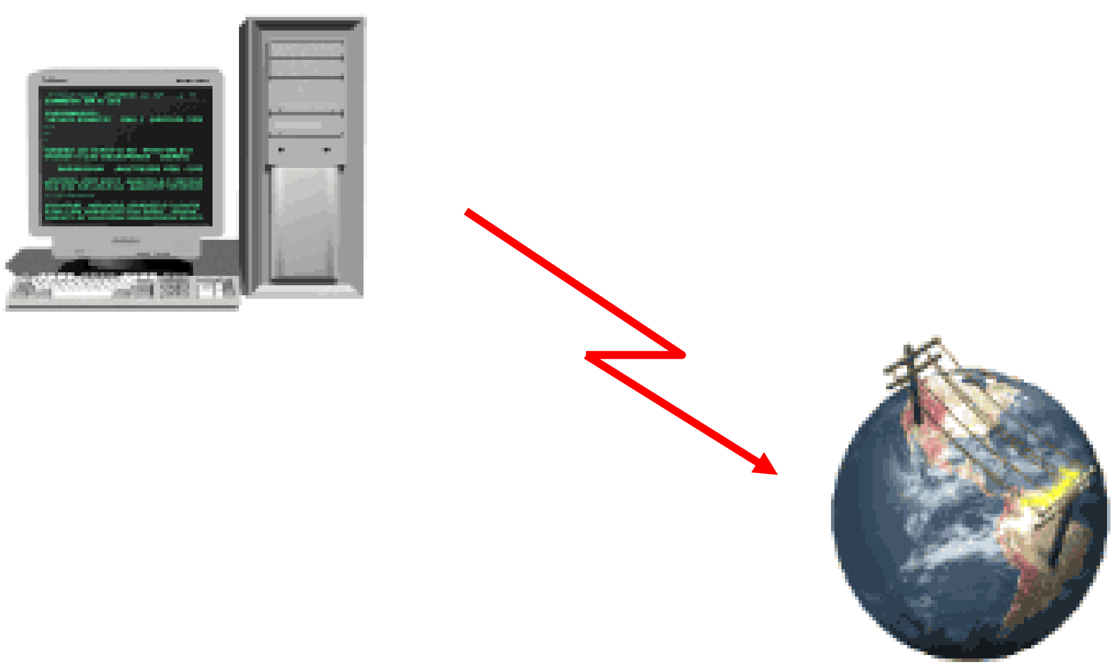
Voice over Internet Protocol
(VoIP)

Aristel Networks D1 ITK VoIP Card

Provision – Installation – Programming



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ARISTEL D1ITK CARD

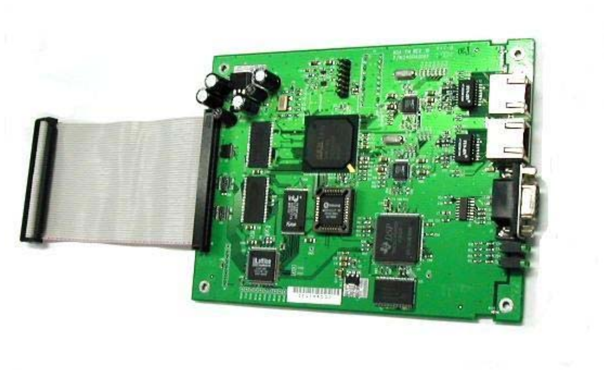
WARNINGS!

IT IS STRONGLY RECOMMENDED THAT THE D1ITK CARD BE CONNECTED TO AN EXCLUSIVE-USE ADSL CONNECTION WITH A STATIC IP ADDRESS.

DIALUP LINKS WILL NOT PROVIDE ENOUGH BANDWIDTH FOR ACCEPTABLE PERFORMANCE.

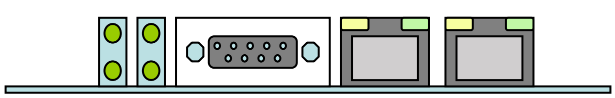
THE VOIP FEATURE UP/DOWN LOADS VERY LARGE AMOUNTS OF DATA IN A SHORT PERIOD OF TIME. THE ADSL CONNECTION SHOULD BE "UNLIMITED UP/DOWN LOAD" OR THE VOIP ECONOMIC ADVANTAGE COULD BE SERIOUSLY ERODED.

CONNECTING A NETWORK TO THE LAN PORT WILL INCREASE TRAFFIC OVER THE ADSL LINK AND REDUCE THE PERFORMANCE OF THE VOIP. THE HEAVIER THE TRAFFIC, THE MORE DELAYS/INTERRUPTIONS WILL BE EXPERIENCED



D1ITK CARD

Make sure the following contents are included in the package
1 Unit – System card
1 Unit – RS-232 cable
4 stand-offs for mounting the card in an expansion position



RS-232 cable

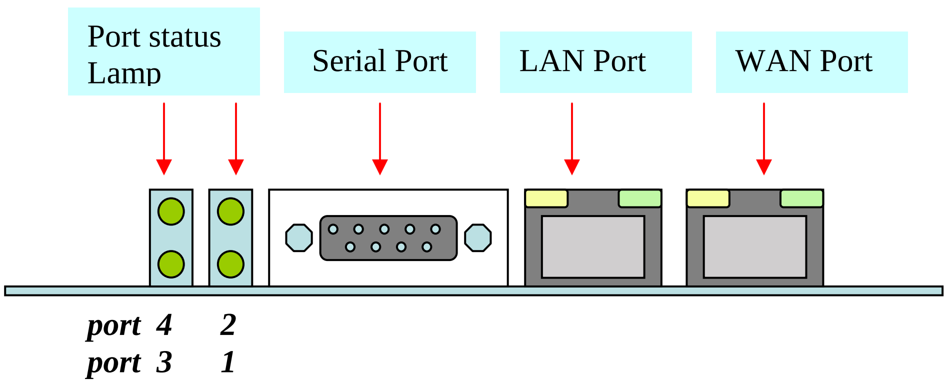
INSTALLATION

The D1ITK card provides 4 channels (trunks) access to the network. i.e. 4 simultaneous VOIP calls can be made from one card. More than 1 card maybe fitted if required.

OVER-VIEW OF COMPLETE INSTALLATION

1. Unpack all contents of the System package.
2. Mount the card as the first trunk card or as an expansion trunk card.
3. Plug the card into the DV-38/96 main board using a CO board connector.
4. Connect the Local Area Network (LAN) cable to the RJ-45 port at the front of the System using cat 5 patch leads
   Physical installation is now complete.
5. Connect the RS-232 cable to PC's serial port. (for 1st time use configuration)
6. Power on DV-38/96
7. Configure the card via the serial port:-
   Installation programming is now complete.

Voice System Card Panel:



Port satus LED functions

Off = channel idle On = channel in use

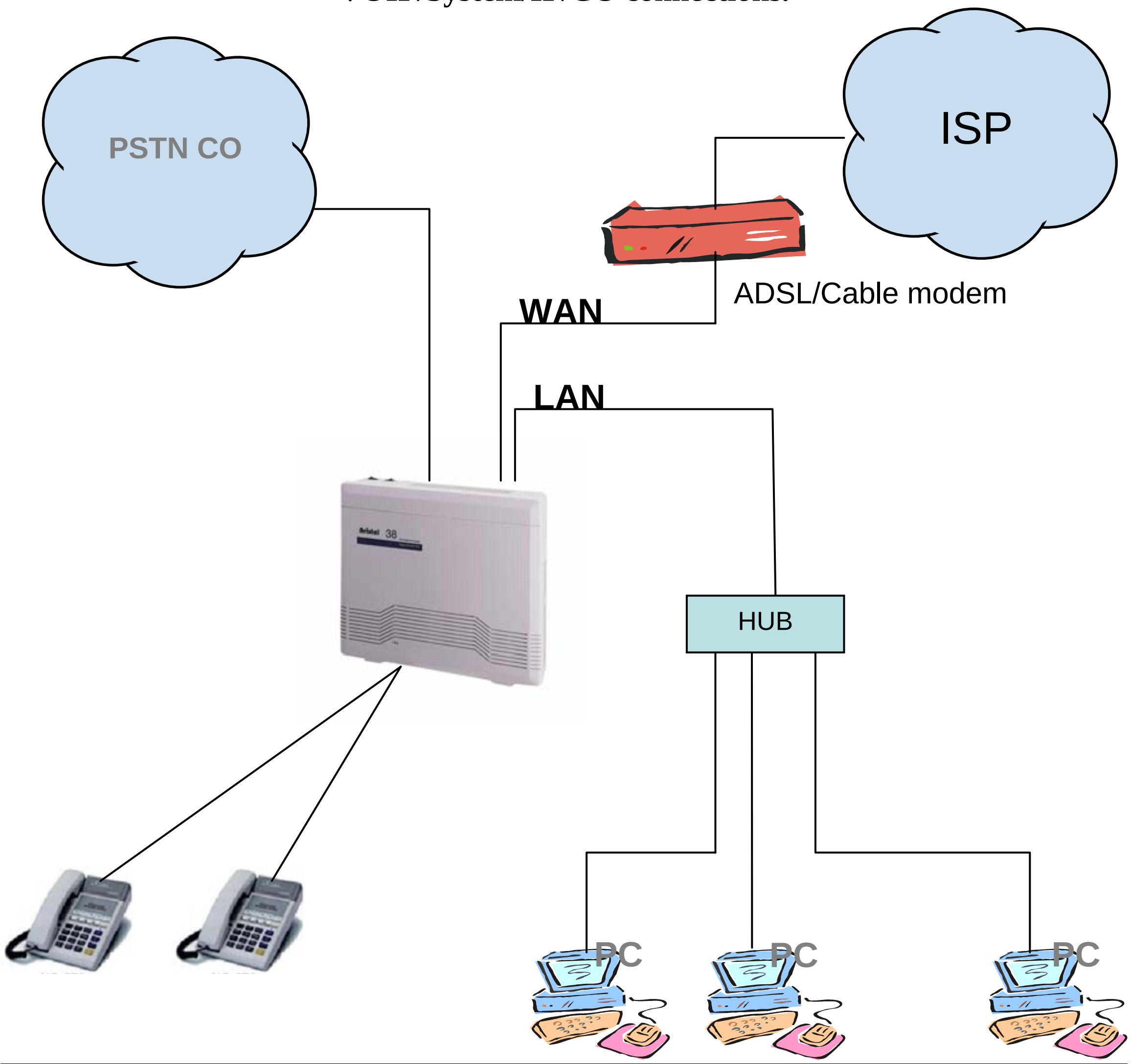
WAN and LAN LED indicator functions.

LED	Light	Activity
Link use	*Green*	*LAN is receiving/transmitting data. Network connection is OK*
Collision	*Yellow*	*Data collisions have occurred during transmission*

Serial port details

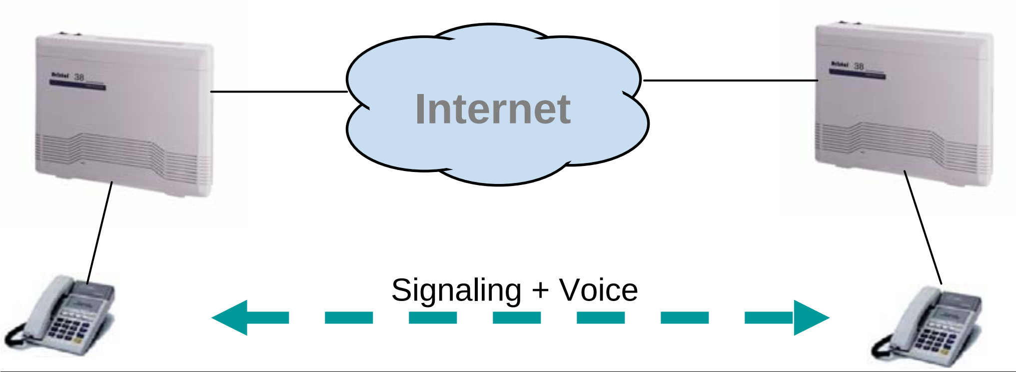
Speed: 38400 bps
Data bits: 8
Parity check: none
Stop bit: 1
Flow control: none

VOIP/System/IP/CO connections.



Note!
Connection of a network to the LAN port will reduce the performance of the VOIP channels.

Peer to Peer connection



Peer to Peer programming

The VOIP systems have static IP addresses. The Gatekeeper's IP address **MUST** be 0.0.0.0 (disabled)

In Gateway Configuration

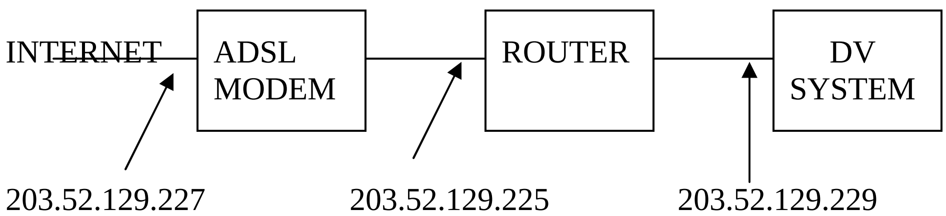
In Network Parameters

Enter your IP address of the DV system (eg 203.52.129.229)

Enter the Default Router address (eg 203.52.129.225)

Enter the network mask data (eg 225.225.225.248)

The example given below shows the information needed.



In H.323 Related Functions

Set the VOIP Protocol = H.323

In Phone/Line Dialling Plan

Remove all port numbers

Set the port numbers to the desired numbers (eg 98372300)

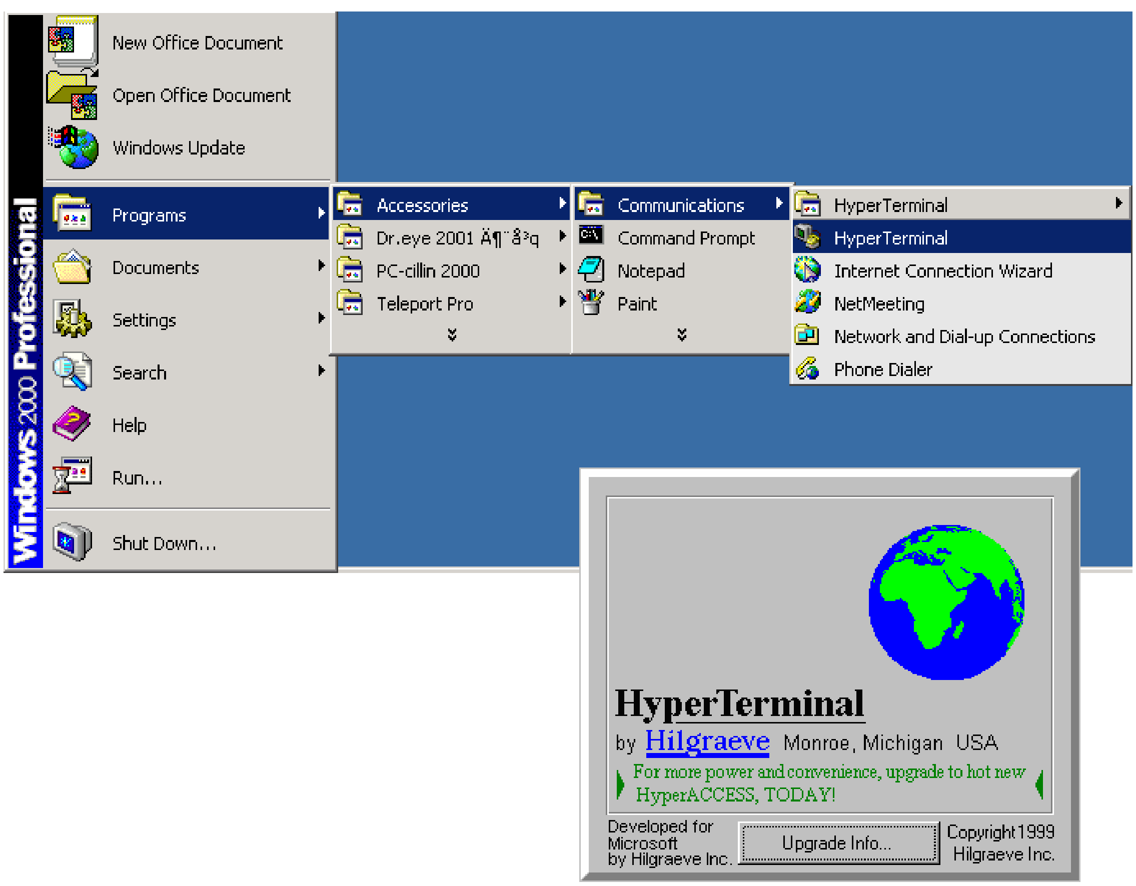
Save the data and restart the system.

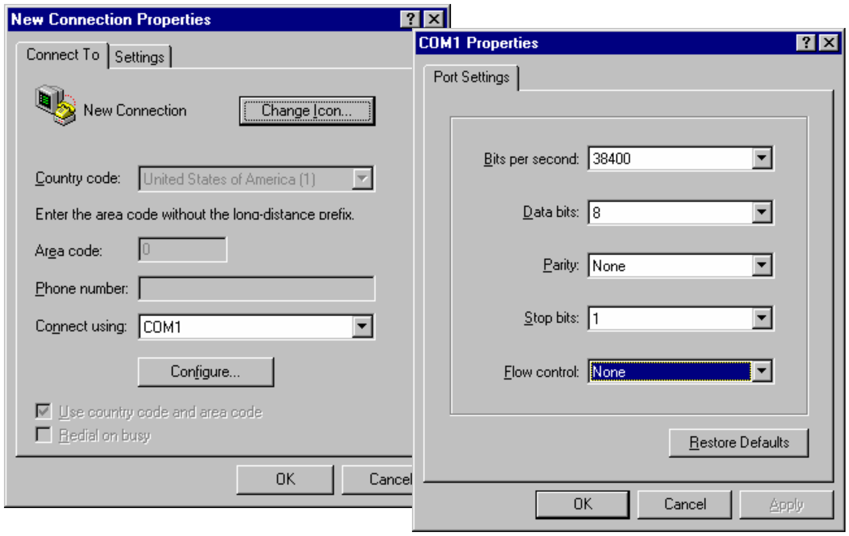
To call another system, you dial the designated port number of the other system.
The last three (for example) digits can be passed onto the remote system for "in-dialling" into the system.
i.e. Strip off the first 5 digits and the last three 300 will be passed onto the remote system. See programming the card for full details.

Configuring the system through the RS232 serial port.

The system can be configured through the serial connection, via a telnet and web browser. Before connecting to the network, you have to setup the IP address for the system. MS Windows's HyperTerminal is a good choice to configure the system. The following settings are the values for serial terminal connection to the system.

- Speed: 38400 bps
- Data bits: 8
- Parity check: none
- Stop bit: 1
- Flow control: none





You must then enter the command:- login
The screen may not echo your command but will respond as below when you enter the Login password: aristel

If the setting is as listed, the system will show the following menu in the Console/Telnet.

Notice:

- **The default login password is “aristel”**

```
---------------------------------------------------

---- Aristel (1A )        Version 1.1.3 (5409)  ----

---------------------------------------------------

1. Set WAN Port network parameters.

2. Set H.323/SIP related functions.

3. Set Port's Dialing Number.

4. Set Hot Line Calls.

5. Set Direct Dialing Numbers.

6. Show All Numbering Plan.

7. Voice processing Control.

8. System access Control.

9. Advanced Configurations.
```

You may key in the selected item and press enter to process the command. For telnet users, you will get the prompt, please enter telnet's password.

```
telnet 10.10.10.51
```

You can change the default password under the "8. System access Control":

Notice:

1. **The default Telnet IP address is "10.10.10.51".**
2. **The default Telnet password is "aristel" .**
3. **Pressing Ctrl + 'a' will reset all setting values back to factory default values.**

Set Network Parameters

The first thing you need to do is set the IP address, please key in ‘1’ and enter your data to set the network parameters.

```
My IP = 10.10.10.51
Default Router IP = 10.10.10.254
Network Mask = 255.255.255.0
My MAC address = 0:4:f:11:11:22
CDR Server IP = 0.0.0.0

1. Change My IP address.
2. Change the Default Router IP address.
3. Change the Network Mask.
4. Enable DHCP Client.
5. Enable PPPoE Client.
6. Ping.
7. Pass Through NAT/Firewall
8. Change CDR Server IP. (Set IP=0 will disable it.)

'Esc' to Main Menu.
Choose one item :
```

You can set fixed IP address by choosing PPPoE method, please contact your network administrator to find the settings for your network

After the above steps have been done:

- Press “Esc” to return to the main menu.
- Press ‘f’ to save current configurations to ROM.
- Then press “g” to restart the System system.

If all the network settings are correct, the system will connect to the network.

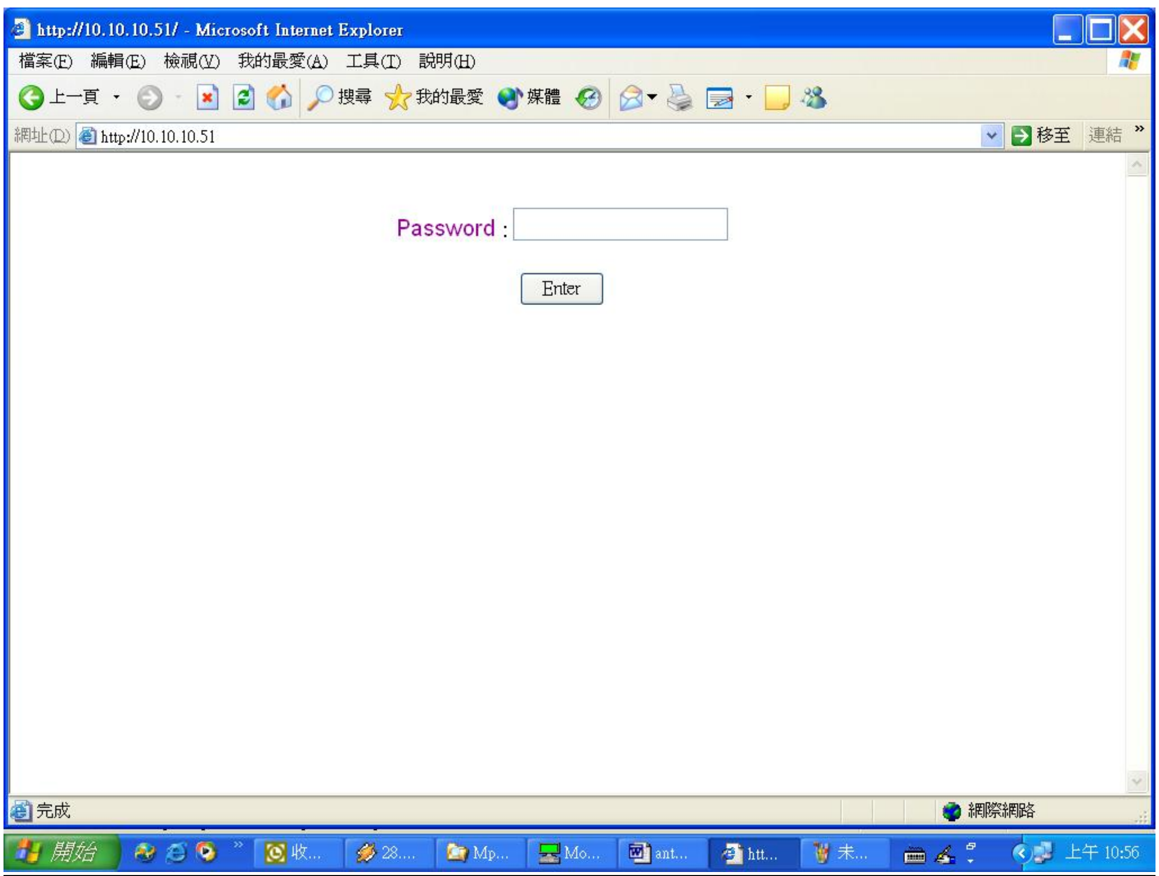
Using Web Browser To Configure System System

Connecting to the system from web browser

When the system is connected to the network, you can easily configure the system by using a web browser

Step:

- In the web browser, key in the IP address of your system (e.g. http://10.10.10.51) as a URL, then the browser will connect to the System and you can see the following web page:
- NOTE! The computer must not be connected to the LAN output of the system. Any computer connected to the Internet not via the system may be used

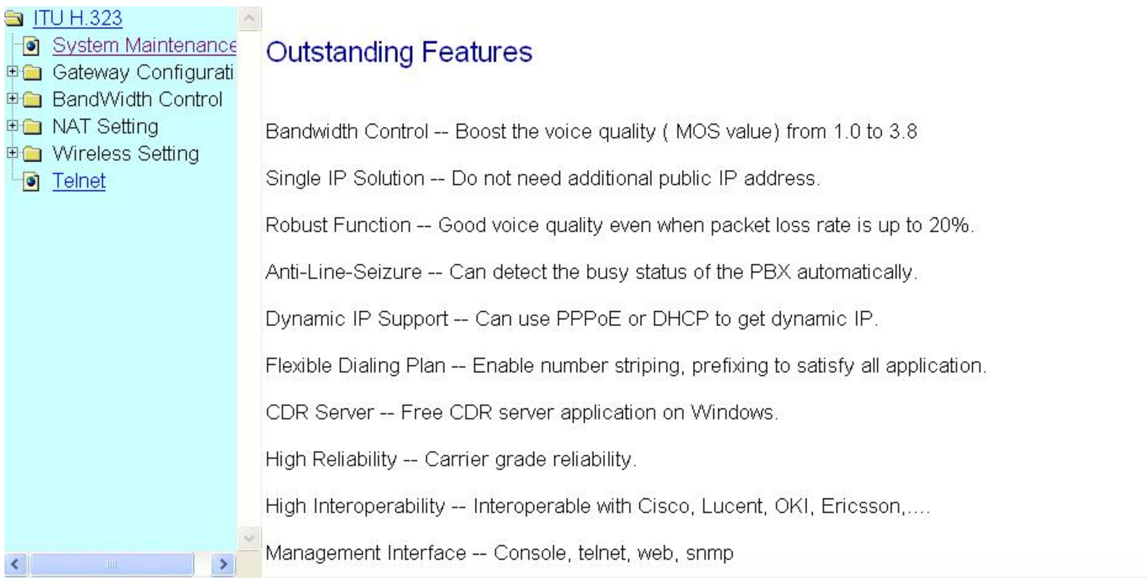


Notice:

1. **The default password is "aristel".**
2. **The default IP for System is "10.10.10.51".**

Change configuration of the system

After you have completed the login steps, you will see the following page. Now you can change the system settings. The main menu is on the left side, and it includes three main categories "System Configuration", "Bandwidth Control" (Voice System Bandwidth Management-Series) and "SNMP Statistics". They will be explained in the following sections.



Steps:

- Click on "System Maintenance" from the menu, and move to the next page.

System System Maintenance

In this page, you have 3 choices. :-
1. Setting the system to default configuration,
2. saving current configuration
3. restarting the System.

Steps:

- Press the “SET” buttonto restore the settings to default values.
- Press the “SAVE” buttonto write current settings to flash memory.
- Press the “RESTART” button,to restart the system.

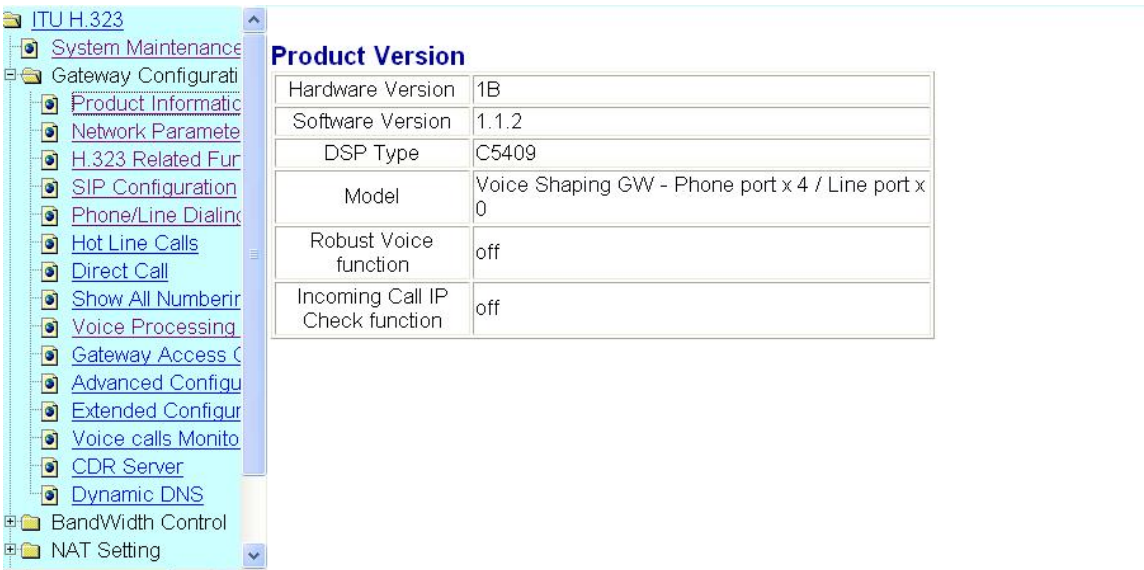
Notice:

“Save” every time after you change settings, then “Restart” the system.

System Configuration

Product Version

If “Product Version“ is under the “System Configuration” menu, you will see the following page. In this page, you can find the information related to this System product, including Hardware version, Software version, Model, DSP type, etc.



Steps:

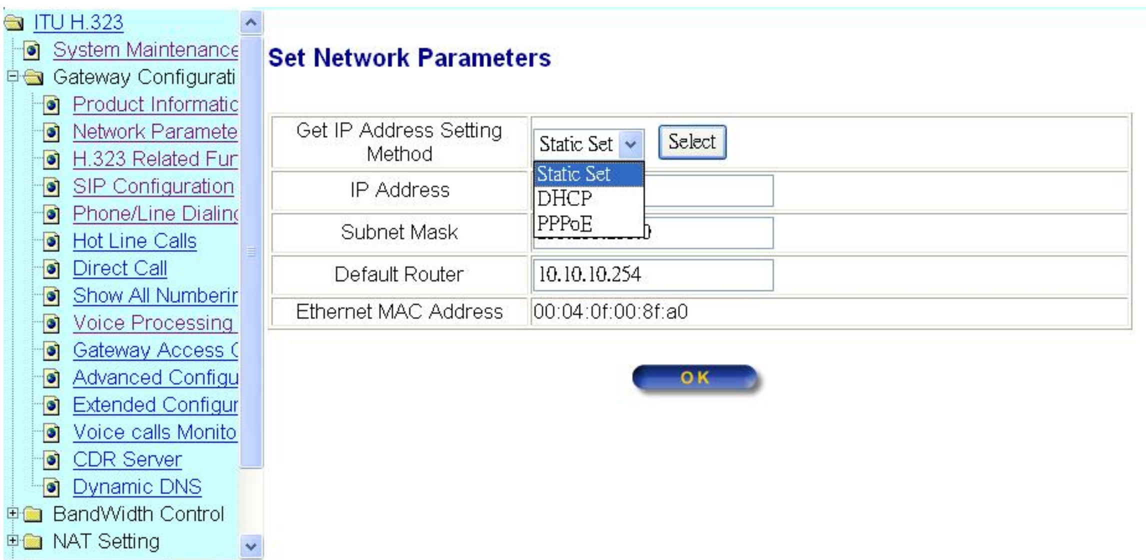
- Click “Product Version” under the “System Configuration” menu.

Network Parameters

By clicking on "Network Parameters" under the "System Configuration" menu, you can change network parameters.

Steps:

- Select from the combo box the method for getting the IP address, it must be "Static set", "DHCP". Press "SET" button to confirm it.
- When "Static Set" is selected, key in the IP address, subnet mask and default router to corresponding field. Press "OK" button to change these parameters.
- Press "OK" button to confirm the setting. Notice that, if this setting is changed, you need to save and restart the system to make the system work properly.



Settings for Console/Telnet only (Not normally required. Consult with Aristel Networks before altering ant settings)

1. Set the network parameters.
   - 1-6 Ping.
   - 1-7 Pass Through NAT/Firewall
     - 1-7-1 Enable Pass Through NAT/Firewall Function.
     - 1-7-2 Change Base Port.
     - 1-7-3 Select the Type of Public IP of NAT/Firewall.
     - 1-7-4 Set the Public IP of NAT/Firewall.

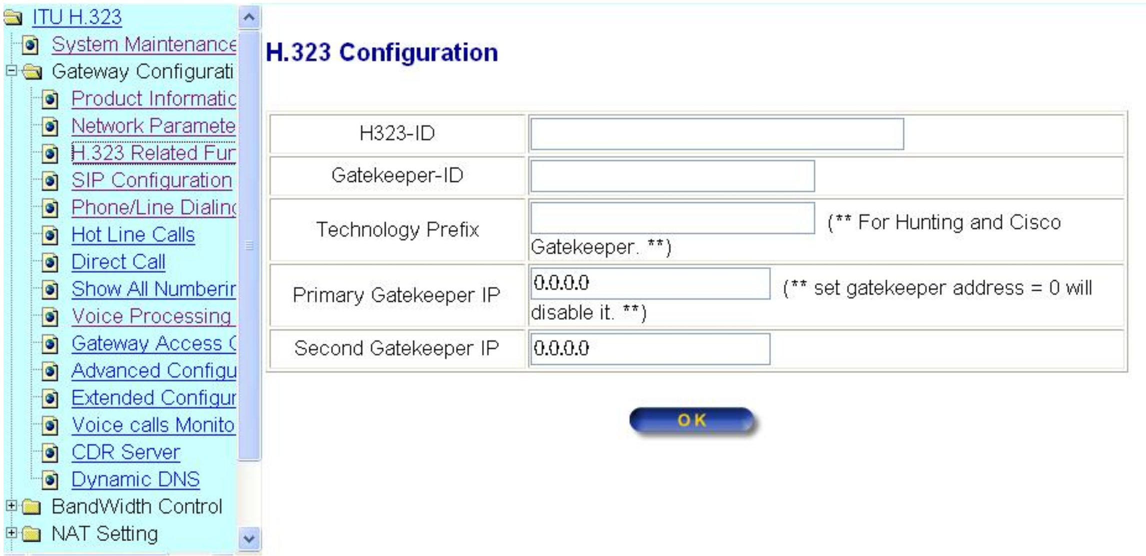
Passing-through NAT/Firewall

- Enable Pass Through NAT/Firewall Function
- Please set the Base Port, the system will use only those TCP/UDP ports range from Base Port to Base Port + 20, for example, if the Base Port is 1719, then only the ports 1719 to 1738 are used by all the VoIP TCP/UDP connections. In order to let the system passing through the NAT device, the NAT device must enable the "virtual server" function to map the port range from 1719~1738 to the system. Thus, any TCP/UDP connections within this port range (1719~1738) will connect to the system.
- Select the type of Public IP of NAT/Firewall, the type is fixed IP or dynamic IP. If the NAT device is using fixed IP address, then the system needs to set this fixed public IP address. If the NAT device is using **dynamic IP address**, the system will need the help of the **CDR server software**. The CDR server software can tell the system the NAT device's dynamic IP address.
- If the NAT device is using dynamic IP address, the system must register to a gatekeeper to work properly.
- If there is only one system behind the NAT device and the NAT device is using a fixed IP address, you don't need the gatekeeper.
- Please be noticed that some kind of NAT device could try to do H.323 address translation, but doing not correctly, in this situation, the NAT pass-through function could fail.

Notice:

1. **You can use console/telnet to control the system and ping others if you would like to ping another device from the system.**
2. **Please go to menu 1-6 under the console/telnet to do so.**

8.3.3 H323 Configuration



Usage:

THE USE OF THE GATEKEEPER FUNCTION IS STRONGLY NOT RECOMMENDED. CONSULT WITH ARISTEL NETWORKS IF REQUIRED

- The “H323-ID” is the H323-ID of this system when registering to the gatekeeper
- The “Gatekeeper-ID” is the gatekeeper’s ID, usually you do not need to set this field unless the gatekeeper needs this value.
- **Gatekeeper backup:** There are two gatekeeper address fields, one is primary, the other is secondary. If this system does not want to register to any gatekeeper, just set value 0 to the primary gatekeeper address. If the primary gatekeeper address is not 0, the system will register to the primary gatekeeper. If the second gatekeeper is not 0, the system will try to register to the second gatekeeper when the registering to primary gatekeeper has failed., i.e. if both the primary gatekeeper and second gatekeeper addresses are present, the system will try to register to these two gatekeeper alternatively. This way, the system can have the gatekeeper backup function.
- **Hunting function:** The “Technology prefix” is for hunting usage, where many systems require to be hunted by a same number, you can assign the “Technology prefix” here to register this prefix number to the gatekeeper. The gatekeeper will hunt those systems with the same prefix address alternatively and skip those systems that have no available ports. The system must also assign the local dialing plan in the “direct dial plan” field to tell this system how to handle the “prefix address”. For example, if the prefix address is 001, then there must be an element in the “direct dial plan”, suppose the dial plan is leading number = 001, destination = port 1,2,3. Then the system will report to the gatekeeper that this system has three available ports. Any time the available ports 1,2,3 are changes, the status will be reported to

the gatekeeper. This hunting mechanism must co-work with the embedded gatekeeper.

- If the system is going to be used as a gatekeeper, just press "ENABLE" button to enable gatekeeper function on this system. (Gatekeeper models only). When this system is used as a gatekeeper, all the system functions are still the same.

Gatekeeper related functions

Gatekeeper backup functions:

- The system can assign both the primary gatekeeper and the secondary gatekeeper to enable the GK backup function.
- The system will try to find the next available GK, when the current GK has no response for 30 seconds.
- The system will switch between these two GK when the current GK has no response.

Technology Prefix: (For Hunting or Using Cisco Gatekeeper)

- Used by hunting or Cisco System/Gatekeeper
- This prefix will register to gatekeeper. Examples: 001, 002, or 01#, 02#, 03# for Cisco gatekeeper.
- When this prefix is set, all other numbers will not register to gatekeeper.
- All systems with the same prefix are hunted together.

Hunting function:

- The (Technology) prefix address if assigned, will enable the hunting function of the system.
- All the systems with the same prefix address are hunted cyclically.
- The hunting numbers must be added in the system (direct dialing number) to find the destination ports.
- The system will report current available ports to the GK when any port status is change.
- The GK will hunt cyclically unless one system has no available ports.

2. Set H.323/SIP related functions.
   2-1 Select Default VoIP protocol - 1.H323, 2.SIP: 2

```
--------------------------------------------------------------------------
                    Use VoIP Protocol  = SIP
                    Proxy Server is : None.
                    Proxy Server Registering Period = 360 sec.

--------------------------------------------------------------------------

1. Set Default VoIP protocol.
2. Set SIP proxy server.
3. Set Registering Period.

'ESC' to Upper Menu
Choose one item( 1 - 3 ) :
```

There are two parts in the menu selections needed to be set for the SIP protocol usage.

SIP Configuration

Select main menu selection “2. Set H.323/SIP related functions.”, and then select submenu item “1”. Select Default VoIP protocol - 1.H323, 2.SIP. to select “2” the SIP protocol configuration, the configuration display is :

```
--------------------------------------------------------------------------
                    Use VoIP Protocol  = SIP
                    Proxy Server is : None.
                    Proxy Server Registering Period = 360 sec.

--------------------------------------------------------------------------

 1. Set Default VoIP protocol.
 2. Set SIP proxy server
 3. Set Registering Period.
 'Esc' to Upper Menu.
 Choose one item( 1-4 ) :
```

The first menu item is to select the "Default VoIP protocol". This selection will define the VoIP signaling protocol used for all outgoing calls. If the "default VoIP protocol" selected is H.323, then all the outgoing call will use H.323 protocol to call out. If the "default VoIP protocol" selected is SIP, then all the outgoing call will use SIP protocol to call out. But if any number is defined in "direct dialing plan", that specific number will use it's defined protocol.

The "2. Set SIP proxy server." selection is to set the IP address/port of the proxy server. When this proxy server is set, the system will try to register to the proxy server no matter what's the selection of "Default VoIP protocol".(This situation is also true for H.323 registering, i.e. if the H.323 gatekeeper is defined, this system will register to the specific gatekeeper even if the "Default VoIP protocol" selected is SIP).

The "3. Set Registering Period." is to set the interval that the system will re-register to the proxy server when previous registering is successful. If the registering is failed, the system will re-register every 10 seconds.

Direct Dialing Plan for SIP

> If the "Default VoIP protocol" selected is the SIP protocol, all the outgoing VoIP calls will go to the proxy server using SIP protocols except some specific numbers defined in the direct dialing plan.

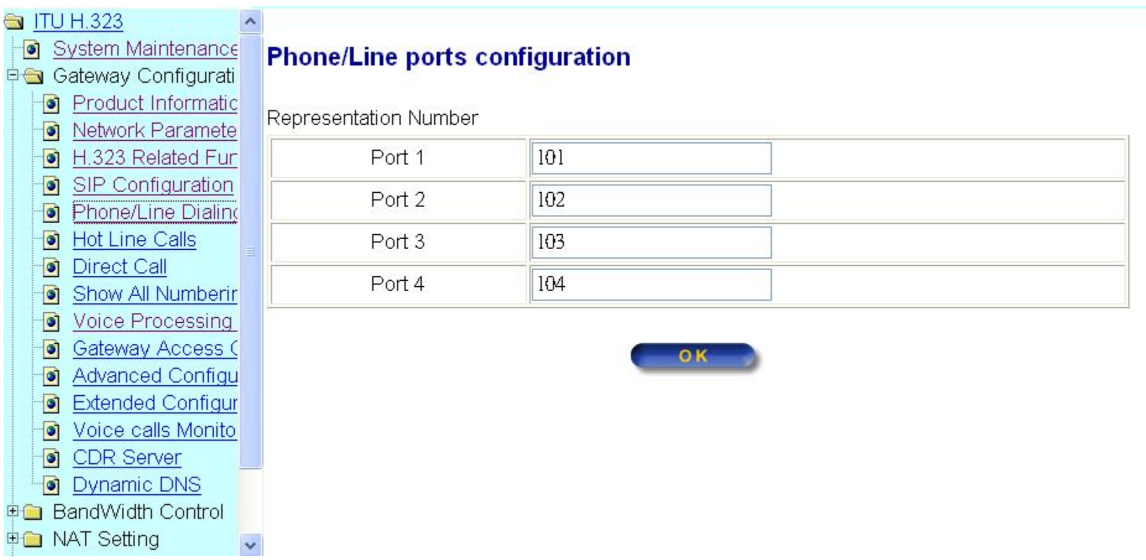
The system can set direct dialing plan for outgoing SIP calls no matter what's the "default VoIP protocol". This could be done by select "5. Set Direct Dialing Numbers." in the main menu. When adding a new plan to the direct dialing plan, after key in all the necessary numbers, then select "Choose Destination - 1: Tel Port, 2: IP address: 3. Domain name to complete dial plan setting.

Phone/Line Dialing Plan

This sectiondescribes the dialing plan of the system. Since there are many different systems, the dialing plan could be divided into the following three major groups.

Phone/Line Dialing Plan

This configuration page is used to configure the numbering plan for those VoIP systems ports only.



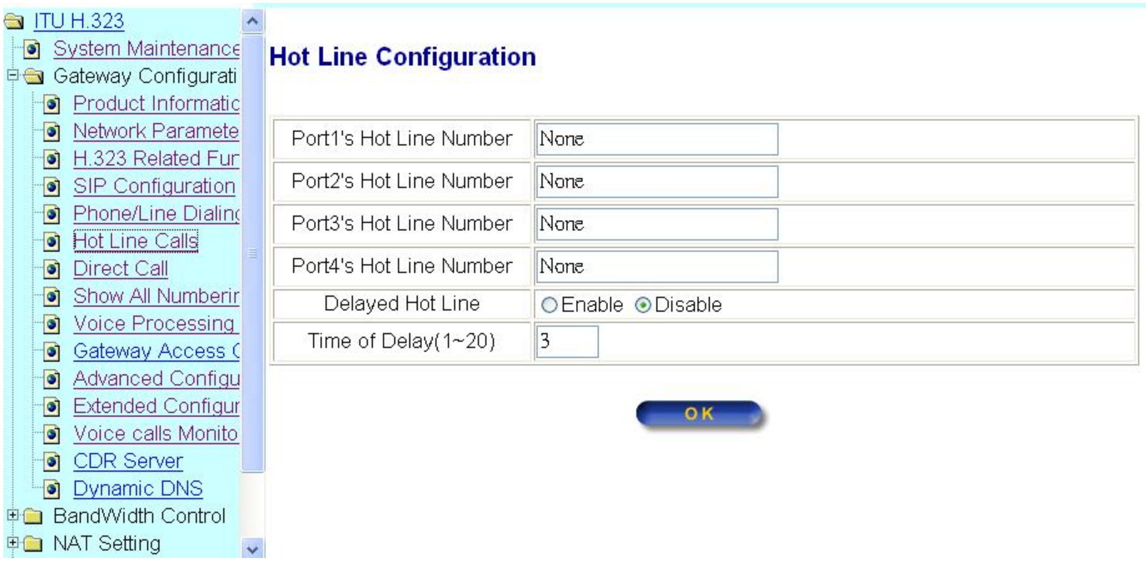
Representation Number

Phone/Line Number:

The representation number is the virtual telephone number of the VoIP port.

Hot Line Calls

The purpose of the "Hot Line Calls" setting is to let user do a "dial free" call. The configuration page is shown below:
If any hot line number is configured for a port, when the port-connected telephone is off hook, the pre-configured hot line number will be automatically dialed out. This port cannot make a phone call to any other number if its "hot line" number is set.



Steps:

- Input hot line number for port 1 to port 4 respectively.
- Press "OK" button to save this setting.

Settings for Console/Telnet only

Set Hot Line Calls.

Default Number of Empty call is:

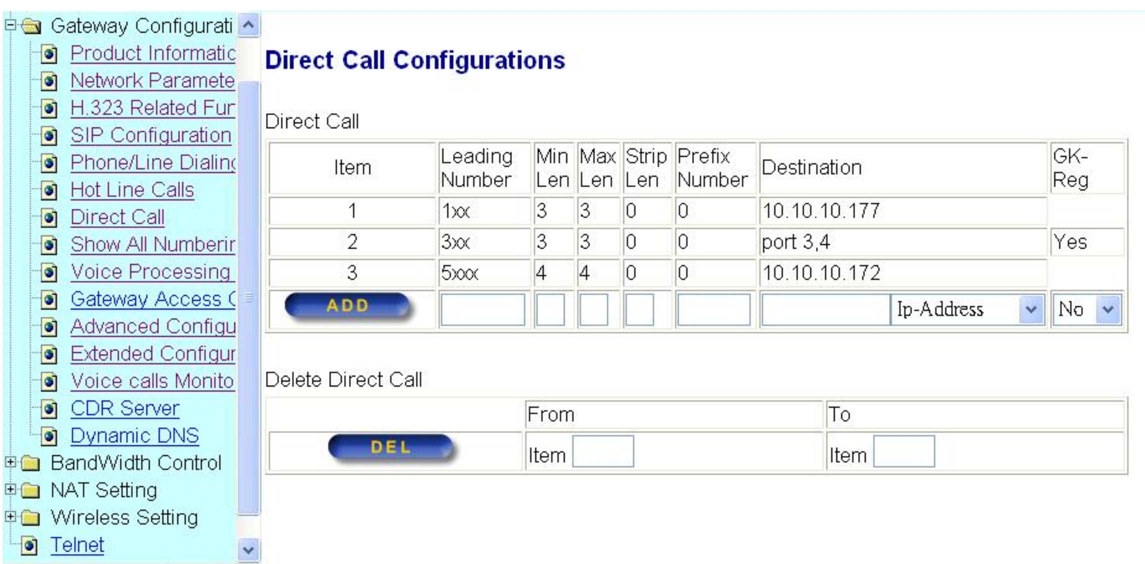
Some times, an incoming call from a network might not carry a called party number (no destination number, empty number), in this situation, the system could give this call a default number to be used as the called party number. This default number should be a port's number, and then the phone set that connects to the specified port will ring.

Direct Call

The purpose of "Direct Call" setting is to let you create a proprietary dialing plan when this System is not registered to any gatekeeper. This setting can also assign a dialing plan to local ports (including prefix strip, prefix addition).
Through this setting, you can directly map a number to a specific system (IP address).

- In the "Direct Call Configurations" settings: (see Appendix C for details)

  - "Leading Number" is the leading digits of the dialing number which can be passed on to the remote PBX.
  - "Min Length" and "Max Length" is the min/max allowed length you can dial.
  - "Strip Length" is the number of digits that will be stripped from beginning of the dialed number. (eg. to pass on a DID extension number)
  - "Prefix Number" is the digits that will be added to the beginning of the dialed number.
  - "Destination" is the IP address of the destination System that owns this phone number. If the "Destination" is "Tel-port", this is for local dial plan setting.
  - **"GK-Reg**" can let you decide if this dialing plan will be registered to the gatekeeper or not when the destination is "Tel-port". This field could give the system more flexibility. If you want other systems to call to this system, you must set "yes" value in the "GK-Reg" field.

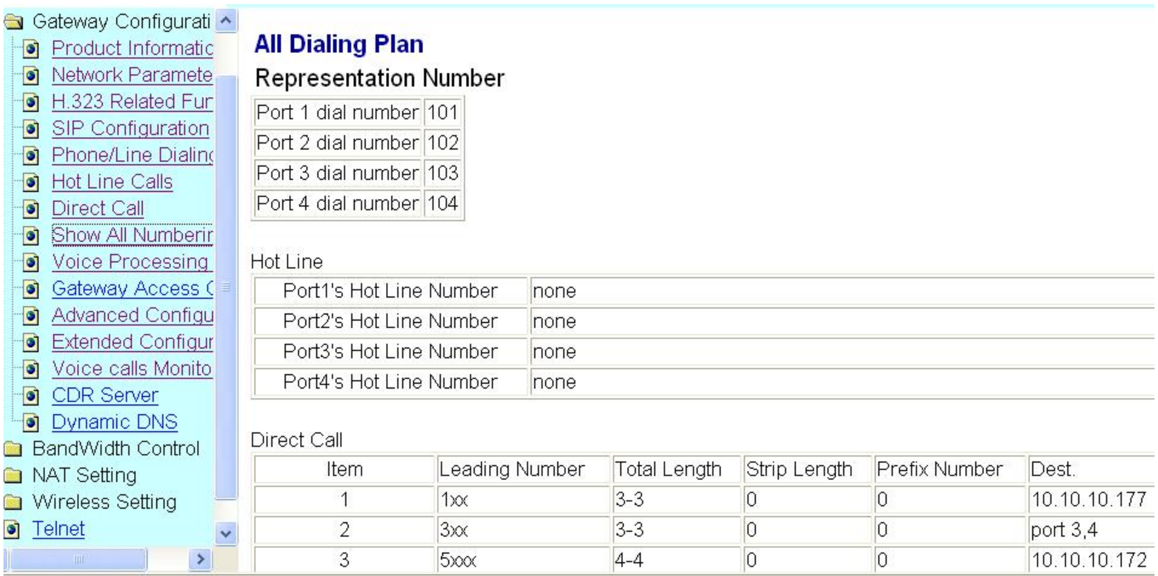


Steps:

- If you want to add a direct call item, fill all the blank fields of “Direct Call” table.
- Select form combo box to choose “Tel-port” or “IP Address”.
- If “Tel-port” is chose, this direct call is out through local analog port (local dial plan).
- If “Tel-port” is chose, you can also set “yes” or “no” in the “**GK-Reg**” field to decide if the “leading number” will register to the gatekeeper or not.
- If “IP Address” is selected, this direct call number is out through the network.
  Notice: this setting is effective only when this System is not registering to another gatekeeper.
- Press “OK” button to add an item.
- If you want to delete a direct call item, input the items you want to delete. Press “DEL” button to delete it.

Show All Numbering Plan

All the numbering plan of this System system will be shown by this selection, as displayed below:



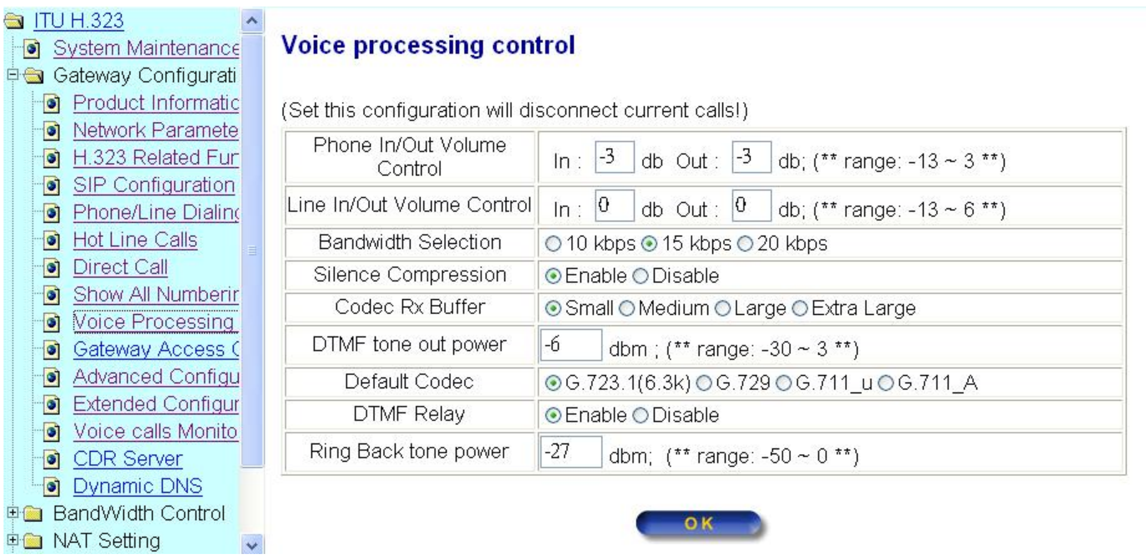
Steps:

- Click “Show All Numbering Plan” in left-side menu.

Voice processing control

This selection is to configure the System and adjust any voice related parameters. **Please do not change any default settings if you are not sure about the meanings of the functions.**

The following page is the configuration page when "Voice Processing Control" has been selected:



Steps:

- Click "voice processing Control" in left-side menu.
- Input the correct value, and make the needed selection.
- Press "OK" button.

The following explains the meanings of all parameters:

Phone In/Out Volume Control:

This function is used for the control of the phone volume. The range is –13dB to 3dB and the default value is –6 dB. When the value is increased, the volume will be increased accordingly. The " **In**" direction is from System to phone set, and the "**Out**" direction is from phone set to System.

Line In/Out Volume Control:

This function is used for the control of the line volume. The range is –13dB to 3dB and the default value is –6 dB. When the value is increased, the volume will be increased accordingly. The " **In**" direction is from System to phone set, and the "**Out**" direction is from phone set to System.

Bandwidth Selection:

This selection is used to choose the bandwidth required in the TCP/IP network. The IP headers are included in the bandwidth. The required bandwidth is dependent on the voice length sent for each packet. The bigger the bandwidth is, the smaller the voice length and hence the smaller the voice delay will be. For 20 kbps, a packet is sent every 30 ms and it has 24 bytes of data in payload. For 15 kbps, a packet is sent every 60 ms and it has 48 bytes of data in payload. For 10 kbps, a packet is sent every 90 ms and it has 72 bytes of data in payload. The maximum bandwidth selection will have the minimum voice delay.

Silence Compression:

If this function is enabled, when silence occurs for a period of time, no data will be sent across the network during this period in order to save bandwidth.

CODEC Rx Buffer:

The Rx buffer is also called the jitter buffer. This buffer is used to decrease the effect of voice packet jitter (delay variation) caused by the network. When the size of this buffer is increased, the jitter effect will be decreased.

DTMF tone out power:

After the VoIP call is connected, when you dial a digit, this digit is sent to the other side by DTMF tone. If the voice path is through many stages (e.g. many PBX), the DTMF tone could be too small to be recognized by the other side. You can adjust this DTMF tone power to increase the recognition. The range of this option is –30 dBm to 3 dBm and the default value is –6 dBm.

Default CODEC:

The CODEC is used to compress the voice signal into data packets. Each CODEC has different bandwidth requirement. There are four kinds of CODEC available; they are G.723, G.729, G.711_u and G.711_A. The default value is G.723.

DTMF Relay:

After the VoIP call is connected, when you dial a digit, this digit is sent to the other side by DTMF tone. There are two methods of sending the DTMF tone. The first is “in band”, that is, sending the DTMF tone in the voice packet. The other is “out band”, that is, sending the DTMF tone as a signal. Sending DTMF tone as a signal could tolerate more packet loss caused by the network. If this selection is enabled, the DTMF tone will be sent as a signal.

Ring Back tone power:

When a call is made from the other side, the other side will send back a tone to inform us, it is called “ring back tone”. The power of ring back tone will influence the correctness of judging the busy tone. The range is –50dBm to 0 dBm, and the default value is –27 dBm.

Settings for Console/Telnet only

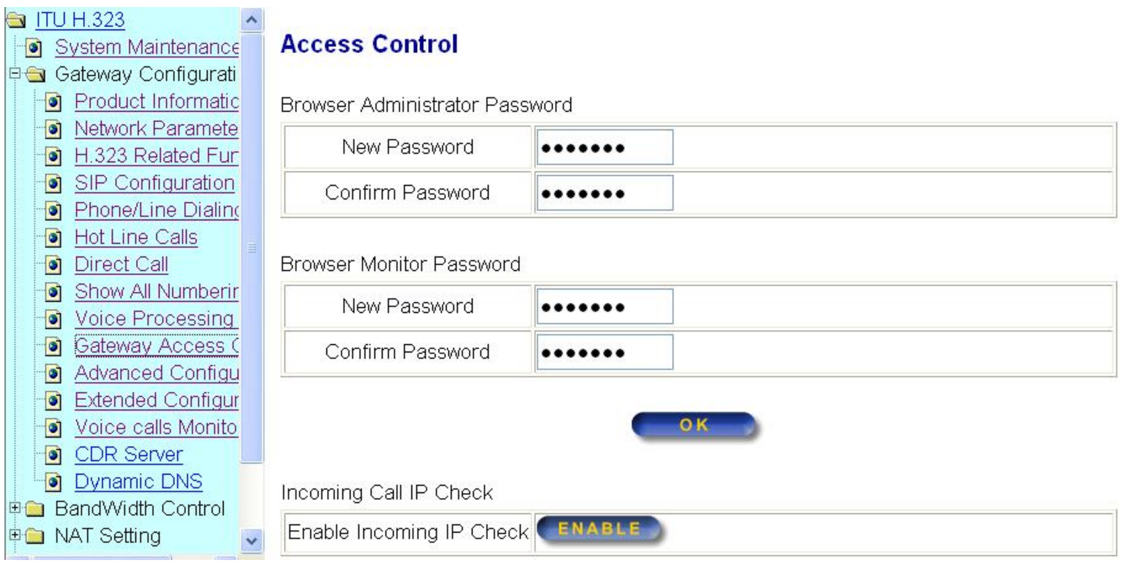
Voice processing Control.

Disconnect All Current Calls.

This function can let the system manager disconnect all active calls of the line ports. Only the calls in the line ports can be disconnected.

System Access Control

The configuration will let you change the password for web browsing and let the System deny some calls from specific IP addresses.



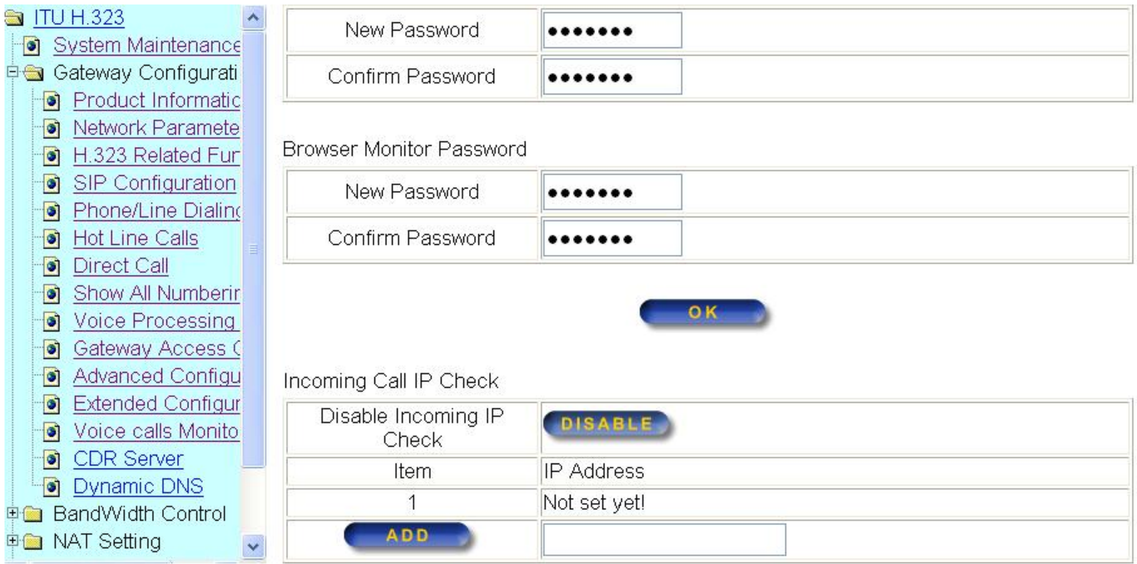
Steps:

- Click “System Access Control” in left-side menu.
- Enter the new password in both “New Password” and “Confirm Password” fields
- Press “OK” button.

When the “ENABLE” button of “Enable incoming IP Check” is clicked, you can add items for incoming IP check.

Notice:

When “Enable incoming IP Check” is enabled, only those IP addresses set in this table are allowed to call into this System. If this function is enabled and the table is empty, no callers are allowed to call in.



Steps:

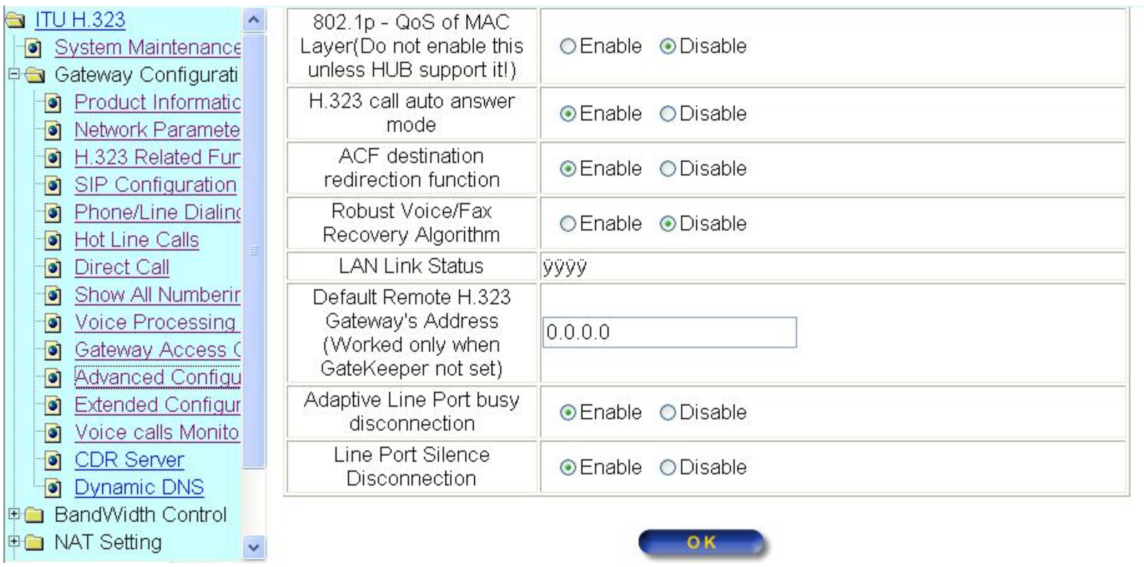
- Press “DISABLE” button again to disable this function.
- To add an item, enter the IP address you want to check.
- Press “OK” button to add it.

Settings for Console/Telnet only

System access Control.
Change the Console/Telnet login Password
Change the Console/Telnet user login Password.
Change the Browser login Password
Change the Browser use login Password
Choose to Disable Browser Function.
Choose to Disable Telnet Function.
Set Caller ID Number

Advanced Configurations

There are some advanced configurations about the System, and they can be set in this page. The following page will be displayed when the "Advanced Configuration" selection is clicked:



Steps:

- Click "Advanced Configuration" in left-side menu.

TOS - QoS of IP Layer:

This selection is to enable the IP layer “Type of Server” QoS function. This function can ensure voice quality if the network can support IP layer QoS.

802.1p - QoS of MAC Layer:

To enable this function, you must have this System connected to a switching hub supporting 802.1p. Do not enable this function if the hub/switch does not support 802.1p.

H.323 call auto answer mode:

When a VoIP call is coming in, the System will ring a specific phone set. The H.323 call signaling part could be connected or alerting during this ringing period. If this selection is enabled, the H.323 signaling part is connected during the ringing period. The benefit of this situation is that the remote side will hear the status of the specific port. That is, the remote side will hear ring back tone if the System is really ringing the phone set. If the phone set is busy, the remote side will hear busy tone. The disadvantage of this situation is that the H.323 connected time is not the real voice call connected time. So, if billing is recorded for this System, this function should be disabled.

ACF destination redirection function:

If this function is enabled, the Gatekeeper will translate any called number into any other number. This could give the gatekeeper more dialing plan flexibility.

Robust Voice/Fax Recovery Algorithm:

When this function is enabled, the quality of the voice/fax is boosted. But enabling this function will increase the bandwidth usage by about 4 kbps per channel.

LAN Link Status:

This selection is for displaying the LAN interface status.

Adaptive Line Port busy disconnection:

When a call has hung up from the line port connected to a PBX, the PBX will send a busy tone to the System. The System will automatically disconnect the call when receiving the busy tone. The function must be enabled.

Line Port Silence Disconnection:

If this function is enabled, a connected call will be disconnected if two-way silence is detected for a period of time (30 seconds) for the line port.

Settings for Console/Telnet only

Advanced Configurations.

Disable H.245 Fast Capability Exchange.
Please set this to the default value (enabled).

Run Line Port Busy Tone learning.

This function allows the system to learn a special busy tone that can not be learned by the system automatically, for example, busy tone with very long on/off duration(longer than 1 second). Normally, you do not need to do this.

Run Line Port Ringing Tone learning.

This function is provided for called party off-hook detection for the line port. When a VoIP call is incoming from another system and called through the line port to a phone set in the PSTN network, this system can know when the called phone set is off-hook by detecting the alerting (ringing) tone. The ringing tone learning can remember two kinds of ringing patterns and even “two cycle” ringing pattern.

This function can be used to generate correct billing records if the line port is connected to PSTN line. In order to let this function work correctly, the H.323 auto-answer mode must be disabled and the “Ringing tone detect duration” must be suitably set (about 15 seconds).

Enable the Line Port Power Dynamic reduction ability.

This function is designed to help reduce failed busy-tone detections caused by echoes. If enabled, this function will cause some uncomfortable voice quality. In most application, please disable this function.

FXO Answer Delay Time (Set Ringing Tone Detect duration.)

This function is provided for called party off-hook detection for the line port. When a VoIP call is incoming from other system and called through the line port to a phone set in the PSTN network, this system will know when the called phone set is off-hook by detecting the alerting (ringing) tone. This ring tone must first be learned by “9-c. run line port ringing tone learning”. This detect duration must be longer than the duration the PSTN network generate the ringing back tone.

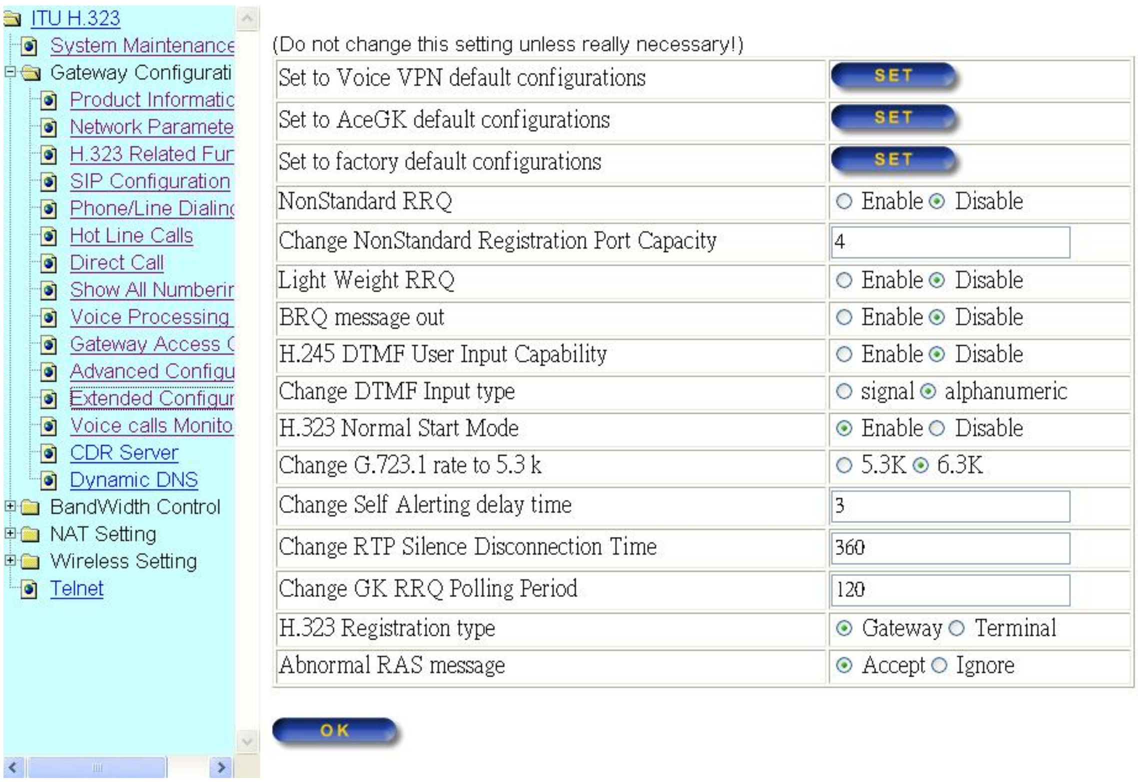
Line Port (FXO) Characteristics

The line ports in all the system series have the following characteristics:

1. Busy tone adaptive learning/disconnection (tone duration < 900 ms).
2. Busy tone manual learning/disconnection (tone duration < 3000 ms).
3. Alerting tone manual learning can learn one cyclic or two cyclic patterns.
4. Off hook detection:
    a. If the phone is off hook immediately, it will be detected after a "ring tone detection duration".
    b. If the phone is off hook after some alerting tone, it will be detected after a "ring cyclic" (max of [ring-on, ring-off]).
5. Alerting forever detection: If alerting duration exceeds 90 seconds, the FXO port will automatically disconnect it:
    a. Even if altering tone not learned.
    b. Even if the alerting pattern is one cyclic or two cyclic
6. If called side disconnected occurs before 13 seconds, the FXO busy tone disconnection would work after 13 seconds (to prevent dial tone like busy tone problem).

Extended Configurations

The System has some Extended configurations that can be set in this page. The following page will be shown when the “Extended Configuration” selection is clicked:



Steps:

- Click “Extended Configurations” in left-side menu.

NonStandard RRQ:

This selection is to enable the RRQ message containing nonstandard elements (Non Standard Identifier) for special interoperability purpose. Should be disabled for most cases.

Change NonStandard Registration Port Capacity:

This selection is to set the special “port capacity” for this system for special interoperability purposes. This setting has no effect if “NonStandard RRQ” is disabled.

Light Weight RRQ:

This selection allows the registration message (RRQ) to contain only necessary information for “keep alive” purposes; this could save network bandwidth. Default is disabled.

BRQ Message Out:

This selection enables the System to send out a BRQ (Bandwidth Request) message to the gatekeeper when the System is really connected on the voice channel. The gatekeeper could use BRQ message for billing purpose.

H.245 DTMF User Input Capability:

This selection enables the System to contain “DTMF capability” in the H.245 capability exchange message. This capability element is for special interoperability purposes with other system. Default is disabled.

Change DTMF Input Type:

The DTMF tone could be sent during a voice call by using the H.245 user-user message. This selection could further select the carrying method of using “signal” format or “Alphanumeric” format in this DTMF message. The default value is “Alphanumeric” format.

H.323 Normal Start Mode:

This selection could force the System to use normal start mode or fast start mode when establishing a VoIP call. Many other systems only support normal start mode, enable this selection when it is necessary. The default is disabled (using fast start mode).

Change G.723.1 rate to 5.3k:

This selection can force the G.723.1 CODEC to use 5.3k bps speeds instead of default 6.3k speeds.

Change Self Alerting delay time:

This setting is for the ring-back tone delay time when a voice call is proceeding. Too short delay could confuse the caller. The default is 3 seconds.

Change RTP Silence Disconnection Time:

This selection enables the System polling of the voice packets (RTP packets) to make sure that the connected two parties are still under normal operation. If one side is powered down during the conversation, the System will do a forced disconnection at the other side after this set time duration.

Change GK RRQ Polling Period:

The value is based on seconds (e.g. 120 means 120 seconds). The system will register to gatekeeper after a polling period of time. When the setting value is 120 then the system will register to gatekeeper every 120 seconds.

H.323 Registration type:

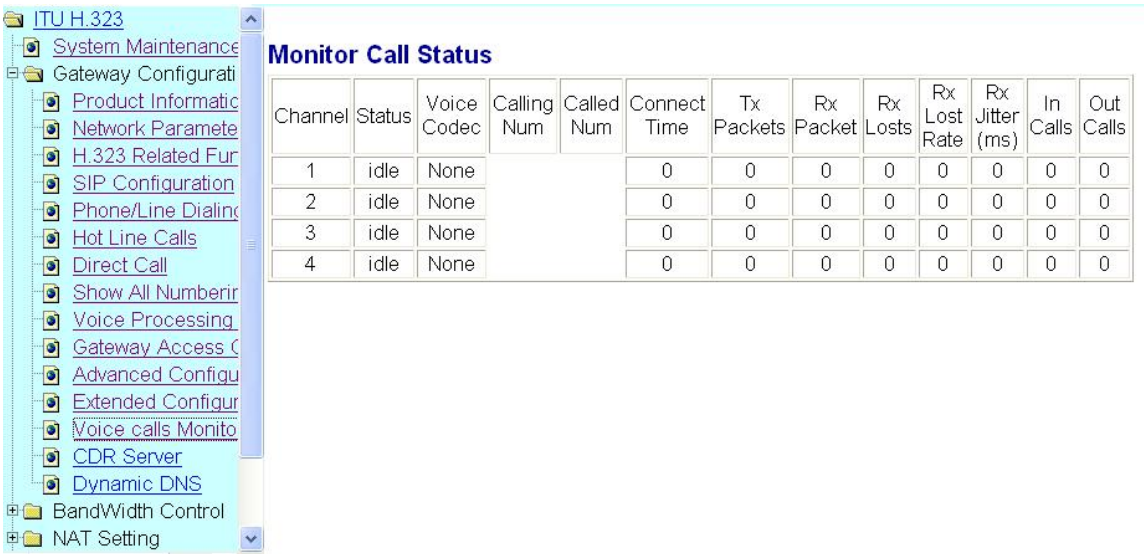
There are 2 choices for this setting. “System” means it will act as the VoIP system. “Terminal” means it will act as the IP phone terminal.

Abnormal RAS message:

When an RAS messages from gatekeeper is unrecognizable, the system can choose “Accept” or “Ignore” for responding to the abnormal RAS message.

Voice calls Monitoring

This selection will display concurrent call status of this System. This status is refreshed every 5 seconds.



Channel	Status	Voice Codec	Calling Num	Called Num	Connect Time	Tx Packets	Rx Packet	Rx Losts	Rx Lost Rate	Rx Jitter (ms)	In Calls	Out Calls
1	idle	None			0	0	0	0	0	0	0	0
2	idle	None			0	0	0	0	0	0	0	0
3	idle	None			0	0	0	0	0	0	0	0
4	idle	None			0	0	0	0	0	0	0	0

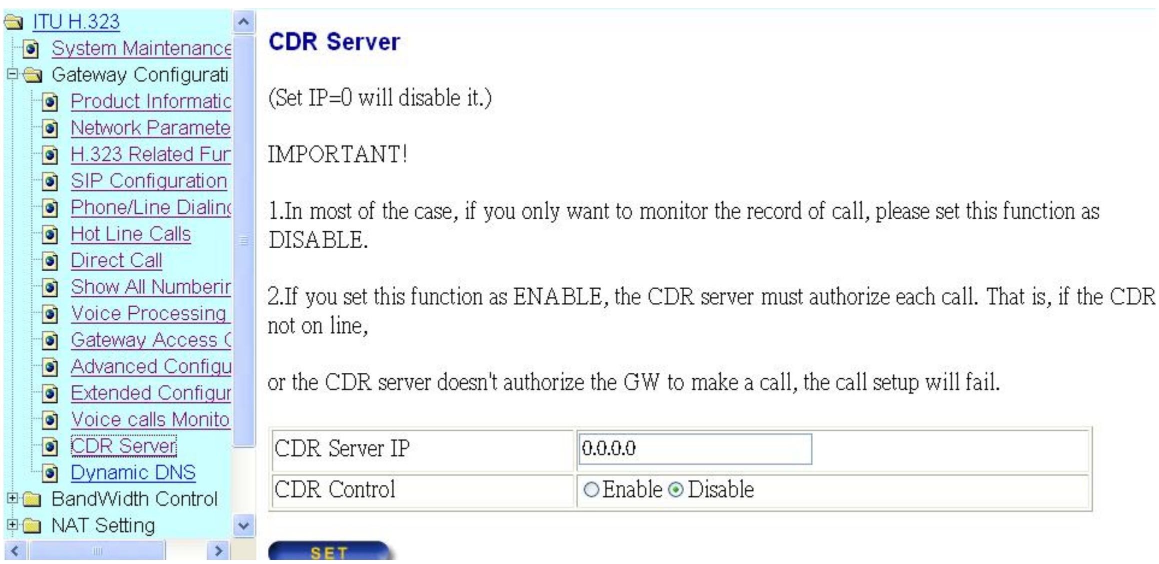
Steps:

- Click “Voice Call Monitoring” in left-side menu.

CDR Server

The System can generate call detail records of the VoIP calls and send them to a CDR server. The only thing what you should do is to set an IP address of CDR server. If the CDR server is not running, the operation of the System is not affected. The call detail record includes calling/called systems IP, calling/called telephone numbers and call duration, … etc. The CDR server is a Windows application freely provided by system.

CDR Control:
This system can be fully controlled by the CDR server if the "CDR Control" is enabled. If this is enabled, the system must connect to the CDR server and the CDR server must enable this system, otherwise, this system cannot make any calls. If the "CDR Control" is enabled, but no CDR server is connected, this system cannot make any calls.



PROGRAMMING THE DV SYSTEM

The only programming that may be required, is to provide DSS buttons for the VOIP "trunks" or to provide "LCR-trunk selection" so that the VOIP trunks are selected when the remote systems call numbers are dialed.

Providing DSS buttons

This is programmed in zone 500 for the particular groups used by the handsets. The buttons will be programmed as per a normal trunk or CO buttons.

e.g. For a DKP50 series handset, use either group 1 (no call appearance buttons) or group 2 (with call appearance buttons)

Providing LCR trunk selection

The system can be programmed to select a particular group of COs (trunks) depending on the digits dialed by the extension.

Zone 253 must be programmed to 03

Speed dial codes 150~199 can be programmed for the digits to be dialed by the extension when trunk selection is to be made.

Example

Zone 253 = 03

Zone 404 code 150 = 0011 and is set to use CO 02

If an extension dials 0011, the system will select a CO from the group 02 in zone 603 for the call. This will happen if the extension

a) dials 9 for a line
b) manually selects the line by pressing a CO button
c) selects a CO by using #4??

All other calls will select a CO from the normal CO group allocated to the extension.

50 codes may be input between the codes 150~199. The code can be up to 8 digits in length.

If the Speed Dial Code is set for 00 trunk selection, a CO from group 05 will be selected

DIGIT SET IN CO SELECT ITEM IN ZONE 404 CODES 150~199	GROUP SELECTED IN ZONE 603
00	05
01	01
02	02
03	03
04	04
05	05
06	06
07	07
08	08
09~12	05

NOTE!

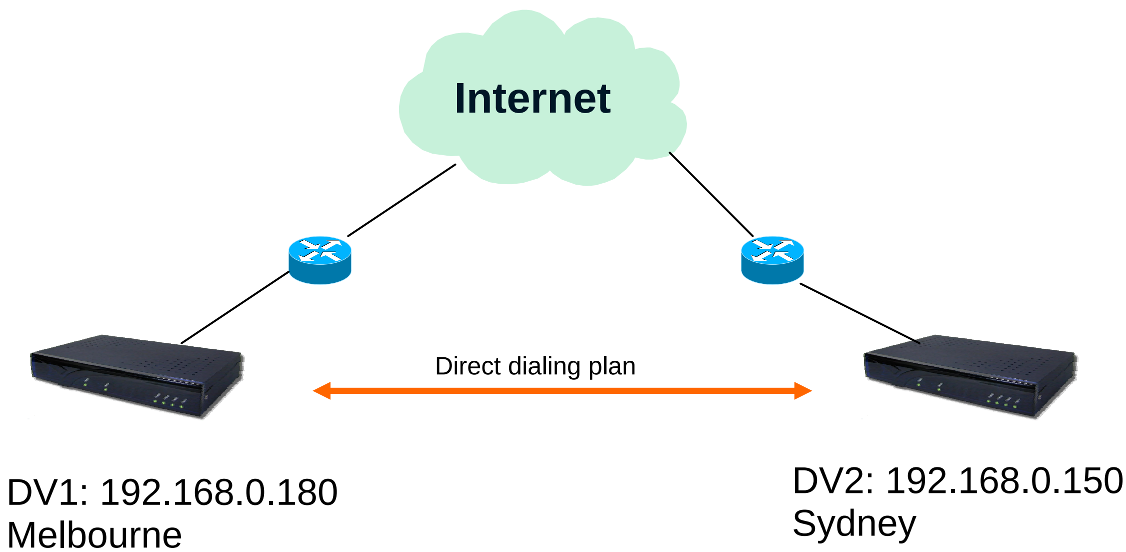
LCR SOFTWARE IS NOT REQUIRED FOR THIS FEATURE

Connecting to the remote end PSTN/ISDN network.

The trunk or trunks used by the VoIP can be answered by the DV system Auto-Attendant in the normal manner. The caller can then press the CO access code (usually 0 or 9) followed by the password (Z301, code 03 P-DISA) and the incoming VoIP trunk will then have access to unrestricted Telco dial tone at the remote end.

i.e. Calling from Australia to Taiwan. Answered by the Taiwan end AA and dialing (for example) 9 1234, will result in the Australian user receiving unrestricted Taiwan dial tone. One VoIP trunk can be programmed for AA working and the other three for normal working.

Gateway programming example



GW1 Melbourne 192.168.0.180
Select 1 to set WAN port network parameters

```
Login password :
--------------------------------------------------
-------  Aristel (1A)  Version 1.1.3 (5409)  -------
--------------------------------------------------
1. Set WAN Port network parameters.
2. Set H.323/SIP related functions.
3. Set Port's Dialing Number.
4. Set Hot Line Calls.
5. Set Direct Dialing Numbers.
6. Show All Numbering Plan.
7. Voice processing Control.
8. Gateway access Control.
9. Advanced Configurations.
a. Set to the Default Configurations.
b. Voice calls Monitoring.
c. Extended Configurations.
d. BandWidth Control/Shaping.
e. NAT Configurations.
f. Save Current Configurations to Rom.
g. Restart the System.
h. Logout.
--------------------------------------------------
  H323 Direct Mode.  NAT Function : ON
--------------------------------------------------
Key in 1 - h:1
```

Select 2 to set GK's IP address

```
Login password :
 ---------------------------------------------------
 -------  Aristel (1A)  Version 1.1.3 (5409)  -------
 ---------------------------------------------------
 1. Set WAN Port network parameters.
 2. Set H.323/SIP related functions.
 3. Set Port's Dialing Number.
 4. Set Hot Line Calls.
 5. Set Direct Dialing Numbers.
 6. Show All Numbering Plan.
 7. Voice processing Control.
 8. Gateway access Control.
 9. Advanced Configurations.
 a. Set to the Default Configurations.
 b. Voice calls Monitoring.
 c. Extended Configurations.
 d. BandWidth Control/Shaping.
 e. NAT Configurations.
 f. Save Current Configurations to Rom.
 g. Restart the System.
 h. Logout.
 ---------------------------------------------------
    H323 Direct Mode.  NAT Function : ON
 ---------------------------------------------------
 Key in 1 - h:2
```

```
----------------------------------------------------
Key in 1 - h:(Need to save and restart !) 2


---------------------  H.323 Related Parameter  ------------------------
Use VoIP Protocol          = H.323
The Gateway's H323-ID      =
The GateKeepr's address    = (1)0.0.0.0, (2)0.0.0.0
(** Set GateKeeper address = 0 will disable it. **)
Status : registered 61.219.20.6
The Gatekeeper's RAS port  = 1719
The Gateway's Q.931 port   = 1720
--------------------------------------------------------------------------

1. Set Default VoIP protocol.
2. Change The Gateway's H323-ID.
3. Change GateKeeper's IP address.
4. Change GateKeeper ID.
5. Change Prefix Address(Hunting).
6. Change Gatekeeper's RAS port.
7. Change Gateway's Q.931 port.

'ESC' to Upper Menu
Choose one item( 1 - 7 ) :
```

Select 3 to check that the port's number is blank

```
Login password :
---------------------------------------------------
-------  Aristel (1A)  Version 1.1.3 (5409)  -------
---------------------------------------------------
1. Set WAN Port network parameters.
2. Set H.323/SIP related functions.
3. Set Port's Dialing Number.
4. Set Hot Line Calls.
5. Set Direct Dialing Numbers.
6. Show All Numbering Plan.
7. Voice processing Control.
8. Gateway access Control.
9. Advanced Configurations.
a. Set to the Default Configurations.
b. Voice calls Monitoring.
c. Extended Configurations.
d. BandWidth Control/Shaping.
e. NAT Configurations.
f. Save Current Configurations to Rom.
g. Restart the System.
h. Logout.
---------------------------------------------------
   H323 Direct Mode.  NAT Function : ON
---------------------------------------------------
Key in 1 - h:3
```

FXO Port 1's Number FXO Port 2's Number

FXO Port 3's Number FXO Port 4's Number

'Esc' to Main Menu.
Choose one port (1-4) to set its number:

Select 5 to set Direct dialing numbers

```
Login password :
--------------------------------------------------
-------  Aristel (1A)  Version 1.1.3 (5409)  -------
--------------------------------------------------
1. Set WAN Port network parameters.
2. Set H.323/SIP related functions.
3. Set Port's Dialing Number.
4. Set Hot Line Calls.
5. Set Direct Dialing Numbers.
6. Show All Numbering Plan.
7. Voice processing Control.
8. Gateway access Control.
9. Advanced Configurations.
a. Set to the Default Configurations.
b. Voice calls Monitoring.
c. Extended Configurations.
d. BandWidth Control/Shaping.
e. NAT Configurations.
f. Save Current Configurations to Rom.
g. Restart the System.
h. Logout.
--------------------------------------------------
  H323 Direct Mode.  NAT Function : ON
--------------------------------------------------
Key in 1 - h:5
```

```
-------------------------------------------------
Key in 1 - h:(Need to save and restart !) 5

Query Priority : Gk First.
Current Direct Dialing Plans Are :
-------------------------------------------------------------------------
    Leading_No. Total_Len Strip_Len Prefix_No. Destination      GK-Regst
 1 : 12345678       8- 8            8         None        1,2,3,4            No
 2 : 87654321       8- 8            0         None        192.168.0.150
---------------------------------------------------------------------------

** There are no DDNS Dialing Mapping **

1. Add Direct Dial Plan.
2. Delete Direct Dial Plan.
3. Set Query Order.

'ESC' to Upper Menu
Choose one item( 1 - 3 ) :3
```

Select 3 to change the query priority to dial plan first

Key in 1 - h:5

Query Priority : Dial Plan First.
Current Direct Dialing Plans Are :

```
--------------------------------------------------------------------------
    Leading_No. Total_Len Strip_Len Prefix_No. Destination      GK-Regst
 1 : 12345678      8- 8          8         None       1,2,3,4             No
 2 : 87654321      8- 8          0         None       2 192.168.0.150
--------------------------------------------------------------------------
```

** There are no DDNS Dialing Mapping **

1. Add Direct Dial Plan.
2. Delete Direct Dial Plan.

'ESC' to Upper Menu
Choose one item(1 - 2) :

GW2 Sydney 192.168.0.150
Select 1 to set WAN port network parameters

```
Login password :
---------------------------------------------------
-------  Aristel (1A)  Version 1.1.3 (5409)  -------
---------------------------------------------------
1. Set WAN Port network parameters.
2. Set H.323/SIP related functions.
3. Set Port's Dialing Number.
4. Set Hot Line Calls.
5. Set Direct Dialing Numbers.
6. Show All Numbering Plan.
7. Voice processing Control.
8. Gateway access Control.
9. Advanced Configurations.
a. Set to the Default Configurations.
b. Voice calls Monitoring.
c. Extended Configurations.
d. BandWidth Control/Shaping.
e. NAT Configurations.
f. Save Current Configurations to Rom.
g. Restart the System.
h. Logout.
---------------------------------------------------
  H323 Direct Mode.  NAT Function : ON
---------------------------------------------------
Key in 1 - h:1
```

Select 1,2,3 to set network parameter

--------------- WAN Port Network Parameters -------------------------------
My IP = 192.168.0.150
Default Router IP = 192.168.0.01
Network Mask = 255.255.255.000
My MAC address = 0:4:f:aa:10:e
WAN Link Status = Auto-Negotiation Mode
CDR Server IP = 0.0.0.0
CDR Call Record Post Status = Enable
IP Server Address = 0.0.0.0
----

1. Change My IP address.
2. Change the Default Router IP address.
3. Change the Network Mask.
4. Enable DHCP Client.
5. Enable PPPoE Client.
6. Ping.
7. Pass Through NAT/Firewall.
8. Change CDR Server IP. (Set IP=0 will disable it.)
9. Disable CDR Call Record Post.

a. Change IP Server Address. (Set IP=0 will disable it.)
b. Dynamic DNS Configure.
c. Wan Port Speed Configure.

'ESC' to Upper Menu
Choose one item(1 - c) :**1,2,3**

Select 3 to check port's number it has to be blank

```
Login password :
 ---------------------------------------------------
 -------  Aristel (1A)  Version 1.1.3 (5409)  -------
 ---------------------------------------------------
 1. Set WAN Port network parameters.
 2. Set H.323/SIP related functions.
 3. Set Port's Dialing Number.
 4. Set Hot Line Calls.
 5. Set Direct Dialing Numbers.
 6. Show All Numbering Plan.
 7. Voice processing Control.
 8. Gateway access Control.
 9. Advanced Configurations.
 a. Set to the Default Configurations.
 b. Voice calls Monitoring.
 c. Extended Configurations.
 d. BandWidth Control/Shaping.
 e. NAT Configurations.
 f. Save Current Configurations to Rom.
 g. Restart the System.
 h. Logout.
 ---------------------------------------------------
    H323 Direct Mode.  NAT Function : ON
 ---------------------------------------------------
 Key in 1 - h:3
```

```
-----------------------------------------------------------------
  FXO Port  1's Number                     FXO Port  2's Number

  FXO Port  3's Number                     FXO Port  4's Number

-----------------------------------------------------------------
'Esc' to Main Menu.
Choose one port ( 1-4 ) to set its number:
```

Select 5 to set Direct dialing numbers

```
Login password :
--------------------------------------------------
-------  Aristel (1A)  Version 1.1.3 (5409)  -------
--------------------------------------------------
1. Set WAN Port network parameters.
2. Set H.323/SIP related functions.
3. Set Port's Dialing Number.
4. Set Hot Line Calls.
5. Set Direct Dialing Numbers.
6. Show All Numbering Plan.
7. Voice processing Control.
8. Gateway access Control.
9. Advanced Configurations.
a. Set to the Default Configurations.
b. Voice calls Monitoring.
c. Extended Configurations.
d. BandWidth Control/Shaping.
e. NAT Configurations.
f. Save Current Configurations to Rom.
g. Restart the System.
h. Logout.
--------------------------------------------------
  H323 Direct Mode.  NAT Function : ON
--------------------------------------------------
Key in 1 - h:5
```

```
-------------------------------------------------------
Key in 1 - h:5

Query Priority : Dial Plan First.
Current Direct Dialing Plans Are :
---------------------------------------------------------------------------------
    Leading_No. Total_Len Strip_Len Prefix_No. Destination
GK-Regst
 1 : 87654321      8- 8              8      None            1,2,3,4
No
 2 : 12345678      8- 8              0      None       192.168.0.180
---------------------------------------------------------------------------------

** There are no DDNS Dialing Mapping **

1. Add Direct Dial Plan.
2. Delete Direct Dial Plan.

'ESC' to Upper Menu
Choose one item( 1 - 2 ) :
```

Don’t forget to save the program after the changes

f. Save Current Configurations to Rom.
g. Restart the System.