

# **NDC2210S-iPHONE USER MANUAL**

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After phone is boot-up and obtains the TFTP Server address through DHCP Sever, then phone will produce "toot-toot" sound to prompt entering "Config ID". After inputting ID through the numeric keyboard and "#" for ending, phone will auto-download the config file from TFTP Server. If the download is successful, phone will be re-boot; if failed, 15s later phone will enter into default standby status; if you do not want to download, you can press "#" directly for entering into the default standby status; if the download is not completed or the downloaded config file "autoupdate module" has no config parameter in the "config file name", phone will still prompt inputting "Config ID" after re-boot. After phone enters into the default standby status, pressing "\*\*\*47"key will make the phone auto-broadcast its own IP address.

## **Function**

1. Provide a Backup SIP Server
2. Support NAT, Firewall
3. Support DHCP assigning IP address, etc automatically
4. Support PPPoE (used while connecting ADSL, cable modem)
5. It can update the program through HTTP ,FTP and TFTP
6. Check the dynamic voice; Soft the noise; Buffer technique of voice
7. Hold Function
8. Hotline Function
9. Speed-dial
10. Call-forward, Three-way conference call
11. DND (Do Not Disturb), Black List, Limit List
12. Auto-answer.
13. Set through standard Web Browser
14. Remote Management Function
15. Classification management for common user's password and super user's password.
16. Broadcast the IP address, Vlan ID, version number and phone number.
17. Cordless Handset, 1.9GHz Operation, with Handshake Technology

## **Standard and Protocols**

- ◆ IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- ◆ PPPoE
- ◆ DHCP Client and Server
- ◆ Support G.711a/u, G729, G7231 5.3/6.3 audio Codec
- ◆ SIP RFC3261, RFC 2543
- ◆ Support IAX2
- ◆ TCP/IP: Internet transfer and control protocol
- ◆ RTP: Real-time Transport Protocol
- ◆ RTCP: Real-time Control Protocol
- ◆ VAD/CNG save bandwidth
- ◆ Telnet: Internet's remote login protocol
- ◆ DNS: Domain Name Server
- ◆ TFTP: Trivial File Transfer Protocol

## 1. Introduction

This is the user manual of NDC2210S. Some configuration should be done before use the NDC2210S phone, then it can work normally. This manual will illustrate how to set the phone through keyboard and web service.

### 1.1 Overview of Hardware

#### 1.1.1

The two RJ-45 network interface support the 10/100M Ethernet. The default WAN interface is a DHCP Client server. User connect the WAN interface to ADSL or switch, and connect the LAN interface to the computer. You can use the administrator's user name "admin" and password "admin" to login and set.

#### 1.1.2

Only the WAN interface supports the POE.

### 1.2 Overview of Software

| Network Protocol  | Tone   |
|---|--|
| <ul style="list-style-type: none"> <li>I SIP v1(RFC2543)</li> <li>    v2(RFC3261)</li> <li>I IP/TCP/UDP/RTP/RTCP</li> <li>I IP/ICMP/ARP/RARP/SNTP</li> <li>I TFTP Client/DHCP Client/PPPOE Client</li> <li>I Telnet/HTTP Server</li> <li>I DNS Clients</li> </ul> | <ul style="list-style-type: none"> <li>I Ring Tone</li> <li>I Ring Back Tone</li> <li>I Dial Tone</li> <li>I Busy Tone</li> </ul>                              |
|   | <b>Phone Function</b>  |
|   | <ul style="list-style-type: none"> <li>I Volume Adjustment</li> <li>I Speed dial key</li> <li>I Phonebook</li> </ul>   |
| <b>Codec</b>  | <b>IP Assignment</b>   |
| <ul style="list-style-type: none"> <li>I G.711: 64k bit/s(PCM)</li> <li>I G.723.1: 63k/5.3k bit/s</li> <li>I G.726: 16k/24k/32k/40k bit/s(ADPCM)</li> <li>I G.729A: 8k bit/s(CS-ACELP)</li> <li>I G.729B: adds VAD &amp; CNG to G.729</li> </ul>                  | <ul style="list-style-type: none"> <li>I IP (Static IP)</li> <li>I DHCP</li> <li>I PPPoE</li> </ul>  |
| <b>Voice Quality</b>  | <b>Security</b>  |
| <ul style="list-style-type: none"> <li>I VAD: Voice activity detection</li> <li>I CNG: Comfortable noise generator</li> <li>I LEC: Line echo canceller</li> <li>I Packet Loss Compensation</li> <li>I Adaptive Jitter Buffer</li> </ul>                           | <ul style="list-style-type: none"> <li>I HTTP 1.1 basic/digest authentication for Web setup</li> <li>I MD5 for SIP authentication (RFC2069/RFC2617)</li> </ul> |
|   | <b>QoS</b>   |
|   | <ul style="list-style-type: none"> <li>I QoS field</li> </ul>  |
| <b>Call Function</b>  | <b>NAT Traversal</b>   |
| <ul style="list-style-type: none"> <li>I Call Hold</li> <li>I Call Waiting</li> </ul>   | <ul style="list-style-type: none"> <li>I STUN</li> </ul>   |
|   | <b>Configuration</b>   |

|                             |                         |
|-----------------------------|-------------------------|
| I Call Forward              | I Web Browser           |
| I Caller ID                 | I Console/Telnet        |
| I 3-way conference          | I Keypad                |
| <b>DTMF</b>                 | <b>Firmware Upgrade</b> |
| I DTMF RELAY                | I TFTP                  |
| I DTMF RFC 2833             | I HTTP                  |
| I DTMF SIP Info             | I FTP                   |
| <b>SIP Server</b>           |                         |
| Provide a Backup SIP Server |                         |

## 2. NDC2210S phone keyboard

### 2.1 Keyboards functions

#### 2.1.1 Function Table of Base Keyboard

| Name      | Status  | Function   |
|-----------|---------|--|
| Store     | On-hook | To enter storage mode for speed dialing  |
| Flash     | On-hook | To enter the deleting mode for pressing down 3 seconds   |
| Redial    | Dialing | Re-dial the last called number   |
| Locate    | On-hook | It's calling handset for press down for a little time and registering with handset for pressing down 3 seconds |
|           | Call    | calling the handset  |
| Volume +  | Call    | Increase the volume  |
| Volume -  | Call    | Decrease the volume  |
| Speak     |         | Hands free   |
| Mute      | Call    | Mute   |
| M1~M10    | On-hook | 10 speed dial numbers  |
|           | Dialing |  |
| Voicemail | On-hook | Pick up voicemail  |
|           | Dialing |  |
| 1         | Dialing | "1"  |
| 2         | Dialing | "2"  |
| 3         | Dialing | "3"  |
| 4         | Dialing | "4"  |
| 5         | Dialing | "5"  |
| 6         | Dialing | "6"  |
| 7         | Dialing | "7"  |
| 8         | Dialing | "8"  |
| 9         | Dialing | "9"  |
| 0         | Dialing | "0"  |
| *         | Dialing | "*"  |
| #         | Dialing | It can be regarded as the first number being dialed out or the end mark for ending number.                     |

#### 2.1.2 Function table of handset keyboard

### 2.2 Keyboard Function

| Item     | Status  | Function                       |
|----------|---------|--------------------------------|
| ON/OFF   |         | On/off Hook switch of Handset  |
| Hold     | Call    | Call Waiting                   |
| Mute     | Call    | Mute                           |
| Redial   | Dialing | Redial the number of last time |
| Volume + | Call    | Increase the volume            |
| Volume - | Call    | Reduce the volume              |

|       |          |  |
|-------|----------|--|
| Flash | Off-Hook | Quickly cut off the current line and off-hook again.                                       |
| 1     | Dialing  | "1"  |
|       | On-Hook  | Long pressing can pick up the message  |
| 2     | Dialing  | "2"  |
|       | On-Hook  | Long pressing can call the number stored in mem6 on the base unit                          |
| 3     | Dialing  | "3"  |
|       | On-Hook  | Long pressing can call the number stored in mem7 on the base unit                          |
| 4     | Dialing  | "4"  |
|       | Off-Hook | Long pressing can call the number stored in mem8 on the base unit                          |
| 5     | Dialing  | "5"  |
|       | On-Hook  | Long pressing can call the number stored in mem9 on the base unit                          |
| 6     | Dialing  | "6"  |
|       | On-Hook  | Long pressing can call the number stored in mem10 on the base unit                         |
| 7     | Dialing  | "7"  |
| 8     | Dialing  | "8"  |
| 9     | Dialing  | "9"  |
| 0     | Dialing  | "0"  |
| *     | Dialing  | "*"  |
|       | On-Hook  | Long pressing for 3 seconds entering into the registration status of the handset           |
| #     | Dialing  | It can be regarded as the first number being dialed out or the end mark for ending number. |

### 2.2.1 Store numbers

Press "STORE" key to enter into the setting state of storing keys, and input storing number, then press the corresponding storing key to store the numbers.

### 2.2.2 Voice Control

Pressing "VOL+" to increase the volume, and press "VOL-" to decrease the volume.

### 2.2.3 Hold Function

It is used for holding the current line or forwarding to other handset, for example:

When you use the #1 handset and want to use the #2 in a call, now you need to hold the #1 handset then pick up the #2 handset.

### 2.2.4 Register handset

Under On-Hook status, pressing the "Locate" key on the base unit for three seconds to enable the handset register mode, and the Charge indicator light flashes at the same time ;then press "\*"key for three seconds to enable the handset



register mode, then ON/OFF indicator light on the handset ,and will give the prompt tone for successful registration.

#### 2.2.5 Delete handset

Keep pressing the "FLASH" key on the base unit for three seconds, the Speaker will give "beep" sound, then

Press "0" on the base unit to delete all the handset.

Press "1" on the base unit to delete the first handset.

Press "2" on the base unit to delete the second handset.

Press "3" on the base unit to delete the third handset.

Press "4" on the base unit to delete the fourth handset.

After pressed the number keys, Charge indicator light will flashes, which means the handset is being deleted by the phone. When Charge indicator light stops flashing, it means that the deletion is finished.

#### 2.2.6 Call--forward

The "Enable Call Transfer" in the "Call Service Setting" should be in chosen mode (the default one is chosen mode). Suppose the user of NDC2210S is user A, and user B called user A through VoIP , during the communication, user B want to make a call to user C, then user A should press FWD, after that user B can call user C through dialing numbers.

#### 2.2.7 Three way conference call

##### 2.2.7.1 Three way conference call of the handset

1) Choosing the "Enable Three Way Call" of "Call Service Setting" (the default option status of the phone), suppose the user of the NDC2210S phone is A, and if user B calls user A through VoIP phone, user B needs to make 3-way conference call with user A and C, then user A can use the second line to call user C, the call auto-maintain with user B, during the conversation with C, pressing down "\*\*", then can make 3-way conference call.

2) Choosing the "Enable Three Way Call" of "Call Service Setting" (the default option status of the phone), suppose the user of the NDC2210S phone is A, and if user B calls user A through VoIP phone, user B needs to make 3-way conference call with user A and C, then user A can use the second line to call user C, the call auto-maintain with user B, during the conversation with C, pressing down "Conf", then can make 3-way conference call.

##### 2.2.7.2 Three way conference call of the base unit

Choosing the "Enable Three Way Call" of "Call Service Setting" (the default option status of the phone), suppose the user of the NDC2210S phone is A, and if user B calls user A through VoIP phone , user B needs to make 3-way conference call with user A and C, then user A can use the second line to call user C, the call auto-maintain with user B, during the conversation with C, pressing down "\*\*" can make 3-way conference call.

#### 2.2.8 Line Switch

##### 2.2.8.1 Handset Line Switch

1) Suppose the line1/2 of handset is talking, the line2/1 is holding, if you want to implement line switch, first you need to press down "Hold ", then press down

"ON/OFF" of the choice line. You can implement line switch.

2) Suppose the line1/2 of handset is holding, the line2/1 is dialing, if you want to switch to line1/2 talking, first hang up line2/1, then press down "ON/OFF" of Line1/2 or press "HOLD". You can implement line switch.

#### 2.2.8.2 Base unit Line Switch

Suppose the line1/2 of the base unit is talking, line2/1 is holding, press down "line2/1" directly and switch to line2/1 talking.

## 2.3 Functions and setting catalog

### 2.3.1 Menu catalog

- 1) Network
- 2) Call Feature
- 3) SIP
- 4) DSP
- 5) System
- 6) Other Setting

### 2.3.2 Network

#### 2.3.2.1 LAN

- 1) Bridge Mode
- 2) IP
  - ◆Switch
  - ◆DNS Relay
- 3) Netmask
- 4) DHCP Server
  - ◆Switch
  - ◆DNS Relay
- 5) NAT
  - ◆Switch
  - ◆FTPalg
  - ◆PPPTPalg

#### 2.3.2.2 WAN

- 1) Status
- 2) Static Net
  - ◆IP
  - ◆NetMask
  - ◆Gateway
  - ◆DNS
  - ◆DNS2
- 3) PPPOE
  - ◆User name
  - ◆Password
- 4) QoS

### 2.3.3 Call Feature

#### 2.3.3.1 Phone-number

- 1) Public SIP

- 2) Private SIP
- 2.3.3.2 Limit-List
  - 1) Current
  - 2) ADD
  - 3) DEL
- 2.3.3.3 Black-List
  - 1) Current
  - 2) ADD
  - 3) DEL
- 2.3.3.4 FastCall
- 2.3.3.5 Three Talk
- 2.3.3.6 Call-Transfer
- 2.3.3.7 Call-Waiting
- 2.3.3.8 Call-Forward
  - 1) Condition
  - 2) SIP
    - ◆Transfer Num
    - ◆Transfer IP
    - ◆Port
- 2.3.3.9 Dial-Rule
  - 1) End With “#”
  - 2) Fixed Length
    - ◆Switch
    - ◆Length

## **2.3.4 SIP**

- 2.3.4.1 Reg Status
  - 1) Public Reg
  - 2) Private Reg
- 2.3.4.1 Reg Switch
  - 1) Public
  - 2) Private
- 2.3.4.2 Server
  - 1) Public
  - 2) Private
- 2.3.4.3 Domain
  - 1) Public
  - 2) Private
- 2.3.4.4 User Agent
  - 1) Public
  - 2) Private
- 2.3.4.5 Detect-server
- 2.3.4.6 Dtmf-mode
- 2.3.4.7 Interval-time
- 2.3.4.8 Swap-server

- 2.3.4.9 RFC-version
- 2.3.4.10 Signal-Port
- 2.3.4.11 Stun
  - 1) Switch
  - 2) Addr
  - 3) Port
  - 4) Expire Time

### **2.3.5 DSP**

- 2.3.5.1 Codec
- 2.3.5.2 Handdown-time
- 2.3.5.3 Dtfm-Volume
- 2.3.5.4 Input-Volume
- 2.3.5.5 Output-Volume

### **2.3.6 System**

- 2.3.6.1 Save
- 2.3.6.2 Reboot
- 2.3.6.3 Set Default

### **2.3.7 Other Setting**

- 2.3.7.1 Syslog
  - 1) Switch
  - 2) Server-IP
  - 3) Server-Port

### **2.3.8 Setting catalog**

1. Pressing down the "Menu" key to enter the setting status of speed dial key and input the needed saving No., then press the corresponding speed dial key to save.
2. Pressing down "Menu" above 3 seconds to enter the setting status, and the default keyword is 123, then you can press "Enter" to enter, "Menu" to exit.
3. When modifying the setting, press "Redial" to enter modification status, and "0" is to make no choice, "1" is to make choice, "Enter" is to confirm the modification, "Menu" is to quit the modification. After finished the modification setting, we will save it on "Save" menu. After rebooting all the settings will be go into effect.

### 3. Through web browser to set phone

Plug one end of the network line to the network card port of the computer, the other end to Lan port of the phone, phone will obtain the IP address automatically, open IE, input the IP address on Address column, then enter into the Web Setting Page. The method of obtaining the dynamic IP address is: under the on-hook status, press “\*\*47#”, then phone will broadcast the current IP address.

#### 3.1 Logon

The default user name and password are admin/admin and guest/guest. and, admin/admin is super user name and password, guest/guest is common user name and password. Logon interface is as follows:

Username:

Password:

#### 3.2 Current state

This page layout shows the work state of VoIP phone. The network part shows the connection state of WAN interface and LAN interface and the network setting; the work state of Public SIP service of VoIP part, and here you can see the registration and whether registered to the server or not. The Phone Number part shows the telephone numbers in Private SIP server and Public SIP server.

## Current Status

| Network   |                         |             |              |
|---|-------------------------|-------------|--------------|
| WAN   |                         | LAN         |              |
| Connect Mode                                      | DHCP                    | IP Address  | 192.168.10.1 |
| MAC Address                                       | 00:19:f3:00:8b:f6       | DHCP Server | OFF          |
| IP Address  | 192.168.21.106          |             |              |
| Gateway   | 192.168.21.1            |             |              |
| Phone Number                                      |                         |             |              |
| SIP LINE 1  | 210@61.156.234.90 :5060 |             | Registered   |
| SIP LINE 2  | @ :5060                 |             | Unapplied    |
| Version: 9602SIP V1.8.90-603 Aug 18 2009 17:37:23 |                         |             |              |

### 3.3 Network

#### 3.3.1 Wan Config

WAN port network setting page.

Support static IP, dynamic obtain IP and PPPoE.

### WAN Configuration

| Wan Status      |                   |
|-----------------|-------------------|
| Active IP       | 192.168.0.10      |
| Current Netmask | 255.255.255.0     |
| Current Gateway | 192.168.0.1       |
| MAC Address     | 00:19:f3:00:5b:ae |
| Get MAC Time    | 20080728          |

  

| WAN Setting                          |                                       |                             |
|--------------------------------------|---------------------------------------|-----------------------------|
| Static <input type="radio"/>         | DHCP <input checked="" type="radio"/> | PPPOE <input type="radio"/> |
| <input type="button" value="APPLY"/> |                                       |                             |

Configure Static IP:

| WAN Setting                             |   |                             |
|---|---|-----------------------------|
| Static <input checked="" type="radio"/> | DHCP <input type="radio"/>                  | PPPOE <input type="radio"/> |
| Static IP Address                       | <input type="text" value="192.168.1.179"/>  |                             |
| Netmask                                 | <input type="text" value="255.255.255.0"/>  |                             |
| Gateway                                 | <input type="text" value="192.168.1.1"/>    |                             |
| DNS Domain                              | <input type="text"/>                        |                             |
| Primary DNS                             | <input type="text" value="202.96.134.133"/> |                             |
| Alter DNS                               | <input type="text" value="202.96.128.68"/>  |                             |
| <input type="button" value="APPLY"/>    |   |                             |

----Enable *Static*;

----Set NDC2210S's IP address in the *IP Address*;

----Set netmask in the *Netmask* field;

----Set router IP address in the *Gateway*;

----DNS Domain:

----Set local DNS server in the *Preferred DNS* and the *Alternate DNS*.

Configure to dynamic obtain IP

----Enable *DHCP*;

If there is DHCP server in your local network, NDC2210S will automatically obtain WAN port network information from your DHCP server.

Configure PPPoE:

| WAN Setting                          |  |
|--------------------------------------|--|
| Static <input type="radio"/>         | DHCP <input type="radio"/>             |
|                                      | PPPOE <input checked="" type="radio"/> |
| PPPOE Server                         | <input type="text" value="ANY"/>       |
| Username                             | <input type="text" value="user123"/>   |
| Password                             | <input type="password" value="*****"/> |
| <input type="button" value="APPLY"/> |  |

----Enable **PPPoE**

----**PPPoE server**: Enter "ANY" if no specified from your ITSP.

----Enter PPPoE username and password in the **username** and **password**.

NDC2210S will automatically obtain WAN port network information from your ITSP if PPPoE setting and the setup are correct.

**Notice:** If user accesses the IP phone through WAN port. He/She should use the new IP address to access the IP phone when the WAN port address was changed.

### 3.3.2 LAN Config

**LAN IP /Netmask:** Set the IP and Netmask for the LAN

**DHCP Server:** Enable DHCP service in LAN port; after user changed LAN IP, phone will automatically modify DHCP Lease Table and save the configure according to IP and Netmask, DHCP server configure won't take effect unless you reboot the device.

**NAT:** Enable NAT.

**Bridge Mode:** Enable this option to switch to bridge mode. IP phone won't assign IP for its LAN port in bridge mode and its LAN and WAN port will be in the same network. **(This setting won't take effect unless you save the config and reboot the device)**

## LAN Configuration

| Lan Set                              |  |
|--------------------------------------|--|
| LAN IP                               | <input type="text" value="192.168.10.1"/>  |
| Netmask                              | <input type="text" value="255.255.255.0"/> |
| DHCP Service                         | <input type="checkbox"/>                   |
| NAT                                  | <input type="checkbox"/>                   |
| Bridge Mode                          | <input checked="" type="checkbox"/>        |
| <input type="button" value="APPLY"/> |  |

### 3.4 VoIP

#### 3.4.1 SIP Config

Setting page of public SIP server:

## SIP Configuration

SIP Line Select

SIP 1

Basic Setting

|                 |                                     |                            |  |
|-----------------|-------------------------------------|----------------------------|--|
| Register Status | Registered                          | Display Name               | <input type="text"/>                     |
| Server Address  | 61.156.234.90                       | Proxy Server Address       | <input type="text"/>                     |
| Server Port     | 5060                                | Proxy Server Port          | <input type="text"/>                     |
| Account Name    | 6009                                | Proxy Username             | <input type="text"/>                     |
| Password        | ••••••••                            | Proxy Password             | <input type="text"/>                     |
| Phone Number    | 6009                                | Domain Realm               | <input type="text"/>                     |
| Enable Register | <input checked="" type="checkbox"/> | Message Waiting Indication | Disable <input type="button" value="v"/> |

**Register Server Addr:** Register address of public SIP server;

**Register Server Port:** Register port of public SIP server, default port is 5060;

**Register Username:** Username of your SIP account (Always the same as the phone number);

**Register Password:** Password of your SIP account.

**Proxy Server Addr:** IP address of proxy SIP server (SIP provider always use the same IP for register server and proxy server, in this case you don't need to configure the proxy server information.);

**Proxy Server Port:** Signal port of SIP proxy;

**Proxy Username:** proxy server username;

**Proxy Password:** proxy server password;

**Domain Realm:** SIP domain, enter the SIP domain if any, otherwise NDC2210S will use the proxy server address as SIP domain.

**Local SIP port:** Local SIP register port, default 5060;

**Phone Number:** Phone number of your SIP account;

**Enable Register:** Enable/Disable SIP register. NDC2210S won't send register info to SIP server if disable register.

**Message Waiting Indication:** Set Disable/Enable message waiting through pull-down menu, including:

Disable: MWI is disabled, even if received NOTIFY message from the server indicating new voicemail, phone will not prompt.;

Enable (Subscribe): MWI is enabled and SUBSCRIBE will be sent, if the server sends NOTIFY message indicating new voicemail received, MWI LED will blink to prompt.



Enable(No Subscribe): MWI is enabled, but phone will not send SUBSCRIBE, if the server sends NOTIFY message indicating new voicemail received, MWI LED will also blink to prompt.

SIP1 Subscribe will still be used in config file: config option "0" means "Disable", "1" means "Enable (Subscribe)", "2" means "Enable (No Subscribe)".

### Advanced SIP Setting

| Advanced SIP Setting                 |   |         |                          |   |
|--------------------------------------|---|---------|--------------------------|---|
| Register Expire Time                 | <input type="text" value="60"/>             | seconds | Forward Type             | <input type="text" value="Off"/>          |
| NAT Keep Alive Interval              | <input type="text" value="60"/>             | seconds | Forward Phone Number     | <input type="text"/>                      |
| User Agent                           | <input type="text" value="Voip Phone 1.0"/> |         | Server Type              | <input type="text" value="common"/>       |
| Signal Key                           | <input type="text"/>                        |         | DTMF Mode                | <input type="text" value="DTMF_RFC2833"/> |
| Media Key                            | <input type="text"/>                        |         | RFC Protocol Edition     | <input type="text" value="RFC3261"/>      |
| Local Port                           | <input type="text" value="5060"/>           |         | Transport Protocol       | <input type="text" value="UDP"/>          |
| Ring Type                            | <input type="text" value="Type 1"/>         |         | Subscribe Expire Time    | <input type="text" value="300"/> seconds  |
| Enable Subscribe                     | <input type="checkbox"/>                    |         | Enable URI Convert       | <input checked="" type="checkbox"/>       |
| Enable Keep Authentication           | <input type="checkbox"/>                    |         | Signal Encode            | <input type="checkbox"/>                  |
| NAT Keep Alive                       | <input type="checkbox"/>                    |         | Rtp Encode               | <input type="checkbox"/>                  |
| Enable Via rport                     | <input checked="" type="checkbox"/>         |         | Enable Session Timer     | <input type="checkbox"/>                  |
| Enable PRACK                         | <input type="checkbox"/>                    |         | Answer With Single Codec | <input type="checkbox"/>                  |
| Long Contact                         | <input type="checkbox"/>                    |         | Auto TCP                 | <input type="checkbox"/>                  |
| Click To Talk                        | <input type="checkbox"/>                    |         |                          |   |
| <input type="button" value="APPLY"/> |   |         |                          |   |

**Register Expire Time:** Register Expire time, default is 60 seconds. NDC2210S will auto configure this expire time as the server recommends if it is different from the SIP server.

NAT Keep Alive Interval: NAT keep alive interval time, the default is 60 seconds;

Forward Type: the type of call forward; (the default is OFF);

OFF: Disable call forward;

Busy: If the phone is busy, it will forward to the appointed phone;

No answer: If no answer, it will forward to the appointed phone;

Always: The caller always forward to the appointed phone.

**Forward Photo Number:** Call the forwarded phone number.

Signal Key: Setting Signal Key; In order to prevent blocking, cooperate platform to encrypt signal, input key here,

Media Key: Setting Media Key;

Subscribe Expire Time: Config the time of sending subscription message; Each interval time, sending a subscription message. Mainly subscribe other's state or voice message.

Enable URI Convert: Enable URI Convert function, Exchange '#' to '%23' and sending out .

Enable Keep Authentication: Whether allow the phone support register with authentication directly to send or not, Such device do not certification requirement and response with server every time, After the server received a request with the

registration of certification , The server can directly reply confirmation message.

**Detect Interval Time:** Set server detect interval time, if phone open SIP detect server function, it will detect server every time whether or not to respond.

**Signal Encode:**Open signal encrypt;

**Rtp Encode:** Open voice encrypt;

**Enable Via rport:** Whether support the RFC3581 or not. Enable Via rport is used in inside network,and needs SIP server support. To maintain network equipment with net device outside of NAT connection.

**Enable Session Timer:** Whether the phone support the RFC4028 or not .At a certain period of time to refresh conversation ,in case a long time ,the other side drop off or break off, after refresh ,the other side did not respond ,we will hang up.

**Enable PRACK:**Whether the phone support SIP-prack function (used mainly in the CRBT). When you receive 183 signal, you can send prack message. Recommend to use the default configuration.

**Answer With Single Codec:** When called to do, Only in response to Codec.

**Long Contact:** Config Contact field bring more parameter.

**Auto TCP:** When th message body exceeds 1300 bytes, automatic transmission using the TCP protocol,to protect the availability of transmission.

**Detect Interval Time:** Co-work with the Auto Detect Server, if Auto Detect Server is enable, NDC2210S will periodically detect whether the SIP server is available according this setting.

**Encrypt Key:** The particular service system decrypts of the key, matching with the server Type usage, the key provide by the particular service system supplier, default is empty

**Server Type:** The particular service system supplier carries out the sign and speeches to encrypt, default is common;

**DTMF Mode:** DTMF signal sending mode: support RFC2833, DTMF\_RELAY (inband audio) and SIP info.

**RFC Protocol Edition:** Current NDC2210S SIP version. Set to RFC 2543 if the gate need to communicate to devices (such as CISCO5300) using the SIP 1.0. Default is RFC 3261.

### 3.4.2 Iax2 Config

Setting page of public IAX server:

## IAX2 Configuration

| IAX2                                 |                          |
|--------------------------------------|--------------------------|
| Register Status                      | Unregistered             |
| IAX2 Server Addr                     | <input type="text"/>     |
| IAX2 Server Port                     | 4569                     |
| Account Name                         | <input type="text"/>     |
| Account Password                     | <input type="text"/>     |
| Phone Number                         | <input type="text"/>     |
| Local Port                           | 4569                     |
| Voice Mail Number                    | 0                        |
| Voice Mail Text                      | mail                     |
| Echo Test Number                     | 1                        |
| Echo Test Text                       | echo                     |
| Refresh Time                         | 60 Seconds               |
| Enable Register                      | <input type="checkbox"/> |
| Enable G.729                         | <input type="checkbox"/> |
| IAX2(Default Protocol)               | <input type="checkbox"/> |
| <input type="button" value="APPLY"/> |                          |

**IAX Server Addr:** Register address of public IAX server;

**IAX Server Port:** Register port of public IAX server, default port is 4569;

**Account Name:** Username of your SIP account (Always the same as the phone number);

**Account Password:** Password of your IAX account.

**Local port:** Signal port of local, default port is 4569;

**Phone Number:** Phone number of your IAX account;

**Voice mail number:** If the IAX support voice mail, but your username of the voice mail is letters which you can not input with the ATA, then you use the number to stand for your username;

**Voice mail text:** if IAX support voice mail, config the domain name of your mail box here.

**Echo test number:** If the platform support echo test, and the number is test form, config the test number to replace the text format The echo test is to test the working status of terminals and platform;

**Echo test text:** echo test number in text format;

**Refresh time:** IAX refresh time;

**Enable Register:** enable or disable register;

**Enable G.729:** Using G.729 speech coding mandatory consultations

**IAX2(Default Protocol):** Set IAX2 as the default protocol , if not the system will choose SIP as default;

### 3.5 Advance

#### 3.5.1 DHCP Server

DHCP server manage page.

User may trace and modify DHCP server information in this page.

### DHCP Service

| DHCP Leased Table                  |                                     |               |                         |               |              |                                       |
|------------------------------------|-------------------------------------|---------------|-------------------------|---------------|--------------|---------------------------------------|
| Leased IP Address                  |                                     |               | Client Hardware Address |               |              |                                       |
| DHCP Lease Table                   |                                     |               |                         |               |              |                                       |
| Name                               | Start IP                            | End IP        | Lease Time              | Netmask       | Gateway      | DNS                                   |
| lan                                | 192.168.10.1                        | 192.168.10.30 | 1440                    | 255.255.255.0 | 192.168.10.1 | 192.168.10.1                          |
| DHCP Lease Table Setting           |                                     |               |                         |               |              |                                       |
| Lease Table Name                   | <input type="text"/>                |               |                         |               |              |                                       |
| Start IP                           | <input type="text"/>                |               |                         |               |              |                                       |
| End IP                             | <input type="text"/>                |               |                         |               |              |                                       |
| Lease Time                         | <input type="text"/> (minute)       |               |                         |               |              |                                       |
| Netmask                            | <input type="text"/>                |               |                         |               |              |                                       |
| Gateway                            | <input type="text"/>                |               |                         |               |              |                                       |
| DNS                                | <input type="text"/>                |               |                         |               |              |                                       |
| <input type="button" value="Add"/> |                                     |               |                         |               |              |                                       |
| DHCP Lease Table Delete            |                                     |               |                         |               |              |                                       |
| Lease Table Name                   | <input type="text" value="lan"/>    |               |                         |               |              | <input type="button" value="Delete"/> |
| DNS relay Setting                  |                                     |               |                         |               |              |                                       |
| DNS Relay                          | <input checked="" type="checkbox"/> |               |                         |               |              | <input type="button" value="APPLY"/>  |

DHCP Lease Table: display the IP–MAC corresponding table that the server distributed.

**Lease Table Name:** Lease table name.

**Start IP:** Start IP of lease table.

**End IP:** End IP of lease table. Network device connecting to the NDC2210S LAN port can dynamic obtain the IP in the range between start IP and end IP.

**Lease Time:** DHCP server lease time.

**Netmask:** Netmask of lease table.

**Gateway:** Default gateway of lease table

**DNS:** Default DNS server of lease table.

**DNS Relay:** Enable DNS relay function.

User may use below setting to add a new lease table.

**Notice:** This setting won't take effect unless you save the config and reboot the device

### 3.5.2 NAT

Advance NAT setting. Maximum 10 items for TCP and UDP port mapping.

## NAT Configuration

| Protocol Set                                  |   |  |                      |
|---|---|--|----------------------|
| <input checked="" type="checkbox"/> IPsec ALG | <input checked="" type="checkbox"/> FTP ALG | <input checked="" type="checkbox"/> PPTP ALG |                      |
| <input type="button" value="APPLY"/>          |   |  |                      |
| NAT Table                                     |   |  |                      |
| Inside IP                                     | Inside TCP Port                             | Outside TCP Port                             |                      |
| Inside IP                                     | Inside UDP Port                             | Outside UDP Port                             |                      |
| NAT Table Option                              |   |  |                      |
| Transfer Type                                 | TCP   | Outside Port                                 | <input type="text"/> |
| Inside Ip                                     | <input type="text"/>                        | Inside Port                                  | <input type="text"/> |
| <input type="button" value="Add"/>            |   | <input type="button" value="Delete"/>        |                      |
| <input type="button" value="DMZ Config"/>     |   |  |                      |

**IPsec ALG:** Enable/Disable IPsec ALG;

**FTP ALG:** Enable/Disable FTP ALG;

**PPTP ALG:** Enable/Disable PPTP ALG;

**Transfer Type:** Transfer type using port mapping.

**Inside IP:** LAN device IP for port mapping.

**Inside Port:** LAN device port for port mapping.

**Outside Port:** WAN port for port mapping.

Click **Add** to add new port mapping item and **Delete** to delete current port mapping item.

### DMZ Config:

| DMZ Table  |                      |
|--|----------------------|
| Outside IP   | Inside IP            |
|  |                      |
| DMZ Table Option   |                      |
| Outside IP   | <input type="text"/> |
| Inside IP  | <input type="text"/> |
| Outside IP   | <input type="text"/> |
| <input type="button" value="Add"/> <input type="button" value="Delete"/> |                      |

### 3.5.3 STUN

This page is used to set the private sip server, stun server, and back up sip server information.

## STUN Configuration

| STUN Set                             |                      |
|--------------------------------------|----------------------|
| STUN NAT Transverse                  | FALSE                |
| STUN Server Addr                     | <input type="text"/> |
| STUN Server Port                     | 3478                 |
| STUN Effect Time                     | 50 Seconds           |
| Local SIP Port                       | 5060                 |
| <input type="button" value="APPLY"/> |                      |

  

| Set Sip Line Enable Stun                  |                          |
|---|--------------------------|
| SIP 1 <input type="button" value="Load"/> |                          |
| Use Stun                                  | <input type="checkbox"/> |
| <input type="button" value="APPLY"/>      |                          |

STUN Server setting: SIP STUN is used to realize SIP penetrates through NAT, when the phone configures IP and port of STUN server (default is 3478) and select Enable SIP Stun, common SIP server can be used to realize the phone to penetrate through NAT. In this way, If you have common SIP proxy and STUN server parked public network, it is all right, but STUN only support three NAT ways: FULL CONE, restricted, port restricted;

**STUN Server Addr:** configure stun server address;

**STUN Server Port:** configure stun server port default 3478

**STUN Effect Time:** stun detect NAT type circle, unit: minute.

**Local SIP Port:** SIP port of this phone.

**Load:** Load the choices of SIP line.

**Use Stun:** Stun. Set the Stun that allows/forbids use user setting.

### 3.5.4 Net Service

## Net Service

| Service Port                         |       |
|--------------------------------------|-------|
| HTTP Port                            | 80    |
| Telnet Port                          | 23    |
| RTP Initial Port                     | 10000 |
| RTP Port Quantity                    | 200   |
| <input type="button" value="APPLY"/> |       |

**HTTP Port:** configure HTTP transfer port; default is 80. User may change this

port to enhance system's security. When this port is changed, please use <http://xxx.xxx.xxx.xxx:xxxx/> to reconnect.

**Telnet Port:** configure telnet transfer port, default is 23.

**RTP Initial Port:** RTP initial port.

**RTP Port Quantity:** Maximum RTP port quantity, default is 200

**Notice:**

Settings in this page won't take effect unless save and reboot the device.

If you need to change telnet port or HTTP port, please use the port greater than 1024, because ports under 1024 is system remain ports.

HTTP service firbids if HTTP is set to 0.

### 3.5.5 Firewall settings

Firewall setting page. User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the internet.

## Firewall Configuration

**Firewall Type**

In\_access Enable
  Out\_access Enable

---

**Firewall Input Rule Table**

| Index | Deny/Permit | Protocol | Src Addr | Src Mask | Des Addr | Des Mask | Range | Port |
|-------|-------------|----------|----------|----------|----------|----------|-------|------|
|       |             |          |          |          |          |          |       |      |

---

**Firewall Output Rule Table**

| Index | Deny/Permit | Protocol | Src Addr | Src Mask | Des Addr | Des Mask | Range | Port |
|-------|-------------|----------|----------|----------|----------|----------|-------|------|
|       |             |          |          |          |          |          |       |      |

---

**Firewall Set**

|               |  |          |  |                                    |
|---------------|--|----------|--|------------------------------------|
| Input/Output  | <input type="button" value="Input"/>     | Src Addr | <input style="width: 90%;" type="text"/> | <input type="button" value="Add"/> |
| Deny/Permit   | <input type="button" value="Deny"/>      | Des Addr | <input style="width: 90%;" type="text"/> |                                    |
| Protocol Type | <input type="button" value="UDP"/>       | Src Mask | <input style="width: 90%;" type="text"/> |                                    |
| Port Range    | <input type="button" value="more than"/> | Des Mask | <input style="width: 90%;" type="text"/> |                                    |

---

**Rule Delete**

|              |                                      |                     |  |                                       |
|--------------|--------------------------------------|---------------------|--|---------------------------------------|
| Input/Output | <input type="button" value="Input"/> | Index To Be Deleted | <input style="width: 90%;" type="text"/> | <input type="button" value="Delete"/> |
|--------------|--------------------------------------|---------------------|--|---------------------------------------|

Access list support two type limits: Input access limit or output access limit.

Each type supports 10 items maximum.

NDC2210S firewall filter is base WAN port. So the source address or input destination address should be WAN port IP address.

Configuration:

**In\_access enable** : enable in\_access rule;

**Out\_access enable**: enable out\_access rule;

**Input/Output:** Specify current adding rule is input rule or output rule.

**Deny/Permit:** Specify current adding rule is deny rule or permit rule.

**Protocol Type:** Protocol using in this rule: TCP/IP/ICMP/UDP.

**Port Range:** Port range of this rule.

**Src Addr:** Source address. It can be a specific IP address or network address.

**Dest Addr:** Destination address. It can be a specific IP address or network address.

**Src Mask:** Source address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID.

**Des Mask:** Destination address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID.

### 3.5.6 VLAN Configuration

Below is QoS Configuration page:

## QoS Configuration

The screenshot shows a 'QoS Set' configuration page. It includes several checkboxes and input fields. 'VLAN ID Check Enable' is checked, while 'VLAN Enable' and 'DiffServ Enable' are unchecked. 'Voice/Data VLAN differentiated' is set to 'Undifferentiated'. 'DiffServ Value' is set to '0x b8'. 'Voice 802.1P Priority' is set to '0' (range 0-7), and 'Data 802.1P Priority' is also set to '0' (range 0-7). 'Voice VLAN ID' is set to '256' (range 0-4095), and 'Data VLAN ID' is set to '254' (range 0-4095). An 'APPLY' button is at the bottom.

NDC2210S phone implement QoS based on 802.1p, The QoS is used to mark the network communication priority in the data link/MAC sub-layer. NDC2210S will sort the packets using the QoS and sends it to the destination.

**VLAN Enable:** If enable the VLAN service, the second layer will realize separate voice, signal and data transmission. To realize separating voice and data transmission by setting IP precedence of ToS area of voice transmission. To reach upper layer switch or router have priority to transfer voice transmission. (The prerequisite is the upper layer switch or router has to identify ToS area.)

**VLAN ID:** Dispose VLAN ID is add a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 1~4094.

**DiffServ Enable:** If enable the VLAN service, it indicates use DSCP mode to realize three layers QoS. This moment, the DSCP of SIP signals which between NDC2210S Phone and MGC. It will use Class Selector 5 (The value is 0xA0). And the DSCP of mediums information (In RTP packets) would be used the values of DiffServ Value field.

**DiffServ Value:** The value range:

0x28,0x30,0x38,0x48,0x50,0x58,0x68,0x70,0x78,0x88,0x90,0x98,0xb8.defa



ult is 0xb8 ,0xb8 stands for best fast transmission; 28-38 is guarantee for the transmission priority for the 1st rank , 48-58 is guarantee for the transmission priority for the 2nd rank, 68-78 is guarantee for the transmission priority for the 3rd rank, 88-98 is guarantee for the transmission priority for the 4th rank.

**802.1P Priority:** The priority of 802.1p

### 3.5.7 Digital Map

Digit map is a set of rules to determine when the user has finished dialing.

## Digital Map Configuration

NDC2210S support below digital map:

Digital Map is based on some rules to judge when user end their dialing and send the number to the server. NDC2210S support following digital map:

----End With "\#": Use # as the end of dialing.

----Fixed Length:: When the length of the dialing is matched, the call will be sent.

----Timeout:: Specify the timeout of the last dial digit. The call will be sent after timeout

----Prefix:: User define digital map:

[ ]: Represents the range of digit, can be a range such as [1-4], or use comma such as [1,3,5], or use a list such as [234]

x : Represents any one digit between 0~9

Tn: Represents the last digit timeout. n represents the time from 0~9 second, it is necessary. Tn must be the last two digit in the entry. If Tn is not included in the entry, we use T0 as default, it means system will sent the number immediately if the number matches the entry.

Example:

8[2-9]xxxxx: : All number from 8200000 to 8899999 will be sent

immediately.

955xx: 5 digits numbers begin with 9 will be sent immediately.

10060: Number 10060 will be sent immediately

22xxxxxT1: 7 digits numbers begin with 22 will be sent after one second

39[3,9]xxxx: 7 digits numbers begin with 393 or 399 will be sent immediately.

### 3.5.8 Call Service Settings

User configure the value add service such as hotline, call forward, call transfer, call waiting, 3-way conference call, auto-answer, etc in this page.

## Call Service

### Call Service Setting

|                       |                                     |                     |  |
|-----------------------|-------------------------------------|---------------------|--|
| Hot Line              | <input type="text"/>                | Warm Line Time      | <input type="text" value="0"/> (0~9 seconds)   |
| P2P IP Prefix         | <input type="text"/>                | No Answer Time      | <input type="text" value="20"/> (0~60 seconds) |
| Do Not Disturb        | <input type="checkbox"/>            | Auto Answer         | <input type="checkbox"/>                       |
| Enable Call Transfer  | <input checked="" type="checkbox"/> | Ban Outgoing        | <input type="checkbox"/>                       |
| Enable Three Way Call | <input checked="" type="checkbox"/> | Enable Call Waiting | <input checked="" type="checkbox"/>            |
|                       |                                     | Accept Any Call     | <input checked="" type="checkbox"/>            |

### Black List

Black List

|                      |                                    |                                       |
|----------------------|------------------------------------|---------------------------------------|
| <input type="text"/> | <input type="button" value="Add"/> | <input type="button" value="Delete"/> |
|----------------------|------------------------------------|---------------------------------------|

### Limit List

Limit List

|                      |                                    |                                       |
|----------------------|------------------------------------|---------------------------------------|
| <input type="text"/> | <input type="button" value="Add"/> | <input type="button" value="Delete"/> |
|----------------------|------------------------------------|---------------------------------------|

**Hotline:** configure hotline number.

**Warm Line Time:** Set waiting time for the user picking up the phone to dial hotline number, the config range is 0-9s, default is 0s.

If warm line time is 0s, then hotline number will be sent right away after off-hook;

If the range is 1-9s, take 3s as an example, hotline number will be sent immediately 3s later without pressing any key.

As long as any key is pressed within the setting time, timer will be suspended.

**Auto Answer:** Enable/disable auto answer function.

**No Disturb:** DND, do not disturb, enable this option to refuse any calls.

**Ban Outgoing:** Enable this to ban outgoing calls.

**Enable Call Transfer:** Please refer to [Value add service](#) for detail.

**Enable Call Waiting:** Enable/disable Call Waiting

**Enable Three Way Call:** Please refer to [Value add service](#) for detail.

**Accept Any Call:** If this option is disable, NDC2210S refuse the incoming call when the called number is different from NDC2210S's phone number.

**No Answer Time:** no answer call forward time setting.

**Black List:** incoming call in these phone numbers will be refused.

**Limit List:** outgoing calls with these phone numbers will be refused

### 3.5.9 Memory Key

This page layout shows the number setting of Voice mail and speed-dial key.

## PHONE

The screenshot displays a configuration page for a phone system. It is divided into two main sections: 'Interface Configuration' and 'Memory Key Setting'. The 'Interface Configuration' section has a yellow header and contains a text input field for 'MWI Number' with the value '\*97' and an 'APPLY' button below it. The 'Memory Key Setting' section also has a yellow header and contains a table with 10 rows, labeled 'Memory 1' through 'Memory 10'. The first four rows have values: 6003, 6009, 6011, and 6012. The remaining six rows are empty. An 'APPLY' button is located at the bottom of this section.

| Interface Configuration              |                                   |
|--------------------------------------|-----------------------------------|
| MWI Number                           | <input type="text" value="*97"/>  |
| <input type="button" value="APPLY"/> |                                   |
| Memory Key Setting                   |                                   |
| Memory 1                             | <input type="text" value="6003"/> |
| Memory 2                             | <input type="text" value="6009"/> |
| Memory 3                             | <input type="text" value="6011"/> |
| Memory 4                             | <input type="text" value="6012"/> |
| Memory 5                             | <input type="text"/>              |
| Memory 6                             | <input type="text"/>              |
| Memory 7                             | <input type="text"/>              |
| Memory 8                             | <input type="text"/>              |
| Memory 9                             | <input type="text"/>              |
| Memory 10                            | <input type="text"/>              |
| <input type="button" value="APPLY"/> |                                   |

### 3.5.10 MMI Filter

MMI filter is used to make access limit to NDC2210S phone.

When MMI filter is enable. Only IP address within the start IP and end IP can access IP phone.

## MMI Filter

| MMI Filter Table                               |                                      |  |
|--|--------------------------------------|--|
| Start IP                                       | End IP                               | Option   |
| 192.168.1.2                                    | 192.168.1.100                        | <input type="button" value="Modify"/><br><input type="button" value="Delete"/> |
| MMI Filter Table Set                           |                                      |  |
| Start IP                                       | End IP                               | <input type="button" value="Add"/>   |
|  |                                      |  |
| MMI Filter Table Set                           |                                      |  |
| <input checked="" type="checkbox"/> MMI Filter | <input type="button" value="APPLY"/> |  |

### 3.5.11 DSP

This page mainly completes voice configuration.

## DSP Configuration

| DSP Set                              |  |                 |  |
|--------------------------------------|--|-----------------|--|
| First Codec                          | <input type="text" value="g711Ulaw64k"/> | Second Codec    | <input type="text" value="g723"/>          |
| Third Codec                          | <input type="text" value="g729"/>        | Fourth Codec    | <input type="text" value="g711Alaw64k"/>   |
| Default Ring Type                    | <input type="text" value="Type 1"/>      | Handdown Time   | <input type="text" value="200"/> ms        |
| Input Volume                         | <input type="text" value="3"/> (1-9)     | Output Volume   | <input type="text" value="9"/> (1-9)       |
| Handfree Volume                      | <input type="text" value="5"/> (1-9)     | Ring Volume     | <input type="text" value="2"/> (1-9)       |
| G729 Payload Length                  | <input type="text" value="20ms"/>        | Signal Standard | <input type="text" value="United States"/> |
| VAD                                  | <input type="checkbox"/>                 |                 |  |
| <input type="button" value="APPLY"/> |  |                 |  |

**CODEC:** select the prefer CODEC; support ulaw, alaw, G729 and G7231 5.3/6.3

**Signal Standard:** Signal standard for different area.

**Handdown Time:** Hand down detect time.

**Input Volume:** Handset in volume.

**Output Volume:** Handset out volume.

**Handfree Volume:** Hand free volume

**G729 Payload Length:** G729 payload length

**VAD:** Enable/disable Voice Activity Detection

### 3.5.12 VPN

This page is VPN setting page, the IP phone support the VPN with UDP and L2TP protocol .The parameters is as below

**VPN IP:** After VPN registered successfully, VPN server will give an IP address

to the terminal. If there is a IP address shown on terminal (except for 0.0.0.0),it means your VPN has registered

UDP Tunnel

**VPN Server Addr:** Register to the address of VPN server

**VPN Server Port:** Register to the port of VPN server

**Server Group ID:** Group ID of UDP VPN

**Server Area Code:** Area code of VPN server

L2TP

**VPN Server Addr:** Register to the address of VPN server

**VPN User Name:** L2TP VPN username

**VPN Password:** L2TP VPN password

**UDPTunnel:** use the UDP to visit VPN

**L2TP:** use the L2TP to visit VPN

**Enable VPN:** Enable the VPN server, you must choose UDP or L2TP type in advance

Notice: At the present, L2TP only support L2TP VPN server under Linux, UDP only support a private UDP VPN server.

## VPN Configuration

|   |                                      |                            |                                    |
|---|--------------------------------------|----------------------------|------------------------------------|
| VPN IP                                      |                                      |                            |                                    |
| 0.0.0.0                                     |                                      |                            |                                    |
| VPN Mode                                    |                                      |                            |                                    |
| <input checked="" type="radio"/> UDP Tunnel |                                      | <input type="radio"/> L2TP |                                    |
| <input type="checkbox"/> Enable VPN         |                                      |                            |                                    |
| UDP Tunnel                                  |                                      |                            |                                    |
| VPN Server Addr                             | <input type="text" value="0.0.0.0"/> | VPN Server Port            | <input type="text" value="80"/>    |
| Server Group ID                             | <input type="text" value="VPN"/>     | Server Area Code           | <input type="text" value="12345"/> |
| L2TP  |                                      |                            |                                    |
| VPN Server Addr                             | <input type="text"/>                 | VPN User Name              | <input type="text"/>               |
| VPN Password                                | <input type="text"/>                 |                            | <input type="text"/>               |
| <input type="button" value="APPLY"/>        |                                      |                            |                                    |

### 3.6 Dial-Peer dial rule setting

Please refer to [How to use dial rule](#) for detail.

## Dial-Peer

### Dial Peer Table

| Number | Destination     | Port | Mode | Alias           | Suffix    | Del Length |
|--------|-----------------|------|------|-----------------|-----------|------------|
| 2T     | 255.255.255.255 | 5060 | SIP  | del             | no suffix | 1          |
| 3T     | 0.0.0.0         | 5060 | SIP  | del             | no suffix | 1          |
| 123    | 0.0.0.0         | 5060 | SIP  | all:06332221015 | no suffix | 0          |
| 0T     | 0.0.0.0         | 5060 | SIP  | rep:86          | no suffix | 1          |
| 11     | 192.168.0.11    | 5060 | SIP  | no alias        | no suffix | 0          |

### Add Dial Peer

Phone Number

Destination (optional)

Port(optional)

Alias(optional)

Call Mode

Suffix(optional)

Delete Length (optional)

### Dial Peer Option

### 3.7 Config Manage

**Save Config:** Save current settings.

**Clear Config:** Restore to default settings.

**Backup Config:** Backup the config file, via point the right key of mouse-à save target as....-à will pop a save window, then type the config file name in the File name (the file type is text file)

**Update Configuration:** Update the current configuration through configuration files.

**Notice:**

Clearing config in admin mode, all settings restores to factory default; clear config in guest modem, all settings except sip, advance sip restore to factory default.

## Configuration

The screenshot displays a web interface titled "Configuration" with four distinct sections, each highlighted with a yellow header:

- Save Configuration:** Contains the instruction "Press the 'Save' button to save the configuration files !" and a "Save" button.
- Backup Configuration:** Contains the instruction "Save all Network and VoIP settings." and a link "Right Click here to Save as Config File (.txt)".
- Clear Configuration:** Contains the instruction "Press the 'Clear' button to Clear the configuration files !" and a "Clear" button.
- Update Configuration:** Contains a "Select file" label, a text input field, a "浏览..." (Browse...) button, a file type filter "(\*.txt)", and an "Update" button.

### 3.8 Update Firmware

#### 3.8.1 Update

##### Web Update:

Update the application or configuration files of the phone. The application document is .z format, and the configuration files is .cfg format.

Through clicking on the "browse" button to open the upgrade file or configuration file, then click on "Update" button. After the upgrade, The phone will automatically restart.

**Notice: When upgrading, WEB page can not be closed.**

##### FTP Update:

Upload/download the configure file with FTP or TFTP server, or download firmware from FTP or TFTP server

Back up configure file to your FTP/TFTP server.

configure use .cfg extension.

The Type includes two parts of config file export and config file import

Config file export: export the config file

Config file import: import the config file

The phone phone support FTP and TFTP auto update, the gateway will auto obtain the configure file from your update server if configured. To obtain the original configure file, you can use the FTP/TFTP back up as describe above. Configure file using module structure, user may remain the concerned modules and remove other modules. Put the configure file in the root directory of update server when finish editing.

## Update Configuration

Web Update

Select file  浏览... (\*.z,\*.txt,\*.au) Update

FTP Update

Server

Username

Password

File Name

Type Application update ▼

Protocol FTP ▼

APPLY

### 3.8.2 Auto Update

**Current Version:** the system will display the current version number

**Server Address:** FTP/TFTP server address

**Username:** FTP server user name

**Password:** FTP server password

**Config File Name:** The name of configuration file

**Config Encrypt Key:** The encrypt key of confirmation file

**Protocol Type:** The protocol type that used for upgrading

**Update Interval Time:** The interval time that the terminals search for new configuration file.

**Update Mode:** auto provision mode; Disable: not auto update, Update after reboot: auto update after reboot, Update at time interval: auto update after a certain time

Configure file version was in the <<VOIP CONFIG FILE>> Version 1.0007 and <GLOBLE CONFIG MODULE> CongFile Version

For instance:

Gateway original version is:

<<VOIP CONFIG FILE>>Version: 1.0000

<GLOBLE CONFIG MODULE> CongFile Version: 6

User may edit the configure file version to:

<<VOIP CONFIG FILE>>Version: 1.0007

<GLOBLE CONFIG MODULE> CongFile Version: 7



## Autoprovision

| Auto Update Setting                  |  |
|--------------------------------------|--|
| Current Config Version               | 2.0002                                 |
| Server Address                       | <input type="text" value="0.0.0"/>     |
| Username                             | <input type="text" value="user"/>      |
| Password                             | <input type="password" value="****"/>  |
| Config File Name                     | <input type="text"/>                   |
| Config Encrypt Key                   | <input type="text"/>                   |
| Protocol Type                        | <input type="button" value="FTP"/>     |
| Update Interval Time                 | <input type="text" value="1"/> Hour    |
| Update Mode                          | <input type="button" value="Disable"/> |
| <input type="button" value="APPLY"/> |  |

### 3.9 System Manage

#### 3.9.1 Account Manage

Set web access account or keypad password of NDC2210S.

## Account Configuration

| Set Keyboard Password  |   |            |       |      |       |         |  |
|--|---|------------|-------|------|-------|---------|--|
| Keyboard Password  | <input type="password" value="..."/> <input type="button" value="Set"/>     |            |       |      |       |         |  |
| User Set   |   |            |       |      |       |         |  |
| <table><thead><tr><th>User Name</th><th>User Level</th></tr></thead><tbody><tr><td>admin</td><td>Root</td></tr><tr><td>guest</td><td>General</td></tr></tbody></table> | User Name   | User Level | admin | Root | guest | General |  |
| User Name  | User Level  |            |       |      |       |         |  |
| admin  | Root  |            |       |      |       |         |  |
| guest  | General   |            |       |      |       |         |  |
| Add User   |   |            |       |      |       |         |  |
| User Name  | <input type="text"/>  |            |       |      |       |         |  |
| User Level   | <input type="button" value="Root"/>   |            |       |      |       |         |  |
| Password   | <input type="password"/>  |            |       |      |       |         |  |
| Confirm  | <input type="password"/>  |            |       |      |       |         |  |
| <input type="button" value="Submit"/>  |   |            |       |      |       |         |  |
| Account Option   |   |            |       |      |       |         |  |
| <input type="button" value="admin"/>   | <input type="button" value="Delete"/> <input type="button" value="Modify"/> |            |       |      |       |         |  |

#### 3.9.2 Syslog config

Set the system log

**Server IP:** set the syslog server address

**Server Port:** set the syslog server port

**MGR Log Level:** set the MGR log level  
**SIP Log Level:** set the SIP log level  
**IAX2 Log Level:** set the IAX2 log level  
Please click "apply" after setting

### Syslog Configuration

| Syslog Set                           |                                      |
|--------------------------------------|--------------------------------------|
| Server IP                            | <input type="text" value="0.0.0.0"/> |
| Server Port                          | <input type="text" value="514"/>     |
| MGR Log Level                        | <input type="text" value="None"/> ▼  |
| SIP Log Level                        | <input type="text" value="None"/> ▼  |
| IAX2 Log Level                       | <input type="text" value="None"/> ▼  |
| Enable Syslog                        | <input type="checkbox"/>             |
| <input type="button" value="APPLY"/> |                                      |

### 3.9.3 Phone Book

### Phone Book

| Phonebook Table                |  |                                       |                                    |
|--------------------------------|--|---------------------------------------|------------------------------------|
| Index                          | Name                                   | Number                                | Type                               |
| Add Phone Book                 |  |                                       |                                    |
| Name                           | <input type="text"/>                   |                                       | <input type="button" value="Add"/> |
| Number                         | <input type="text"/>                   |                                       |                                    |
| Ring Type                      | <input type="text" value="Default"/> ▼ |                                       |                                    |
| Phone Book Option              |  |                                       |                                    |
| <input type="text" value="▼"/> | <input type="button" value="Delete"/>  | <input type="button" value="Modify"/> |                                    |

### 3.9.4 Time Set

This page layout is the setting of time of phone.

**Server:** type the IP address of time server

**Timezone:** select correct time zone in list box

**Timeout:** longest response time for SNTP

**Daylight:** daylight saving time

**SNTP:** select SNTP server

**12 Hours Format:** select 12 hours format

**Manual Config:** time setting

## SNTP Configuration

| SNTP Time Set                        |  |
|--------------------------------------|--|
| Server                               | <input type="text" value="209.81.9.7"/>                                  |
| Time Zone                            | <input type="text" value="(GMT-07:00)Mountain Time(U.S. &amp; Canada)"/> |
| Time Out                             | <input type="text" value="21600"/> (seconds)                             |
| 12 Hours Systems                     | <input type="checkbox"/>   |
| SNTP                                 | <input checked="" type="checkbox"/>                                      |
| <input type="button" value="APPLY"/> |  |

  

| Daylight Timeset                     |                                     |                                      |
|--------------------------------------|-------------------------------------|--------------------------------------|
| Enable Daylight                      | <input type="checkbox"/>            |                                      |
| Time shift (minutes)                 | <input type="text" value="60"/>     |                                      |
| Time Zone                            | Start Date                          | End Date                             |
| Month                                | <input type="text" value="March"/>  | <input type="text" value="October"/> |
| Week                                 | <input type="text" value="5"/>      | <input type="text" value="5"/>       |
| Day                                  | <input type="text" value="Sunday"/> | <input type="text" value="Sunday"/>  |
| Hour                                 | <input type="text" value="2"/>      | <input type="text" value="2"/>       |
| Minute                               | <input type="text" value="0"/>      | <input type="text" value="0"/>       |
| <input type="button" value="APPLY"/> |                                     |                                      |

### 3.9.5 MMI SET

Set the greeting information on LCD.

## MMI Configuration

| Greeting Message Set                 |   |
|--------------------------------------|---|
| Greeting Message                     | <input type="text" value="VOIP PHONE"/> |
| <input type="button" value="APPLY"/> |   |

### 3.9.6 Logout & Reboot

Logout: logout the Web entry.

Reboot Phone: logout the entry, and reboot the phone. When user modifies any config of the phone, it will take effect after being rebooted, you can enter into this layout and click "Reboot". And the phone will be rebooted automatically.

**Note:** Reboot IP phone, some setting needs to reboot to make it works. Please always save config before reboot, otherwise the setting will return to previous setting.

# Logout & Reboot System

Logout

Press the "Logout" button to Logout Phone !

Logout

Reboot Phone

Press the "Reboot" button to reboot Phone !

Reboot

## 4. Operating Method for Dialing

### 4.1 How to dial IP Phone

You can make a call after being made a proper setting on your phone. Please confirm whether all the net wires are connected correctly.

If you want to make a call, you can make it after dialing the number and then pressing "#".

You can find IP address through the menu.

Modifying the IP address of the computer, and making it the same net as NDC2210S.

Inputting the IP address of NDC2210S in the browser, and then you can visit the setting layout of NDC2210S after pressing the Enter key; super user account is admin/admin; common user account is guest/guest.

### 4.2 Set the phone being connected to server

#### 4.2.1 Set the WAN interface

The connection ways of entering the Network à WAN Config layout phone of the net port:

NDC2210S could be connected to Internet by using the static IP, DHCP IP, or PPPoE dialing.

As for the specific configuration please see 3.3.1 WAN Config.

#### 4.2.2 SIP setting:

### SIP Configuration

**SIP Line Select**

SIP 1

---

**Basic Setting**

|                 |                                     |                            |  |
|-----------------|-------------------------------------|----------------------------|--|
| Register Status | Registered                          | Display Name               | <input type="text"/>                     |
| Server Address  | 61.156.234.90                       | Proxy Server Address       | <input type="text"/>                     |
| Server Port     | 5060                                | Proxy Server Port          | <input type="text"/>                     |
| Account Name    | 6009                                | Proxy Username             | <input type="text"/>                     |
| Password        | ●●●●●●●●                            | Proxy Password             | <input type="text"/>                     |
| Phone Number    | 6009                                | Domain Realm               | <input type="text"/>                     |
| Enable Register | <input checked="" type="checkbox"/> | Message Waiting Indication | Disable <input type="button" value="v"/> |

Enter into the **VoIP à SIP Config** to set the layout config and sip account information:

**Register Server Addr:** Register address of public SIP server

**Register Server Port:** Register port of public SIP server. **default port is 5060**

**Register Username:** Username of your SIP account (Always the same as the phone number)

**Register Password:** Password of your SIP account.

**Phone Number:** Phone number of your SIP account

----choose Enable Register;

You can dial VoIP phone when the WAN interface and IAX protocol are being set correctly.

**Message Waiting Indication:** Set Disable/Enable message waiting through pull-down menu, including:

**Disable:** MWI is disabled, even if received NOTIFY message from the server indicating new voicemail, phone will not prompt.;

**Enable (Subscribe):** MWI is enabled and SUBSCRIBE will be sent, if the server sends NOTIFY message indicating new voicemail received, MWI LED will blink to prompt.

**Enable(No Subscribe):** MWI is enabled, but phone will not send SUBSCRIBE, if the server sends NOTIFY message indicating new voicemail received, MWI LED will also blink to prompt.

SIP1 Subscribe will still be used in config file: config option "0" means "Disable", "1" means "Enable(Subscribe)", "2" means "Enable (No Subscribe)".

### 4.2.3 IAX setting

#### IAX2 Configuration

| IAX2                                 |                          |
|--------------------------------------|--------------------------|
| Register Status                      | Unregistered             |
| IAX2 Server Addr                     | <input type="text"/>     |
| IAX2 Server Port                     | 4569                     |
| Account Name                         | <input type="text"/>     |
| Account Password                     | <input type="text"/>     |
| Phone Number                         | <input type="text"/>     |
| Local Port                           | 4569                     |
| Voice Mail Number                    | 0                        |
| Voice Mail Text                      | mail                     |
| Echo Test Number                     | 1                        |
| Echo Test Text                       | echo                     |
| Refresh Time                         | 60 Seconds               |
| Enable Register                      | <input type="checkbox"/> |
| Enable G.729                         | <input type="checkbox"/> |
| IAX2(Default Protocol)               | <input type="checkbox"/> |
| <input type="button" value="APPLY"/> |                          |

**IAX Server Addr:** Register address of public IAX server

**IAX Server Port:** Register port of public IAX server, default port is 4569  
**Account Name:** Username of your SIP account (Always the same as the phone number)

**Account Password:** Password of your IAX account.

**Local port:** Signal port of local, default port is 4569

**Phone Number:** Phone number of your IAX account

----choose Enable Register;

----if you use IAX account to make a call, please choose IAX (Default Protocol), if you fail to choose it, then you can use SIP account to make a call again.

----if you use G..729 to arrange it, please choose Enable G..729

You can dial VoIP phone when the WAN interface and IAX protocol are being set correctly.

**Note:** please choose Save Config in the Config Manage after setting the information, or the existing setting information will be failed after rebooting.

#### 4.3 How to use the dial rule?

NDC2210S provide flexible dial rule, with different dial-rule configure, user can easily implement the following function:

----Replace, delete or add prefix of the dial number.

----Make direct IP to IP call

----Place the call to different servers according the prefix.

You can click "Add" to add a new dial rule. Below is the detail setting of the dial-rule:

**Phone Number:** The Number suit for this dial rule, can be set as full match or prefix match. Full match means that if the number user dialed is completely the same as this number, the call will use this dial-rule. Prefix match means that if prefix of the number that the user dials is the same as the prefix, the call will use this dial-rule, to distinguish from the full match case, you need to add "T" after the prefix number in the phone number setting.

**Call Mode:** support SIP.

**Destination (optional):** call destination, can be IP or domain. Default is 0.0.0.0, in this case the call will be routed to the Public SIP server. If you set the destination to 255.255.255.255, then the call will be routed to the private SIP server. Also you can key other address here to make direct IP calls

**Port (optional):** Configure the port of the destination, default is 5060 in SIP

**Alias (optional):** Set up the Alias. We support four Alias as below. Alias need to co-work with the *Del Length*:

add:xxx, add prefix to the phone number, can set to reduce the dial length.

all: xxx, replace the phone number with the xxx, can use as speed dial function.

Del: delete the first N numbers. N is set in the *Del Length*

rep:xxx, replace the first N numbers. N is set in the *Del Length*. For Example: User wants to place a call 86633-8215555, then can set the *phone number* in the dial rule as 0633T, and set the *Alias* as rep: 86633, and set the *Del Length* to 4. Then all calls begin with 0633 will be changed to 86633 xxxxxxxx.

**Suffix (optional):** Configure suffix, show no suffix if not set

## Dial-Peer

| Dial Peer Table |                 |      |      |                 |           |            |
|-----------------|-----------------|------|------|-----------------|-----------|------------|
| Number          | Destination     | Port | Mode | Alias           | Suffix    | Del Length |
| 2T              | 255.255.255.255 | 5060 | SIP  | del             | no suffix | 1          |
| 3T              | 0.0.0.0         | 5060 | SIP  | del             | no suffix | 1          |
| 123             | 0.0.0.0         | 5060 | SIP  | all:06332221015 | no suffix | 0          |
| 0T              | 0.0.0.0         | 5060 | SIP  | rep:86          | no suffix | 1          |
| 11              | 192.168.0.11    | 5060 | SIP  | no alias        | no suffix | 0          |

  

| Add Dial Peer                         |                                      |
|---------------------------------------|--------------------------------------|
| Phone Number                          | <input type="text"/>                 |
| Destination (optional)                | <input type="text"/>                 |
| Port(optional)                        | <input type="text"/>                 |
| Alias(optional)                       | <input type="text"/>                 |
| Call Mode                             | SIP <input type="button" value="v"/> |
| Suffix(optional)                      | <input type="text"/>                 |
| Delete Length (optional)              | <input type="text"/>                 |
| <input type="button" value="Submit"/> |                                      |

  

| Dial Peer Option                    |   |
|-------------------------------------|---|
| 2T <input type="button" value="v"/> | <input type="button" value="Delete"/> <input type="button" value="Modify"/> |

Instance:

**2T rule:** If the call starts with 2, the first 2 will be deleted, and the rest number will be sent to private SIP server.

**3T rule:** If the call starts with 3, the first 3 will be deleted, and the rest number will be sent to public SIP server.

**123 rule:** Dial 123 and will send 06332221015 to your server. It is used as speed dial function.

**0T rule:** If the call begins with 0, the first 0 will be replaced by 86. It means that if you dial 06332221015 and AG-188 will send 866332221015 to your server.

**11 rule:** when you dial 11, the call will send to 192.168.0.11, suit for LAN application without set up a sip server.

### 4.4 Voice mail

When there are new messages, the MWI will give out prompt according to the options of "Message Waiting Indication" in web page layout.

If "Disable" is selected, MWI function is disabled, even if received NOTIFY message from the server indicating new voicemail, phone will not prompt;

If "Enable (Subscribe)" is selected, MWI function is enabled and Subscribe will be sent, if the server sends NOTIFY message indicating new voicemail received,



MWI LED will blink to prompt;

If "Enable (No Subscribe)" is selected, MWI is enabled, but Subscribe will not be sent, if the server sends NOTIFY message indicating new voicemail received, MWI LED will also blink to prompt. After the voicemails are picked-up, MWI LED will stop blinking.

## Compliance Statement

### FCC Caution:

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

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.

### FCC Radiation Exposure Statement:

The device complies with FCC radiation exposure limits set forth for uncontrolled environment. The base should be installed and operated with minimum distance 20 cm between the radiator & your body. This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.

### Note:

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 radiocommunications. 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.