โปรเฟสชั่นแนล พี. เอ. บี. เอ็กซ์.
30/270 ซอยนวมินทร์ 80 แขวงนวลจันทร์ เขตบึงกุ่ม กรุงเทพฯ 10230
Hotline : 084-920-5065 Tel : 02-519-1718 , 02-107-3057
E.Mail : info@pfpbx.com , jirasak_service@hotmail.com
www.pfpbx.com

Setup Reference Guide for KX-NS1000 to SBC interconnection

Method of connection by "PPPoE and Global IP address directly" (i.e. SBC is the Perimeter Router device.)

Panasonic IP-PBX (KX-NS1000 Version2 series),
Media5 Session Border Controller (Mediatrix501 series SBC)

Version 1.0 (PSNJ) 11th.March 2013

Attention: The content of this document is made up by verification results. **It is no guarantee**.

Models Used during verification:
Panasonic IP-PBX KX-NS1000 (Ver2)
Media5s SBC Mediatrix501 (Firmware 5.35-M4)
Panasonic SIP Phone KX-UT series SIP telephones (Version 01.221)

Panasonic System Networks Co., Ltd.

Change history

Version NO.	CONTENTS OF REVISION	Support	PSN (Japan)	
		Checked by	Checked by	Author
Ver. 1.0	First edition			Oonishi Mar.6th 2013

Table of Contents

Table of Contents

Trademarks

• Microsoft, Windows, Windows XP, Vista, 7 and Internet Explorer are either registered trademarks or trademarks of Microsoft Corporation in the United States and/or other countries.
• All other trademarks identified herein are the property of their respective owners.
• Microsoft product screen shot(s) reprinted with permission from Microsoft Corporation.

[Important matter]

Configuration Advice

You have to configure the SBCs SIP Trunk settings If you have a SIP trunk connection need between PBX and ITSP (Internet Telephony Service Provider).
As necessary, refer the additional volume of "Setup Guide for Media5 SBC and NS1000 Ver2 WAN Scenario Ver1.0"."

1. Introduction

[Note]

The SIP remote extension(s) are registered to the V-UTEXT32 of NS1000 (Ver.2), it's not the registration of V-SIPEXT32. We can register the UT-SIP extension as V-UTEXT32 through the SBC by NS1000 Ver2. This Setup Reference Guide describes mainly using the V-UTEXT32.

Objective:

A Session Border Controller is required to supplement existing IP-PBX functionality.
It will provide the means of establishing a simple remote office connection
(Allowing the use of remote SIP extensions of the IP-PBX without the need for a PPTP, IPSEC, GRE or Hosted VPN Solution). **** Please Note: HTTPS/SSL is VPN Technology ****

This Setup Reference Guide describes the configuration to interconnect between the Panasonic IP-PBX (KX-NS1000 Version2 series), the Media5 Session Border Controller (Mediatrix501 series SBC), and remote SIP Extensions (Panasonic KX-UT series).
The items above are interconnected using SIP, TR069 (CWMP) and NTP protocol. The global IP address (also known as public IP address) of the main office is used to interconnect them.

Results (confirmed operation):
1-1 Receiving and making a Call
Calls between extensions are possible. The Caller ID (internal phone number) is displayed on the LCD screen of Panasonic UT-SIP Extension and SIP Extension.
Incoming calls from PBX trunk lines also display the Caller ID (according to system settings).

1-2 Conversation with G.722, G.711 and G.729
Use of the above codec is possible, providing PBX settings allow this. (e.g. KX-NS1000 (V-UTEXT) settings)

1-3 Placing a call on-hold and retrieving a Call that is on-hold
These features are confirmed by KX-NS1000 control.

1-4 Transferring Call
The transferring of a Call to another destination is confirmed by KX-NS1000 control.

1-5 Call forwarding (V-UTEXT32 Registered)
These features are confirmed by KX-NS1000 control.
*** Note*** This feature does not work as using registration of SIP extension(V-SIPEXT32).
Restriction on the use of standard SIP Extension (V-SIPEXT32).

2. Approach to Interconnection

(1) For the Panasonic IP-PBX, the Virtual UT SIP Extension (V-UTEXT32) is used to interconnect the IP-PBX to a remote UT SIP extension (remote office) via the SBC. The SBC is installed as the main router in the head office. For this setting of the SBC, WAN and LAN (ET1) interface are used. All SIP traffic between the IP-PBX and the internet is routed through the Mediatrix SBC. The SBC is set-up a DHCP server and also as a NAT device.

(2) The SBC operates to ensure correct interconnection between the IP-PBXs V-UTEXT32 virtual circuit card and the Remote office UT SIP Extension.
The SBC provides the following functions:

- Remote office UT SIP and SIP extensions address resolution and address translation within SIP messages.
- Head office (any PBX extension) and the remote office (UT SIP Extensions) can be seamlessly connected by the use of an IP-PBX UT SIP extension.
- Little or no dependence on the setting of the Router of the Remote-Office.

(3) We recommend that you consider the bandwidth of Internet access in each country, to change the priority of voice Codec G.729 the remote side.

(4) We recommended that you will be use the Port number 5060 of receiving of SIP in SBC.
The SBC is likely to have some interoperability issues when using different SIP port of this.
The SBC will check all SIP messages and modify them even if as use the SIP Trunk in the PBX.
It means the SBC receiving Port Number is "5060" for SIP-Extension and also SIP Trunk.
Therefore we strongly recommend that you set-up the port number 5060 of SBC, due to the specification of the Media5 SBC.

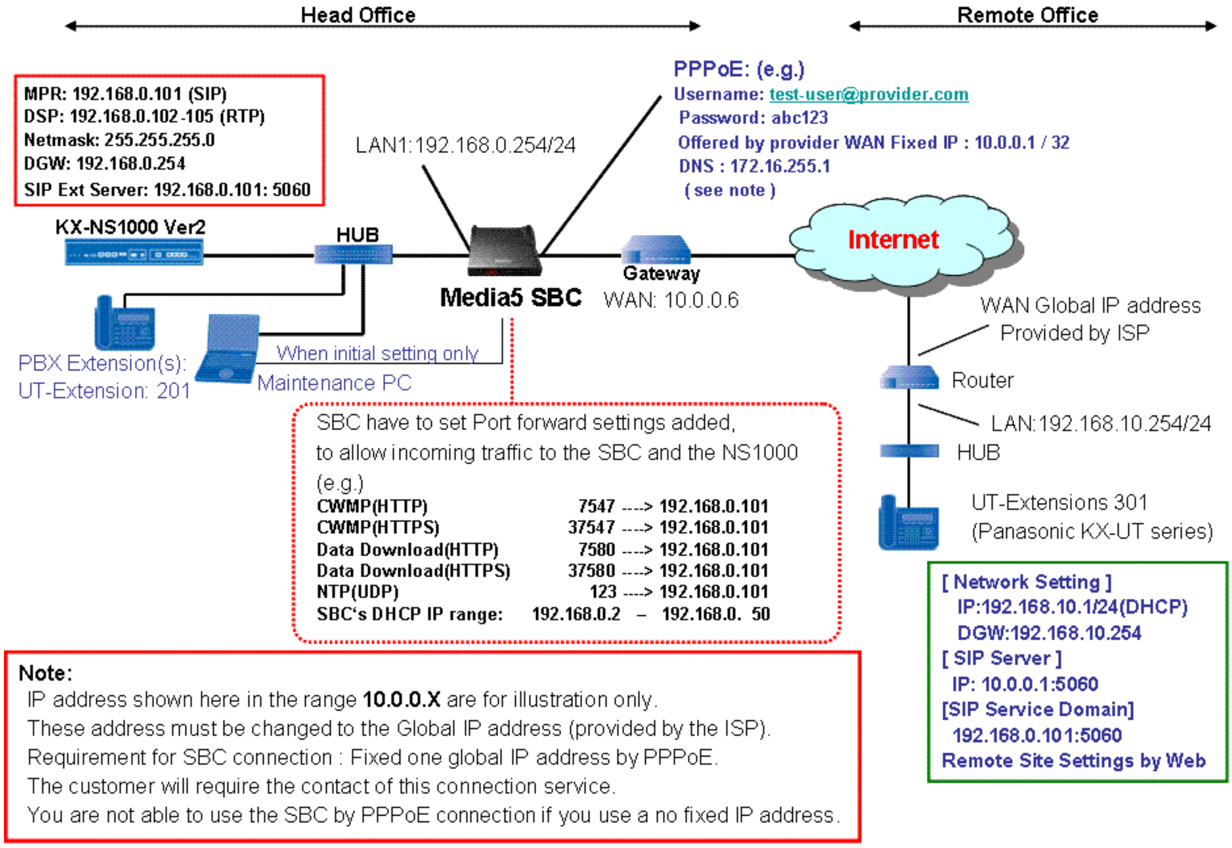
(5) About Interoperate with Remote SIP Extension and SIP Trunk connection for ITSP.
You have to configure the SBCs SIP Trunk settings if you have a SIP trunk connection need between PBX and ITSP (Internet Telephony Service Provider).
As necessary, refer the additional volume of Setup Reference Guide for KX-NS1000 to SBC SIP Trunking.

(6) The NS1000 has protocol HTTPS and HTTP for UT-SIP Phone registration.
The NS1000 can support up to 20 remote extensions at the same time when using the HTTPS protocol.
The protocol is described the HTTPS type as example in this Setup Reference Guide.

3. System configuration example

3.1 Diagram of system configuration example

Method of Connection by PPPoE and Global IP address directly



3.2 Settings:

This section describes the network address scheme. Refer to later sections regarding entry of these and other settings.

3.2-1 SBC – Contents of Main Network Settings (Example)

Item	Configuration example	Description
PPPoE Fixed WAN IP address	10.0.0.1 (change to global address)	Mandatory (Information offered by provider) *Need to the External IP of SBC settings
Username	test-user@provider.com	Mandatory (Information offered by provider)
Password	abc123	Mandatory (Information offered by provider)
LAN nterface1:IP address	192.168.0.254	SBC LAN fixed-IP address
LAN interface 1: Netmask	255.255.255.0	Subnet mask
Receiving SIP port	5060	SIP port used
Used RTP port	35000 - 35999	Use for RTP streams.
Primary DNS	172.16.255.1	Mandatory (Information offered by provider)
DHCP IP Range from To	192.168.0.2 - 50	SBC's DHCP Server: Enable

3.2-2 SBC - Contents of Port Redirection (also known the Port forwarding) Settings (Example)

Protocol	Port number	Destination	Description
CWMP(HTTP)	7547 (TCP)	192.168.0.101	Send CWMP to PBX (PBX LAN IP address)
CWMP(HTTPS)	37547 (TCP)	192.168.0.101	Send CWMP to PBX (PBX LAN IP address)
SIP-MLT Data Download(HTTP)	7580 (TCP)	192.168.0.101	Send Data to PBX (PBX LAN IP address)
SIP-MLT Data Download(HTTPS)	37580 (TCP)	192.168.0.101	Send Data to PBX (PBX LAN IP address)
NTP	123 (UDP)	192.168.0.101	Send NTP to PBX (PBX LAN IP address)

3.2-3 IP-PBX (NS1000) - Contents of Main Network Settings (Example)

PBXs IP Address/Ports Settings

Item	Configuration example	Description
PBX MPR IP address	192.168.0.101	Example only (Fixed IP)
PBX DSP IP address	192.168.0.102 - 105	Example only (Fixed IP)
Net Mask	255.255.255.0	Example only
Gateway	192.168.0.254	SBC LAN IP address
DNS Settings (Preferred DNS IP Address)	172.16.255.1 or 192.168.0.254 (SBC LAN)	Information offered by provider or SBC LAN IP address
PBX DHCP Server Feature Starting IP address Ending IP address	 192.168.0.51 192.168.0.100	If required Note) Set the different IP range from SBC’s DHCP IP range.

3.2-4 IP-PBX (NS1000) - Confirmation of current each [Port Number] on Site Property (Example)

Port Number Item	Configuration example	Description
UDP Port No. for SIP Extension Server	5060	Default (SIP Port Number)
CWMP (HTTP) Port No. for SIP-MLT	7547	Default
CWMP (HTTPS) Port No. for SIP-MLT	37547	Default
Data Transmission Protocol (HTTP) Port No. for SIP-MLT	7580	Default
Data Transmission Protocol (HTTPS) Port No. for SIP-MLT	37580	Default

3.2-5 IP-PBX (NS1000) - Configure the SIP Extension parameter on Site Property (Example)

SIP Extension Item	Configuration example	Description
NAT - CWMP Server IP Address	10.0.0.1 (Change to Global IP)	Default: empty (Set PPPoE Fixed IP address)
NAT - CWMP Server (HTTP) Port No.	7547	Default
NAT - CWMP Server (HTTPS) Port No.	37547	Default
NAT - SIP-MLT Data Download Server (HTTP) Port No.	7580	Default
NAT - SIP-MLT Data Download Server (HTTPS) Port No.	37580	Default
NAT - SIP Proxy Server IP Address	10.0.0.1 (Change to Global IP)	Default: empty (Set PPPoE Fixed IP address))
NAT - SIP Proxy Server Port No.	**5060**	**Default:15060** **Recommended changes**
NAT - NTP Server IP Address	10.0.0.1	Default: empty (Set PPPoE Fixed IP address)
NAT - NTP Server Port No.	123	Default
NAT - Keep Alive Packet Type	Blank UDP	Default You can select REGISTER Or None
NAT - Keep Alive Packet Sending Interval Time (s)	20	Default: 20 (sec) *Note 1
NAT - SIP Register Expire Time (s)	20	Keep Alive Packet Type: REGISTER only

*Note 1: This interval must be shorter than the NAT binding time of the router. The default value is appropriate in most cases.

3.2-6 IP-PBX (NS1000) - Configure the Options of Recommended P2P Group (Example)

System options P2P Group (2.9-Option 7)	Configuration example	Description
Priority Voice 1	G729	Default: G729
Priority Voice 2	G711	Default: G711
Priority Voice 3	None	Default: G722

3.2-7 IP-PBX (NS1000) - Configure the Group of P2P (Example)

P2P Group (3.10)	Configuration example	Description
P2P Group P2P Group Name Bandwidth Control	1 Empty Disable	Default Default: Empty Default: Disable
P2P Group P2P Group Name Bandwidth Control	2 Remote Office Enable	Default Example Default: Disable

3.2-8 IP-PBX (NS1000) - Contents of Remote UT Extension (SIP-MLT) Settings (Example)

Item	Configuration example	Description
[Port Property Main] SIP Extension Number Password P2P Group **[Option tab]** Codec Priority **[Remote Place tab]** Phone Location Protocol for Remote SIP-MLT	 301 1234 2 *1st: G729A / 2nd:G711A/ 3rd:G722Mu / *4th: G722 Remote HTTPs	 Example Default Default:1 *1st: G722 / 2nd:G711A/ 3rd:G722Mu / *4th: G729A Default: Local Default: HTTP
[Port Property Main] SIP Extension Number Password P2P Group **[Option tab]** Codec Priority **[Remote Place tab]** Phone Location Protocol for Remote SIP-MLT	 302 1234 2 *1st: G729A / 2nd:G711A/ 3rd:G722Mu / *4th: G722 Remote HTTPs	 Example(If required) Default Default:1 *1st: G722 / 2nd:G711A/ 3rd:G722Mu / *4th: G729A Default: Local Default: HTTP

3.2-9 Maintenance PC - Contents of Network Settings example

Item	Configuration example	Description
Maintenance PC IP address	192.168.0.200 (Example)	(DHCP or fixed ; For fixed, confirm usable IP address first)
Subnet Mask	255.255.255.0	Example
Gateway		Unused (in Fixed IP)
DNS		Unused (in Fixed IP)

3.2-10 Remote Office UT Extension - Contents of automatically downloaded settings via TR069.

(Example) * Note1: Unless specifically instructed to do so, please do not directly configure the UT-SIP Phone via the web as this will interfere with the configuration settings delivered by the PBX.

Item	Configuration example	Description
UT SIP Phone: IP address	DHCP(Example, 192.168.10.1)	
UT SIP Phone: Netmask	DHCP(Example, 255.255.255.0)	
UT SIP Phone: Gateway	DHCP(Example, 192.168.10.254)	
Registrar Server Address	10.0.0.1 (Change to global IP address)	(Set Head office WAN address of assigned to the SBC.)
Registrar Server Port	5060	SBC SIP receiving port
Proxy Server Address	10.0.0.1 (Change to global IP address)	(Set Head office WAN address of assigned to the SBC.)
Proxy Server Port	5060	SBC SIP receiving port
SIP Service Domain	192.168.0.101:5060	Example PBX SIP Server Domain Need to add a :port number
SIP source port	25060	Source port for outgoing SIP * Measures for SIP ALG function in Remote router.
NAT Identity Keep Alive Interval	20 (second)	Example (Default: 20)
NAT Identity Supports Rport	Yes	Example (Default: Yes)
SIP extension Number Password	301 pass301	Example
SIP extension Number Password	302 pass302	Example (if required)

3.2-11 Remote Office Existing Router - Contents of main network settings

Item	Configuration example	Description
WAN global IP address	Fixed IP or It will provide different IP address from ISP every time.	Existing remote office router WAN IP address.
LAN IP address	192.168.10.254	Existing remote office router LAN IP address

3.2-12 Remote office router contents of port forward settings

It is not necessary to change any settings of the Router of the remote office when using a SIP phone with “Keep-Alive” capability. (e.g.) Panasonic KX-UT series SIP Phones.

UT series SIP Phones can send the Keep Alive messages to the SBC (Blank UDP packets).

3.3 Media5 SBC Configuration Sheet (Connection type: PPPoE Connection)

Section	Part	Item	Setting value	Description
Home	Active Profile	Security	Low	Select
Configuration	Network Config	Operational mode	Router	Default
	WAN	ET0 used as	Outside	Select
	WAN	Access type	**PPPoE**	Select
	WAN	**User**	**test-user@provider**	Example
	WAN	**Password**	**abc123**	Example
	DNS Server	IP Address	172.16.255.1	Example
		2nd (DNS Server Address)		If required
	LAN	IP Address	192.168.0.254	
	LAN	Subnet Mask	255.255.255.0	
	DHCP Server	Enable		
		From: / To:	192.168.0.2 -50	
SIP Server	Allow to Register	Inside users	All	Select
		Outside users	All	Select
		Allow outgoing calls from	All	Select
Advanced	Advanced SIP set	Far End Nat Traversal (FENT)	Select the check	
		Detect endpoints behind same NAT	Clear the check	
Advanced	Authorized Users	Method	REGISTER	
		URI	*	Enter
		Direction	Inbound	
		Allow	Select the check	
		Authentication	Select the check	
		Authentication User IDs	*	Enter
Advanced	Authorized Users	Method	INVITE	
		URI	*@192.168.0.101	Enter
		Direction	Inbound	
		Allow	Select the check	
		Authentication	Select the check	
		Authentication User IDs	*	Enter
Advanced		Reuse received nonces	Clear the check	
	SIP Proxy	SIP Server UFP port numbers	5060	
	Advanced	RTP media port range	35000-35999	
		Allow RTP in reverse direction	Select the check	
		Reuse port number with same session	Select the check	
		Force Real Username on registration	Select the check	
	Trusted Networks	Check box	Clear the check	P-Asserted-ID

3.3 Media5 SBC Configuration Sheet (2/2) (Connection type: PPPoE Connection)

Section	Part	Item	Setting value	Description
SECURITY Profile:Low	Port redirection	Outside Port	Inside Host	
	TCP	Local port: **7547** (CWMP port / HTTP)	192.168.0.101	Remote UT-SIP Protocol type. Select type: HTTP
	TCP	Local port: **37547** (CWMP port / HTTPS)	192.168.0.101	Remote UT-SIP Protocol type. Select type: HTTPs
	TCP	Local port: **7580** (Data download / HTTP)	192.168.0.101	Remote UT-SIP Protocol type. Select type: HTTP
	TCP	Local port: **37580** (Data download / HTTPs)	192.168.0.101	Remote UT-SIP Protocol type. Select type: HTTPs
	UDP	Local port: **123** (NTP port)	192.168.0.101	Remote UT-SIP time server

3.4 SBC Firmware Revision

Section	Installed Firmware	
Device Information	5.35-M4	or Later

3.5 KX-NS1000 and UT-Extension Firmware Revision

Section	Installed Firmware	
KX-NS1000 IP-PBX Version	2.02039	or Later
KX-UT Phone　Version Information	01.160	or Later

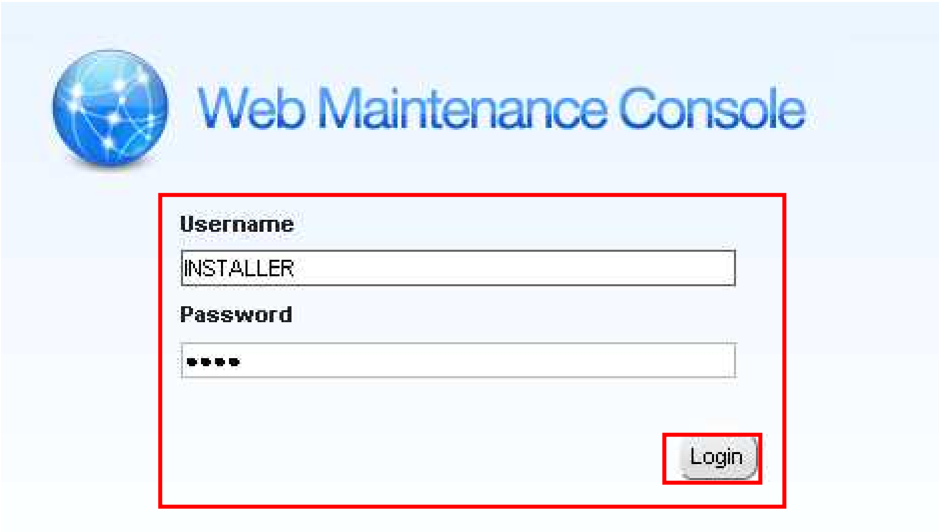
4. Initial set-up of the NS1000

(Note) The SIP remote extension(s) are registered to the V-UTEXT32 of NS1000 (Ver.2)

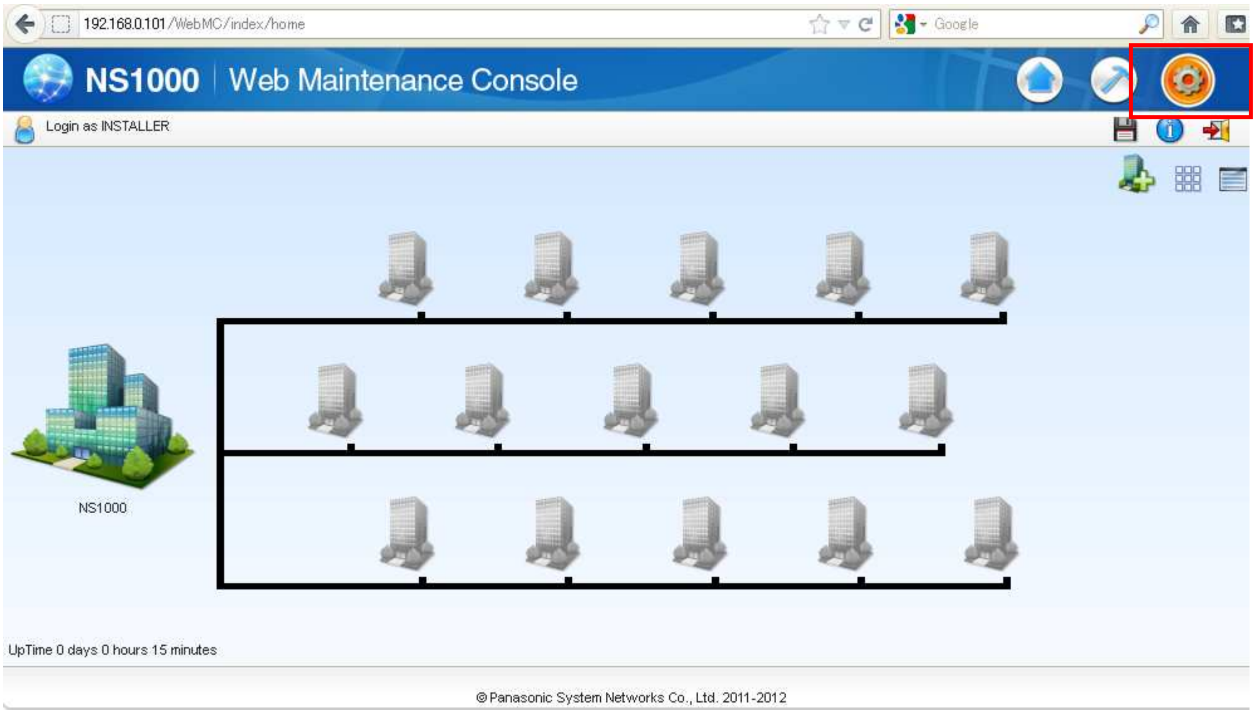
4.1 Start up software of web browser. (Internet Explorer Version 7 or later, Mozilla Firefox 6 or later)

4.2 Access the KX-NS1000 Web Maintenance Console page (using previously read IP address).
e.g. http://192.168.0.101/

4.3 Enter Username: **INSTALLER**, Password:**1234** ---> Next, click on [**Login**].

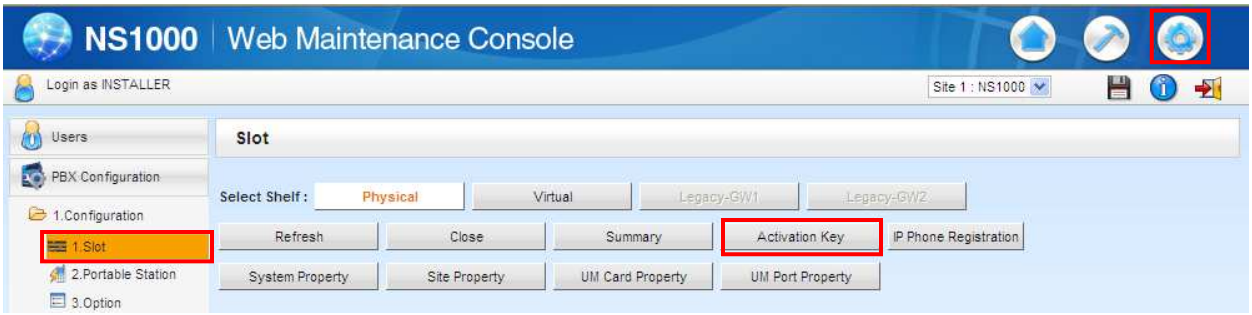


4.4 Access to initial web page (HOME) and Click on [**Setup**].



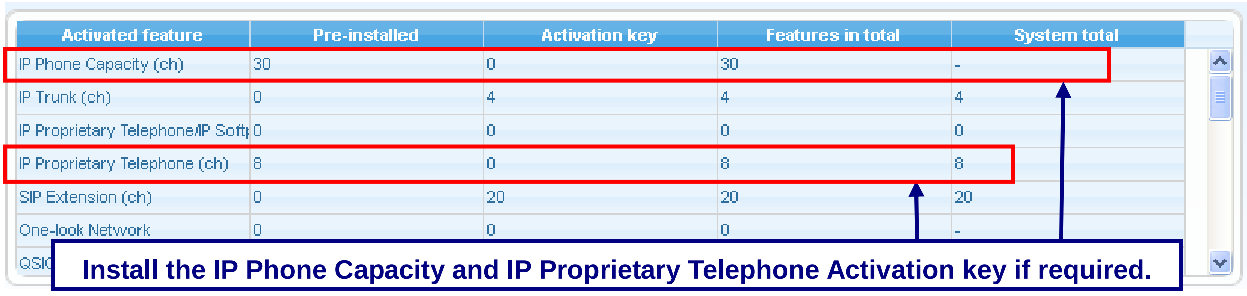
4.5 Confirmation of Activation Key (To install if you need the Activation key.)

Click on [**PBX Configuration**] --> [**1.Configuration**] --> [**1.Slot**] --> [**Activation Key**]



4.6 Confirmation of "**IP Phone Capacity** and **IP Proprietary Telephone Activation (ch)** " key

(In this case, IP Phone Capacity (ch):30 / IP Proprietary Telephone/IP:0 (Note *1).



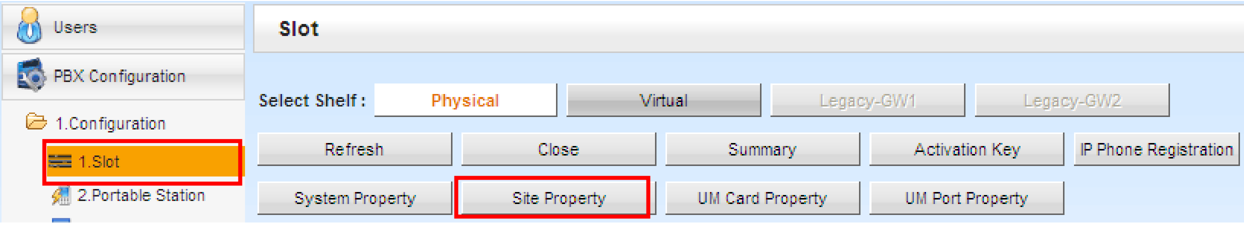
Activated feature	Pre-installed	Activation key	Features in total	System total
IP Phone Capacity (ch)	30	0	30	-
IP Trunk (ch)	0	4	4	4
IP Proprietary Telephone/IP Softp	0	0	0	0
IP Proprietary Telephone (ch)	8	0	8	8
SIP Extension (ch)	0	20	20	20
One-look Network	0	0	0	-
QSIG				

***Note *1): 30 IP-Extension can be installed to NS-1000 without extra Activation Keys, but for connecting IP Extension itself. User must purchase IP-Telephones Activation Keys when expand over 8 Telephones.**

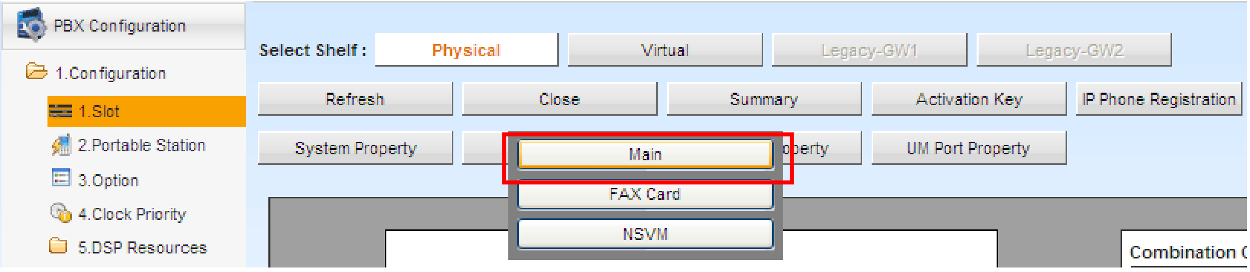
4.7 Confirm, then click [**OK**] to close page.

4.8 Click on [**PBX Configuration**] --> [**1.Configuration**] --> [**1.Slot**]

--> Move mouse over [**Site Property**]

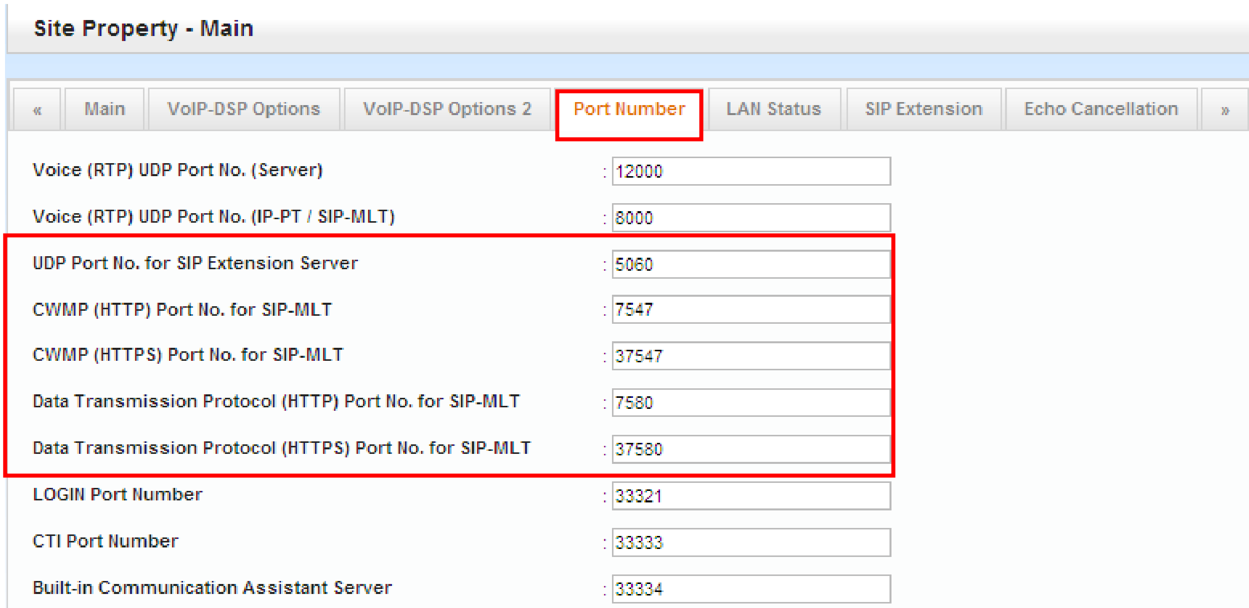


4.9 Select [**Main**] menu.



4.10 Click on [**Port Number**], --> Confirmation of current parameter value

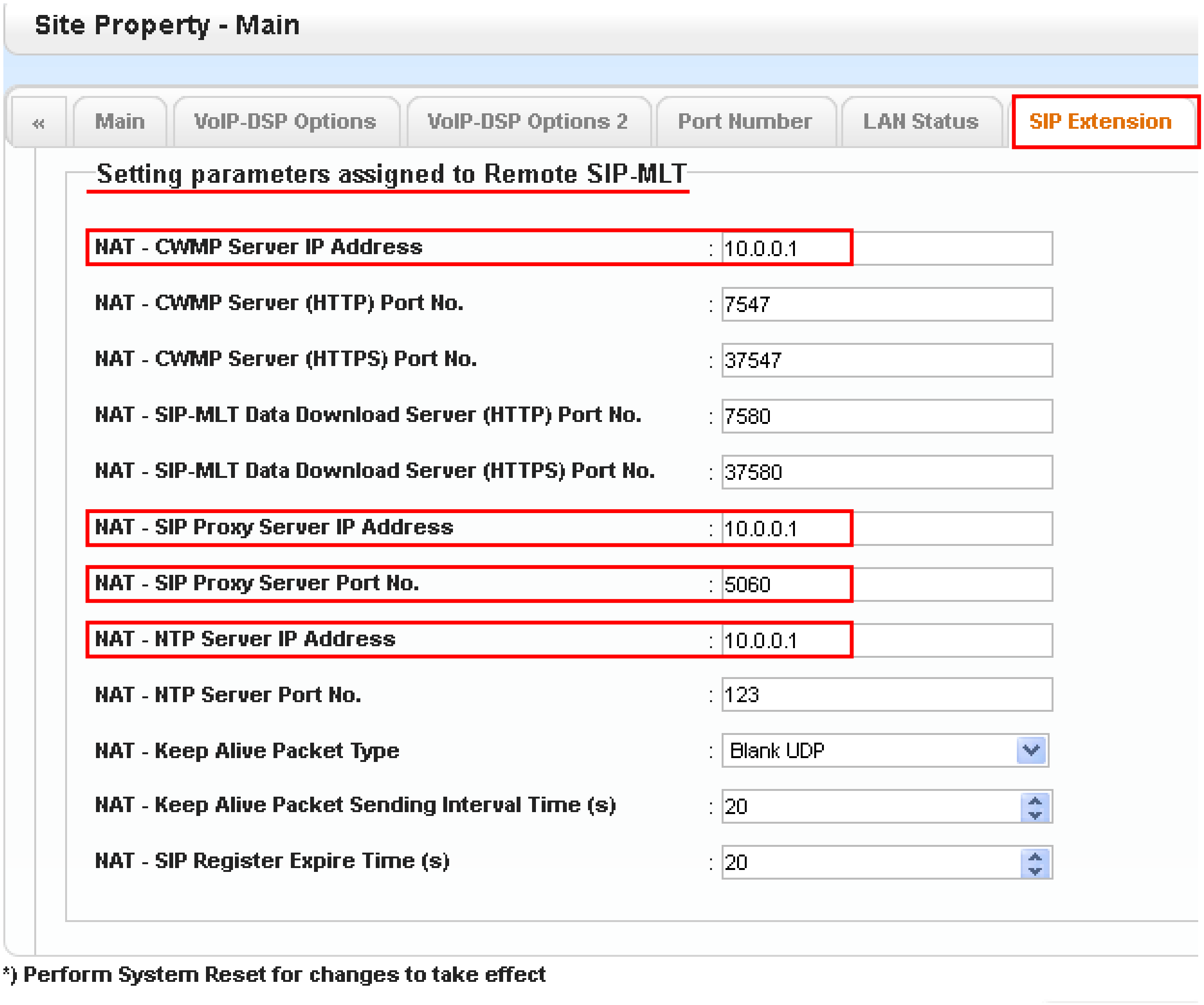
1. [UDP Port No. for SIP Extension Server]: **5060** (Default)
2. [CWMP (HTTP) Port No. for SIP-MLT]: **7547** (Default)
3. [CWMP (HTTPS) Port No. for SIP-MLT]: **37547** (Default)
4. [Data Transmission Protocol (HTTP) port No. for SIP-MLT]: **7580** (Default)
5. [Data Transmission Protocol (HTTPS) port No. for SIP-MLT]: **37580** (Default)



Note: These each parameter of PBX in LAN side are using default value in this example.

4.11 Configure SIP Extension into the IP-PBX (NS1000) for Remote SIP Extension.

Click on [**SIP Extension**] --> Edit the each parameters



4.12 Click on [**Apply**] and [**OK**] to close page.

--> *) **Perform System Reset for changes to take effect**

[Setting parameters assigned to Remote SIP-MLT] (Example)

NAT - CWMP Server IP Address : **10.0.0.1** (This is an example, Change to Global IP)

NAT - CWMP Server (HTTP) Port No. : **7547** (This is a default value.)

NAT - CWMP Server (HTTPS) Port No. : **37574** (This is a default value.)

NAT - SIP-MLT Data Download Server (HTTP) Port No.: **7580** (This is a default value.)

NAT - SIP-MLT Data Download Server (HTTPS) Port No: **37580** (This is a default value.)

NAT - SIP Proxy Server IP Address: **10.0.0.1** (This is an example, Change to Global IP)

NAT - SIP Proxy Server Port No.: 5060 (Recommended changes) *Note Refer to 2. Approach (4)

NAT - NTP Server IP Address: **10.0.0.1** (This is an example, Change to Global IP)

NAT - NTP Server Port No.: **123** (This is a default value.)

NAT - Keep Alive Packet Type: **Blank UDP** (This is a default value)

NAT - Keep Alive Packet Sending Interval Time (s): **20** (This is a default value)

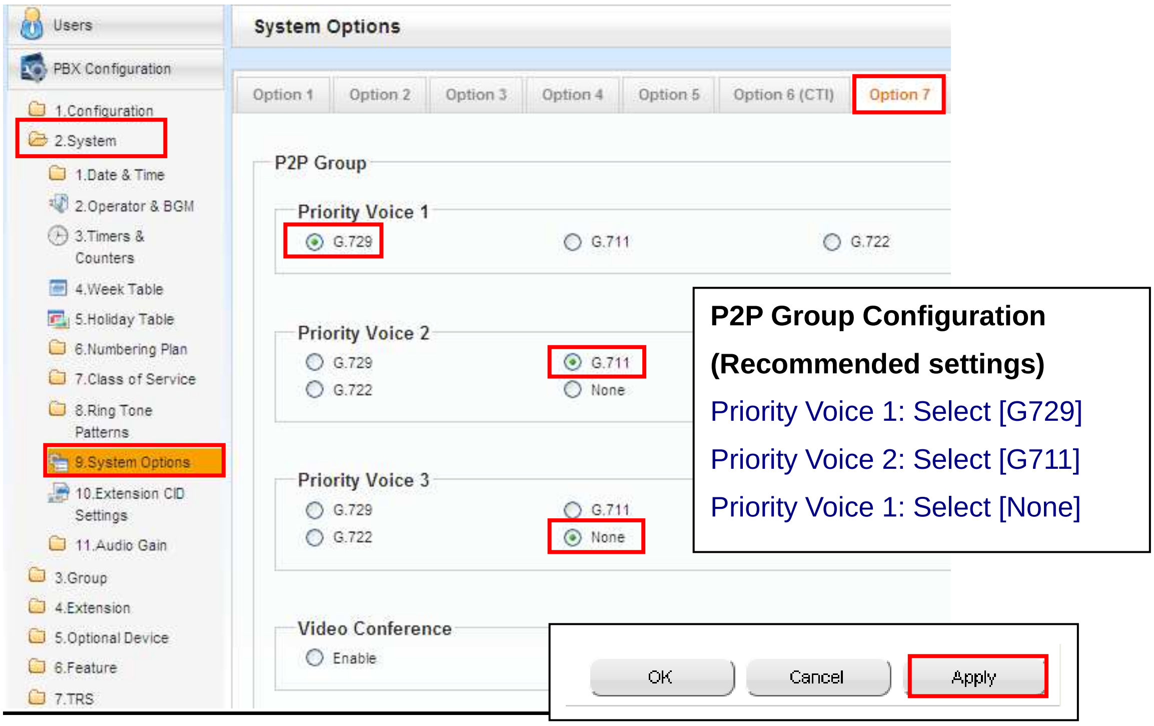
NAT - SIP Register Expire Time (s):**20** (This is a default value.)

4.13 Configure the P2P Group recommended settings. (Example)

*Note) We recommend that you consider the bandwidth of Internet access in each country, to change the priority of voice Codec G729 the remote side.

Click on [**2.System**] --> [**9.System Options**] --> [**Option7**]

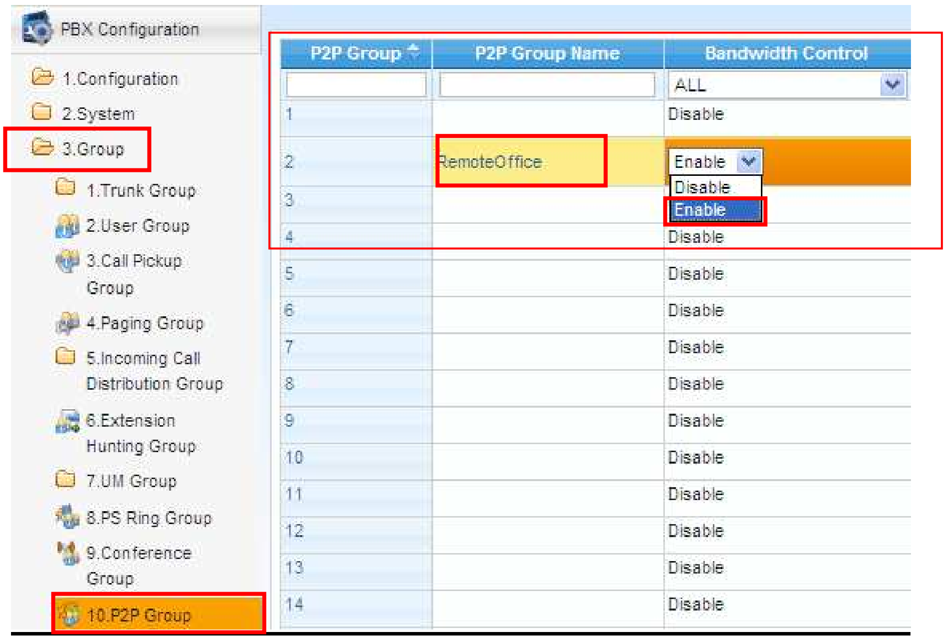
Configure each Priority Voice of P2P. --> Click on [**Apply**].



4.14 Click on [**3.Group**] --> [**10.P2P Group**] --> Enter Group Number "**2**" (Example)

Enter the P2P Group Name: **Remote Office** (Example)

Select [**Enable**] in the [**Bandwidth Control**] column for P2P Group that will be used at a remote site.



4.15 Click on [**Apply**] --> Click on [**OK**]

5. Procedure for Installing Remote SIP Phone (Remote V-UTEXT32).

There are 2 methods to install UT-SIP Phones (V-UTEXT32) at same local site as PBX and at remote site.

[Method 1]
Connect the UT-SIP Phone to the PBX, register the UT-SIP Phone to the PBX, and then configure remote V-UTEXT32 settings using Web Maintenance Console.

[Method 2]
Configure the UT-SIP Phone remote settings using the Web user interface of the UT-SIP Phone. You do not have to connect the UT-SIP Phone to the PBX when using this method.

* Note)
1. A KX-NS1000 can work with only one SBC. Also, multiple sites can share an SBC.

2. KX-UT series SIP Phones can communicate over a NAT (Network Address Translation)-enabled network only when communicating via an SBC from the KX-NS1000 to which the KX-UT series SIP Phones are registered.

3. When an SBC is in use, packets from P2P communication also go through the SBC. Therefore, the number of maximum calls is limited according to the maximum number of calls of the SBC.

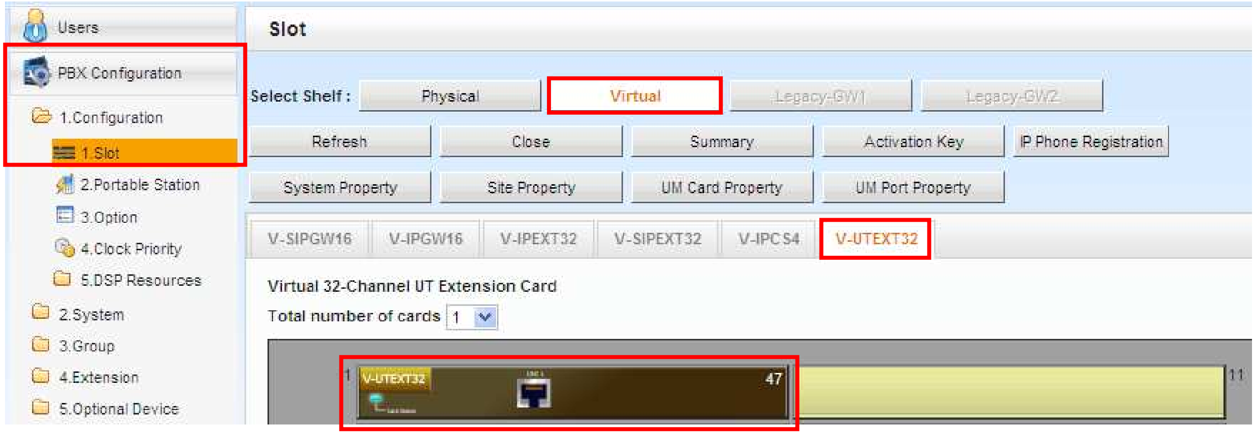
4. When installing KX-UT series SIP Phones at a remote site where the time zone is different, those KX-UT series SIP Phones will not match the Daylight Saving Time, and Time Display of the remote site. The KX-UT series SIP Phones will act according to the time setting of the KX-NS1000 to which the SIP Phones are registered.
However, if the KX-UT series SIP Phones are registered to a V-SIPEXT card and if an NTP server is specified by the SIP Phone, the Daylight Saving Time and Time Display match the KX-UT series SIP Phone setting.

5.1 Procedure for Method 1 with KX-UT Series SIP Phones.

Configure the V-UTEXT32 card for KX-UT series SIP Phone registration.

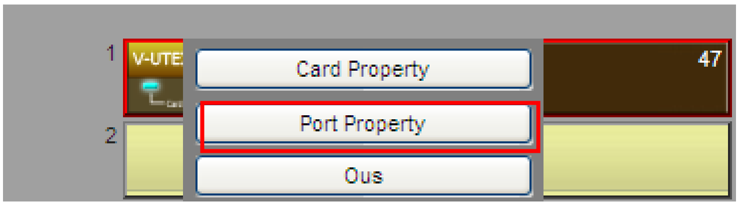
* Note) This procedure differs according to the IP Terminal Registration Mode already set to your KX-NS1000.For details about how to configure the V-UTEXT32 card in each mode, refer to "Installation Manual 5.9.1 Registering IP Telephones".

5.1-1 Click on [**PBX Configuration**] --> [**1.Configuration**] --> [**1.Slot**] --> [**Virtual**] --> [**V-UTEXT32**]

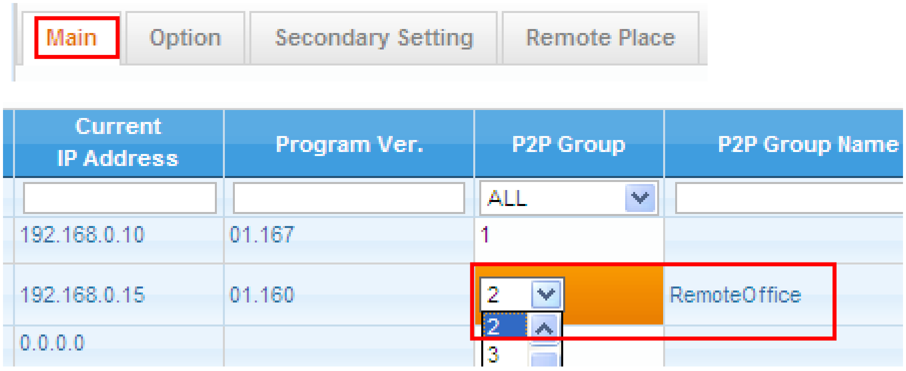


5.1-2 Move the mouse pointer over the **V-UTEXT32** card (Virtual UT Extension Card).

A menu will be shown under the mouse pointer. --> Click on [**Port Property**].



5.1-3 Select [**2**] (Example) in the P2P Group column for each UT-SIP Phone that will be used at a remote site.



5.1-4 Configure the [UT Codec Priority]

Click on [**Option**] tab

Select UT Codec Priority- **1st: G729A**

Select UT Codec priority- **4th: G722**

Main | Option | Secondary Setting | Remote Place

Extension Number	Extension Name	Connection	System Speed Dial Download	SIP QoS -DSCP	RTP QoS -DSCP	UT Codec Priority - 1st	UT Codec Priority - 2nd	UT Codec Priority - 3rd	UT Codec Priority - 4th	Packet Sampling Time
		ALL	ALL	ALL	AL	ALL	ALL	ALL	ALL	ALL
201		INS	Enable	0	0	G.722	G.711A	G.711Mu	G.729A	20ms
301		INS	Enable	0	0	G.729A	G.711A	G.711Mu	G.722	20ms

5.1-5 Configure the [**Remote Place**]

Click on [**Remote Place**] tab

Select [**Remote**] in [**Phone Location**] column for SIP Phone that will be used at a remote site.

Select [**HTTPs**] in [**Protocol for Remote SIP-MLT**] column for SIP Phone that will be used at a remote site.

Main | Option | Secondary Setting | Remote Place

ID	Site	Shelf	Slot	Port	Extension Number	Extension Name	Connection	Phone Location	Web-MC Ability	Protocol for Remote SIP-MLT
		ALL					ALL	ALL	ALL	ALL
1	1	Virtual	47	1	201		INS	Local	Enable	HTTP
2	1	Virtual	47	2	301		INS	Remote	Enable	HTTPs
3	1	Virtual	47	3	105		Fault	Local	Enable	HTTP
4	1	Virtual	47	4	106		Fault	Local	Enable	HTTP

5.1-6 Click on [**Apply**] and [**OK**]

OK | Cancel | Apply

5.1-7 Reboot UT-Phone by "**Power reset or RESET command**" with UT- Phone manually.

The UT-SIP Phone will download Remote site settings automatically.

5.1-8 Please wait until UT-SIP Phone is received the Remote Extension configuration.

The UT-SIP Phone will download the Remote Configuration parameters.

The UT-SIP Phone will be shown on display as following message.

Connection error (90002)
Check server and set it.

*** Note)**

Depending on the model of the existing Router, you may be able to connect to the PBX.

5.1-9 After confirming remote connection to the PBX, re-pack the KX-UT series SIP Phone, and then send it to the remote site. The UT-SIP Phone completed the settings.

Note:

When the KX-UT series SIP Phone is connected at the remote site, it should start normally.
If the KX-UT series SIP Phone cannot connect normally, import the configuration file of "UT_ACS_HTTPS_01NS1000.cfg or UT_ACS_NS1000.cfg" again as with "Procedure for method 2" using the Web user interface after initialize.

5.1-10 Unpack the UT-SIP Phone and connect it to the LAN.

The UT-SIP Phone will connect to the Head office PBX via SBC.

And, the UT-SIP Phone will be shown as following on display (Example).

9 OCT 12:00 TUE **301**

5.1-11 Please check the Basic outgoing and incoming calls.

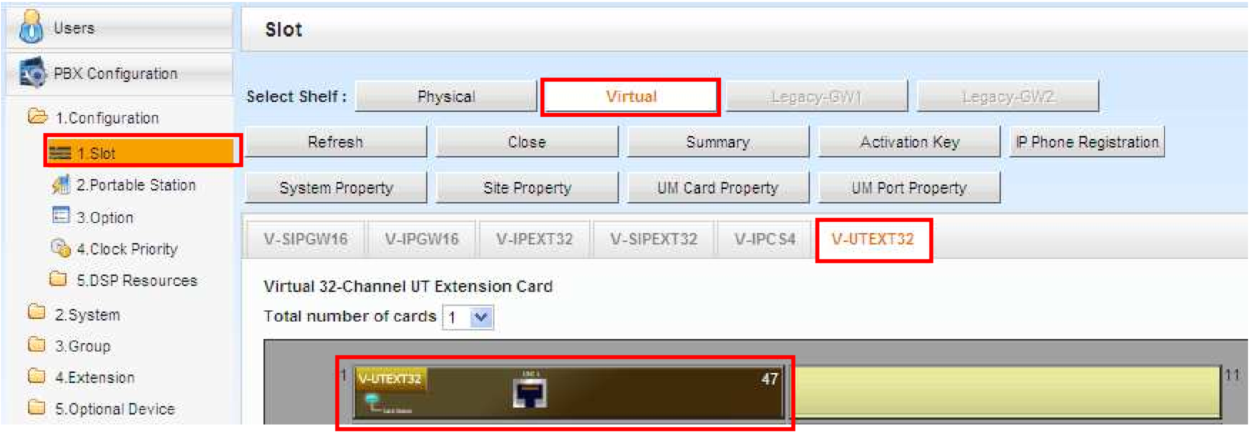
5.2 Procedure for Method 2 with KX-UT Series SIP Phones.

* Note)

Configure the SIP Phone remote settings using the Web user interface of the SIP Phone.

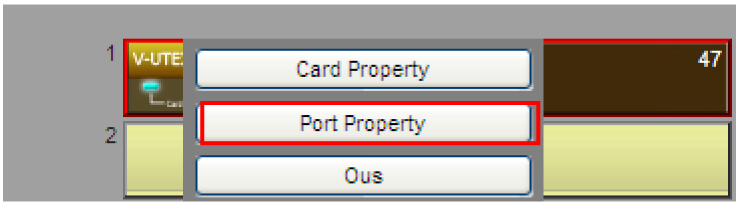
You do not have to connect the SIP Phone to the PBX when using this method.

5.2-1 Click on [**PBX Configuration**] --> [**1.Configuration**] --> [**1.Slot**] --> [**Virtual**] --> [**V-UTEXT32**]

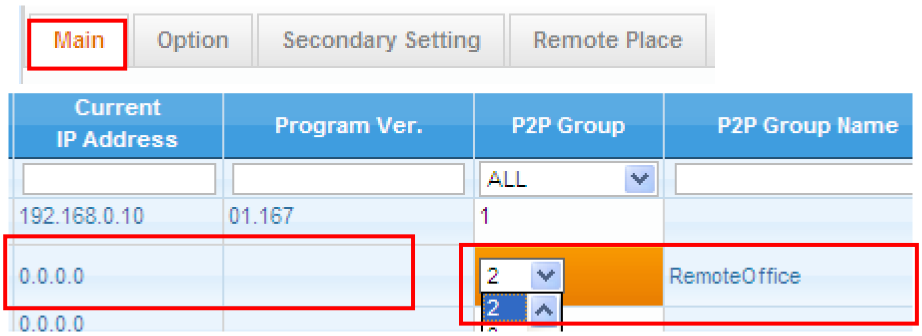


5.2-2 Move the mouse pointer over the **V-UTEXT32** card (Virtual UT Extension Card).

A menu will be shown under the mouse pointer. --> Click on [**Port Property**].



5.2-3 Select [**2**] (Example) in the P2P Group column for each UT-SIP Phone that will be used at a remote site.

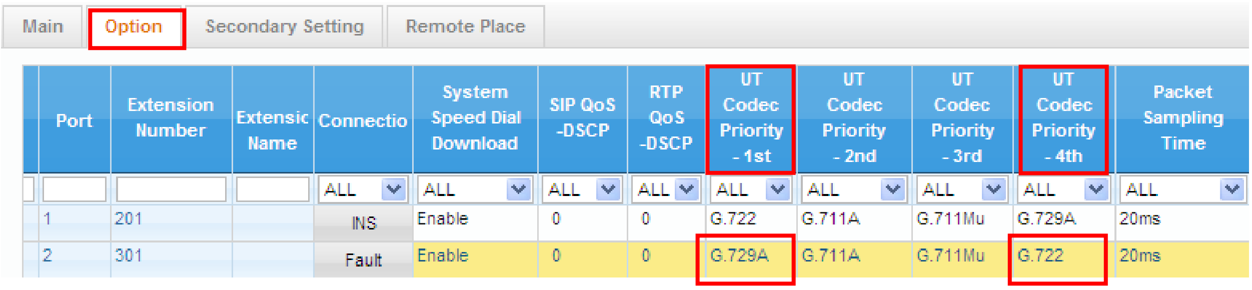


5.2-4 Configure the [**UT Codec Priority**]

Click on [**Option**] tab

Select UT Codec Priority- **1st: G729A**

Select UT Codec priority- **4th: G722**

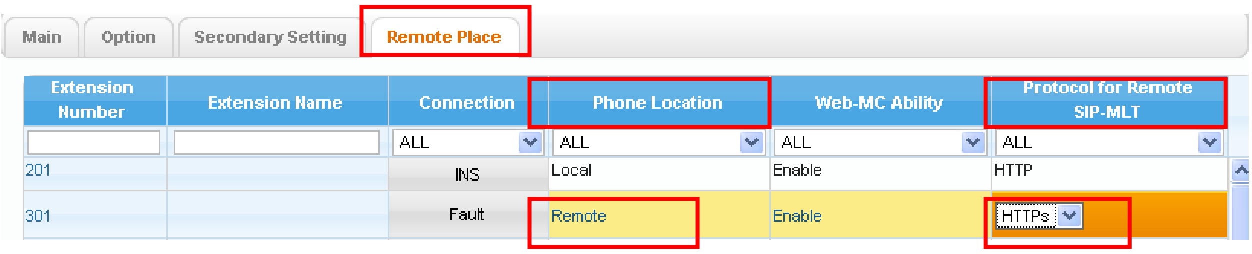


5.2-5 Configure the [**Remote Place**]

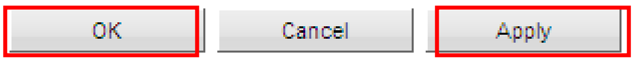
Click on [**Remote Place**] tab

Select [**Remote**] in [**Phone Location]** column for SIP Phone that will be used at a remote site.

Select [**HTTPs**] in [**Protocol for Remote SIP-MLT**] column for SIP Phone that will be used at a remote site.



5.2-6 Click on [Apply]



5.2-7 Save the System Data.

Click on [**Save System Data**] 

5.2-8 Have to make the [**UT_ACS_HTTPS_01NS1000.cfg**] file on using NS1000.

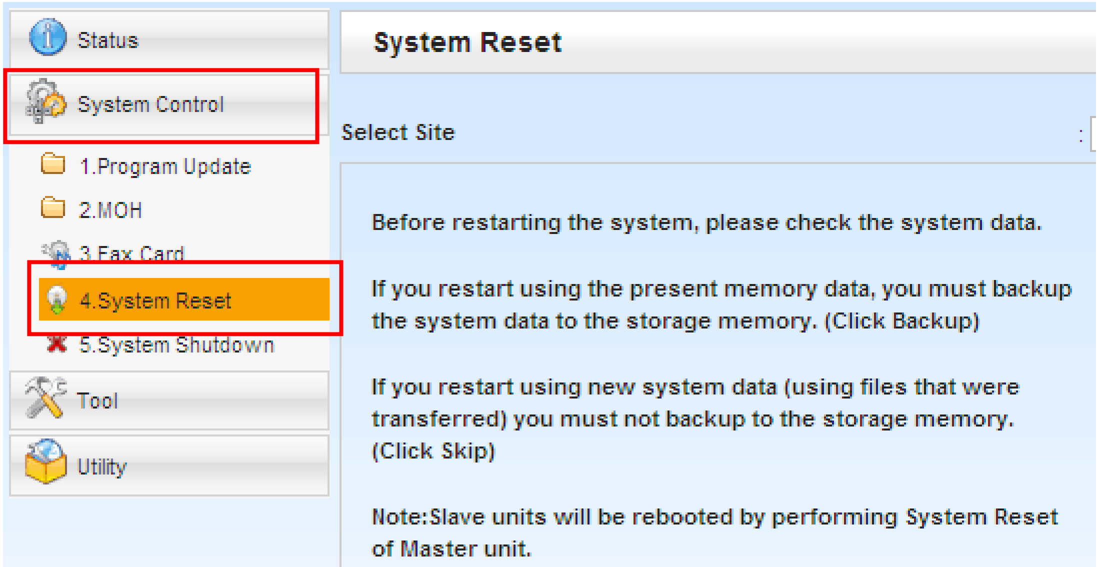
Note)* The [UT_ACS_HTTPS_01NS1000.cfg] is made at system startup.

Therefore we have to reboot the NS-1000 only once, but we do not need this every time.

(Need when configure the [Setting parameters assigned to Remote SIP-MLT] Site Property)

Click on [**Maintenance**] -->

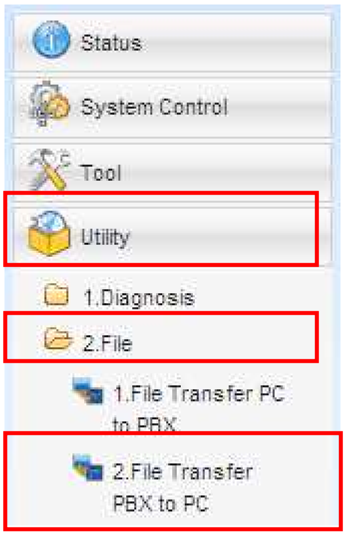
Click on [**System Control**] --> [**4.System Rest**] --> [**Backup**] (Just in case) --> [**OK**] --> [**OK**]



5.2-9 Access the KX-NS1000 Web **Maintenance Console page again**, after the PBX re-starting.

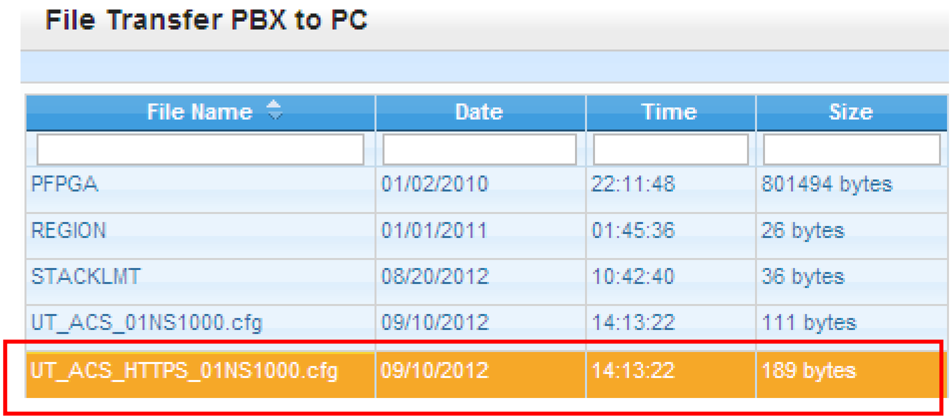
5.2-10 Access the [Maintenance Page]

Click on [**Utility**] --> [**2.File**] --> [**2.File Transfer PBX to PC**]



5.2-11 Click on [**Next Page**] and Please find the [**ACS_File**].

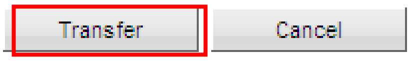
5.2-12 Click on [**UT_ACS_HTTPS_01NS1000.cfg**] line.



File Transfer PBX to PC

File Name	Date	Time	Size
PFPGA	01/02/2010	22:11:48	801494 bytes
REGION	01/01/2011	01:45:36	26 bytes
STACKLMT	08/20/2012	10:42:40	36 bytes
UT_ACS_01NS1000.cfg	09/10/2012	14:13:22	111 bytes
UT_ACS_HTTPS_01NS1000.cfg	09/10/2012	14:13:22	189 bytes

5.2-13 Click on [**Transfer**]



5.2-14 Save as to in Maintenance's PC folder.

File Name: **UT_ACS_HTTPS_01NS1000.cfg**

5.2-15 Distribute to the PC to install this file.

5.2-16 The NS1000 completed for the settings of remote UT-SIP extension.

Next, we have to access to UT-SIP Phone web setting page.

5.2-17 Allow to access the UT SIP Phone's web page

Enter [**Setting**] on UT-SIP Phone --> Enter [**#**], [**5**], [**3**], [**4**] --> Select [**On**] --> [**Enter**]

5.2-18 Confirm the assigned IP address for UT-SIP Phone.

Click on [**Setting**] on UT-SIP Phone --> Select [**Information Display**] --> [**Enter**]
--> Select [**IP Address**] **192.168.10.1** (Example) --> Enter [**CANCEL]** Key
Access the UT-SIP Phone web page. **http://192.168.10.1/** (Example)

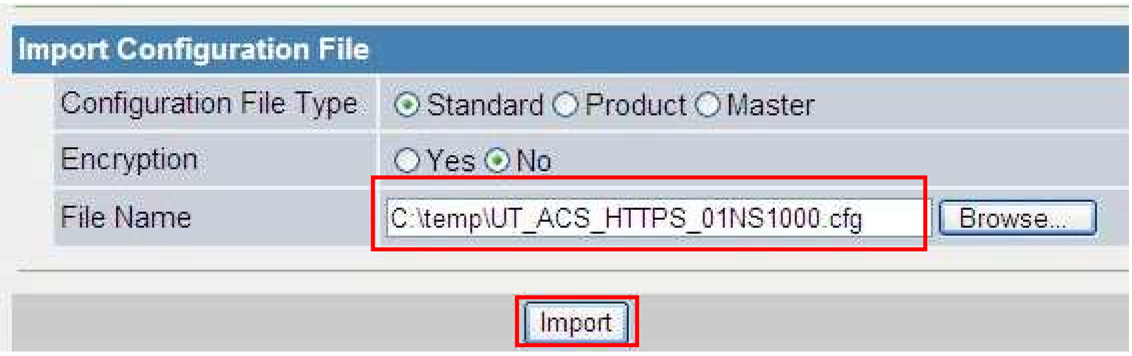
5.2-19 [Operator Login]

Username: **instoperatoruserid**
Password: **instpass**

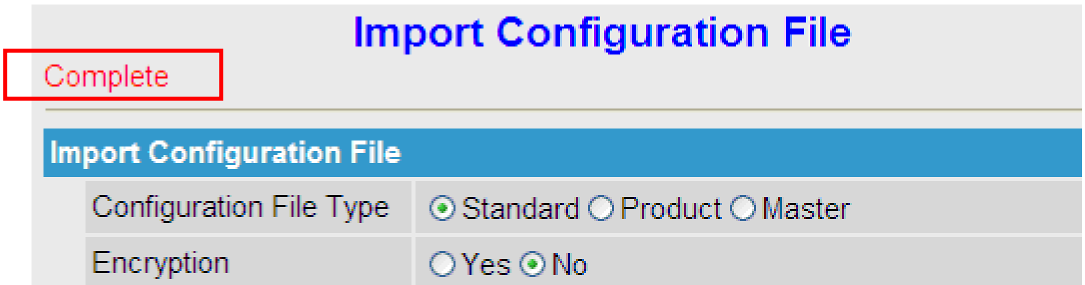
5.2-20 Click on [**Maintenance**] and then Click on [**Browse…**]

5.2-21 Find and Select the [**UT_ACS_HTTPS_01NS1000.cfg**] file

And Click on [**Import]**



5.2-22 Confirm [**Complete**]



You will now download the remote UT-SIP Phone (V-UTEXT32) configuration.

5.2-23 Register the UT-SIP Phone (V-UTEXT32) by NS1000 registration.

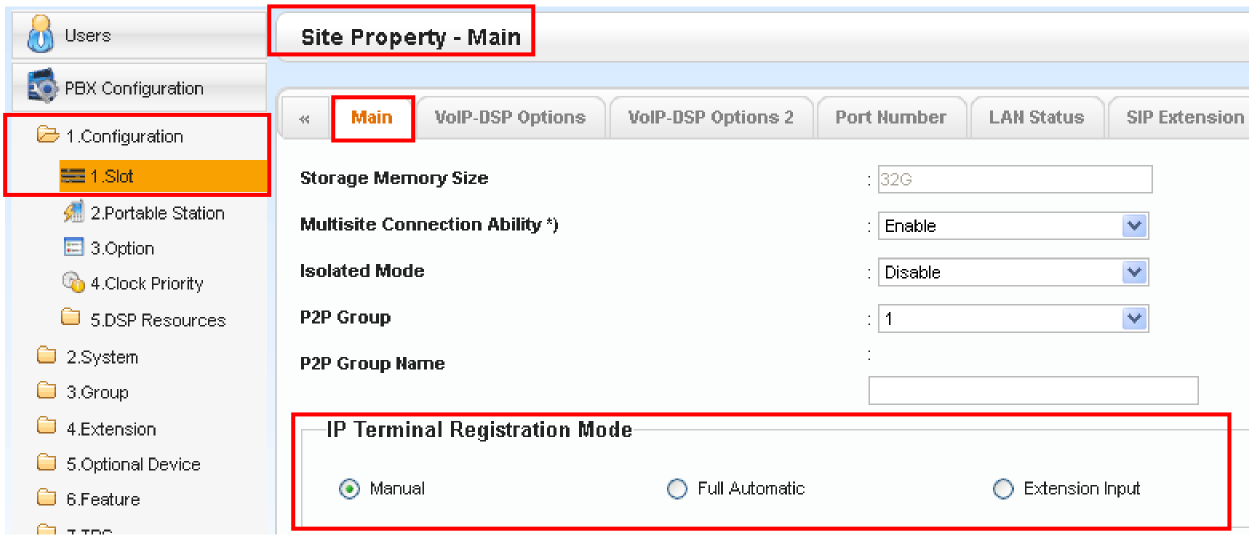
5.3 Registering IP Telephones

After the programming of the PBX and IP telephones is finished (refer to "5.8 Assigning Networking Information to IP Telephones" in the Installation Manual), the IP telephones must be registered to the PBX.

The procedure for registering IP telephones differs according to the IP terminal registration mode specified during the Easy Setup Wizard. This setting can also be changed in the Site Property - Main screen of the Web Maintenance Console (refer to "9.5.1 PBX Configuration - [1-1] Configuration - Slot - Site Property - Main - Main - IP Terminal Registration Mode" in the PC Programming Manual).

Refer to the following table:

	1.Full Automatic mode	2. Extension Input mode	3. Manual Mode
UT Series (V-UTEXT32)	**Yes**	**No**	**Yes**



5.4 Full Automatic Mode

If networking settings have been completed, when IP-PTs or KX-UT series SIP Phones are connected to the same network as the PBX, they will be registered automatically. No registration procedure is required.

5.5 Extension Number Input Mode

For KX-UT Series SIP Phones

If networking settings have been completed, when KX-UT series SIP Phones are connected to the same network as the PBX, they will be registered automatically as same as when they are registered in Full Automatic mode. No registration procedure is required.

***Note)**

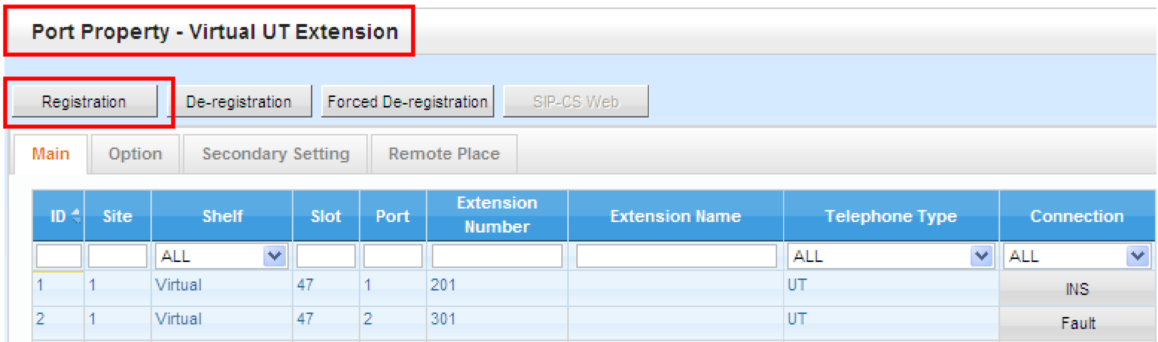
UT series do not support “Extension Input Mode”, so even if you set registration mode to “Extension Input Mode”, the way of registration is same as “Full Automatic Mode”.

Please refer “Full Automatic” explanation.

5.6 Manual Mode

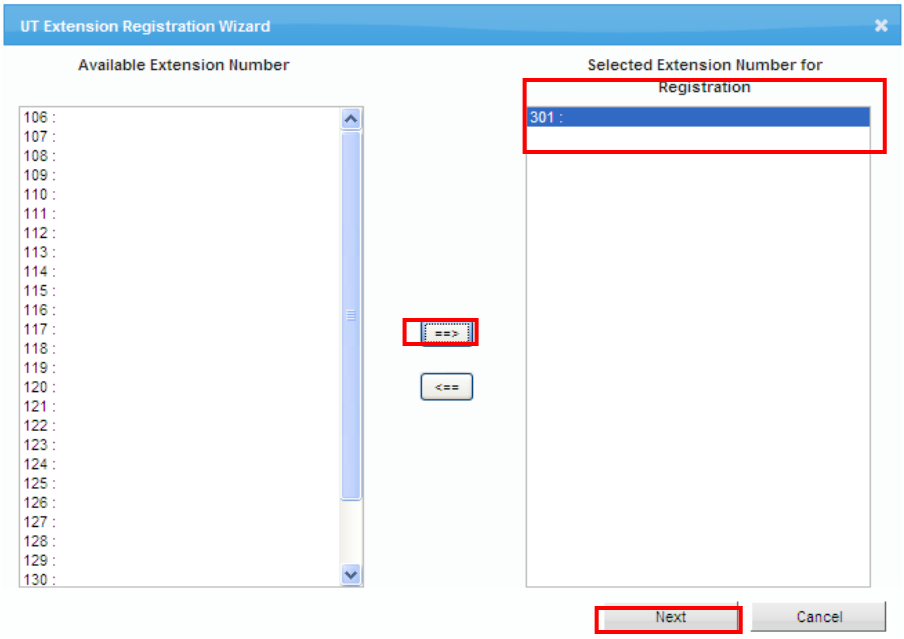
5.6-1 Manual Mode (Example)

Select the Port Property – Virtual UT Extension --> Click on [**Registration**]

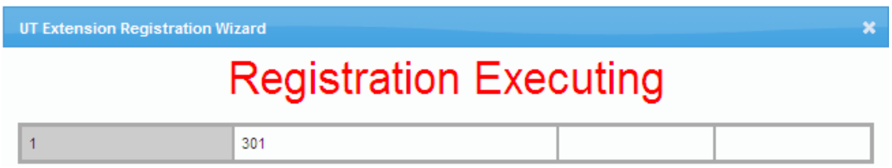


5.6-2 **Select Extension Number** for Registration (Example)

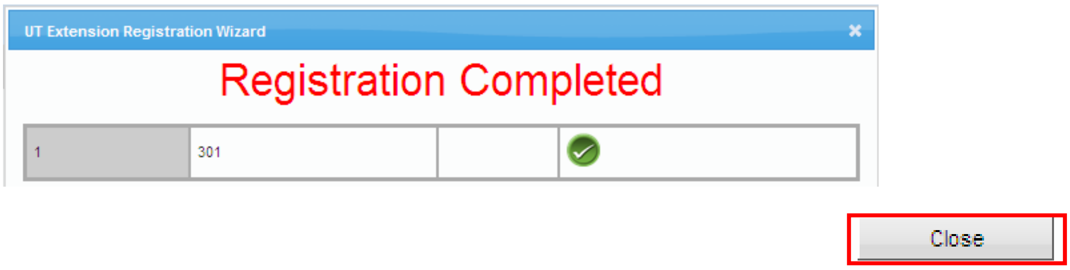
Click on [**Next**] --> [**Next**]



5.6-3 Wait a [**Registration Executing**]



5.6-4 Confirm [**Registration Completed**] and click on [**Close**]

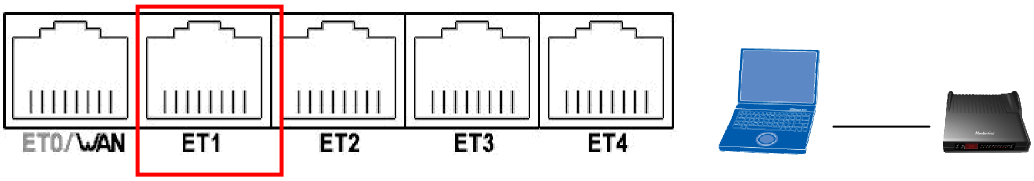


6. Initial setting of the Mediatrix SBC (Mediatrix 500 series)

6.1 In Preparation of Network

6.1-1 The SBC has a default IP address of 192.168.0.1, Subnet mask: 255.255.255.0

Connect the ET1 of SBC and maintenance PC Network directly.



The SBC's DHCP server function is running with the SBC, it's default setting.

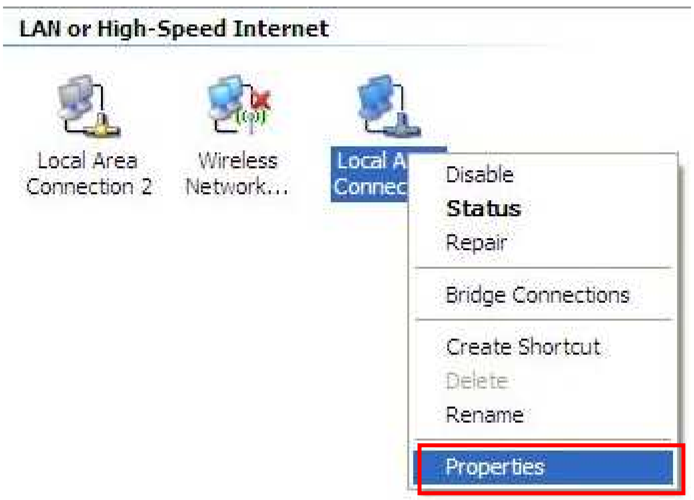
In this document, the Network setting is described using obtain an IP configuration automatically.

As a matter of course you can use static IP address.

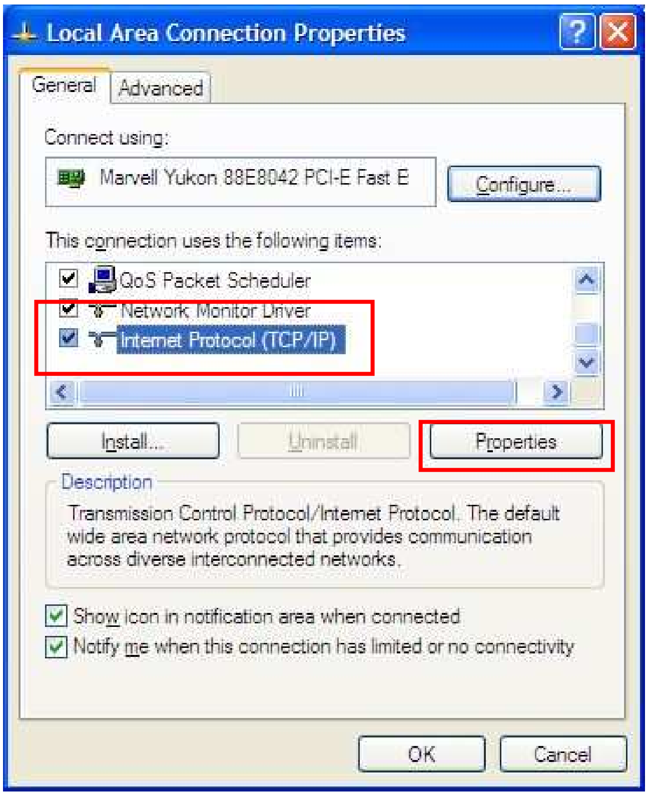
6.1-2 Confirmation of PC LAN settings to allow setup of Mediatrix SBC

[**View Network Connections**] Select the LAN in use.

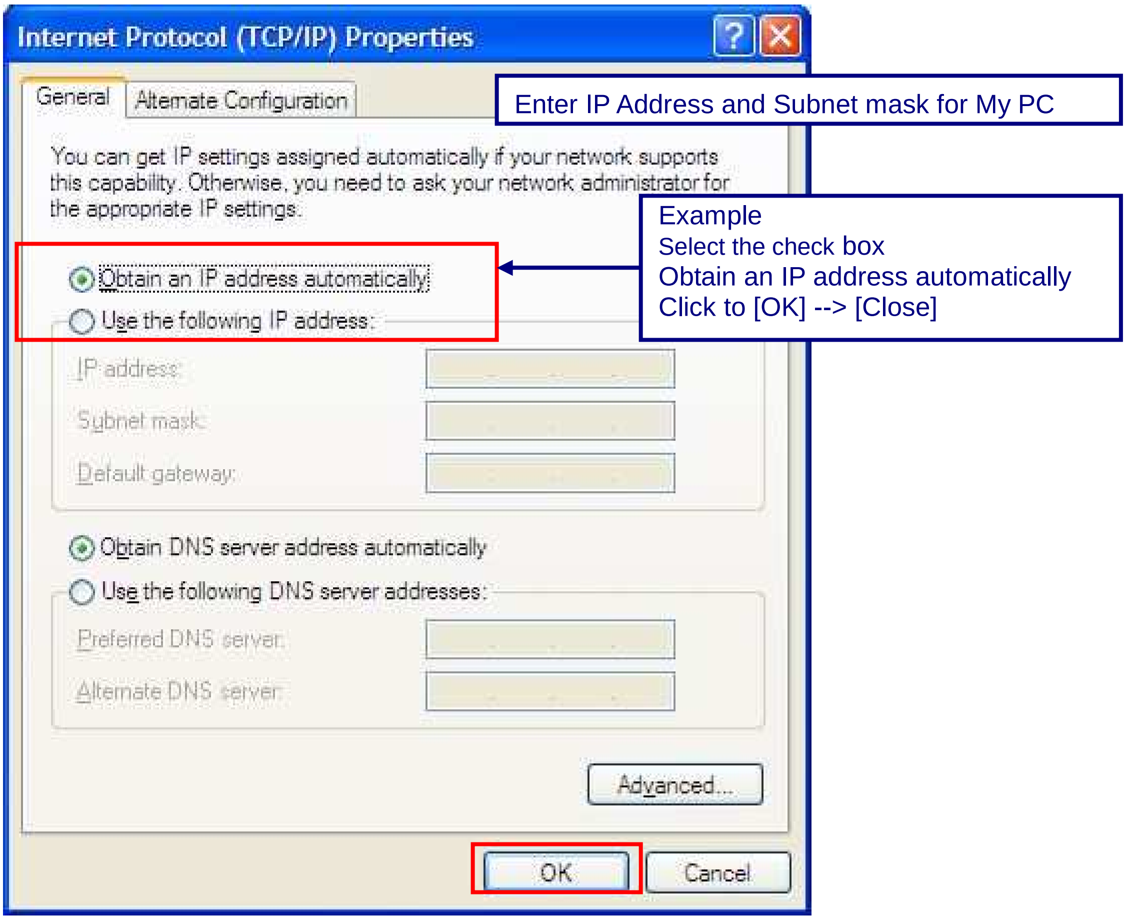
6.1-3 [**Local Area Connection Properties**] – Right click and Select the [**Properties**].



6.1-4 Select [**Internet Protocol (TCP/IP)**] and Click on [**Properties**].



6.1-5 Confirmation of Network Properties and Click on [**OK**].



6.1-6 Program start up [**Command Prompt**] (Start --> Accessories --> Command Prompt)

--> Enter "**ipconfig /all**" and to check the **currently IP Address**.

```
Command Prompt

C:\>ipconfig /all

Windows IP Configuration

        Host Name . . . . . . . . . . . . : PC-PCC09060398E
        Primary Dns Suffix  . . . . . . . :
        Node Type . . . . . . . . . . . . : Unknown
        IP Routing Enabled. . . . . . . . : No
        WINS Proxy Enabled. . . . . . . . : No
        DNS Suffix Search List. . . . . . : local.lan

Ethernet adapter Wireless Network Connection:

        Media State . . . . . . . . . . . : Media disconnected
        Description . . . . . . . . . . . : Intel(R) WiFi Link 5100 AGN
        Physical Address. . . . . . . . . : 00-22-FA-98-E8-30

Ethernet adapter Local Area Connection:

        Connection-specific DNS Suffix  . : local.lan
        Description . . . . . . . . . . . : Marvell Yukon 88E8042 PCI-E Fast Eth
ernet Controller
        Physical Address. . . . . . . . . : 00-24-81-58-78-EE
        Dhcp Enabled. . . . . . . . . . . : Yes
        Autoconfiguration Enabled . . . . : Yes
        IP Address. . . . . . . . . . . . : 192.168.0.31
        Subnet Mask . . . . . . . . . . . : 255.255.255.0
        Default Gateway . . . . . . . . . : 192.168.0.1
        DHCP Server . . . . . . . . . . . : 192.168.0.1
        DNS Servers . . . . . . . . . . . : 192.168.0.1
        Lease Obtained. . . . . . . . . . : Friday, July 13, 2012 1:10:39 PM
        Lease Expires . . . . . . . . . . : Friday, July 13, 2012 5:10:39 PM

C:\>_
```

6.1-7 Enter [**ping 192.168.0.1**] then confirm the replying from the Mediatrix SBC.

```
C:\>ping 192.168.0.1

Pinging 192.168.0.1 with 32 bytes of data:

Reply from 192.168.0.1: bytes=32 time<1ms TTL=128
Reply from 192.168.0.1: bytes=32 time<1ms TTL=128
Reply from 192.168.0.1: bytes=32 time<1ms TTL=128
Reply from 192.168.0.1: bytes=32 time<1ms TTL=128

Ping statistics for 192.168.0.1:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
Approximate round trip times in milli-seconds:
    Minimum = 0ms, Maximum = 0ms, Average = 0ms
```

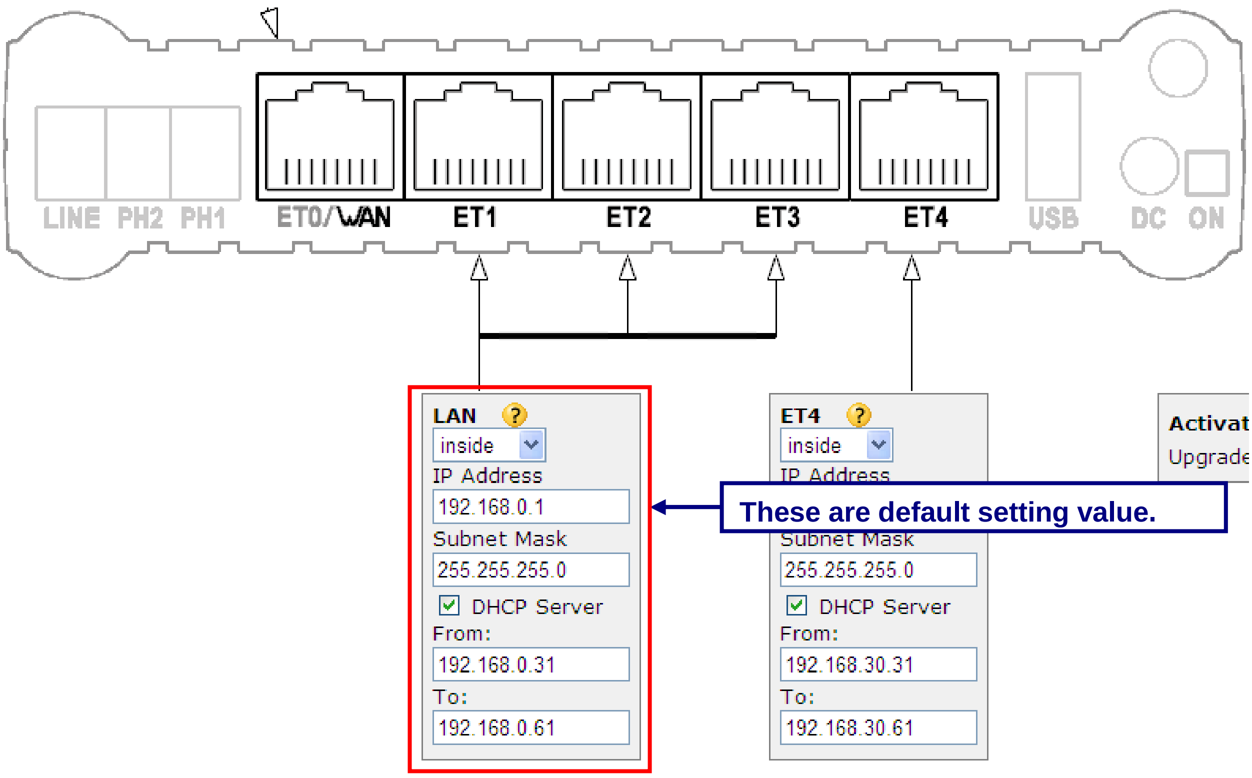
6.2 In Network Configurations (1)

6.2-1 Access to Web Home and Click on [**Log in**]. Example **http://192.168.0.1/**

6.2-2 To Enter Network Password Username: **admin** / Password: **admin** (Default).

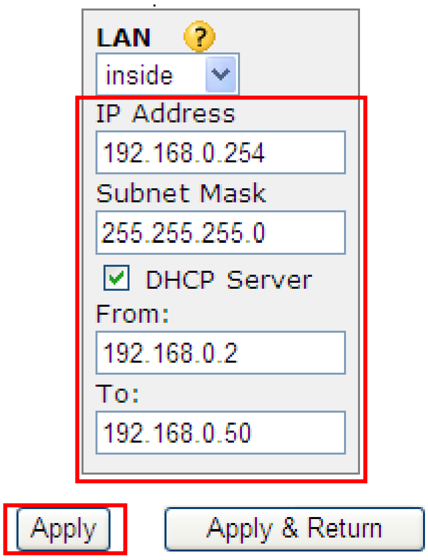
6.2-3 Access to initial web page (HOME) --> Click on [**Network**]

6.2-4 Confirmation of LAN settings.



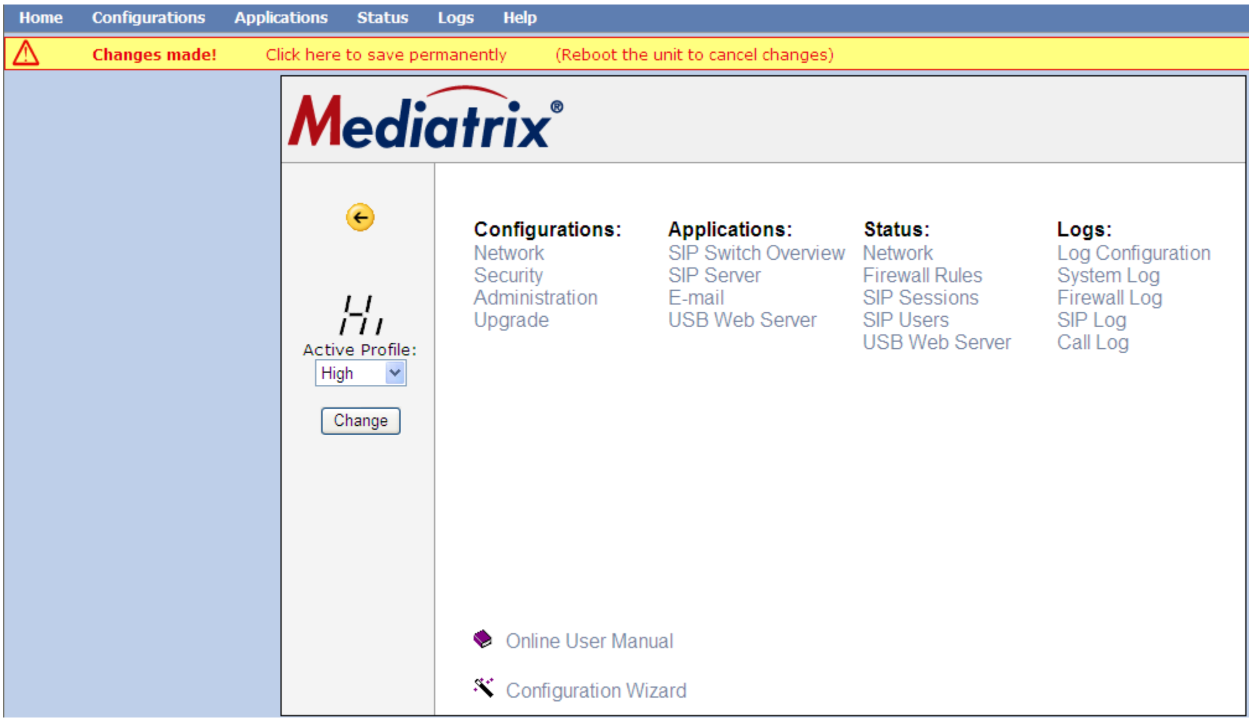
6.2-5 Configure **LAN IP Address**, **Subnet Mask and DHCP Server Range** (From To).

Click on [**Apply**]

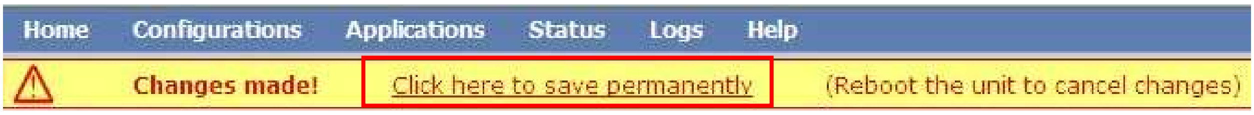


6.2-6 Access to Web Home with New IP address and Click on **[Log in**].

Example **http://192.168.0.254/**

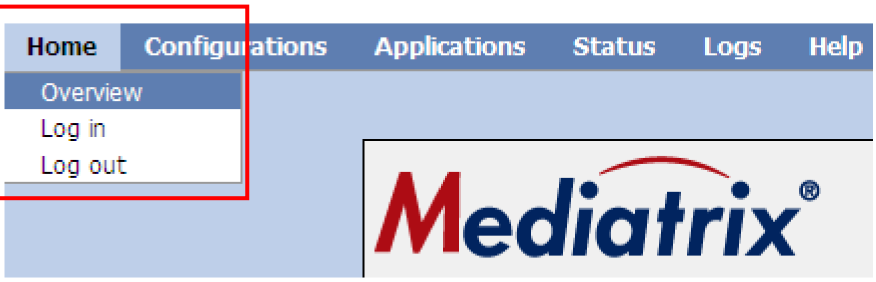


6.2-7 Click on [**Click here to save permanently**]



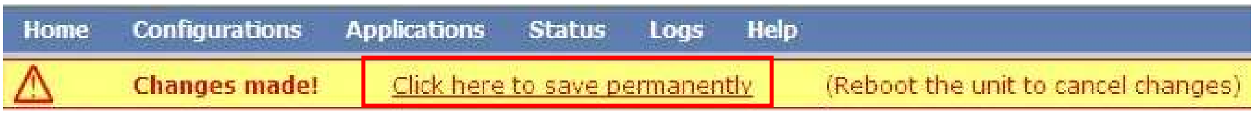
6.3 In Network Configurations (2)

6.3-1 Move mouse over [**Home**] and Select [**Overview**]

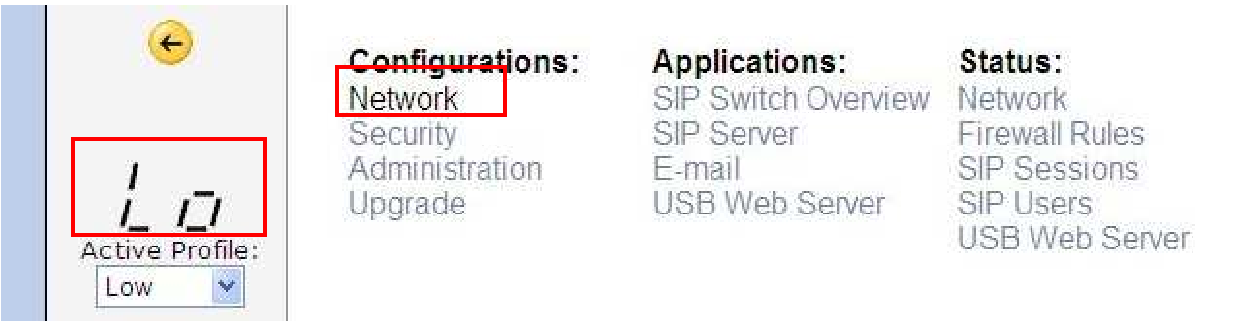


6.3-2 Select Active Profile: [**Low**] and Click on [**Change**]

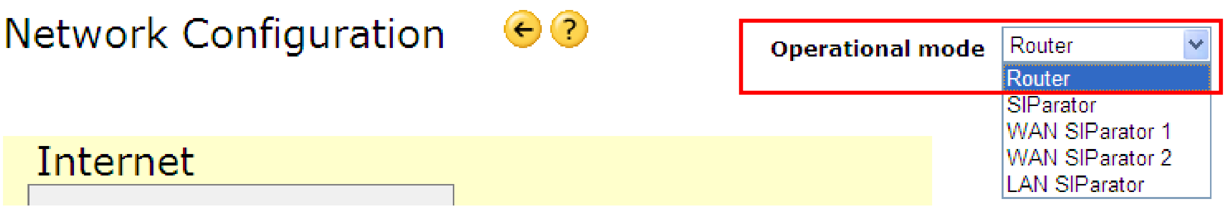
6.3-3 Click on [**Click here to save permanently**] and then Click on [**Network**].



5.3-4 Confirmation of Active Profile: **[Lo]** and Click on [Network].



6.3-5 Confirmation or Selection of Operational mode: **[Router]**



6.3-6 Network Configuration

[ET0 Settings]-- Select the Access type: [**PPPoE**]

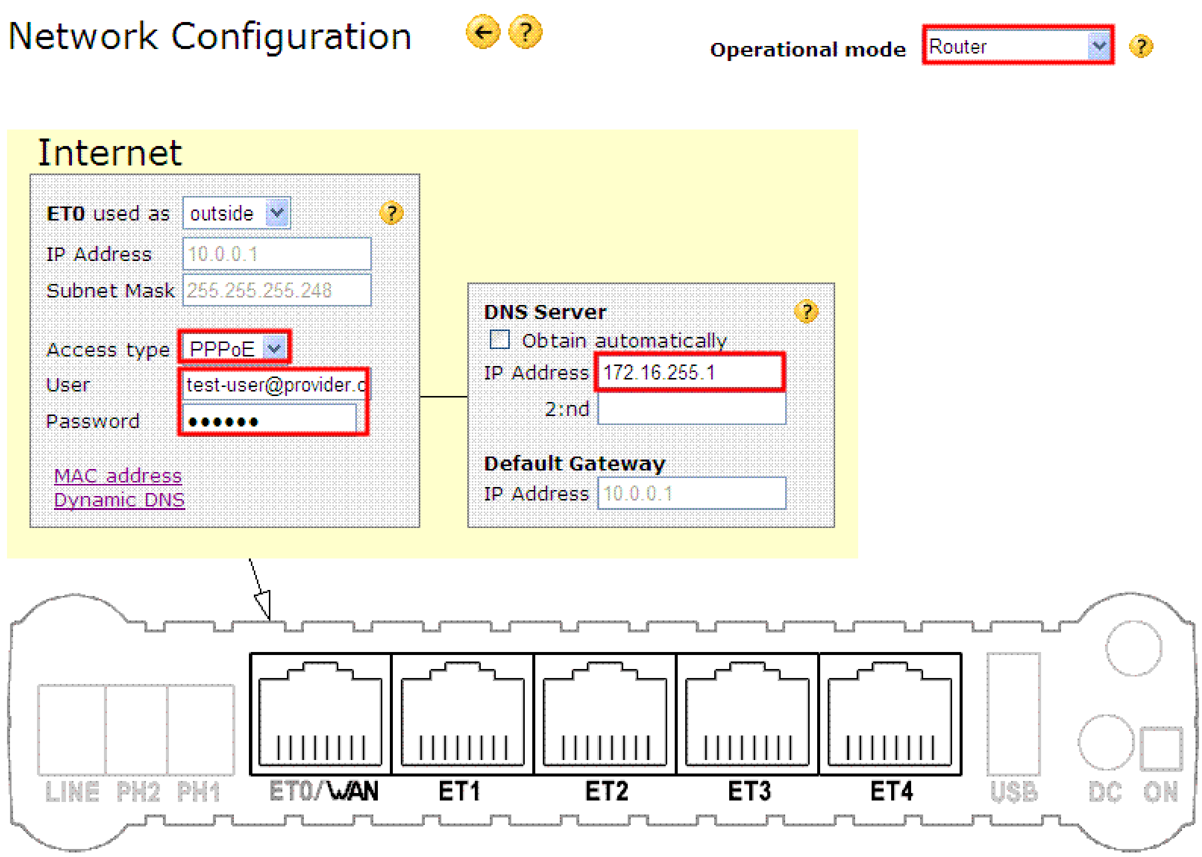
-- User: **test-user@provider.com**

-- Password: **abc123**

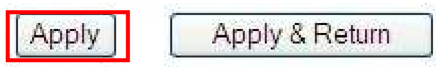
[DNS Server]

-- Clear the check box [Obtain automatically] Example (if it be provided)

-- IP Address: **172.16.255.1** Example, (Change to Global IP)

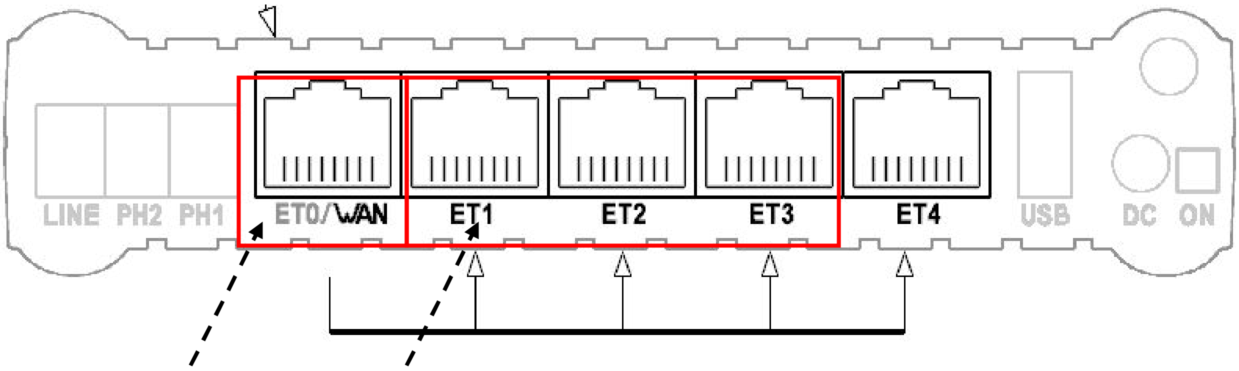


6.3-7 Click on [**Apply**]



6.3-8 Click on [**Click here to save permanently**]

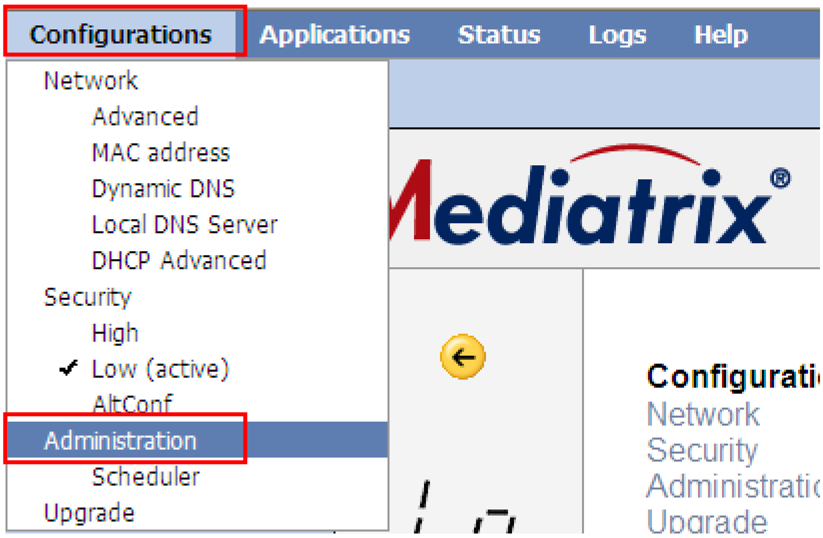
6.3-9 Connect to SBC **ET1** (to ET4) and Maintenance PC for existing LAN segment.



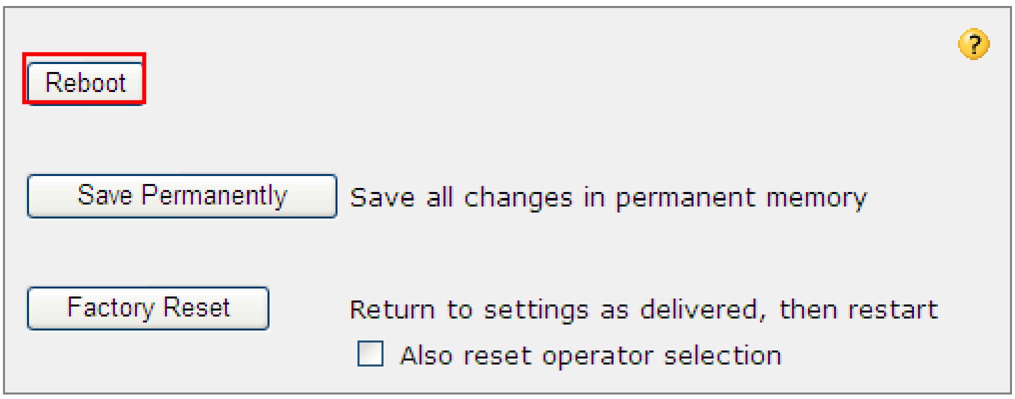
[Note] Connect the WAN. **Connect to the LAN**

6.3-10 Reboot the SBC (Recommended operations)

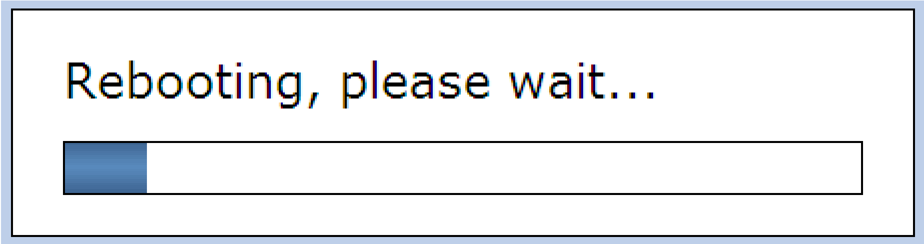
Select [**Configurations**] --> [**Administration**]



6.3-11 Click on [**Reboot**]

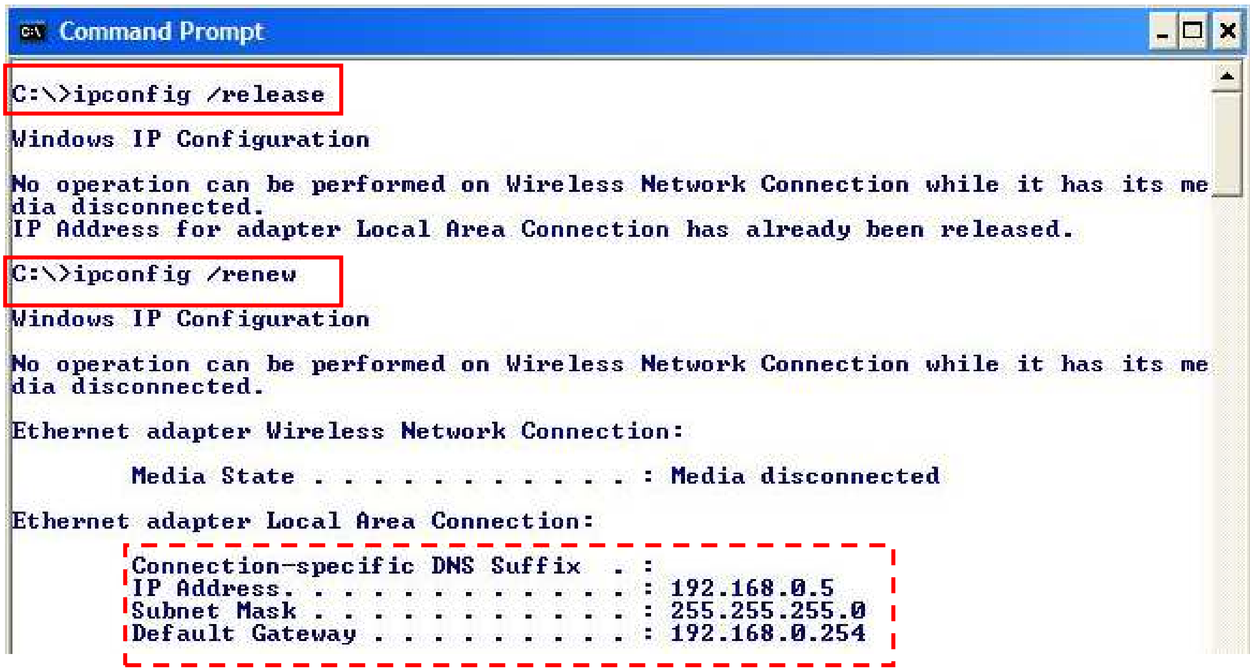


6.3-12 Wait for Rebooting.

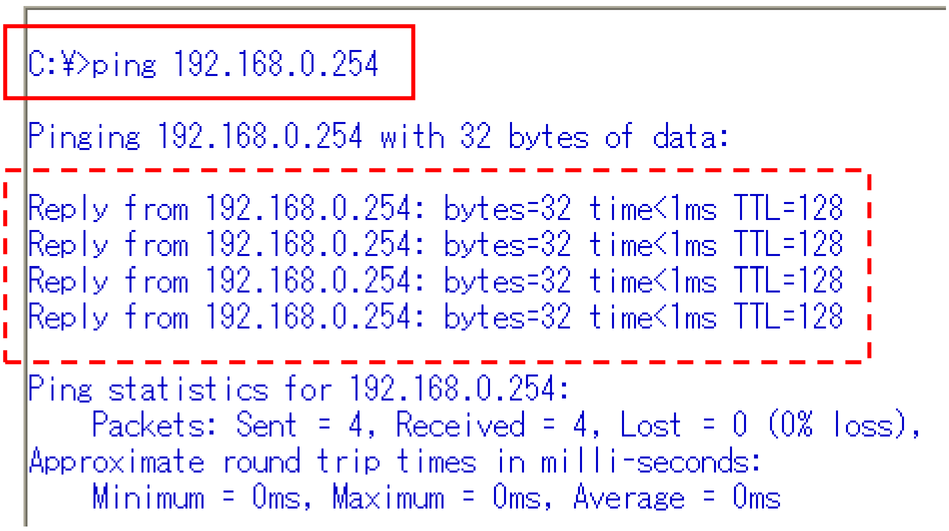


6.3-13 After rebooting --> Configure the IP Address, execute the release and renew.

(Example, dynamic addressing)



6.3-14 Enter the **ping 192.168.0.254** on Command Prompt. ---> Confirmation of Reply.



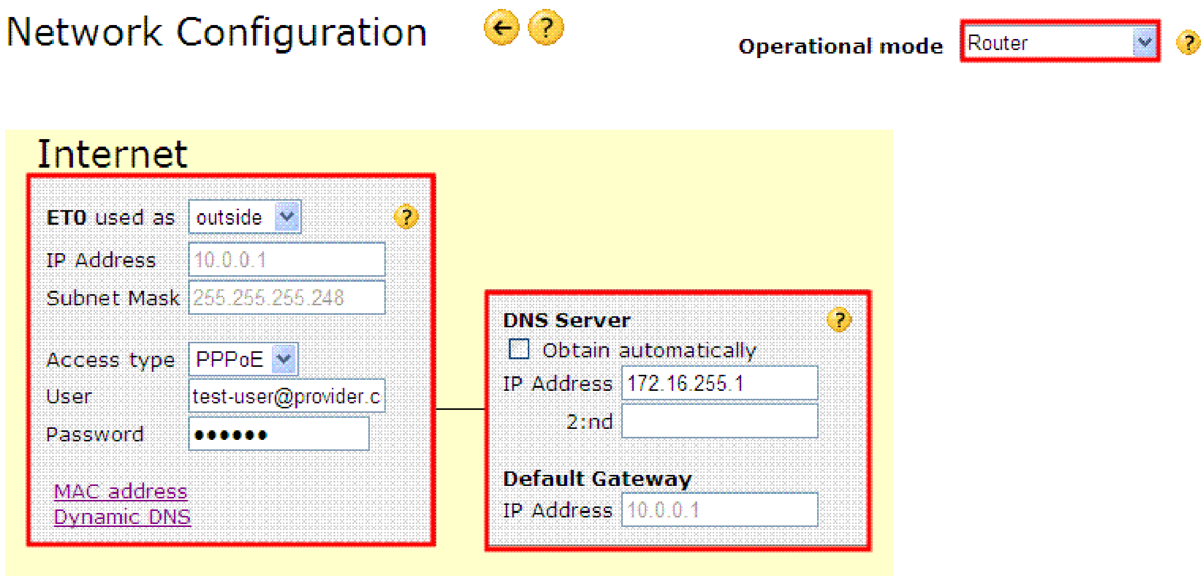
6.3-15 Confirm the Network Configuration

Access to web using new IP address and login again. http:// 192.168.0.254/ (Example)

Click on Configuration [**Network**]

And then confirmation of Operational mode: **[Router]**

And ET0 settings / DNS / Default Gateway / SIP Routing Trough Extern Firewall settings.



Network Configuration

[ET0 Settings]-- Select the Access type: [**PPPoE**]

-- User: **test-user@provider.com**

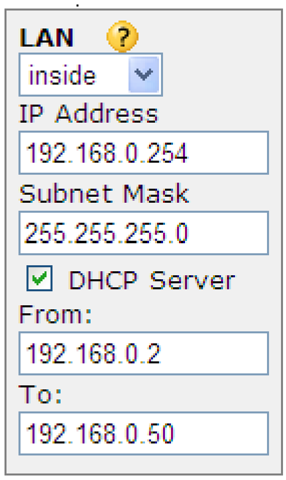
-- Password: **abc123**

[DNS Server]-- IP Address: **172.16.255.1** Example, (Change to Global IP)

LAN Configuration

[IP Address]: **192.168.0.254** / Subnet Mask **255.255.255.0** (Example)

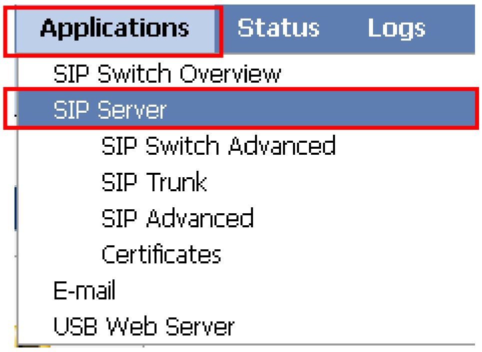
[DHCP Server]: Enable / From: **192.168.0.2 – 192.168.0.50** (Example)



[Note] Need to factory-reset the SBC if you need to select the operational mode after once select it.

6.4 In SIP Server Setting

6.4-1 Move Mouse over [Applications] --> and Select [SIP Server]



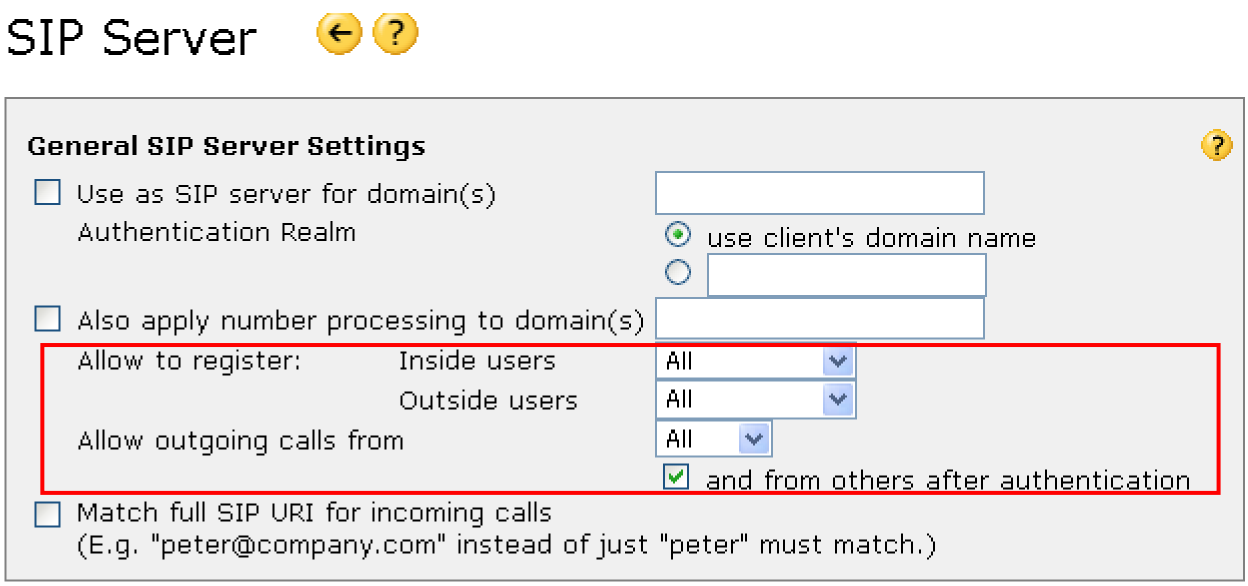
6.4-2 Select Allow to register

-1. Inside Users: [**All**] (Default)

-2. Outside Users: [**All**]

-3. Allow outgoing calls from: [**All**]

-4. Select the **check box** [**and from others after authentication**] (Default)

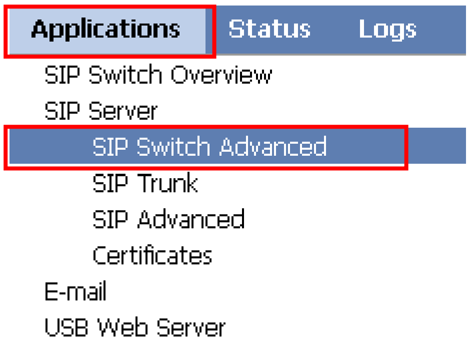


6.4-3 Click on [**Apply**]

6.4-4 Click on [**Click here to save permanently**]

6.5 In SIP Switch Advanced

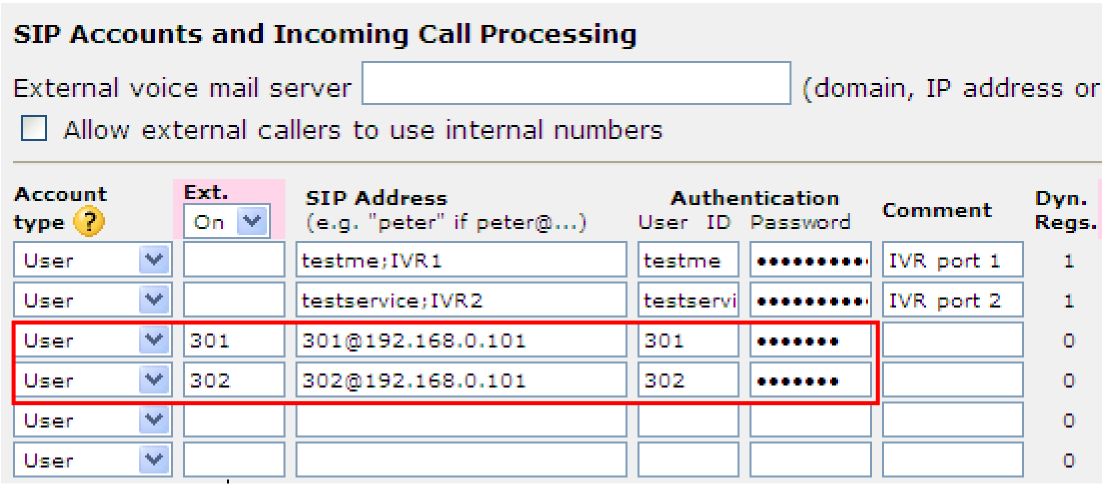
6.5-1 Move Mouse over [**Applications**] --> and Select [**SIP Switch Advanced**]



6.5-2 Enter the Authorized User. Example

Ext: 301/ SIP Address: 301@192.168.0.101/ User ID: 301/ Password: 1234

Ext: 303/ SIP Address: 302@192.168.0.101/ User ID: 302/ Password: 1234

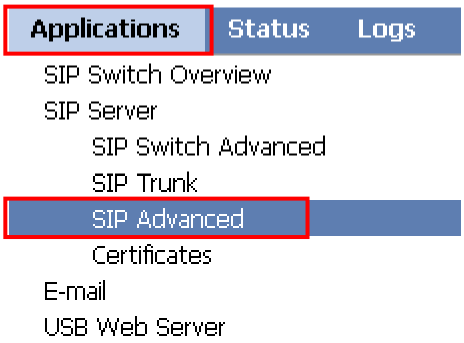


6.5-3 Click on [**Apply**]

6.5-4 Click on [**Click here to save permanently**]

6.6 In SIP Advanced

6.6-1 Move Mouse over [**Applications**] --> and Select [**SIP Advanced**]



6.6-2 Configuration of Advanced SIP Settings

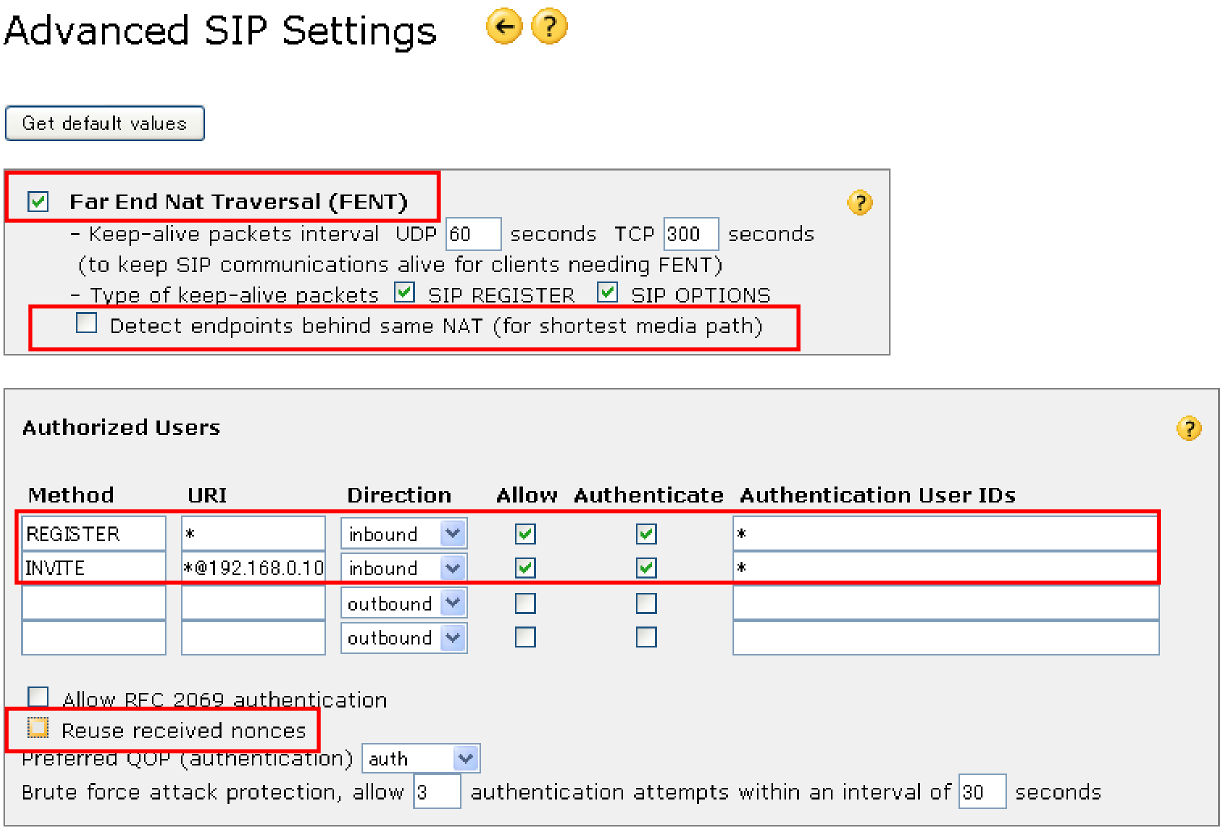
-1 Select the **check box** [**Far End Nat Traversal (FENT)**]

-2 **Clear the check box** [**Detect endpoints behind same NAT (for shortest media path)**]

-3 Enter the Authorized Users:

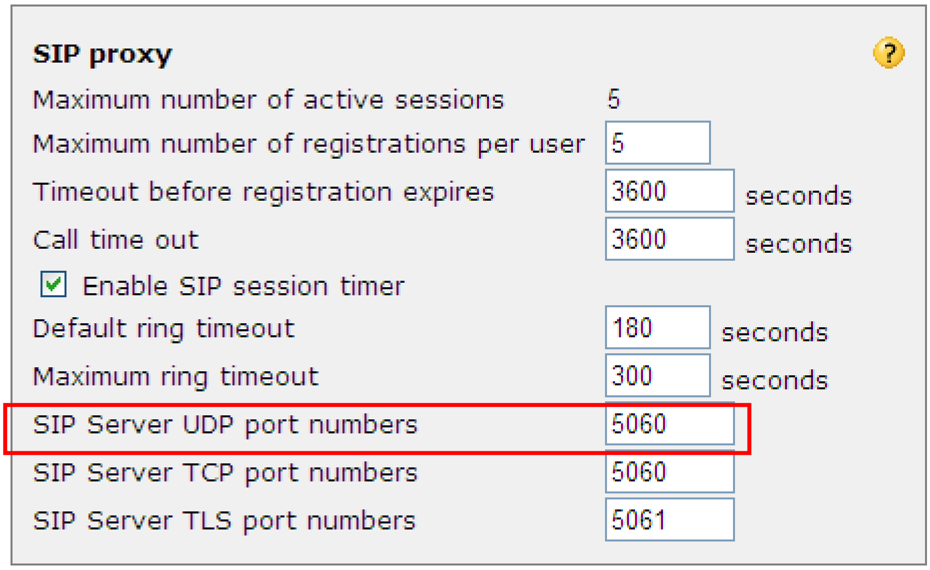
Method	URI	Direction	Allow/Authenticate	Authentication User ID
REGISTER	*	Inbound	Check / Check	*
INVITE	*@192.168.0.101	Inbound	Check / Check	*

-4 **Clear the check box** [**Reuse received nonces**]



6.6-3 Configuration of SIP Proxy

Enter the SIP Server UDP port number: **5060** (Default: 5060)

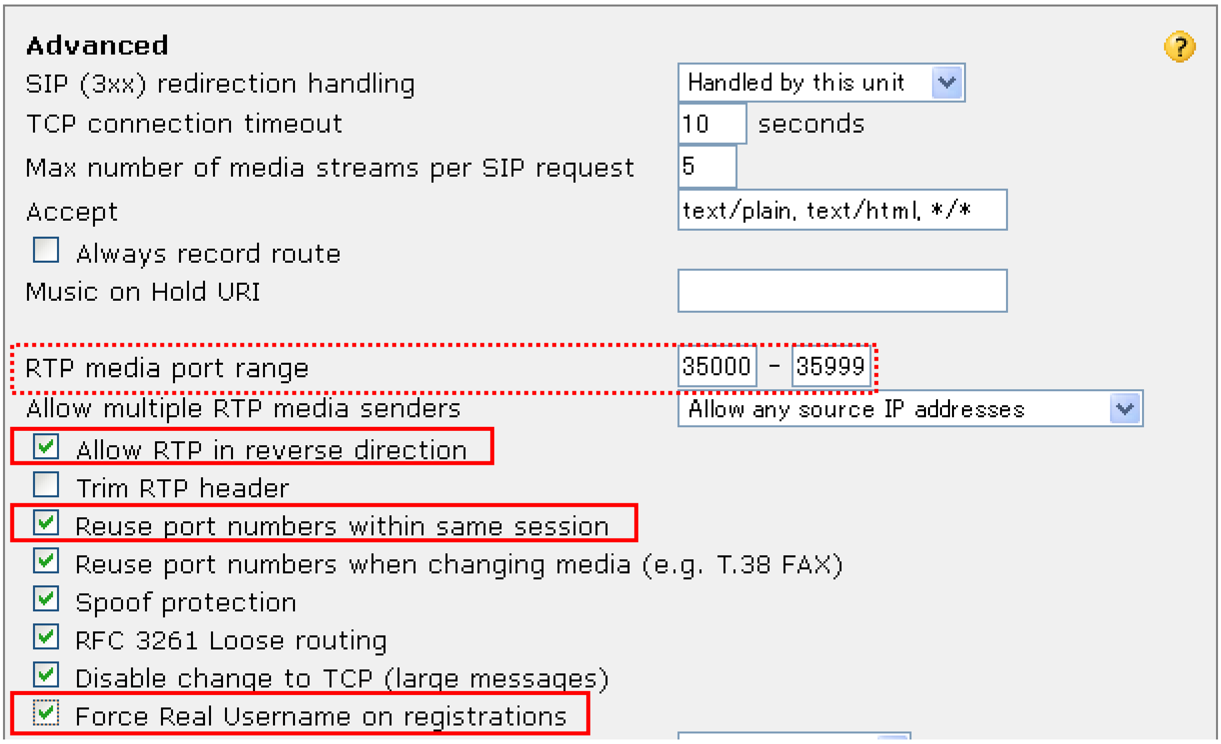


6.6-4 Configuration of Advanced and you can confirm the RTP port range in this page.

-1 Select the **check box** [**Allow RTP in reverse direction**]

-2 Select the **check box** [**Reuse port numbers within same session**]

-3 Select the **check box** [**Force Real Username on registrations**]

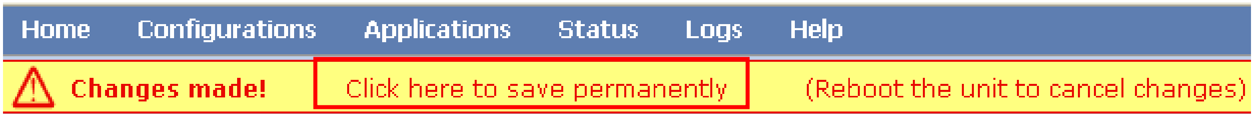


6.6-5 Configuration of Trusted networks

Clear the check box [**Enable**]

6.6-6 Click on [**Apply**]

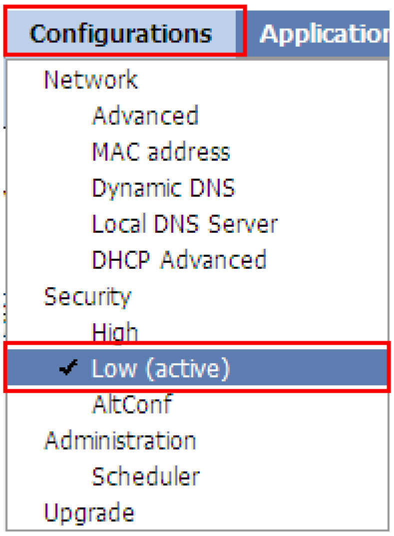
6.6-7 Click on [**Click here to save permanently**]



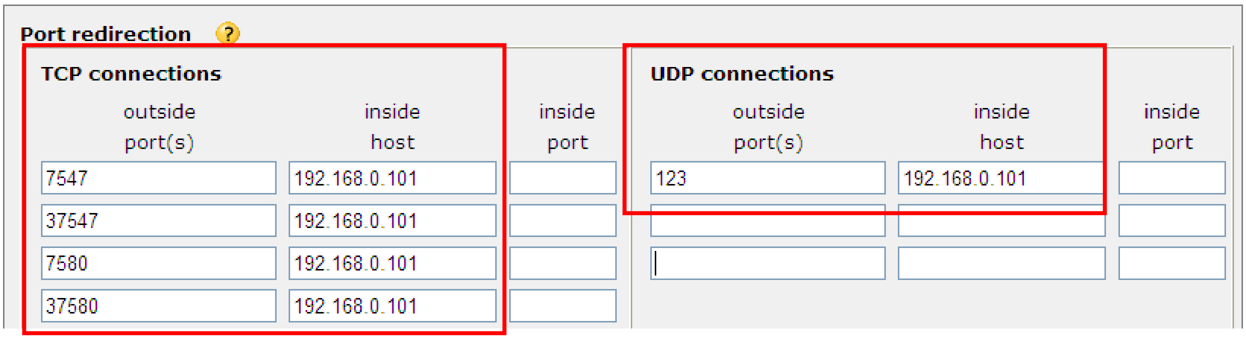
6.7 Configure the Port Redirection

6.7-1 Move the mouse pointer over the [**Configurations**].

A menu will shown under the mouse pointer --> Click on [**Security Low (active)**].



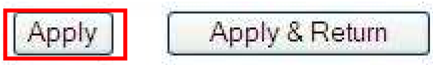
6.7-2 Configure Port redirection into the NS1000 for TCP connections and UDP connections.



Contents of Port redirection (also known as Port forwarding) Settings (Example)

Protocol	Out side Port (s)	Inside Host	Description
CWMP(HTTP)	7547 (TCP)	192.168.0.101	Send CWMP to PBX (PBX LAN IP address)
CWMP(HTTPS)	37547 (TCP)	192.168.0.101	Send CWMP to PBX (PBX LAN IP address)
SIP-MLT Data Download(HTTP)	7580 (TCP)	192.168.0.101	Send Data to PBX (PBX LAN IP address)
SIP-MLT Data Download(HTTPS)	37580 (TCP)	192.168.0.101	Send Data to PBX (PBX LAN IP address)
NTP	123 (UDP)	192.168.0.101	Send NTP to PBX (PBX LAN IP address)

6.7-3 Click on [**Apply**]



6.7-4 Click on [**Click here to save permanently**]

7. Operation

Try the basic calls.
We confirm the following operation by settings in this Reference Guide.

7-1 Incoming Call and making Call

The Caller ID is displayed on the LCD screen of Panasonic UT-SIP Extension and SIP Extension.

7-2 Conversation with G.722 G.711 and G.729

The more than single codec is already set in KX-NS1000 (V-SIPEXT)

7-3 Holding Call and retrieving Call held

These features are confirmed by KX-NS1000 control.

7-4 Transferring Call

The transferring Calls are confirmed by KX-NS1000 control.

7-5 Call forwarding (V-UTEXT32 Registered)

These features are confirmed by KX-NS1000 control.
*** Note*** This feature does not work as using registration of SIP extension (V-SIPEXT32).
Restriction on the use of standard SIP Extension (V-SIPEXT32).

8. How to register 3rd party SIP Phones

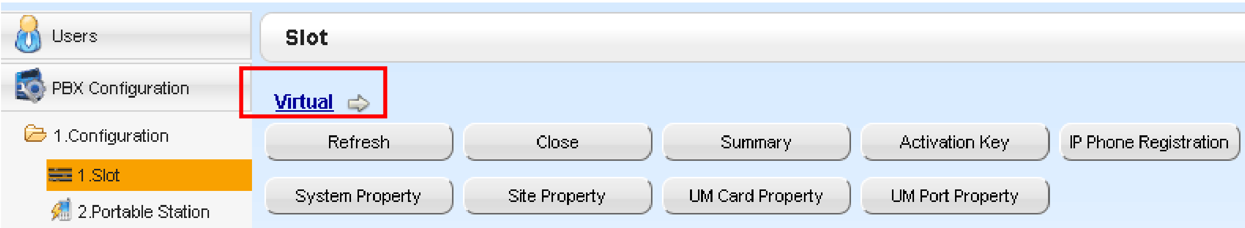
Procedure for Installing Remote SIP Phone (Remote V-SIPEXT32) if required.

This PBX supports the use of 3rd party SIP Phones connected from a remote office over an IP network through an SBC.
SIP Phones can be set up by simply connecting the Phones to the LAN at the remote office.
If the customer has needs, we can register the 3rd party SIP Phones.
For example, the Media5fone. They have to set a registration method of V-SIPEXT.

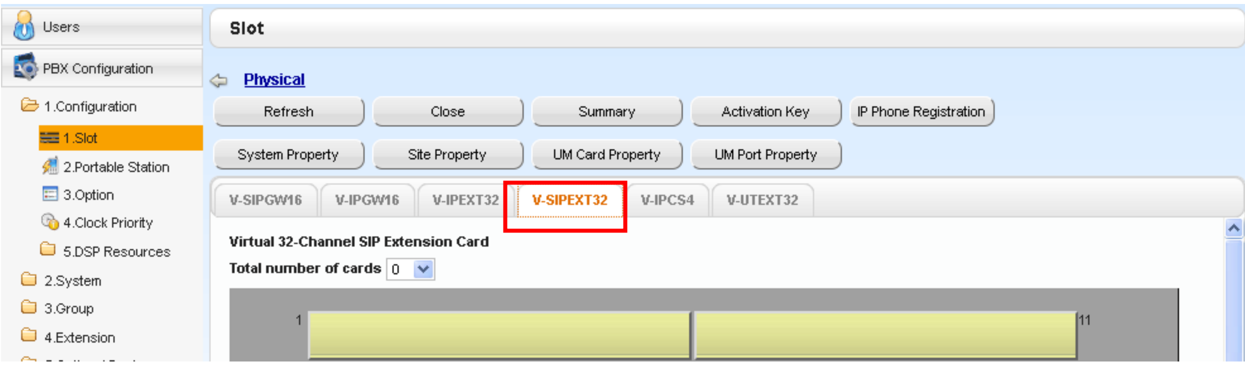
8.1 How to make the new SIP Extension (Example)

8.1-1 Configuration of the SIP Extension into the IP-PBX.

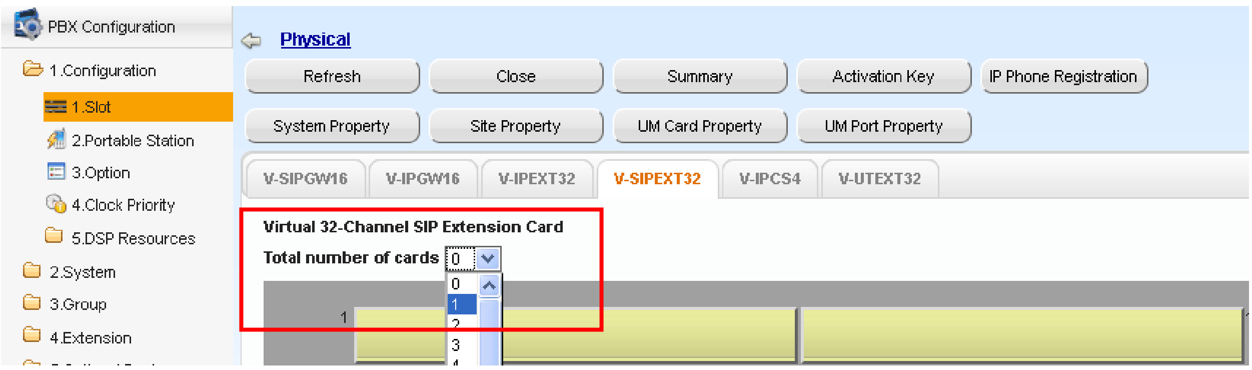
Click on [**Virtual**]



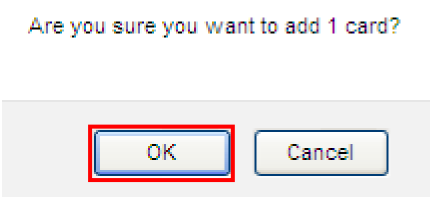
8.1-2 Click on [**V-SIPEXT32**]



8.1-3 Click on [**Total number of cards**] and Select: **1** (Example)



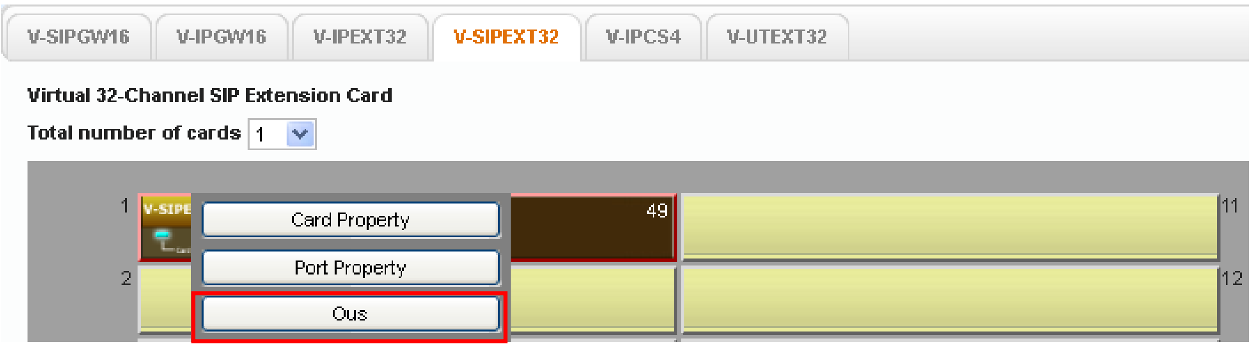
8.1-4 Click on [**OK**].



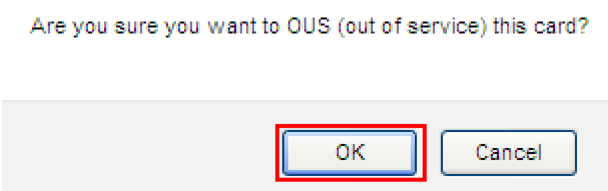
8.1-5 Configuration of V-SIPEXT32 Virtual slot.

Move mouse over installed [**V-SIPEXT32**] card

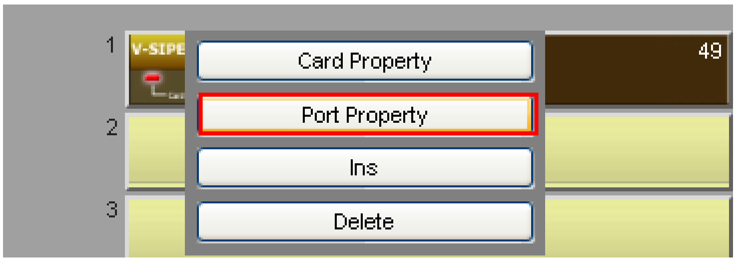
8.1-6 Select [**OUS**]



8.1-7 Click on [**OK**].

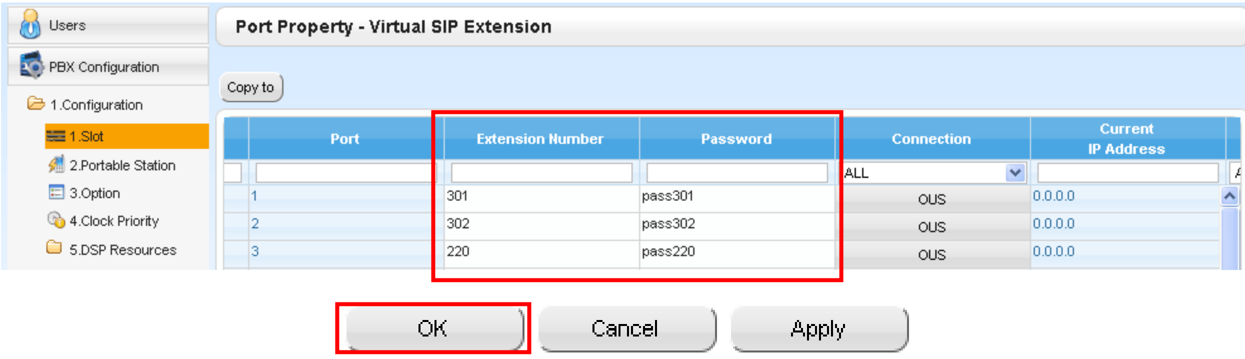


8.1-8 Select [**Port Property**]

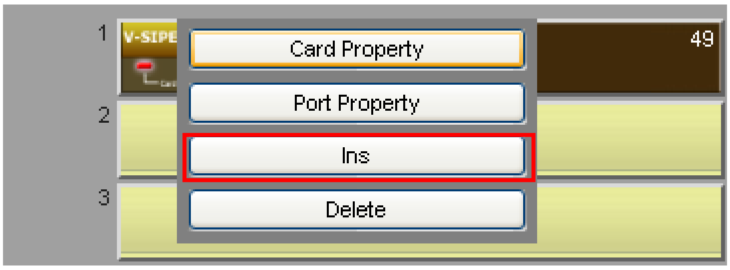


8.1-9 Edit the Extension Number and Password fields (click on them to enter data).

(Example, Extension Number: **301**, Password: **pass301**) --> Click on [**OK**]



8.1-10 Select [**INS**]

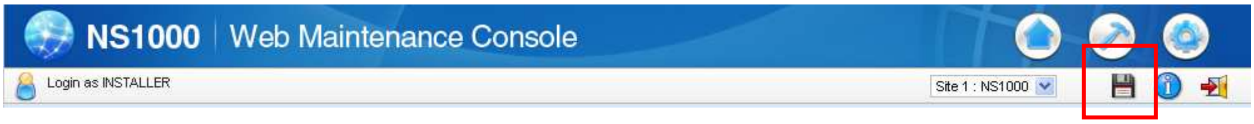


Contents of PBX main SIP Extension settings

Item	Configuration example	Description
SIP Extension port	5060 (Default)	Does not change it Need a System reset if setting change
SIP extension Number Password	301 pass301	Example
SIP extension Number Password	302 pass302	Example (If required).

8.1-11 Save the System Data

Click on [Save System Data icon]



[Note]

If you networking settings to change you need "system reset" that click on [System Reset]

During system rebooting, the PBX cannot use.

The PBX preparation completed.

9. Configure the Remote Office SIP Extension Settings if required only (Example). (Here is described as sample for Panasonic KX-UT SIP Phone(KX-UT123)).

We have to configure the SIP terminal via web in case using registration of V-SIPEXT (SIP-SLT)

Connect the SIP-terminal to the LAN. The following explanation assumes the LAN supports DHCP.

(e.g. DHCP server has given the SIP terminal the address 192.168.10.1).

9.1 Login and confirmation of info.

9.1-1 On the telephone, press [**Setting** or **Setup**] --> Select the [**Network Settings**] --> Press [**Enter**] --> Select the [**Embedded web**] --> Press [**ENTER**] --> Select [**ON**] --> Press [**ENTER**] --> [**Back**] --> [**Bac**k]. Or press [**Setting** or **Setup**] [#],[**5**],[**3**],[**4**] Select [**On**] Press [**Enter**]

9.1-2 Confirmation of current IP Address.

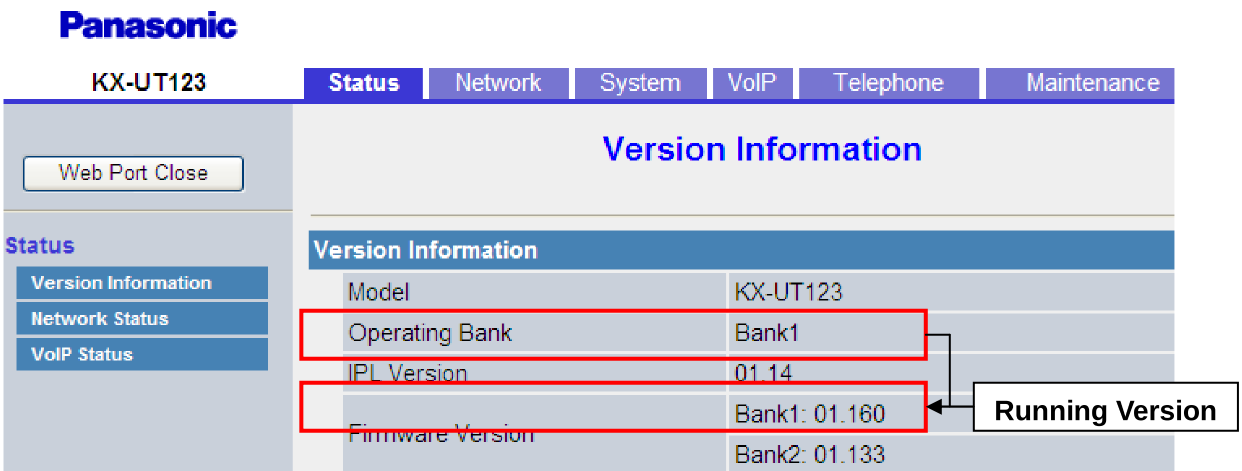
On the telephone, press [**Setting** or **Setup**] --> Select the [**Information Display**] --> Press [**ENTER**] --> Select the [**IP Address**] confirmation IP Address **192.168.10.1** (Example)

9.1-3 Access the SIP Terminal's web page (using previously read IP address).

e.g. http://192.168.10.1/ User Name: **admin** / Password: **adminpass** --> Click on [**OK**]

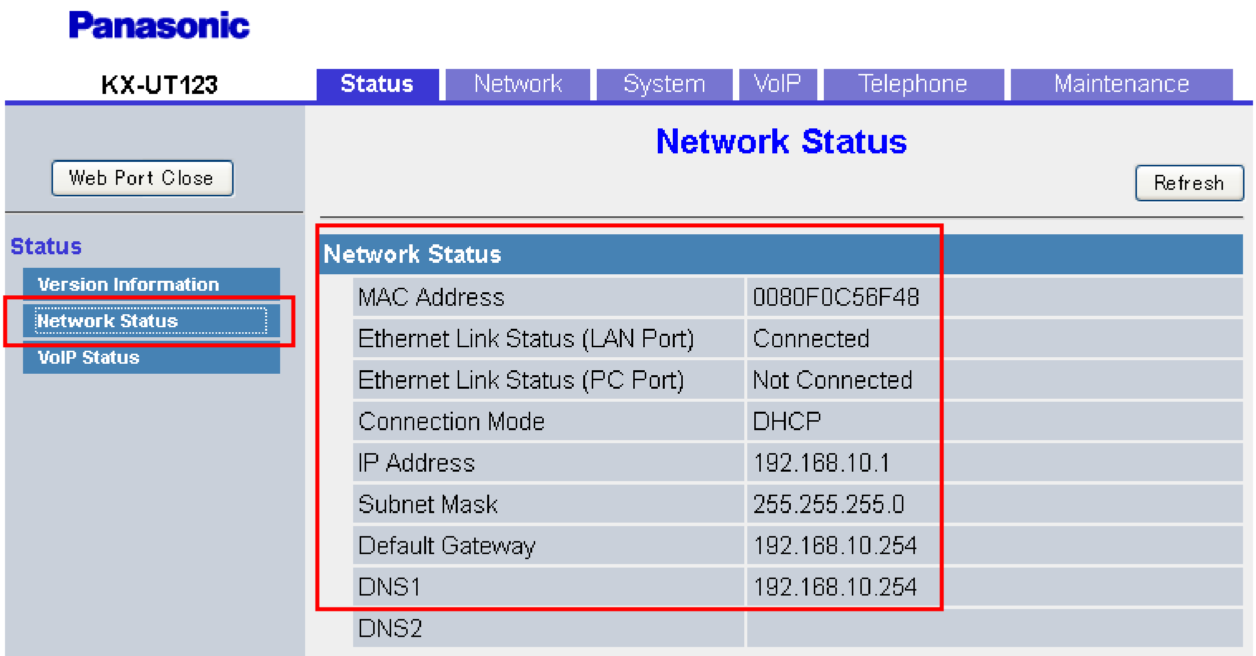
9.1-4 Confirmation of **Version Information**: In this case, 01.160 (Operating Bank: Bank1)

(Software version must be at or later than the version shown)



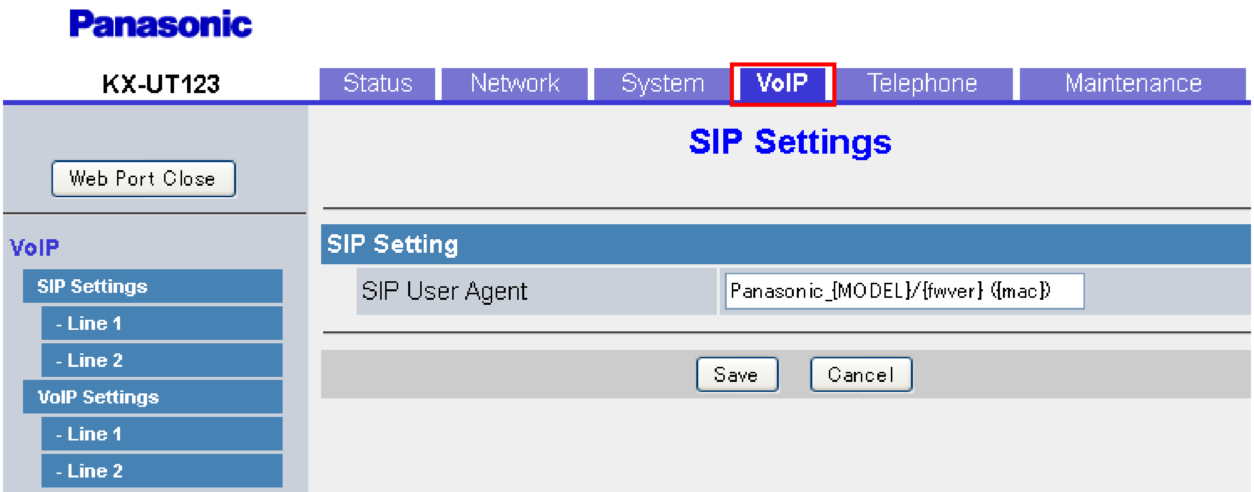
9.1-5 Confirm the Status of the Network: (DHCP has setup detail OK)

Click on [Network Status]

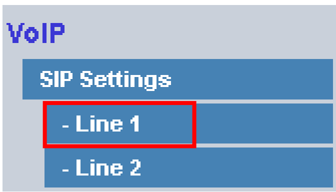


9.2 In VoIP Setting

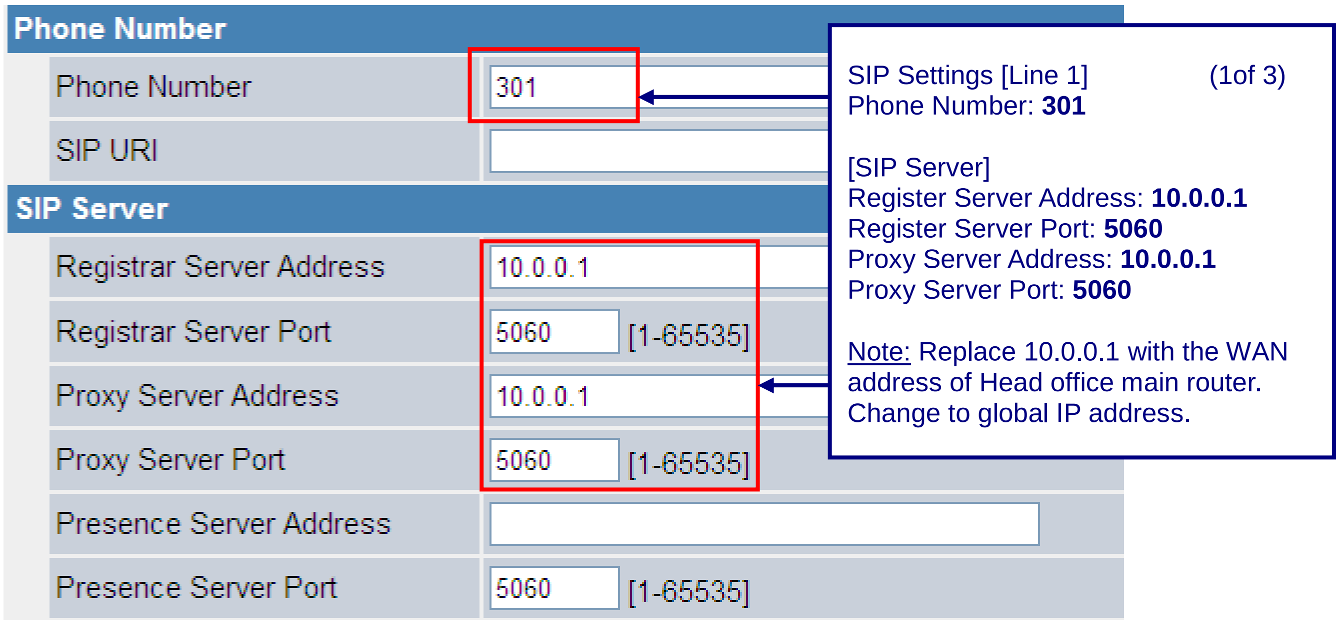
9.2-1 Click on [**VoIP**].



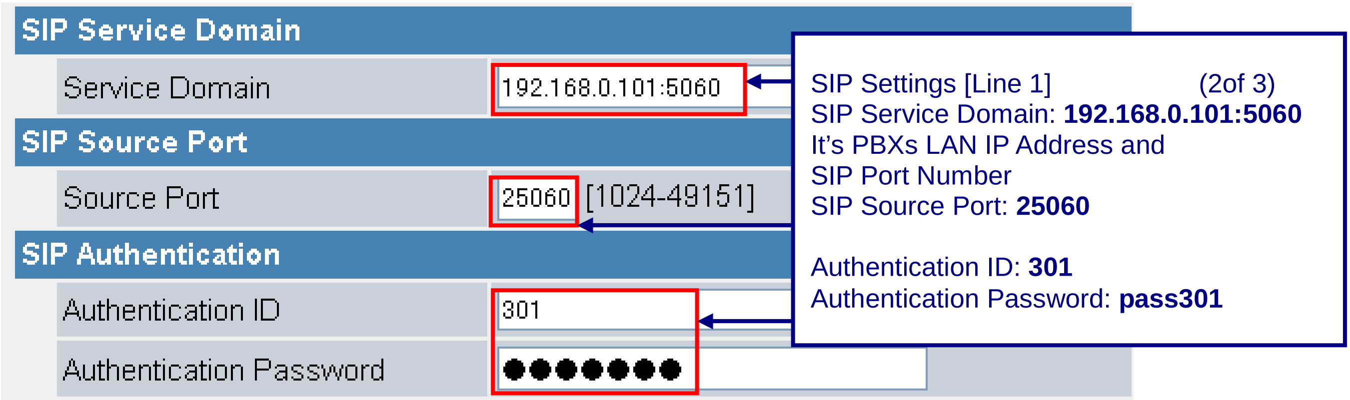
9.2-2 Click on [**Line 1**]



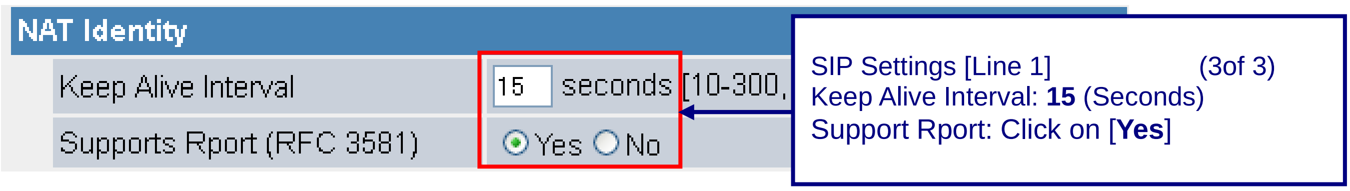
9.2-3 Configure the SIP Settings (1 of 3)



9.2-4 Configure the SIP Settings (2 of 3)



9.2-5 Configure the SIP Settings (3 of 3)



9.2-6 Click on [**Save**]

9.2-7 Check the [**Complete**] Message.

The configuration is completed!

10. Further SBC Information and Configuration

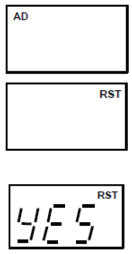
All documents are available online on the Mediatrix Download Portal at https://support.mediatrix.com/DownloadPlus/Download.asp.
Or on the web site at the following link
http://www.mediatrix.com/en/sessionbordercontroller Under the documentation tab.

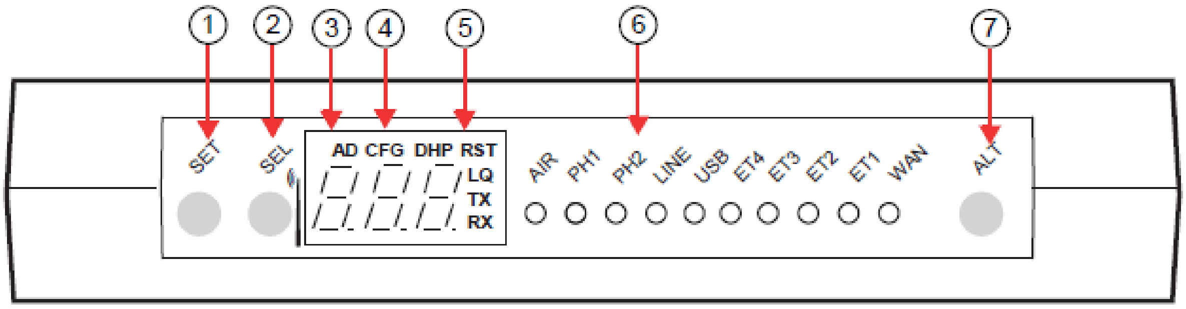
11. Management

11.1 Reset SBC to Factory Defaults

If you wish to you can reset all settings to their original values, so your Mediatrix 500 Series unit is setup the same way as when delivered from the factory.

1. Press and hold [SET] (1) pressed for 3 seconds, to enter setup mode.
2. Press [SEL] (2) repeatedly until ”RST” appears in the display.
3. Press [SET](1).
4. The question ”Clear all?” appears, and then ”no”.
5. Press [SEL](2) to choose ”YES”.
6. Press [SET](1).

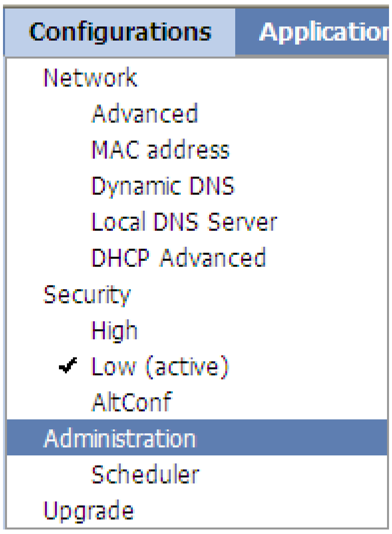




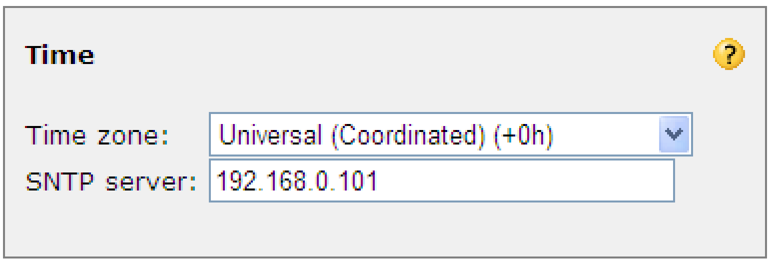
11.2 Time Setting

Time setting will be useful for analyzing some kind of problems.

11.2-1 Move the mouse [Configurations] --> [Administration] --> in the Time section.



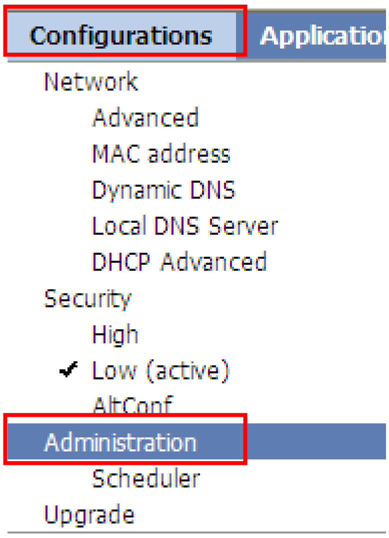
11.2-2 Configure the Time Server IP address, this IP is PBX IP address.(Example)



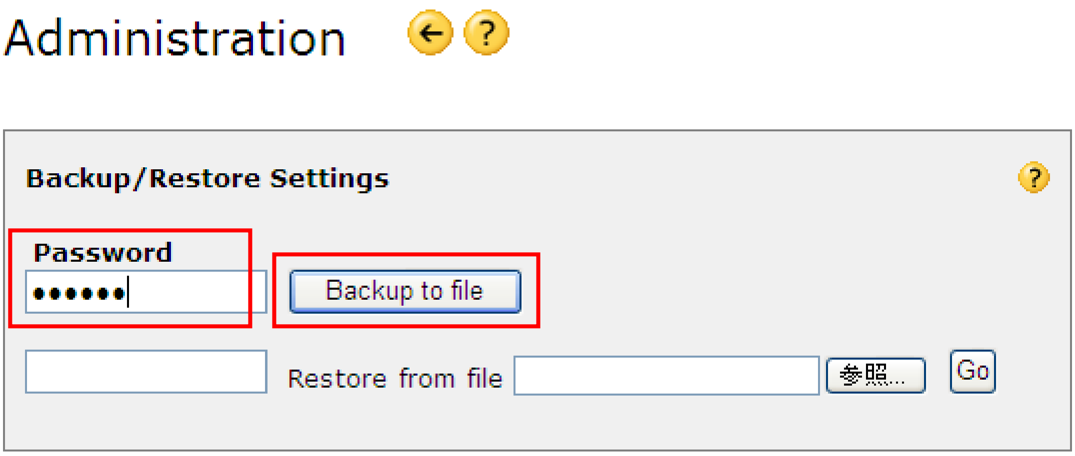
11.2-3 Click on [Apply]

11.3 SBC Configuration Backup

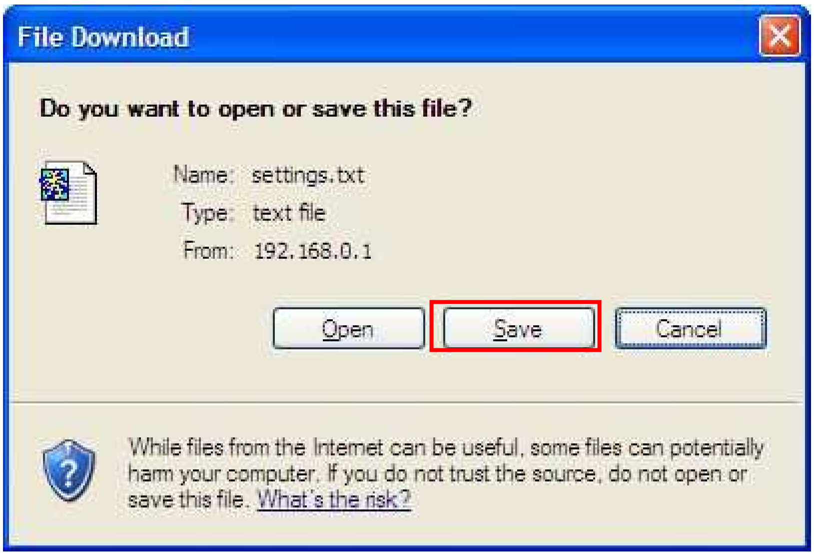
11.3-1 Move Mouse over [Configurations] and Select [Administration].



11.3-2 Enter the Password: 123456 and then Click on [Backup to file]. Example

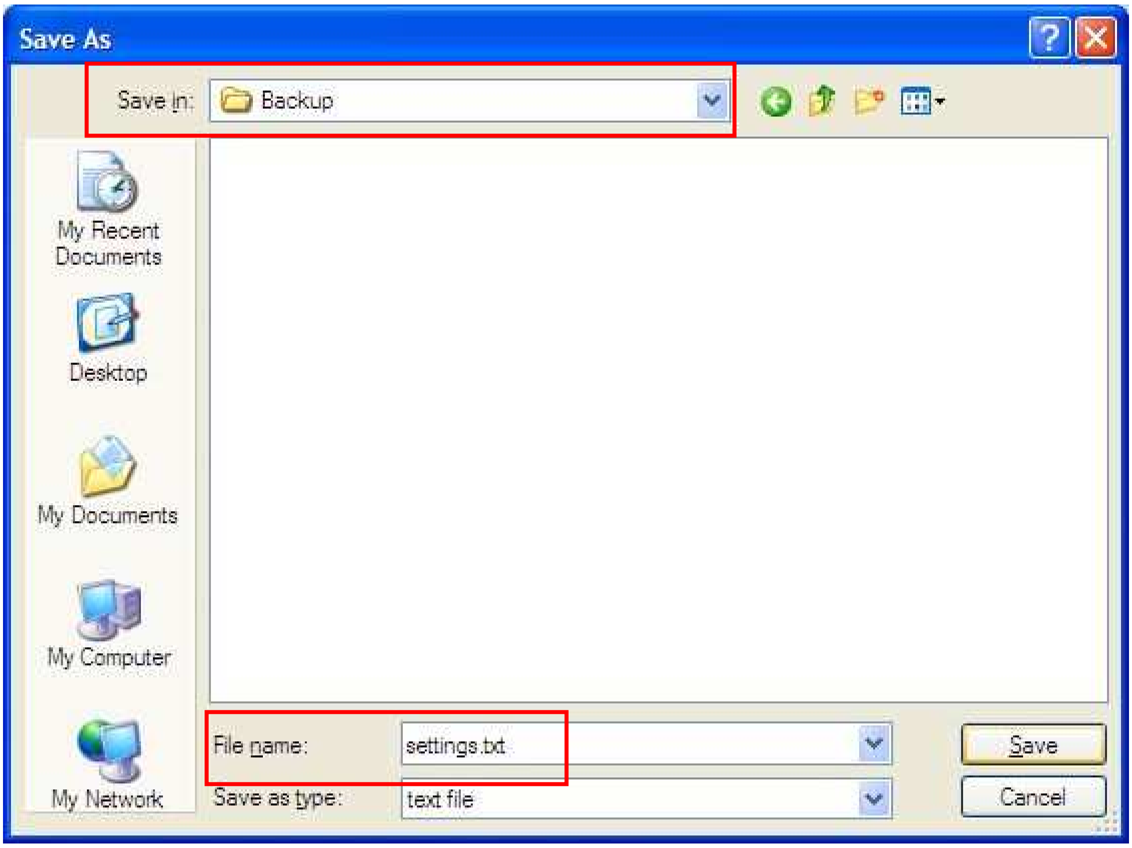


11.3-3 Click on [Save]



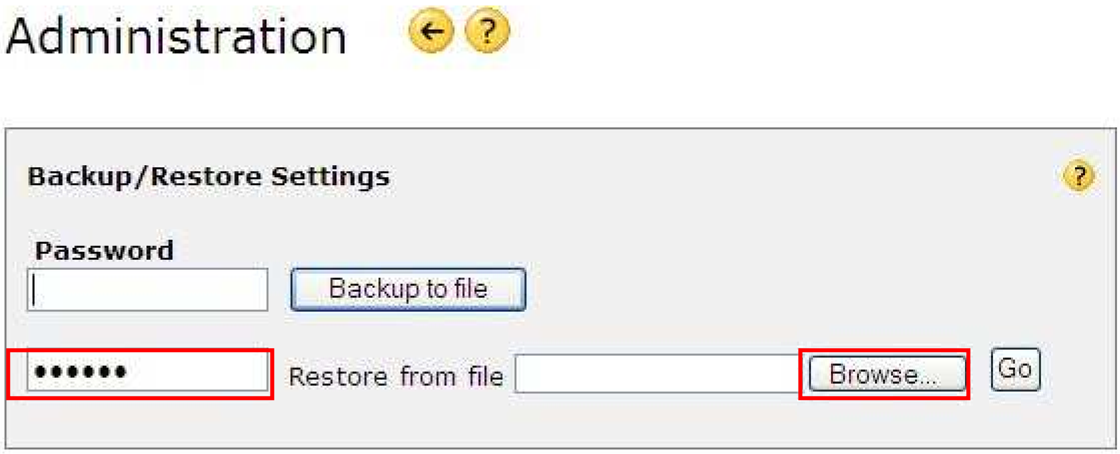
11.3-4 Save As

Select the Save Folder and Enter the File name [settings.txt] Example(Default).

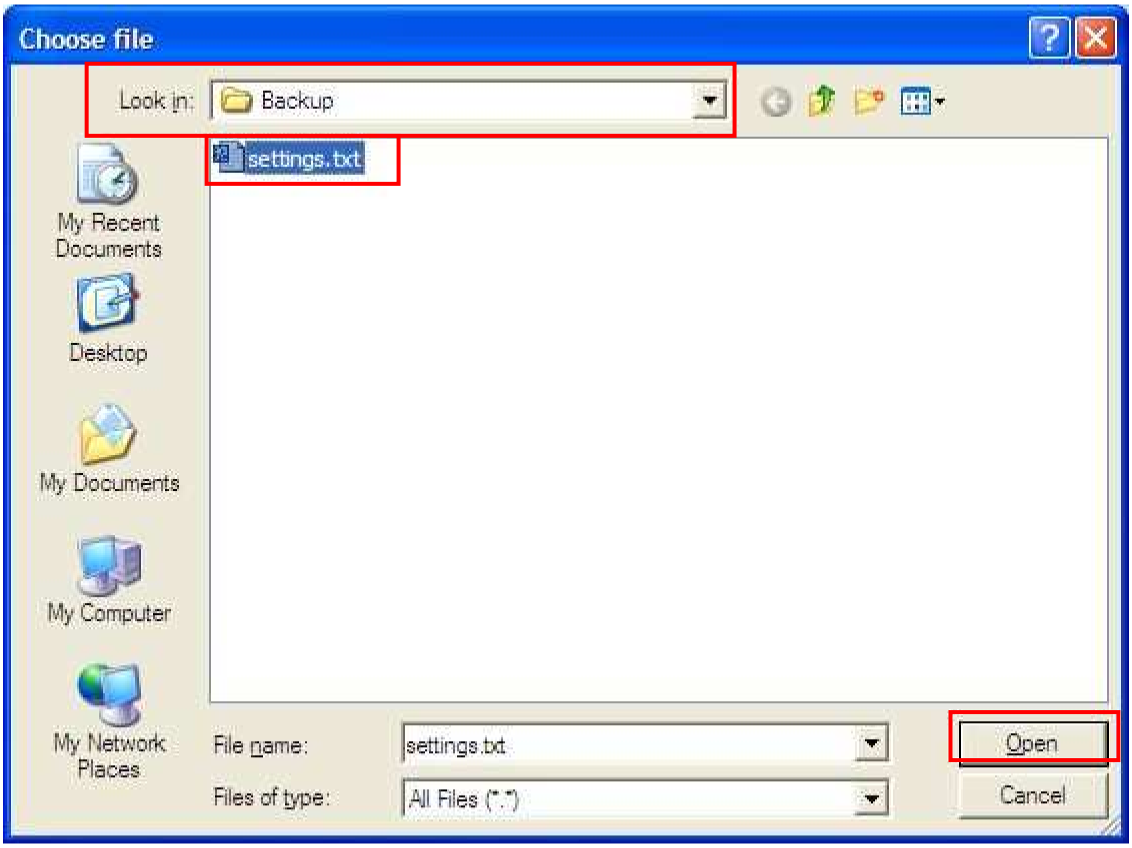


11.4 Restore Settings

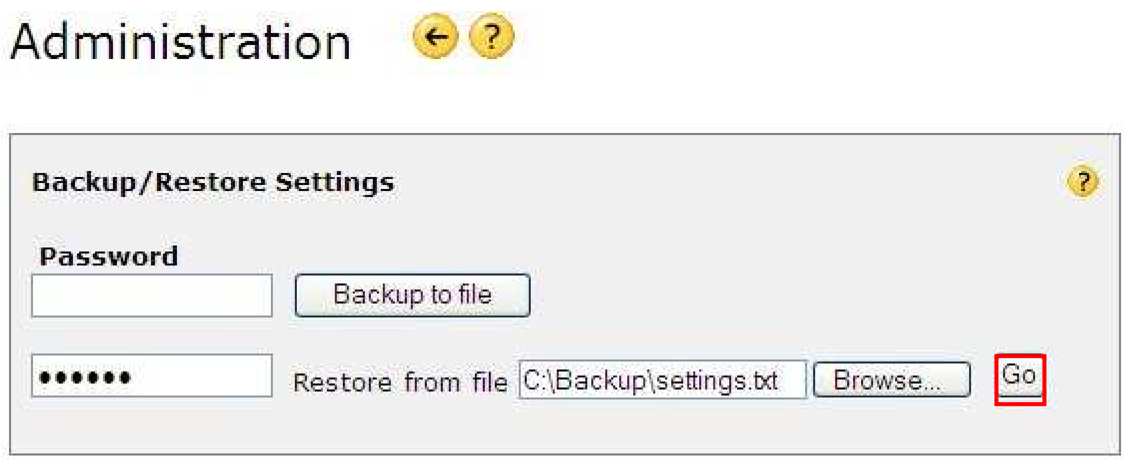
11.4-1 Enter the Password: **123456** (When at saving) and then Click on [**Browse…**].



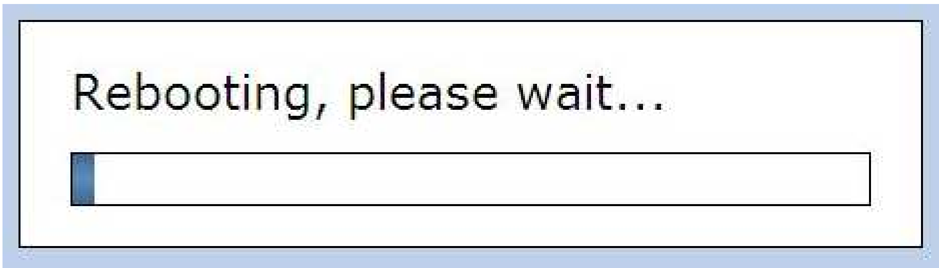
11.4-2 Choose file: **settings.txt** (Example) and then Click on [**Open**].



11.4-3 Click on [**Go**]



11.4-4 Rebooting, please wait… after the restore was successful.



11.5 Reset the UT-SIP Phone to Factory default.

Press [**Settings**] [**#**],[**1**],[**3**],[**6**] [**Enter**] --> Select [**Yes**] press [**Enter**]

11.6 Allow the access to web page on UT-SIP Phone.

Press [**Settings**] [**#**],[**5**],[**3**],[**4**] [**Enter**]

12 Troubleshooting

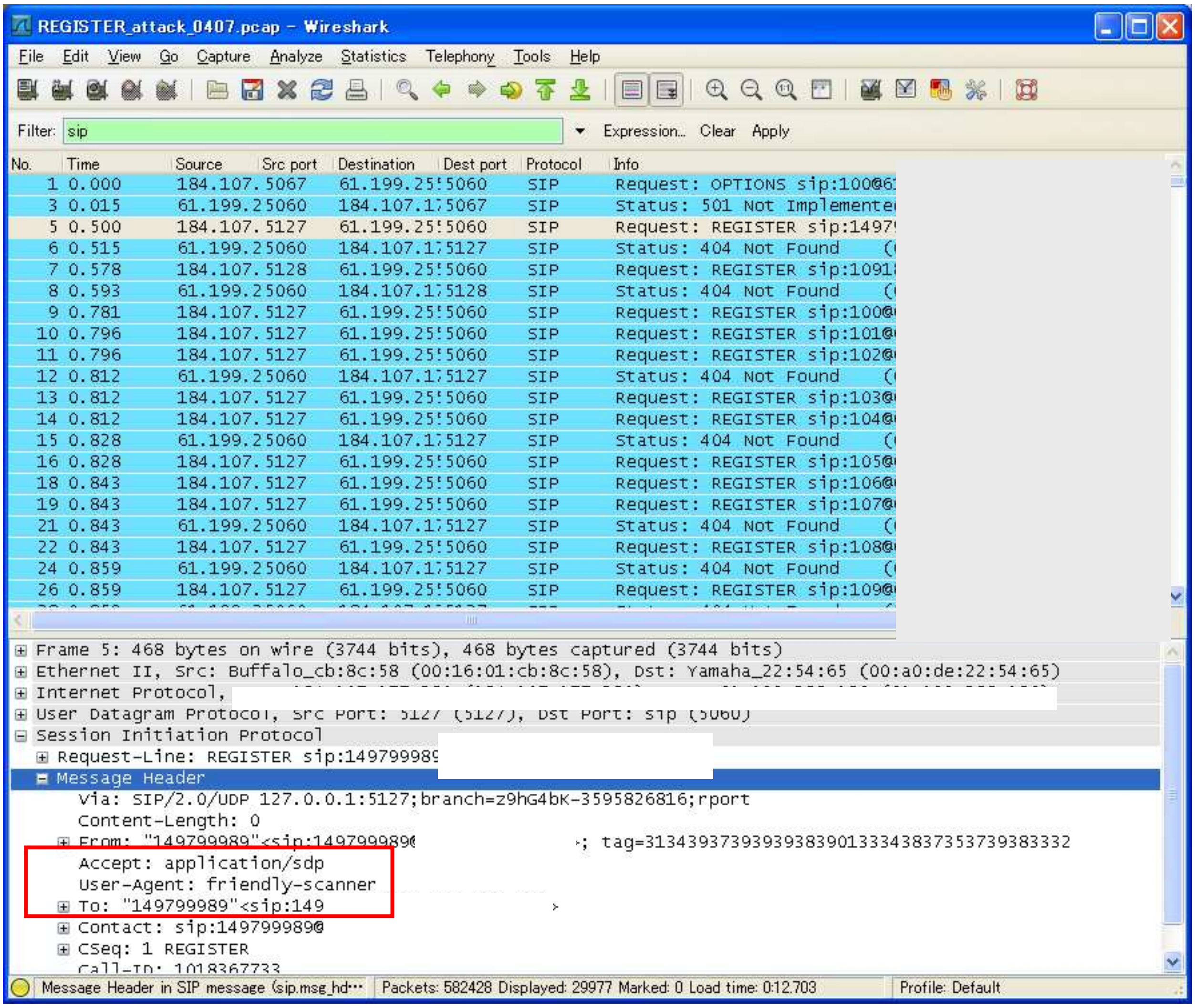
12.1 REGISTER Flood Attack

The Figure below shows a REGISTER Flood attack example.

The attack begins with OPTIONS message. Then, the attacker sends a great many REGISTER messages. The source address changes irregularly.

The symptom of this type of attack is the PBX temporarily becomes un-responsive,

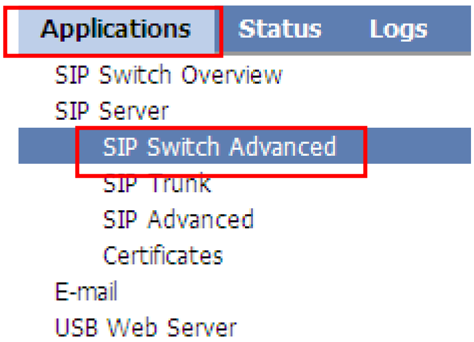
(It is very busy sending “404 Not Found” messages until the attacks over).



Countermeasure:

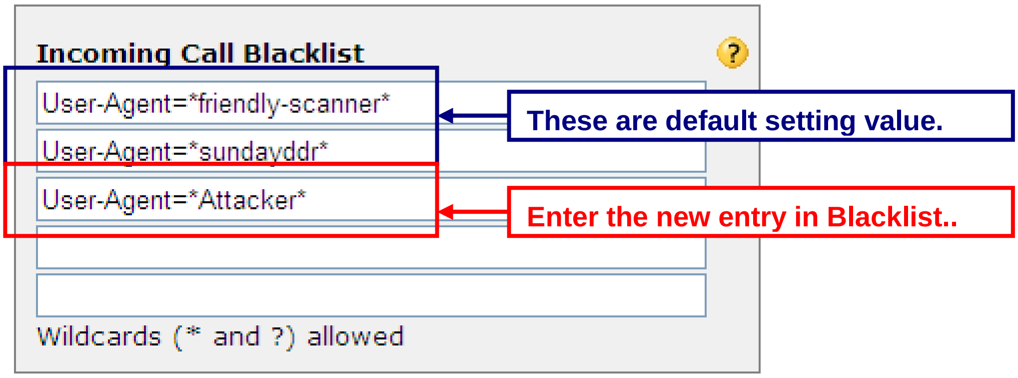
In the Switch Advanced, Configure a new entry in the Incoming Call Blacklist from captured packets.

12.1-1 Move mouse over Applications in SIP Advanced.



12.1-2 Configure a new entry in the Incoming Call Blacklist from captured packets.

User-Agent=*Attacker* (Example)



12.1-3 Click on [Click here to save permanently]

12.2 When UT-SIP Phone is repeated a reboot at remote site.

See section 5

1. Check the remote connection protocol whether it match or not.
2. Check the selected Phone location whether it match or not.

13. Appendix

13.1 SBC Configuration Check Sheet (PPPoE and Global IP address directly) (1/2)

Section	Part	Item	Setting value	Description
Home	Active Profile	Security	Low	Select
Configuration	Network Config	Operational mode	Router	Default
		Access type	**PPPoE**	Select
		Username		Offered by provider
		Password		Offered by provider
	DNS Server	IP Address		DNS or Main Router IP
		2nd (DNS server IP Address)		If required
	Default Gateway	IP Address		Main Router LAN IP
	SIP Routing Through Extern Firewall	Media Ports (Default 35000-35999)		Must much RTP Port forward setting of main router
		Outside IP		Existing main router Mapped SBCs IP
SIP Server	Allow to Register	Inside users	All	Select: All
		Outside users	All	Select: All
		Allow outgoing calls from	All	Select : All
Advanced	Advanced SIP set	Far End Nat Traversal (FENT)	Select the check	
		Detect endpoints behind same NAT	Clear the check	
	Authorized Users	Method	REGISTER	
		URI	*	
		Direction	Inbound	Select: Inbound
		Allow	Clear the check	
		Authentication	Clear the check	
		Authentication User IDs	*	
	Authorized Users	Method	INVITE	
		URI	*@PBX IP	
		Direction	Inbound	Select: Inbound
		Allow	Select the check	
		Authentication	Select the check	
		Authentication User IDs	*	
Advanced		Reuse received nonces	Clear the check	
		Allow RTP in reverse direction	Select the check	
		Reuse port number with same session	Select the check	
		Force Real Username on registration	Select the check	
	Trusted Networks	Check box	Clear the check	

13.1 SBC Configuration Check Sheet (PPPoE and Global IP address directly) (2/2)

Section	Part	Item	Setting value	Description
Security	Port redirection	Outside port(s)	Inside host	
	TCP	7547		PBX IP address
	TCP	37547		PBX IP address
	TCP	7580		PBX IP address
	TCP	37580		PBX IP address
	UDP	123		PBX IP address