

Setup of UT670 for connecting to SIP server

(KX-UT670)

No. 21-001

Dec 22 , 2011

Ver.1.0

Panasonic Corporation

Abstract about this document

This document describe about general setting procedure for connecting UT670 to SIP server.

Revision history

Date	Version	Revision	Firmware version
Dec. 22, 2011	Ver. 1.0	Initial Release	All versions

Step1 : Access the Web user interface

1. Confirm the IP address of KX-UT670.

1-1 : [Menu] (Upper left of sheet key) →[Settings] → [About phone] → [Status]

1-2 : Confirm the IP address displayed on LCD.

2. Embedded Web.

2-1 : [Menu] (Upper left of sheet key) →[Settings] → [Network]
→[Embedded web] →select [On]

3. Access the Web user interface.

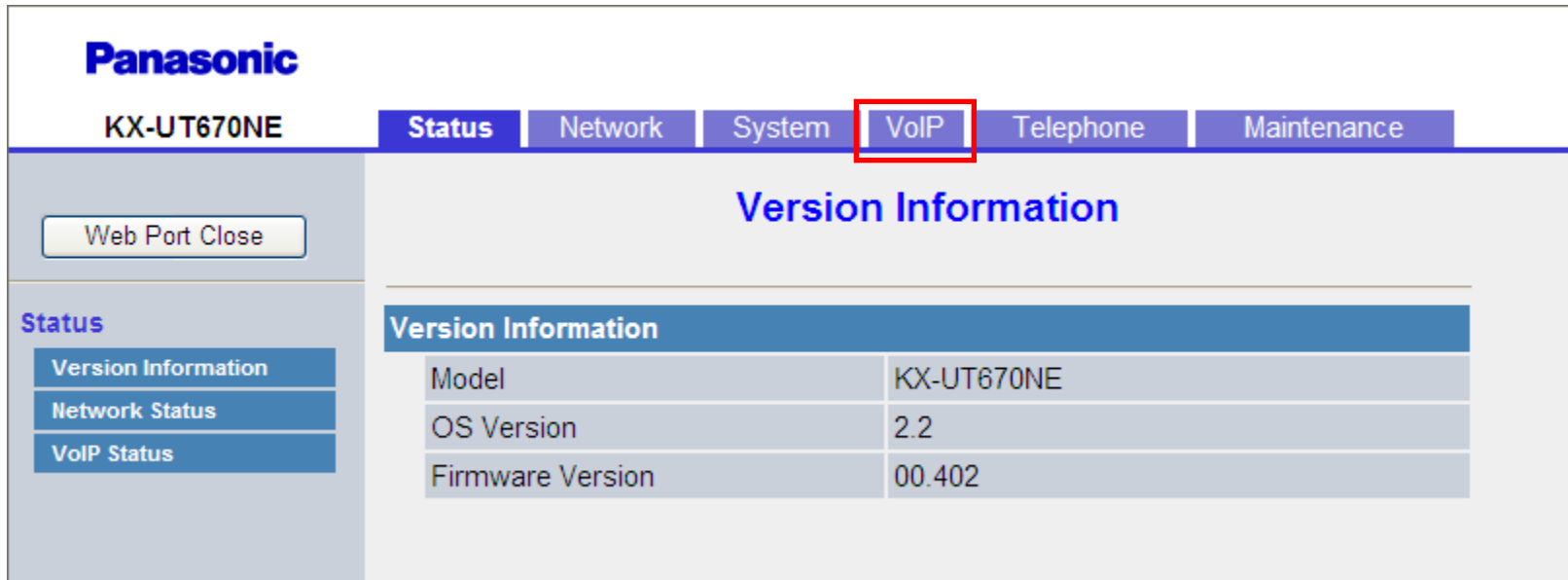
3-1 : Open your Web browser.

3-2 : Enter your KX-UT670 URL to Web browser
(<http://your KX-UT670 IP address>) .

3-3 : Enter user name (**admin**) and password (**adminpass**) and click “OK”.

3-4 : The Web user interface window is displayed.
Configure the settings for the unit as desired.

Step2 : Select VoIP tag



The screenshot displays the Panasonic KX-UT670NE web interface. At the top left, the Panasonic logo and model number "KX-UT670NE" are visible. A navigation bar contains several tabs: "Status", "Network", "System", "VoIP", "Telephone", and "Maintenance". The "VoIP" tab is highlighted with a red rectangular box. Below the navigation bar, the "Version Information" section is active, showing a table with the following data:

Version Information	
Model	KX-UT670NE
OS Version	2.2
Firmware Version	00.402

On the left side of the interface, there is a "Status" menu with options for "Version Information", "Network Status", and "VoIP Status". A "Web Port Close" button is also present in the top left area.

1. Click "VoIP" tag.

Step3 : Select Line 1 tag

The screenshot shows the Panasonic KX-UT670NE web interface. At the top, the model name 'KX-UT670NE' is displayed next to a 'Web Port Close' button. Below this, a navigation menu includes 'Status', 'Network', 'System', 'VoIP', 'Telephone', and 'Maintenance'. The 'VoIP' tab is active, showing 'SIP Settings'. In the left sidebar, under 'VoIP', there are two sections: 'SIP Settings' and 'VoIP Settings'. Under 'SIP Settings', there are six line tags: '- Line 1', '- Line 2', '- Line 3', '- Line 4', '- Line 5', and '- Line 6'. The '- Line 1' tag is highlighted with a red border. The main content area shows the 'SIP Setting' for Line 1, with the 'SIP User Agent' field containing the text 'Panasonic_{MODEL}/{fwwer}({mac})'. A red warning message states: 'The phone reboots automatically if you change the settings on this screen.' At the bottom of the main content area, there are 'Save' and 'Cancel' buttons.

1. Click “Line 1” tag of SIP Settings.

Step4 : SIP server address settings

SIP Settings [Line 1]	
Phone Number	
Phone Number	<input type="text" value="601"/>
SIP URI	<input type="text"/>
SIP Server	
Registrar Server Address	<input type="text" value="192.168.1.54"/>
Registrar Server Port	<input type="text" value="5060"/> [1-65535]
Proxy Server Address	<input type="text" value="192.168.1.54"/>
Proxy Server Port	<input type="text" value="5060"/> [1-65535]
Presence Server Address	<input type="text"/>
Presence Server Port	<input type="text" value="5060"/> [1-65535]
Outbound Proxy Server	
Outbound Proxy Server Address	<input type="text"/>

1. Enter the **Phone Number** consisting of 0–9, *, and #.
2. Enter **your SIP server address** to
Resister Server Address and Proxy Server Address.

Step5 : Enter the ID and Password

SIP Authentication	
Authentication ID	1. <input type="text" value="601"/>
Authentication Password	2. <input type="password" value="●●●"/>
DNS	
Enable DNS SRV lookup	<input checked="" type="radio"/> Yes <input type="radio"/> No
SRV lookup Prefix for UDP	<input type="text" value="_sip_udp."/>
SRV lookup Prefix for TCP	<input type="text" value="_sip_tcp."/>
Transport Protocol for SIP	
Transport Protocol	<input checked="" type="radio"/> UDP <input type="radio"/> TCP
Timer Settings	
T1 Timer	<input type="text" value="500"/> <input type="button" value="v"/> milliseconds
T2 Timer	<input type="text" value="4"/> <input type="button" value="v"/> seconds

1. Enter your Authentication ID.

2. Enter your Authentication Password.

Step6 : Save the settings

Quality of Service (QoS)	
SIP Packet QoS (DSCP)	<input type="text" value="0"/> [0-63]
SIP extensions	
Supports 100rel (RFC 3262)	<input type="radio"/> Yes <input checked="" type="radio"/> No
Supports Session Timer (RFC 4028)	<input type="text" value="0"/> seconds [60-65535, 0: Disable]
NAT Identity	
Keep Alive Interval	<input type="text" value="0"/> seconds [10-300, 0: Disable]
Supports Rport (RFC 3581)	<input type="radio"/> Yes <input checked="" type="radio"/> No
Security	
Enable SSAF (SIP Source Address Filter)	<input type="radio"/> Yes <input checked="" type="radio"/> No

1. Click the “Save”.
2. After save, confirm the Status. If Registration is successful, Phone service is “OK”.

[Menu] (Upper left of sheet key) → [Settings] → [About phone] → [Status]

- The setting is finished -