Intelligent Mobile Extension Cloud Server

Talk Server PRO WP1264
Base Setting Manual

TALK SERVER Pro
TELECOM & DATACOM

嘉騰科技股份有限公司
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http://www.jia-teng.com

Federal Communication Commission Interference Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures: • Reorient or relocate the receiving antenna. • Increase the separation between the equipment and receiver. • Connect the equipment into an outlet on a circuit different from that to which the receiver is connected. • Consult the dealer or an experienced radio/TV technician for help.

FCC Caution: This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Non-modification Statement: Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

FCC Radiation Exposure Statement:

This equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment. This equipment should be installed and operated with minimum distance 20cm between the radiator & your body.

Limited Channels fixed for use in the US:

IEEE 802.11b or 802.11g or 802.11n(HT20) operation of this product in the U.S. is firmware-limited to Channel 1 through 11. IEEE 802.11n(HT40) operation of this product in the U.S. is firmware-limited to Channel 3 through 9.

Hardware Specification：

Dimension：360X120X44mm

Antenna：Two detachable 7dBi antenna

Frequency Band：2.400GHz~2.484GHz

Input voltage：AC 100-240V 50Hz/60Hz 0.3A

Output voltage：DC 12V 1A

Operation environment：0°C ~ 60°C

Storage Humidity：10% ~ 90%

Interface：RJ-45 Port X1(WAN)、RJ-45 Port x4(LAN)、Reset/WPS button、USB Port X1、AC X1

Wireless Specification：

Wireless data transmission rate：

802.11b/802.11g/802.11n(40MHz) Maximum Transfer Rate 300Mbps

Wireless security support：64/128 bit WEP、WPA(TKIP with IEEE 802.1x)、WPA2(AES with IEEE 802.1x)、WPAPSK/WPA2PSK

Router Specification:

Router Specification：WAN Mode：Static、DHCP、PPPoE

Internet Sharing：DHCP Server、NAT、Port Mapping

Firewall：IP/Port filtering、Block Port Scan、Block SYN Flood

clock synchronization：NTP Client

QoS：Voice first

IP PBX Specification:

A. PBX Features

a. IVR

- Call busy voice prompt
- No answer voice prompt
- Call transfer voice prompt
- Voice prompt for voice mail operation
- Voice prompt for call transfer
- Voice prompt for system setup

b. Auto Attendant

- 6 sets of Auto Attendant configuration
- Using first DTMF for AA procedure identification
- Auto Attendant procedure tree support
- Voice prompt of different Auto Attendant procedures could be recorded separately
- Remote voice mail
- Call transfer to extension number
- Configuration of dialing plan for incoming call being transferred to external number
- On Duty/Break Duty/Off Duty modes

c. Dialing plan

- 64 sets of configuration of call dialing plan
- 64 sets of configuration of Trunk dialing plan

d. Call Features

- Support G.711A, G.711μ-law, G.723, G.729 codec
- D.I.L. (Direct In Line)
- Remote station (NAT Transversal)
- Call transfer to extension or SIP Trunk number
- Call forward/Busy forward to extension or SIP Trunk number
- Direct record AA content through IP Phone
- Personal answering information function support
- Voice mail setup through IP Phone
- Phone command
- Call holding
- Call waiting
- Three ways conference (Phone support is necessary)
- Support blind transfer
- Support DTMF transfer
- Broadcast(bundle with CM5000)
- H.264/MPEG4/H263(passthrough)

B. SIP Server

a. SIP Proxy Server

- Build in SIP Proxy Registration Server
- Support SIP v2(RFC3261, 3262, 3263, 3264)
- Compliant with IETF SIP standards
- Support up to 64 Subscribers
- Support up to 20 concurrent calls
- NAT support

b. SIP Trunk

- Support up to 12 SIP Trunk registrations
- Support Trunk dialing plan
- DTMF support RFC2833/SIP information/inband

C. Management

- Web configuration
- HTTP firmware upgrade
- User/admin two tiers password protection
- Configuration file backup and recovery
- SNMP-Client

D. Network Features

- DHCP server and client support
- NAT Setting
- PPPoE support
- Wireless
- QOS
- Firewall

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Configuration

About Jia Teng

Since its establishment in 2006, JiaTeng has adhered to the generous experience of developing software products and the ability to integrate manufacturing process vertically. We are devoted to offering customers everything from innovative designs to systematized manufacturing services. The business scope contains management information and billing systems for telecommunication operators, mobile phones, wireless communication products, and networking products.

It is JiaTeng's vision to become the strategic partner of the telecommunication operator. Under its great leadership and staff efforts, JiaTeng has already become one of the important strategic partners within Asian-Pacific telecommunication operations. Besides providing management information systems for telecommunication operators, JiaTeng also, in accordance with the demand, supplies Feature Phone and Smart Phone in creating a great harvest for future networking business.

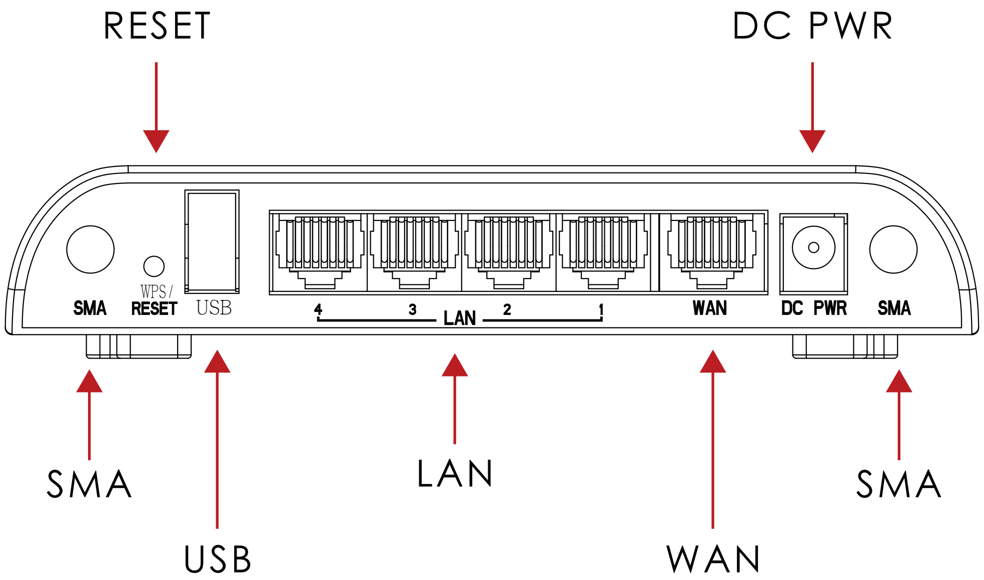
Login

1. Please install WIFI Antenna first, and then connect with the power
2. Plug ADSL Modem into WAN
3. Connect your RJ 45 cable into LAN

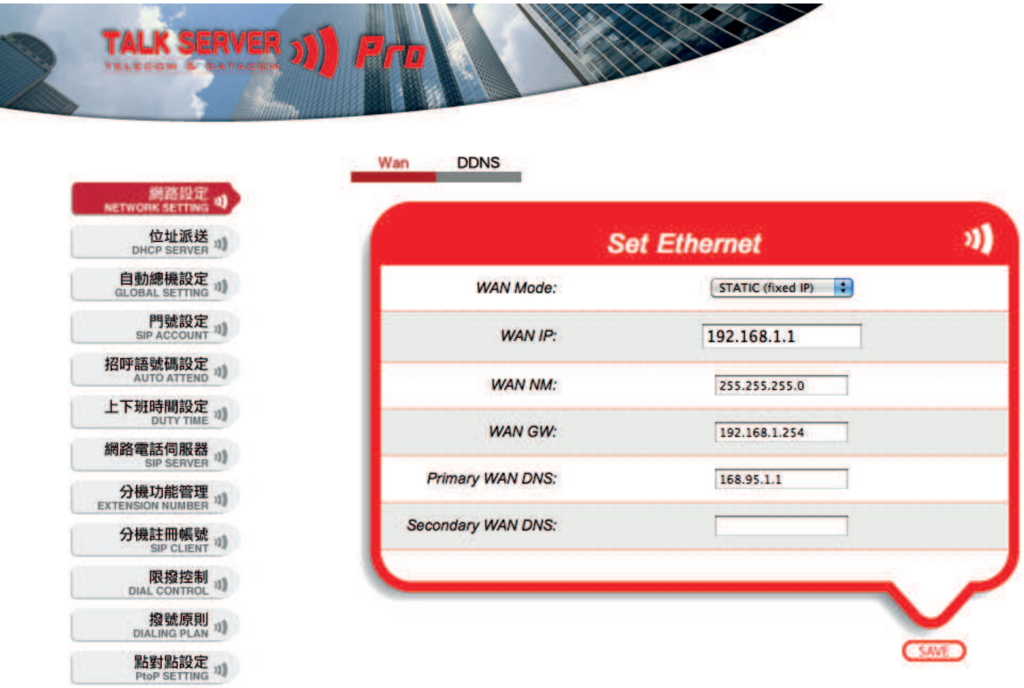
Start your browser to input http://192.168.100.1, then press “ENTER”.

Input Account: admin

Password: 123456



NETWORK SETTING



WAN Mode:

There are 3 modes of WAN Mode: STATIC/DHCP/PPPoE

Please select your suitable mode according to ADSL data

WIN DNS:

Please input the IP address of DNS according to ADSL data.

If user has applied the SIP account of Telecom, please input the assigned IP address of Telecom.

Finish all setting items, press **SAVE** putton.

DHCP SERVER

DHCP Server Switch:

Enable & disable for DHCP Server Switch. Default is "Enable".

Please so not change to be "Disable", otherwise user cannot login the system.

Lan IP: Default for 192.168.100.1

Netmask: Default for 255.255.255.0

Please do not change or amend the setting, otherwise user cannot login the system.

Finish all setting items, press **SAVE** putton.

GLOBAL SETTING

Web Login Username:

Web login Password:

Web login Password (check): Input the Password again

Number Length:

There are 2 available choices (3 digits/4 digits) for Number Length of Extension.

IP PBX password: Set the Password of IP PBX

EXT '9' function: **EXT '0' function:**

FUN_NULL: No Function

SIP TRUNK: register the SIP Account to call out

CO_GW: connect the PSTN interface of VoIP Gateway to call out

Duty Auto Attend:

There are 2 available choices Enable/ Disable for duty setting

Default is Disable and adopt the greeting of On Duty AA.

Change to be Enable will add the selection of Break Duty AA & Off Duty AA.

according to the assigned time of duty time, it will automatically to select Greet of Auto Attend to operate the setting.

Build-in: 3 items for greeting

400 On Duty / 401 Break Duty / 402 Off Duty

There is another item without playing greeting, code: No IVR, ring directly to operator.

To recode or amend the new message must use phone set. Here are the operations: Pick up the phone set and press *50#, then hear the reminding message: Input the password (default:1234)

after input 1234, hear the reminding message: input the operator number

User can add one new Greeting or amend the default one (ex 400)

after input, hear the reminding message: record the Greeting when finished press #

GLOBAL SETTING

E-mail notification:

There are 4 items for setting to notice voice mail by E-mail

E-Mail:

E-Mail User Name:

E-mail Password:

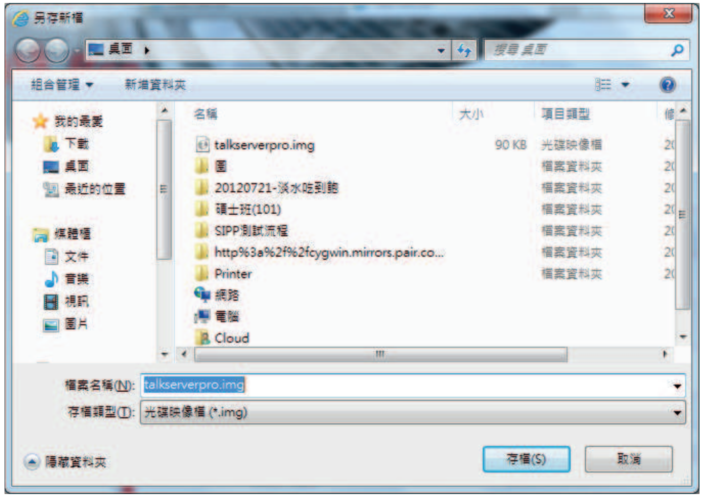
SMTP Server: The IP address of Mail server.

NTP Setting: Time Server setting

Back up: Save the file of new setting

Select **Make Backup File**, then appear as the following picture Make Backup File

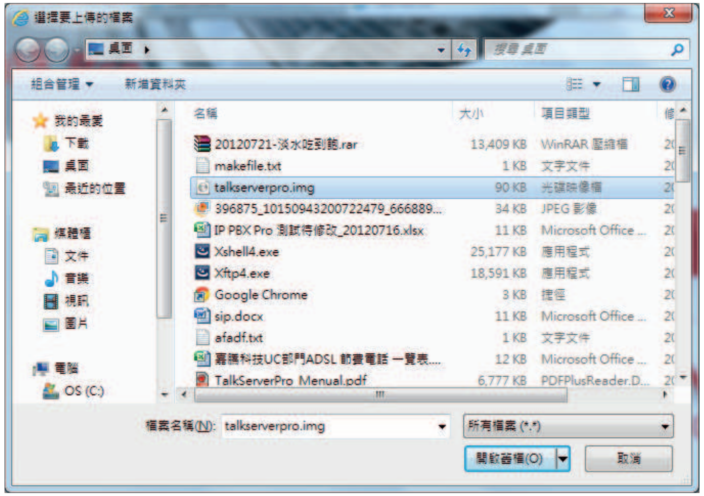
Press **Download** to download the latest setting file, then appear as the following picture.



Finish all, press **Save** putton

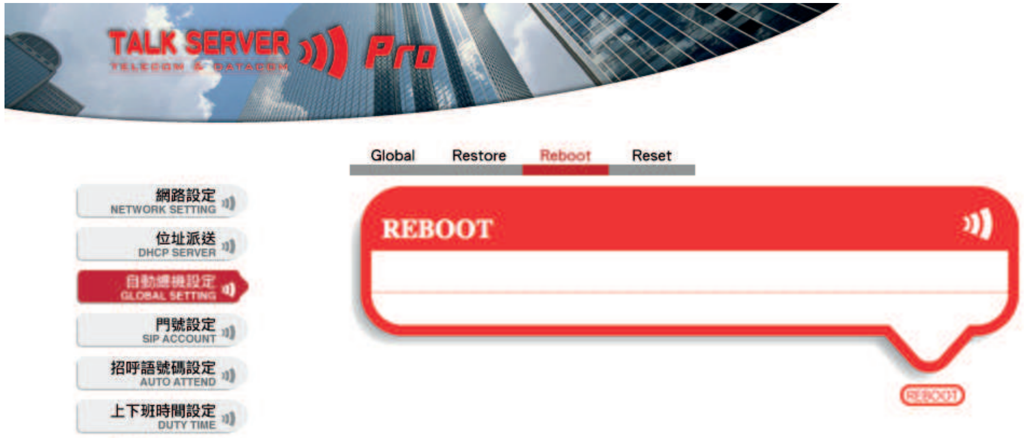
Restore: re-save the setting file of system

Select the restore file, then the system will appear as the following picture



Finish all and press **Upload** putton to send the setting file
※ After Upload the setting file, please reboot manually to make the setting work.

Reboot: restart device



Reset: back to the default
※ This setting will clear all current setting and back to default

SIP ACCOUNT

This page can afford checking list of SIP account (Trunk).
Uni Talk: 2 SIP account available
Uni Talk plus: 4 SIP account available.
Talk server Pro: 12 SIP account available.

Number column:
Register column: appear the SIP account or Telecom Number
Mode column: appear the register mode
Group column:
Select column:

DELETE putton:

Answer Mode: 3 mode (ON / OFF / DID)

ON	Start up the line
OFF	Close the line
DID	Ring directly to the assigned extension. It will add **To num:** for assigned extension.

Duty Auto Attend :
2 selections (Enable / Disable). Default is Disable.
※ refer to the Duty Auto Attendant of **GLOBAL SETTING**

On Duty AA: default is none.
※ refer to the Duty Auto Attendant of **GLOBAL SETTING**

※ Each SIP account can set the Auto Attendant and Greeting individually. Normal usages do not start any setting, otherwise it will imitate the Duty Auto Attendant of **GLOBAL SETTING**

Register ID:
Register Username:
Register Password:
Authorise Name:
Proxy Address:
Server IP:
Expire Time:
DTMF Type:

Finish all setting items, press **SAVE** putton.

AUTO ATTEND

Build-in: 3 items for greeting
400 On Duty / 401 Break Duty/ 402 Off Duty
There is another item without playing greeting, code: None IVR, ring directly to operator.

None IVR: Without playing greeting
400: On Duty
401: Break Duty
402: Off Duty

ADD To add new code

DELETED delete code

Select AUTO ATTEND or press ADD to enter web page.

Edit Auto Attend

Auto Attend Number: Set or amend greeting of Auto Attendant

Operator Ring Mode: 2 selections (Sequential / Round Robin)
Seguential Each ringing start from Operator(1),
jump to the next one if Operator(1) is busy
Round Robin The first ringing for Operator(1),
the second ringing for Operator(2), the third ringing for Operator(3),
the fourth ringing for Operator(1).
Jump to the next if Operator (1) is busy.

Operator(1): point out will show all extensions
Operator(2): point out will show all extensions
Operator(3): point out will show all extensions
Operator(4): point out will show all extensions
Operator(5): point out will show all extensions

There are 4 items for incoming call, the process after press phone set
'0' Function: '9' Function: '*' Function:
Another four available selections:
None / To Operator/ To Voice Mail/ To Auto Attendant

None	Close
To Operator	directly to operator
To Voice Mail	enter into voice mail (※extra buying function)
To Auto Attendant	Enter greeting of AA to be greeting of layer 2

Call Waiting Times: default is 30 seconds

Dial Error Times: default 3 times for dial error

Finish all setting items, press **SAVE** putton.

DUTY TIME

The process for Duty Auto Attend

Working hours: On Duty AA

The time **Before Noon** and the time **After Noon**

Noon rest: Break Duty AA

The interval of noon rest

After working Hours: Off Duty AA

The time after working hours

Finish all setting items, press **SAVE** putton.

網路電話伺服器 SIP SERVER

Local Port: Assigned the port of SIP protocol
The registered number for SIP Client must use the same port.

Register Expire: default for 300 second

RTP Port: Real-time Transport Protocol Port

RFC2833 Payload type: RTP Payload for DTMF Digits,
Telephony Tones and Telephony Signals
Reference setting for Telecom Specification
Ex: FETnet use 97, this section must fill in 97.
Be careful, the registered units must fill in 97, too.

G711 Packets/Frame: default for 20
G729 Packets/Frame: default for 20
G723 Packets/Frame: default for 30
※These values have been tuned, no need for amendment.

Default DTMF Type: 2 selections RFC2833 / SIP_INFO
※ Must be compatible with Telecom specification.
※ the registered units must use the same values. Default for RFC2833

Codec Support: 3 selections for voice codec G.711 / G.723 / G.729
※Must be compatible with Telecom specification.
※the registered units must use the same values. Default for G.729.

Finish all setting items, press **SAVE** putton.

EXTENSION NUMBER

Talk Server Pro: available for 64 extensions
Uni-Talk: available for 8 extensions

S_NO column: Point out the S_NO to enter the WEB

Number column: Show the extension number

Transfer mode column: Show the status of transfer

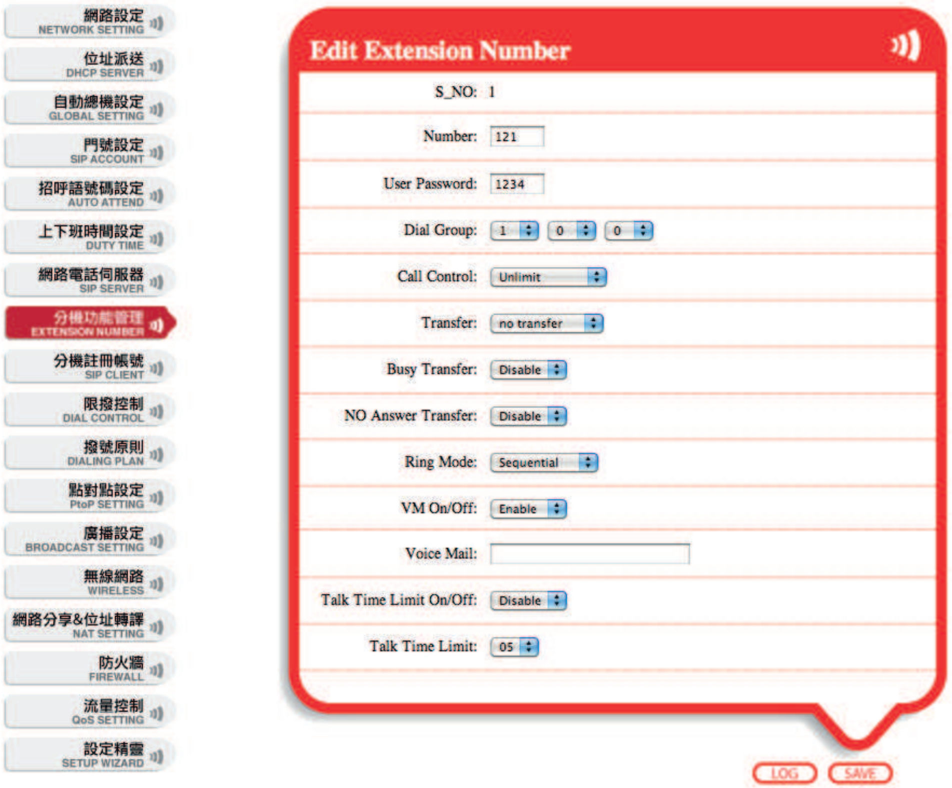
Select column: Point out the square and select delete, then user can delete

ADD putton:
To add new extension, point out the ADD to enter the WEB.

DELETE putton:
Delete the extension once user points out at the Select column.

Point out the number of S_NO column or press ADD to enter WEB page.

Edit Extension Number



S_NO: No need for amendment

Number: set the extension number

User Password: Set the password of extension,
if phone command(*6#,*12#, *15#..,) is in process.

Dial Group: Max for 3 groups

Call Control: the limit of extension
4 selections: Unlimit / NoInternation / NoDomestic / NoOutline
Unlimit: without any control
NoInternation: limit for international
NoDomestic: limit for long distance
NoOutline: only for extension call
NoInternation / NoDomestic:
limit table for calling, must enter **CALL CONTROLL** page

Transfer:
Four available selections: no transfer / to v_mail / to user / to outside
No transfer: default no transfer
to v_mail: extra buying functions
to user: transfer to other extensions,
this selection will appear Transfer Number column
to outside: transfer to CO number,
this selection will appear Transfer Number column

Busy Transfer: Transfer to other extensions if busy
2 selections (Disable / Enable): default is Disable
Transfer Number will appear when enable,
fill in the transferred extension number.

No Answer Transfer: Transfer to the other extension if no answer.
2 selections (Disable / Enable): default is Disable

Ring Mode: Multi phone set to use the same extension for ringing
2 selections (Ring All / Single User)
Ring All: Ring all extensions
Single User: only ring one

VM On/Off: To set if the extension has the right to use voice mail.
2 selections (Enable / Disable): Default is Enable

Talk Time Limit On/Off:
allow the limit time of Talking on extension or not.
2 selections (Enable / Disable): Default is Disable

Talk Time Limit: set the limit time of Talking on extension

Finish all and press **SAVE** putton to save the setting file.

Press the **LOG** putton to check the talking records of extension.

SIP CLIENT

S_NO column: To show the number

Register ID column: show the registered number and status of register
Point out to enter WEB

RingSwitch column: To show the status of ringing
（Enable: allow ringing / Disable: Not allow ringing）

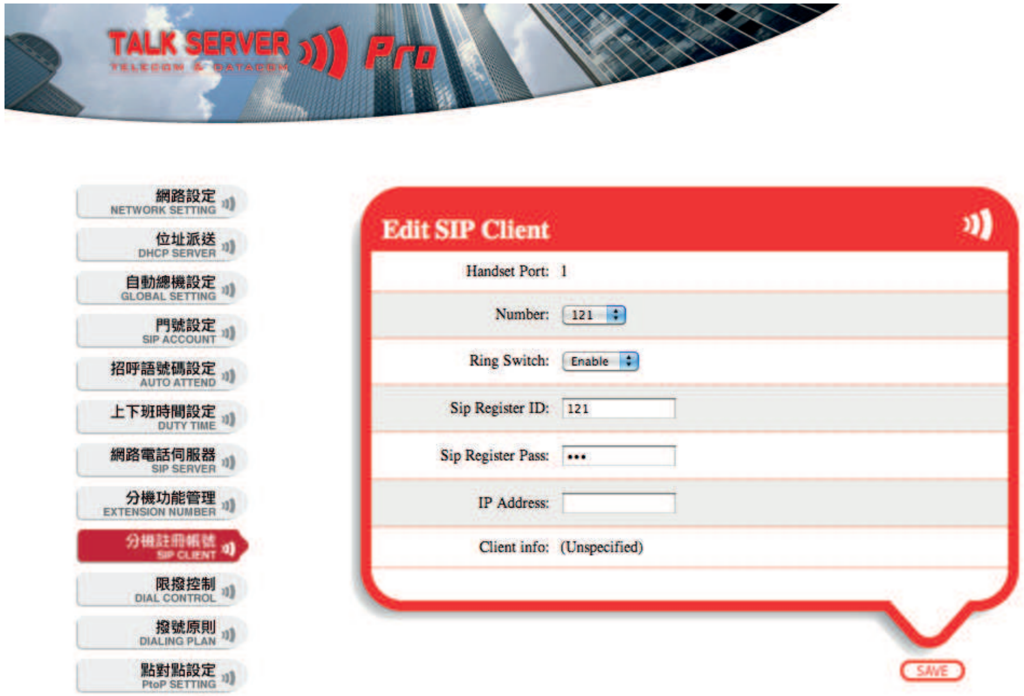
Number column: Show the extension number of registered account

Select column: Point out the square and select delete

ADD putton: To add new extension, point out the ADD to enter the WEB.

DELETE putton:
Delete the extension once user points out at the Select column.

Edit SIP Client



Handset Port: Number

Number: Select the extension number

Ring Switch: 2 selections (Enable / Disable)
Default is Enable to allow to ring. If the selection is Disable,
the registered number of SIP client will be DND (Do not disturb).

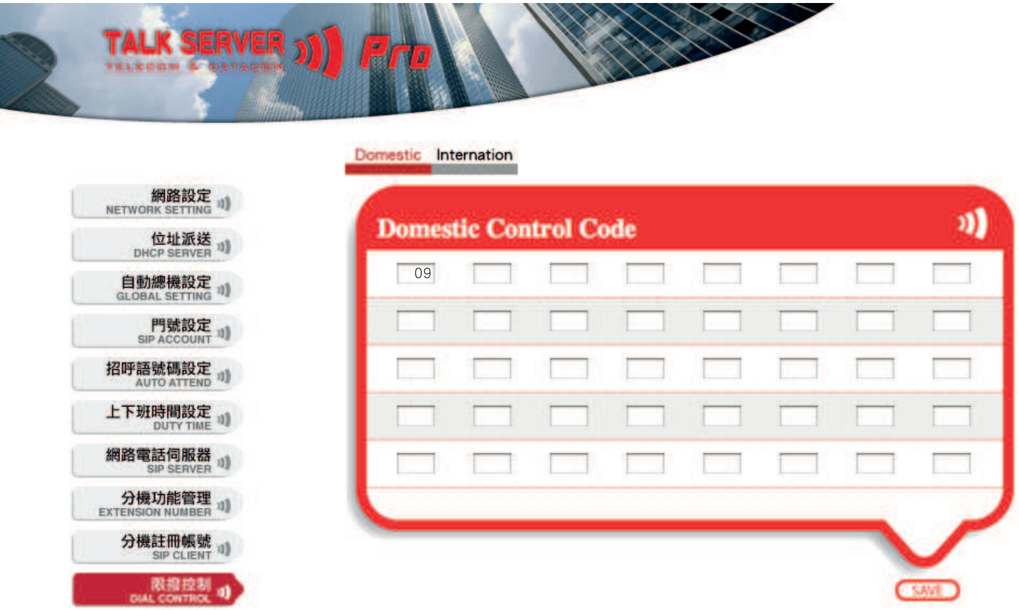
SIP Register ID: Set the number of registered account to let phone recognize the other units

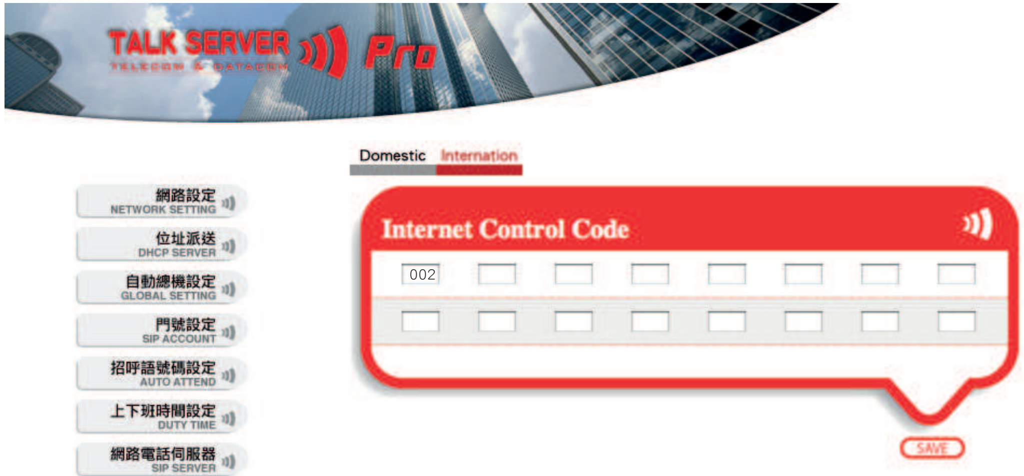
SIP Register Pass: Set the password

Client info: Show the IP Address of SIP Client

Finish all and press **SAVE** putton to save the setting file.

DIAL CONTROL





Dial Control Table

The extension will according to previous number of these 2 table(domestic & international).
If the first 2 digits of out-going number match with dial control, the phone cannot call out。

Phone command (Function code)

Press 9 for outgoing call

***6#** Pick up the incoming call, if the other extension is ringing.

***12#** Direct transfer without any limits to other extensions/Trunk line/Voice mail

***15#** Busy transfer to other extensions.

***16#** transfer to other extensions if no one answer.

***14#** **s**et DND mode

***25#** Hear the message of your own phone set

***2#** Hear the message of the other phone set.

***31#** Paging to the same group