



Elastix Server(Free) with MV-374

Voip GSM Gateway

This is the configuration setting up Elastix with PORTech gateways

1. IP Setting

Elastix Server

IP Address : 192.168.0.6

Incoming port : 5060

Firmware : 2.4.0

MV-374 GSM Gateway

IP address : 192.168.0.100

Dialpeer port : 5060

Mobile_1 sip port :5064

Mobile_2 sip port :5066

Mobile_3 sip port :5068

Mobile_4 sip port :5070

Firmware : V10.206

SJ-phone

Register the Soft phone to Elastix Server Extension: 6000

2. Application example

Outbound:

The example is to register SJ-phone to Elastix Server Extension: 6000.

After registration, please enter any destination number and dial out via Elastix server to MV-374.

Then please select any free GSM channel from MV-374 Dial Peer to dial out.

Inbound

User use cell phone or land line to dial out any GSM number of MV-374.

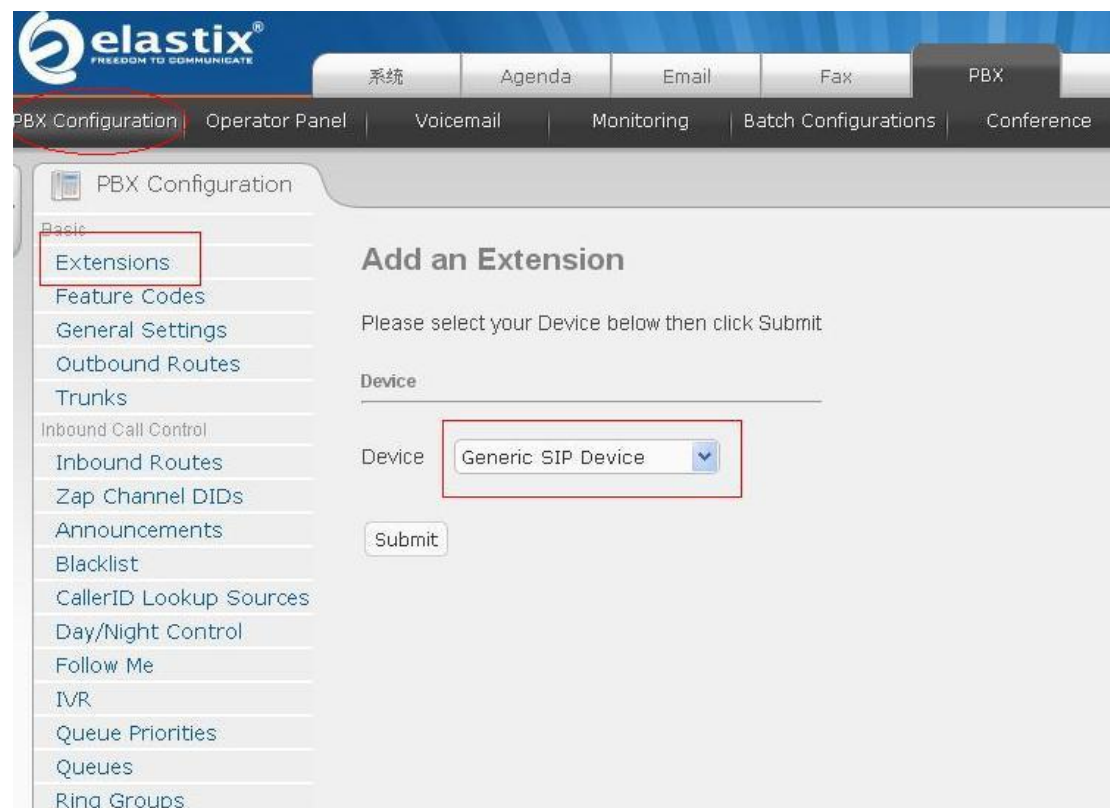
MV-374 route the call via Elastix server Extension 6000 (SJ-phone) and Extension 6000 (SJ-phone) will ring and show the cell phone number or caller ID of land line

3. Elastix Server setup

Step1. Add an Extension

E.g. We create 5 Extension on Elastix server: "6000" "8001" "8002" "8003" "8004", Please follow below diagram as Extension 8001

The main setting is showed in red circle; other settings are not necessary



The screenshot displays the Elastix web interface for PBX Configuration. The top navigation bar includes tabs for '系统', 'Agenda', 'Email', 'Fax', and 'PBX'. Below this, a secondary navigation bar contains 'PBX Configuration', 'Operator Panel', 'Voicemail', 'Monitoring', 'Batch Configurations', and 'Conference'. The main content area is titled 'Add an Extension' and includes a sidebar with a list of configuration options: 'Basic' (Extensions, Feature Codes, General Settings, Outbound Routes, Trunks), 'Inbound Call Control' (Inbound Routes, Zap Channel DIDs, Announcements, Blacklist, CallerID Lookup Sources, Day/Night Control, Follow Me, IVR, Queue Priorities, Queues, Ring Groups), and 'PBX Configuration'. The 'Add an Extension' form prompts the user to 'Please select your Device below then click Submit'. The 'Device' dropdown menu is set to 'Generic SIP Device' and is highlighted with a red circle. A 'Submit' button is located below the dropdown.

PBX Configuration

Basic

- Extensions
- Feature Codes
- General Settings
- Outbound Routes
- Trunks

Inbound Call Control

- Inbound Routes
- Zap Channel DIDs
- Announcements
- Blacklist
- CallerID Lookup Sources
- Day/Night Control
- Follow Me
- IVR
- Queue Priorities
- Queues
- Ring Groups
- Time Conditions
- Time Groups

Internal Options & Configuration

- Conferences
- Languages
- Misc Applications
- Misc Destinations
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- VoiceMail Blasting

Remote Access

- Callback
- DISA

Option

- Unembedded freePBX

Add SIP Extension

Add Extension

User Extension

Display Name

CID Num Alias

SIP Alias

Extension Options

Outbound CID

Ring Time

Call Waiting

Call Screening

Pinless Dialing

Emergency CID

Assigned DID/CID

DID Description

Add Inbound DID

Add Inbound CID

Device Options

This device uses sip technology.

secret

dtmfmode

The following Extension setting ("8002" "8003" "8004" and "6000") is same as above setup.

Step2. Add a trunk



The screenshot displays the PBX Configuration interface. On the left is a navigation menu with the following items: Basic, Extensions, Feature Codes, General Settings, Outbound Routes, Trunks (highlighted with a red box), Inbound Call Control, Inbound Routes, Zap Channel DIDs, Announcements, Blacklist, CallerID Lookup Sources, Day/Night Control, Follow Me, IVR, Queue Priorities, Queues, Ring Groups, Time Conditions, Time Groups, and Internal Options & Configuration. The main content area is titled "Add a Trunk" and contains a list of options, each with a green plus icon in a circle: Add SIP Trunk (highlighted with a red box), Add DAHDI Trunk, Add Zap Trunk (DAHDI compatibility mode), Add IAX2 Trunk, Add ENUM Trunk, Add DUNDI Trunk, and Add Custom Trunk.

- Basic
 - Extensions
 - Feature Codes
 - General Settings
 - Outbound Routes
 - Trunks
- Inbound Call Control
 - Inbound Routes
 - Zap Channel DIDs
 - Announcements
 - Blacklist
 - CallerID Lookup Sources
 - Day/Night Control
 - Follow Me
 - IVR
 - Queue Priorities
 - Queues
 - Ring Groups
 - Time Conditions
 - Time Groups
- Internal Options & Configuration
 - Conferences
 - Languages
 - Misc Applications
 - Misc Destinations
 - Music on Hold
 - PIN Sets
 - Paging and Intercom
 - Parking Lot
 - System Recordings
 - VoiceMail Blasting
- Remote Access
 - Callback
 - DISA

Edit SIP Trunk

Delete Trunk dialpeer

In use by 1 route

General Settings

Trunk Name:

Outbound Caller ID:

CID Options:

Maximum Channels: → MV374=4 , MV378=8

Disable Trunk: Disable

Monitor Trunk Failures: Enable

Dialed Number Manipulation Rules

(prepend) + prefix | match pattern

+ Add More Dial Pattern Fields

Clear all Fields

Dial Rules Wizards:

Outbound Dial Prefix:

Outgoing Settings

Trunk Name:

PEER Details:

```
host=192.168.0.100 → MV ip and dialpeer port
port=5060
type=peer
```

MV-374 is 4 ports GSM channels, and it require to be 4 GSM trunk (GSM 1, GSM 2, GSM3, GSM4)

The screenshot displays the 'PBX Configuration' interface for editing a SIP Trunk. On the left is a navigation menu with categories: Basic, Inbound Call Control, and Internal Options & Configuration. The main content area is titled 'Edit SIP Trunk' and includes a 'Delete Trunk GSM1' button. A warning message states: 'WARNING: This trunk is not used by any routes!'. Below this is the 'General Settings' section, which contains the following fields: 'Trunk Name' (set to 'GSM1'), 'Outbound Caller ID' (set to '0981086653'), 'CID Options' (set to 'Allow Any CID'), 'Maximum Channels' (set to '1'), 'Disable Trunk' (checkbox for 'Disable' is checked), and 'Monitor Trunk Failures' (checkbox for 'Enable' is unchecked). The 'Dial Number Manipulation Rules' section includes a rule editor with '(prepend) + prefix | match pattern' and buttons for '+ Add More Dial Pattern Fields' and 'Clear all Fields'. It also features a 'Dial Rules Wizards' dropdown menu (set to '(pick one)') and an 'Outbound Dial Prefix' input field.

Outgoing Settings

Trunk Name:

PEER Details:

```
host=192.168.0.100 — MV ip
port=5064 — mobile 1 sip port
type=peer
```

Incoming Settings

USER Context:

USER Details:

```
type=friend
secret=PT8001 > MV_mobile 1 sip setting
username=8001
qualify=yes
nat=yes
canreinvite=no
context=from-pstn
host=192.168.0.100 — MV ip
port=5064 — mobile 1 sip port
```

The following trunk setting (GSM2, GSM3 and GSM4) is same as above setup.

Step3. Outbound Routes

Basic

- Extensions
- Feature Codes
- General Settings
- Outbound Routes**
- Trunks

Inbound Call Control

- Inbound Routes
- Zap Channel DIDs
- Announcements
- Blacklist
- CallerID Lookup Sources
- Day/Night Control
- Follow Me
- IVR
- Queue Priorities
- Queues
- Ring Groups
- Time Conditions
- Time Groups

Internal Options & Configuration

- Conferences
- Languages
- Misc Applications
- Misc Destinations
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- VoiceMail Blasting

Remote Access

- Callback

Edit Route

⊖ Delete Route GSM

Route Settings

Route Name:

Route CID: Override Extension

Route Password:

Route Type: Emergency Intra-Company

Music On Hold?

Time Group:

Route Position:

Additional Settings

PIN Set:

Dial Patterns that will use this Route

+ Add More Dial Pattern Fields

Dial patterns wizards:

Trunk Sequence for Matched Routes

Step4. Inbound Routes

a. DID Number : 100

b. Set destination to Extension 6000

Outbound Routes

Trunks

Inbound Call Control

Inbound Routes

Zap Channel DIDs

Announcements

Blacklist

CallerID Lookup Sources

Day/Night Control

Follow Me

IVR

Queue Priorities

Queues

Ring Groups

Time Conditions

Time Groups

Internal Options & Configuration

Conferences

Languages

Misc Applications

Misc Destinations

Music on Hold

PIN Sets

Paging and Intercom

Parking Lot

System Recordings

VoiceMail Blasting

Remote Access

Callback

DISA

Option

Unembedded freePBX

← Edit Extension 6000 (6000)

Edit Incoming Route

Description:

DID Number:

Caller ID Number:

CID Priority Route:

Options

Alert Info:

CID name prefix:

Music On Hold:

Signal RINGING:

Pause Before Answer:

Privacy

Privacy Manager:

Language

Language:

CID Lookup Source

Source:

Fax Detect

Detect Faxes: No Yes

Set Destination

Extensions

4. MV-374 GSM Gateway Setup

Step1. Mobile to Lan Settings

Mobile 1, 2

Page: 1

Item	CID	URL	Select
0	*	100	<input type="checkbox"/>
1		Inbound Routes DID Number	<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Step2. Lan to Mobile Route

Mobile 1, 2

Page: 1

Item	URL	Call Num	Select
0	*	#	<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Delete Selected Delete All Reset

Step3. On mobile 1, 2, 3, 4, please edit the SIP From to "Tel/Tel(Not Std)

The screenshot shows a configuration page for a network device. On the left is a navigation menu with the following items: Dial Peer, Route, Mobile, Status, Settings, Fwd Settings, SMS Agent, SIM Setting, Operator Setting, Cell Info, USSD, Network, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The 'Mobile' section is selected, and a dropdown menu at the top shows 'Mobile 1.2' selected. The main content area is divided into two sections: 'Mobile 1' and 'Mobile 2'. Both sections have a status indicator 'ON' selected. The 'SIP From' field in both sections is highlighted with a red box and set to 'Tel/Tel (Not Std)'. Other fields include VoIP Tx Gain, VoIP Rx Gain, LAN Dialtone Vol, CODEC Tx Gain, CODEC Rx Gain, Answer delay, CLID Presentation, Restart dial fails, Mobile PIN Code, Dial Prefix, LAN Answer Mode, and Init AT Cmd.

Mobile 1	Mobile 2
VoIP Tx Gain: 9 (0-12)	VoIP Tx Gain: 9 (0-12)
VoIP Rx Gain: 11 (0-15)	VoIP Rx Gain: 11 (0-15)
LAN Dialtone Vol: 9 (0-12)	LAN Dialtone Vol: 9 (0-12)
Routing Range: 0 ~ 49 (0-49)	Routing Range: 0 ~ 49 (0-49)
CODEC Tx Gain: 6 (0-7)	CODEC Tx Gain: 6 (0-7)
CODEC Rx Gain: 6 (0-7)	CODEC Rx Gain: 6 (0-7)
SIP From: Tel/Tel (Not Std)	SIP From: Tel/Tel (Not Std)
Answer delay: 0 (0-15)	Answer delay: 0 (0-15)
CLID Presentation: OFF	CLID Presentation: OFF
Restart dial fails: 1 (0-15)	Restart dial fails: 1 (0-15)
Mobile PIN Code: On	Mobile PIN Code: On
Dial Prefix:	Dial Prefix:
LAN Answer Mode: Answered	LAN Answer Mode: Answered
Init AT Cmd:	Init AT Cmd:

Mobile 3, 4 ▾

VoIP Tx Gain:	<input type="text" value="9"/> (0~12)	VoIP Rx Gain:	<input type="text" value="11"/> (0~15)
LAN Dialtone Vol:	<input type="text" value="9"/> (0~12)		

Mobile 3 ON OFF

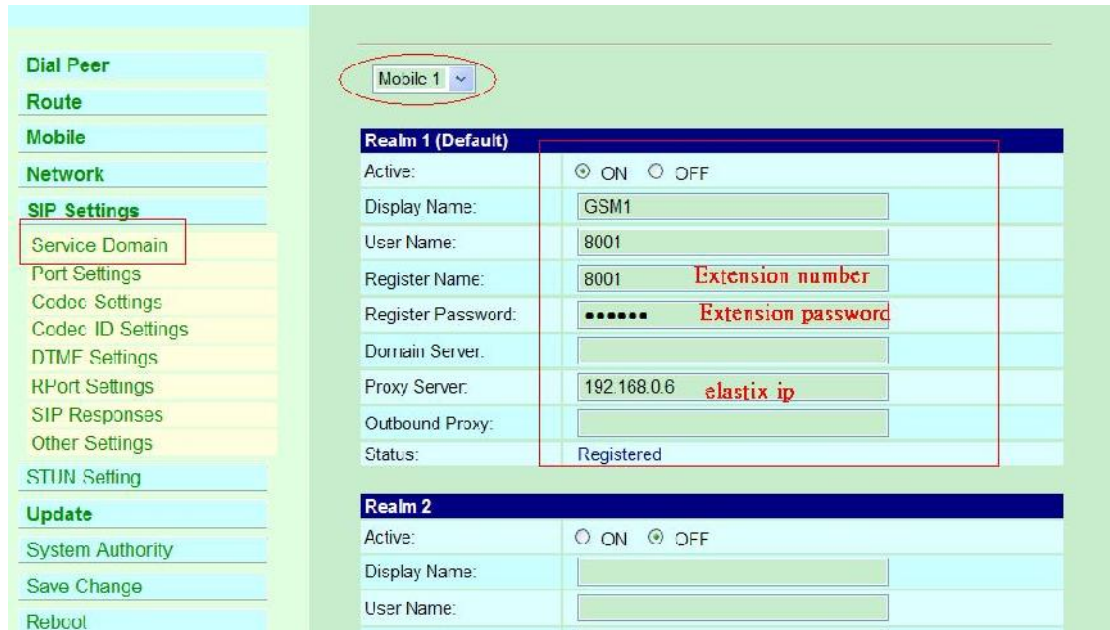
Routing Range	<input type="text" value="0"/> ~ <input type="text" value="49"/> (0~49)		
CODEC Tx Gain:	<input type="text" value="6"/> (0~7)	CODEC Rx Gain:	<input type="text" value="6"/> (0~7)
SIP From:	<input type="text" value="Tel/Tel (Not Std)"/> ▾	Answer delay	<input type="text" value="0"/> (0~15)
CLID Presentation	<input type="radio"/> OFF <input checked="" type="radio"/> ON	Restart dial fails	<input type="text" value="1"/> (0~15)
Mobile PIN Code:	On <input type="checkbox"/> Code: <input type="text"/>	Confirmed:	<input type="text"/>
Dial Prefix	<input type="text"/>	LAN Answer Mode	<input type="text" value="Answered"/> ▾
Init AT Cmd	<input type="text"/>		

Mobile 4 ON OFF

Routing Range	<input type="text" value="0"/> ~ <input type="text" value="49"/> (0~49)		
CODEC Tx Gain:	<input type="text" value="6"/> (0~7)	CODEC Rx Gain:	<input type="text" value="6"/> (0~7)
SIP From:	<input type="text" value="Tel/Tel (Not Std)"/> ▾	Answer delay	<input type="text" value="0"/> (0~15)
CLID Presentation	<input type="radio"/> OFF <input checked="" type="radio"/> ON	Restart dial fails	<input type="text" value="1"/> (0~15)
Mobile PIN Code:	On <input type="checkbox"/> Code: <input type="text"/>	Confirmed:	<input type="text"/>
Dial Prefix	<input type="text"/>	LAN Answer Mode	<input type="text" value="Answered"/> ▾
Init AT Cmd	<input type="text"/>		

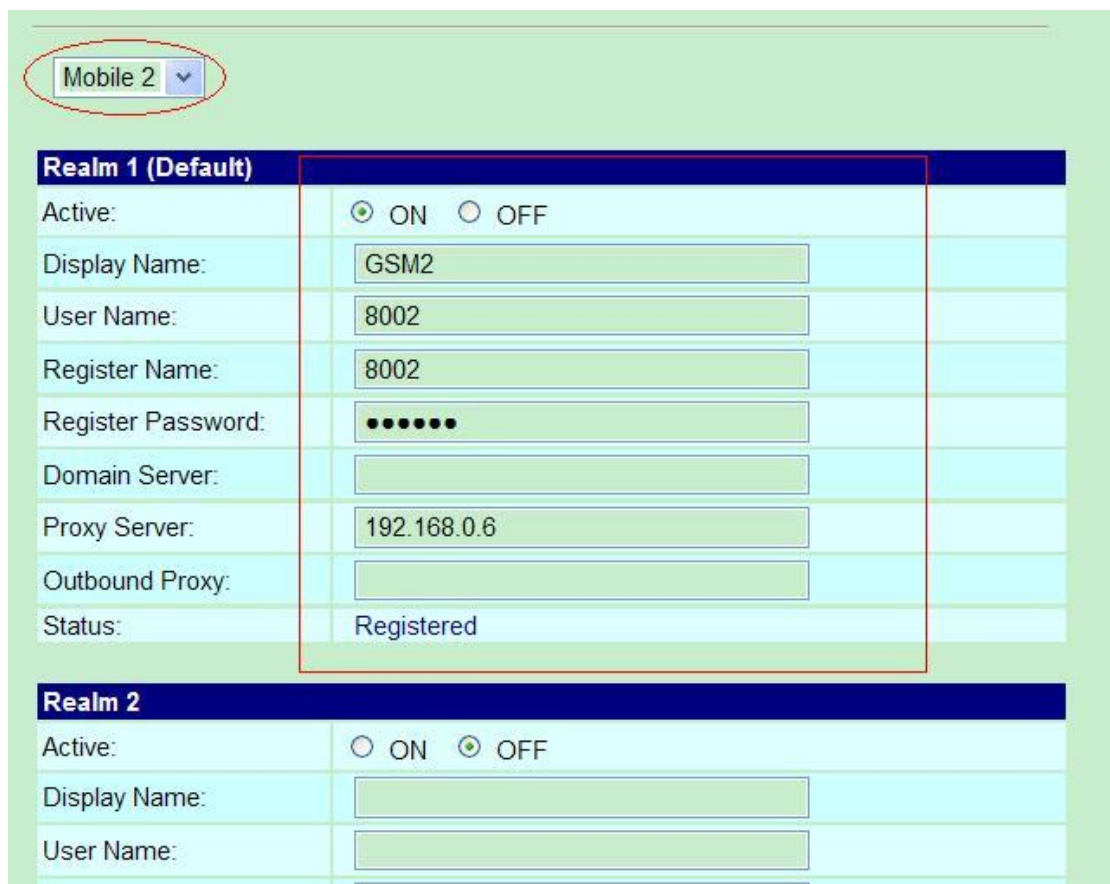
Step 4 . SIP Settings

On service domain, please register the Elastix server Extension to “8001” “8002” “8003” “8004”



The screenshot shows the Asterisk SIP Settings interface for 'Mobile 1'. The left sidebar contains a menu with 'SIP Settings' highlighted. The main content area shows the configuration for 'Realm 1 (Default)'. The 'Active' checkbox is checked. The 'Display Name' is 'GSM1'. The 'User Name' is '8001'. The 'Register Name' is '8001' with a red annotation 'Extension number'. The 'Register Password' is masked with dots and has a red annotation 'Extension password'. The 'Proxy Server' is '192.168.0.6' with a red annotation 'elastix ip'. The 'Status' is 'Registered'. Below this, 'Realm 2' is shown with 'Active' unchecked.

Mobile 1	
Realm 1 (Default)	
Active:	<input checked="" type="radio"/> ON <input type="radio"/> OFF
Display Name:	GSM1
User Name:	8001
Register Name:	8001 <i>Extension number</i>
Register Password: <i>Extension password</i>
Domain Server:	
Proxy Server:	192.168.0.6 <i>elastix ip</i>
Outbound Proxy:	
Status:	Registered
Realm 2	
Active:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Display Name:	
User Name:	



The screenshot shows the Asterisk SIP Settings interface for 'Mobile 2'. The left sidebar contains a menu with 'SIP Settings' highlighted. The main content area shows the configuration for 'Realm 1 (Default)'. The 'Active' checkbox is checked. The 'Display Name' is 'GSM2'. The 'User Name' is '8002'. The 'Register Name' is '8002'. The 'Register Password' is masked with dots. The 'Proxy Server' is '192.168.0.6'. The 'Status' is 'Registered'. Below this, 'Realm 2' is shown with 'Active' unchecked.

Mobile 2	
Realm 1 (Default)	
Active:	<input checked="" type="radio"/> ON <input type="radio"/> OFF
Display Name:	GSM2
User Name:	8002
Register Name:	8002
Register Password:
Domain Server:	
Proxy Server:	192.168.0.6
Outbound Proxy:	
Status:	Registered
Realm 2	
Active:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Display Name:	
User Name:	

Mobile 3 ▾

Realm 1 (Default)

Active:	<input checked="" type="radio"/> ON <input type="radio"/> OFF
Display Name:	<input type="text" value="GSM3"/>
User Name:	<input type="text" value="8003"/>
Register Name:	<input type="text" value="8003"/>
Register Password:	<input type="password" value="••••••"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text" value="192.168.0.6"/>
Outbound Proxy:	<input type="text"/>
Status:	Registered

Realm 2

Active:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Display Name:	<input type="text"/>
User Name:	<input type="text"/>

Mobile 4 ▾

Realm 1 (Default)	
Active:	<input checked="" type="radio"/> ON <input type="radio"/> OFF
Display Name:	<input type="text" value="GSM4"/>
User Name:	<input type="text" value="8004"/>
Register Name:	<input type="text" value="8004"/>
Register Password:	<input type="password" value="••••••"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text" value="192.168.0.6"/>
Outbound Proxy:	<input type="text"/>
Status:	Registered

Realm 2	
Active:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Display Name:	<input type="text"/>
User Name:	<input type="text"/>
Register Name:	<input type="text"/>
Register Password:	<input type="password"/>