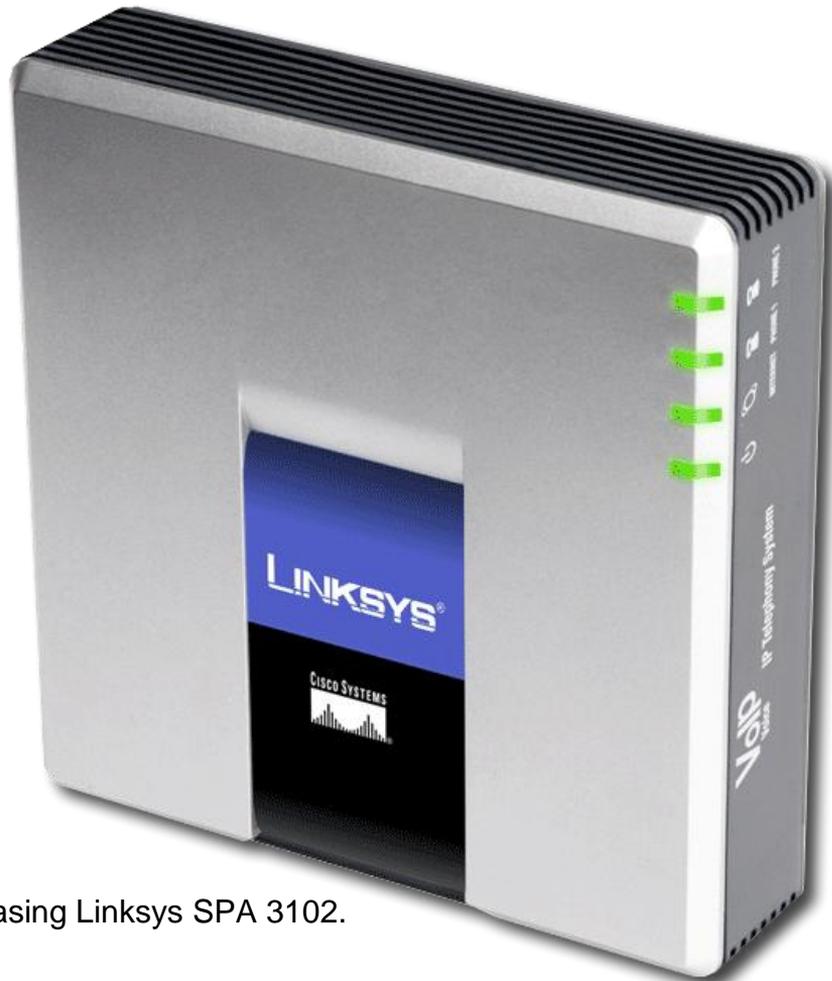


Linksys SPA3102



Thank you for purchasing Linksys SPA 3102.

Package Contents:

- SPA3102 Body
- AC Adapter
- Modular Cable
- Ethernet Cable
- Instruction Manual

Disclaimer

- Product functions and specifications are subject to change without prior notice.
- Quality of call (e.g. choppy line, noise, etc.) may be affected by company's network environment.
- Our company does not assume any responsibilities for a difference in call quality due to personal hearing impairment.
- Our company is not responsible for service that may not be available outside the country.
- Linksys SPA3102 may temporarily suspend the service for routine maintenance without prior notice.
- We would not assume any liability for product failure due to device malfunction.

Linksys SPA 3102

Analog Calls Routed over the Internet

Internet telephone service is now more accessible than before. The SPA 3102 Voice Gateway is a part of Cisco Small Business Voice Gateways and ATAs. It allows automatic routing of

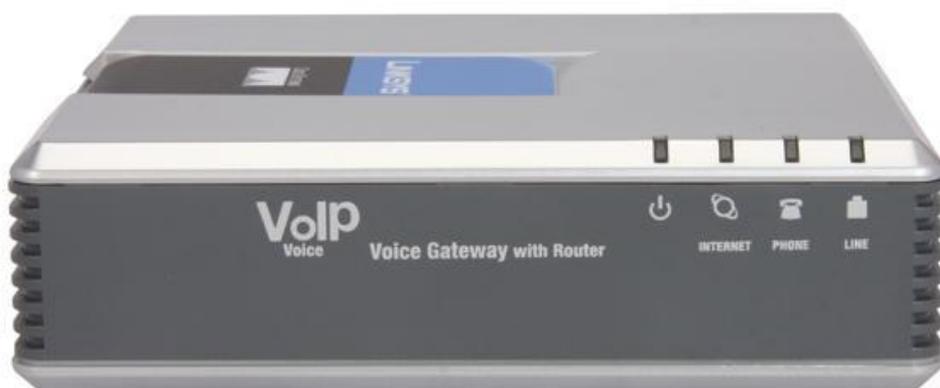
local calls from mobile phones and landlines to Voice over Internet Protocol (VoIP) service providers, and vice versa. Reduce or eliminate long-distance charges by first calling the local number on the SPA3102.

Calls can be safely redirected to a traditional carrier via standard analog interface in the event of power loss or Internet access is unavailable.

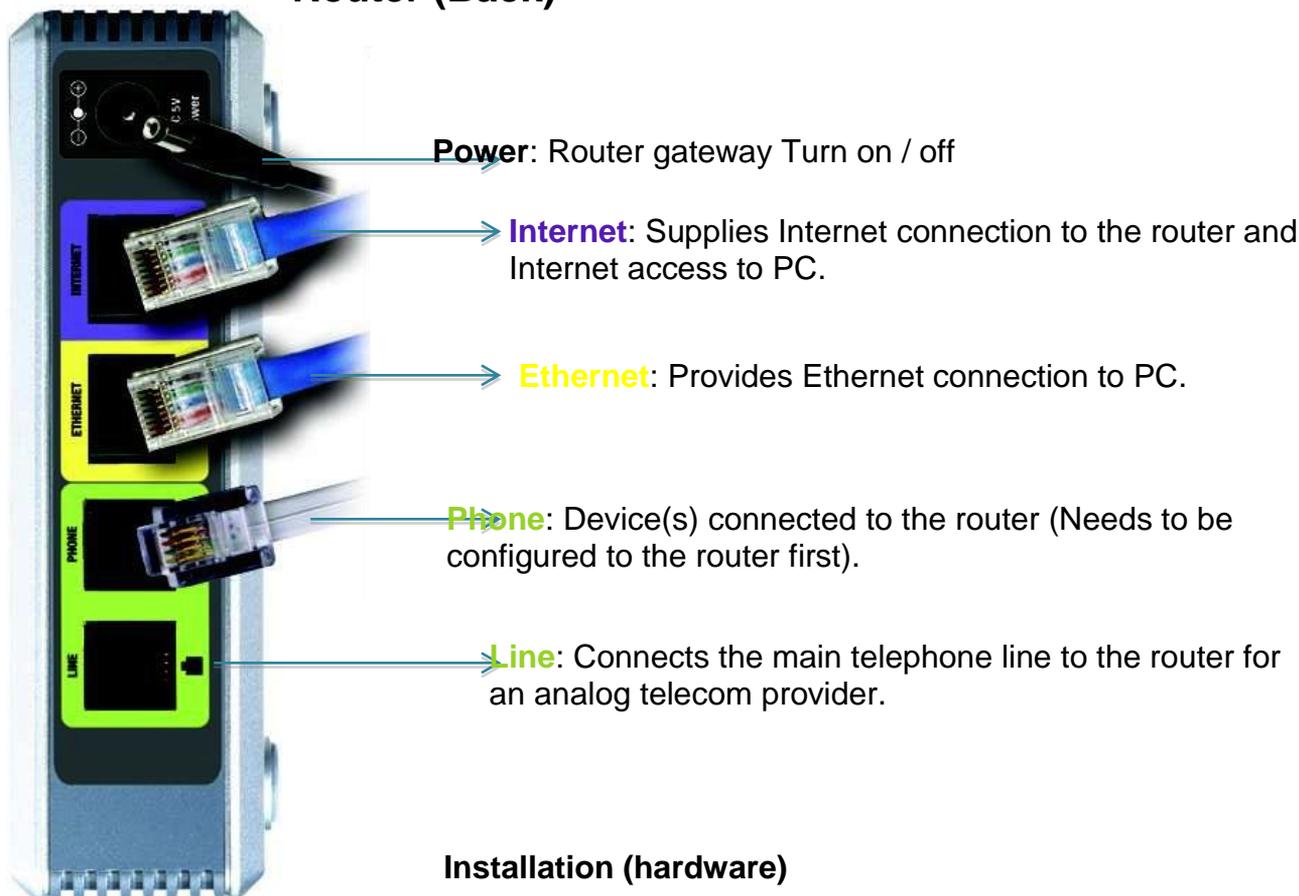
Additional Features

- One RJ-11 POTS (Plain Old Telephone Service) FXS port for an existing analog phone or fax machine connection.
- One PSTN FXO port for a Telco or PBX circuit connection.
- Two 100 BaseT RJ-45 Ethernet interfaces for home or office LAN connection and an Ethernet connection to a broadband modem or router.
- FXS and FXO lines that can be independently configured.
- Secured encryption-based methods for communication, provisioning and servicing.

Router (Front)



Router (Back)



1. Connect the power cable to your SPA3102.

2. Connect the network cable from your SPA3102 to your DHCP enabled modem / router OR connect the network cable from your modem to the SPA3102 WAN port and connect your PC via Ethernet cable to the LAN port of the SPA3102.
 3. Connect an analog phone to the [**Phone**] port in the SPA3102.
 4. Turn the power on.
 5. Pick up the phone and dial [* * * *] (You should hear a voice saying Linksys Configuration Menu.)
 6. Dial [**110#**]. (Write down the IP address that is returned.)
 7. Dial [**7932#**], then dial [**1#**] to enable the web server, then press [**1**] to save.
 8. Hang up.
 9. Connect your **PSTN** line to the [**Line**] port of the SPA3102.
- * You can opt to upgrade firmware.

Configuring SPA3102 via Web Interface

1. Access the web Interface by opening a web browser and type [**http://IPAddress of SPA3102**].
* By default the SPA 3102 does not have a username/password associated with it. If you are prompted for a username and password, please enter the default username and password: [**user**].
2. Click [**Admin Login**] → click [**Advanced**].

LINKSYS[®]
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Linksys Phone Adapter Configuration

Router | Voice

Status | Wan Setup | [Admin Login](#) | basic | [advanced](#)

Product Information

Product Name:	SPA-3102	Serial Number:	FM600F61
Software Version:	5.1.7(GW)	Hardware Version:	1.1.5
MAC Address:	000E08CC	Client Certificate:	Installed
Customization:	Open		

System Status

Current Time:	5/29/2008 13:44:14	Elapsed Time:	00:11:33
Wan Connection Type:	DHCP	Current IP:	192.168.
Host Name:	SipuraSPA	Domain:	
Current Netmask:	255.255.255.0	Current Gateway:	192.168.
Primary DNS:	192.168.		
Secondary DNS:			
LAN IP Address:	192.168.0.1	Broadcast Pkts Sent:	0
Broadcast Bytes Sent:	0	Broadcast Pkts Recv:	26
Broadcast Bytes Recv:	7101	Broadcast Pkts Dropped:	0
Broadcast Bytes Dropped:	0		

[Undo All Changes](#) | [Submit All Changes](#)

[Admin Login](#) | basic | [advanced](#)

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3. Click [**Voice**] → Click [**Regional**].

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Linksys Phone Adapter Configuration

Router | **Voice**

On [Distinctive Ring Pattern] set [Ring1 Cadence] to [60 (1/2)1]

Miscellaneous

Set Local Date (mm/dd):	30/01/08	Set Local Time (HH/mm):	19:05
Time Zone:	GMT+09:00	FXS Port Impedance:	600
Daylight Saving Time Rule:	start=4/1/7;end=10/-1/7;save=1		
FXS Port Input Gain:	-3	FXS Port Output Gain:	-3
DTMF Playback Level:	-16	DTMF Playback Length:	.1
Detect ABCD:	yes	Playback ABCD:	yes
Caller ID Method:	Bellcore(N.Amer,China)	Caller ID FSK Standard:	bell 202
Feature Invocation Method:	Default	More Echo Suppression:	no

Undo All Changes Submit All Changes

User Login basic | advanced

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On [Miscellaneous] set the following:
[Set Local Date (mm/dd)]
[Set Local Time (HH/mm)]
[Time Zone]: Ex. [GMT + 09:00]

To manually set NTP Server:

1. Click [Admin Login] → click [Router].
2. Click [Wan Setup].
3. Go to [Optional Settings] → in [Primary NTP Server] type [ntp.jst.mfeed.ad.jp].

* Click [Submit All Changes] to save changes made.

4. Click [Voice] → Click [Line 1].

Router		Voice	
Info	System	SIP	Provisioning
Regional		Line 1	
PSTN Line		User 1	
PSTN User		User Login	
basic		advanced	
Line Enable:	yes		
Streaming Audio Server (SAS)			
SAS Enable:	no		
SAS Inbound RTP Sink:			
SAS DLG Refresh Intvl:	30		
NAT Settings			
NAT Mapping Enable:	no		
NAT Keep Alive Enable:	yes		
NAT Keep Alive Msg:	\$NOTIFY		
NAT Keep Alive Dest:	\$PROXY		
Network Settings			
SIP ToS/DiffServ Value:	0x68		
SIP CoS Value:	3 [0-7]		
RTP ToS/DiffServ Value:	0xb8		
RTP CoS Value:	6 [0-7]		
Network Jitter Level:	very high		
Jitter Buffer Adjustment:	disable		
SIP Settings			
SIP Transport:	UDP		
SIP Port:	5060		
SIP 100REL Enable:	no		
EXT SIP Port:			
Auth Resync-Reboot:	yes		
SIP Proxy-Require:			
SIP Remote-Party-ID:	yes		
SIP GUID:	no		
SIP Debug Option:	none		
RTP Log Intvl:	0		
Restrict Source IP:	no		
Refer Target Bye Delay:	0		
Referer Bye Delay:	4		
Refer-To Target Contact:	no		
Sticky 183:	no		
Auth INVITE:	no		
Call Feature Settings			
Blind Attn-Xfer Enable:	no		
MOH Server:			
Xfer When Hangup Conf:	yes		
Proxy and Registration			
Proxy:	VoIP Server		
Outbound Proxy:			
Use Outbound Proxy:	no		
Use OB Proxy In Dialog:	yes		
Register:	yes		
Make Call Without Reg:	no		
Register Expires:	3600		
Ans Call Without Reg:	no		
Use DNS SRV:	no		
DNS SRV Auto Prefix:	no		
Proxy Fallback Intvl:	3600		
Proxy Redundancy Method:	Normal		
Voice Mail Server:			
Mailbox Subscribe Expires:	2147483647		
Subscriber Information			
Display Name:	Unique ID		
User ID:	VoIP Server		
Password:	Password		
Use Auth ID:	yes		
Auth ID:	Unique ID		
Mini Certificate:			
SRTP Private Key:			

Set the following:

[NAT Settings]

NAT Keep Alive Enables: Select **[Yes]**.

[Network Settings]

Network Jitter Level: Select **[Very High]**.

Jitter Buffer Adjustment: Select **[Disable]**.

[Proxy Registration]

Proxy: Enter **[VoIP Server]**.

[Subscriber Information]

Display Name, User ID, Auth ID: Enter **[Unique ID]**.

Password: Enter the password provided during registration. If you changed your password, enter your new password.

User Auth ID: Select **[Yes]**.

Supplementary Service Subscription

Call Waiting Serv:	no	Block CID Serv:	yes
Block ANC Serv:	yes	Dist Ring Serv:	yes
Cfwd All Serv:	yes	Cfwd Busy Serv:	yes
Cfwd No Ans Serv:	yes	Cfwd Sel Serv:	yes
Cfwd Last Serv:	yes	Block Last Serv:	yes
Accept Last Serv:	yes	DND Serv:	yes
CID Serv:	yes	CWCID Serv:	yes
Call Return Serv:	yes	Call Redial Serv:	yes
Call Back Serv:	yes	Three Way Call Serv:	no
Three Way Conf Serv:	yes	Attn Transfer Serv:	yes
Unattn Transfer Serv:	yes	MWI Serv:	yes
VMWI Serv:	yes	Speed Dial Serv:	yes
Secure Call Serv:	yes	Referral Serv:	yes
Feature Dial Serv:	yes	Service Announcement Serv:	no

Audio Configuration

Preferred Codec:	G711u	Silence Supp Enable:	no
Use Pref Codec Only:	yes	Silence Threshold:	medium
G729a Enable:	yes	Echo Canc Enable:	no
G723 Enable:	yes	Echo Canc Adapt Enable:	yes
G726-16 Enable:	yes	Echo Supp Enable:	no
G726-24 Enable:	yes	FAX CED Detect Enable:	yes
G726-32 Enable:	yes	FAX CNG Detect Enable:	yes
G726-40 Enable:	yes	FAX Passthru Codec:	G711u
DTMF Process INFO:	yes	FAX Codec Symmetric:	yes
DTMF Process AVT:	yes	FAX Passthru Method:	ReINVITE
DTMF Tx Method:	AVT	DTMF Tx Mode:	Strict
FAX Process NSE:	yes	Hook Flash Tx Method:	None
FAX Disable ECAN:	yes	Release Unused Codec:	yes
FAX Enable T38:	yes	FAX T38 Redundancy:	1
FAX Tone Detect Mode:	caller or callee	Symmetric RTP:	yes

Gateway Accounts

Gateway 1:		GW1 NAT Mapping Enable:	no
GW1 Auth ID:		GW1 Password:	
Gateway 2:		GW2 NAT Mapping Enable:	no
GW2 Auth ID:		GW2 Password:	
Gateway 3:		GW3 NAT Mapping Enable:	no
GW3 Auth ID:		GW3 Password:	
Gateway 4:		GW4 NAT Mapping Enable:	no
GW4 Auth ID:		GW4 Password:	

VoIP Fallback To PSTN

Auto PSTN Fallback: yes

Dial Plan

Dial Plan: (<0000:>xx.<:@gw0>|xx.|*x.|**x.|***x.|*x.*x.|*x.*x.*x.|#x.|#x.*x.|#x.*x.*x.)
Emergency Number:

Enable IP Dialing: no

FXS Port Polarity Configuration

Idle Polarity: Forward
 Caller Conn Polarity: Forward
 Callee Conn Polarity: Forward

Undo All Changes Submit All Changes

User Login basic advanced

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[Supplementary Service Subscription]
 Call Waiting Serv: Select [No].
 Three Way Call Serv: Select [No].

[Audio Configuration]
 User Pref Codec Only: Select [Yes].
 DTMF Tx Method: Select [AVT].
 FAX Disable ECAN: Select [Yes].
 FAX Enable T38: Select [Yes].
 Silence Supp Enable: Select [No].
 Echo Canc Enable: Select [No].
 Echo Supp Enable: Select [No].
 FAX Passthru Codec: Select [G711u].
 FAX Passthru Method: Select [ReINVITE].

[Dial Plan]
 Dial Plan: Enter (<0000:>xx.<:@gw0>|xx.|*x.|**x.|***x.|*x.*x.|*x.*x.*x.|#x.|#x.*x.|#x.*x.*x.)

5. Click [Voice] → Click [PST Line].

LINKSYS®
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Router | **Voice** | Info | System | SIP | Provisioning | Regional | Line | **PSTN Line** | User 1 | PSTN User | [User Login](#) | [Help](#) | [Advanced](#)

Line Enable: yes

NAT Settings
 NAT Mapping Enable: no
 NAT Keep Alive Enable: no
 NAT Keep Alive Msg: \$NOTIFY
 NAT Keep Alive Dest: \$PROXY

Network Settings
 SIP ToS/DiffServ Value: 0x68
 RTP ToS/DiffServ Value: 0xb8
 Network Jitter Level: very high
 SIP CoS Value: 3 [0-7]
 RTP CoS Value: 6 [0-7]
 Jitter Buffer Adjustment: disable

[NAT Settings]
 NAT Keep Alive Enables:
 Select **[No]**.
[Network Settings]
 Network Jitter Level:
 Select **[Very High]**.
 Jitter Buffer Adjustment:
 Select **[Disable]**.

Audio Configuration

Preferred Codec: G711u
 Use Pref Codec Only: yes
 G729a Enable: yes
 G723 Enable: yes
 G726-16 Enable: yes
 G726-24 Enable: yes
 G726-32 Enable: yes
 G726-40 Enable: yes
 DTMF Process INFO: yes
 DTMF Process AVT: yes
 DTMF Tx Mode: Strict
 FAX Process NSE: yes
 FAX Disable ECAN: yes

Silence Supp Enable: no
 Echo Canc Enable: no
 Echo Canc Adept Enable: yes
 Echo Supp Enable: no
 FAX CED Detect Enable: yes
 FAX CNG Detect Enable: yes
 FAX Passthru Codec: G711u
 FAX Codec Symmetric: yes
 FAX Passthru Method: ReINVITE
 DTMF Tx Method: AVT
 Release Unused Codec: yes
 Symmetric RTP: yes

[Audio Configuration]
 Use Pref Codec Ony: Select **[Yes]**.
 FAX Disabule ECAN: Select **[Yes]**.
 Silence Supp Enable: Select **[No]**.
 Echo Canc Enable: Select **[No]**.
 Echo Supp Enable: Select **[No]**.
 FAX Passthru Codec: Select **[G711u]**.
 FAX Passthru Method: Select **[ReINVITE]**.
 DTMF Tx Method: Select **[AVT]**.

PSTN-To-VoIP Gateway Setup

PSTN-To-VoIP Gateway Enable: no
 PSTN Ring Thru Line 1: yes
 PSTN CID For VoIP CID: no
 PSTN Caller Auth Method: none
 PSTN PIN Max Retry: 3
 PSTN CID Number Prefix:

[PSTN-To-VoIP Gateway Setup]
 PSTN-To-VoIP Gateway Enable: Select **[No]**.

* Click **[Submit All Changes]** to save changes made.

Router Configuration

* This setting applies only to SPA3102.

1. Click [Admin Login] → Click [Router] → Click [Wan Setup].

LINKSYS
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Linksys Phone Adapter Configuration

Router Voice
Status **Wan Setup** Lan Setup Application [User Login](#) [Basic](#) | [Advanced](#)

Internet Connection Settings
Connection Type: DHCP

Static IP Settings
Static IP: NetMask:
Gateway:

PPPoE Settings
PPPOE Login Name: PPPOE Login Password:
PPPOE Service Name:

Optional Settings
HostName: Domain:
Primary DNS: Secondary DNS:
DNS Server Order: Manual DNS Query Mode: Parallel
Primary NTP Server: ntp.jst.mfeed.ad.jp Secondary NTP Server:
DHCP IP Revalidate Timer: 0 minutes

MAC Clone Settings
Enable MAC Clone Service: no Cloned MAC Address:

Remote Management
Enable WAN Web Server: no WAN Web Server Port: 80

QoS Settings
QoS Policy: Always On
QoS QDisc: NONE Maximum Uplink Speed: 128 (Kbps)

VLAN Settings
Enable VLAN: no VLAN ID: 1 [0x000-0xFFFF]

[Undo All Changes](#) [Submit All Changes](#)

[User Login](#) [Basic](#) | [Advanced](#)

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[Internet Connection Settings]

[Static IP Settings]

[PPPoE Settings]

[Optional Settings]

Primary NTP Server: Enter [ntp.jst.mfeed.ad.jp].

[MAC Clone Settings]: Select [Yes].

* MAC Address is a 12-digit code assigned to a unique piece of hardware for identification.

[Remote Management]: Select [Yes].

* This depends on the assigned web server port.

* Click [Submit All Changes] to save changes made.



Precautions & Safety Instructions



The following instructions help use Linksys SPA3102 properly and prevent accidents. Failure to follow these instructions would void the product's warranty and accident may happen.



- Immediately unplug the device from the power supply if there is a strange smell and smoke. Don't use the device and immediately contact the dealer for repair. Do not disassemble or modify. The device may malfunction, cause electric shock and fire.



- Do not insert metal and keep the device dry. The device may malfunction, cause electric shock and fire.



- Do not touch the device when lightning occurs. Doing so may cause an electrical shock. Don't place near a flammable material. This may cause fire or explosion.



- Holding the phone on wet hands may cause electric shock. Don't place the device in an extreme hot or cold place. Sudden temperature change may cause malfunction of the device.



- Do not place in a damp, dusty or direct sunlight. This may trigger an explosion of the device. Place in an even and stable location.



- Do not place the phone and earpiece near an absorbent and magnet material. This may cause malfunction.



Initialization Method

Clear all data initialization process. You must set all the required information to start initializing your service again.

1. Note the (000B82000xxxx) MAC address located at the bottom of the unit.
2. Press the **[MENU]** button and select **[Config]**. Then, select **[Factory Reset]** and confirm by selecting **[OK]**.

PROBLEM	CAUSE	SOLUTION
General Failure	<p>Plug and unplug from the power source. (Do this a few times).</p> <p>Wait at least one minute before you again plug the device to the power source.</p> <p>Please provide your number and broadband products.</p>	
No ringer sound	<p>Ethernet cable is disconnected.</p> <p>AC adapter is disconnected.</p> <p>Did you change the volume settings?</p> <p>Did you change the settings on the display?</p>	<p>Check the connection of the Ethernet cable.</p> <p>Check the connection of the AC adapter.</p> <p>Adjust the volume on the volume button.</p> <p>Revert the changed settings. Disconnect the Ethernet cable (Please reboot).</p>
Call is interrupted by noise	Are there any abnormalities in the network environment?	Check your network environment.