

**VOIP
User's Manual**

VOIP Solution

What is VOIP? Internet Voice, also known as Voice over Internet Protocol (VOIP), is a technology that allows you to make telephone calls using a broadband Internet connection instead of a regular (or analog) phone line. Some services using VOIP may only allow you to call other people using the same service, but others may allow you to call anyone who has a telephone number - including local, long distance, mobile, and international numbers. Also, while some services only work over your computer or a special VOIP phone, other services allow you to use a traditional phone through an adaptor.

Foreword

VOIP allows you to make telephone calls using a computer network, over a data network like the Internet. VOIP converts the voice signal from your telephone into a digital signal that travels over the internet then converts it back at the other end so you can speak to anyone with a regular phone number. When placing a VOIP call using a phone with an adapter, you'll hear a dial tone and dial just as you always have. Some gateways at the recipient side can convert phonecall back into voice signal, you need go through a VOIP to PBX gateway if you'd like the phone call from VOIP to regular phone number.

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1. Unpacking Information

Check List

Carefully unpack the package and check its contents against the checklist.

Package Contents

- VOIP 2FXS with Router Adapter
- Diskette User Manual
- AC to DC Power Adapter

Please inform your dealer immediately for any missing, or damaged parts. If possible, retain the carton, including the original packing materials, Use them to repack the unit in case there is a need to return for repair.

2. Installation

Hardware Installation

This chapter describes how to install the VOIP and establishes network connections. You may install the VOIP on any level surface (e.g, a table or shelf). However, please take note of the following minimum site requirements before you begin.

Pre-installation Requirements

Before you start actual hardware installation, make sure you can provide the right operating environment, including power requirements, sufficient physical space, and proximity to other network devices that are to be connected. Verify the following installation requirement:

- Power requirements: DC12V/1A or above.
- The VOIP should be located in a cool dry place, with at least 10cm/4in of space at the front and back for ventilation.
- Place the VOIP out of direct sunlight, and away from heat sources or areas with a high amount of electromagnetic interference.
- Check if network cables and connectors needed for installation are available

General Rules

Before making any connections to the VOIP, note the following rules:

- Ethernet Port (RJ-45)
All network connections to the Modem Ethernet port must be made using Category 5 UTP for 100Mbps; Category 3,4 UTP for 10Mbps
No more than 100 meters of cabling may be use between the MUX or HUB and an end node.
- Phone Port (RJ-11)
All Phone set connections to the RJ-11Port made using 24~28 Gauge phone wiring.

Connecting the VOIP

The VOIP has two ETHERNET port which support connection to Ethernet operation. The devices attached to these ports must support auto-negotiation or 10Base-T OR 100Base-TX unless they will always operate at half duplex.

Using WAN port connect to devices such as XDSL modem or router. Using LAN connect to devise such as NIC or switch.

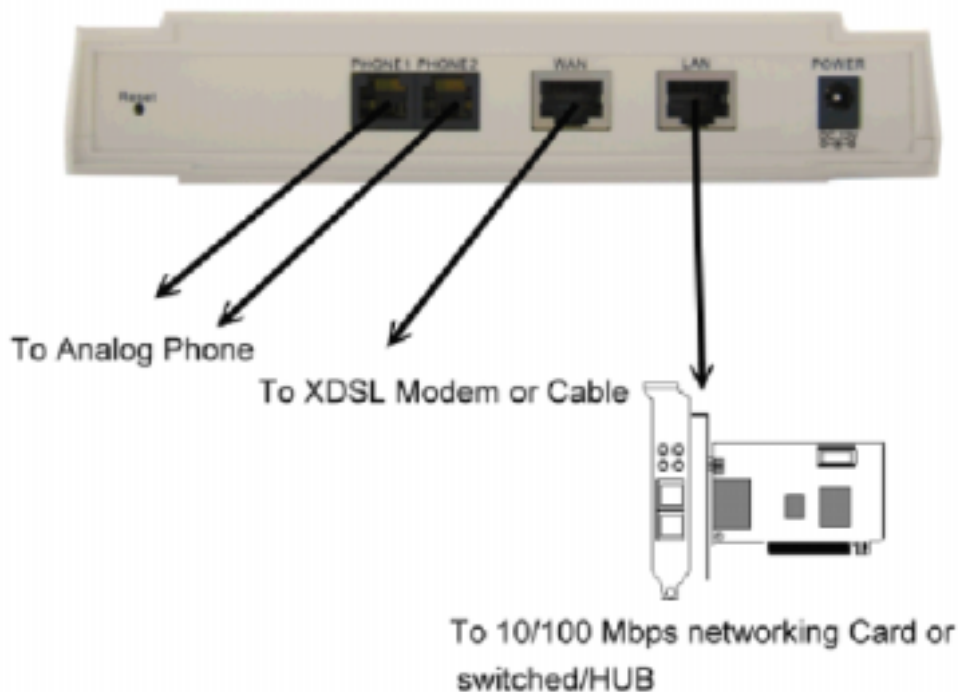
The VOIP has two RJ-11 ports, which support connection to two of analog phone set.

Using Phone 1or Phone 2 port to connect to analog phone set.

Connecting the RJ-11/RJ-45 Ports

1. The VOIP RJ-11 ports support 2 of traditional analog phone set with IP voice transmissions. Please see (Figure 1.0).
2. The VOIP RJ-45 ports support 10/100Mbps auto negotiation and auto MDIX functions, one WAN Port for connecting to XDSL Modem or Cable modem, one LAN port for connecting to PC networking Card or switched/HUB. Please see (Figure 1.0).

Figure 1.0 VOIP use as adapter to connect RJ-11 and the LAN card inside the PC



3. Do not plug a RJ-11 phone jack connector into the Ethernet port (RJ-45 port). This may damage the modem. Instead, use only twisted-pair cables with RJ-45 connectors that conform to FCC standard.

3 Hardware Description

This section describes the important parts of the VOIP. It features the front indicators and rear connectors.

3.1 Front Indicators

The following figure shows the front panel.



Figure Chapter 2.2 Front Indicators

Six LED indicators.

At a quick glance of the front panel, it will be easy to tell if the modem has power, signal from its Ethernet RJ-45 port or there is phone line signal RJ-11 port

Front Indicators

LED Description and Operation

The Modem has three LED indicators.

LEDs	Status	Descriptions
Power Good LED	Steady Green	It will light up (ON) to show that the product power is good, and system reset OK.
SYS LED	Steady Green Flashing (booting)	It will light up after flashing twice, when System boot OK.
Ethernet (WAN/LAN LED)	Steady Green Flashing (LINK/ACT)	Each RJ45 station port on the Ethernet is assigned a LED light for monitoring port "Good Linkage". LED is normally OFF after the power on operation, but will light up steadily to show good linkage. And Flashing to show data transmission.
Phone set dialing status (P1/P2 LED)	Steady Green Flashing (Ringing)	RJ11 station port on the P1/P2 is assigned a LED light for dialing function OK, when you pick up handset If LED flashing indicate getting a ringing

Rear Panel

The following figure shows the rear connectors
Figure Chapter 2.3 Rear Connectors



VOIP Rear Side Connectors

Connectors	Description	Type
Rest button	For reboot system	push switch
Phone1/2	For connecting to the telephone	RJ-11
WAN	For connecting to XDSL/Cable modem	RJ-45
LAN	For connecting to a PC networking card or switched/HUB	RJ-45
FG	For connecting to AC ground or ignored Terminator Power	
	For connecting to DC12V/1A or above power adapter	2.0m/m plug

Power On

1. Check if the modem is properly connected
2. Verify the power LED is steadily on

Web Login information:

account : admin

Password : voip

Default Web IP address: 192.168.1.1

Home

VoIP v01(2FXS)

Home WAN LAN SIP CODECS System Download Configuration Reset Logout

Home

Welcome to the CS6220 VoIP v01(2FXS) download and configuration utility.
Select from the configuration options in the menu on the top.

System Information

System Uptime:	0 days, 0h 1m 33s
NTP time:	NTP Time Not Available
LAN IP Address:	192.168.1.1 (Static)
MAC Address:	00:05:6e:00:1d:89
Serial Number:	
Security:	Password installed
Application Code Version:	VR 4.1Beta2 (MSCS (20001) Build-Date: Oct 6 2005
Downloader Code Version:	US 1.0Beta2 (MSCS VR40)

System Uptime: specifies the amount of time, which the system has been up. This time is reset every time the system is reset.

LAN IP Address: indicates the IP Address of your LAN.

MAC address: MAC address is the address of your MAC.

Security: for your password, which is configured in the “System” section.

Application Code Version: tells the version of the application code which you are using.

Download Code Version: tells the version of the download code which you are using.

WAN

WAN status

Home **WAN** LAN SIP CODECS System Download Configuration Reset Logout

WAN Status

WAN Settings
PPPoE
MAC Spoofing

WAN Status

Interface Status

Enabled:	Yes
Service:	Routed
Protocol:	Ethernet
Interface Status:	Up
Link Status:	10M bps, Full Duplex

Network Settings

Dynamic IP Assignment:	NO
IP Address:	192.168.16.250
MAC Address:	00:05:6e:00:1d:69
Subnet Mask:	255.255.255.0
Default Gateway:	192.168.16.1
DNS Address:	80.0.0.1
DNS Address 2:	
Domain Name:	
Priority Tag:	Not set
Broadcast limit:	100% (of downstream bit rate)
Multicast limit:	100% (of downstream bit rate)

Interface Status: these are the details of your interface's status.

Enabled: "Yes", lets you know that your interface is enabled and ready to be used.

Service: either "Routed or Bridged", tell you the level of your interface's connection.

Protocol: refers to how you are transmitting data. (i.e. Ethernet)

Interface Status: either "Up" or "Down".

Under Network Settings: these are the details of your network settings.

Dynamic IP Assignment: "Yes" or "No", depending on whether or not you are using a dynamic IP.

IP address: your specified IP.

MAC address: Your specified MAC address.

Subnet Mask: indicates the IP address of your mask.

Default Gateway: is the IP address of the gateway. The gateway IP could be retrieved from DHCP offer in DHCP mode, or be set up manually in fixed IP mode.

DNS address: refers to the address of your dynamic name server, if applicable.

Priority Tag: Priority Tag value encoded in the Ethernet header in outgoing packets.

WAN Configuration

- 1. Device Operating Mode:** you choose either “Router” or “Bridged” depending on your operation.
- You will check either “**Obtain WAN configuration dynamically**” or “**Specify fixed WAN configuration**”.

When you choose “**Obtain WAN configuration dynamically**”, the information is detected automatically through DHCP.

If you choose “**Specify fixed WAN configuration**”, you are required to enter the IP address, IP of the Sub mask, IP of the Gateway, and IP of the DNS Server, if applicable.

3. Multicast Limits:

Broadcast Limit: the value specifies the maximum limit on the percentage of broadcast packets which will be bridged to the destination

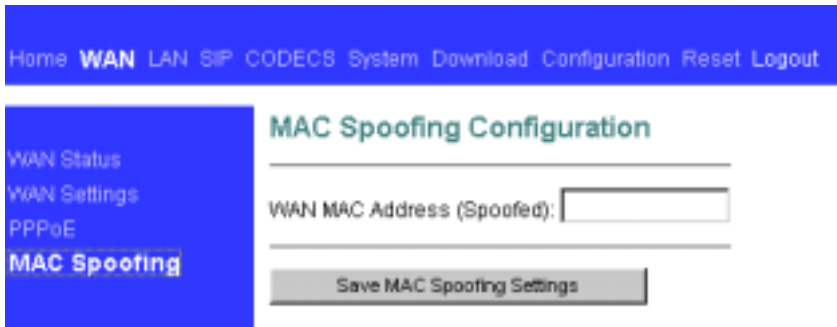
interface (as a percentage of the source side bandwidth)

Multicast Limit: the value specifies the maximum limit on the percentage of multicast packets which will be bridged to the destination interface (as a percentage of the source side bandwidth)

WAN PPPoE Configuration

1. **Enable PPPoE:** “Yes” or “No”, to enable/disable PPPoE
2. Under “**Authentication**”, you enter the username and password given by your ISP.
3. **Settings:**
 - Echo Timeout:** the duration between PPP echo requests sending to server.
 - Echo Count:** the number of unanswered PPP echo requests before PPP connection is closed.

MAC Spoofing Configuration

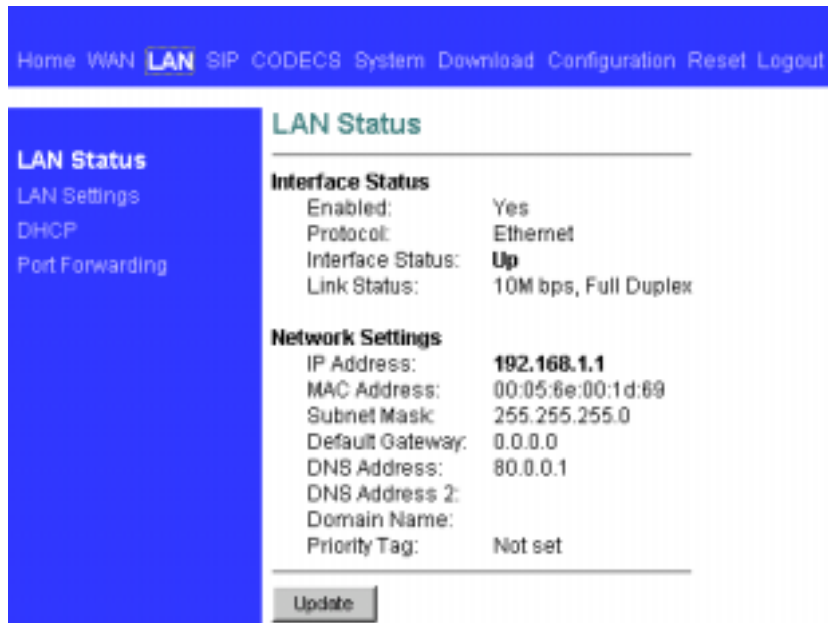


WAN MAC Address (Spoofed):

Only available when devices under the router mode. The spoofed MAC address to be used by the device's WAN interfaces, the Ethernet address of the outgoing packets from the WAN interface would be replaced with this address. If blank, the WAN interfaces will use the value of MAC

LAN

LAN Status



This page shows current status of LAN interface, including the IP address and other network setting the interface is currently using.

LAN Configuration

Home WAN **LAN** SIP CODECS System Download Configuration Reset Logout

LAN Status
LAN Settings
 DHCP
 Port Forwarding

LAN Configuration

Network Settings

IP Address:

Subnet Mask:

Multicast Limits

Broadcast limit: % (of Ethernet connection bitrate)

Multicast limit: % (of Ethernet connection bitrate)

PHY Speed mode:

1. Under “Network Settings”, you enter the IP address and subnet mask of your network.

2. **Multicast Limits:**

Broadcast Limit: the value specifies the maximum limit on the percentage of broadcast packets, which will be bridged to the destination interface (as a percentage of the source side bandwidth)

Multicast Limit: the value specifies the maximum limit on the percentage of multicast packets, which will be bridged to the destination interface (as a percentage of the source side bandwidth)

DHCP Server Configuration

Home WAN LAN SIP CODECS System Download Configuration Reset Logout

LAN Status
LAN Settings
DHCP
Port Forwarding

DHCP Server Configuration

Server Settings

Enabled Disabled

Client IP Address Range: 192.168.1. -

Client Network Information

Domain Name:

DNS Server 1: 2:

Static Address Assignments

Identify Using	Host Identifier	Internal Address
<input type="text" value="Hostname"/>	<input type="text"/>	192.168.1. <input type="text"/>

These configuration parameters are for the device's internal DHCP server.

1. Server Setting: “Yes” or “No”, to enable/disable DHCP

Client IP Address Range: Minimum and Maximum limit on the DHCP IP address pool

2. Client Network Information

Domain Name: LAN domain name provided to DHCP clients during the OFFER process.

DNS Server: This statically assigned DNS server IP address will be provided to clients during the OFFER process.

3. Static Address Assignment

Up to eight static DHCP address assignments can be configured. To add a static IP assignment, enter the LAN device's **host name** (must be unique in the private network) and/or **MAC address**. Specify the **Internal address** to be assigned and press the "Add" button.

Port Forwarding Configuration

Home WAN **LAN** SIP CODECS System Download Configuration Reset Logout

LAN Status
LAN Settings
DHCP
Port Forwarding

Port Forwarding Configuration

Reserved Ports
The following ports have been reserved by the CPE, and may not be forwarded to the LAN
68, 5060-5070, 8000-8015, 7001-7005, 80, 23, 161, 1480-12000

Port Forwarding to LAN

Port Range	Protocol	Destination Address
<input type="text"/> - <input type="text"/>	<input type="text" value="Both"/>	192.168.1. <input type="text"/>

DeMilitarized Zone
If specified, packets which port are not listed above will be forwarded to this DMZ host
192.168.1.

1. Under “**Reserved Ports**”, specified are the ports, which cannot be forwarded to the LAN.

2. Under “**Port Forwarding to LAN**”, you enter the specifications, which you will be forwarding to the LAN, including **port range**, **protocol** (Both, TCP or UDP), and **destination IP address**.

Click on “**Save NAPT Settings**” to save your configurations.

SIP

SIP Configuration

Home WAN LAN **SIP** CODECS System Download Configuration Reset Logout

SIP Server Configuration

Primary Server Settings	Secondary Server Settings
(Current Server: 192.168.16.251 : 5060 ; Domain: 192.168.16.251)	(Current Server: : 0 ; Domain:)
* Address: <input type="text" value="192.168.16.251"/> (IP or FQDN)	* Address: <input type="text"/> (IP or FQDN)
* Port: <input type="text" value="5060"/>	* Port: <input type="text" value="5060"/>
Domain Name: <input type="text"/>	Domain Name: <input type="text"/>
<input type="checkbox"/> Send Registration Request with Expire Time <input type="text" value="3600"/>	<input type="checkbox"/> Send Registration Request with Expire Time <input type="text"/>
Outbound Proxy IP: <input type="text"/> (IP or FQDN)	Outbound Proxy IP: <input type="text"/> (IP or FQDN)
Outbound Proxy Port: <input type="text" value="5082"/>	Outbound Proxy Port: <input type="text" value="5082"/>

RTP Port Number Setting(5000~65535) -

NAT Traversal Settings

NONE
 UPnP Control Point
 STUN Server IP: (IP or FQDN) STUN Server Port:

Gateway Settings

Dial Plan:

use as a quick dial function * use as a quick dial function
 To enable # to be recognized as dial number To enable * to be recognized as dial number

* Leaving a setting blank will force the unit to use the information obtained via DHCP and/or DNS
 ** Leaving a setting blank will disable server redundancy function

1. Under “SIP Server Settings”, you enter the **server address, port, domain name, and expiration time unit**, if you choose to send registration request with an expiration time.

2. Gateway Settings

- Dial Plan: refer to appendix D of this guide
- **#use as a quick dial function:** If this box is checked, the dialed digits would be sent out when “#” key is pressed.
- **Enable # to be recognized as dial number:** allow “#” key to be appeared in the INVITE request URI
- **Enable # to be recognized as dial number:** allow “#” key to be appeared in the INVITE request URI

SIP Extensions

Home WAN LAN **SIP** CODECS System Download Configuration Reset Logout

Server
Extensions
 User 1
 User 2
 OOB Signaling
 ToS/DiffServ
 Tone
 Ring
 Service Code

SIP Extensions

- Support PRACK method with provisional response reliability
- Encode SIP URI with user parameter
- Session Timer use UPDATE method
- Call Hold using c=0.0.0.0 (RFC 2543) in SDP
- enable Global Number support (E.164)
- send NOTIFY for REFER request
- send Message Waiting Indicator (MWI) SUBSCRIBE command
- No Authorization Header in re-REGISTER
- Check existence of To Tag in INVITE 2xx response

SIP Timers

- Send INVITE with Timer header value: Seconds
- SIP Session Timer value: Seconds
- Conditional Call Forwarding Timer: Seconds

Inter Digit Timer: Seconds.

SIP T1 Timer: Milliseconds

SIP T2 Timer: Milliseconds

SIP T4 Timer: Milliseconds

This page allows specification of the SIP signaling stack behavior under certain scenarios.

If you wish for the SIP stack implement reliable transmission of provisional responses according to RFC 3262 (using the PRACK method), check the option “Support PRACK method with provisional responses reliability”.

If you wish for the SIP stack to include the user parameter “user = phone” in the SIP URI header(s), check the option “Encode SIP URI with user parameter”.

If you wish for the SIP stack to send INVITE message with “Timer” header field present, check the option “Send INVITE with Timer header value” and enter the Timer header value.

If you wish for the SIP stack to implement a session timer according to “draft-sip-session-timer”, select the option “SIP Session Timer value”, and enter the session time-out value.

Press “Save SIP Extension Setting” to save the new values.

RTP Telephone Event Configuration

Home WAN LAN **SIP** CODECS System Download Configuration Reset Logout

Server
Extensions
User 1
User 2
OOB Signalling
ToS/DiffServ
Tone
Ring
Service Code

RTP Telephone Event Configuration

Send DTMF Events

RFC2833 signalling using payload value:

Regenerate OOB DTMF tone

Save OOB Settings

This sub-page allows configuration of the out-of-band signaling options for SIP. Select whether OOB telephone event signaling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signaling. For additional information please refer RFC2833.

ToS/DiffServ

Home WAN LAN **SIP** CODECS System Download Configuration Reset Logout

Server
Extensions
User 1
User 2
OOB Signalling
ToS/DiffServ
Tone
Ring
Service Code

ToS/DiffServ

Call Signalling Packets: (2 Hex digit byte value)

RTP Packets: (2 Hex digit byte value)

Save ToS/DiffServ Settings

This sub-page is used to configure the Type-of-Service/Diffserv byte values which are to be used in the IP header of all transmitted SIP signaling packets and RTP packets. The ToS/DiffServ byte values are entered as two-digit hexadecimal values. If no special ToS/DiffServ value is to be used for a particular traffic type, enter “00” or leave the setting empty.

Press “**Save ToS/DiffServ Settings**” to save these new settings.

Tone

Home WAN LAN **SIP** CODECS System Download Configuration Reset Logout

Server
Extensions
User 1
User 2
OOB Signalling
ToS/DiffServ
Tone
Ring
Service Code

Tone Configuration

Dial Tone:

Recall Dial Tone:

Confirm Tone:

Ring Back Tone:

Busy Tone:

Reorder Tone:

Receiver-Off-Hook Tone:

Message-Waiting Indicator Tone:

Call-Waiting Indicator Tone:

Save Tone Settings

Dial Tone: The tone you hear when you pick up handset.

Recall Dial Tone: The tone when you hold caller and prepare to make another call.

Confirm Tone: The tone after you've set up some service, like DND (Do Not Disturb), Call Forwarding, etc.

Ring Back Tone: The audible ringing you hear before caller pick up and answer your call.

Busy Tone: The tone indicates the number you dialed is in busy now.

Recorder Tone: The tone you hear if you dial an invalid number or the call is not available.

Receiver-Off-Hook Tone: The tone to alert you to place the handset on-hook.

Message-Waiting-Indicator Tone: The tone to notify you to call for message box.

Call-Waiting-Indicator Tone: The tone to make you aware of the second incoming call while you are in conversations.

Ring

Home WAN LAN **SIP** CODECS System Download Configuration Reset Logout

Server
Extensions
User 1
User 2
OOB Signalling
ToSID#Serv
Tone
Ring
Service Code

Ring Configuration

Default Ring:

Call-Waiting
Reminder Ring:

Distinctive Ring Configuration

Distinct Ring 1:

Distinct Ring 2:

Distinct Ring 3:

Distinct Ring 4:

Distinct Ring 5:

Distinct Ring 6:

Distinct Ring 7:

Distinct Ring 8:

- **Ring Configuration:**

Default Ring: Default ring cadence when the phone rings.

Call-Waiting Reminder Ring: Ring cadence of Call-Waiting Reminder Ring

- **Distinctive Ring Configuration:**

Distinctive Ring 1-8: Ring cadences provided for distinctive function.

You may customize them according to the fixed format.

For example:

ON (500), OFF (500), R

Will cause 500 milliseconds ring on, then 500 milliseconds off, and repeat steadily.

Service Code

Home WAN LAN SIP CODECS System Download Configuration Reset Logout

Server
Extensions
User 1
User 2
OOB Signalling
ToS/Dial/Service
Tone
Ring
Service Code

Service Code Configuration

Conditional Call Forwarding:

Call Forwarding On:

Call Forwarding Off:

Do Not Disturb On:

Do Not Disturb Off:

Call Transfer:

Call Return:

Speed Dial:

- use *xx# or #xx# format, xx=01-99

Save Service Code Settings

Service Code Configuration:

Conditional Call Forwarding: * 70#

Call Forward On: * 72#

Call Forward Off: # 72#

Do Not Disturb On: * 74#

Do Not Disturb Off: #74#

Call Transfer: * 98#

Call Return: * 69#

Speed Dial: * 68

Note: We recommend a general user to change the services code values only if you have meet conflict with the setting of service provider.

CODEC

Audio/CODEC Configuration

Home WAN LAN SIP **CODECS** System Download Configuration Reset Logout

CODECS

Audio/CODEC Configuration

CODECS

Selected	Silence Suppression
<input checked="" type="checkbox"/> G711U	<input type="text" value="ON"/>
<input checked="" type="checkbox"/> G711A	<input type="text" value="ON"/>
<input type="checkbox"/> G723	<input type="text" value="ON"/>
<input type="checkbox"/> G726	<input type="text" value="ON"/>
<input type="checkbox"/> G729	<input type="text" value="ON"/>

Packetization

Jitter Buffer

Adaptive Jitter Buffer: (maximum playout delay in milliseconds)

Fixed Jitter Buffer: (fixed playout delay in milliseconds)

Automatically switch to Fixed Jitter Buffer upon fax/modem tone detection

1. CODECS: configure the silence suppression to your desired settings.

2. Packetization: configure the packet sending increments.

3. Jitter Buffer: configure the timing of the voice buffering.

Selection between adaptive or fixed jitter buffer. Default = ADAPTIVE

Set the adaptive jitter buffer maximum playout delay. Default = 100ms

or Fixed jitter buffer playout delay. Default = 40ms

Whether or not to automatically switch from an adaptive jitter buffer to a fixed jitter buffer upon VOIP tone detection

Click on “**Save CODEC Configuration**” to save the configurations made.

SYSTEM

Set Security Password

Home WAN LAN SIP CODECS **System** Download Configuration Reset Logout

Security
Timeout
Localization
Handset
SNMP
Service Access

Set Security Password

Password is currently installed

Account: admin

Old password:

New password:

Confirm new password:

Configure a password for the system.

Timeout

Home WAN LAN SIP CODECS **System** Download Configuration Reset Logout

Security
Timeout
Localization
Handset
SNMP
Service Access

Set Web System Timeout

HTTP Authentication Timeout: (Seconds)

In HTTP Authentication Timeout field, input timeout value you want. Then press “Change Time” button. After HTTP Authentication timeout value expired, it will redirect to password protected page.

Localization

The screenshot shows a web interface with a blue header bar containing navigation links: Home, WAN, LAN, SIP, CODECS, **System**, Download, Configuration, Reset, Logout. On the left is a blue sidebar menu with links: Security, Timeout, **Localization**, Handset, SNMP, Service Access. The main content area is titled "Localization" and contains the following fields:

- Country: A dropdown menu currently showing "United States".
- NTP Server: An empty text input field.
- Time Zone: A dropdown menu currently showing "(GMT-12:00) Eniwetok, Kwajalein".
- Adjust clock for daylight savings: An unchecked checkbox.

At the bottom of the form is a "Save Localization Settings" button.

Choose the correct country for a proper impedance match, as well as the NTP Server, and Time Zone. Check the “**Adjust clock for daylight savings**”, when applicable.

Click on “**Save Localization Settings**”, to save your configurations.

Handset

The screenshot shows a web interface with a blue header bar containing navigation links: Home, WAN, LAN, SIP, CODECS, **System**, Download, Configuration, Reset, Logout. On the left is a blue sidebar menu with links: Security, Timeout, Localization, **Handset**, SNMP, Service Access. The main content area is titled "Media Hub Handset Configuration" and contains the following fields:

- Control Timer Values: A section header.
- Hook Flash Timer Min: A text input field followed by "Milliseconds".
- Hook Flash Timer Max: A text input field followed by "Milliseconds".

Below the input fields is a note: ***Please enter a multiple of 10.(ex:10,20,30...)**

At the bottom of the form is a "Save Handset Settings" button.

This page allows user to configure the hook time interval.

SNMP Configuration

Home WAN LAN SIP CODECS **System** Download Configuration Reset Logout

Security
Timeout
Localization
Handset
SNMP
Service Access

SNMP Configuration

SNMP Trap Configuration

IP address: Trap Community:

SNMP Community Configuration

Read Community: Write Community:

SNMP System Configuration

System Description:

System Objectid:

1. SNMP Trap Configuration

IP address: Trap host IP address

Trap Community: The community name used by the SNMP manager to verify traps. The default value is “public”

2. SNMP Community Configuration

Read Community: The community name used by the SNMP manager when reading SNMP data items from a client MIB. The default value is “public”

Write Community: The community name used by the SNMP manager when setting SNMP data items in a client’s MIB. The default value is “public”

3. SNMP System Configuration

System Description: Description of the unit (e.g. “John’s phone”)

System Object Id: A vendor’s enterprise ID

Service Access Configuration

	LAN	WAN
HTTP (Web access):	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
SNMP:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
TELNET:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

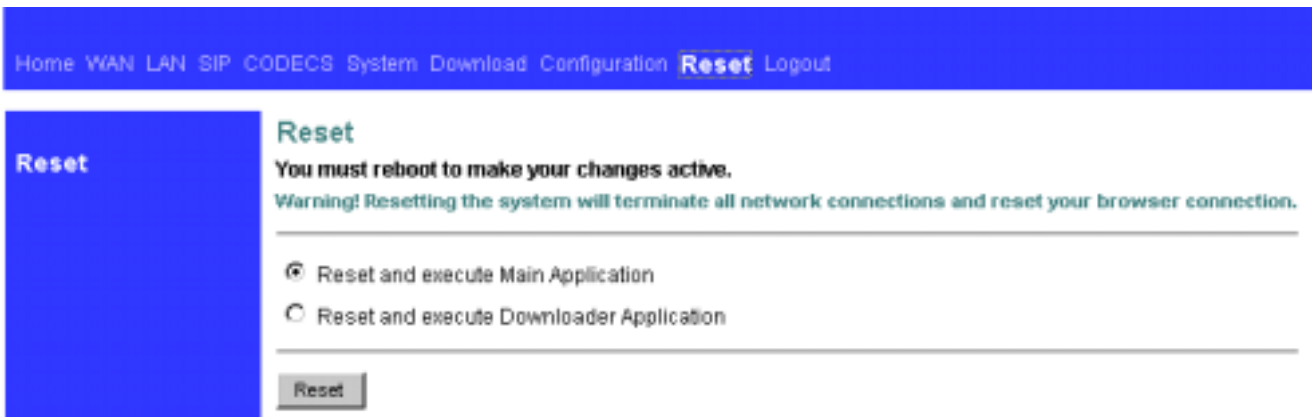
Check the proper boxes enabling LAN and WAN for the **HTTP**, **SNMP**, and **VOIP Discovery**.

Click on “**Save Service Access Settings**”, to save the configurations.

Download

For both **HTTP** and **TFTP** methods, the device will reboot itself into the downloader mode if the main application is executing, and proceed with the ROM file download and permanent write of the application to the device’s flash memory. After the download is completed, the download status page will be displayed.

Reset



The screenshot shows a web interface with a blue header bar containing navigation links: Home, WAN, LAN, SIP, CODECS, System, Download, Configuration, **Reset**, and Logout. Below the header, there is a blue sidebar with the word "Reset" in white. The main content area has a green heading "Reset" followed by a warning: "You must reboot to make your changes active. Warning! Resetting the system will terminate all network connections and reset your browser connection." Below this, there are two radio button options: "Reset and execute Main Application" (which is selected) and "Reset and execute Downloader Application". At the bottom of the form is a "Reset" button.

Chose the “**Reset and execute Main Application**” option, for execution of the main application which you have configured, once you reset the system.

Chose the “Reset and execute Downloader Application” option to download, once you reset the system.

Appendix A: Dial Plans

The SIP code will allow provisioning (via web browser) of the dial plan. A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the gatekeeper for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

The formal syntax of the dial plan is described by the following notation:

Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"

Timer ::= "T" | "t"

Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c" | "D" | "d"

Range ::= "X" | "x" -- matches any digit

| "[" Letters "]" -- matches any of the specified letters

Letters ::= Sub range | Sub range Letters

Sub range ::= Letter -- matches the specified letter

| Digit "-" Digit -- matches any digit between first and last

Position ::= Letter | Range

String Element ::= Position -- matches any occurrence of the position

| Position "." -- matches an arbitrary number of occurrences

including 0

String ::= String Element | String Element String

String List ::= String | String "|" String List

Dial Plan ::= String | "(" String List ")"

A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan, if the result is under qualified (partial matches at least one entry) then it will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match) then send the string to the gatekeeper and clear the dial string. The Timer T is activated when it is all that is required to produce a match. The period of timer T is 4 seconds. For example a dial plan of (xxxT|xxxxx) will match immediately if 5 digits are entered, it will also match after a 4-second pause when 3 digits are entered.

● Sample Dial Plans

Simple Dial Plan

Allows dialing of 7 digit numbers (e.g. 5551234) or an operator on 0. Dial plan is (0T|xxxxxxx)

Non-dialed Line Dial Plan

As soon as handset is lifted, the unit contacts the gatekeeper (used for systems where DTMF detection is done in-call). Dial plan is (x.) i.e. match against 0 (or more) digits.

Note: the dot ‘.’

Complex Dial Plan

Local operator on 0, long distance operator on 00, four digit local extension number starting with 3,4 or 5, seven digit local numbers are prefixed by an 8, two digit star services (e.g. 69), ten digit long distance prefixed by 91, and international numbers starting with 9011+variable number of digits.

Dial plan for this is:

(0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)

Appendix B: Cable Requirement

A CAT 3,4 or 5 UTP (unshielded twisted pair) cable is typically used

To connect the Ethernet device to the modem.

A 10Base-T cable often consists of four pairs of wires, two of which are used for transmission. The connector at the end of the 10Base-T cable is referred to as an RJ-45 connector and it consists of eight pins.

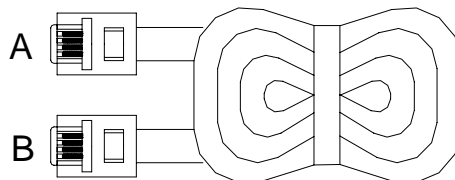
The Ethernet standard uses pins 1,2,3 and 6 for data transmission purposes.

Table RJ-45 Ethernet Connector Pin out Assignments

PIN	MNEMONIC	FUNCTION
1	TX+	Ethernet differential Transmit signal(+)
2	TX-	Ethernet differential Transmit signal(-)
3	RX+	Ethernet differential receive signal(+)
4	NC	Unused
5	NC	Unused
6	RX-	Ethernet differential receive signal(-)
7	NC	Unused
8	NC	Unused

Standard telephone wire of any gauge or type-flat, twisted or quad is used to connect the Modem to the telephone network. A telephone cable typically consists of three pairs of wires, one of which is used for transmission. The connector at the end of the telephone cable is called RJ-11 connector and it consists of six pins. POTS (plain old telephone services) use pins 3 and 4 for voice transmission. A telephone cable is shown below.

Figure Telephone cable



The A and B connectors on the rear of the modem are RJ-11 connectors. These connectors are wired identically. The RJ-11 connectors have six positions, two of which are wiring, The Modem uses the center two pins. The pin out assignment for these connectors is presented below.

Table RJ-11 Pin out Assignments

Pin#	MNEMONIC	FUNCTION
1	NC	Unused
2	NC	Unused
3	TIP	POTS
4	RING	POTS
5	NC	Unused
6	NC	Unused_

Appendix C: Product Specification

Product Name VOIP with Router Adaptor
Application Home networking solution
Product Specification

- **Hardware**

- **Digital Signal Processors & Control Processor**

- System On Chip (SOC) for Network Processing and DSP Application
 - MIPS-X5 unified RISC and DSP core (up to 180 DSP MIPS)
 - 384K bytes on-chip RAM, 16-way interleaved with single cycle access
 - 16-K byte cache
 - Low power, 1.8V core voltage, 3.3V I/O voltage
 - 2M bytes flash memory

- **I/O**

- 2 Standard 10/100 Base-TX RJ 45 interface for 2 FXS model
 - 2 RJ 11 Loop Start interfaces for FXS

- **Mechanical, Environment & Power**

- Dimension: L x W x H = 176mm x 143mm x 27mm
 - Operating temperature: 0°C to 50°C (32 to 122 F)
 - Operating humidity: 10% to 95% (non-condensing)
 - Storage temperature: -10 to 60°C (14 to 140F)
 - AC-to-DC power supply (12VDC, 120 VAC, 60 Hz. For US or 12VDC, 230VAC,50Hz for Europe)

- **Power consumption: 3.5 watt (Typical)**

- **Compliant**

- CE
 - FCC part 15 A

- **Software**

- Compression algorithms: ITU G. 711, G.723, G726, G.729A/B Hybrid echo cancellation G.168 (16 ms)
 - DTMF tone detection/regeneration
 - Comfort Noise Generation (CNG)
 - User programmable Call Progress detection/generation
 - Voice Activity Detection (VAD)

- **Management Tools**

- HTTP Server
 - TFTP and flash memory for remote software download and upgrade

- **SIP Protocol Stack**

- Compliant with SIP v2.0 (RFC 3261)

- **MGCP Protocol Stack**

 - Compliant with MGCP protocol specifications (RFC 3435)**

- **Lan / Wan Functions**

 - Tagging VLAN(IEEE802.1q)**

 - QOS (IEEE802.1p)**

 - DHCP Server**

 - PPPOE**

 - NAT**

 - Firewall**

 - SNMP(V1/V2)**

Appendix D: Troubleshooting

● **Diagnosing the VOIP's Indicators**

The VOIP can be easily monitored through its comprehensive panel indicators. These indicators assist the network manager in identifying problems the hub may encounter. This section describes common problems you may encounter and possible solutions

1. Symptom: Power LED indicator does not light up (green) after power on.
Cause: Defective External power supply
Solution: Check the power plug by plugging in another that is functioning properly. Check the power cord with another device. If these measures fail to resolve the problem, have the unit power supply replaced by a qualified distributor.
2. Symptom : WAN/LAN LED indicator does not light up (green) after making a connection.
Cause: Network interface (e.g, a network adapter card on the attached device), network cable, or switch port is defective.
Solution:
 - 2.1 Power off for the VOIP.
 - 2.2 Verify that the switch and attached device are powered on.
 - 2.3 Be sure the cable is plugged into both the switch and corresponding device.
 - 2.4 Verify that the proper cable type is used and its length does not exceed specified limits.
 - 2.5 Check the VOIP on the attached device and cable connections for possible defects.
 - 2.6 Replace the defective VOIP or cable if necessary.
3. Symptom : Phone 1/2 LED indicator does not light up (green) after making a connection.
 - 3.1 Be sure the phone wire is plugged into both the VOIP and phone set.
 - 3.2 Be sure the phone set is analog type and on hang on status
 - 3.3 Check the VOIP on the phone set and cable connections for possible defects
 - 3.4 Replace the defective VOIP or phone set if necessary

● System Diagnostics

Power and Cooling Problems

If the POWER indicator does not turn on when the power cord is plugged in, you may have a problem with the power outlet, power cord, or internal power supply as explained in the previous section. However, if the unit power is off after running for a while, check for loose power connections, power lost or surges at the power outlet, and verify that the fan on back of the unit is unobstructed and running prior to shutdown. If you still cannot isolate the problem, then the internal power supply may be defective. In this case, contact your dealer.

Installation

Verify that all system components have been properly installed. If one or more components appear to be malfunctioning (e.g., the power cord or network cabling), test them in an alternate environment where you are sure that all the other components are functioning properly.

Transmission Mode

The default method of selecting the transmission mode for RJ-45 ports is 10/100 Mbps ETHERNET, for RJ-11 port are Voice. Therefore, if the Link signal is disrupted (e.g., by unplugging the network cable and plugging it back in again, or by resetting the power), the port will try to reestablish communications with the attached device via auto-negotiation. If auto-negotiation fails, then communications are set to half duplex by default. Based on this type of industry-standard connection policy, if you are using a full-duplex device that does not support auto-negotiation, communications can be easily lost (i.e., reset to the wrong mode) whenever the attached device is reset or experiences a power fluctuation. The best way to resolve this problem is to upgrade these devices to a version that supports Ethernet and analog phone set.

Physical Configuration

If problems occur after altering the network configuration, restore the original connections, and try to track the problem down by implementing the new changes, one step at a time. Ensure that distance of cable and other physical aspects of the installation do not exceed recommendations

System Integrity

As a last resort verify the switch integrity with a power-on reset. Turn the power off and then on several times. If the problem still persists after you have completed all the preceding diagnoses, contact your dealer

Appendix E : Compliance and Safety Information

FCC Radio Frequency Interference Statement

This equipment has been tested to comply with the limits for a computing device, pursuant to Part 15 of FCC rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment can generate, use and radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by taking one or more of the following measures :

1. Reorient or relocate the receiving antenna.
2. Increase the distance between the equipment and receiver.
3. The equipment and the receiver should be connected to outlets on separate circuits.
4. Consult the dealer or an experienced radio/television technician for help.

Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

If this telephone equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the proper functioning of your equipment. If they do, you will be notified in advance in order for you to make necessary modifications to maintain uninterrupted service.

This equipment may not be used on coin service provided by the telephone company. Connection to party lines is subject to state tariffs.

Important Safety Instructions

Caution : The direct plug-in wall transformer serves as the main product for disconnecting. The socket outlet shall be installed near the product and be readily accessible.

Caution : Use only the power supply included with this product. In the event the power supply is lost or damaged : In the United States, use only with CSA certified or UL listed Class 2 power supply, rated 5Vdc 1A or above.

IN Europe, use only with CE certified power supply, rated 5Vdc 1A or above.

Do not use this equipment near water, for example in a wet basement.

Avoid using a telephone during an electrical storm. There may be a remote risk of electrical shock from lightning.

Do not use the telephone to report a gas leak in the vicinity of the leaking area.

If you experience trouble with this unit, please contact customer service at the address and phone listed below. **DO NOT DISASSEMBLE THIS EQUIPMENT.** It does not contain any user serviceable components.

FCC Warning

This equipment has been tested to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment can generate, use, and radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful

interference in which case the user will be required to correct the interference at owner's expense.

CE Mark Warning

This is a CE class A product. In a domestic environment, this product may cause radio interference in which case the user may be required to take adequate measures.

Warranty

The original owner of this package will be free from defects in material and workmanship for one-year parts after purchase. For the warranty to apply, you must register your purchase by returning the registration card indicating the date of purchase.

There will be a minimal charge to replace consumable components, such as fuses, power transformers, and mechanical cooling devices. The warranty will not apply to any products which have been subjected to any misuse, neglect or accidental damage, or which contain defects which are in any way attributable to improper installation or to alteration or repairs made or performed by any person not under control of the original owner.

THE ABOVE WARRANTY IS IN LIEU OF ANY OTHER WARRANTY, WHETHER EXPRESS, IMPLIED, OR STATUTORY, INCLUDING BUT NOT LIMITED TO ANY WARRANTY OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE, OR ANY WARRANTY ARISING OUT OF ANY PROPOSAL, SPECIFICATION, OR SAMPLE. SHALL NOT BE LIABLE FOR INCIDENTAL OR CONSEQUENTIAL DAMAGES. WE NEITHER ASSUMES NOR AUTHORIZES ANY PERSON TO ASSUME FOR IT ANY OTHER LIABILITY.

Note: Please do not tear off or remove the warranty sticker as shown, otherwise the warranty will be void

