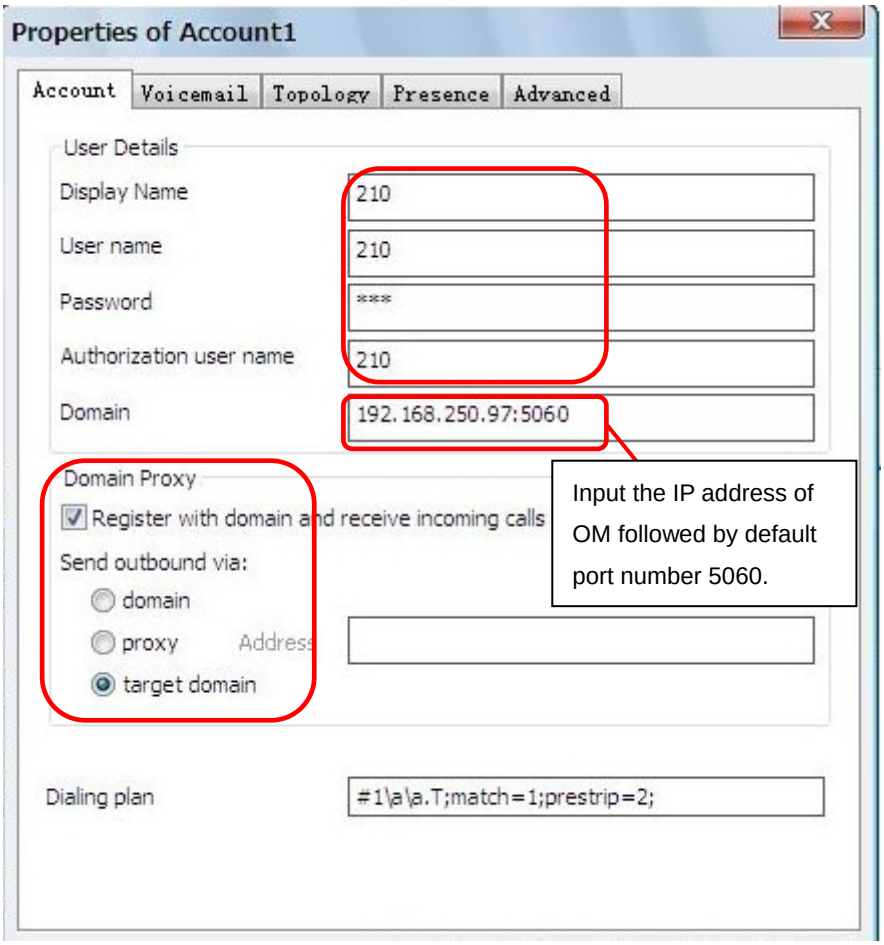
Figure 8-2 X-Lite login interface



Step3 Input registration information of the softphone

Click **Account** and fill out the fields shown in Fig.8-3.

Figure 8-3 X-Lite registration interface



8.2 SIP Phone

Take a SIP phone from Escene Communication as an example to illustrate the setting steps.

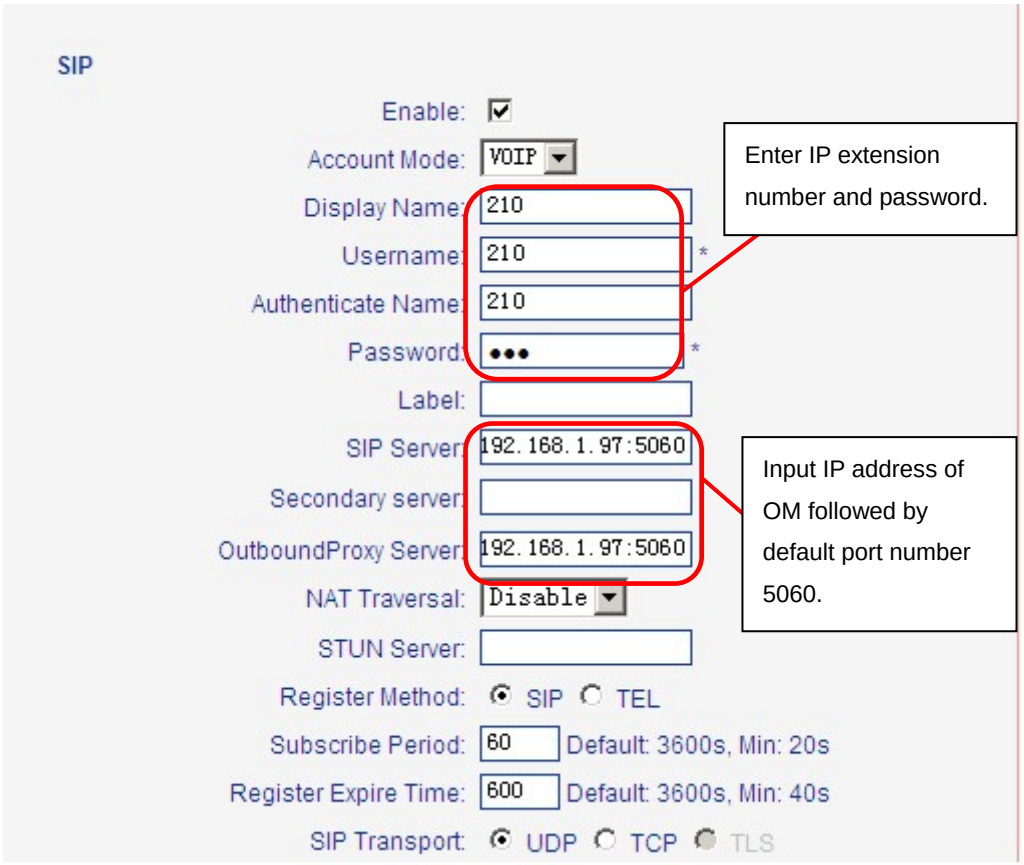
Step1 Assign extension number

Go to the **Extension > IP ext.** page to enter extension number and registration password. See Figure 8-1.

Step2 Configure SIP phone

Enter IP address of SIP phone to login the configuration interface.

Figure 8-4 SIP phone configuration interface



Note

- While configuring IP extension, please enable G.729 codec and select the same DTMF transmission mode written on interface of **Advanced > System**.
- After configuration, go to the **Basic > Status** page to check the status of IP extension.