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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 (Mediatrrix501 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 Mediatrrix501 (Firmware 5.35-M4)

Panasonic SIP Phone KX-UT series SIP telephones (Version 01.221)

Panasonic System Networks Co., Ltd.

Change history

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[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 (Mediatix501 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).

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

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.

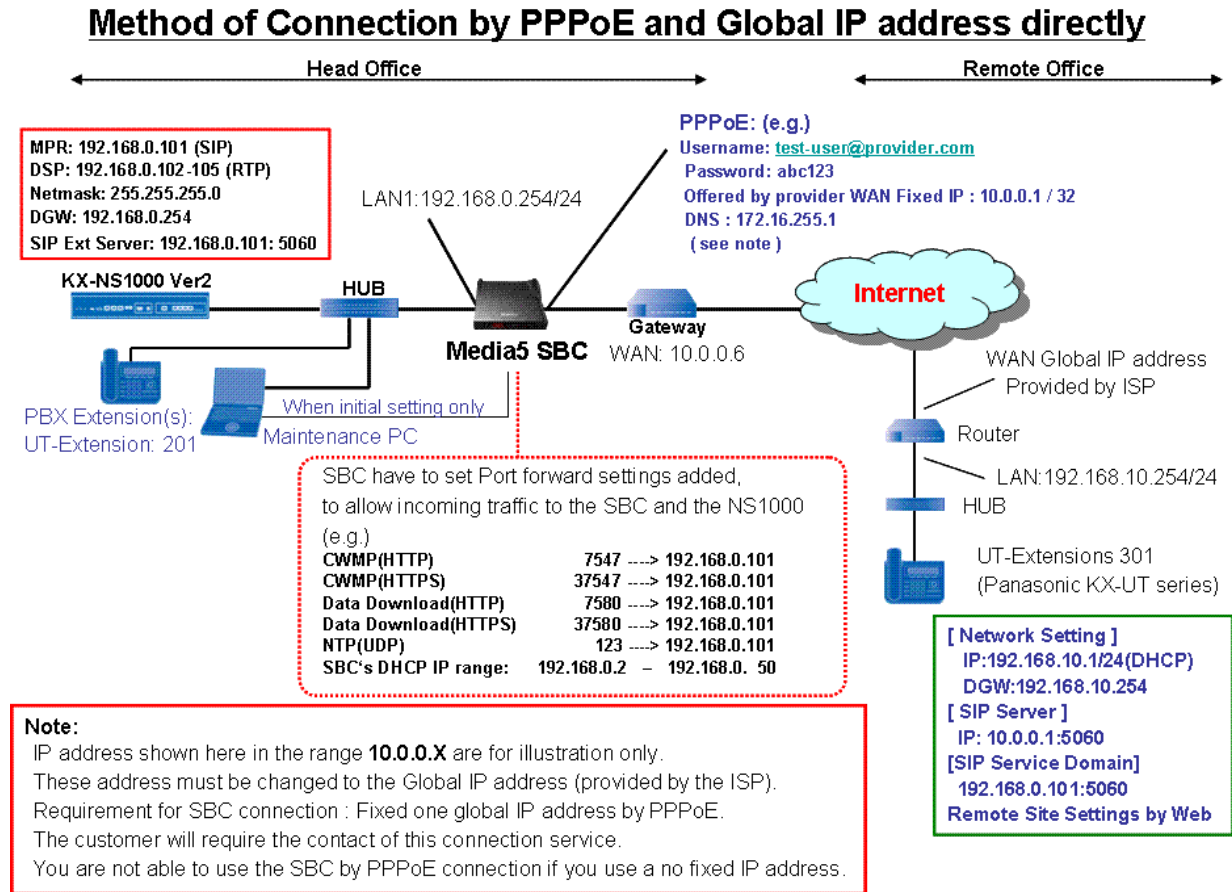
- (4) We recommended that you will be use the Port number 5060 of receiving of SIP in SBC.
The SBC is likely to have some interoperability issues when using different SIP port of this. The SBC will check all SIP messages and modify them even if as use the SIP Trunk in the PBX. It means the SBC receiving 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



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	10.0.0.1	Mandatory (Information offered by provider)
Fixed WAN IP address	(change to global address)	*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		If required
Starting IP address	192.168.0.51	Note) Set the different IP range from SBC's DHCP IP range.
Ending IP address	192.168.0.100	

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	1	Default
P2P Group Name	Empty	Default: Empty
Bandwidth Control	Disable	Default: Disable
P2P Group	2	Default
P2P Group Name	Remote Office	Example
Bandwidth Control	Enable	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	301	Example
Password	1234	Default
P2P Group	2	Default:1
[Option tab]		
Codec Priority	*1st: G729A / 2nd:G711A/ 3rd:G722Mu / *4th: G722	*1st: G722 / 2nd:G711A/ 3rd:G722Mu / *4th: G729A
[Remote Place tab]		
Phone Location	Remote	Default: Local
Protocol for Remote SIP-MLT	HTTPs	Default: HTTP
[Port Property Main]		
SIP Extension Number	302	Example(If required)
Password	1234	Default
P2P Group	2	Default:1
[Option tab]		
Codec Priority	*1st: G729A / 2nd:G711A/ 3rd:G722Mu / *4th: G722	*1st: G722 / 2nd:G711A/ 3rd:G722Mu / *4th: G729A
[Remote Place tab]		
Phone Location	Remote	Default: Local
Protocol for Remote SIP-MLT	HTTPs	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	301	Example
Password	pass301	
SIP extension Number	302	Example (if required)
Password	pass302	

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

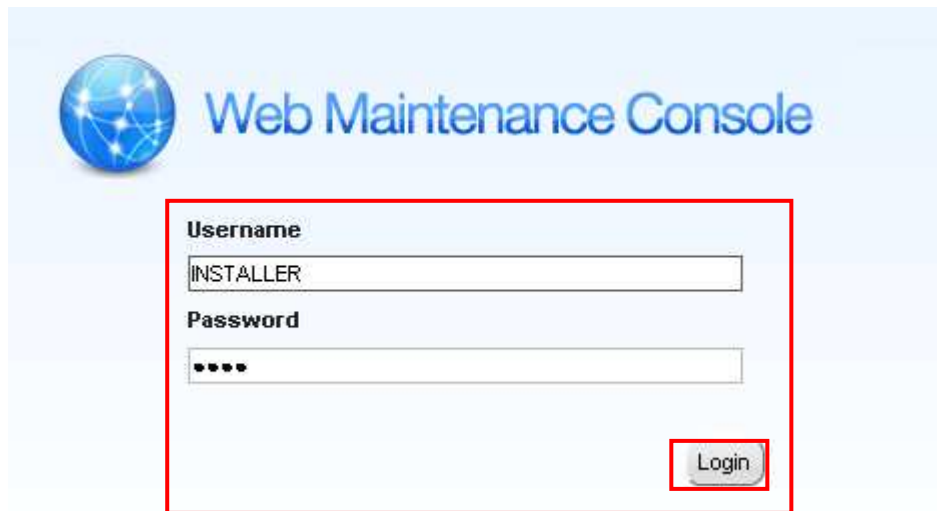
(Note) The SIP remote extension(s) are registered to the V-UTEXT32 of NS1000 (Ver.2)

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

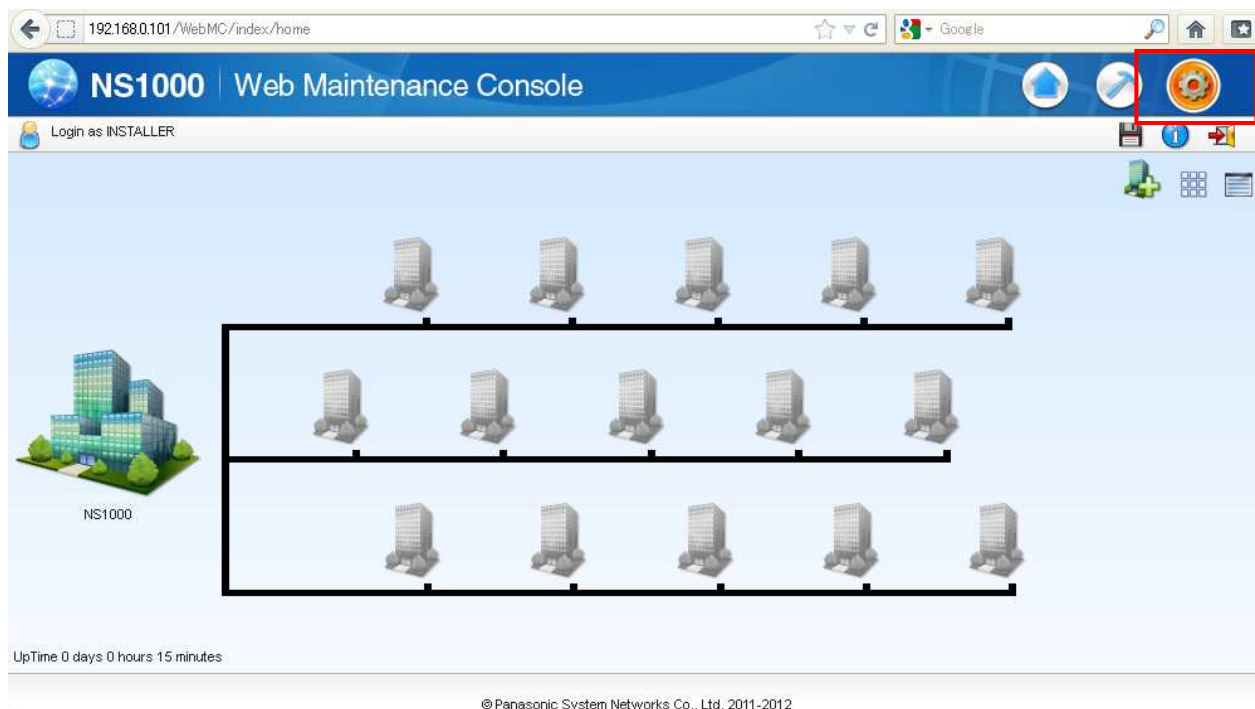
4.2 Access the KX-NS1000 Web Maintenance Console page (using previously read IP address).

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

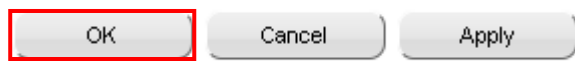
(In this case, IP Phone Capacity (ch):30 / IP Proprietary Telephone/IP:0 (Note *1).

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 Soft	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	-

Install the IP Phone Capacity and IP Proprietary Telephone Activation key if required.

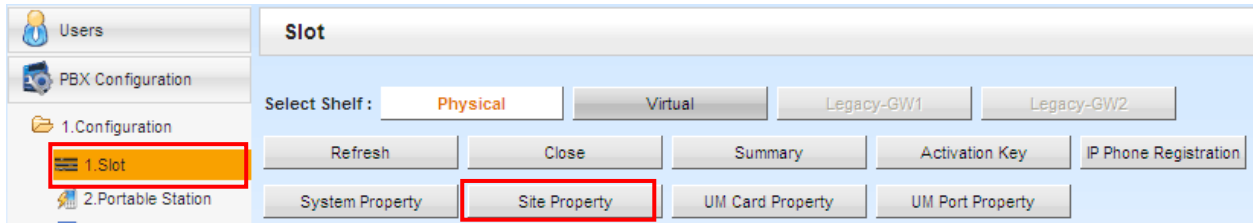
***Note *1): 30 IP-Extension can be installed to NS-1000 without extra Activation Keys, but for connecting IP Extension itself. User must purchase IP-Telephones Activation Keys when expand over 8 Telephones.**

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

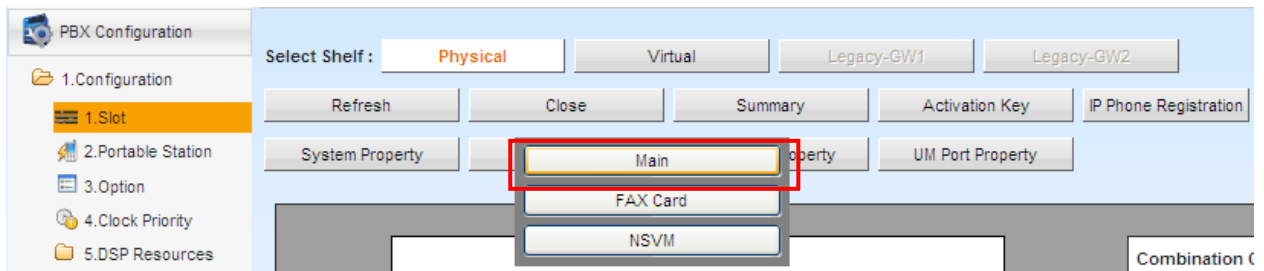


4.8 Click on [PBX Configuration] --> [1.Configuration] --> [1.Slot]

--> Move mouse over [Site Property]

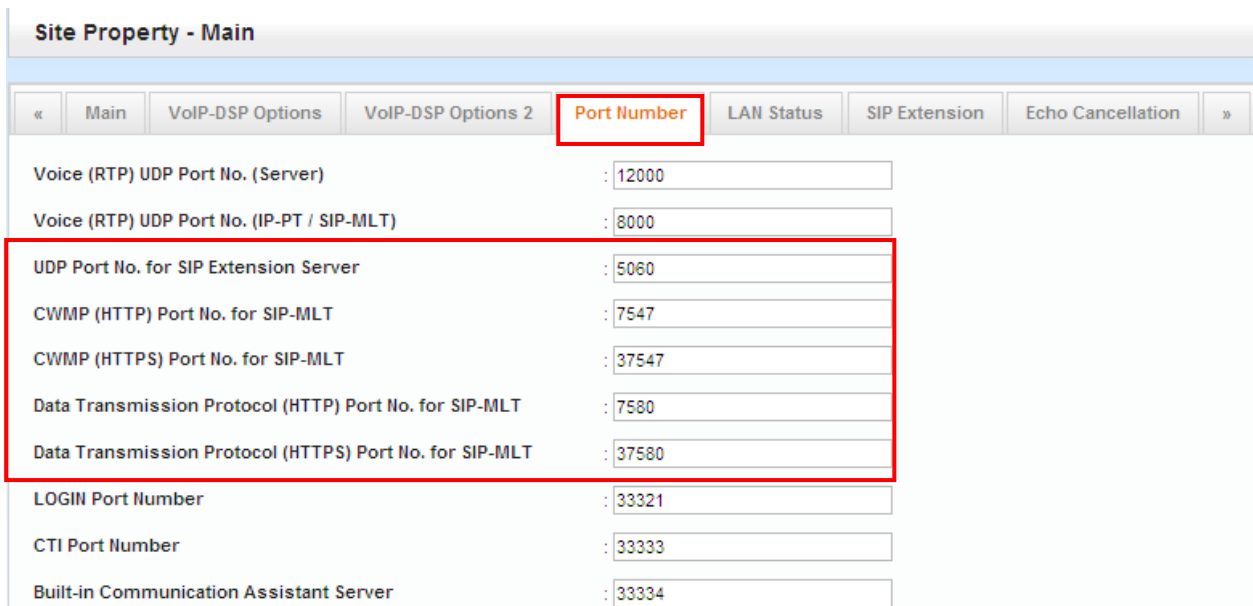


4.9 Select [Main] menu.



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

Site Property - Main

« Main VoIP-DSP Options VoIP-DSP Options 2 Port Number LAN Status **SIP Extension**

Setting parameters assigned to Remote SIP-MLT

NAT - CWMP Server IP Address	:	10.0.0.1
NAT - CWMP Server (HTTP) Port No.	:	7547
NAT - CWMP Server (HTTPS) Port No.	:	37547
NAT - SIP-MLT Data Download Server (HTTP) Port No.	:	7580
NAT - SIP-MLT Data Download Server (HTTPS) Port No.	:	37580
NAT - SIP Proxy Server IP Address	:	10.0.0.1
NAT - SIP Proxy Server Port No.	:	5060
NAT - NTP Server IP Address	:	10.0.0.1
NAT - NTP Server Port No.	:	123
NAT - Keep Alive Packet Type	:	Blank UDP
NAT - Keep Alive Packet Sending Interval Time (s)	:	20
NAT - SIP Register Expire Time (s)	:	20

*) Perform System Reset for changes to take effect

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

OK Cancel Apply

--> *) 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]**.

P2P Group Configuration (Recommended settings)

Priority Voice 1: Select [G729]

Priority Voice 2: Select [G711]

Priority Voice 1: Select [None]

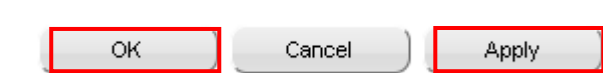
4.14 Click on **[3.Group]** --> **[10.P2P Group]** --> Enter Group Number "2" (Example)

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.

P2P Group	P2P Group Name	Bandwidth Control
		ALL
1		Disable
2	RemoteOffice	Enable
3		Enable
4		Disable
5		Disable
6		Disable
7		Disable
8		Disable
9		Disable
10		Disable
11		Disable
12		Disable
13		Disable
14		Disable

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)

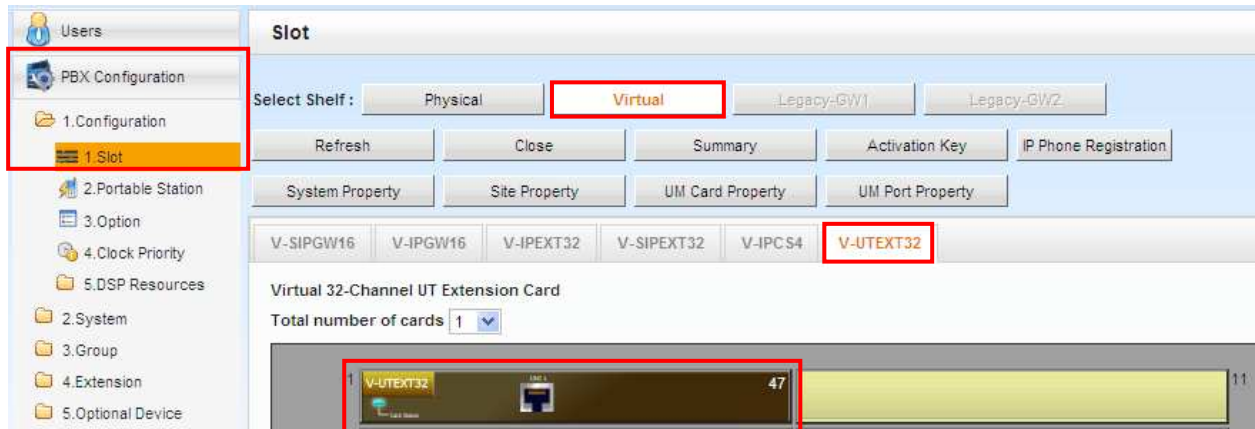
1. A KX-NS1000 can work with only one SBC. Also, multiple sites can share an SBC.
2. KX-UT series SIP Phones can communicate over a NAT (Network Address Translation)-enabled network only when communicating via an SBC from the KX-NS1000 to which the KX-UT series SIP Phones are registered.
3. When an SBC 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.

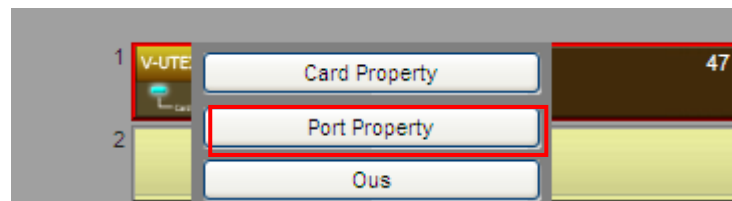
* Note) This procedure differs according to the IP Terminal Registration Mode already set to your KX-NS1000. For details about how to configure the V-UTEXT32 card in each mode, refer to "Installation Manual 5.9.1 Registering IP Telephones".

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

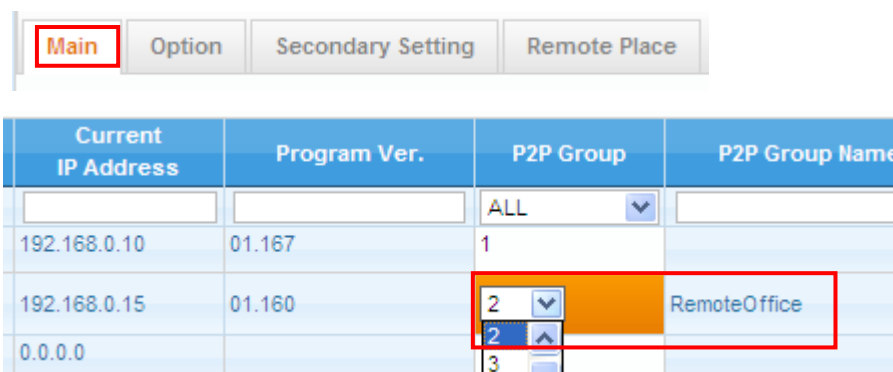


5.1-2 Move the mouse pointer over the V-UTEXT32 card (Virtual UT Extension Card).

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

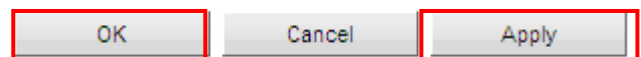
Select [Remote] in [Phone Location] column for SIP Phone that will be used at a remote site.

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]



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.

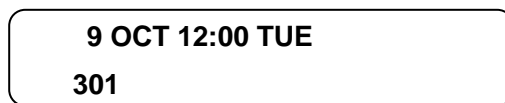
Note:

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).



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

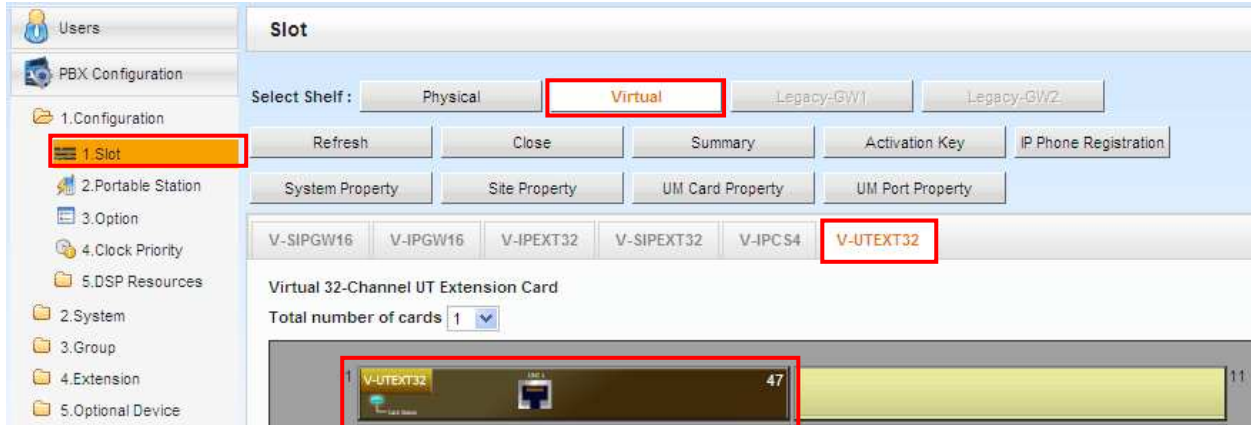
5.2 Procedure for Method 2 with KX-UT Series SIP Phones.

* Note)

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

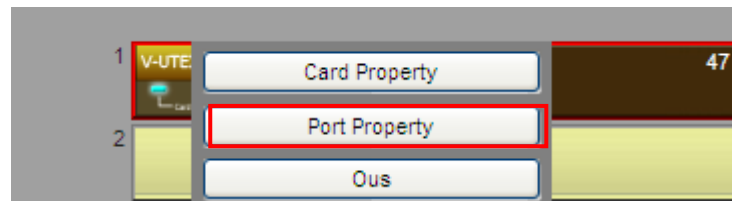
You do not have to connect the SIP Phone to the PBX when using this method.

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.

Main			
Current IP Address	Program Ver.	P2P Group	P2P Group Name
192.168.0.10	01.167	ALL	
0.0.0.0		2	RemoteOffice
0.0.0.0		2	

5.2-4 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												
Port	Extension Number	Extensic Name	Connectio	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	ALL	ALL	ALL	ALL	ALL	ALL	
1	201		INS	Enable	0	0	G.722	G.711A	G.711Mu	G.729A	20ms	
2	301		Fault	Enable	0	0	G.729A	G.711A	G.711Mu	G.722	20ms	

5.2-5 Configure the [Remote Place]

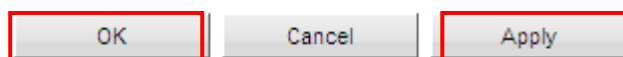
Click on [Remote Place] tab

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.

Main Option Secondary Setting Remote Place						
Extension Number	Extension Name	Connection	Phone Location	Web-MC Ability	Protocol for Remote SIP-MLT	
		ALL	ALL	ALL	ALL	
201		INS	Local	Enable	HTTP	
301		Fault	Remote	Enable	HTTPs	

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.

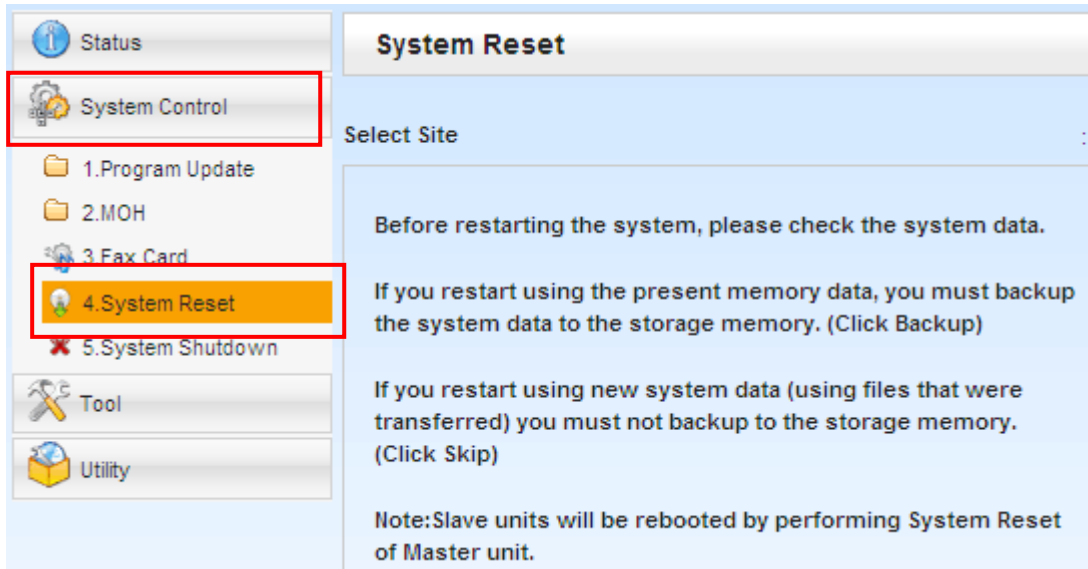
Therefore we have to reboot the NS-1000 only once, but we do not need this every time.

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

Click on [Maintenance] -->



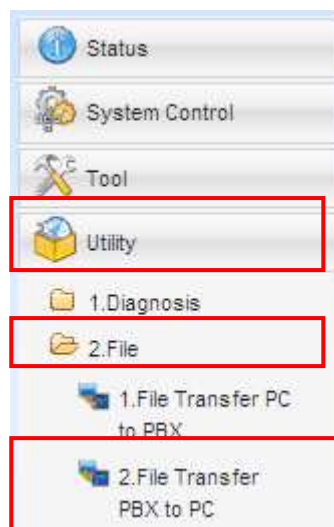
Click on [System Control] --> [4.System Rest] --> [Backup] (Just in case) --> [OK] --> [OK]



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
PFGGA	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...]**

The screenshot shows the Panasonic KX-UT113 web interface. At the top, there are navigation tabs: Status, Network, System, VoIP, Telephone, and Maintenance (highlighted with a red box). Below the tabs is a 'Web Port Close' button. On the left, a 'Maintenance' menu is visible with options: Import Configuration File, Export Configuration File, Export Web Settings, Firmware Maintenance, Local Firmware Update, and Provisioning Maintenance. The main content area is titled 'Import Configuration File' and contains the following form fields:

Configuration File Type:	<input checked="" type="radio"/> Standard <input type="radio"/> Product <input type="radio"/> Master
Encryption:	<input type="radio"/> Yes <input checked="" type="radio"/> No
File Name:	<input type="text"/> <input (highlighted="" a="" box)<="" red="" td="" type="button" value="Browse..." with=""/>

At the bottom of the form is an 'Import' button.

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

And Click on **[Import]**

This screenshot shows the 'Import Configuration File' page with the file name field filled with 'C:\temp\UT_ACS_HTTPS_01NS1000.cfg' (highlighted with a red box). The 'Import' button at the bottom is also highlighted with a red box.

Configuration File Type:	<input checked="" type="radio"/> Standard <input type="radio"/> Product <input type="radio"/> Master
Encryption:	<input type="radio"/> Yes <input checked="" type="radio"/> No
File Name:	C:\temp\UT_ACS_HTTPS_01NS1000.cfg <input <="" td="" type="button" value="Browse..."/>

5.2-22 Confirm **[Complete]**

This screenshot shows the 'Import Configuration File' page with a 'Complete' message in a red box at the top left. The form fields below are the same as in the previous screenshot.

Complete (highlighted with a red box)	
Configuration File Type:	<input checked="" type="radio"/> Standard <input type="radio"/> Product <input type="radio"/> Master
Encryption:	<input type="radio"/> Yes <input checked="" type="radio"/> No

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

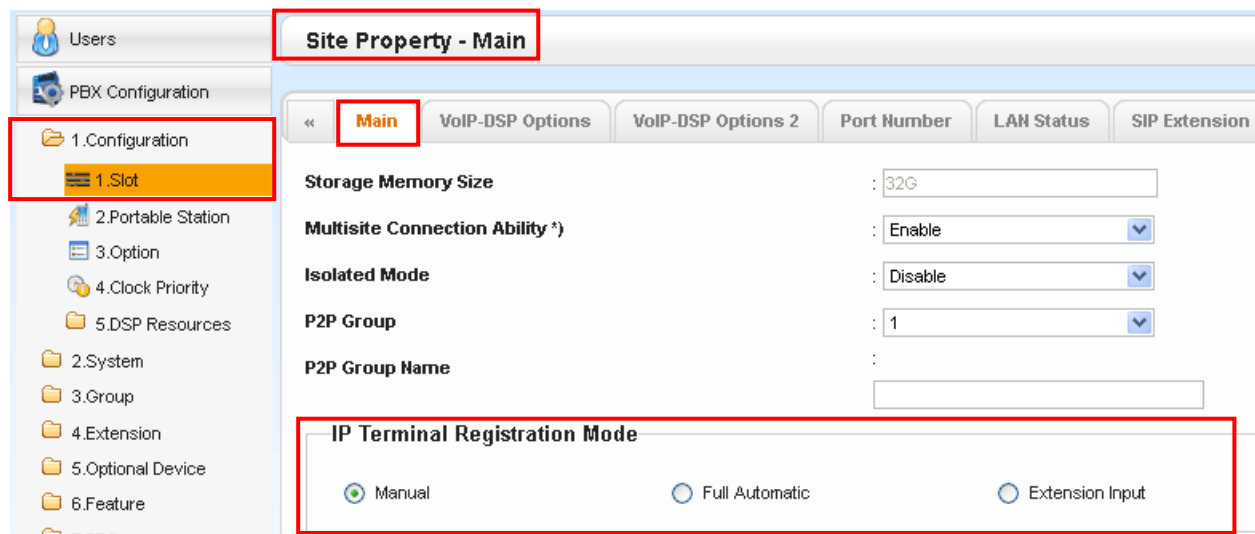
5.2-23 Register the UT-SIP Phone (V-UTEXT32) by NS1000 registration.

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]**

ID	Site	Shelf	Slot	Port	Extension Number	Extension Name	Telephone Type	Connection
1	1	Virtual	47	1	201		UT	INS
2	1	Virtual	47	2	301		UT	Fault

5.6-2 Select Extension Number for Registration (Example)

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

UT Extension Registration Wizard

Available Extension Number

Selected Extension Number for Registration

106 :
107 :
108 :
109 :
110 :
111 :
112 :
113 :
114 :
115 :
116 :
117 :
118 :
119 :
120 :
121 :
122 :
123 :
124 :
125 :
126 :
127 :
128 :
129 :
130 :

301 :

Next Cancel

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

UT Extension Registration Wizard

Registration Executing

1	301
---	-----

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

UT Extension Registration Wizard

Registration Completed

1	301	✓
---	-----	---

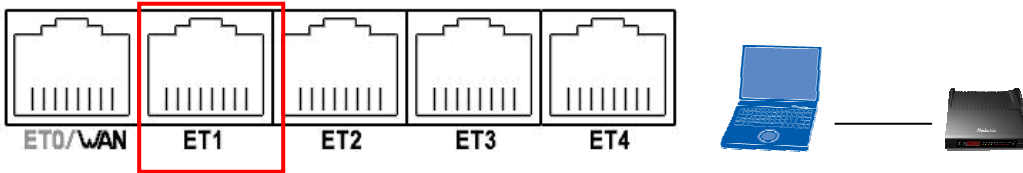
Close

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.

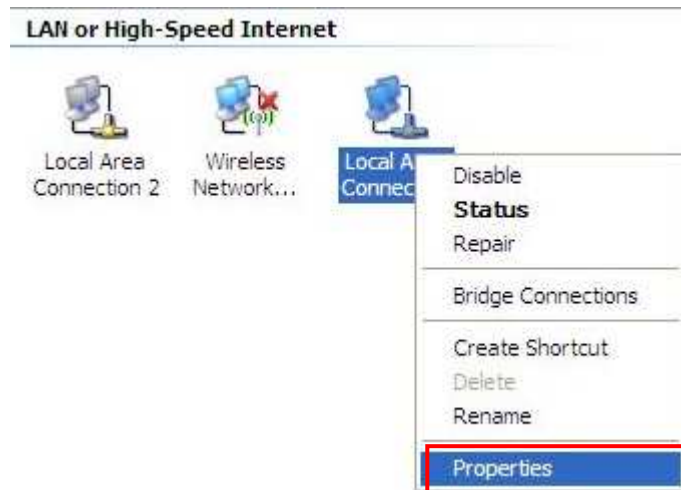
As a matter of course you can use static IP address.

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

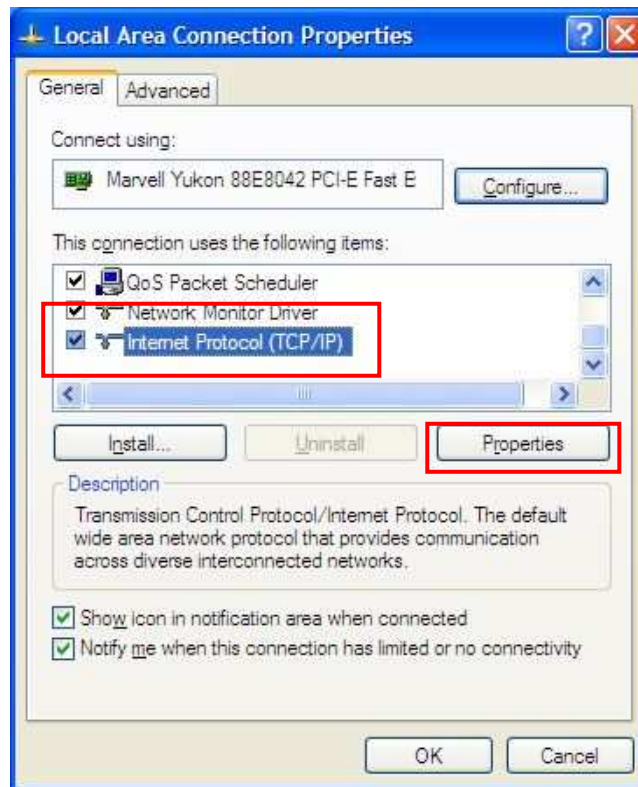
[View Network Connections] Select the LAN in use.



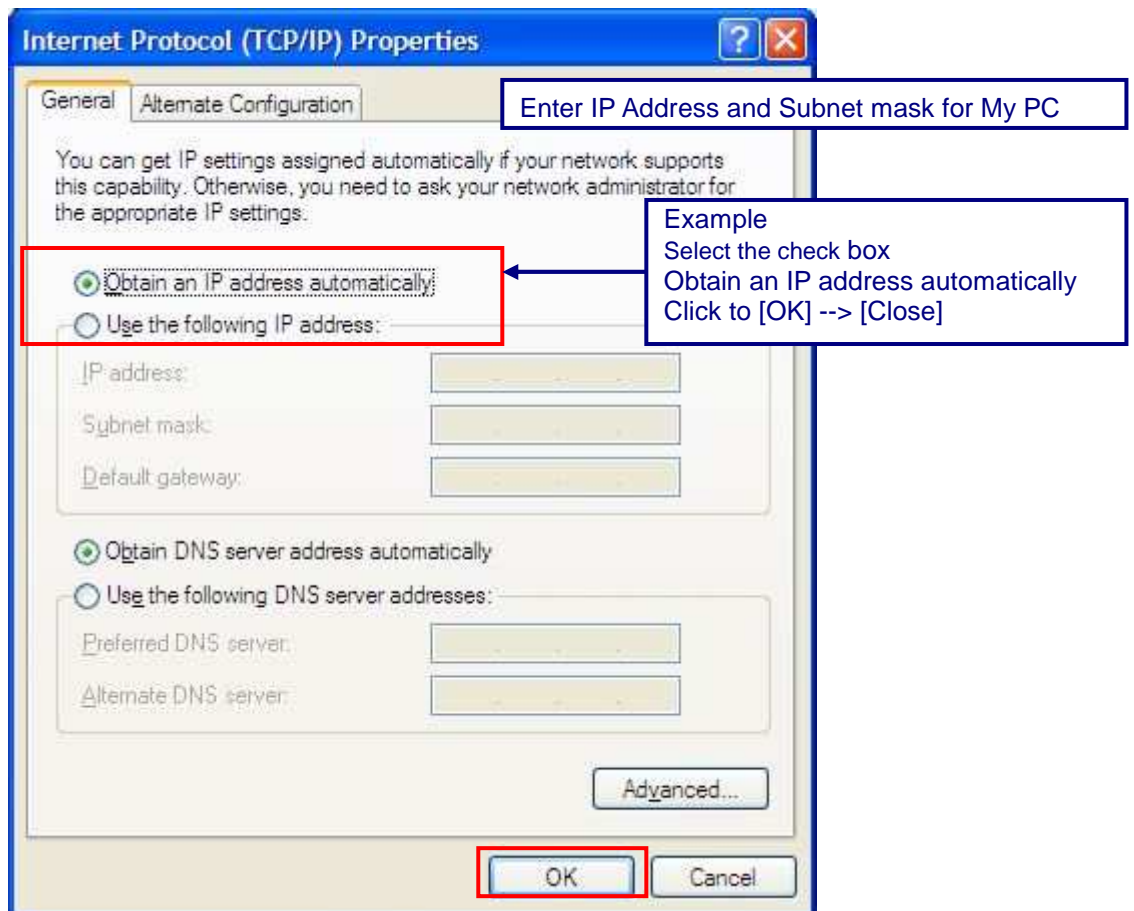
6.1-3 [Local Area Connection Properties] – Right click and Select the [Properties].



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-F0-98-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

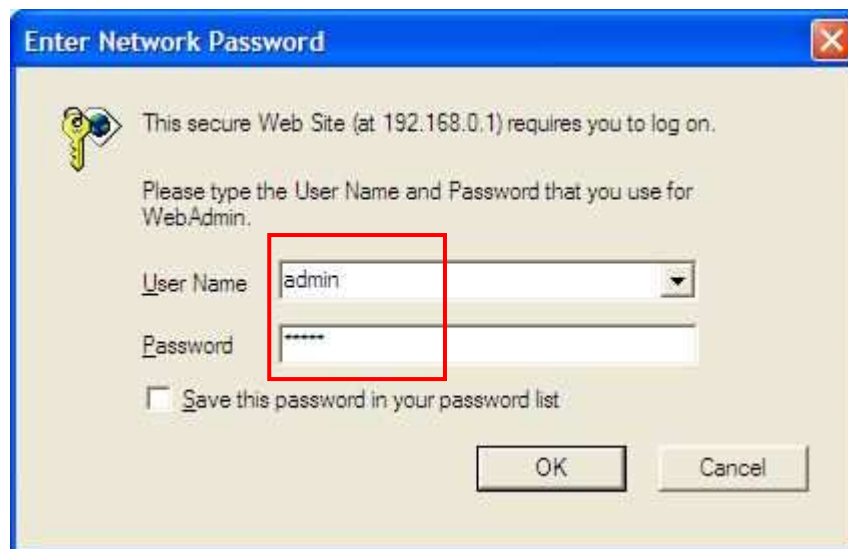
Ping statistics for 192.168.0.1:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
    Approximate round trip times in milli-seconds:
        Minimum = 0ms, Maximum = 0ms, Average = 0ms
```


6.2 In Network Configurations (1)

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



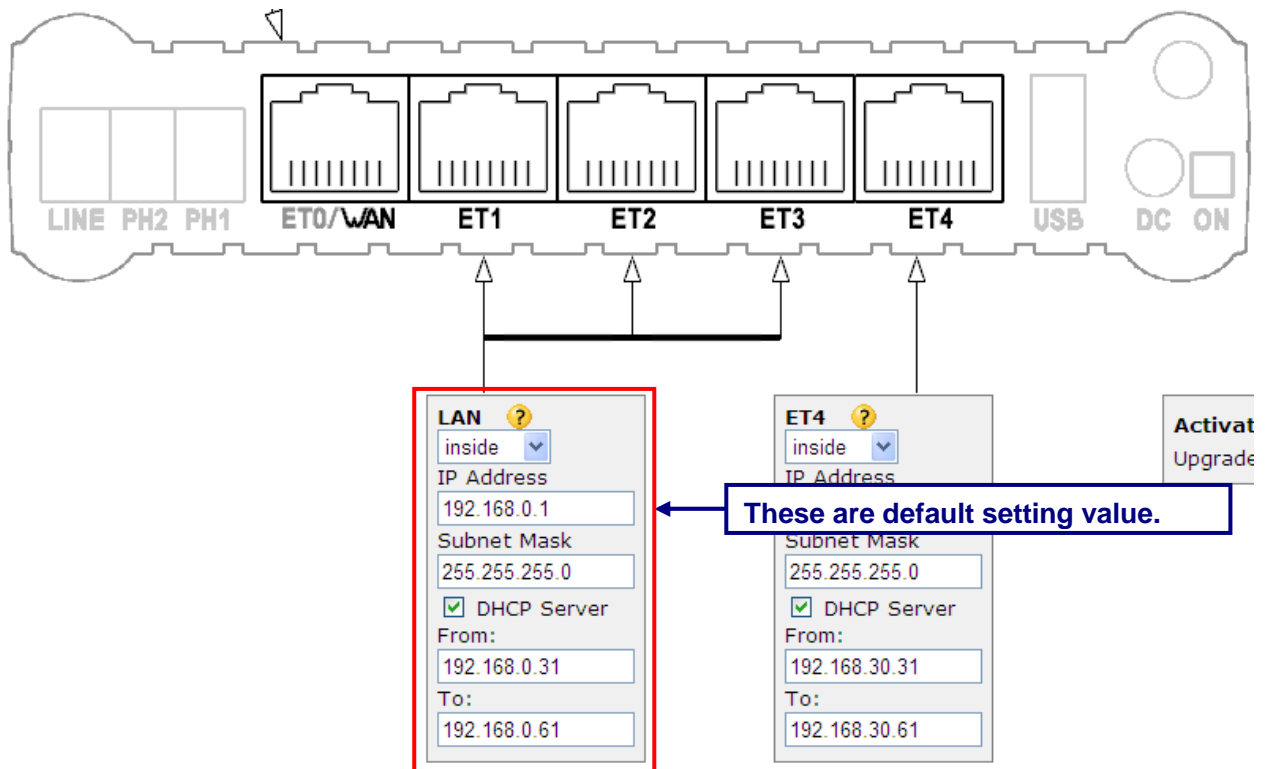
6.2-2 To Enter Network Password Username: **admin** / Password: **admin** (Default).



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

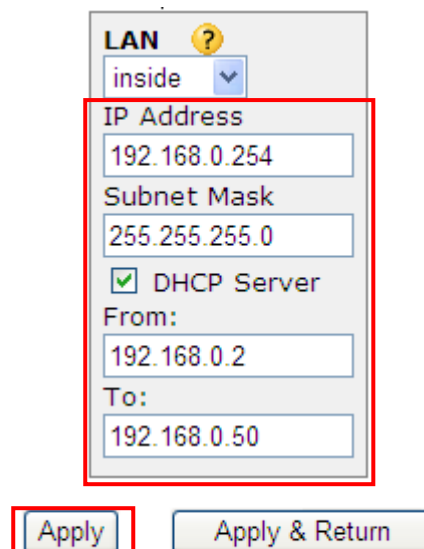


6.2-4 Confirmation of LAN settings.



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/>

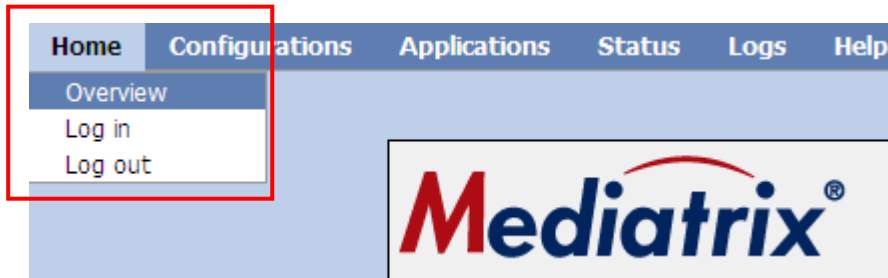
The screenshot shows the Mediatrix web interface. At the top, there is a navigation menu with links for Home, Configurations, Applications, Status, Logs, and Help. Below the menu is a yellow warning banner that reads "Changes made! Click here to save permanently (Reboot the unit to cancel changes)". The main content area features the Mediatrix logo and a sidebar with a back arrow, a profile icon, and an "Active Profile: High" dropdown menu with a "Change" button. The main content area is divided into four columns: Configurations (Network, Security, Administration, Upgrade), Applications (SIP Switch Overview, SIP Server, E-mail, USB Web Server), Status (Network, Firewall Rules, SIP Sessions, SIP Users, USB Web Server), and Logs (Log Configuration, System Log, Firewall Log, SIP Log, Call Log). At the bottom, there are links for "Online User Manual" and "Configuration Wizard".

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

This is a close-up screenshot of the warning banner from the previous image. The banner is yellow and contains the text "Changes made! Click here to save permanently (Reboot the unit to cancel changes)". The link "Click here to save permanently" is highlighted with a red rectangular box.

6.3 In Network Configurations (2)

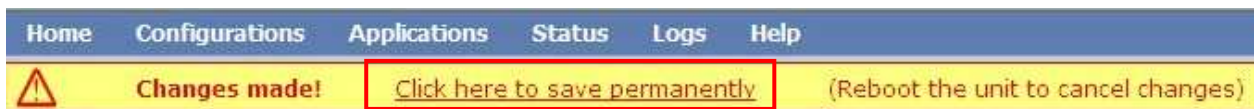
6.3-1 Move mouse over [Home] and Select [Overview]



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



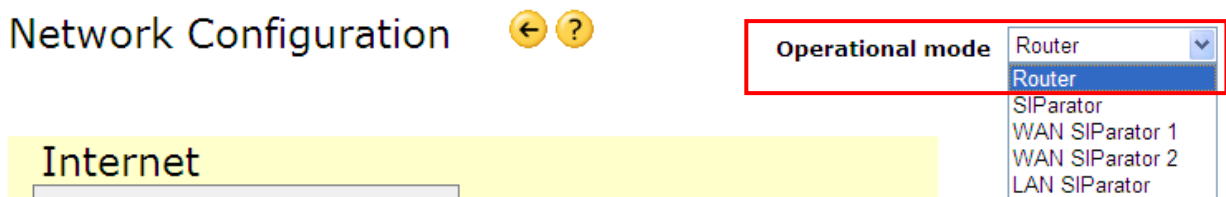
6.3-3 Click on [Click here to save permanently] and then Click on [Network].



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)

Network Configuration



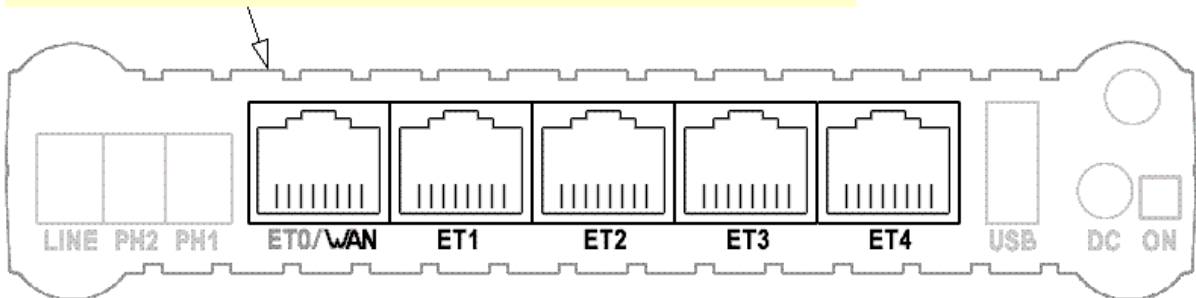
Operational mode

Router

Internet

ET0 used as	outside
IP Address	10.0.0.1
Subnet Mask	255.255.255.248
Access type	PPPoE
User	test-user@provider.c
Password	•••••
MAC address Dynamic DNS	

DNS Server	
<input type="checkbox"/> Obtain automatically	
IP Address	172.16.255.1
2:nd	
Default Gateway	
IP Address	10.0.0.1



6.3-7 Click on [Apply]

Apply

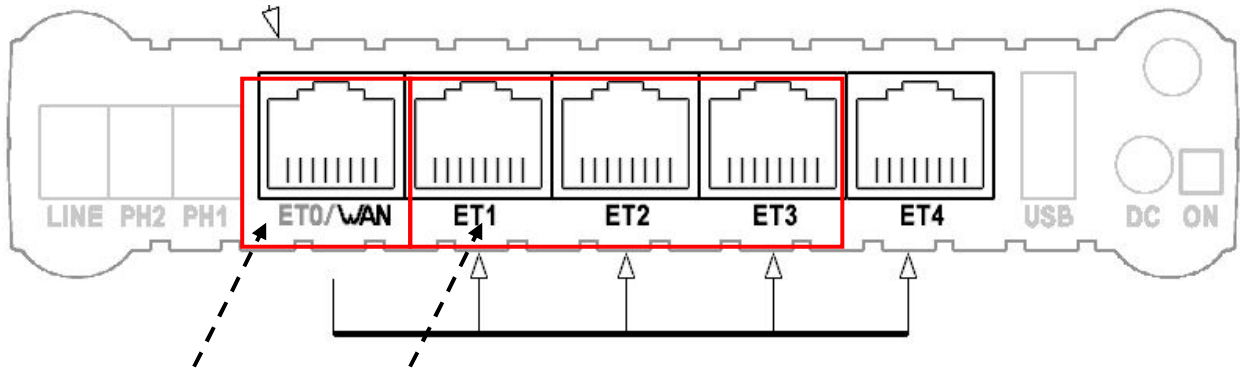
Apply & Return

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

Home Configurations Applications Status Logs Help

Changes made! [Click here to save permanently](#) (Reboot the unit to cancel changes)

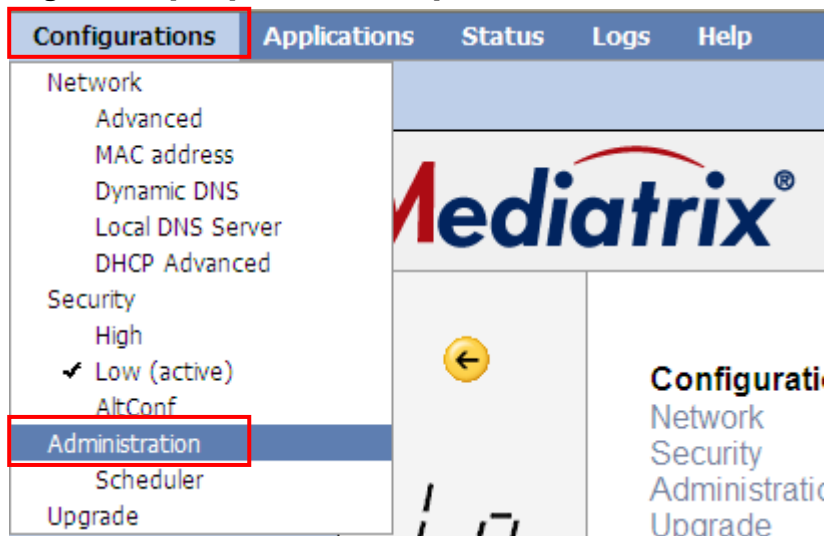
6.3-9 Connect to SBC **ET1** (to ET4) and Maintenance PC for existing LAN segment.



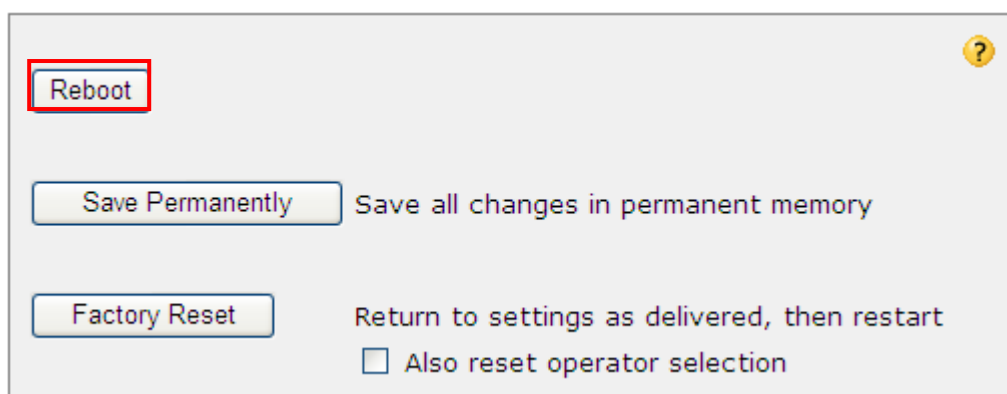
[Note] Connect the WAN. Connect to the LAN

6.3-10 Reboot the SBC (Recommended operations)

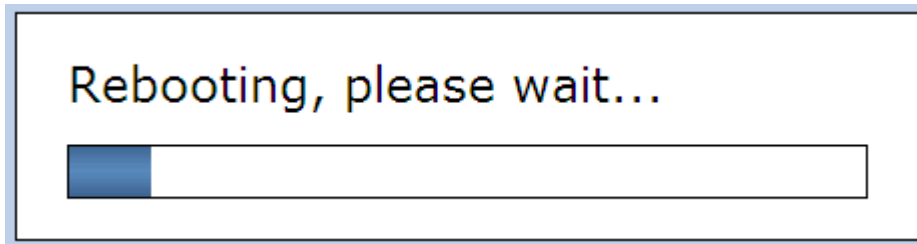
Select **[Configurations]** --> **[Administration]**



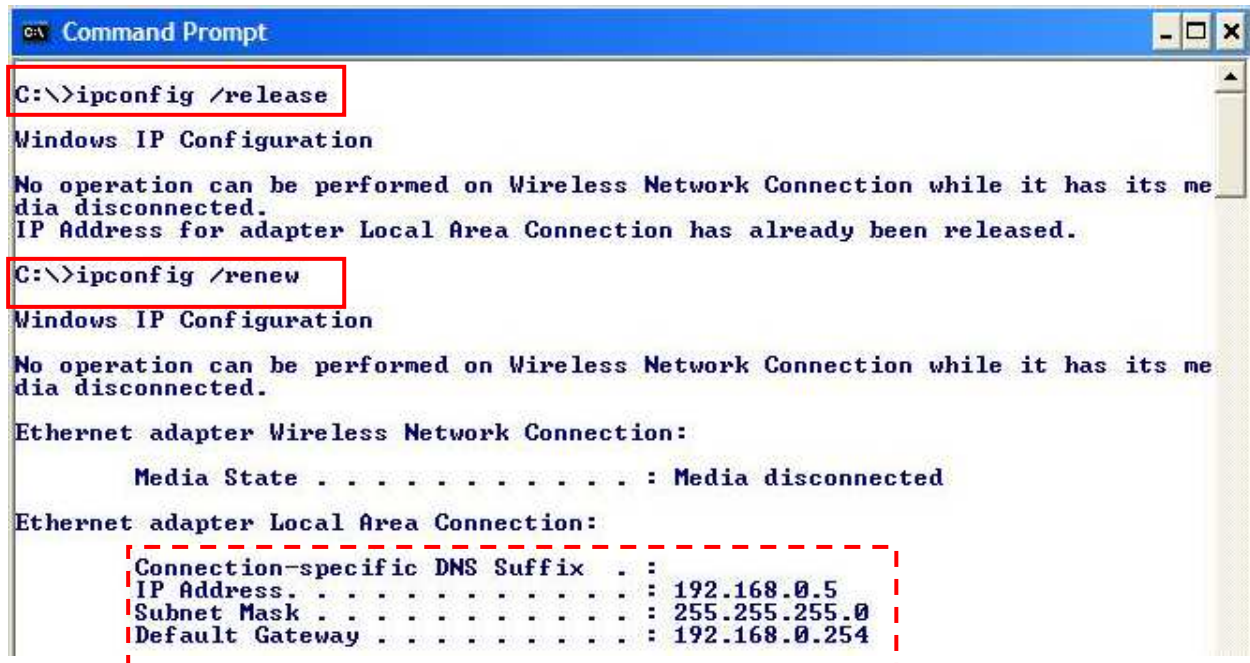
6.3-11 Click on **[Reboot]**



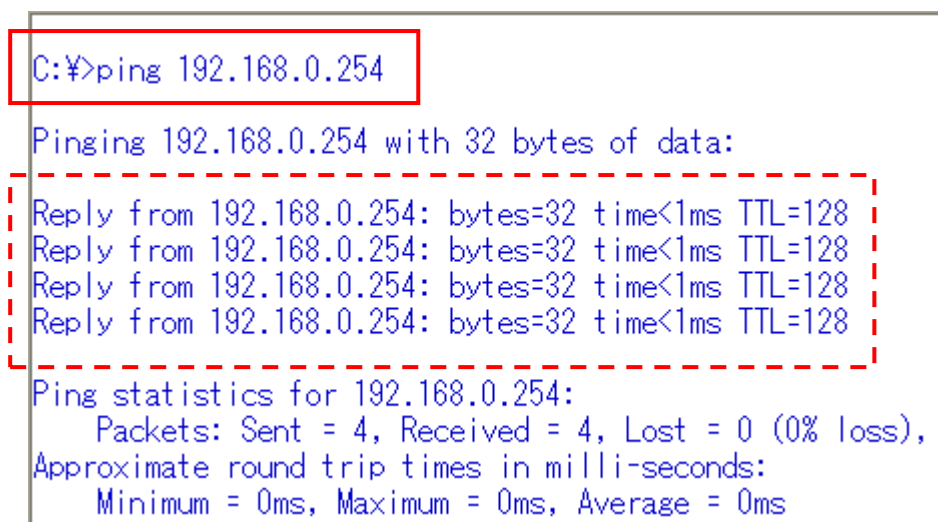
6.3-12 Wait for Rebooting.



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



Operational mode

Router



The screenshot shows the 'Internet' configuration section with two sub-sections highlighted by red boxes. The first sub-section contains: 'ET0 used as' set to 'outside', 'IP Address' as '10.0.0.1', 'Subnet Mask' as '255.255.255.248', 'Access type' as 'PPPoE', 'User' as 'test-user@provider.c', and 'Password' as masked characters. Below these are links for 'MAC address' and 'Dynamic DNS'. The second sub-section is 'DNS Server', with 'Obtain automatically' unchecked, 'IP Address' as '172.16.255.1', and a '2:nd' field. Below that is 'Default Gateway' with 'IP Address' as '10.0.0.1'. A yellow background highlights the entire configuration area.

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)

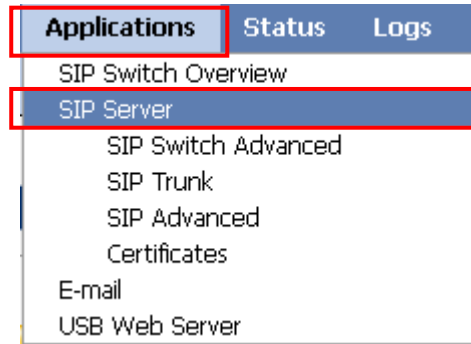
[DHCP Server]: Enable / From: **192.168.0.2** – **192.168.0.50** (Example)

The screenshot shows the 'LAN' configuration section. It includes: 'LAN' title with a help icon, 'inside' selected in a dropdown, 'IP Address' as '192.168.0.254', 'Subnet Mask' as '255.255.255.0', 'DHCP Server' checked, 'From:' as '192.168.0.2', and 'To:' as '192.168.0.50'.

[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)

SIP Server

A screenshot of the 'General SIP Server Settings' configuration page. The page has a light gray background and a yellow question mark icon in the top right corner. The settings are as follows:

- Use as SIP server for domain(s) Authentication Realm: [text input field]
- use client's domain name
- [text input field]
- Also apply number processing to domain(s) [text input field]
- Allow to register:
 - Inside users: All [dropdown menu]
 - Outside users: All [dropdown menu]
- Allow outgoing calls from: All [dropdown menu]
- and from others after authentication
- Match full SIP URI for incoming calls (E.g. "peter@company.com" instead of just "peter" must match.)

The 'Allow to register' and 'Allow outgoing calls from' sections are highlighted with a red border.

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

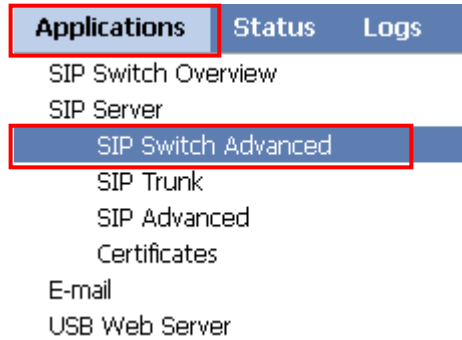


6.4-4 Click on **[Click here to save permanently]**



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

SIP Accounts and Incoming Call Processing

External voice mail server (domain, IP address or
 Allow external callers to use internal numbers

Account type ?	Ext. On ▾	SIP Address (e.g. "peter" if peter@...)	Authentication		Comment	Dyn. Regs.
			User ID	Password		
User ▾		testme;IVR1	testme	IVR port 1	1
User ▾		testservice;IVR2	testservi	IVR port 2	1
User ▾	301	301@192.168.0.101	301		0
User ▾	302	302@192.168.0.101	302		0
User ▾						0
User ▾						0

6.5-3 Click on [Apply]

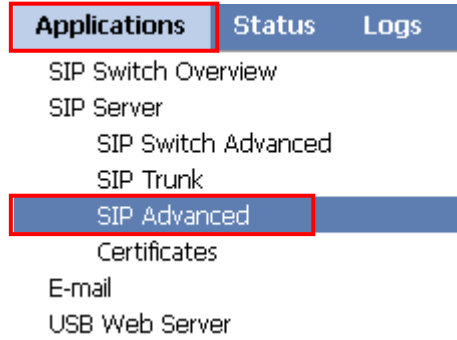


6.5-4 Click on [Click here to save permanently]



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]

Advanced SIP Settings

Get default values

Far End Nat Traversal (FENT) 

- Keep-alive packets interval UDP seconds TCP seconds (to keep SIP communications alive for clients needing FENT)
- Type of keep-alive packets SIP REGISTER SIP OPTIONS
- Detect endpoints behind same NAT (for shortest media path)

Authorized Users 

Method	URI	Direction	Allow	Authenticate	Authentication User IDs
REGISTER	*	inbound	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	*
INVITE	*@192.168.0.101	inbound	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	*
		outbound	<input type="checkbox"/>	<input type="checkbox"/>	
		outbound	<input type="checkbox"/>	<input type="checkbox"/>	

Allow RFC 2069 authentication

Reuse received nonces

Preferred QOP (authentication)

Brute force attack protection, allow authentication attempts within an interval of seconds

6.6-3 Configuration of SIP Proxy

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

SIP proxy	
Maximum number of active sessions	5
Maximum number of registrations per user	<input type="text" value="5"/>
Timeout before registration expires	<input type="text" value="3600"/> seconds
Call time out	<input type="text" value="3600"/> seconds
<input checked="" type="checkbox"/> Enable SIP session timer	
Default ring timeout	<input type="text" value="180"/> seconds
Maximum ring timeout	<input type="text" value="300"/> seconds
SIP Server UDP port numbers	<input type="text" value="5060"/>
SIP Server TCP port numbers	<input type="text" value="5060"/>
SIP Server TLS port numbers	<input type="text" value="5061"/>

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]**

Advanced	
SIP (3xx) redirection handling	Handled by this unit
TCP connection timeout	<input type="text" value="10"/> seconds
Max number of media streams per SIP request	<input type="text" value="5"/>
Accept	<input type="text" value="text/plain, text/html, */*"/>
<input type="checkbox"/> Always record route	
Music on Hold URI	<input type="text"/>
RTP media port range	<input type="text" value="35000"/> - <input type="text" value="35999"/>
Allow multiple RTP media senders	Allow any source IP addresses
<input checked="" type="checkbox"/> Allow RTP in reverse direction	
<input type="checkbox"/> Trim RTP header	
<input checked="" type="checkbox"/> Reuse port numbers within same session	
<input checked="" type="checkbox"/> Reuse port numbers when changing media (e.g. T.38 FAX)	
<input checked="" type="checkbox"/> Spoof protection	
<input checked="" type="checkbox"/> RFC 3261 Loose routing	
<input checked="" type="checkbox"/> Disable change to TCP (large messages)	
<input checked="" type="checkbox"/> Force Real Username on registrations	

6.6-5 Configuration of Trusted networks

Clear the check box [Enable]

Trusted networks (RFC 3325) ?

Enable

Accept identity in From header
Default Privacy Policy

IP address range

Begin	End	Protocol	Trusted certificates	Group
<input type="text"/>	<input type="text"/>	Any <input type="button" value="v"/>	All installed trusted cert <input type="button" value="v"/>	Authenticated <input type="button" value="v"/>
<input type="text"/>	<input type="text"/>	Any <input type="button" value="v"/>	All installed trusted cert <input type="button" value="v"/>	Authenticated <input type="button" value="v"/>

6.6-6 Click on [Apply]

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

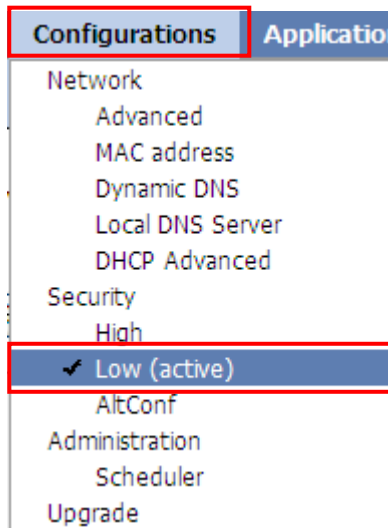
Home Configurations Applications Status Logs Help

Changes made! [Click here to save permanently](#) (Reboot the unit to cancel changes)

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)]**.



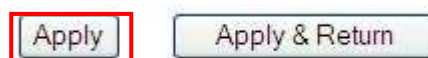
6.7-2 Configure Port redirection into the NS1000 for TCP connections and UDP connections.

Port redirection ?						
TCP connections			UDP connections			
outside port(s)	inside host	inside port	outside port(s)	inside host	inside port	
7547	192.168.0.101		123	192.168.0.101		
37547	192.168.0.101					
7580	192.168.0.101					
37580	192.168.0.101					

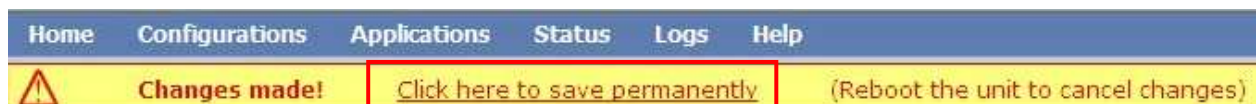
Contents of Port redirection (also known as Port forwarding) Settings (Example)

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.

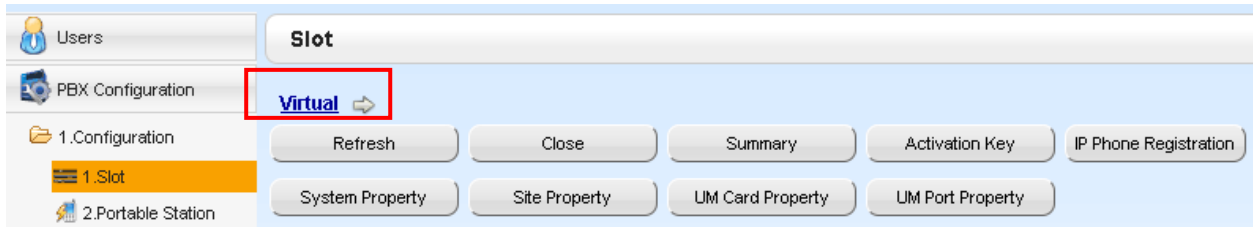
If the customer has needs, we can register the 3rd party SIP Phones.

For example, the Media5fone. They have to set a registration method of V-SIPEXT.

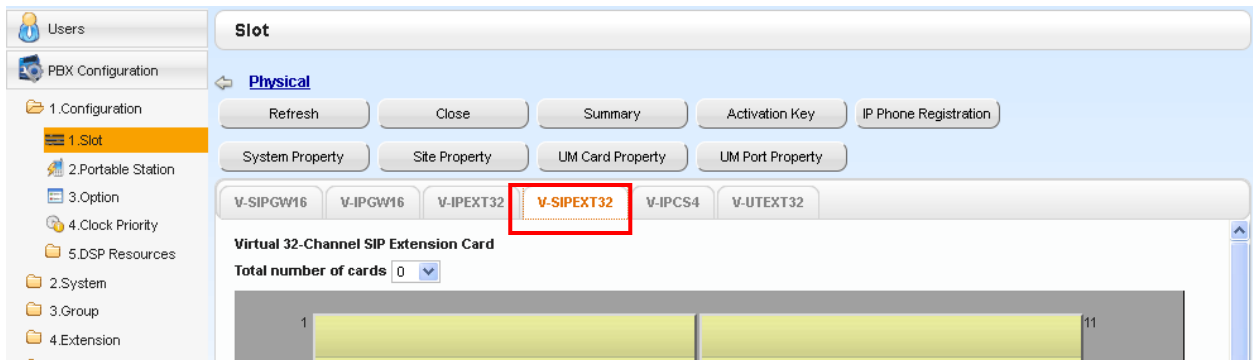
8.1 How to make the new SIP Extension (Example)

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

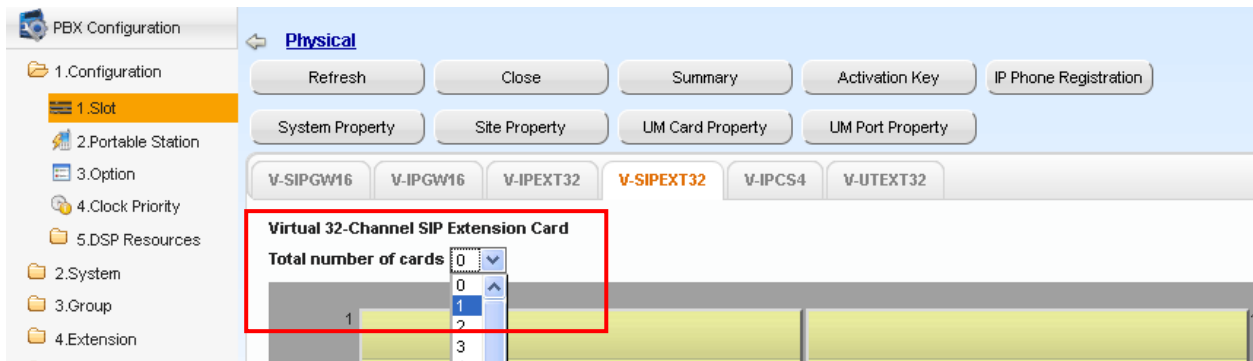
Click on [Virtual]



8.1-2 Click on [V-SIPEXT32]

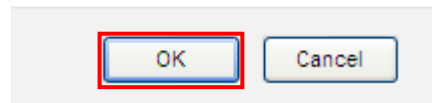


8.1-3 Click on [Total number of cards] and Select: 1 (Example)



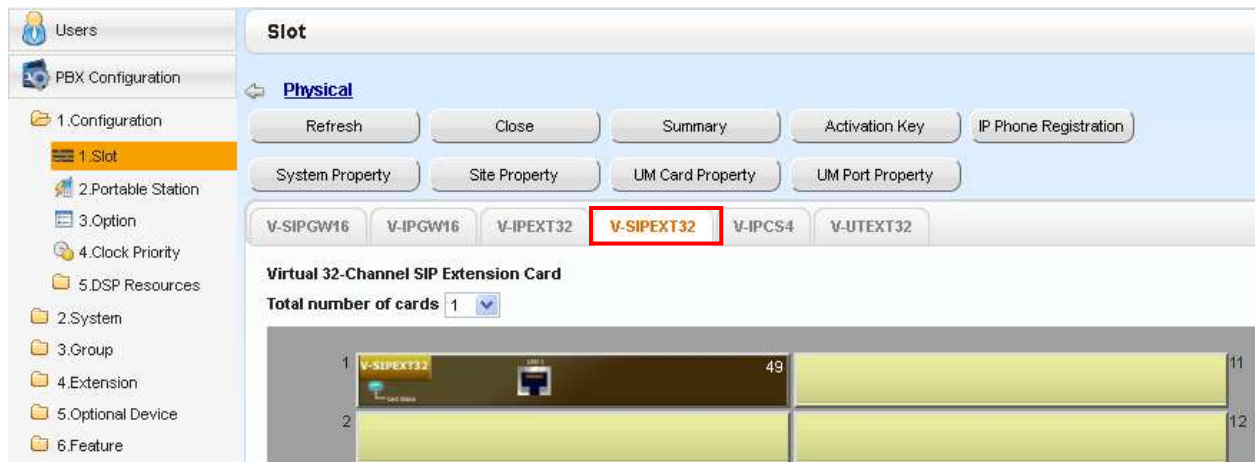
8.1-4 Click on [OK].

Are you sure you want to add 1 card?

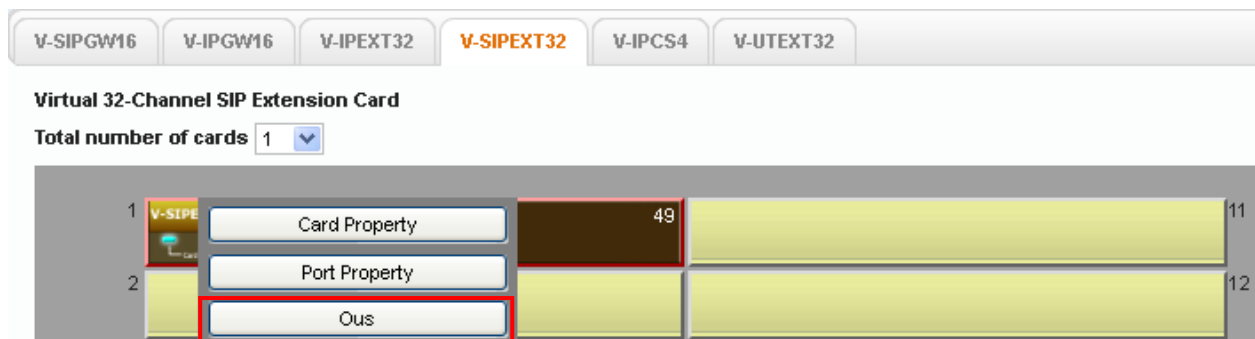


8.1-5 Configuration of V-SIPEXT32 Virtual slot.

Move mouse over installed [V-SIPEXT32] card

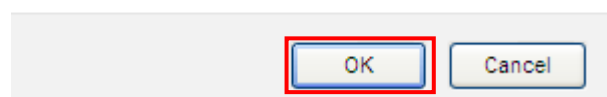


8.1-6 Select [OUS]

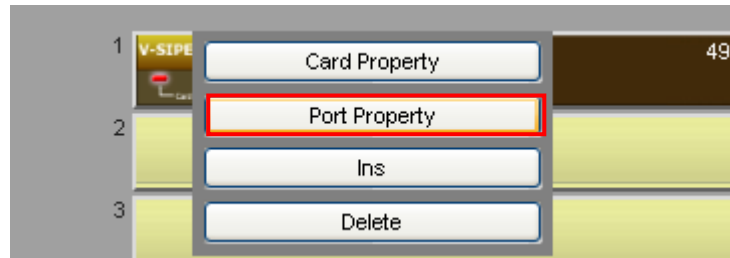


8.1-7 Click on [OK].

Are you sure you want to OUS (out of service) this card?

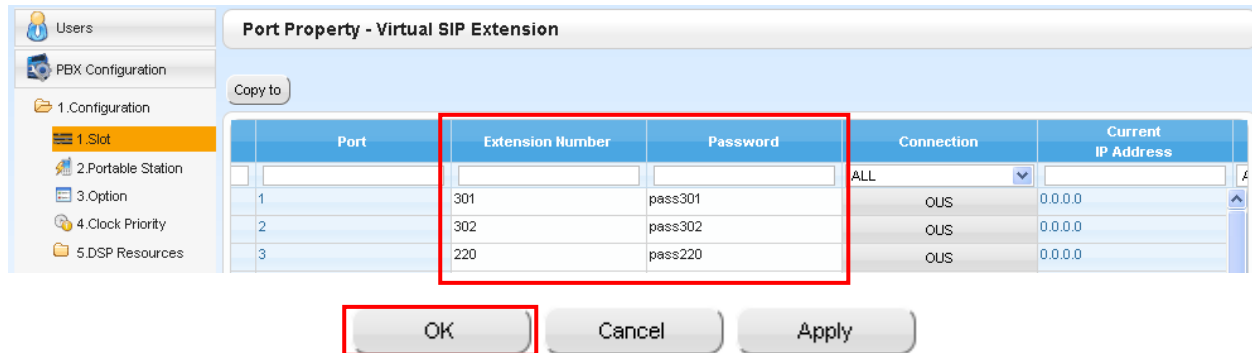


8.1-8 Select [Port Property]

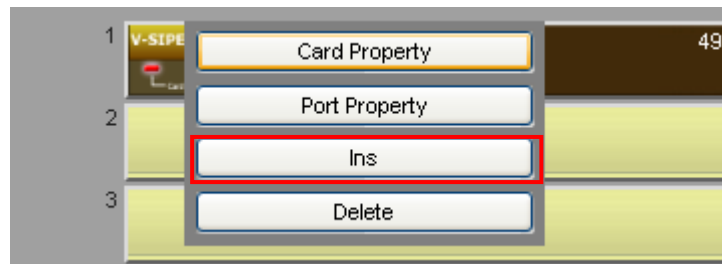


8.1-9 Edit the Extension Number and Password fields (click on them to enter data).

(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.

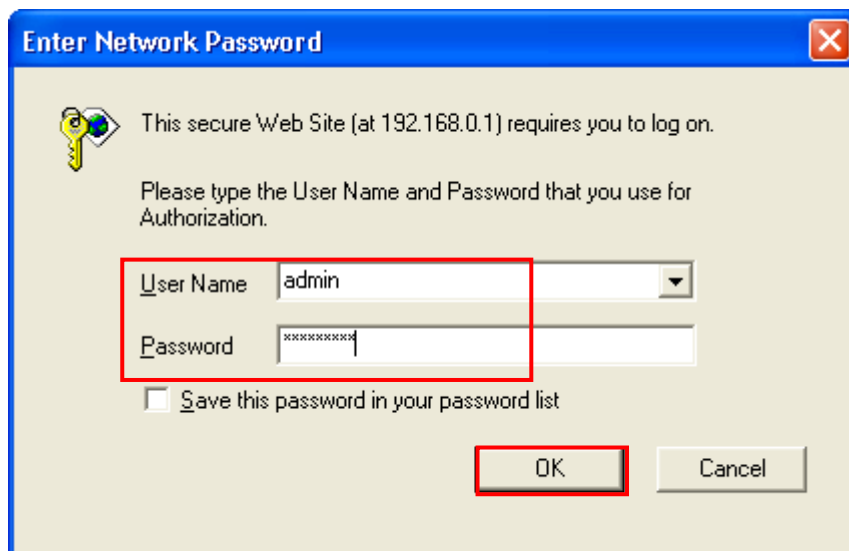
9.1-1 On the telephone, press [**Setting or Setup**] --> Select the [**Network Settings**] --> Press [**Enter**]
 --> Select the [**Embedded web**] --> Press [**ENTER**] --> Select [**ON**] --> Press [**ENTER**]
 --> [**Back**] --> [**Back**]. Or press [**Setting or Setup**] [#],[5],[3],[4] Select [**On**]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

Panasonic

KX-UT123

Status | Network | System | VoIP | Telephone | Maintenance

Version Information

Web Port Close

Status

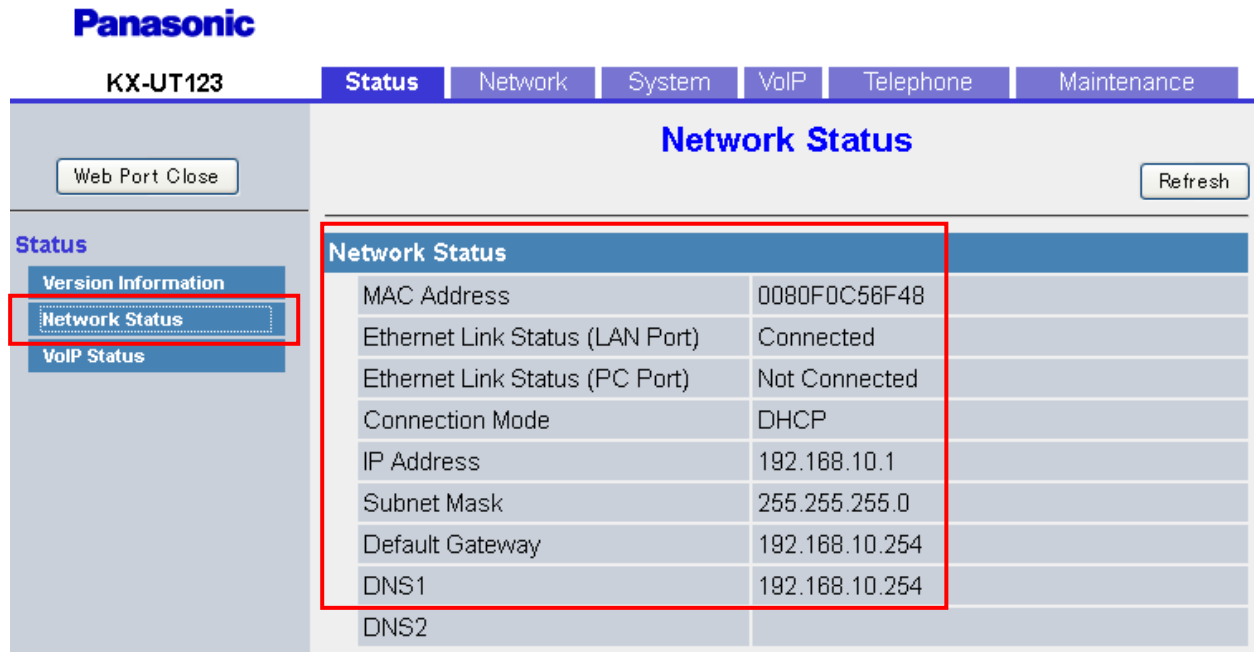
- Version Information
- Network Status
- VoIP Status

Version Information	
Model	KX-UT123
Operating Bank	Bank1
IPL Version	01.14
Firmware Version	Bank1: 01.160
	Bank2: 01.133

Running Version

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

Click on [Network Status]

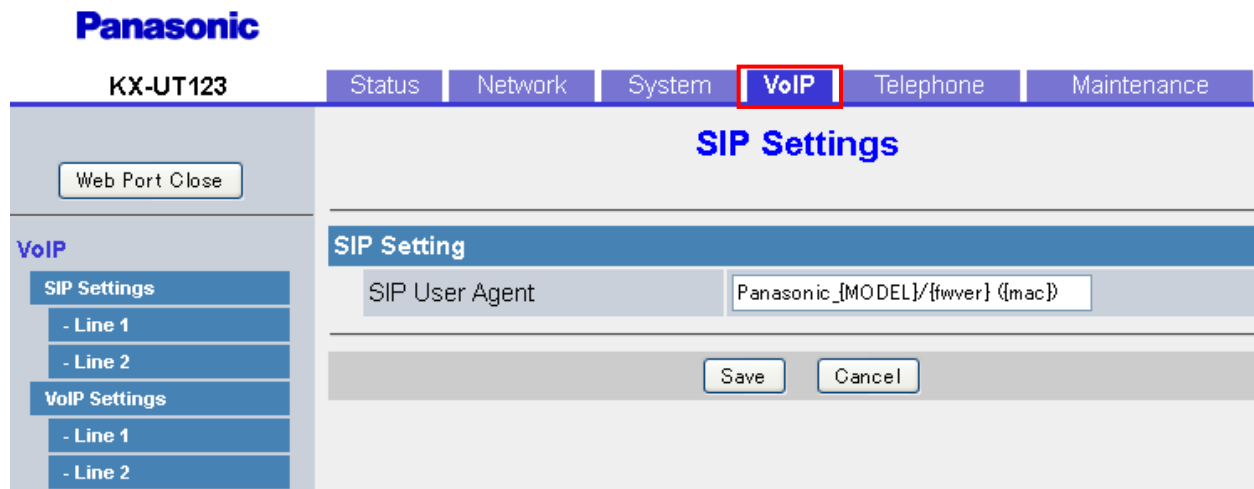


The screenshot shows the Panasonic KX-UT123 web interface. The top navigation bar includes 'Status', 'Network', 'System', 'VoIP', 'Telephone', and 'Maintenance'. The 'Network' tab is active, displaying the 'Network Status' page. A 'Web Port Close' button is in the top left, and a 'Refresh' button is in the top right. On the left sidebar, under 'Status', the 'Network Status' option is highlighted with a red box. The main content area shows a table of network parameters:

Network Status	
MAC Address	0080F0C56F48
Ethernet Link Status (LAN Port)	Connected
Ethernet Link Status (PC Port)	Not Connected
Connection Mode	DHCP
IP Address	192.168.10.1
Subnet Mask	255.255.255.0
Default Gateway	192.168.10.254
DNS1	192.168.10.254
DNS2	

9.2 In VoIP Setting

9.2-1 Click on [VoIP].



The screenshot shows the Panasonic KX-UT123 web interface with the 'VoIP' tab selected. The 'SIP Settings' page is displayed. The 'VoIP' sidebar on the left has 'SIP Settings' selected, with sub-options for '- Line 1' and '- Line 2'. The main content area shows the 'SIP Setting' for the selected line, with the 'SIP User Agent' field containing the text 'Panasonic_{MODEL}/{fwver} ({mac})'. 'Save' and 'Cancel' buttons are located at the bottom of the form.

9.2-2 Click on [Line 1]



This is a close-up of the VoIP sidebar from the previous screenshot. It shows the 'SIP Settings' section with two sub-items: '- Line 1' and '- Line 2'. The '- Line 1' item is highlighted with a red box, indicating it is the selected configuration.

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

Phone Number	
Phone Number	301
SIP URI	
SIP Server	
Registrar Server Address	10.0.0.1
Registrar Server Port	5060 [1-65535]
Proxy Server Address	10.0.0.1
Proxy Server Port	5060 [1-65535]
Presence Server Address	
Presence Server Port	5060 [1-65535]

SIP Settings [Line 1] (1of 3)
 Phone Number: **301**

[SIP Server]
 Register Server Address: **10.0.0.1**
 Register Server Port: **5060**
 Proxy Server Address: **10.0.0.1**
 Proxy Server Port: **5060**

Note: Replace 10.0.0.1 with the WAN address of Head office main router. Change to global IP address.

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

SIP Service Domain	
Service Domain	192.168.0.101:5060
SIP Source Port	
Source Port	25060 [1024-49151]
SIP Authentication	
Authentication ID	301
Authentication Password	●●●●●●●●

SIP Settings [Line 1] (2of 3)
 SIP Service Domain: **192.168.0.101:5060**
 It's PBXs LAN IP Address and SIP Port Number
 SIP Source Port: **25060**

Authentication ID: **301**
 Authentication Password: **pass301**

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

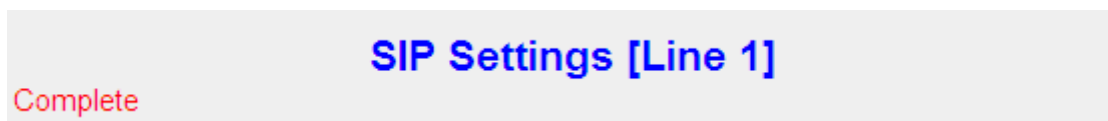
NAT Identity	
Keep Alive Interval	15 seconds [10-300]
Supports Rport (RFC 3581)	<input checked="" type="radio"/> Yes <input type="radio"/> No

SIP Settings [Line 1] (3of 3)
 Keep Alive Interval: **15** (Seconds)
 Support Rport: Click on **[Yes]**

9.2-6 Click on [Save]

Save
Cancel

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

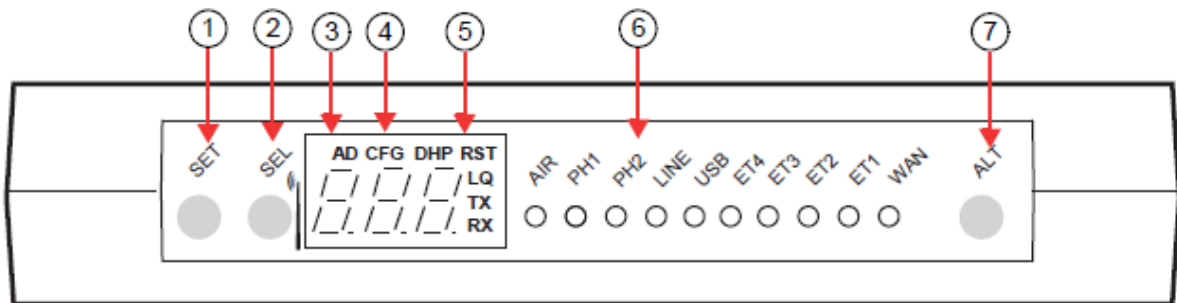
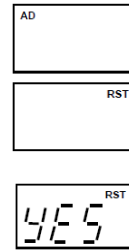
<http://www.mediatrix.com/en/sessionbordercontroller> Under the documentation tab.

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.

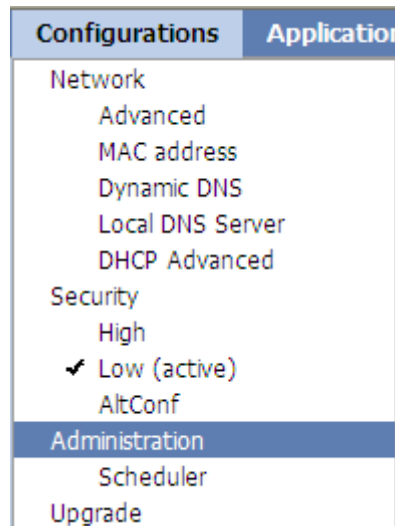
1. Press and hold [SET] (1) pressed for 3 seconds, to enter setup mode.
2. Press [SEL] (2) repeatedly until "RST" appears in the display.
3. Press [SET](1).
4. The question "Clear all?" appears, and then "no".
5. Press [SEL](2) to choose "YES".
6. Press [SET](1).



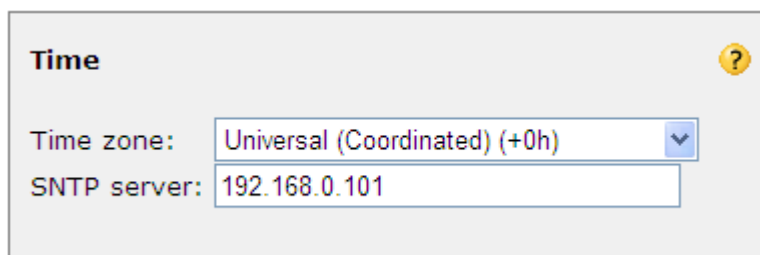
11.2 Time Setting

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

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



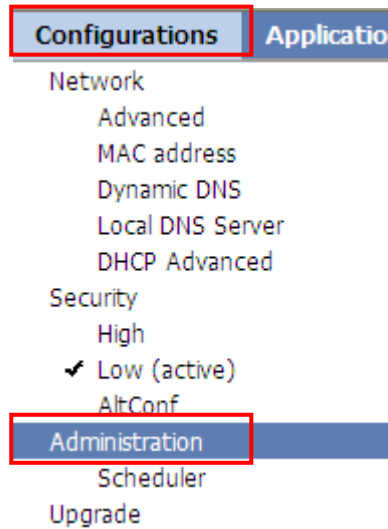
11.2-2 Configure the Time Server IP address, this IP is PBX IP address.(Example)



11.2-3 Click on [Apply]

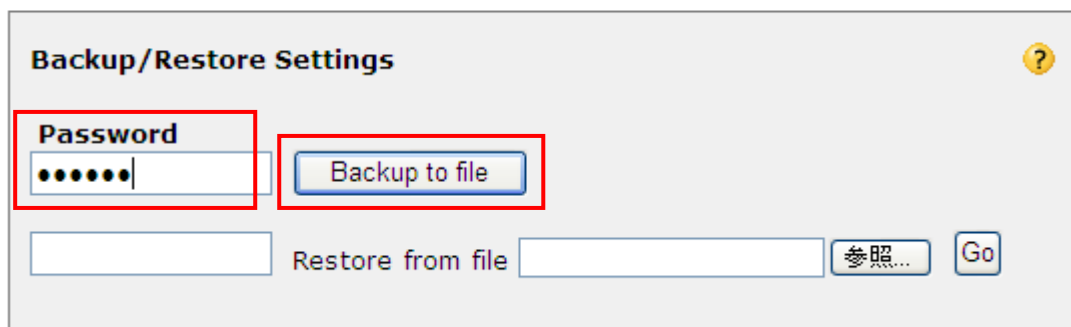
11.3 SBC Configuration Backup

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

Administration ← ?

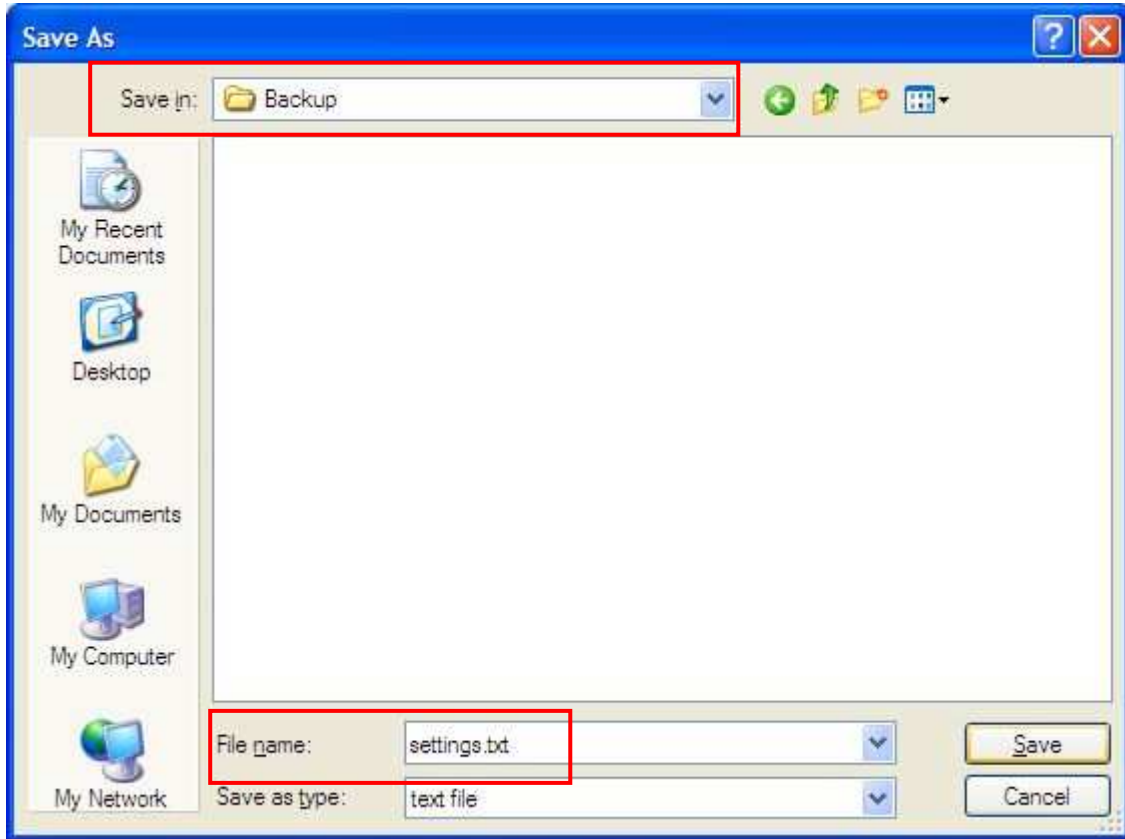


11.3-3 Click on [Save]



11.3-4 Save As

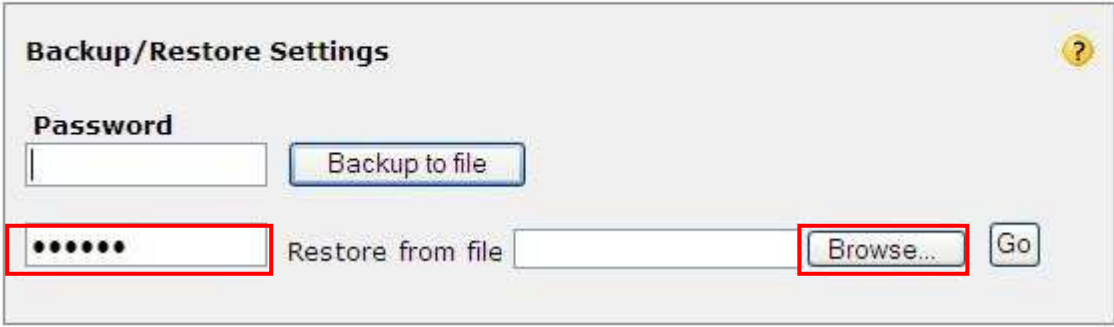
Select the Save Folder and Enter the File name [settings.txt] Example(Default).



11.4 Restore Settings

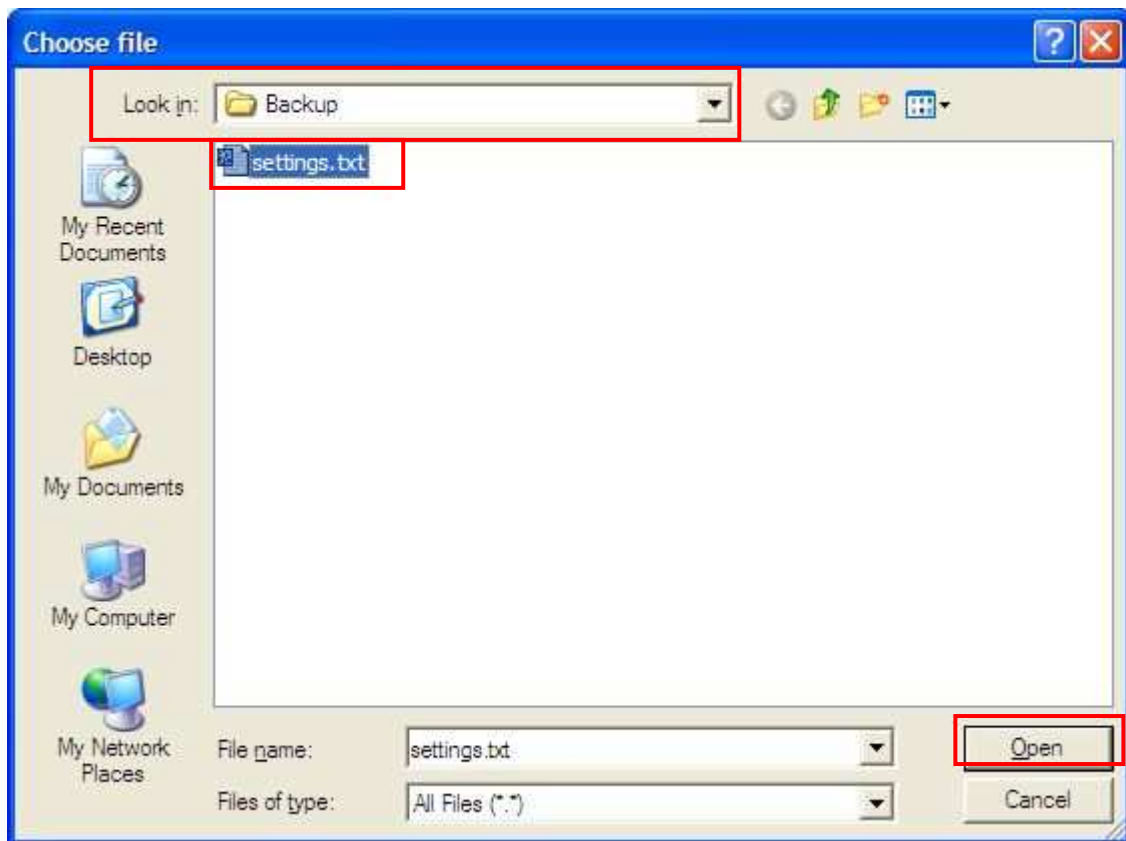
11.4-1 Enter the Password: **123456** (When at saving) and then Click on [**Browse...**].

Administration



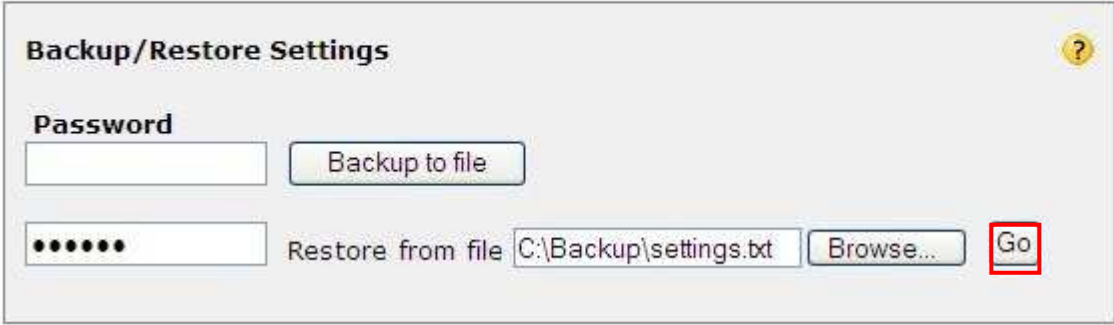
The screenshot shows the 'Backup/Restore Settings' dialog box. It has a title bar with a question mark icon. The main area contains a 'Password' field with a 'Backup to file' button to its right. Below this, there is a field with six dots (representing a password) and a 'Restore from file' label. To the right of this field is a 'Browse...' button and a 'Go' button. Red boxes highlight the password field and the 'Browse...' button.

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



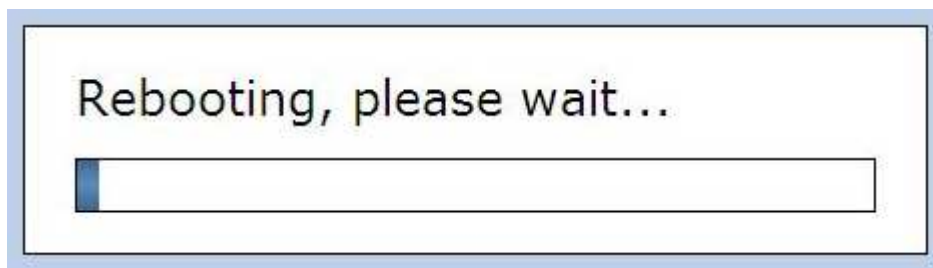
11.4-3 Click on [Go]

Administration



The screenshot shows a dialog box titled "Backup/Restore Settings" with a help icon in the top right corner. It contains two password input fields. The first field is empty, and the second field contains six dots. To the right of the first field is a "Backup to file" button. To the right of the second field is the text "Restore from file" followed by a text box containing "C:\Backup\settings.txt", a "Browse..." button, and a "Go" button which is highlighted with a red rectangular border.

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]

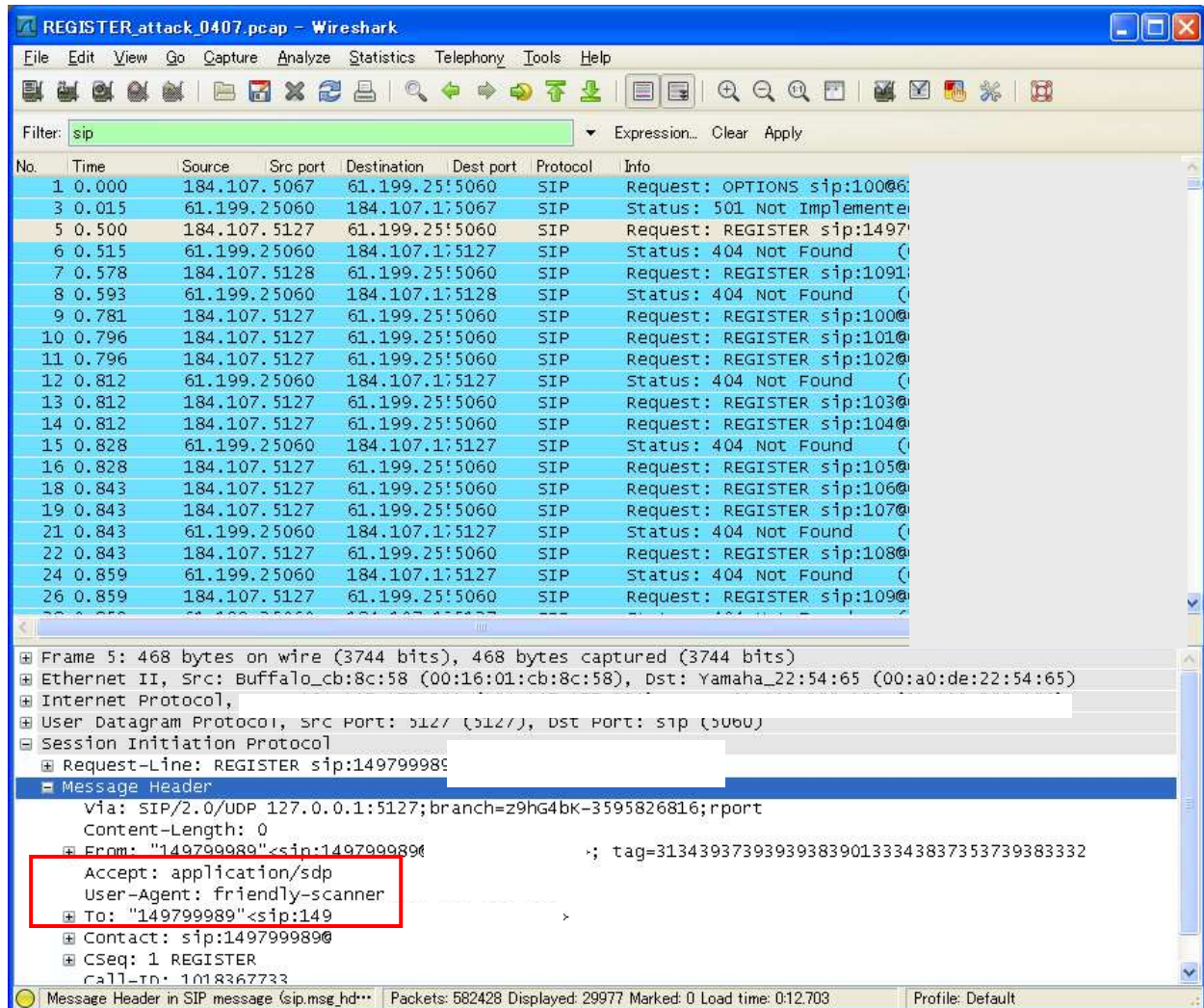
12 Troubleshooting

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.

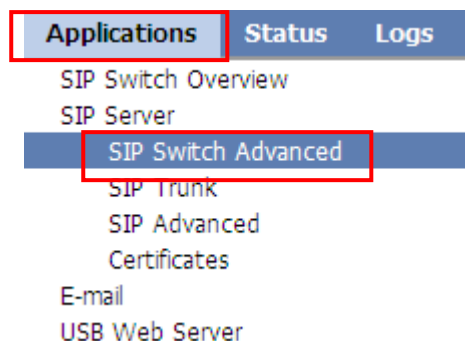
The symptom of this type of attack is the PBX temporarily becomes un-responsive, (It is very busy sending "404 Not Found" messages until the attacks over).



Countermeasure:

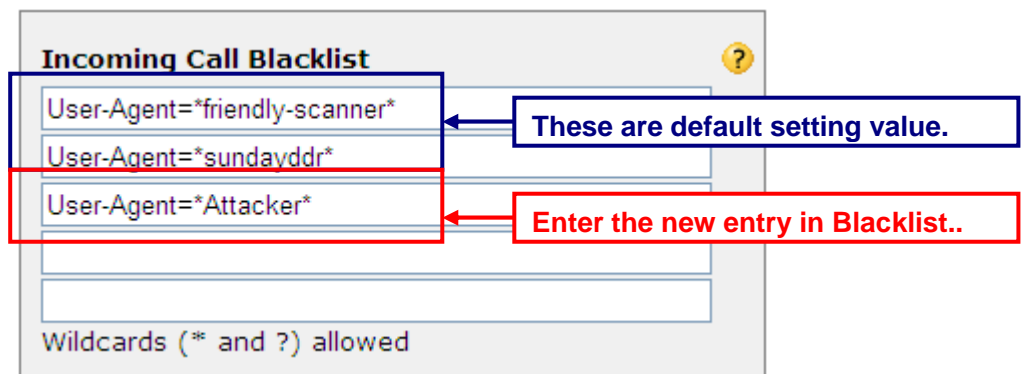
In the Switch Advanced, Configure a new entry in the Incoming Call Blacklist from captured packets.

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