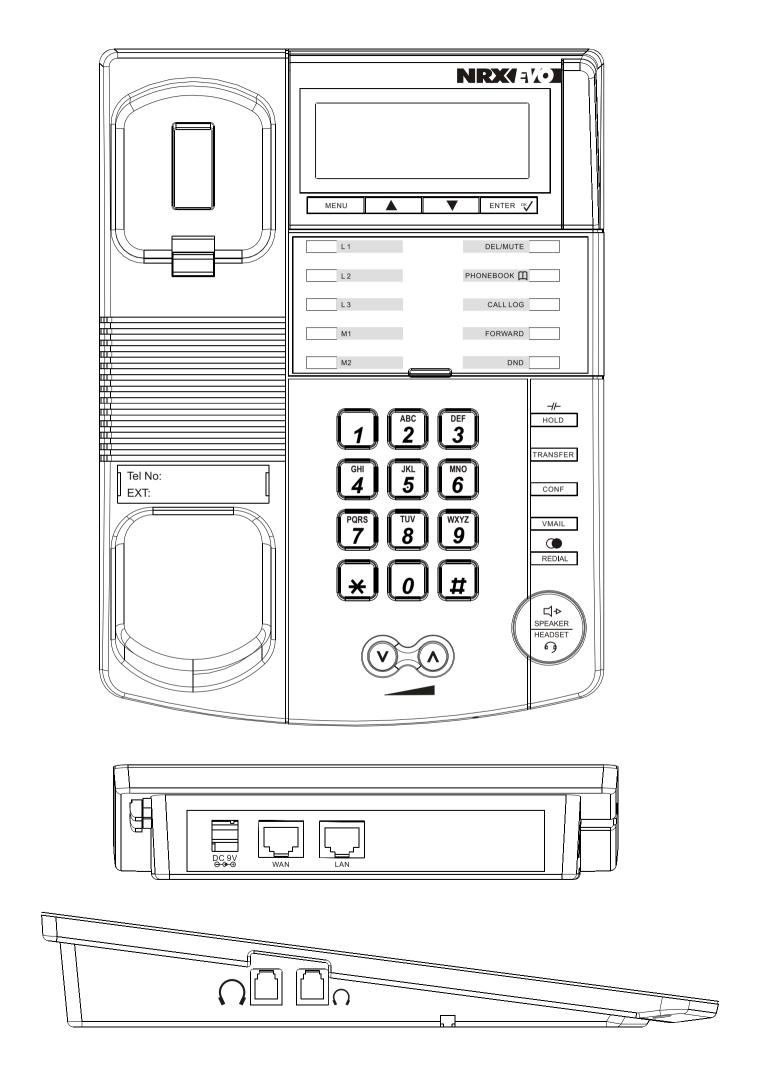
NRXAD



NRX EVOIP 1 USER GUIDE

ecom.com WWW. NTX-tel



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Quick Start Guide

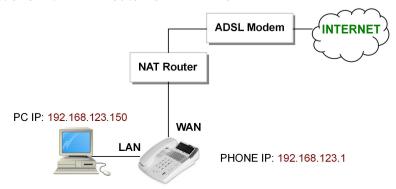
Step 1: Broadband (ADSL/Cable Modem) Connections for NRX EVolP 1

1. Connect one end of handset cord to the telephone handset and the other end to the handset socket located on the underside of the telephone.

NOTE: You will need to remove the wall mount bracket to access this socket.

- 2. Connect the RJ45 WAN port to a ADSL NAT Router.
- 3. Connect the RJ45 LAN port to Notebook PC LAN port using a Category 5 LAN cable.
- 4. Connect DC Power Adaptor. The LCD Panel will start showing 'Loading Program!'
- 5. The LCD panel will show Date, Time and No service without SIP registration, or <phone number> after successful SIP registration.
- 6. Pick up the phone, and the LCD panel will show IP Dialing.. 1. Please hang up.
- 7. Press MENU, 4,5 from the keypad to check the LAN IP address (default: 192.168.123.1) for NRX EVoiP telephone.
- 8. The **MENU** key is used for menu/escape, the **UP**, **DOWN** and **ENTER** keys for navigating the menu system, phonebook or Call log.

Figure A. ADSL Connections with NAT Router for NRX EVoIP 1



Quick Start Guide

Step 2: Configurations from Keypad

Keypad Settings for Password and SIP Account:

- Press **MENU**, **7**, **4** and the LCD panel will show Password:
- Enter password (Default: admin) i.e. press 2 two times, 3 two times, 6 two times, 4 four times, 6 three times, then press **ENTER**. Note: The password must be entered first to access the other settings.
- Press **MENU**, **5**, **ENTER** to configure First realm for SIP account.

Keypad Settings for IP Status:

- IP Addresses for WAN and LAN Ports: Press MENU, 4, 5
- WAN port Status : Press MENU, 4, 1, 1 2
- 3. LAN port Status : Press MENU , 4 , 2

Example: Network Settings for Fixed IP by Keypad (M=MENU; E=ENTER)

For Network Settings, there are three choices for IP type in the WAN Settings; Fixed IP, DHCP, and PPPoE. For Fixed IP, the user must key in a static fixed IP address, Subnet Mask, and Gateway IP with DNS Server IP address.

- 1 Fixed IP: Press M 4 E 1 E M M M to select Fixed IP and reboot.
- Set IP Address: Press M_4_E_2_1 and enter 192*168*101*112 then E_M_M_M_M to reboot. Set Subnet Mask: Press M_4_E_2_2 and enter 255*255*255*000 then E_M_M_M_M to reboot. 2.
- Set Gateway IP: Press M 4 E 2 3 and enter 192*168*101*001 then E M M M M M to reboot.
- 5. Set DNS Server IP address as the following page if necessary.
- Power the unit OFF and then ON again. 6.



Quick Start Guide

Step 3: Configurations from PC Web Browser

- Press **MENU**, 4,5 from keypad to get WAN and LAN IP addresses.
- Enter the IP address into your Web Browser i.e Given the IP address 192.168.123.1 for LAN port, enter http://192.168.123.1:9999 into the Web browser to display login page for Web configuration.
- 3. Enter username and password and click LOGIN.

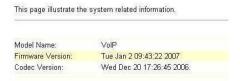
Default:

Username: admin Password: admin



5. The Web browser will show the following System Information page.

System Information



Quick Start Guide

6. You need to set up the following web configuration: Phone, Network WAN/LAN Settings, SIP Settings. Remember to submit, save and reboot for new configuration.

7. For WAN settings:

DHCP:

This is for Dynamic IP address and is commonly selected for WAN IP address when NAT is selected. Once selected, the IP address, Subnet mask, and Gateway IP will be automatically assigned. Remember to click Submit button to get effective.

PPPoE:

This is for ADSL IP address when PPPoE protocol is used. Once selected, the user needs to enter the PPPoE username and password, and the WAN port will get an IP automatically.

Bridge LAN mode:

The Bridge LAN mode can be selected when NAT is not needed. In this case, the WAN and LAN port are transparent as Ethernet Bridge Hub. Note that the embedded NAT is enabled at default.





Quick Start Guide

8. SIP Settings: You may configure up to 3 SIP registration accounts with the 3 realms in Service

Domain Settings.

Realm 1 (Default for 1st SIP registration account)

Active: ON
Display Name: 206
User Name: 206
Register Name: 206
Register Password: 1234

Proxy Server: 220.228.43.172:5070 Domain Server: 220.228.43.172:5070

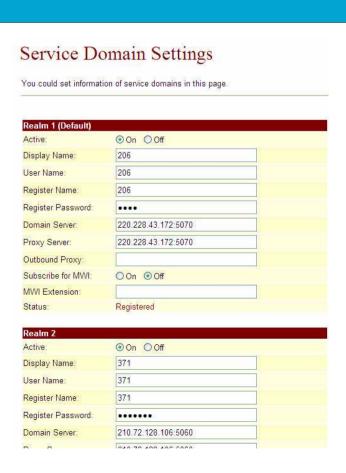
Outbound Proxy:

Realm 2 & 3 (for 2nd & 3rd SIP registration accounts)

Active: ON
Display Name: 371
User Name: 371
Register Name: 371
Register Password: 1234567

Proxy Server: 210.72.128.106:5060 Domain Server: 210.72.128.106:5060

Outbound Proxy:



Quick Start Guide

SIP Account Number Selections:

With 3 SIP account registered numbers, the NRX EVoIP telephone may receive all the incoming calls from the 3 SIP servers. Realm 1, 2, and 3 are used to configure the registered numbers for the 1st, 2nd, and 3rd SIP servers, respectively. The telephone may register different phone numbers to different SIP servers.

You can select the different lines by pressing LINE 1, LINE 2 or LINE 3 on the telephone. Line 1 is used at default. When one or more of the lines are registered the LED indicator will switch on.

9. The LCD will display the registered <phone number> after successful SIP registration to the 1st SIP server at default.

Step 4: Making Point-To-Point SIP Calls

- 1. Select one of the three SIP servers. Pick up the handset or press the 'Speaker/Headset' button, you should hear a dial tone.
- 2. Press 123456# to call the party with the number 123456 registered in the SIP server. Note # is used to send out the call immediately. In a moment, you should hear the ring back tone, and wait for the called party to answer.

Note: If you have difficulties in configuring your NRX EVoIP telephone, please refer to the last chapter for trouble shooting.



Introduction

1. Introduction

The NRX EVoIP is an IP Telephone with SIP Protocols for Voice over IP (VoIP) applications. Connecting to the Internet, the NRX EVoIP can make a VoIP call over the Internet. The NRX EVoIP provides one WAN port for Internet ADSL connections, one LAN port for PC connection, and two RJ11 connectors for optional Headset. With an embedded NAT/DHCP server, the unit can be easily configured for different network requirements by using a IE Web browser or the telephone keypad.

Note: The NRX EVoIP telephone requires an IP address, a subnet mask, and its gateway Router IP address to connect to Internet. These three are available from your Internet service provider. The NRX EVoiP can use PPPoE or DHCP to automatically get an assigned dynamic IP from the ITSP. Please refer to Section 8 Configurations by Web Browser for detailed information.

2. Features

The NRX EVoIP has the following features:

- Two LED Indicators: REG for registration, VMS for voice mail
- RJ45 x 2 for WAN and LAN ports + RJ11 x 2 for Headset ports
- Configurations by Web Browser and Telephone Keypad
- Embedded NAT/DHCP Server
- PPPoE/DHCP Client for Dynamic IP plus NAT, DNS, and DDNS Clients
- Support STUN server for NAT Traversal
- Support registrations for up to 3 SIP accounts
- Hot Line Mode

Standard Compliances

- Dial Plan Settings
- Speaker Phone, Voice Mail (VMS) for incoming IP calls
- Call Forward/Transfer/Waiting/Hold, and 3-Way Conference Call features
- Auto Configurations by TFTP, HTTP, or FTP server
- Remote Firmware Upgraded with HTTP or TFTP server by Web PC
- Direct IP/URL Dial without SIP Proxy or Dial number via SIP server
- Telephone features: Volume adjustment, Phone book, Redial
- Function Keys: Speed Dial, 3 SIP Registration Lines, DND & VMS
- Out-Band DTMF (RFC 2833) / In-Band DTMF / Send DTMF SIP Info

3. Standard Compliances

The NRX EVoIP Phone supports for the following standards:

- VoIP Protocol: IETF RFC3261 and RFC 2543 for SIP
- SIP Authentication: IETF RFC2069 and RFC 2617 for MD5
- Speech Codec: ITU-T G.711, G.723, G.729A/B, VAD and CNG
- Echo Cancellation: ITU-T G.165/168



Packing Contents

4. Packing Contents

Inside the NRX EVoIP box you should find:

- (1) One NRX EVoIP telephone (Including Handset & Handset Cord)
- (2) One AC to DC Power Adaptor (9~12VDC/1A)
- (3) One User Manual
- (4) One network cable

Please check if the packing is damaged or any component is missing. If so, please contact your distributor.

5. LED Indicators

On the top right hand side of the NRX EVoIP telephone, there are two LED indicators.

REG: "Red On" indicates successful SIP registration.

VMS: "Red Flashing" indicates there are voice messages for incoming VoIP calls.

To hear the message, press the Speaker Phone key then VMS function key.

Installation & SIP Configuration

6. Installations & SIP Configurations

- 1. Connect the NRX EVoIP RJ45 WAN port to NAT Router using a Category 5 LAN cable.
- 2. Connect RJ45 LAN port to your PC using a Category 5 LAN cable.
- 3. Connect RJ11 Headset port to a headset if available.
- 4. Connect DC Power Adaptor. The LCD Panel will start displaying "Loading Program" for about 5 seconds, and the LEDs will be ON and start initializations.
- The LCD panel will show Date, Time and No service without SIP registration, or <phone number> after successful SIP registration.
- 6. Pick up the phone, and you will hear a dial tone and the LCD panel will show IP Dialing.. 1. If you hear a busy tone, please check if the WAN port is connected and hang up.
- 7. Press Menu_7_4, and test (i.e. press 8 two times_3 three times_7 five times_8 two times_enter) to enter password first. Press Menu_4_5 from keypad to read WAN and LAN IP addresses. The default LAN IP address is 192.168.123.1.
- 8. Go to Chapter 8 and you may enter this IP address in IE Web browser for web configurations. Configure and register the NRX EVoIP into your SIP server. There are more details and examples for VoIP application of SIP registrations in Chapter 10.
- 9. If NRX EVoIP has successfully registered in the SIP server, the LED REG will turn ON.
- 10. Pick up the phone, and press 123456# to call the party with the number 123456 registered in the SIP server. Note that # will dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapsed. In a moment, you should hear a ring back tone, and wait for answer.

7. Default Reset by Using Keypads

Press Menu / 7.Administrator / 3.Default setting / 1.Load default by using Menu and arrow keys to



Configuration by Web Browser

reset back to factory defaults, and the LCD panel will show Set Default.... In 5 seconds, the LCD will start showing Loading Program and System Initialized. The NRX EVoIP will reset to default and restart.

Press Menu / 7.Administrator / 7.Restart to reboot the NRX EVoIP. In 5 seconds, the LCD will start showing Loading Program and System Initialized.

8. Configurations by Web Browser

You may enter the IP address from a Web browser to configure the NRX EVoIP telephone.

- 1. Enter the IP address into the browser address window followed by **:9999** For example http://192.168.123.1:9999
- 2. The following Web page should be displayed on PC. If you have difficulties accessing the Web page from the PC Web browser, please refer to Chapter 9 for trouble shooting.



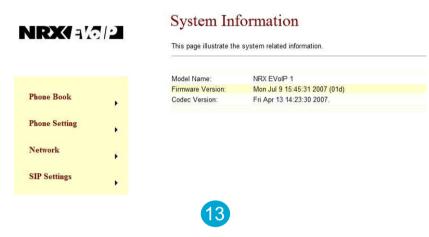
8.2. Please enter the username and password into the blank fields. The default settings are:

Username: admin Password: admin

8.3. Click the "Login" button to enter the System information page for web configuration. Whenever you change the setting in each Web page, you need to click the "Submit" button on the page, and then click "Save Change" on the tool bar followed by the 'Save' Button confirmation to save into the phones non-volatile memory.

System Information

- 8.4. The system information shows firmware version, Codec, etc.
- 8.5. You can click the buttons, for example Phone Book, at the left hand side to configure the NRX EVoIP telephone.



Configuration by Web Browser

Phone Book

Phone Book Settings:

- 8.6. You can add/delete names up to maximum 140 entries in Phone book list.
- 8.7. To add a phone name, you need to enter the position, the name, and the phone URL. When you have finished, click the "Add Phone" button.
- 8.8. To delete a phone name, select the record then click "Delete Selected" button.
- 8.9. To delete all the phone names, please click "Delete All" button.
- 8.10. To dial from the phonebook, the records can either be accessed by pressing the 'Phonebook' button and then scrolling through the records until you have the required record, then pressing the 'Speakerphone/ Headset' button or lifting the handset. Alternatively, the records can be speed dialed from the keypad as follows:

Example 1: Name: 333, URL: 192.168.1.100

Press 333# on telephone, the phone at 192.168.1.100 will start ringing.

Example 2: Name: 201, URL: 14081234567

Press 201# on telephone, the phone will call the registered number 14081234567.

Example 3: Name: 301, URL: 192.168.1.100:5062

Press 301# on telephone, the phone at 192.168.1.100 with port 5062.

Example 4: No Name 401 in phone book.

Press 401# on telephone, the phone will call 401.

NOTE: The # key acts as a send button when pressed after a number has been entered, If the # key is not pressed, the number is dialled automatically after five seconds.

NRXAMP



Phone Book

You could add/delete items in current phone book Phone Book Page: page 1 ▼ Phone Name Select 0 1 Tom tom@voip.sipserver.com 2 333 192.168.1.100 3 6 8 Delete Selected Delete All Reset Add New Phone Position: $(0 \sim 139)$ Name: URL: Add Phone Reset



Configuration by Web Browser

Speed Dial Phone List

8.11. You can add/delete Speed Dial number up to maximum 10 entries in Speed Dial Phone List.

8.12. If you need to add a phone number into the Speed Dial list, you need to enter the position, the name, and the URL. Click the "Add Phone" button to finish the setting.

8.13. To delete a phone number, please select the phone number you want to delete then click "Delete Selected" button. If you want to delete all phone numbers, please click "Delete All" button

Example 1: Press 1# from keypad to Speed Dial the phone number 2 immediately.

Example 2: Press M1 or M2 function key will Speed Dial the phone number 1 or 2 immediately.

NRX(4/6/2



Speed Dial Phone List

You could set the speed dial phones in this page.

Add Phone Reset

Phone	Name	URL	Select
0			
1	Kevin	01234567890	
2			
3			
4			
5			
6			
7			
8			
9			
Delete S		Delete All Reset	
	(0	~9)	
lame:			
IRL:	7.1		

Phone Setting

Call Forward Setting:

8.14. You can select the forward mode and enter the forward URL.

All Forward: All incoming call will forward to the URL you choose.

Busy Forward: The incoming call will forward to the URL when you are busy and on the phone.

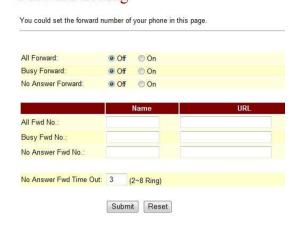
No Answer Forward: The incoming call will forward to the URL when no answer.

8.15. You need to set the Time Out ring which will initiate No-Answer forwarding to the number you choose. When you finished the setting, please click the "Submit" button.





Forward Setting





Configuration by Web Browser

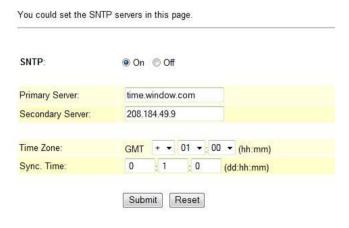
SNTP Settings

8.16. You can setup the primary and the second SNTP Server IP Addresses to get the date/time information. You may also set the Time Zone, and how often the phone synchronizes. When you finished the setting, please click the "Submit" button.





SNTP Settings



Volume Settings

8.17. You can setup the Handset Volume/Gain, Ringer Volume, and Speaker Volume/Gain in this page. Handset/Speaker Volume is to set the volume hearing from the handset/Speaker. Handset Gain is to set the volume send out from the handset. Speaker Gain is to set the volume send out from the microphone at the front of the NRX EVoIP.

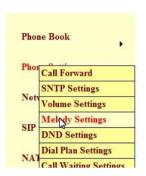
Melody Settings

8.18. You can select the ringer melody type for the incoming call.









Volume Setting

You could set the volume of your phone in this page.

Handset Volume: 10 (0~10)

Speaker Volume: 10 (0~10)

Ringer Volume: 6 (0~10)

Handset Gain: 10 (0~10)

Speaker Gain: 9 (0~10)

Submit Reset

Ringer Settings



You could set your favorite ringer in this page



Configuration by Web Browser

DND Settings

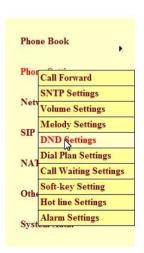
8.19. You can setup the DND (Do Not Disturb) to keep the phone silent. You can choose DND Always or DND period.

DND Always: All incoming call will be blocked until the feature is disabled.

DND Period: Set a time period and the phone will be blocked during the time period. If the "From" time is larger than the "To" time, the block time will from Day 1 to Day 2.

When you finished the setting, please click the Submit button.





DND Setting

You could set the do not disturb period of your phone in this page.

DND Always: On Off

DND Period: On Off

From: OO ON (hh:mm)

To: OO (hh:mm)

Dial Plan Settings

8.20. Dial plan and auto dial timer settings can be set on this page. The dial plan allows you to map the dialing into an easy-to-remember phone number system. The auto dial timer specifies the elapse time between the dialing digits. When Drop prefix is ON and the dialing prefix is matched, the prefix will be dropped and replaced by the rule digits and followed by the rest of dialing digits. When Drop prefix is OFF and the dialing prefix is matched, the rule digits will be added before the dialing digits in accord with the settings.

Example 1: Drop Prefix: No, Replace rule 1: 002, 8613+8662

- a) Pressing 8613xxx will result in dialing out 002+8613+xxx.
- b) Pressing 8662xxx will result in dialing out 002+8662+xxx.

Example 2: Drop Prefix: Yes, Replace rule 2: 006, 002+003+004+005+007+009

- a) Pressing 002xxx will result in dialing out 006+xxx.
- b) Pressing 003xxxx will result in dialing out 006+xxxx.

Example 3: Drop Prefix: No, Replace rule 3: 009, 12

a) Pressing 12xxx will result in dialing out 009+12xxx.

Example 4: Drop Prefix: No, Replace rule 4: 007, 5xxx+35xx+21xx

- a) Pressing 5xxx will result in dialing out 007+5xxx.
- b) Pressing 534 will result in dialing out 534 (not matched for the rest 3 digits).
- c) Pressing 35xx will result in dialing out 007+35xx.
- d) Pressing 356 will result in dialing out 356 (not matched for the rest 2 digits).
- e) Pressing 35668 will result in dialing out 35668 (not matched for the rest 2 digits).



Configuration by Web Browser

Example 5: Dial Now: *xx+#xx+11x+xxxxxxxx

- a) Pressing *00, *01, *02 .. *99 will result in dialing out the same *xx immediately.
- b) Pressing #00, #01, #02 .. #99 will result in dialing out the same #xx immediately.
- c) Pressing 110, 111, .. 119 will result in dialing out 11x immediately.
- d) Pressing 12345678 (8 digits) will result in dialing out 12345678 immediately. This implies that phone numbers with 9 or more digits are prohibited.
- 8.22. Auto Dial Timer: The inter-digit timer. Default is 5 seconds.
- 8.23. When you finish the setting, please click the Submit button.
- 8.24. Click the Save button. The changes you have made will be saved into the system and the system will reboot automatically.





Dial Plan

You could the set the dial plan in this page Drop prefix : O Yes O No Replace rule 1: 002 + 8613+8662 Drop prefix : Yes

 No 006 + 002+003+004+007+009 Drop prefix: Replace rule 3: 009 Drop prefix : Yes No Replace rule 4: 007 + 5xxx+35xx+21xx Dial now: Auto Dial Time: 5 (3~9 sec) Use # as send key:

Yes
No Use * for IP dialing:

Yes No Submit Reset

Call Waiting Setting:

8.25. When call waiting is set ON, you will hear an interrupt tone over the phone to remind that there is an incoming call from the third party.



Soft Key Setting:

8.26. You can configure the VMS key (Voice Mail) setting to work with an IP PBX on this page.



Configuration by Web Browser

When registered with an IP PBX with an incoming voice message, the LED VMS will start flashing. To hear

the message, press the Speaker Phone key or lift the handset then press the VMS function key.

Hot Line Settings:

8.29. You can enable/disable and set a Hot Line phone number in this page.

8.30. When Hot Line mode is enabled, you just pick up the phone and the NRX EVoIP phone will call the Hot line number immediately. The default for Hot Line mode is disabled.

8.31. Hot-Line Mode is very convenient for IP calling to a Public Switching Telephone Network (PSTN) number through FXO Gateway.

Alarm Settings:

8.32. You can configure the Alarm setting on this page.



Network

8.33. You can check the Network status, and configure the WAN, LAN, DDNS, and VLAN settings in this section.

Network Status:

8.34. You can check and show the current Network setting in this page.





Network Status

This page shows current status of network interfaces of the system.

DHCP Client	
192.168.1.5	
255.255.255.0	
192.168.1.1	
168.95.192.1	
168.95.1.1	
	192.168.1.5 255.255.255.0 192.168.1.1 168.95.192.1

nterface 1		
Гуре:	DHCP Server	
P:	192.168.123.1	
Mask:	255.255.255.0	
Gateway:	192.168.123.1	
DNS Server 1:	168.95.192.1	
ONS Server 2:	168.95.1.1	



Configuration by Web Browser

WAN Settings:

Submit button.

8.35. The WAN setting is used to configure the Ethernet port connects to the ADSL Modem/Router. 8.36. The default setting is NAT mode for IP Phone, and this enables the embedded NAT router between the LAN port and PC port. You may change to Bridge Mode if you need NOT use the embedded NAT router. When setting to Bridge Mode, the WAN and the LAN ports will be bridged. 8.37. There are three selections for WAN: Fixed IP. DHCP Client, and PPPoE modes. This WAN setting is for the LAN port when set in NAT mode. The WAN default is at DHCP Client Mode. 8.38. For Fix IP Mode, please make sure the IP address. Net Mask, Gateway, and DNS settings are suitable in your current network environment. 8.39. For PPPoE Mode, you have to enter correct username and password to get the IP address from your Internet Service Provider. 8.40. When you finished the setting, please click the





WAN Settings

You could configure the WAN settings in this page.

IP Type:	○ Fixed IP ● DHCP Client ○ PPPoE
IP:	192.168.1.5
Mask:	255.255.255.0
Gateway:	192.168.1.1
DNS Server1:	168.95.192.1
DNS Server2:	168.95.1.1
MAC:	0009f37734e8
Host Name:	NRX EVoIP 1

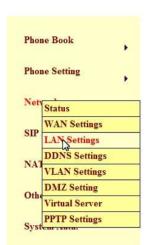
Submit Reset

LAN Settings:

8.41. The default IP address is 192.168.123.1 for the NRX EVoIP phone, with Net Mask 255.255.255.0., and DHCP Server enabled. The IP addresses for DHCP are from 150 to 200.

8.42. Connect your PC to the PC port, set your PC as DHCP Client mode, and then the PC will get an IP address from the NRX EVoIP automatically.

8.43. When you finished the setting, please click the Submit button.



NRX





Configuration by Web Browser

DDNS Settings:

8.44. You can configure the DDNS setting on this page. You need to have the DDNS account and input the informations properly. You need a DDNS account with a public IP address then others can call you via the DDNS account. Most of the VoIP applications work with a SIP Proxy Server as well as DDNS Server. When you finished the setting, please click the Submit button.

DDNS Settings

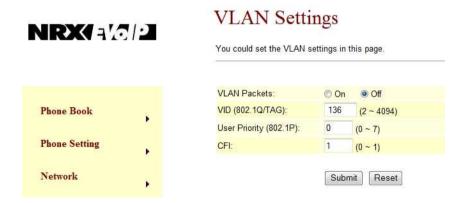
DDNS:	⊙ On ○ Off
Host Name:	voipphone.dyndns.org
User Name:	test
Password:	••••
E-mail Address:	
DDNS Server:	
DDNS Server List:	members.dyndns.org
Туре:	dyndns
Wild Card:	on 💌
BACKMX:	○ On
Off Line:	○On ⊙Off

VLAN Settings:

- 8.45. There are two parts for VLAN settings. One is to set for VoIP packets related to the NRX EVoIP, and the other is for the VLAN setting in the NAT Mode.
- 8.46. There are two kinds of destination packets coming to the WAN port, one is VoIP packets for NRX EVoIP, and the other will go through the WAN port to the LAN port.
- 8.47. VLAN Packets: If you enable VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be checked with the IP Address and the VID.
- 8.48. VID: Please set your VID in accordance with your service provider.
- 8.49. User Priority: Defines user priority with eight (2³) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually, this will be defined by your service provider.
- 8.50.CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.
- 8.51 When you enable the VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the NRX EVoIP IP address and the same VID will be accepted by the phone. If the incoming packets with the NRX EVoIP IP address but different VID then the packets will be discarded by phone. The Other incoming packets with different IP address will go through the WAN port to the LAN port.

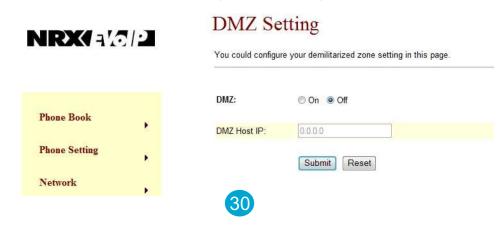


Configuration by Web Browser



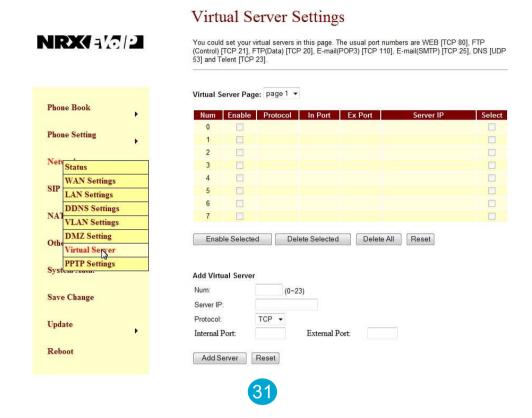
DMZ Setting:

8.52. The DMZ can be enabled/disabled and configured in this page.



Virtual Server:

8.53. The Virtual Server IP and Port numbers can be configured in this page.



Configuration by Web Browser

PPTP Setting:

8.54. The PPTP Server can be set ON/OFF in this page.



SIP Settings:

Service Domain Settings

8.55. You can setup the Service Domain, Port Settings, Codec Settings, RTP Setting, RPort Setting and Other Settings for SIP Proxy Server registrations on this page.

8.56. You may register up to three SIP Servers for three Realms in the NRX EVoIP telephone. You can receive the incoming calls from all the three SIP Servers. For outgoing calls, you may select the registration SIP server by pressing the appropriate line button, and then call the associated registration phone number.

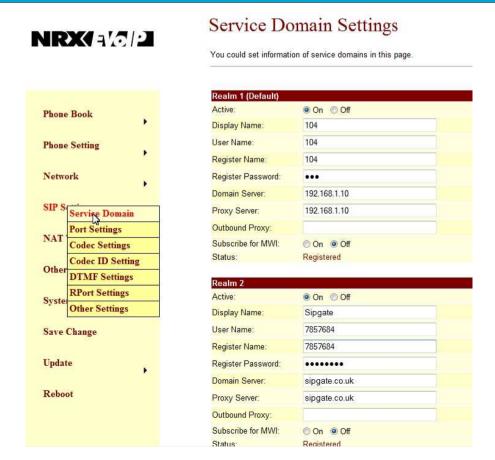
8.57. Click "Active" ON to enable the Service Domain, then enter the following items:

- 8.58. Display Name: enter the name you want to display.
- 8.59. User Name: enter the User Name given by your ITSP.
- 8.60. Register Name: enter the Register Name given by your ITSP.
- 8.61. Register Password: enter the Register Password given by your ITSP.
- 8.62. Domain Server: enter the Domain Server given by your ITSP.
- 8.63. Proxy Server: enter the Proxy Server given by your ITSP.
- 8.64. Outbound Proxy: enter the Outbound Proxy of ITSP. If not provided, you may skip this.
- 8.65. When it shows "Registered" in the Register Status, it indicates a successful registration to the ITSP, and the "REG" LED will turn ON. The NRX EVoIP is then ready for VoIP call.
- 8.66. After you finished the setting, please click the "Submit" button.

NOTE: The User Name and Register Name are also referred to as SIP ID. Many voice over IP service providers will only provide a SIP ID, if this is the case then enter this as both the User Name and Register Name. It is also recommended that when registering with a voice over IP provider that the MWI indication is turned off.



Configuration by Web Browser



Port Settings:

8.67. The SIP Port and RTP Port numbers are default at 5060 and 60000, respectively. The RTP port number must be even number. If you have more than one VoIP phones under the same NAT router, it is recommended that different RTP port numbers be assigned to each of IP Phones.



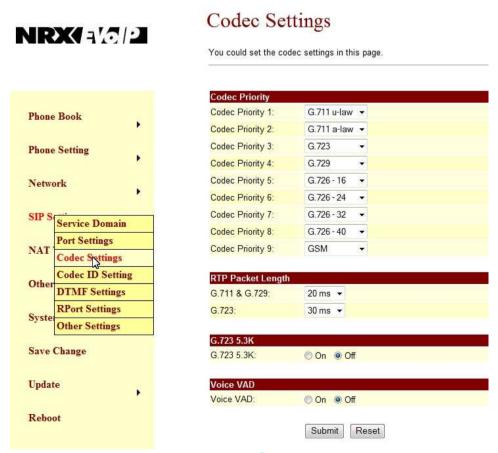
Codec Settings:

8.68. You can setup the Codec priority, RTP packet length, and VAD function in this page. When you finished the setting, please click the Submit button.

NOTE: It is suggested that you do not alter these settings from the default telephone setup unless you have technical expertise in this area.

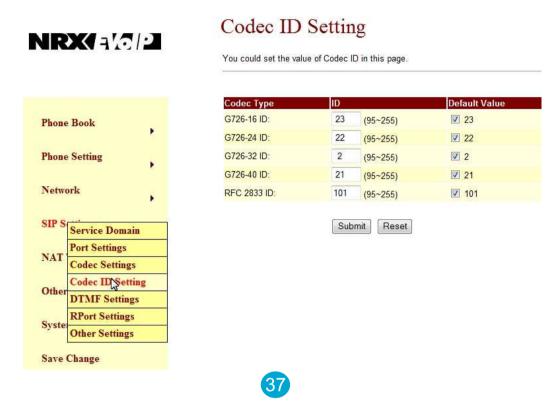


Configuration by Web Browser



Codec ID Settings:

8.69. You can set the Codec ID to meet the other device's requirement. When you finished the setting, please click the Submit button.



Configuration by Web Browser

DTMF Settings:

8.70. You can setup the options for DTMF function in this page. The options include RFC2833 (Outband DTMF), Inband DTMF, and Send DTMF SIP info. If you are making two-stage callings for extension to PSTN, you might need to select RFC2833 DTMF option.



RPort Function:

8.71. You can enable/disable the RPort in this page. To change this setting, please follow your ISP information. When you finished the setting, please click the Submit button.

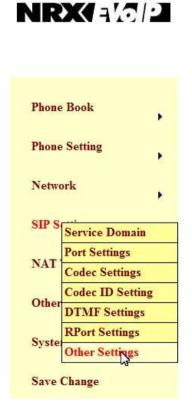


Other Settings:

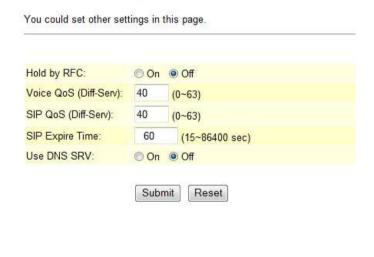
8.72. You can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. Higher value for voice packets will get higher priority to the Internet. It is required that all the Internet routes are with the QoS function for voice priority.



Configuration by Web Browser



Other Settings

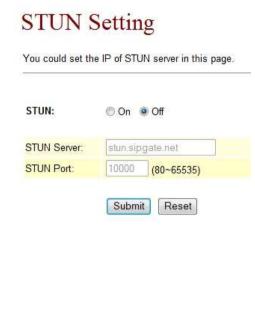


NAT Transversal:

STUN Setting

8.73. You can Enable/Disable and configure STUN Server IP address in this page. This function helps IP Phone working properly behind NAT. To change these settings please follow your ISP information. When you finished the setting, please click the Submit button.







Configuration by Web Browser

Auto Configuration Setting:

8.75. Auto Configuration function can be used to download a configuration file stored on a TFTP, HTTP, or FTP server. This function must work in conjunction with the Auto Config Server. After enabling the function, please click the "Submit" button. Remember to click "Save" in the Save Change section. The IP Phone will then reboot and automatically download the configurations from the corresponding server. Note that the TFTP download works only for a public IP address.



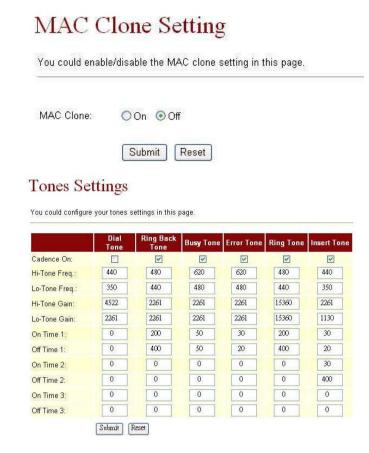


MAC Clone Setting:

8.76. The MAC Clone function is to clone the MAC when only one MAC is available from ITSP. This is to share with the PC using the same MAC. When you finished settings, please click the Submit button.

Tones Settings:

8.77. The Tone setting can be adjusted to generate Dial tone, Ring tone, Ring Back tone, and Busy tone for different countries. When you finished with the settings, please click the Submit button.

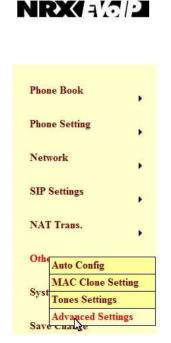




Configuration by Web Browser

Advanced Setting:

8.78. The advanced settings might be useful for some network requirements. The ICMP function is to echo when someone ping this device. This can prevent hackers from attacking the device by not echoing. When you finished the setting, please click the Submit button.





System Authority:

8.79. The user login name and password can be changed in this page.

System Authority

You could change the login username/password in this page.

New username:

New password:

Confirmed password:

Submit Reset

Save Change:

8.80. If you want to save the changes you have made for new setting in the NRX EVoIP Phone, You have to click the Save button. After you click the Save button, the phone will automatically restart and the new setting will implemented.

Save Changes

You have to save changes to effect them.

Save Changes: Save



Configuration by Web Browser

Update:

Update Firmware

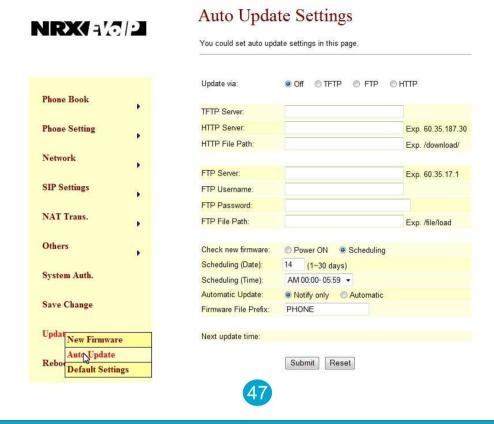
- 8.81. The IP Phone provides two methods, HTTP or TFTP, to update new firmware as follows:
- 8.82. Select the firmware code type, Risc or DSP code. (Usually Risc code)
- 8.83. Click the "Browse" button to choose the updated file location for HTTP download, or
- 8.84. Select TFTP and enter the IP address of TFTP server for firmware download, then click the "Update" button.

Update Firmware



Auto Update Settings:

8.85. The IP Phone provides two methods, HTTP or TFTP, to update new firmware as the following steps:



Configuration by Web Browser

Restore Default Settings:

8.86. You can restore the NRX EVoIP to factory default on this page. Click the Restore button, then the phone will restore to default and automatically restart again.

Restore Default Settings

You could click the restore button to restore the factory settings.

Restore default settings: Restore

Reboot:

8.87. If you want to restart the NRX EVoIP Phone, you can just click the Reboot button, and then the phone will restart automatically.

Reboot System

You could press the reboot button to restart the system.

Reboot system: Reboot

Menu Shortcuts

MENU shortcuts:

To use the menu shortcuts press the 'MENU' button followed by the keypad numbers shown.

- 1. Phone Book
 - 1.1 Search: Search Phone Book.
 - 1.2 Add entry: Add new phone number to phone book.
 - 1.3 Speed dial: Add speed dial phone number to speed dial list.
 - 1.4 Erase all: Erase all phone number from Phone Book.
- 2. Call history
 - 2.1 Incoming calls: Show all incoming call.
 - 2.2 Dialed numbers: Show all dialed call.
 - 2.3 Erase record: Delete call history.
 - 2.3.1 All: Delete all call history.
 - 2.3.2 Incoming: Delete all incoming call.
 - 2.3.3 Dialed: Delete all dialed out call.
- 3. Phone setting
 - 3.1 Call forward:
 - 3.1.1 All Forward:
 - 3.1.1.1 Activation: To Enabled/Disabled this function.
 - 3.1.1.2 Number: Forward to a registered or URL Number.
 - 3.1.2 Busy Forward.
 - 3.1.2.1 Activation: To Enabled/Disabled this function.
 - 3.1.2.2 Number: Forward to a registered or URL Number.



Menu Shortcuts

- 3.1.3 No Answer Forward.
 - 3.1.3.1 Activation: To Enabled/Disabled this function.
 - 3.1.3.2 Number: Forward to a registered or URL Number.
- 3.1.4 Ring Timeout: Set the Ring times to start the No Answer Forward function.
- 3.2 Do not Disturb
 - 3.2.1 Always: Block all calls
 - 3.2.2 By Period: Block calls by the period time
 - 3.2.3 Period Time: Set the start time and end time to Block calls.
- 3.3 Alarm Setting:
 - 3.3.1 Activation: Enable/Disable alarm
 - 3.3.2 Alarm Time: Set the alarm time
- 3.4 Date/Time setting:
 - 3.4.1 Date & Time: Set the IP Phone Date and Time.
 - 3.4.2 SNTP setting:
 - 3.4.2.1 SNTP: Enabled / Disable SNTP.
 - 3.4.2.2 Primary SNTP: Set Primary SNTP server IP address or URL.
 - 3.4.2.3 Secondary SNTP: Set Secondary SNTP server IP address or URL.
 - 3.4.2.4 Time zone: Set Time zone.
 - 3.4.2.5 Adjustment Time: Set adjustment time period.
- 3.5 Volume and Gain
 - 3.5.1 Handset volume: Set Handset volume from 0~15 (max.) for you to hear.
 - 3.5.2 Speaker volume: Set Speaker phone volume from $0\sim15$ (max.) for you to hear.
 - 3.5.3 Handset Gain: Set Handset Gain from 0~15 (max.) for remote site to hear.
 - 3.5.4 Speaker Gain: Set Speakerphone Gain from 0~15 (max.) for remote site to hear.
- 3.6 Ringer:
 - 3.6.1 Ringer volume: Ringer volume selection from 0~15 (max.).

Menu Shortcuts

3.7 Auto Dial: Auto Dial time selection from 3~9 seconds.

4. Network

4.1 WAN Setup:

4.1.1 IP Type:

4.1.1.1 Fixed IP client

4.1.1.2 DHCP client:

4.1.1.3 PPPoE client:

4.1.2 Fixed IP setting:

4.1.2.1 Host IP

4.1.2.2 Subnet mask

4.1.2.3 Gateway IP

4.1.3 PPPoE setting:

4.1.3.1 User name

4.1.3.2 Password

4.2 LAN Setup:

4.2.1 Bridge

4.2.2 NAT

4.3 DNS Server:

4.3.1 Primary DNS

4.3.2 Secondary DNS

4.4 VLAN:

4.4.1 Activation

4.4.2 VID

4.4.3 Priority

4.4.4 CFI

4.5 Status: Show IP addresses of WAN, LAN and MAC address (use UP/Down keys).



Menu Shortcuts

5. SIP Settings

To set the SIP setting from keypad, you have to press Menu_7_4 to input the password first, or the SIP setting may not be allowed to access.

5.1 Service Domain

5.1.1 First realm

5.1.1.1 Activation:

5.1.1.2 User name:

5.1.1.3 Display name:

5.1.1.4 Register name:

5.1.1.5 Register password:

5.1.1.6 Proxy server: Proxy Server IP Address

5.1.1.7 Domain server: Domain Server IP Address

5.1.1.8 Outbound proxy: Outbound Proxy IP Address

5.2 Codec

5.2.1 Codec type

5.2.1.1 G.711 uLaw: G.711 uLaw

5.2.1.2 G.711 aLaw: G.711 aLaw

5.2.1.3 G.723: G.723.1

5.2.1.4 G.729: G.729A

5.2.1.5 G.726-16: G.726 16Kbps

5.2.1.6 G.726-24: G.726 24Kbps

5.2.1.7 G.726-32: G.726 32Kbps

5.2.1.8 G.726-40: G.726 40Kbps

5.2.2 VAD: Voice Activity Detection Enable/Disable.

Menu Shortcuts

5.3 RTP Setting:

5.3.1 Outband DTMF: Outband DTMF Enabled/Disabled.

5.3.2 Duplicate RTP

5.3.2.1 No duplicate: No resend voice packets.

5.3.2.2 One duplicate: Resend voice packets once.

5.3.2.3 Two duplicate: Resend voice packets twice.

- 5.4 RPort Setting: RPort Enabled/Disabled.
- 5.5 Hold by RFC: Hold by RFC3261 Enabled/Disabled.
- 5.6 Status: Use Up/Down keys to show the SIP Proxy register status.

6. NAT Transversal

6.1 STUN setting

6.1.1 STUN: STUN Enabled/Disabled.

6.1.2 STUN server: Server IPAddress or URL.

7. Administrator

7.1 Auto Config

7.1.1 Config Mode: Select Disable/TFTP/FTP/HTTP for auto config function with server.

7.1.2 TFTP server: Set the TFTP server IP address.

7.1.3 FTP server: Set the FTP server IP address.

7.1.4 FTP Login Name: Set the login name to the FTP server.

7.1.5 FTP Password: Set the Password to the FTP server.

7.2 Upgrade System: You can restore to the default setting.

7.2.1 Upgrade Now: Select Yes/No to upgrade with the upgrade Server.

7.2.2 Upgrade via: Select Disable/TFTP/FTP/HTTP to do upgrade.

7.2.3 Status:

7.2.4 Reset Time: Set Yes/No to reset time.

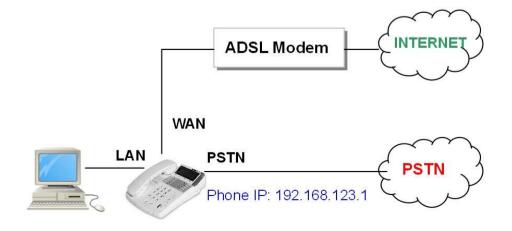


Voip Application Examples

- 7.3 Default setting: To load/abort the default setting.
- 7.4 System Authority: Must enter the password first for SIP setting. Default is "admin".
- 7.5 Version: This shows the firmware version.
- 7.6 Watch Dog: This enables Watch Dog function for debugging.
- 7.7 Restart: This function will restart your IP Phone.

10. VolP Application Examples:

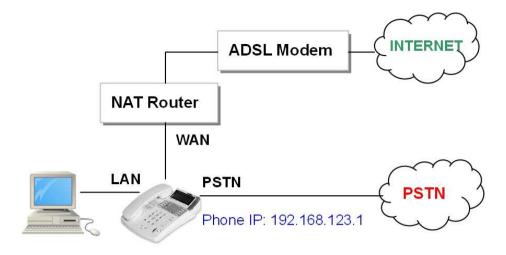
A. ADSL Connections without NAT Router for NRX EVolP





Voip Application Examples

B. ADSL Connections with NAT Router for NRX EVolP



Example 1: SIP-to-SIP Calling/Answering

Application:

The SIP-to-SIP calling works when both calling and answering parties are registered to SIP server with given registered phone numbers. The ADSL connections can be as in either Diagrams A or B. Both parties are registered to SIP server under NAT router. For Diagram A without NAT router, you may select NAT mode to enable the embedded NAT router. For Diagram B with external NAT router, you may select Bridge mode to disable the embedded NAT.



Voip Application Examples

Configuration:

- 1. Select either "NAT" or "Bridge" in accord with your network in "WAN settings" page,
- 2. Select "DHCP Client" to automatically get an IP address from NAT router.
- 3. Remember to click the "Submit" button,
- 4. Select Active "ON" in the "SIP settings / Service Domain" page,
- 5. Enter the Register Name, Register Password, Proxy Server, and Outbound Proxy,
- 6. Select "ON" in the "STUN setting", if Outbound Proxy is NOT available.
- 7. Upon successful SIP registration, the REG LED indicator will be ON and the LCD will show registered <phone number>.

Calling:

- 8. Pick up the phone, and you should hear a dial tone for VoIP mode.
- 9. Press 1688# or 1688 to call the party with the registered SIP phone number 1688. Note that # key will dial out the number immediately. Dialing without # will not dial out until the auto dial timer default=5 seconds) elapsed.

Example 2: SIP to Direct IP Calling

Applications:

The application is for the calling party with ADSL connection as in either Diagrams A or B. The calling party is registered to SIP server with either fixed real IP or private IP under NAT router. The answering party is with fixed real IP. You may also use the Phone Book to call Direct IP Address.

Configuration:

1. Same as in Example 1.

Voip Application Examples

- 2. Select "ON" in "STUN setting" page, if Outbound Proxy is NOT available.
- 3. Upon successful SIP registration, the LCD will show registered <phone number>.

Calling:

- 4. Press Speakerphone key, and you should hear a dial tone.
- 5. Press 211*21*191*4# or 211*21*191*4 to call the party with the real IP address of 211.21.191.4. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

Example 3: Direct IP to Direct IP Calling/Answering

Application:

The applications are for ADSL connection without NAT router as in Diagram A. Both parties are with fixed real IP. The Direct IP calling works when both calling and answering parties are with known fixed IP. SIP server registrations are not required in this application.

Configuration:

- 1. Select "Fixed IP" in the "Network / WAN settings" page,
- 2. Enter the items of IP, Subnet Mask, Gateway IP,
- 3. Click the "Submit" button.

Calling:

- 4. Pick up the phone, and you should hear a dial tone.
- 5. Press 211*21*191*4# or 211*21*191*4 to call the party with the real IP address of 211.21.191.4. Note that # key will dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapsed. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.



Voip Application Examples

Example 4: Direct IP to Direct IP Calling within NAT Router

Application:

For the calling party in ADSL connection with NAT router as in Diagram B, this Direct IP calling can work when the answering parties are with fixed private IP addresses within the same VPN network, or with fixed real IP addresses.

Configuration:

- 1. Select "Fixed IP" in the "Network / WAN settings" page,
- 2. Enter the items of IP, Subnet Mask, Gateway IP,
- 3. Click the "Submit" button.

Calling:

- 4. Pick up the phone, and you should hear a dial tone
- 5. Press 192*168*1*51# or 192*168*1*51 to call the party with the private IP address of 192.168.1.51. Press 211*21*191*4 to call the party with the real IP address of 211.21.191.4. In a moment, you should hear a ring back tone, and wait for the called party to answer.

Voip Application Examples

Example 5: 3-Way Conference Call, Call Transfer, Call Waiting, Hold

Application:

The Call Transfer and 3-Way Conference Call applications are for calls among Parties A, B, and C. Three parties are registered to SIP server with either fixed real IP or private IP. There are two kinds of call transfer; Blind Transfer and Attendant Transfer.

Blind Transfer:

- 1. Party A calls Party B.
- 2. While in conversation, Party B may press Transfer key, and should hear a dial tone.
- 3. Party B enters [Party C number] # and then hang up to transfer to Party C.

Attendant Transfer:

- 1. Party A calls Party B.
- 2. While in conversation, Party B may press Transfer key, and should hear a dial tone.
- 3. Party B enters [Party C number] # and talks to Party C.
- 4. When Party B then hangs-up, Party A will transfer and connect to Party C.

3-Way Conference Call:

- 1. Party A calls Party B.
- 2. While in conversation, Party B may press Hold key to hold the call, and should hear a dial tone.
- 3. Party B calls Party C.
- 4. While in conversation, Party may press Conf. key to join in Party A for a three-way conference.

Call Waiting Application:

When a new call is coming while you are talking, you will hear an interrupt "dodo" tone and you can press Hold key to answer the new incoming call. You may press Hold key to switch back to the previous call.



Trouble Shooting

Call Hold Application:

You may press Hold key to hold the current call for a while, then press Hold key again to resume conversations.

11. Trouble Shooting for Web Configurations:

11.1. DO NOT HEAR DIAL TONE?

When you pick up the phone and hear a busy tone, it indicates the WAN port is NOT connected. The LCD will show Ethernet Error! Make sure the ADSL Ethernet cable is connected to the WAN port of NRX EVoIP phone and Power Reset again.

11.2. CAN NOT ACCESS WEB PAGE?

IE Web Browser is a useful tool to configure NRX EVoIP phone. When you have difficulties in accessing the default IP address http://192.168.123.1 of the phone, the most likely reason is that your PC might have different subnet IP settings from 192.168.123.xxx. In this case, you must change NRX EVoIP phone's IP address to the same subnet as your PC and NAT router.

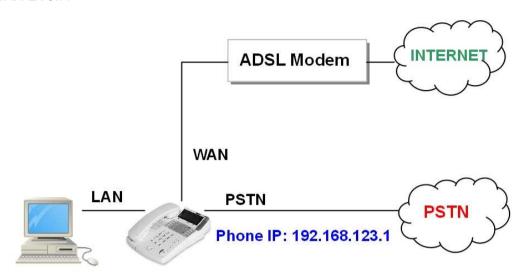
Example: To change the NRX EVoIP IP address to the same subnet as PC and NAT router

- 1. Set the phone to DHCP Client mode. The NRX EVoiP phone will reboot, and LED will start flashing to get an IP address from NAT DHCP server.
- 2. Press Menu_4_5 to read IP Addresses for WAN and LAN Ports, for example, 192.168.62.51.
- 3. Enter from IE web browser http://192.168.62.51:9999 to login to the web page for configurations

Trouble Shooting

11.3. CONFIGURE PC'S IP SETTINGS FOR VP306 EMBEDDED NAT FUNCTION?

If you don't have a router to connect both your PC and NRX EVoIP for sharing the only one IP address from ADSL/Cable modem, you should enable the embedded NAT function inside the NRX EVoIP phone. You need to change your PC's IP settings to recognize the NRX EVoIP as the default gateway. In this case, you should enable the embedded NAT router of the NRX EVoIP to provide more than one IP addresses for the PC and NRX EVoIP.





Trouble Shooting

Example: To change your PC IP address to the same subnet as 192.168.123.1 for the NRX EVoIP

- 1. As in Window 2000 (my computer),
 - At "Network and Dialup Connections", right click on "Local Area Connection", then click on property.
 - The "Local Area Connection Properties" window will pop up.
 - Double click on "Internet Protocol (TCP/IP)".
 - The "Internet Protocol (TCP/IP) Properties" window will pop up.
 - Click on "Use the following IP Address".
 - Enter IP: 192.168.123.50 (50 can be any number other than 1, which is used by the NRX EVoIP).
 - Enter Subnet mask: 255.255.255.0.
 - Enter Default gateway: 192.168.123.1
 - Click on OK button.
- 3. You will lose internet connection at this time.
- 4. At IE browser, type http://192.168.123.1
- 5. Follow the example in "Advanced Settings for Embedded NAT" for web login.
- 6. At LAN setting, turn on DHCP server.
- 7. At WAN setting, choose "DHCP client" to work with your ADSL/Cable modem.
- 8. Save change, wait for the NRX EVoIP to reboot.
- 9. Change your PC's "Internet Protocol (TCP/IP) Properties" back to "obtain an IP address automatically".
- 10. You may press Menu_4_5 to read IP Addresses for WAN and LAN Ports.

Warranty

12. WARRANTY

Trojan Telecom has built the NRX EVoIP telephone to a high standard. Our warranty reflects our belief that during it's working life you should not experience any mechanical failures.

Trojan Telecom will at all times use a sensible and supportive attitude towards warrantable returns, working with you in identifying 'no fault found'.

The following terms and conditions apply:

- 1) Where we find a genuine warranty failure, Trojan Telecom will replace the faulty instrument with a one-for-one replacement.
- 2) It is the responsibility of the user to return the faulty telephone to Trojan Telecom. We ask you to cover this cost and we will return your replacement telephone at our expense.
- 3) We reserve the right to repair the faulty item or replace it with a similar telephone of the same make.
- 4) Discontinued models under warranty will be replaced with a similar or more featured telephone.
- 5) We reserve the right to charge for items considered to be chargeable under fair wear and tear.
- 6) User misuse or any modification carried out to the NRX EVoIP telephone by the customer is not considered a manufacturing fault or component defect. Therefore it is not covered by the warranty.
- 7) Accidental damage such as liquid spillage or user damage will not be covered under the terms of the warranty.
- 8) When returning the telephone please ensure you fully complete the returns authorisation form, failure to do so could cause



Warranty

you unnecessary delay.

The terms of this warranty do not effect your statutory rights. Warranty applies to UK customers only.

Faulty units should be returned to our service centre at the following address, together with the completed return form on the opposite page.

SERVISCOMM UNIT 2, RED BARNES WAY McMullen Road, Darlington DL1 2RR UK



Ist Floor
Amphenol Complex
Thanet Way
Whitstable
Kent
CT5 3JF
ENGLAND

Tel: (+44) (0)1227 275357 Fax: (+44) (0)1227 272932

e-mail: info@trojantel.com

REF: NRXEVO350140607

CE