Administrator's Manual

J&R SIP Phones

J&R Technology Ltd

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Notice

This manual provides the system administrator with a description for setting up and configuring the J&R's IP Phones.

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1 Introduction

This manual is intended for the system administrator who is responsible for setting up and configuring the IP Phones.
J&R provides a user-friendly Web interface that runs on a Web browser(Microsoft® Internet Explorer is the recommended browser).
For a detailed description on hardware installation and for operating the phone's call features, refer to the Quick Setup Manual.

Part I Web Configuration

2 Web Interface

This chapter describes the phone's embedded Web server interface that is used for configuring the phone.

2.1 Accessing Web Interface

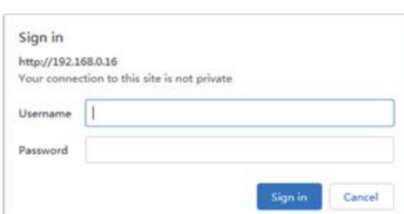
You can use any standard Web browser (such as Microsoft Internet Explorer) to access the phone's Web interface. The IP address used for accessing the Web interface is the phone's IP address, received from a DHCP server or manually configured (static IP address).

To access the phone's Web interface:

1. Connect the WAN port of your phone to the IP network (using the Cable or ADSL modem from your Internet Service Provider).
   Determine the phone's IP address obtained from the DHCP server, you can see the IP address of the phone on LCD screen or you can hear the voice broadcast for IP address when the phone connected the network.(It takes 40~60 seconds to connect the network successfully.)

2. Open a Web browser, and then in the URL address field, enter the phone's IP address (for example, https://192.168.0.2), as displayed below:

The Web login window appears: Web Login Window



3. Enter the User name and Password, and then click OK.
   Note: The administrator's default login user name and password are "admin" and "admin" respectively.
4. If there is no DHCP server, the IP address cannot be automatically obtained. The phone has a second IP address, and the second IP address is static. You can connect to it directly from this IP address. The IP address is 192.168.188.200.

2.2 Getting Started with the Web

The areas of the Web interface are shown below: Web Interface Areas Web Fields

The Web interface is composed of the following main areas:

- Home: opens the Home page
- Log off: closes the Web interface
- Navigation bar: provides tabs for accessing the configuration menus:
- Configuration: provides menus for configuring the phone.
- Management: provides menus for various management tasks such as firmware upgrade and changing the login user name and password.
- Status & Diagnostics: provides menus for displaying information on the status of the phone, such as call history.
- Navigation tree: tree-like, hierarchical structure of menus pertaining to the selected tab on the Navigation bar.
- Configuration pane: displays the configuration parameters pertaining to a selected menu in the Navigation tree.

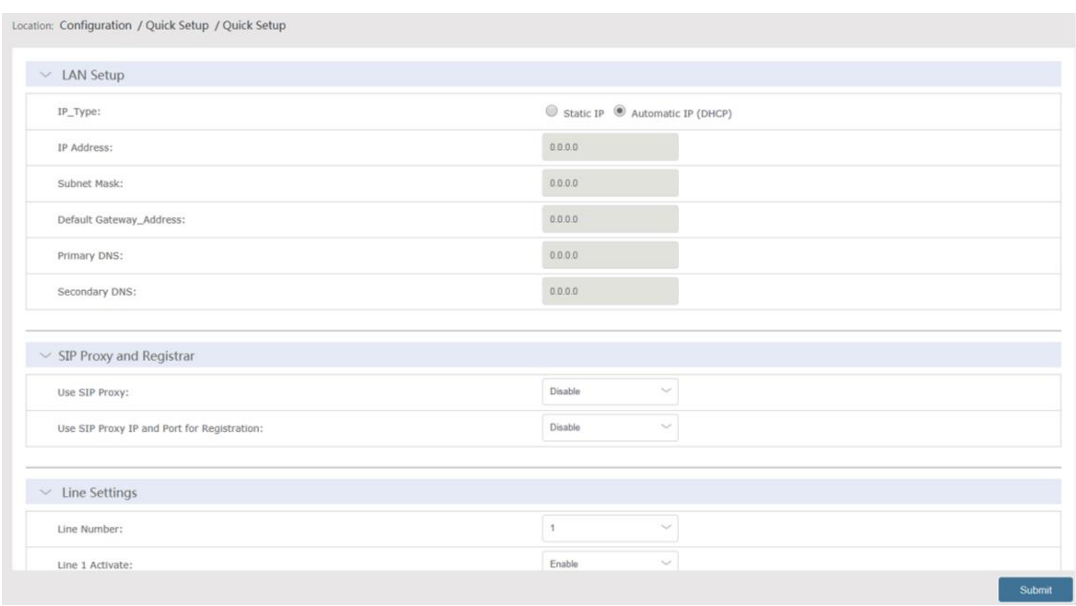
3 Quick Setup

The Web interface provides you with a Quick Setup page that allows you to configure basic features to quickly set up your IP Phone to operational level.

To quickly setup your phone:

1. Access the 'Quick Setup' page (Configuration tab >Quick Setup menu >Quick Setup).

Quick Setup Web Fields



For a description of the parameters on this page, refer to the following:

Parameters under the SIP Proxy and Registrar group see Part II VoIP Settings 9.2.1 on page 10.

Parameters under the LAN Setup group, see Part II VoIP Settings 13.1 on page 18.

4 Configuring Date and Time

4.1 Configuring Date and Time

Generally, a phone retrieves the date and time from a Network Time Protocol (NTP) server when it connects to the Internet. Alternatively, the date and time can be configured manually. NTP is a protocol for distributing the Coordinated Universal Time (UTC) by means of synchronizing te clocks of computer systems over packet-switched, variable-latency data networks.

Note: By default, the network settings are configured for automatic provisioning. However, if you need to change them, you can do it manually as is described in this part.

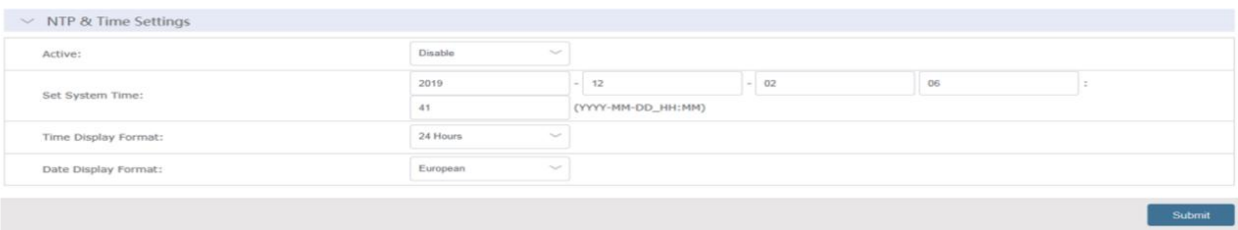
4.1.1 Configuring Date and Time

The date and time can be configured using the Web as described below.

To configure the date and time:

Access the ‘Date and Time’ page (Configuration tab >Advanced Applications menu >Date and Time).

Date and Time Web Fields



Configure the date and time and then click Submit.

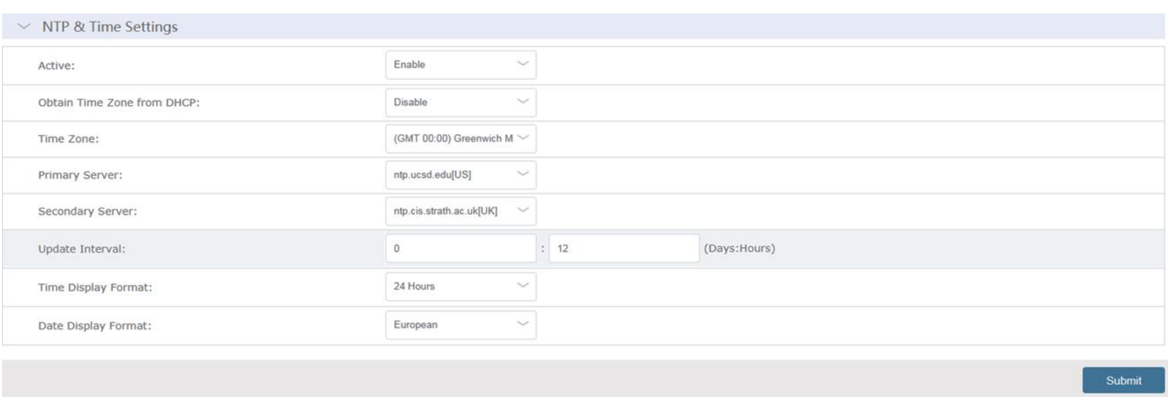
4.1.2 Configuring Network Time Protocol Server

The Network Time Protocol (NTP) server can be configured using the Web as described below.

To configure the Network Time Protocol Server:

1. Access the ‘Date and Time’ page (Configuration tab >Advanced Applications menu >Date and Time).
2. Configure the Network Time Protocol Server according to the parameters, and then click Submit.

NTP and Time Settings Web Fields



Configure the date and time and then click Submit.

Note: If both options (4 and 42) are received, the higher priority is given to Option 42.

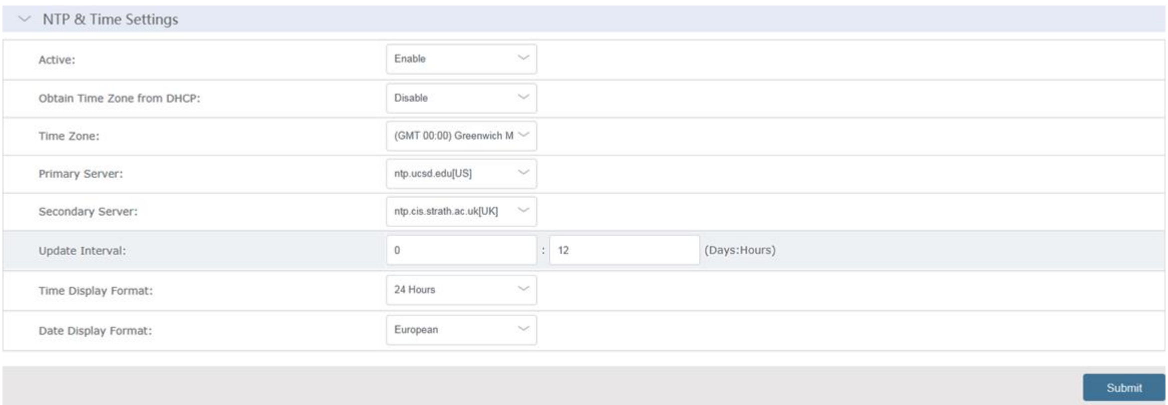
Time Zone – (Option 2)

The IP Phone sends an NTP request to the Primary NTP server. If there is no response, the NTP request is sent to the Secondary NTP server. After obtaining the time from the server it adds the GMT offset in Option 2 and this is the updated system time.

To manually configure the NTP Server and GMT offset via DHCP:

1. Access the 'Date and Time' page (Configuration tab >Advanced Applications menu >Date and Time).
2. From the "Obtain Time Zone From DHCP" drop-down list, select Disable.
3. Configure the NTP and Time Settings, and then click Submit.

NTP and Time Settings Web Fields



Note: These values will have no affect if the time zone is set to be obtained from DHCP. If the Time Zone and NTP servers are manually set, the IP Phone acts the same as described above, but the values are obtained from configuration file and not from DHCP.

5 IP Network Settings

5.1 Configuring Static IP Address

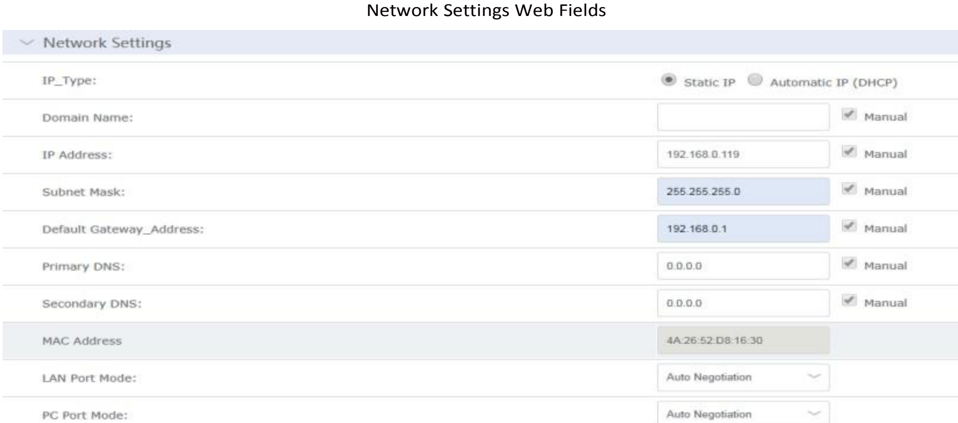
This section describes how to configure IP Network settings using the Web.

The phone's LAN configuration includes defining the method for obtaining an IP address. The phone's IP address can be static whereby the IP address is manually entered, or automatic whereby the IP address is acquired from a DHCP server. For Automatic IP, you can manually define some of the main parameters.

To define the phone's LAN settings:

Access the 'LAN Settings' page (Configuration tab >Network Connections menu >Network Settings).

Network Settings Web Fields



Configure the LAN settings and then click Submit.

6 LAN and PC Port Settings

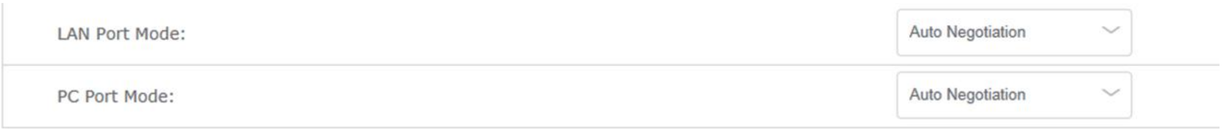
Port settings can be configured using the Web or Configuration File, as described below.

Note: The optional values of the Configuration file parameters are enclosed in square brackets while its corresponding Web values are written outside the square brackets, for example, [1] Enable.

To define the phone's port settings:

Access the 'LAN Settings' page (Configuration tab >Network Connections menu >LAN Settings).

Port Mode Web Fields



Configure Port Mode and then click Submit.

7 VLAN Settings

VLAN settings can be configured using the Web or Configuration File, as described below.

To configure the phone's VLAN settings:

Access the 'Network Settings' page (Configuration tab >Network Connections menu >Network Settings).

Configure Port Mode and then click Submit.

Part II VoIP Settings

8 SIP Settings

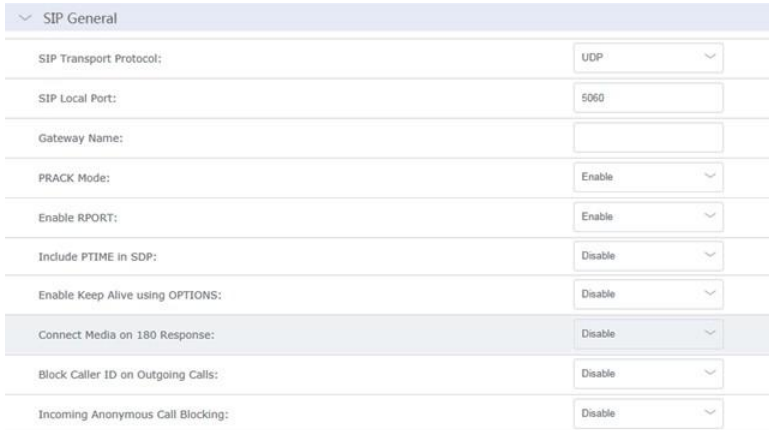
8.1 Configuring General SIP Settings

SIP General settings can be configured using the Web as described below.

To configure the phone's general SIP settings:

Access the 'Signaling Protocol' page (Configuration tab >Voice Over IP menu>Signaling Protocols).

SIP General Web Fields



Configure the General settings and then click **Submit**.

8.2 Proxy and Registration

This section describes the process for configuring Proxy and Registration settings.

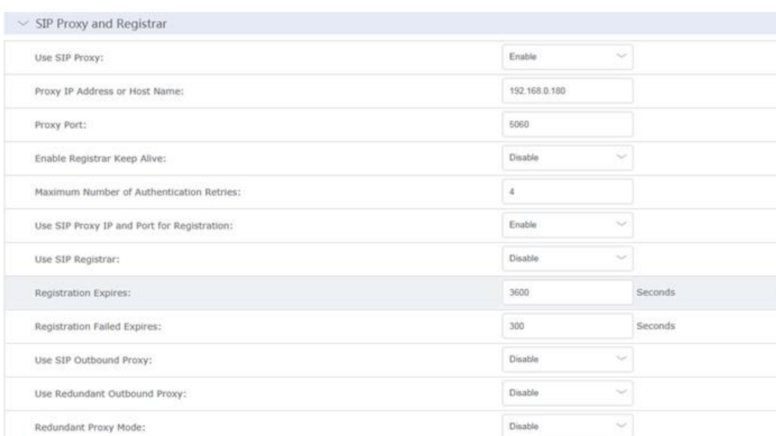
8.2.1 Configuring General Proxy and Registration Settings

Proxy and Registration settings can be configured using the Web as described below.

To configure the phone's Proxy and Registration settings:

Access the 'Signaling Protocol' page (**Configuration** tab >**Voice Over IP** menu>**Signaling Protocols**).

SIP Proxy and Registrar Web Fields



Configure Proxy and Registration and then click Submit.

8.2.2 Configuring Proxy Redundancy

The Redundant Proxy feature allows the configuration of a backup SIP proxy server to increase QoS stability. Once this feature is enabled, the phone identifies cases where the primary proxy does not respond to SIP signaling messages. In these scenarios, the phone registers to the redundant proxy and the phone seamlessly continues normal functionality, without the user noticing any connectivity failure or malfunction with the primary proxy.

The Redundant Proxy feature can operate in one of the following modes:

Asymmetric mode: The primary proxy is assigned a higher priority for registration than the redundant proxy. Once the phone is registered to the primary proxy, it sends keep-alive messages (using SIP OPTIONS messages) to the primary proxy. If the primary proxy does not respond, the phone registers to the redundant proxy, but continues sending keep-alive messages to the primary proxy. If the primary proxy responds to these keep-alive messages, the phone re-registers to the primary proxy.

Symmetric mode: Both proxies are assigned the same priority for registration. Once the phone is registered to a proxy, it sends keep-alive messages to this proxy. The phone switches proxies only once the proxy to which it has registered, does not respond.

To configure Proxy Redundancy:

Access the 'Signaling Protocol' page (Configuration tab >Voice Over IP menu>Signaling Protocols).

Proxy Redundancy Web Fields

Use Redundant Outbound Proxy:	Enable	
Redundant Oubound Proxy Address:	192.168.0.180	
Redundant Outbound Proxy Port:	5060	
Redundant Outbound Proxy Keep Alive Period:	60	Seconds
Redundant Outbound Proxy Symmetric Mode:	Disable	
Redundant Proxy Mode:	Disable	

Configure the SIP Redundancy parameters and then click Submit.

8.3 Configuring SIP Timers

SIP Timers can be configured using the Web as described below.

To configure the phone's SIP timer settings:

Access the 'Signaling Protocol' page (Configuration tab >Voice Over IP menu>Signaling Protocols).

SIP Timers Web Fields

SIP Timers	
Retransmission Timer T1:	500
Retransmission Timer T2:	4000
Retransmission Timer T4:	5000
INVITE Timer:	32000
Session-Expires:	1800
Min-SE:	90

Configure the SIP Timer parameters and then click Submit.

8.4 Configuring SIP QoS

SIP Quality of Service (QoS) can be configured using the Web as described below.

To configure the phone's SIP QoS:

Access the 'Signaling Protocol' page (Configuration tab >Voice Over IP menu>Signaling Protocols).

Quality of Service Web Fields

Quality of Service Parameters		
Type of Service (ToS):	0x60	Hex

Configure the SIP Quality of Service parameters and then click Submit.

9 Dialing Settings

9.1 Configuring General Dialing Settings

General Dialing settings can be configured using the Web as described below.

To define the dialing parameters:

Access the 'Dialing' page (Configuration tab >Voice Over IP menu >Dialing).

General Dialing Web Fields

Dialing Parameters		
Dialing Timeout:	5	Seconds
Phone Number Length:	32	Digits
Enable Dialing Complete Key:	Enable	
Dialing Complete Key:	#	

Configure Dialing options and then click Submit.

9.2 Tones

Tones settings can be configured using the Web as described below.

To configure the dial tones:

Access the 'Dialing' page (Configuration tab >Voice Over IP menu >Dialing).

Tones Web Fields in Dialing Page

Enable Replace Number Sign With Escape Char:	Disable	
Dial Tone Timeout:	30	Seconds
Reorder Tone Timeout:	40	Seconds
No Answer Call Timeout:	60	Seconds
Howler Tone Timeout:	120	Seconds
Secondary Dial Tone:	Enable	
Secondary Dial Tone Key:	9	

Configure Dialing options and then click Submit.

9.3 Configuring DTMF

DTMF can be configured using the Web as described below.

To configure DTMF:

Access the 'Dialing' page (Configuration tab >Voice Over IP menu >Dialing).

DTMF Transport Mode Web Field

DTMF Transport Mode:	RFC 2833

Configure the parameter and then click Submit.

9.4 Configuring Digit Maps and Dial Plans

Digit maps and Dial plans can be configured using the Web as described below.

To configure digit map and dial plan:

Access the 'Dialing' page (Configuration tab >Voice Over IP menu >Dialing).

Digit Map and Dial Plan Web Fields

Digit Map:	
Dial Plan:	

Configure the Digit Maps and Dial Plans parameters and then click **Submit**.

9.5 Configuring Relay Settings

The relay settings can be configured using the Web as described below.

To configure Relay Settings:

Access the ‘Dialing’ page (**Configuration** tab >**Voice Over IP** menu >**Dialing**).

Remote Trigger Key: To choose a key to trigger the relay 1 / relay 2 during two way communication calls. The key could be 0~9, *, #.

Triggering Duration: To set a time for the relay triggering duration. The time could be 1~99 seconds. (E.g. To choose key “1” to trigger relay 1, and set triggering duration time to be 5 seconds. During a two way communication call, when you press key “1” at another side, the relay 1 will be triggered and keeping for 5 seconds before switching back.)

Relay Settings Web Field

Relay 1		
Remote Trigger Key:	1	
Triggering Duration:	50	(1-99 Seconds)
Relay 2		
Remote Trigger Key:	2	
Triggering Duration:	50	(1-99 Seconds)

Configure Relay Settings and then click **Submit**.

9.6 Configuring Auto Answer

Digit maps and Dial plans can be configured using the Web as described below.

To configure Auto Answer:

Access the ‘Dialing’ page (**Configuration** tab >**Voice Over IP** menu >**Dialing**).

Auto Answer Web Fields

Auto Answer	
Activate::	Enable
Time Out:	5

Configure Relay Settings and then click **Submit**.

9.7 Configuring Cycle Dialing

Digit maps and Dial plans can be configured using the Web as described below.

To configure Cycle Dialing:

Access the ‘Dialing’ page (**Configuration** tab >**Voice Over IP** menu >**Dialing**).

Cycle Dialing Web Fields

Cycle Dialing		
Keep Time:	0	(0-999 Seconds)

Configure the parameters and then click **Submit**.

This setting is for speed dial type phones, the keep time is to set a time that outgoing call can not be canceled when you push the button again.

9.8 Configuring Cycle Dialing

Digit maps and Dial plans can be configured using the Web as described below.

To configure Automatic Dialing:

Access the 'Dialing' page (Configuration tab >Voice Over IP menu >Dialing).

Automatic Dialing Web Fields

Automatic Dialing		
Activate:	Enable	
Time Out:	3	Seconds
Destination Phone Number:	1003	

Configure the parameters and then click Submit.

This setting is for auto dial phones, as the setting on the above picture, the phone will dial to 8002 when you lift the handset.

9.9 Configuring Automatic Redial On Busy

Digit maps and Dial plans can be configured using the Web as described below.

To configure Automatic Redial On Busy:

Access the 'Dialing' page (Configuration tab >Voice Over IP menu >Dialing).

Automatic Redial On Busy Web Fields

Automatic Redial On Busy		
Activate:	Enable	
Time Out:	30	Seconds

Configure the parameters and then click Submit.

This setting is for auto dial phones, as the setting on the above picture, the phone will dial to 8002 when you lift the handset.

10 Media Settings

10.1 Configuring RTP Port Range and Payload Type

RTP Port Range and Payload Type can be configured using the Web as described below.

To configure the RTP Port Range and Payload Type:

Access the 'Media Streaming' page (Configuration tab >Voice Over IP menu >Media Streaming).

Automatic Redial On Busy Web Fields

Media Streaming Parameters	
RTP Port Range - Contiguous Series of 4 Ports Starting From:	4000
DTMF Relay RFC 2833 Payload Type:	101

Configure the RTP Port Range and Payload Type parameters and then click Submit.

10.2 Configuring RTP QoS

RTP QoS can be configured using the Web as described below.

To configure RTP QoS:

Access the 'Media Streaming' page (Configuration tab >Voice Over IP menu >Media Streaming).

QoS Web Fields

Quality of Service Parameters		
Type of Service (ToS):	0xb8	Hex

Configure the RTP Qos parameters and then click **Submit**.

10.3 Configuring Codecs

Codecs can be configured using the Web as described below.

To define the Codecs:

Access the 'Media Streaming' page (Configuration tab >Voice Over IP menu >Media Streaming).

QoS Web Fields

Codecs

Codec Priority	Codec Type	Packetization Time (milliseconds)
1st Codec	G.711, 64 Kbps, A-Law	20
2nd Codec	G.711, 64 Kbps, u-Law	20
3rd Codec	G.729, 8 Kbps	20
4th Codec	G.722/16000	20
5th Codec	None	30

Configure the Codecs parameters and then click **Submit**.

11 Gain Settings (Volume Levels)

The procedures for configuring volume levels are described below.

11.1 General Setting

General Setting can be configured using the Web or Configuration File, as described below.

To configure General Setting:

1. Access the 'Gain Settings' page (Configuration tab >Voice Over IP menu >Gain Settings).

General Setting Web Field

Gain Settings	
Tone Volume:	-10dB
Ringer Volume:	0dB

2. Configure the tones volume settings and ringer volume parameters and then click **Submit**.

Tone Volume: Call progress tone volume. The valid range is 1 to 31 (-dB). The default value is 10 (-10dB).

Ringer Volume: Ringing tone volume. The valid range is -31 to 20 dB. The default value is 0 dB.

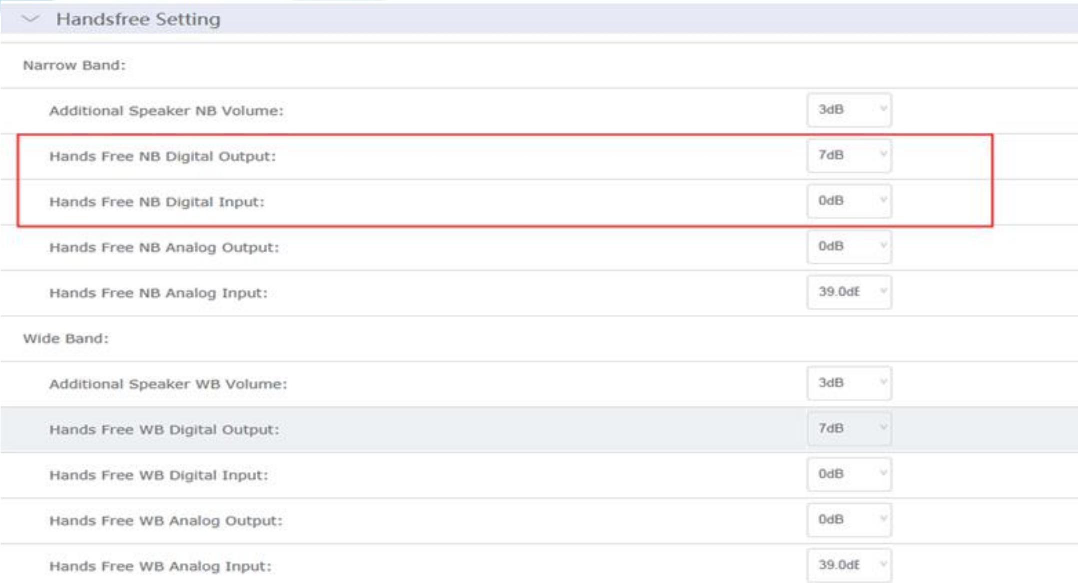
11.2 Configuring Speakerphone (Hands Free) Volume

The speakerphone (Hands Free) volume can be configured using the Web or Configuration File, as described below.

To configure speakerphone (Hands Free) volume:

1. Access the 'Gain Settings' page (Configuration tab >Voice Over IP menu >Gain Settings).

Hands Free Web Fields



2. Configure the Volume settings parameters and then click **Submit**.

Note: To adjust the handset volume, please only change the two parameters in the Red box.

"Handset NB Digital Output" is for handset receiver volume. The valid range is -21 to 15 dB.

"Handset NB Digital Input" is for handset microphone volume. The valid range is -31 to 31 dB.

12 Lines/SIP Accounts Settings

This section describes how to configure lines as described below.

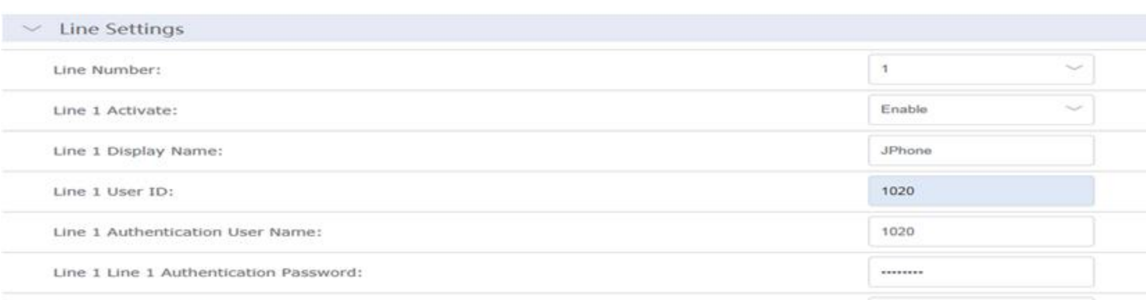
12.1 Configuring Lines using Web and Configuration File

Before you can make a call on your phone, you must configure a phone line. Lines can be configured using the Web as described below.

To configure lines:

Access the 'Line Settings' page (**Configuration** tab >**Voice Over IP** menu >**Line Settings**).

Line Setting Web Fields



Configure the Line settings parameters and then click **Submit**.

Part III Personal Settings

13 Phone Directory

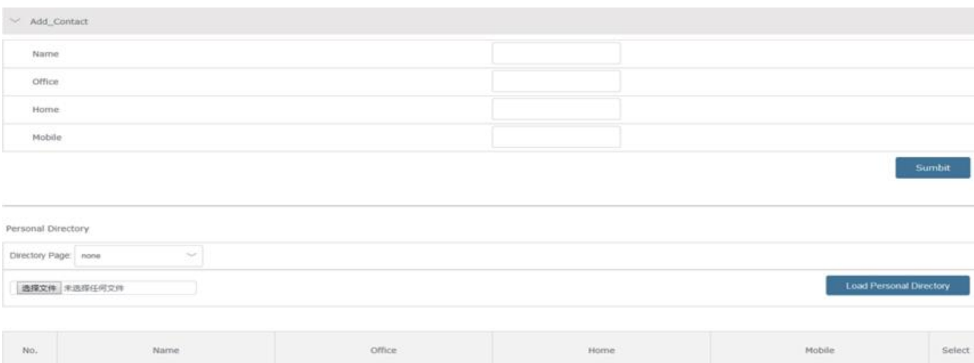
13.1 Local Phone Directory

You can add, edit, or delete directory contacts. A contact's address can be a telephone number, IP address, or domain name. You can also download or upload a personal directory file via the Web interface.

To add a contact to the phone's directory:

Access the 'Directory' page (Configuration tab >Personal Settings menu >Directory).

Add Contact Web Fields



Under the Add Contact group, define the contact:

In the 'Name' field, enter the name of the contact.

In the 'Office', 'Home' and/or 'Mobile' fields, enter the contact's telephone numbers. The contact's number can be defined with an IP address or domain name (e.g. <number>@<IP address or domain name>).

Click Submit; the contact appears in the Directory list at the bottom of the page.

To edit a contact:

If the contact does not appear in the displayed Directory list, then from the 'Directory Page' drop-down list, select the page in the directory that you want displayed.

In the Directory list, click the number that appears in the 'No.' column corresponding to the contact you want to edit; the contact's attributes appear in the Edit Phone group above.

Edit the contact as required, and then click Submit; the contact's new attributes are updated in the Directory list.

To delete a contact:

In the Directory list, mark the 'Select' check box corresponding to the contact you want to delete.

Click Delete. (To delete all contacts, click the Delete All button.)

14 Speed Dial on Function Keys

Speed Dial function is for some of the models which comes with speed dial buttons or keys.

14.1 Adding a Speed Dial

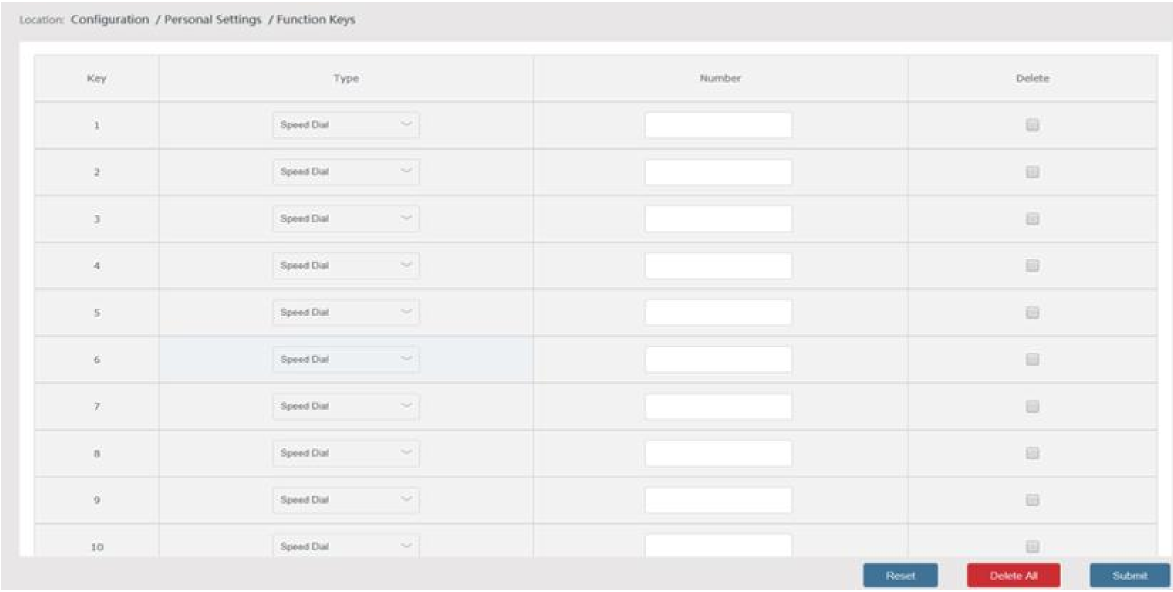
Adding a Speed Dial can be configured using the Web as described below.

To add a speed dial for function keys:

Access the 'Speed Dial' page (Configuration tab >Personal Settings menu >Function Keys).

Note: This function is only for speed dial type phone.

Function Keys Web Fields



In the 'Number' field corresponding to the phone's Speed Dial key (in the 'Button' column), enter the speed dial number to which you want to assign the Speed Dial key, then Click **Submit**. The Speed Dial key are related to the speed dial buttons on the speed dial type phones.

14.2 Cycle Dialing

In the 'Number' field you can assign Max of 4 numbers on one speed dial button, and with a separator " | " between each number, then Click **Submit**.

According to the above picture setting, when you press the speed dial button 2, the phone dials to 1002; and it dials to 1003 if no one answers on 1002; either it dials to 1004 if no one answers on 1003; at last it dials to 1005 if no one answers on 1004.

14.3 Deleting a Speed Dial on Function Key

To delete speed dials, do one of the following:

Deleting selected speed dial entries: select the 'Delete' check boxes corresponding to the speed dials that you want to delete, and then click **Submit**.

All speed dials: Click **Delete All**, and then at the prompt, click **OK**.

To clear (unselect) all selected 'Delete' check boxes: click **Reset**.

Part IV Maintenance

15 Changing User Login Credentials

User Login Credentials can be changed using the Web as described below.

To change the login username and password:

Access the 'System Authorization' page (**Management** tab >**Administration** menu >**Users**).

System Authorization Web Fields

- In the ‘Username’ field, enter a user name.
- In the ‘Password’ field, enter a new password, and then in the ’Confirm Password’ field, re-enter this new password.
- Click Submit; a confirmation box appears.
- Click OK.

16 Restarting Phones

You can use the Web interface to restart your phone.

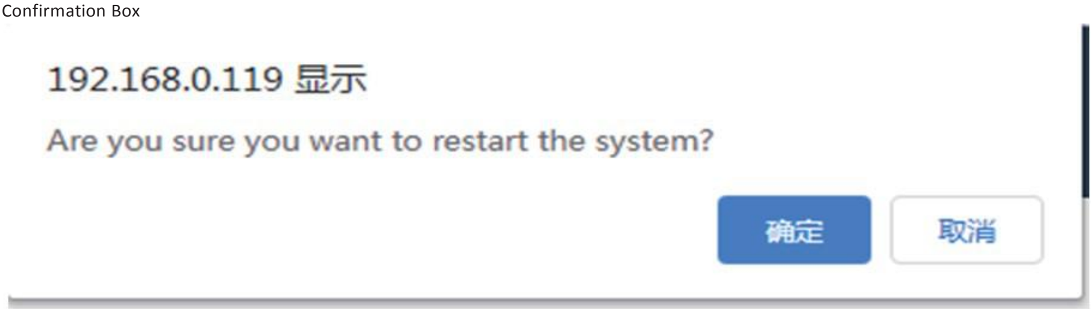
To restart the phone:

Access the ‘Restart System’ page (**Management** tab >**Administration** menu >**Restart System**).

Restart System Web Fields

Click the **Restart** button; a confirmation box appears prompting you to confirm.

Confirmation Box



Click **OK**.

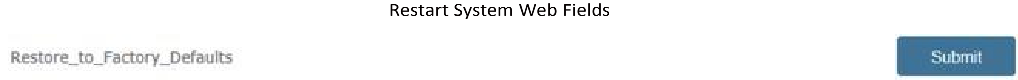
17 Restoring Phone Defaults

You can restore all your phone’s settings to factory default settings using the Web, as described in the procedure below.

To restore the phone to factory defaults:

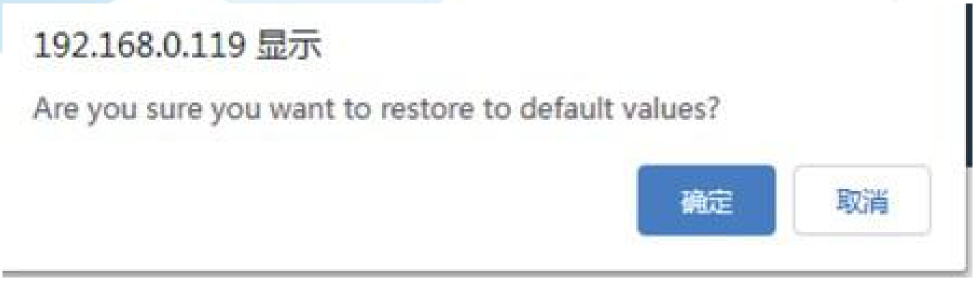
Access the ‘Restore Defaults’ page (**Management** tab >**Administration** menu >**Restore Defaults**).

Restart System Web Fields



Click the **Submit** button; a confirmation box appears prompting you to confirm.

Submit Confirmation Box



Click **OK**.

18 Change the language

You can use the Web to change the language of the phone, as described below. (Only Chinese and English)

To change the phone language

Access the ‘Restore Defaults’ page (Management tab >Administration menu >Restore Defaults).

Restart System Web Fields

Part V Status and Monitoring

19 Viewing Network Status

This section describes how to view the network status using the Web interface.

19.1 LAN Status

The following describes how to view LAN status information.

To view LAN status information:

Access the ‘Network Status’ page (Status & Diagnostics tab >System Status menu >Network Status).

LAN Information Web Fields

LAN Information

IP_Type:	DHCP Client
IP Address:	192.168.0.119
Subnet Mask:	255.255.255.0
Default Gateway_Address:	192.168.0.1
Primary DNS:	192.168.0.1
Secondary DNS:	0.0.0.0
MAC Address	4A:26:52:D8:16:30

19. 2 Port Status

The following describes how to view the Port Mode status.

To view Port Mode status:

Access the 'Network Status' page (Status & Diagnostics tab >System Status menu >Network Status).

Port Mode Status Web Fields

Port_Mode_Status

Attribute	LAN Port	PC Port
Link State:	Up	Down
Negotiation:	Automatic	Automatic
Speed:	100Mbps	N/A
Duplex:	Full	N/A
VLAN Activate:	Disable	Disable
VLAN Id:	N/A	N/A
VLAN Priority:	N/A	N/A

20 Viewing VoIP Status

The following describes how to view VoIP status using the Web interface.

20. 1 Phone Status

This section describes how to view the phone status.

To view the phone status:

Access the 'Network Status' page (Status & Diagnostics tab >System Status menu >VoIP Status).

Restart System Web Fields

Phone Status	
Hook State	On Hook
Audio Device	Ringer

20. 2 Line Status

This section describes how to view the line status.

To view the line status:

Access the 'Network Status' page (Status & Diagnostics tab >System Status menu >VoIP Status).

Line Status Web Fields

Line Status	
Line Number	Line 1
SIP Registration - Registrar 1	Registered
SIP Registration Server	192.168.0.180
DnD	Off
Mute	Off
Forward State	Disabled
Forward Destination	N/A

20.3 Current Call Information

The Web interface displays call information of a currently established call.

To view the line information:

Access the ‘Network Status’ page (Status & Diagnostics tab >System Status menu >Network Status).

Current Call Information Web Fields

Line 1 Call Information	
Call Number	Call 1
CallState	Dialing
Origin	Unknown
Remote Number	-
RemoteID	-
Duration	-
Codec	N/A
Packets Sent	0
Packets Received	0
Bytes Sent	0
Bytes Received	0
Packets Lost	0
Fraction Lost	0.0%
Jitter	0

21 Viewing Call History

You can view a list of received calls, missed calls, and dialed numbers. The call duration of the received and dialed calls is also displayed.

To view call history log:

Access the ‘Call History’ page (Status & Diagnostics tab >History menu >Call History).

Call History Web Fields

Type: Received Calls Page: 1

No.	Name	Number	Time	Duration	Delete
1		2022 Dial	02/12/2019 Monday 08:58:49	00:00:08	

Type: Dialed Calls Page: 1

No.	Name	Number	Time	Duration	Delete
1		1019 Dial	02/12/2019 Monday 08:59:39	00:00:00	

Type: Missed Calls Page: 1

No.	Name	Number	Time	Delete
1		2022 Dial	02/12/2019 Monday 08:58:29	

From the ‘Type’ drop-down list, select the type of call history (i.e., missed calls, received calls, and dialed numbers) that you want to view; the table lists the call history according to the chosen call history type.

You can delete a logged call history entry, by selecting the ‘Delete’ check box corresponding to the entry that you want to delete, and then clicking the Delete button.

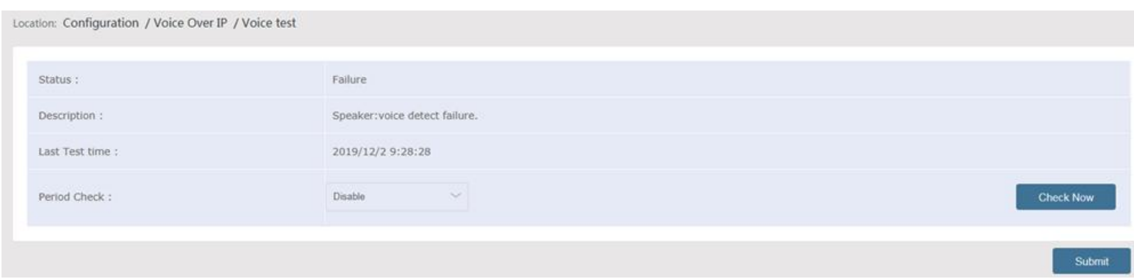
22 Voice test

You can check the phone's MIC and speakers on this page. Detecting whether the speaker and MIC work properly by making a sound

You can see the Status, Description and Last Test time in this web. When you press the Voice check button, the Voice test will start.
You can enable the Period Check or enable the Period Check. If you enable the Period Check, you can set the Check Period.
You can set hourly, Daily, weekly or on Power up only.

Voice test success

Voice test failure



Part VI Appendices

A SIP Support (RFC, Headers)

The following is a list of supported SIP RFCs and Methods that you can use to create for the IP Phone.

Supported IETF RFC’ s

RFC Number	RFC Title
RFC 2327	SDP
RFC 2617	HTTP Authentication: Basic and Digest Access Authentication
RFC 2782	A DNS RR for specifying the location of services
RFC 2833	Telephone event
RFC 3261	SIP
RFC 3262	Reliability of Provisional Responses in SIP
RFC 3263	Locating SIP Servers
RFC 3264	Offer/Answer Model
RFC 3265	(SIP)-Specific Event Notification
RFC 3310	Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)
RFC 3311 (Partially Supported)	UPDATE Method
RFC 3326 (Partially Supported)	Reason header
RFC 3389	RTP Payload for Comfort Noise
RFC 3515	Refer Method
RFC 3605	RTCP attribute in SDP
RFC 3611	RTP Control Protocol Extended Reports (RTCP XR)
RFC 3665	SIP Basic Call Flow Examples
RFC 3711	The Secure Real-time Transport Protocol (SRTP)
RFC 3725	Third Party Call Control
RFC 3842	MWI
RFC 3891	"Replaces" Header
RFC 3892 (Sections 2.1-2.3 and 3 are supported)	The SIP Referred-By Mechanism
RFC 3960 (Partially Supported)	Early Media and Ringing Tone Generation in SIP (partial compliance)
RFC 3966	The tel URI for Telephone Numbers
RFC 4028 (Partially Supported)	Session Timers in the Session Initiation Protocol
RFC 4240	Basic Network Media Services with SIP – NetAnn
draft-ietf-sip-privacy-04.txt (Partially Supported)	SIP Extensions for Network-Asserted Caller Identity using
draft-ietf-sipping-cc-transfer-05	Remote-Party-ID header Call Transfer
draft-ietf-sipping-realtimefax-01	SIP Support for Real-time Fax: Call Flow Examples
draft-choudhuri-sip-info-digit-00	SIP INFO method for DTMF digit transport and collection
draft-mahy-sipping-signaled-digits-01	Signaled Telephony Events in the Session Initiation Protocol

Note: The following SIP features are not supported:

- Preconditions (RFC 3312)
- SDP - Simple Capability Declaration (RFC 3407)
- S/MIME
- Outbound, Managing Client-Initiated Connections (RFC 5626)
- SNMP SIP MIB (RFC 4780)
- SIP Compression – RFC 5049 (SigComp)
- ICE (RFC 5245)
- Connected Identity (RFC 4474)

A.1 SIP Methods

The device supports the following SIP Methods:

Supported SIP Methods

Method	Supported	Comments
INVITE	Yes	
ACK	Yes	
BYE	Yes	
CANCEL	Yes	
REGISTER	Yes	Send only
REFER	Yes	Inside and outside of a dialog
NOTIFY	Yes	
INFO	Yes	
OPTIONS	Yes	
PRACK	Yes	
PUBLISH	Yes	Send only
SUBSCRIBE	Yes	

BIP Phone Specifications

Feature		Details
VoIP Signaling Protocols	★	SIP: RFC 3261, RFC 2327 (SDP)
Data Protocols	★	IPv4, TCP, UDP, ICMP, ARP, DNS and DNS SRV for SIP Signaling
	★	SIP over TLS (SIPS)
	★	802.1p/Q for Traffic Priority and QoS
	★	VLAN Discovery Mechanism (CDP, LLDP)
	★	ToS (Type of Service) field, indicating desired QoS DHCP Client
	★	NTP Client
Media Processing	★	Voice Coders: G.711, G.723.1, G.729A/B, G.722.
	★	Acoustic Echo Cancelation: G.168-2004 compliant, 64-msec tail length
	★	Adaptive Jitter Buffer 300 msec
	★	Voice Activity Detection
	★	Comfort Noise Generation
	★	Packet Lost Concealment
	★	RTP/RTCP Packetization (RFC 3550, RFC 3551), SRTP (RFC3711)
	★	DTMF Relay (RFC 2833)
Telephony Features	★	Call Transfer
	★	3-Way Conferencing (with local mixing)
	★	Redial
	★	Caller ID Notification
	★	Secondary Dial Tone
	★	Call Logs: Missed/Received Calls and Dialed Numbers
	★	Speed Dial
	★	Dial Plan
	★	URL Dialing
	★	Redundant SIP Proxy Mechanism
	★	Remote Conferencing (RFC 4240)

Feature		Details
Hardwares	★	LCD screen: 4 * 16 characters
	★	Connectors interfaces:
	★	2 x RJ-45 ports (10/100BaseT Ethernet) for WAN and LAN
	★	Mounting:
	★	Wall mounting or Flush mounting
	★	Power:
	★	DC jack adapter 12V
	★	Power supply AC 100 ~ 240V
	★	PoE: IEEE802.3af (optional)

Warranty Replacement

1. J&R offers free repair for the following products within 1 year, but additional warranties are available at additional cost:

Scope: ① Hardware damage during shipment.
② Hardware damage under normal using according to standard operation criterion of J&R.
③ Other defects authorized by J&R.

J&R's responsibility for product defects is limited to repair or replacements service as determined by J&R in it sole discretion, J&R is not responsible for direct, special, incidental or consequential damages resulting from any breach of warranty or condition.

2. Operation flow of repair

a. Apply for a valid RMA number from J&R.
b. J&R will check whether the product qualifies for replacement and will respond within one working day via fax or e-mail.
c. Purchaser should send the defective products with original packaging to J&R after obtaining the valid RMA number.
d. Within 1 week days of receipt of the returned products, J&R will send the replacement to the purchaser.

3. Relative explanation:

a. J&R recommends that you return products by reputable freight forwarders in order to avoid damage or loss during shipment.
b. J&R is NOT responsible for any damage or loss of products during shipment.
c. During the warranty replacement period, J&R reserves the ownership of the defective components after replacement.

4. Consider purchasing new equipment:

Customers are encouraged to consider purchase of new equipment under warranty, for both the improved reliability of new product, the possibility of a more cost effective solution, and the warranty coverage provided with the new equipment.

J&R TECHNOLOGY
Acorn Monitoring

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