

1-port H.323/SIP E1/T1 Trunk Gateway

VIP-2100



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Chapter 1 VIP-2100 Introduction

System Description

VIP-2100 is a cost effective solution for VoIP trunk gateway supporting one-, port T1/E1 VoIP trunks that provides voice and fax over IP network. It supports ITU-T H.323 V3, SIP RFC 2543/3261, SNMP V2, Call Detail Record, WEB management and other useful functions to meet customer requirements.

The built-in enhanced IVR (Interactive Voice Response) and Billing Service of VIP-2100 is suitable for prepaid and postpaid service. It can rapidly provide value added service for customers.

VIP-2100 Features:

- Dual SIP/H.323 co-existing
- ITU-T H.323 v3 and H.450 compliance
- SIP RFC 2543/3261 standard compliance
- PSTN signaling: ISDN/PRI, CAS (MFC R2, MFC R1, E&M), **QSIQ**
- Mixed SIP, Gatekeeper and P2P calls
- Support H.323 Gatekeeper register, direct and route calls
- Support SIP outbound proxy, redirect and register server
- Redundant SIP Proxy/Outbound Proxy Server Support (Outbound Active/Active fail over, Register A/A no fail over)
- Support SIP Overload Redirect
- SIP supplemental service - on Hold, Call Transfer (Transferred)
- Built-in phone book and prefix routing for SIP and H.323 P2P calls
- Support H.323 fast connect, early H.245 and H.245 tunneling
- Support H.323 and SIP early media
- VoIP to VoIP calls support – SIP to H.323, SIP to SIP, H.323 to H.323
- Global Trunk-Channel Block out: 0xffffffff (busy block out)
- Intelligent PSTN call routing and in-trunk hunting: reverse rotary, channel mask (default:0xffffffff), ANI prefix match
- **Reset a channel/trunk on the fly**
- Flexible digit manipulation plan
- Support RADIUS Authentication, Authorization and Accounting
- Support access control by ANI, DNIS, IP, Gatekeeper only, proxy only or RADIUS
- SIP UDP/TCP support
- Behind NAT friendly for SIP calls
- Inbound and out of band DTMF transmission
- SIP/H.323 T.38 fax relay up to 14400 bps
- Dynamic call treatment based on DNIS, ANI or collected DTMF
- Grouping DNIS/ANI Number Replacement
- Built-in IVR & call-flow controller for PSTN / VoIP side
- CISCO compatible
- Web-based graphic announcement edit and management

-
- Multiple configuration saving
 - Provides CDR (Call Detail Record)
 - Built-in internal user authentication for prepaid & postpaid users

Technical Specification

Interface

- Two 10/100MB Ethernet Ports (Host & VoIP stream)
- 1 xT1/E1 (**120 Ohm-RJ48C connectors**)
75 Ohm needs external 3rd party BNC/RJ-48C adapter cables

Protocol and Standard

- ITUT H.323 v3 and H.450 compliance
- SIP RFC 2543/3261 compliance

Audio Feature

- Codec -- G.711 A/ μ -Law, G.723.1 (5.3K/6.3K), G.729A, G.729
- Support G.168 echo cancellation
- Configurable audio payload size & adaptive jitter buffer
- Support silent suppression for G.729A, G.723, G.729
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generation)

DTMF Transmission

- Transparent
- H.245 signal/alphanumeric
- H.323 Q.931
- RFC 2833
- SIP INFO

FAX Support

- Automatic voice/fax detection
- T.38 fax relay based on H.323 Annex D
- SIP T.38 fax relay
- Up to G3 fax
- ECM support
- Redundant T.38 packet (0-2)
- CISCO compatible

Built-in IVR & call-flow controller

- Web-based GUI Drag and Drop interface
- Full control of call behavior (one-stage or two-stage dialing)
- IVR functions
- Support time duration play back (Chinese & English)
- Power call information branch
- Collected information validation
- Active disconnect & reconnect without hang up

-
- Selected disconnect cause code & behavior

Management Feature

- OS and program upgradeable
- Console port: RS-232 port
- TELNET
- Full Web management interface & real time monitor
- Front panel LCD
- SNMP v2 (H.341) and SNTP v4 support
- User account management
- Time zone and day light saving support
- Support fixed IP and DHCP
- Support DNS and dynamic DNS

LED indicators for system status

- Power/Storage access indicators
- Front panel LCD (2^{lines} x 16) status display

Power

- 90~240V auto switch

Environmental

- Operation temp: 0° C to 60° C
- Relative humidity: 5% to 95%

Dimension

- 483mm (L) x 450 mm (W) x 44mm(H)

Certification

- CE, FCC, EMI

VIP-2100 Detail Specifications

| Feature | | VIP-2100 |
|---------------------------------|------------------------|---|
| Physical Dimension | | |
| 1 | Width | 483mm |
| 2 | Height | 44mm |
| 3 | Depth | 450mm |
| 4 | Industrial rack mount | Yes |
| 5 | Color | Black |
| 6 | Weight | 8Kg |
| Power / Environmental | | |
| 1 | Power | 90-240V auto switch |
| 2 | Operating temperature | 0~60 C |
| 3 | Relative humidity | 5%~95% |
| Processors & Storage | | |
| 1 | DSP vendor | Intel Pentium, AudioCodes DSP |
| 2 | Operation System | XP Embedded |
| 3 | RAM | 512 MB |
| 4 | Program/Data Storage | 256 MB DOM |
| 5 | OS Upgradeable | Yes |
| 7 | Program Upgradeable | Yes |
| Front Panel Display | | |
| 1 | LED status | Power/DOM/System |
| 2 | LCD status | Yes |
| LAN Interface | | |
| 1 | 10/100 Base Ethernet | 10/100MB Ethernet ports *2 (host & RTP) |
| 2 | IP Address Required | 2 |
| PSTN Interface | | |
| 1 | Customizable E1/T1 CAS | Yes |
| 2 | E1 CAS DTMF | Loop Start FXO Hot-Line |
| 3 | E1 CAS R2 MF | Argentina, Bolivia, Brazil, Chile, China, Czech-Republic, Egypt, India, Indonesia, Israel, ITU, Korea, Malaysia, Mexico, Philippines, Thailand, Uruguay, Venezuela, RomTelcom |
| 4 | E1 ISDN PRI Support | Euro, Australia, Hong Kong, Korea, New Zealand, QSIC |
| 5 | E1/T1 Interface | Selectable |
| 6 | PCM law Support | Alaw/Mulaw selectable |

| | | |
|--|-------------------------------------|--|
| 7 | T1 CAS DTMF/R1MF | E&M Bell Core Feature Group D, Wink Start, E&M Delay Start, E&M Feature Group A Immediate Start, E&M Feature Group B Wink Start, E&M Feature Group D Wink Start(ANI B4 ADDR), E&M Feature Group D Wink Start, E&M Immediate Start, E&M Wink Start, GroundStart FXO, GroundStart FXS, Loop Start FXO, Loop Start FXS, Loop Start FXO Hot-Line |
| 8 | T1 ISDN PRI Support | NI2 ISDN, 5ESS 10 ISDN, DMS100 ISDN, NTT ISDN (INS-1500), Hong Kong, QSIC |
| 9 | Trunk Spans | 1 (T1/E1s) per chassis |
| 10 | Default Trunk Channel Mask | Yes |
| 11 | PSTN Line Hunting | Yes |
| 12 | PSTN Line Hunting Channel Selection | Yes |
| 13 | On the Fly Reset Channel/Trunk | Yes |
| Audio Codec Support | | |
| 1 | G.711 A-law | Yes |
| 2 | G.711 u-law | Yes |
| 3 | G.723.1 | Yes (5.3/6.3K) |
| 4 | G.729A | Yes |
| 5 | Selectable Payload Size - G.711 | 20, 40, 60 ms |
| 6 | Selectable Payload Size - G.723 | 30, 60, 90 ms |
| 7 | Selectable Payload Size - G.729 | 20, 40, 60 ms |
| Fax Transmission | | |
| 1 | Bypass mode | Yes |
| 2 | CISCO Compatible | Yes |
| 3 | ECM Support | Yes |
| 4 | FAX auto-detection | Yes |
| 5 | H.323 Annex D Support | Yes |
| 6 | SIP- T.38 Reinvite | Yes |
| 7 | T.38 During fast connect | Yes |
| 8 | T.38 Redundant Packet | 0-2 |
| 9 | Transparent mode | Yes |
| 10 | Up to G3 FAX | Yes (up to 14400 bps) |
| DTMF Transmission | | |
| 1 | RFC 2833 | Yes |
| 2 | H.245 Alphanumeric mode | Yes |
| 3 | H.245 Signal mode | Yes |
| 4 | Q.931 UUI | Yes |
| 5 | SIP INFO | Yes |
| 6 | Transparent mode | Yes |
| Voice Quality & Echo Cancellation | | |
| 1 | Adaptive Jitter Buffer | Yes |
| 2 | CNG | Yes |

| | | |
|-------------------------------|---------------------------------|--|
| 3 | G.168 (Echo Cancellation) | Yes (32ms) |
| 4 | Gain Control | Yes |
| 5 | Improved Echo Tail Suppression | Yes |
| 6 | Silence Suppression | Yes |
| 7 | VAD | Yes |
| Maintenance | | |
| 1 | Administrative Log | Yes |
| 2 | Auto Daylight Saving | Yes |
| 3 | Customizable Time Zone | Yes |
| 4 | Front Panel LCD Setup | Yes |
| 5 | FTP Server | Yes |
| 6 | HTTP server | Yes |
| 7 | HTTP SSL support | Yes |
| 8 | Multiple configuration | Yes |
| 9 | NTP time synchronization | Yes (SNTP V4) |
| 10 | Password Security | Yes |
| 11 | RS232 | Yes |
| 12 | System Event Log | Yes |
| 13 | Telnet | Yes |
| 14 | Time Zone Support | Yes |
| 15 | User Account Manager | Yes |
| 16 | Web-based GUI | Yes |
| 17 | Web-based Real Time Monitor | Yes |
| 18 | Web-based Voice File Management | Yes |
| Network Management | | |
| 1 | DHCP | Yes |
| 2 | Fixed IP | Yes |
| 3 | DNS | Yes |
| 4 | Dynamic DNS | Yes |
| 5 | Ping | Yes |
| 6 | TOS field setting | Yes (RTP only) |
| 7 | SNMP V2 MIB I & II | Yes |
| 8 | SNMP get command | Yes |
| 9 | SNMP set command | Yes |
| 10 | SNMP Trap | Yes |
| 11 | H.341 MIB Support | Yes |
| 12 | SysLog Support | Yes |
| H.323 Protocol Support | | |
| 1 | H.323 V3 | Yes |
| 2 | H.323 ID | Yes |
| 3 | E.164 ID | Yes |
| 4 | Fast Connect | Yes (selectable for incoming/outgoing) |
| 5 | H.450 | Yes |
| 6 | H.245 Tunneling | Yes |
| 7 | Early H.245 | Yes |
| 8 | Cause Code Mapping | Yes |

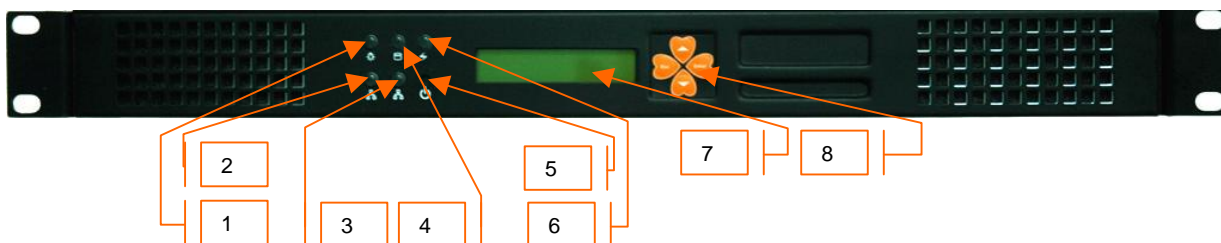
| SIP Protocol Support | | |
|---------------------------------|------------------------------|------------------------------|
| 1 | Cause Code Mapping | Yes |
| 2 | HTTP Digest Authentication | Yes |
| 3 | SIP Call on Hold | Yes |
| 4 | SIP Early Media | Yes |
| 5 | SIP Overload Redirect | Yes |
| 6 | SIP Transfer (unattend) | Yes |
| 7 | SIP Transfer (attend) | Yes |
| 8 | SIP/TCP | Yes |
| 9 | SIP/UDP | Yes |
| 10 | SIP-180/SDP | Yes |
| 11 | SIP-183/SDP | Yes |
| 12 | SIP-PRACK | Yes |
| 13 | SIP-RFC 3261 | Yes |
| 14 | SIP-RFC 3264 (Offer/Answer) | Yes |
| H.323 Gatekeeper Support | | |
| 1 | Gatekeeper Register | Yes |
| 2 | Direct call | Yes |
| 3 | Routed call | Yes |
| 4 | Light weight RRQ | Yes |
| 5 | IRQ: IRR sequence | Yes |
| 6 | Gatekeeper Call only | Yes |
| SIP Proxy Sever Support | | |
| 1 | SIP Outbound Proxy Support | Yes |
| 2 | SIP Redirect Server Support | Yes |
| 3 | SIP Registrar Server Support | Yes |
| 4 | Redundant SIP Proxy Server | Yes |
| 5 | Auto Fail Over | Yes |
| Dial Plan | | |
| 1 | P2P H.323/SIP Call | Yes |
| 2 | GK Call | Yes |
| 3 | SIP Call | Yes |
| 4 | PSTN Call | Yes |
| 5 | Mixed SIP, P2P, GK call | Yes |
| 6 | Build-in Phone Book | Yes |
| 7 | P2P Prefix Routing | Yes |
| 8 | Digits Manipulation | Yes |
| 9 | ISDN Dial Plan by Prefix | Yes (Source & Destination) |
| Call Type Support | | |
| 1 | Call Decision | Dynamic Decided by Call Flow |
| 2 | H.323 to H.323 Call | Yes |
| 3 | H.323 to H.323 Fax Realy | Yes |
| 4 | H.323 to PSTN Call | Yes |
| 5 | H.323 to SIP Call | Yes |
| 6 | H.323 to SIP FAX Relay | Yes |

| | | |
|------------------------|---------------------------------|--|
| 7 | H.323 to SIP FAX Relay | Yes |
| 7 | PSTN to H.323 Call | Yes |
| 8 | PSTN to PSTN Call | Yes |
| 9 | PSTN to SIP Call | Yes |
| 10 | SIP to H.323 Call | Yes |
| 11 | SIP to PSTN Call | Yes |
| 12 | SIP to SIP Call | Yes |
| 13 | SIP to SIP Fax Relay | Yes |
| 14 | VoIP to VoIP RTP unRouted | Yes |
| 15 | VoIP to VoIP RTP Routed | Yes |
| Enhance Service | | |
| 1 | ANI Access List | Yes |
| 2 | DNIS Access List | Yes |
| 3 | DID/DOD | Yes |
| 4 | PSTN Two Stage Dialing | Yes |
| 5 | VoIP Two Stage Dialing | Yes |
| 6 | Intelligent PSTN Call Routing | Yes (Random, Round Robin, Priority) |
| 7 | In-trunk hunting method | Cyclic, random, rotary, reverse cyclic, reverse rotary |
| 8 | Ring Back Tone Generation | Yes (per trunk enable/disable) |
| 9 | Call Progress Tone Support | Yes |
| 10 | Web-based Call Flow GUI | Drag and Drop interface, Full control of call behavior (one-stage or two-stage dialing), IVR functions, Support time duration play back (Chinese & English), Power call information branch, Collected information validation, Active disconnect & reconnect without hang up, Selected disconnect cause code & behavior |
| 11 | Play Credit Time Duration | Yes (Chinese & English) |
| 12 | Play Credit Balance | Yes (Chinese & English) |
| 13 | Almost-time-expired notify tone | Yes |
| 14 | IVR for PSTN | Yes |
| 15 | IVR for SIP | Yes |
| 16 | IVR for H.323 | Yes |
| 17 | IP Access List | Yes |
| 18 | ANI Replacement | Yes |
| 19 | DNSI Replacement | Yes |
| AAA | | |
| 1 | Call detail record (CDR) | Yes |
| 2 | RADIUS Authentication | Yes |
| 2 | RADIUS Authorization | Yes |
| 3 | RADIUS Accounting | Yes |
| 4 | Redundant RADIUS Server Support | Yes, Active/Standby/Auto Failover |
| 5 | PSTN Prepaid Support | Yes |

| | | |
|--------------------------|---------------------------|----------|
| 6 | VoIP Prepaid Support | Yes |
| Embedded AAA | | |
| 1 | Embedded Prepaid Service | Yes |
| 2 | Embedded Postpaid Service | Yes |
| 3 | Point/second Calculation | Yes |
| 4 | Second/point calculation | Yes |
| 5 | Auto Disable/Clean User | Yes |
| 6 | PSTN Prepaid Support | Yes |
| 7 | VoIP Prepaid Support | Yes |
| System Limitation | | |
| 1 | Max DM | 4096 |
| 2 | Max IP ACL | 2048 |
| 3 | Max DNIS ACL | 4096 |
| 4 | Max ANI ACL | 4096 |
| 5 | Max User ACL | 20000 |
| 6 | Max Phone Book Entries | 10000 |
| 7 | Max Call Flow Component | 256 |
| 8 | Max CDR Keep Days | 5 |
| 9 | Max Voice File Storage | 10 hours |
| Manual | | |
| 1 | English User Guide | Yes |

VIP-2100 Appearance Description

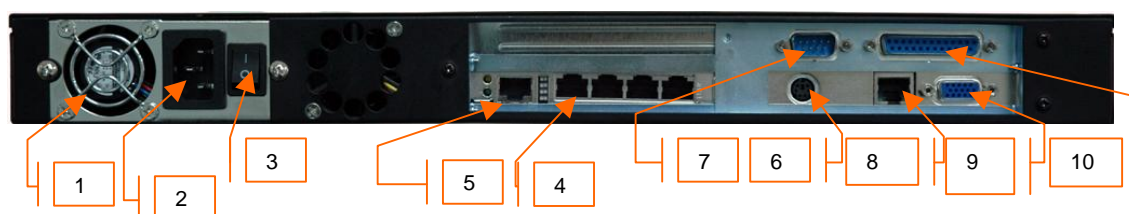
VIP-2100 Front Panel:



Functions:

- 1: Power LED
- 2: Network1 Interface LED
- 3: Network2 Interface LED (not used)
- 4: H/D LCD
- 5: Power Switch
- 6: System Status LED
- 7: LCD Panel
- 8: LCD Touch Panel

VIP-2100 Rear Panel:



Functions:

- 1: Electric Fan
- 2: AC Power outlet
- 3: AC Power switch **(Keep on)**
- 4: Trunk E1/T1 port
- 5: VoIP Ethernet port
- 6: Keyboard/Mouse
- 7: Com1 port
- 8: Ethernet port
- 9: VGA
- 10: print port (not available)

Chapter 2 Logon VIP-2100

After connected E1/T1 & Ethernet cables into the VIP-2100, turned on the power. The first step is to logon the system and set up the IP address.

Before you can use the Browser to setup VIP-2100, you need to have Java Standard Runtime (1_4_1_02) to make it work. Please refer to [Appendix 2 Java plug-in Install](#) for detail.

Logon VIP-2100

Setp1: Start IE5.0 (or later version) to navigate VIP-2100 Management System by typing the default IP address (the default URL is <http://192.168.111.111:10087>). The screen will display **User ID** and **Password** as figure 2.1-1.



The screenshot shows a web browser window with a login form. The form has two input fields: "User ID" with the text "root" entered, and "Password" with "****" entered. Below the fields are two buttons: "Login" and "Cancel".

Figure 2.1-1

● **Note:** The default network IP address is 192.168.111.111 and subnet mask is 255.255.0.0

Step 2: Enter log user name and password (the default user id is root and user password is root). You can manage your user account via web (refer to Section "[Account Manager](#)") later.



The screenshot shows a web browser window with a login form. The form has two input fields: "User ID" with the text "wellgate" entered, and "Password" with "*****" entered. Below the fields are two buttons: "Login" and "Cancel".

Figure 2.1-2

Step 3: The screen shows the Home Page of VIP-2100 as figure 2.1-3.



Figure 2.1-3

Network Configuration

Step 1: After successfully logon to the system, we need to change the network configuration. Click **Control**→**Network** to setup the network parameters as figure 2.2-1.

Network Control

Use DHCP
 Use fixed IP address

| | |
|--------------|----------------|
| IP Address : | 192.168.19.173 |
| IP Netmask : | 255.255.255.0 |
| IP Gateway : | 192.168.19.254 |

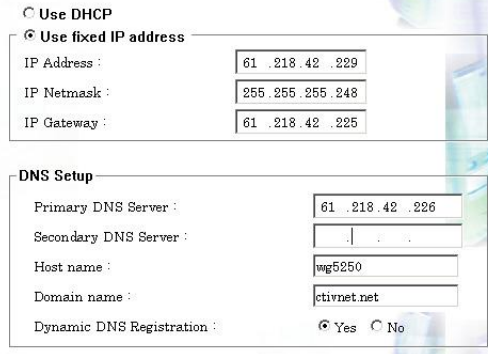
DNS Setup

| | |
|----------------------------|---|
| Primary DNS Server : | 61.218.42.226 |
| Secondary DNS Server : | |
| Host name : | wg5250 |
| Domain name : | ctivnet.net |
| Dynamic DNS Registration : | <input checked="" type="radio"/> Yes <input type="radio"/> No |

Figure 2.2-1

Step 2: Enter the deserved IP address, Submask and default gateway. Apply the change by clicking **apply** button as figure 2.2-2.

Network Control



The screenshot shows a configuration page titled "Network Control". It has two radio buttons: "Use DHCP" (unselected) and "Use fixed IP address" (selected). Below the radio buttons are three input fields: "IP Address" with the value "61 . 218 . 42 . 229", "IP Netmask" with "255 . 255 . 255 . 248", and "IP Gateway" with "61 . 218 . 42 . 225". Below these is a "DNS Setup" section with four input fields: "Primary DNS Server" with "61 . 218 . 42 . 226", "Secondary DNS Server" (empty), "Host name" with "wg5250", and "Domain name" with "ctivnet.net". At the bottom, there is a "Dynamic DNS Registration" section with "Yes" selected and "No" unselected.

Figure 2.2-2

Step 3: When screen shows “**Setup network configuration successfully!**” It means the IP Network setting is successfully changed as figure 2.2-3.



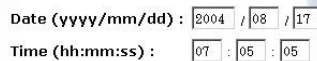
Figure 2.2-3

● **Note:** “*Network Control*” takes around 5-second to apply the new network configuration. Please logon again with new IP address after 5 seconds.

System Time Configuration

Step 1: When re-logon to the new IP address; the next is to setup the system time zone. Click **Control**→**System Time Zone** to setup the system as figure 2.3-1.

System Time Configuration



The screenshot shows a configuration page titled "System Time Configuration". It has two input fields: "Date (yyyy/mm/dd)" with the value "2004 / 08 / 17" and "Time (hh:mm:ss)" with "07 : 05 : 05".

Figure 2.3-1

Step 2: After apply the new time zone, click **Back** to adjust the date and time as figure 2.3-2.

System Time Configuration

Date (yyyy/mm/dd) : 2004 / 08 / 17
Time (hh:mm:ss) : 15 : 00 : 05

Figure 2.3-2

Step 3: Enter current date and time. Apply the change by clicking **Apply** button as figure 2.3-3.

System Time Configuration

Date (yyyy/mm/dd) : 2004 / 08 / 17
Time (hh:mm:ss) : 15 : 00 : 05

Figure 2.3-3

Step 4: The screen will show **“Setup system time successfully!”** It means the **System Time** setting is successfully changed as figure 2.3-4.



Figure 2.3-4

Step 5: If you would like to use SNTP to sync time with a SNTP V4 Server, click **Time Sync** button to setup it as figure 2.3-5.

Time Synchronization Control

Disable
 Enable

Primary SNTP Server : 210.59.163.133
Secondary SNTP Server : 61.220.126.28
Polling Interval (second) : 3600
Retry Interval (second) : 3600
Max Fail Retry : 3

Apply Back

Figure 2.3-5

Account Manager

Step 1: You can manage your user account by click **Control**→**Account Manager**. Add a new user account, Click **New** button as figure 2.4-1.

Account Management

| User ID | Password |
|---------|----------|
| admin | ***** |
| root | ***** |

Figure 2.4-1

Step 2: Enter the new user ID, password, user role and description, as you need. Apply the change as figure 2.4-2.



Figure 2.4-2

Field Description:

- User ID: Login User ID
- Password: Login Password
- Confirm Password: Confirm new password again

Step 3: When screen shows ***“Create user account successfully!”*** It means user account setting is successfully created as figure 2.4-3



Figure 2.4-3

- **Note: The system provides 2 USER ID by default:**
User 1: “root” Password: “root”
User 2: “admin” Password: “admin”

Relogin

Step 1: Click **Control**→**Relogin** to relogin by another user account as figure 2.5-1.

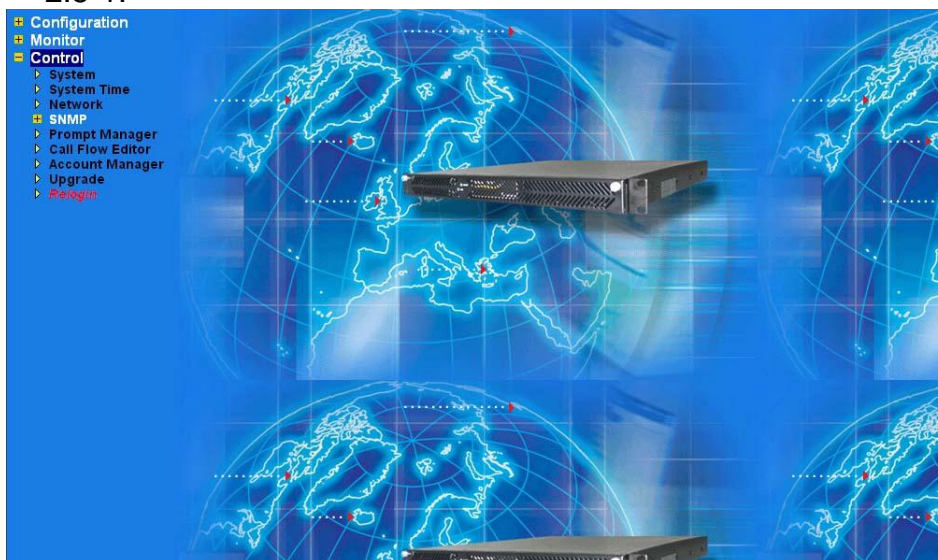


Figure 2.5-1

Step 2: Enter new User ID and Password to relogin the VIP-2100 as figure 2.5-2.



Figure 2.5-2

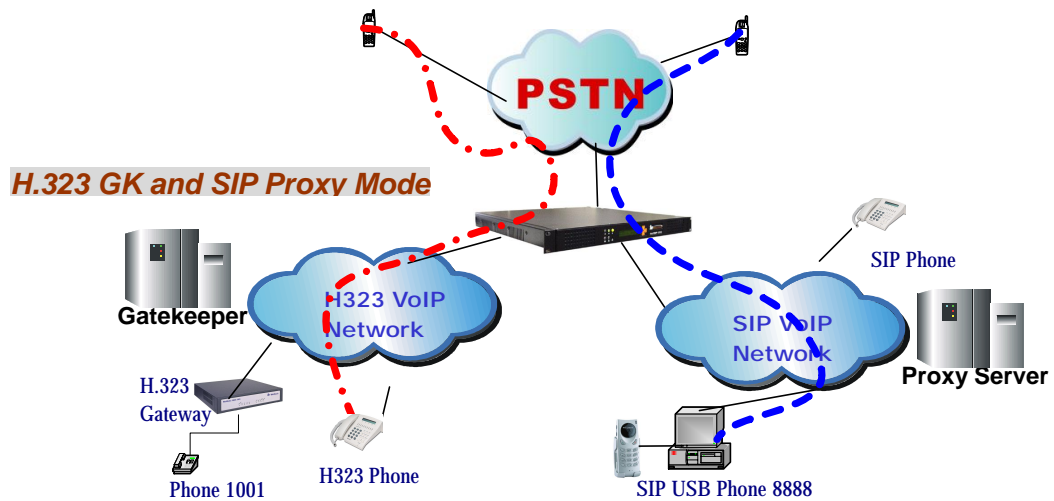
Step 3: The screen shows the Home Page of VIP-2100 as figure 2-5-3.



Figure 2.5-3

Chapter 3 H.323 Gatekeeper and SIP Proxy Mode Configuration

Environment used in this chapter



Process:

PSTN → H.323 Call: DNIS (1001) → Make H.323 - Gatekeeper Call (1001)
→ SIP Call: DNIS (8888) → Make SIP – SIP Proxy Call (8888)

H.323 → DNIS (5932111222) → DM (H.323_in_drop) → Make Call
(0932111222)

SIP → DNIS (11382265699) → DM (SIP_in_drop) → Make Call
(82265699)

Interface Configuration

This section is going to setup the VoIP interface.

Step 1: Now we are going to setup the VoIP interface, click **Configuration**→**Interface** to setup VoIP T1/E1 interface as figure 3.1-1.

Interface Configuration

| Interface ID | Interface Type | Description |
|--------------|----------------|----------------------|
| 0 | 4 E1/T1 | VoIP 4E1/T1 Interfac |

[1]



Figure 3.1-1

Step 2: Double-click the installed interface (i.e Interface ID:0) to config it as figure 3.1-2.

Interface Configuration

| Interface ID | Interface Type | Description |
|--------------|----------------|----------------------|
| 0 | 4 E1/T1 | VoIP 4E1/T1 Interfac |

[1]



Figure 3.1-2

Step 3: Modify the VoIP Interface parameters (i.e. IP Address, Protocol Tag, Subnet Mask and Default gateway) and apply the change by clicking **Apply** as figure 3.1-3.

Interface Configuration

| | | | |
|-------------------|----------------------------------|-------------|---|
| Interface ID : | 0 | Card Slot : | 3 |
| Interface Type : | 4 E1/T1 | | |
| Description : | VoIP 4 E1/T1 Interface 1 | | |
| Serial No : | 242807 | | |
| License Key : | 020519d73e655d430901e35ef2dd1497 | | |
| IP Address : | 192.168.19.174 | | |
| Subnet Mask : | 255.255.255.0 | | |
| Default Gateway : | 192.168.19.254 | | |
| PCM Type : | A-law | | |

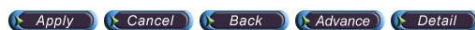


Figure 3.1-3

Frequency changed parameters: (Refer to section “[Interface Configuration](#)” for more detail)

- IP Address: 192.168.19.174
- Subnet Mask: 255.255.255.0
- Default Gateway: 192.168.19.254
- PCM Type: A-law or Mulaw

● **Caution: Subnet Mask does not support Supernet.**

Step 4: After successfully to change the Interface configuration, the screen come back the page of **Interface Configuration** as figure 3.1-4.



Figure 3.1-4

T1/E1 Trunk Configuration

This section is going to setup the PSTN trunk parameters.

Step 1: Select the installed interface to modify the trunk parameter by click **Detail** button as figure 3.2-1.

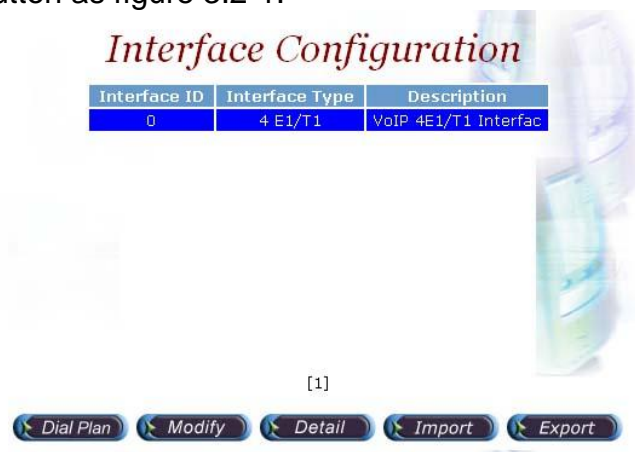


Figure 3.2-1

Step 2: Select the trunk to be modified, and click **Modify** button as figure 3.2-2.

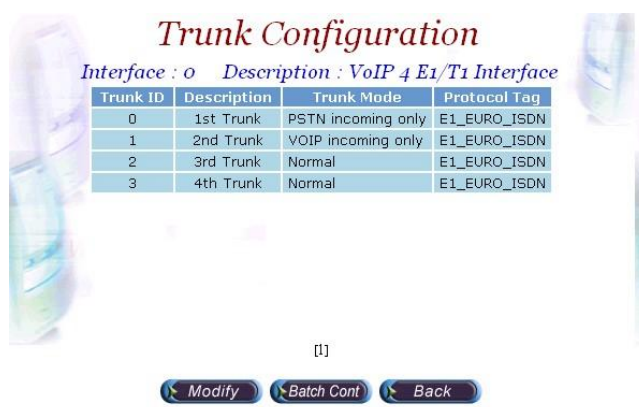


Figure 3.2-2

Step 3: Modify the trunk parameters (i.e. Trunk Type, Termin Side, Trunk Mode, Protocol Tag, Line Code) and apply the change by clicking **Apply** as figure 3.2-3.

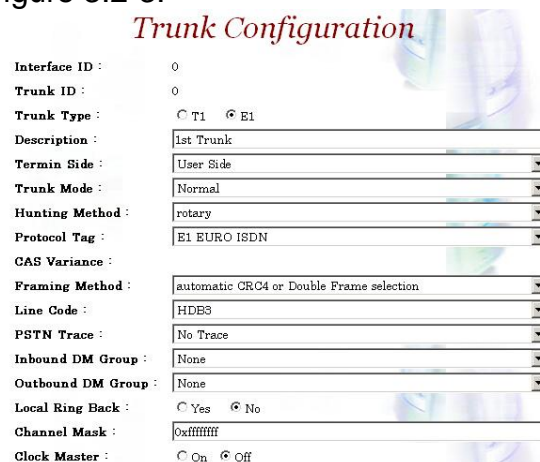


Figure 3.2-3

Frequency Changed Parameters:

- Trunk Type: E1 or T1
- Termin Type: User Side or Network Side
- Trunk Mode: Normal
- Protocol Tag: ISDN protocol used
- Line Code: T1 or E1 line code used

Step 4: After modifications are made to the Trunk Configuration, the screen comes back the page of **Trunk Configuration** as figure 3.2-4.

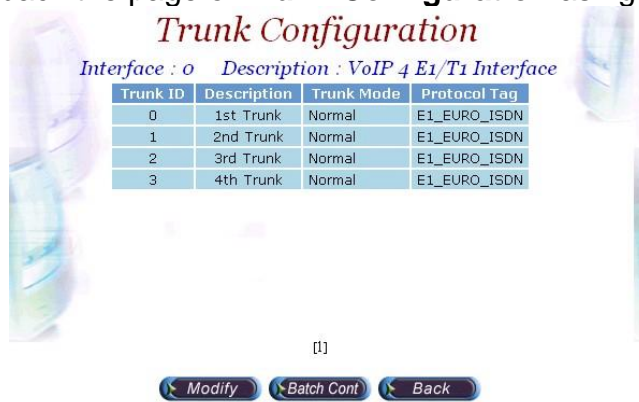


Figure 3.2-4

H.323 Configuration

This section is going setup the H.323 parameter. If you only need SIP calls, you can skip it.

Step 1: Click **Configuration**→**H.323** to setup the H.323 parameters for Gatekeeper related information as figure 3.3-1.

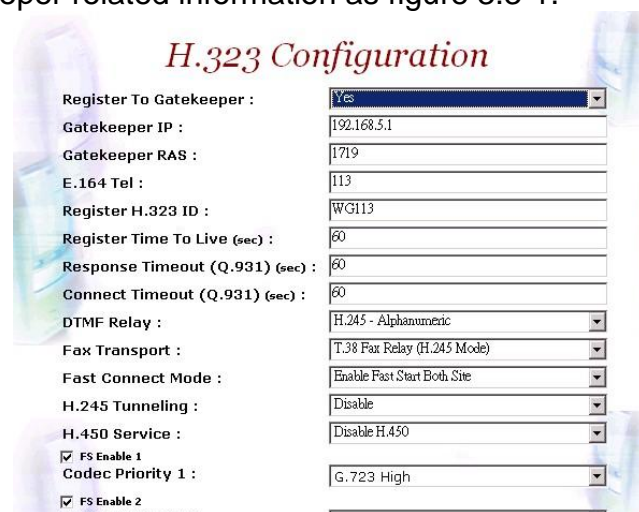


Figure 3.3-1

Frequency used parameters:

- Register to Gatekeeper: Yes
- Gatekeeper IP: 192.168.5.1
- E.164 Tel: 113
- Register H.323 ID: 113

Step 3: You can see the screen display the new configuration of the **H.323 Configuration** as figure 3.3-3.



Figure 3.3-3

SIP Configuration

This section is going setup the SIP parameter. If you only need H.323 calls, you can skip it.

Step 1: Click **Configuration**→**SIP** to setup the SIP parameters for SIP Proxy Server related information as figure.3.4-1.

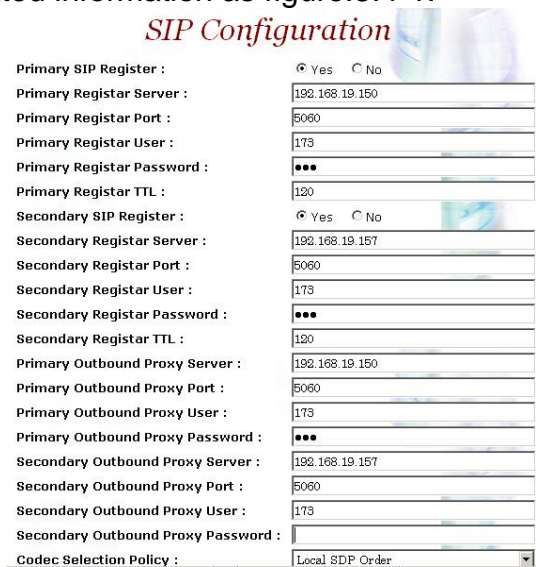


Figure 3.4-1

Frequency used parameters:

- SIP Register: Yes
- Primary Registrar Server: 192.168.19.150
- Primary Registrar Port: 5060
- Primary Registrar User: 173
- Primary Registrar Password: 173

- Primary Outbound Proxy Server: 192.168.19.150
- Primary Outbound Proxy Port: 5060
- Primary Outbound Proxy User: 173
- Primary Outbound Password: 173

Step 3: You can see the screen display the new configuration of the **SIP Configuration** as figure 3.4-2.

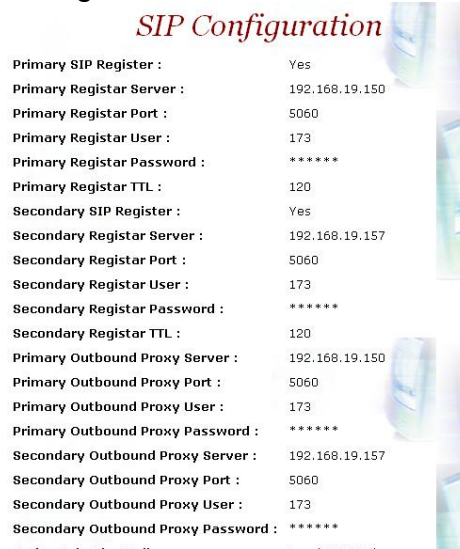


Figure 3.4-2

Digit Manipulation

The purpose of “Digit Manipulation” is to add or drop dialed digits for PSTN or IP side (Interface configuration for PSTN side & H.323 Configuration for IP side) at the selected interface in order to meet local PSTN dialing requirement. It can also be used in **Call Flow Edit** for flexible usage.

Step 1: We introduced the group and interface dependent digital manipulation to meet the customer’s requires. Click **Digit Manipulation** to add a new Digit Manipulation Group, add as figure 3.5-1.



Figure 3.5-1

Step 2: Enter the related parameters and click **Apply** button as figure 3.5-2.



Figure 3.5-2

Field Description:

- Group ID: 0 (DM Group identify)
- Description: H.323: H323 In Drop
SIP: SIP In Drop

Step 3: Click the New created DM and **Detail** button to add digits setting as figure 3.5-3.



Figure 3.5-3

Step 4: Click **New** button to add a new DM rule as figure 3.5-4.



Figure 3.5-4

Step 5: Create a new H.323 DM Group “1” and DM detail is show as follows:

Digit Manipulation List

| Matched Pattern | Group Id | Drop | Insert |
|-----------------|-------------------|------|--------|
| 5 | 1 - H.323 In Drop | 5 | |
| 504 | 1 - H.323 In Drop | 5 | |

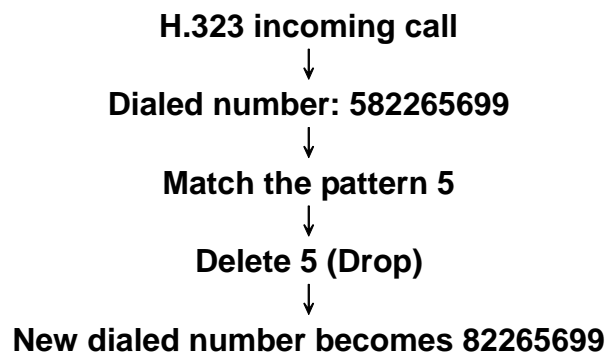
[1]

New Modify Delete Back

Figure 3.5-5

H.323 Incoming Call DM Setting:

- Matched Pattern: 5 (pattern to be matched)
- Group ID: 1-H323 In Drop (belong to this DM group)
- Drop: 5 (drop digits)



Step 5: Also create a new SIP DM Group ‘2” and DM detail is show as follows:

Digit Manipulation List

| Matched Pattern | Group Id | Drop | Insert |
|-----------------|-----------------|------|--------|
| 113 | 2 - SIP In Drop | 113 | |
| 11307 | 2 - SIP In Drop | 113 | |

[1]

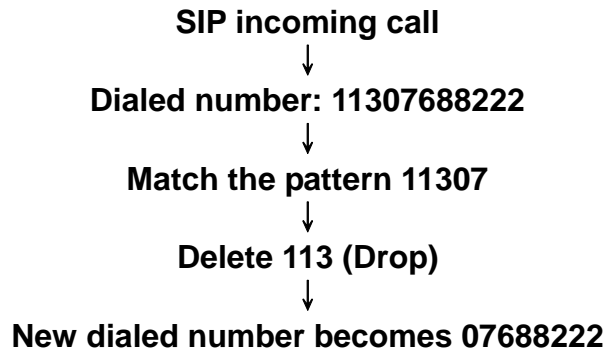
New Modify Delete Back

Figure 3.5-6

SIP Incoming Call DM Setting:

- Matched Pattern: 113 (pattern to be matched)
- Group ID: 1-SIP In Drop (belong to this DM group)

- Drop: 113 (drop digits)



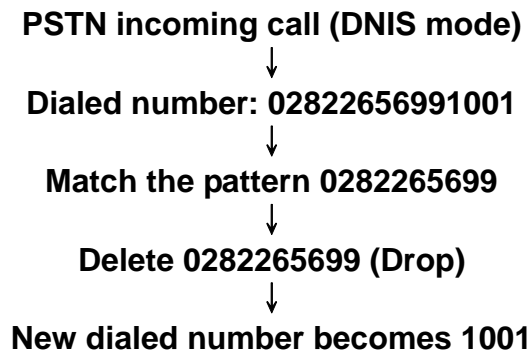
Step 6: Create a PSTN incoming call DM Group “3” and DM detail is show as follows:



Figure 3.5-7

PSTN DM Setting:

- Matched Pattern: 0282265699 (pattern to be matched)
- Group ID: PSTN In Drop (belong to this group id)
- Drop: 0282265699 (drop digits)




● **Note:** *Digit Manipulation have to tapped for PSTN Side (Trunk→ Outbound/Inbound DM Group), VoIP Side (VoIP→*

Outbound/Inbound DM Group) or Call Flow (refer to section “[Call Flow Editor](#)”) to take effect.

Chapter 4 Call Flow Editor

Call Flow Editor is used to control the call behavior including voice prompt, AAA, DM...etc. It requires Java run time to run.

Step 1: Click Control→**Call Flow Editor** to create a Call Flow, click  button to activate IVR Tool as figure 4-1

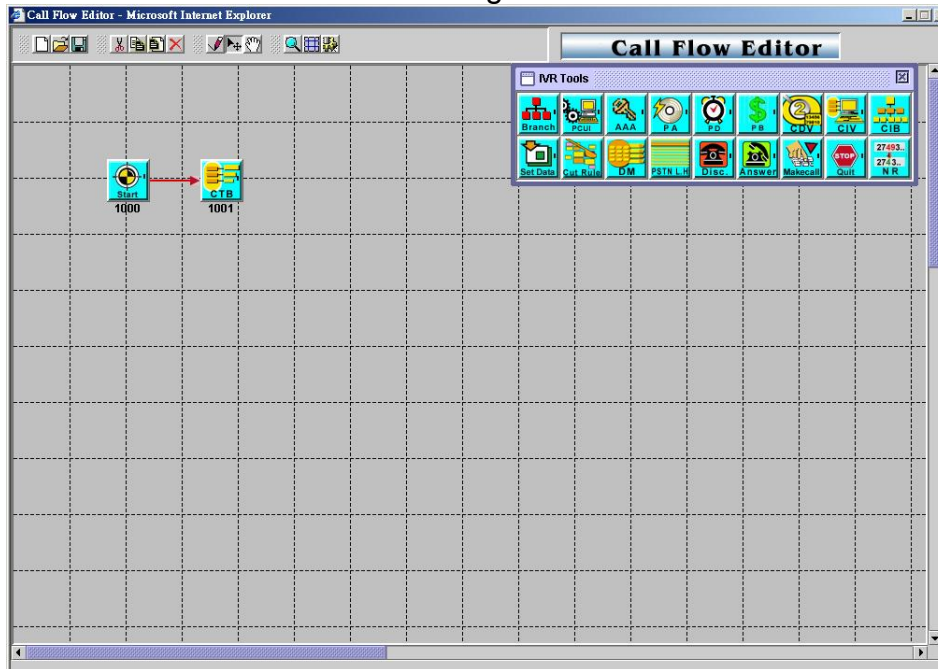















Figure 4-1

Component Description:


-  New: Create a new call flow
-  Load Call Flow: Load call flow from VIP-2100
-  Save: Save a call flow in VIP-2100
-  Cut: Cut a component
-  Copy: Copy a component
-  Paste: Paste a component
-  Delete: Delete a component
-  Line: Connecting 2 components together
-  Select: Select the component at call flow workspace
-  Scroll: Scroll the call flow workspace
-  Zoom: Zoom in or zoom out the workspace
-  View Grid: View or not
-  Show Component Table: Show all component table

Step 2: Drag and prop the required component icon into the workspace as figure 4-2.



Figure 4-2

Right click the component to bring up the component propriety to setup parameter:

-  AAA: Send Authorization or Authentication for validation



The screenshot shows the 'AAA' component property dialog box. The 'Current Component' is set to 1011. The 'Type' is set to 'Authentication'. The 'Fail Other To' field is 1009, 'Prepaid User To' is 1006, and 'Postpaid User To' is 1010. Below these fields is a table with two columns: 'Failed Reason' and 'Failed To'. The table contains one row with 'Invalid PinNumber' in the 'Failed Reason' column and '1009' in the 'Failed To' column. At the bottom of the dialog are 'OK' and 'Cancel' buttons.

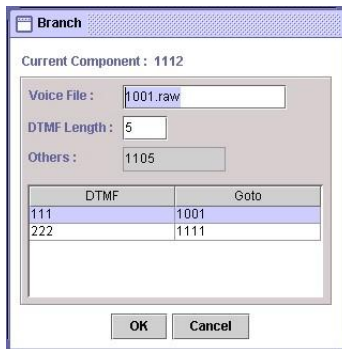
| Failed Reason | Failed To |
|-------------------|-----------|
| Invalid PinNumber | 1009 |

- Type: AAA type selection
 - Authorization: Send RADIUS Authorization packet out
 - Authentication: Send RADIUS Authentication packet out
- Success To: Success to component
- Failed other to: Failed to component

- Failed Reason: Return code from RADIUS server
- **Line Propriety:**
 - Invalid Account
 - Account In Use
 - Zero Balance
 - Account Expired
 - Over Credit Limit
 - Number of Retries Exceeded
 - Insufficient Balance

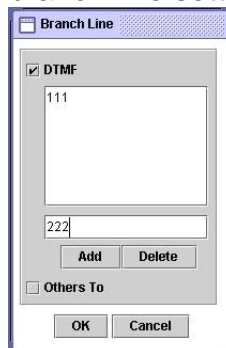
J Note: Detail response attributes, please refer [RADIUS Format Attributes](#)

-  Answer: Answer incoming call (PSTN only)
-  Branch: Play an announcement and branch into different route

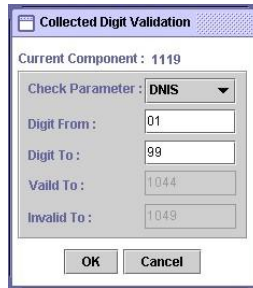


| DTMF | Goto |
|------|------|
| 111 | 1001 |
| 222 | 1111 |

- Voice File: Voice prompt file (“ . raw” format) to be playing
- DTMF Length: Number of DTMF to be receiving
- Others: Default flow if not match
- DTMF: DTMF match pattern
- Goto: The next component if matched
- Line Propriety:
 - Branch Line: DTMF branch line setting

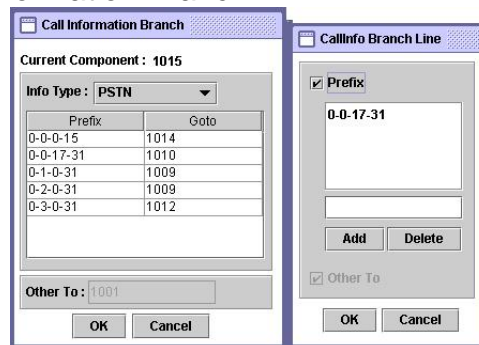


-  CDV: Collected Digit Validation




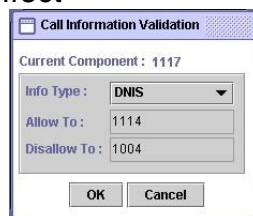
- Check Parameter: Check parameter type (DNIS, ANI....)
- Digit From: Start digit from
- Digit To: End digit to
- Valid To: If the checked variable is success to validate
- Invalid To: If the checked variable is not success to validate

-  CIB: Call Information Branch



- Info Type: Information type selection
 - ANI: Calling Number
 - DNIS: Called Number
 - IP: IP Address or network (e.g. 192.168.0.0)
 - PSTN: E1/T1 trunk and channel filter, format: ***interface id-trunk id- trunk start- trunk stop***
 - Prefix: The prefix to be match
 0-1-17-31:
0: Interface ID (Always 0)
1: Trunk ID: 1
17: Start from B Channel 17
31: Stop from B Channel 31
- Goto: The component to run next
- Call Info Branch Line: ANI, DNIS, IP or PSTN goto setting

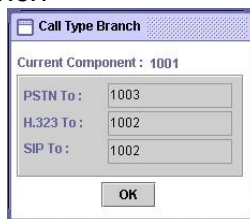
-  CIV: Call Information Validation, the user need setup the ACL for DNIS and IP TO take effect



- Info Type: The infor type to be validation

- DNIS: Called number
- ANI: Calling number
- IP: In coming IP address
- User: User ID
- Allow To: If it is met the ACL defined
- Disallow To: If it is not met the ACL defined

-  CTB: Call Type Branch



Call Type Branch

Current Component : 1001


PSTN To : 1003

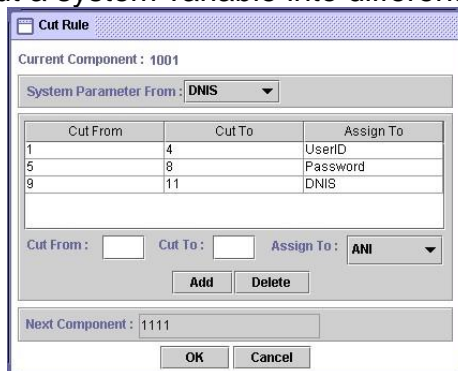
H.323 To : 1002

SIP To : 1002

OK

- PSTN To: Route for PSTN call
- H.323 To: Route for H.323 call
- SIP To: Route for SIP call

-  Cut Rule: Cut a system variable into different parts



Cut Rule

Current Component : 1001

System Parameter From : DNS

| Cut From | Cut To | Assign To |
|----------|--------|-----------|
| 1 | 4 | UserID |
| 5 | 8 | Password |
| 9 | 11 | DNIS |

Cut From : Cut To : Assign To : ANI

Add Delete

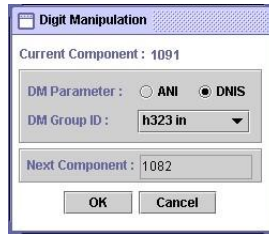
Next Component : 1111

OK Cancel


- Cut From: Cut start digit from (start from 1)
- Cut To: Cut end digit to
- Assign To: Store the cutted result into

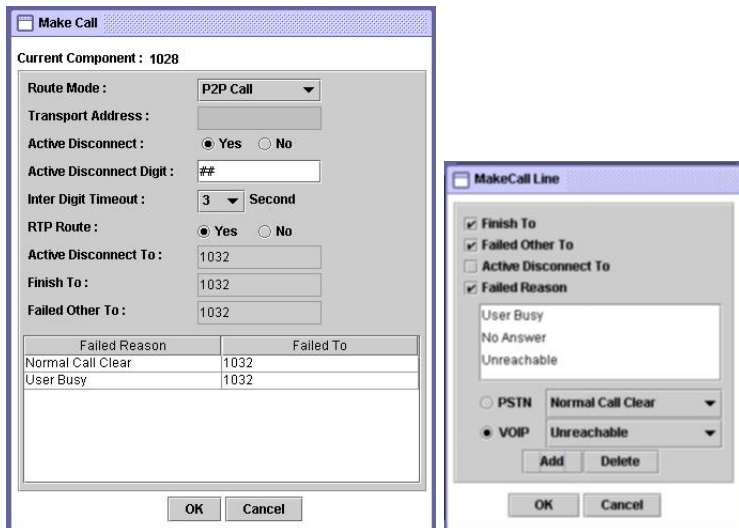
-  Disconnect: Disconnect the call

-  DM: Digit Manipulation



- DM Parameter: Manipulation ANI or DNIS
- DM Group ID: Apply to DM group

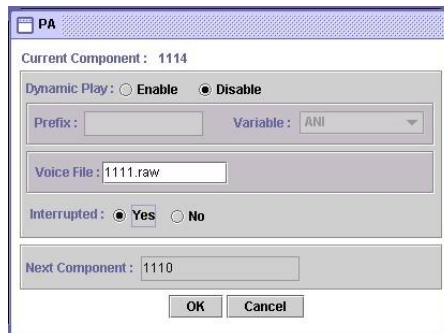
-  MakeCall: Make Call to PSTN or H.323/SIP site




- Route Mode: Gatekeeper Call or P2P Call or PSTN...etc. (for PSTN incoming call, please select the Gatekeeper, P2P Call, or SIP Proxy call TA; for H.323/SIP incoming call, please select the PSTN call)
- Transport Address: When used for “H.323 TA” routing mode, the format used is “Ipaddr:port” (e.g. 192.168.111.50:1720)
- Active Disconnect: Enable PSTN user can actively disconnect the call or not
- Active Disconnect Digit: The DTMF digit to be tread as the disconnect trigger. Only can be used “Active Disconnect” enable
- Active Disconnect To: The next component when active disconnect is occurred
- Inter Digit Timeout: The max time to in seconds to wait between two digits.
- RTP Route: Voice RTP routing over VIP-2100 or not, for VoIP to VoIP call
- Finish To: Successfully connect to remote site
- Failed Other to: The next component when default failed call
- Failed Reason: Failed reason selection
- Failed To: When the failed reason occurred go to

- **Line Propriety:**
 - PSTN: PSTN disconnect reason code:
 - Normal Call Clear
 - User Busy
 - No User Response
 - No Answer
 - Call Reject
 - VoIP: VoIP disconnect reason code:
 - User Busy
 - No Answer
 - Unreachable
 - Other

-  PA: Play Announcement



- Dynamic Play: Dynamic play voice file by combine prefix and variable as the file name
- Enable: Combine prefix to variable as the voice file to play
 - Prefix: Voice file prefix (**e.g. prefix: WT, variable: user1 (contact 201, played voice file is “WT201.raw”)**)
 - Variable: Variable to be appending as the voice file name
- Disable: Use filter voice prompt file
- Voice File: Voice prompt file
- Interrupted: Voice can be interrupted or not

-  PB: Play Balance for prepaid purpose



- Voice File: Voice prompt file
- Language: Play balance language section
 - English
 - Chinese.
- Interrupted: Voice can be interrupted or not


-  PCUI: Prompt and Connect User Information

- Play Type: Dial tone or voice prompt selection
- Voice File: Voice prompt file
- Max DTMF: Maxtor of DTMF to be received.
- Assign To: Result (received DTMF) will be assign to
- End of DTMF: The digit to indicate dial end.
- Interrupted: Voice can be interrupted or not

-  PD: Play Duration for prepaid purpose

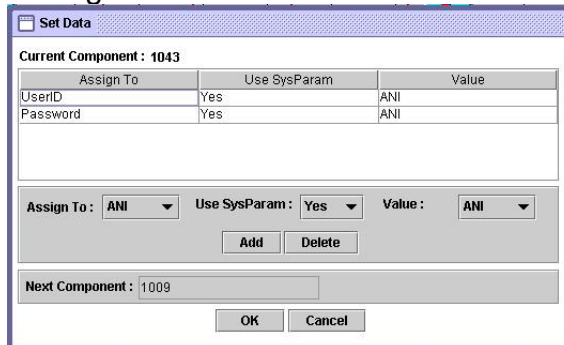
- Voice File: Leading voice prompt file
- Language: Play duration language section
 - English
 - Chinese
- Interrupted: Voice can be interrupted or not

● **Note: The RADIUS servers need to be setup to send H.323/SIP credit time or internal RADIUS must be used.**

-  PSTN L.H: PSTN Line Hunting

- Success To: If fine an available channel by system setup (call hunting)
- Failed To: If not fine an available channel by system setup (call hunting)

-
-  Set Data: Assign value to a variable

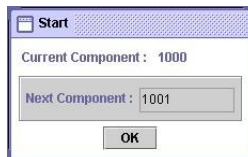


The 'Set Data' dialog box shows the current component as 1043. It contains a table with columns 'Assign To', 'Use SysParam', and 'Value'. The table lists 'UserID' and 'Password', both with 'Yes' in the 'Use SysParam' column and 'ANI' in the 'Value' column. Below the table are three dropdown menus: 'Assign To' (ANI), 'Use SysParam' (Yes), and 'Value' (ANI). There are 'Add' and 'Delete' buttons between the dropdowns. At the bottom, there is a 'Next Component' field with the value 1009, and 'OK' and 'Cancel' buttons.

| Assign To | Use SysParam | Value |
|-----------|--------------|-------|
| UserID | Yes | ANI |
| Password | Yes | ANI |

- Assign To: Assigned variable
- Use SysParam: Use system parameter to replace or not
- Value: ANI, DNIS, User ID or other digits

-  Start: Call flow start

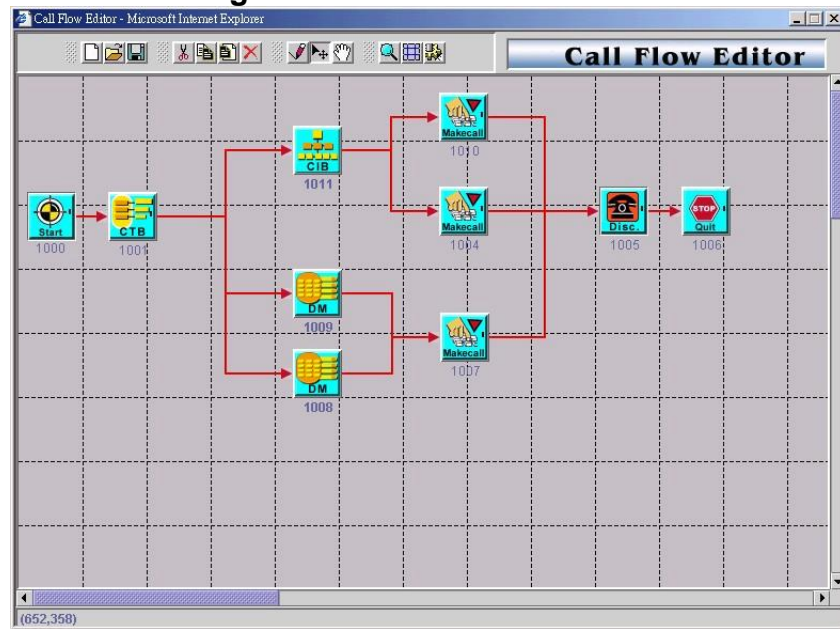


The 'Start' dialog box shows the current component as 1000. It has a 'Next Component' field with the value 1001 and an 'OK' button.




- Next Component

-  Quit: Disconnect calls



Example Call Flow as figure 4-3.





Example Description:





| Components | Contents |
|---|---|
|  Start Component ID: 1000 | Next Component: 1001 |
|  CTB Component ID: 1001 | PSTN To: 1011 H.323 To: 1009 SIP To: 1008 |
|  CIB Component ID: 1011 | Info Type: ANI Prefix: 1 goto: 1010 (H.323 GK call) Prefix: 8 goto: 1004 (SIP Proxy call) |

1011 Route for PSTN call





| | |
|--|---|
|  MakeCall Component ID: 1010 | Route Mode: Gatekeeper Finish To: 1005 Failed Other To: 1005 |
|  MakeCall Component ID: 1004 | Route Mode: SIP Proxy Call Finish To: 1005 Failed Other To: 1005 |

| | |
|--|----------------------|
|  Disc Component ID: 1005 | Next Component: 1006 |
|  Quit Component: 1006 | |

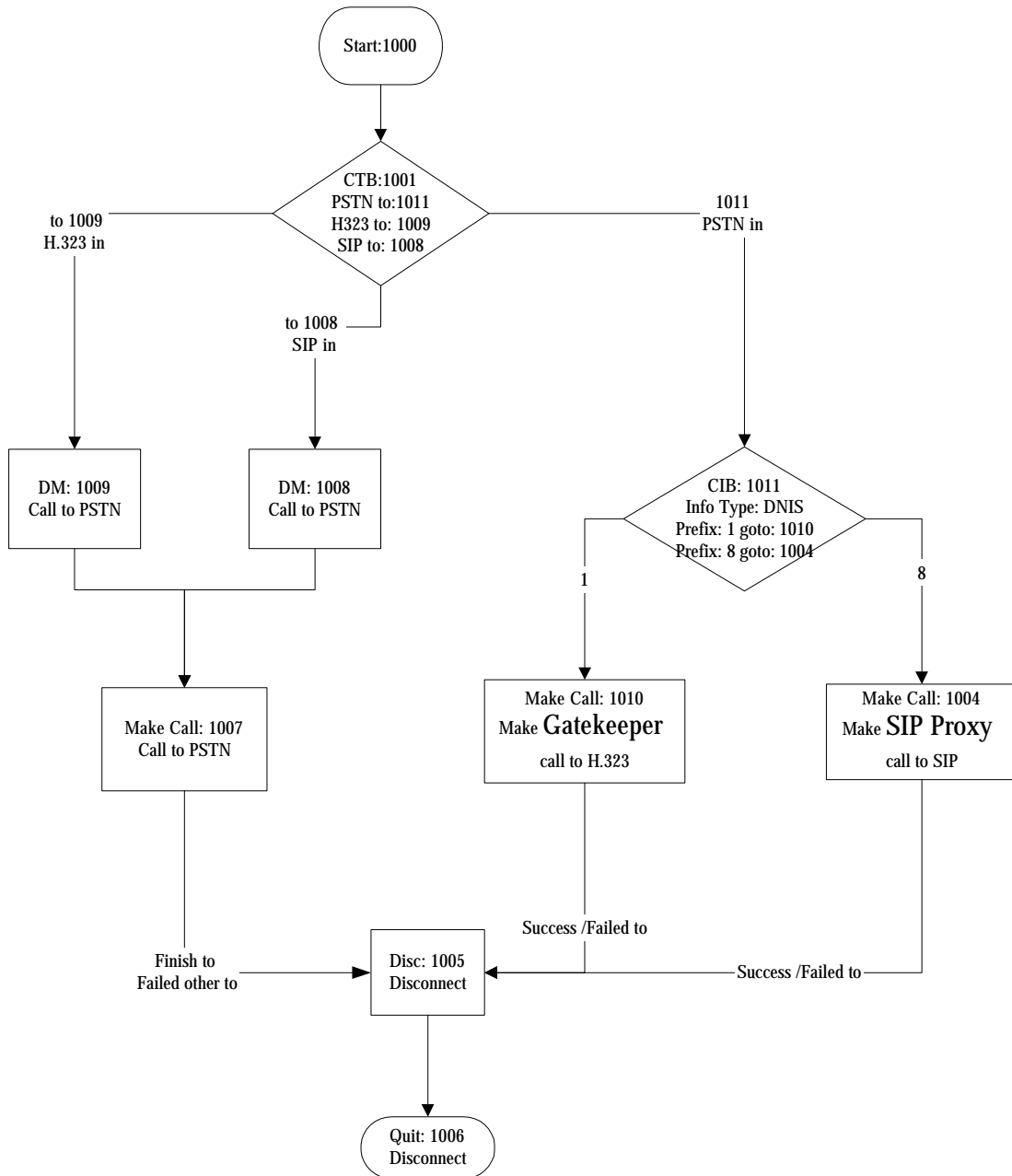
1001 Route for H.323 Gatekeeper call

| | |
|--|--|
|  DM Component ID: 1009 | DM Parameter: DNIS DM Group ID: H.323 In Drop Next Component: 1007 |
|  MakeCall Component ID: 1007 | Route Mode: PSTN Finish To: 1005 Failed Other To: 1005 |
|  Disc Component ID: 1005 | Next Component: 1006 |
|  Quit Component: 1006 | |

1001 Route for SIP Proxy call

| | |
|--|--|
|  DM Component: 1008 | DM Parameter: DNIS DM Group ID: SIP In Drop Next Component: 1007 |
|  MakeCall Component ID: 1007 | Route Mode: PSTN Finish To: 1005 Failed Other To: 1005 |
|  Disc Component ID: 1005 | Next Component: 1006 |
|  Quit Component: 1006 | |

Example Used Call Flow:



Configuration Manager

Configuration Management provides a way to save and reload the system configuration for future use.

Load a Configuration:

Step 1: When you need to load a saved configuration, click a saved configuration (i.e. 04/26/2004 Loading Test) item to load it back as figure 4.1-1.



Figure 3.7-1

Step 2: When screen shows “**Current configuration will lost! Are you sure to load this configuration?**” click on OK button to load he saved configuration to the working configuration as figure 4.1-1.

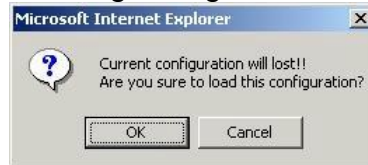


Figure 3.8-2

J Note: It is need to restart the system to take effect of the new-loaded working configuration.

Save the working Configuration:

Step 3: To save the current configuration, select a new created configuration and click **Save** button, when screen shows “**Description**”, please enter the configuration description (i.e. Billing Test) for the saved configuration as figure 4.2-2.



Figure 3.8-3

Step 4: You can see the screen display the changes as figure 4.2-4.

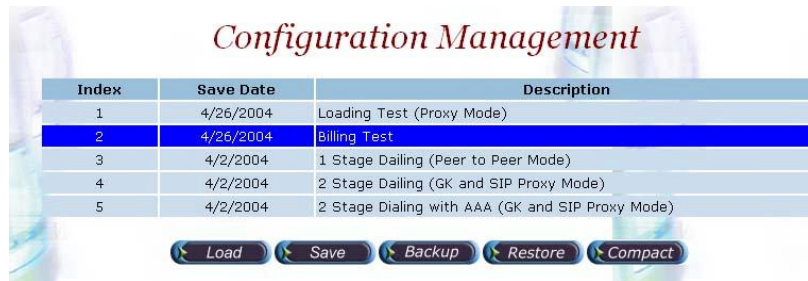


Figure 3.8-4

Backup the working configurations:

Step 5: To backup the running configuration, click on **Backup** button, to back up local hard disk. The whole running configuration will be compress into a zip file (file name: export.zip) and transfer back to local as figure 4.2-2.



Figure 3.8-5

Restore configuration:

Step 6: To restore the backup configuration file, click on **Restore** button, when screen shows **“Import Configuration file”**, select backup file (i.e. c:\export.zip) click on **Import** button to restore the configuration to the working configuration as figure 4.2-2.

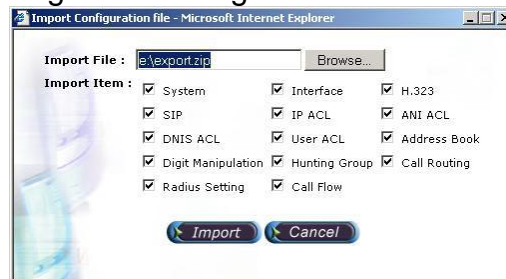


Figure 3.8-6

Compact the database file:

Step 7: In order to optimize the system performance, you can optional compact the database by click **Compact** button as figure 4.1-2.



Figure 3.8-7

J Note: Please make sure that there is no others person to use database concurrently.

Apply Change

When you load a new working configuration, the system must be restarted to take effect.

Step 1: Click **Configuration**→**Apply Change**, the screen show “ ***The change you made need to restart the system for apply please confirm to restart or do it later.***” Click on **OK/Cancel** to restart the system or not as figure 4.3-1.

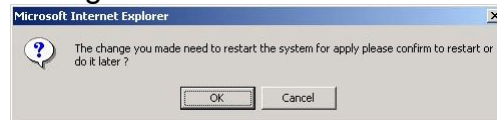
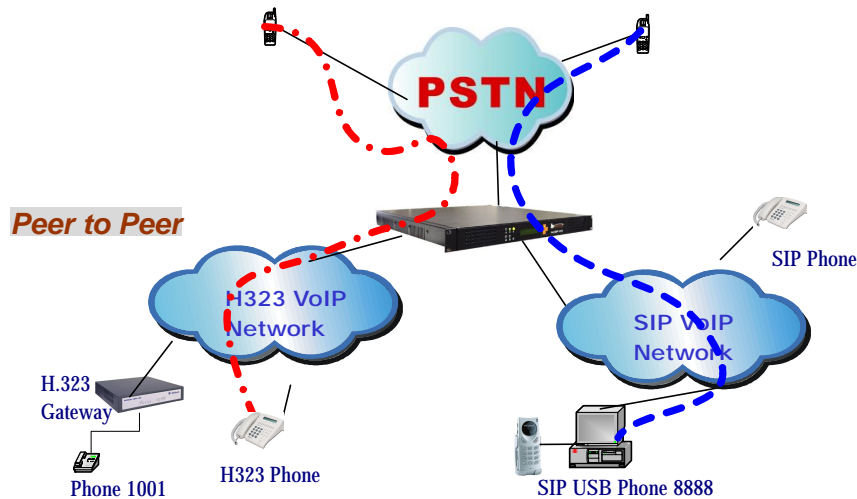


Figure 4.3-1

Chapter 5 Peer to Peer Mode Configuration

Environment used in this chapter



Process:

PSTN → H.323 Call: DNIS (822656991001) → DM (PSTN In Drop) → Make H.323 - Peer to Peer Call (1001)
→ SIP Call: DNIS (822656998888) → DM (PSTN In Drop) → Make SIP - Peer to Peer Call (8888)

H.323 → DNIS (50932123321) → DM (H.323_in_drop) → Make Call (0932123321)

SIP → DNIS (1130028610825123) → DM (SIP_in_drop) → Make Call (0028610825123)

☺ Digit Manipulation: Please refer section "[Digit Manipulation](#)"

Network Configuration

Please refer to section "[Network Configuration](#)"

Account Manager

Please refer to section "[Account Manager](#)"

Interface Configuration

Please refer to section "[Interface Configuration](#)"

H.323 Configuration

Step 1: Change **Register To Gatekeeper** to “No” to enable peer to peer mode as figure 5.1-1.



Figure 5.1-1

Frequency used parameters:

- Register to Gatekeeper: No

SIP Configuration

Step 1: Change **SIP Register** to “No” to enable peer to peer mode as figure 5.2-1.



Figure 5.2-1

Frequency used parameters:

- Primary SIP Register: No

Address Book

For making a **Peer-to-Peer** call, the IP device must have an address record in the phone book for routing.

Step 1: Click **Address Book** adds a new address book for the peer to peer calls, New to add as figure 5.3-1.

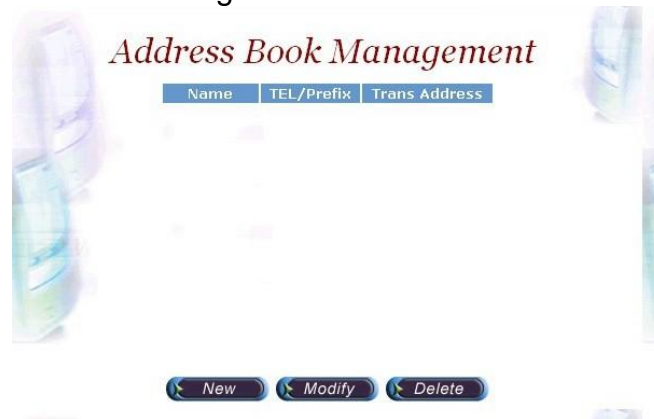


Figure 5.3-1

Step 2: Enter the related parameters and click **Apply** button as figure 5.3-2.

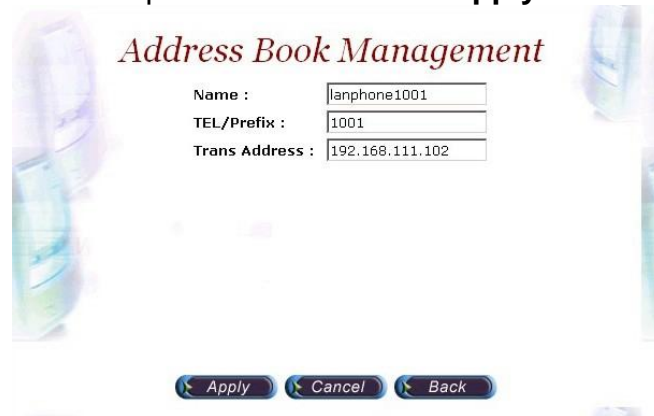
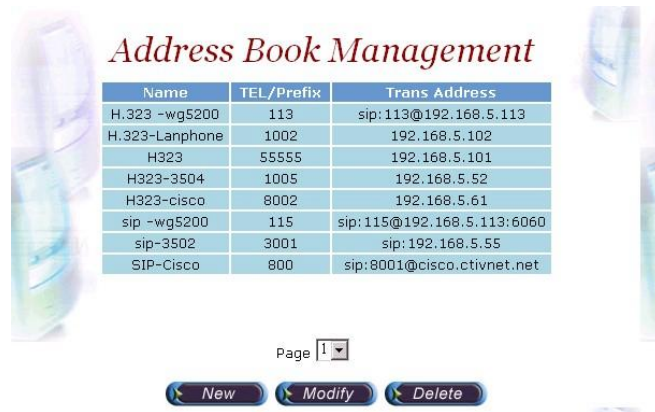


Figure 5.3-2

Field Description:

- Name: H.323 IP Phone or SIP-Cisco
- Tel/Prefix: 1002
- Trans Address:
 - H.323 Call: 192.168.5.102 or 192.168.5.102:1720
 - SIP Call: sip:8001@192.168.5.61 or sip:8001@192.168.5.61:5060 or sip:8001@ctivnet.net

Step 3: You can see the screen displays the new Address Book as figure 5.3-3.



The screenshot shows a web interface titled "Address Book Management". It features a table with three columns: "Name", "TEL/Prefix", and "Trans Address". Below the table, there is a "Page" dropdown menu set to "1" and three buttons labeled "New", "Modify", and "Delete".

| Name | TEL/Prefix | Trans Address |
|----------------|------------|----------------------------|
| H.323 -wg5200 | 113 | sip:113@192.168.5.113 |
| H.323-Lanphone | 1002 | 192.168.5.102 |
| H323 | 55555 | 192.168.5.101 |
| H323-3504 | 1005 | 192.168.5.52 |
| H323-cisco | 8002 | 192.168.5.61 |
| sip -wg5200 | 115 | sip:115@192.168.5.113:6060 |
| sip-3502 | 3001 | sip:192.168.5.55 |
| SIP-Cisco | 800 | sip:8001@cisco.ctivnet.net |

Figure 5.3-3

● **Note:** You must apply the change to take effect for the change.

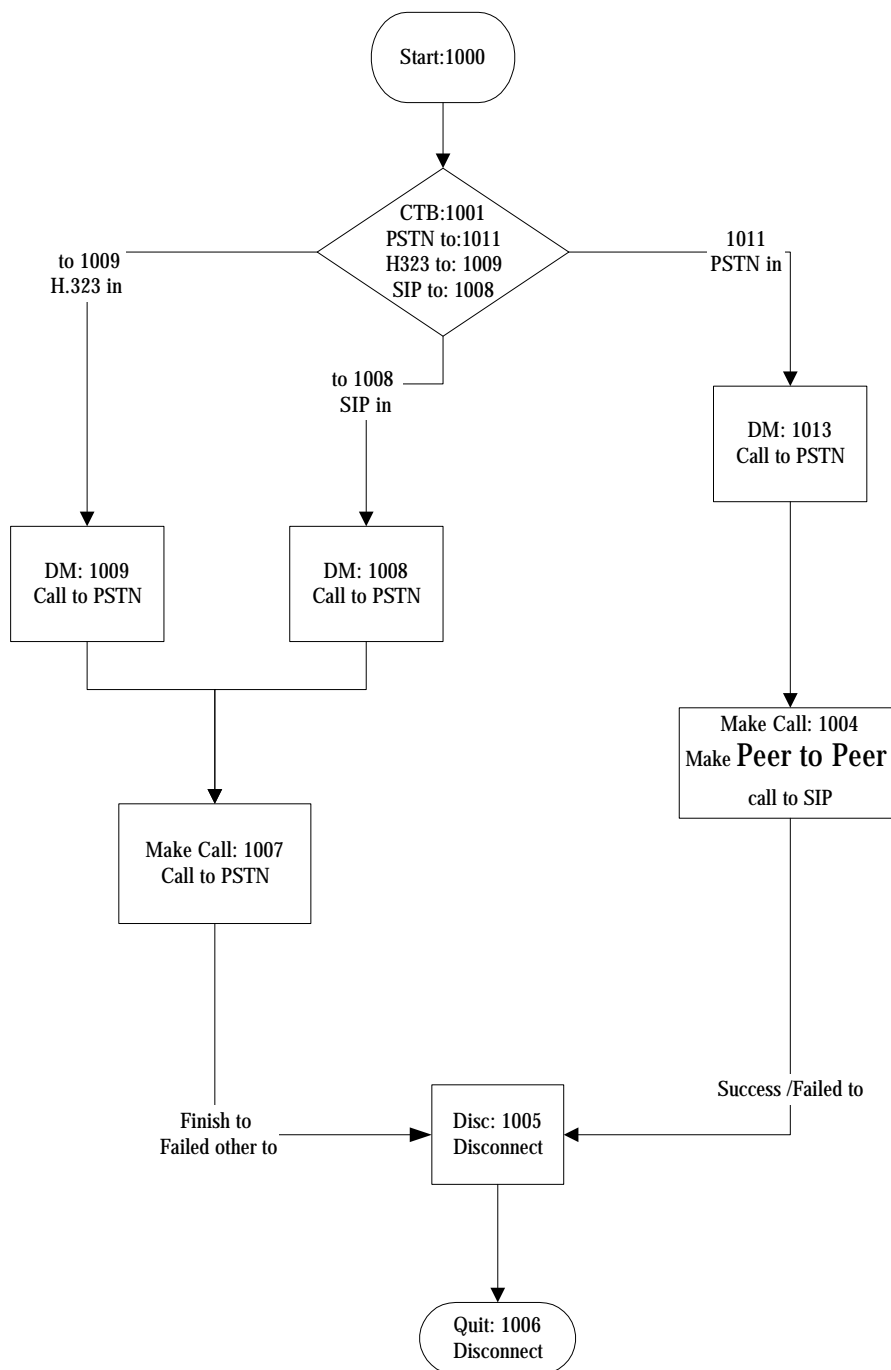
Digit Manipulation

Please refer to section "[Digit Manipulation](#)"

Call Flow Editor

Please refer to section "[Call Flow Editor](#)"

Call Flow (P2P Mode):



Configuration Manager

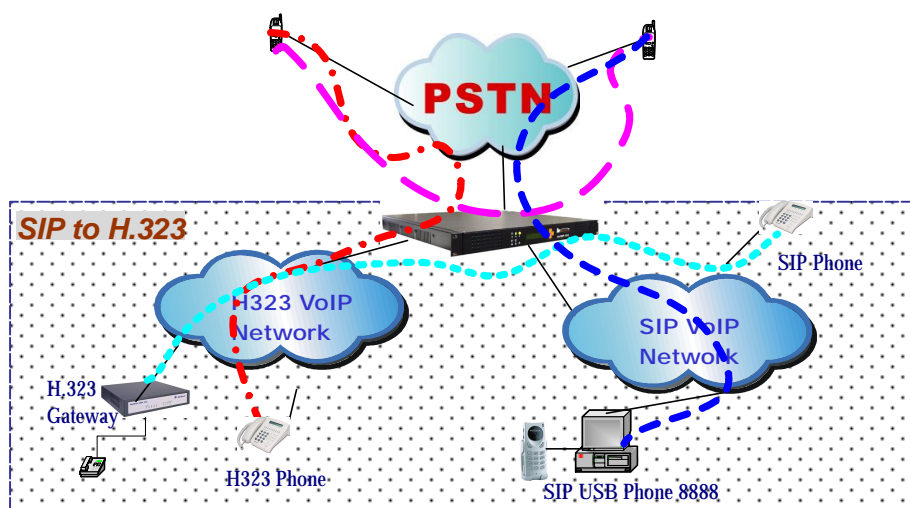
Please refer to section "[Configuration Manger](#)"

Apply Change

Please refer to section “[Apply Change](#)”

Chapter 6 SIP to H.323 Mode Configuration

Environment used in this chapter



Process:

SIP → H.323 Call: DNIS (8861001) → DM (SIP In Drop) → Make H.323 (1001)

H.323 → SIP (8868888) → DM (H.323_in_drop) → Make Call (8888)

☺ Digit Manipulation: Please refer section “[Digit Manipulation](#)”

Network Configuration

Please refer to section “[Network Configuration](#)”

Account Manager

Please refer to section “[Account Manager](#)”

Interface Configuration

Please refer to section “[Interface Configuration](#)”

H.323 Configuration

Please refer to section “[H323 Configuration](#)”

SIP Configuration

Please refer to section “[SIP Configuration](#)”

Address Book

Please refer to section "[Address Book](#)"

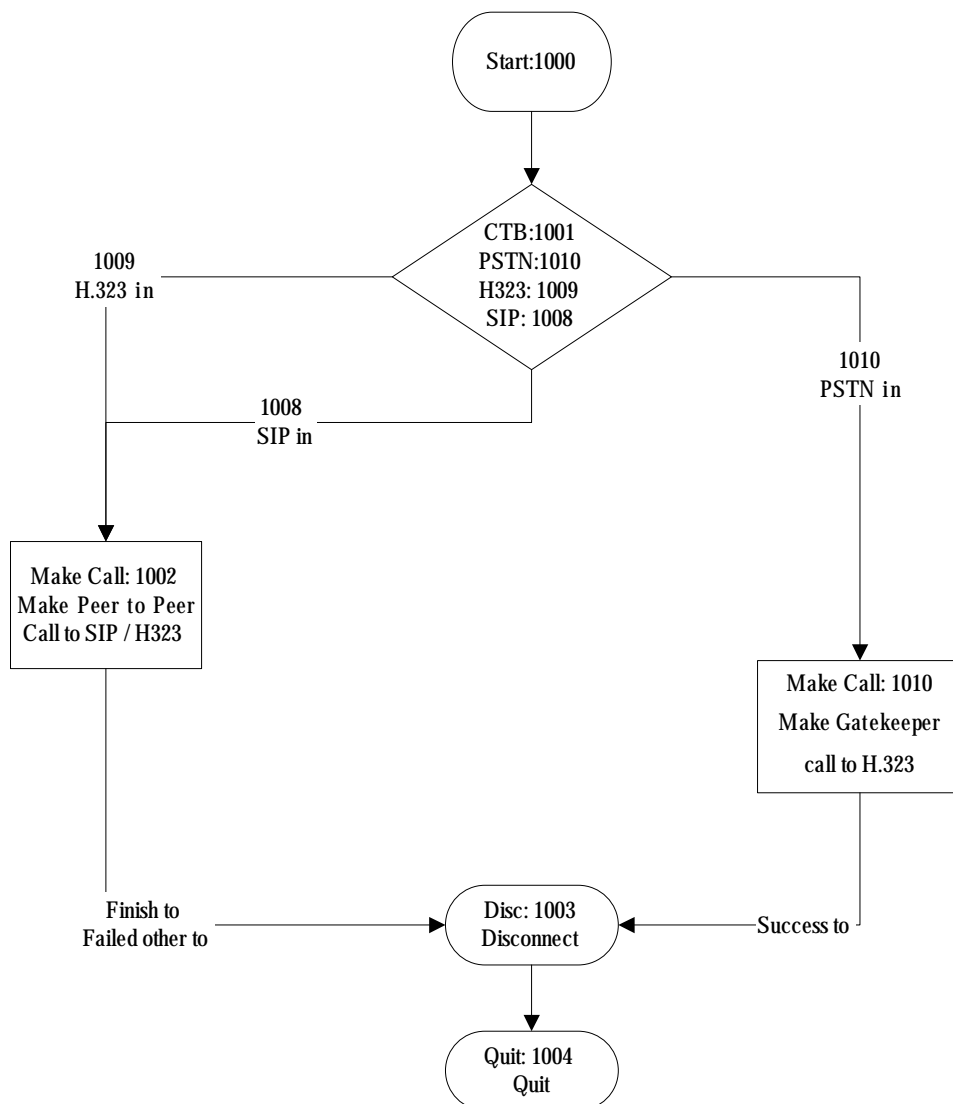
Digit Manipulation

Please refer to section "[Digit Manipulation](#)"

Call Flow Editor

Please refer to section "[Call Flow Editor](#)"

Call Flow (P2P Mode):



Configuration Manager

Please refer to section "[Configuration Manger](#)"

Apply Change

Please refer to section “[Apply Change](#)”

Chapter 7 Advance Configuration Reference

Configuration

System Configuration

Start Path: Configuration→System



Figure 7.1-1

Parameter Description:

- CDR Mode: Call detail record generating mode (Refer to “[Appendix 3 Retrieve CDR Information](#)” for detail file description)
 - File Only: Log CDR into the file only. It can be retrieved by ftp (directory c:\cd cdr).
 - Radius Start/Stop: Log CDR into the file and send RADIUS start/stop billing message out.
 - VoIP: enable VoIP site RADIUS billing message or not.
 - PSTN: enable PSTN site RADIUS billing message or not.
 - Radius Stop: Log CDR into the file and send RADIUS stop billing message out.
 - VoIP: enable VoIP site RADIUS billing message or not.
 - PSTN: enable PSTN site RADIUS billing message or not.
- CDR Keepdays: CDR system keeping days
- Hot Swappable: Hot swappable support (reserved)
- First Digit Timeout: The max to time (in second) waits for receiving the first digit entered (5~20 sec).
- Inter Digit Timeout: The max to time (in second) waits for the between two digits (5~20 sec).
- Debug Level:
 - Critical: Show critical error messages only
 - Warring: Show warring and critical error message only
 - Information: Show information, warring and critical message only
 - Debug: Show all debug messages
 - Full Trace: Show all status and debug messages

● **Note: Please set to “Critical” only, or the whole system performance will be hitted.**

- Time Expired Notify: Seconds to be notifying caller before communication expired. This function is used for Pre-Paid calling card service and must cooperate with RADIUS Server.
- Almost Expired Tone: Communication expired notice tone selection
- Fast Response Timeout: The maximum times to wait for response. It's depended on the network speed.
- No Answer Timeout: The maximum the (in second) to wait the remote party answer (pick up phone)
 - Notify Tone#1:
 - Notify Tone#2:
- Authentication Mode: Authentication by VIP-2100 or RADIUS
 - Internal: Authentication building User ACL
 - External: Authentication by RADIUS
 - Ext. AAA Failure Opt: Bypass or disconnect incoming calls when external
- Version: 5.1

Interface Configuration

Start Path: Configuration→Interface

Interface Configuration

| | | | |
|-------------------|----------------------------------|-------------|---|
| Interface ID : | 0 | Card Slot : | 3 |
| Interface Type : | 4 E1/T1 | | |
| Description : | VoIP 4 E1/T1 Interface 1 | | |
| Serial No : | 242807 | | |
| License Key : | 020519d73e655d430901e35ef2dd1497 | | |
| IP Address : | 192.168.19.174 | | |
| Subnet Mask : | 255.255.255.0 | | |
| Default Gateway : | 192.168.19.254 | | |
| PCM Type : | A-law | | |

Figure 7.2-1

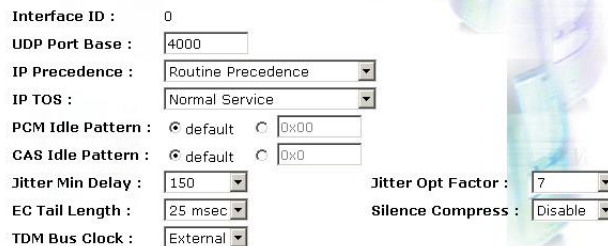
Basic Parameter Description:

- Interface ID: System parameter
- Card slot: System parameter
- Interface Type: System parameter
- Description: System parameter
- Serial No: System parameter
- License Key: System parameter
- IP Address: IP address used for voice RTP stream
- Subnet Mask: Submask (doesn't support super class)
- Default Gateway: Default gateway for routing
- PCM Type: PCM type encoding, E1 A-law; T1 u-law

Advance Interface Configuration:

Start Path: Configuration→Interface →Advance

Advance Interface Configuration



The screenshot shows a configuration window titled "Advance Interface Configuration". The settings are as follows:

| | |
|---------------------|---|
| Interface ID : | 0 |
| UDP Port Base : | 4000 |
| IP Precedence : | Routine Precedence |
| IP TOS : | Normal Service |
| PCM Idle Pattern : | <input checked="" type="radio"/> default <input type="radio"/> 0x00 |
| CAS Idle Pattern : | <input checked="" type="radio"/> default <input type="radio"/> 0x0 |
| Jitter Min Delay : | 150 |
| Jitter Opt Factor : | 7 |
| EC Tail Length : | 25 msec |
| Silence Compress : | Disable |
| TDM Bus Clock : | External |

Figure 7.2-2

Advance Parameter Description:

- Interface ID: System parameter
- UDP Port Base: UDP port used for RTP stream, each channel needs 3 RTP ports and must be started by a multiple of 10
- IP Precedence: Voice package priority setting
 - Routine Precedence
 - Priority Precedence
 - Immediate Precedence
 - Flash Precedence
 - Flash Override Precedence
 - Critical Precedence
 - Internetwork Precedence
 - Network Precedence
- IP TOS: Top of Service with the following priority selection
 - Normal Service
 - Minimize Monetary
 - Maximize Reliability
 - Maximize Thought
 - Minimize Delay
- PCM Idle Pattern: This pattern will be sending on each B channel PCM time slot when the channel is idle (not connected). The default value for A-Law is 0xff and for Mu-Law is 0x55. You only change it when SWITCH need.
- CAS Idle Pattern: When channel is idle, ABCD (CAS) pattern to be applied CAS signaling bus
- Jitter Min Delay: The minimum delay time of Jitter buffer. The range is 0 to 150ms. Default value is 150ms. Which has better voice quality but the delay time will be long.
- Jitter Opt Factor: Jitter buffer optimization factor from 0 to 12. The default value is 7. Set to 0 will have lowest voice delay but have bad

voice quality. Set to 12 will have long voice delay but with better voice quality

- EC Tail Length: Echo Cancellation Length, default value is 25ms
- Silence Compress: Enable silence compress or not
- TDM Bus Clock: TDM Bus clock source
 - Internal: derived from internal oscillator
 - External: derived from external PSTN E1/T1 clock

Dial Plan Configuration

Dial Plan can be used to assign the ISDN number plan based on prefix setting.

Start Path: Configuration→Interface→Dial Plan

Dial Plan Management

| Prefix | Src Num Plan | Src Num Type | Dest Num Plan | Dest Num Type | ApplyTo |
|--------|------------------------|----------------------|---------------------|----------------------|-----------|
| 031 | Unknown Numbering Plan | Unknown Number | ISDN Numbering Plan | International Number | All Trunk |
| 886 | ISDN Numbering Plan | International Number | ISDN Numbering Plan | International Number | Trunk 0 |

Figure 7.3-1

Basic Parameter Description:

- Prefix: Called party number prefix
- Src Num Plan: ISDN Source number plan
- Src Num Type: ISDN Source number type
- Dest Num Plan: ISDN destination number plan
- Dest Num Type: ISDN destination number type
- ApplyTo: Trunks apply to

T1/E1 Trunk Configuration

Start Path: Configuration→Interface→Trunk

Trunk Configuration

Interface ID : 0

Trunk ID : 0

Trunk Type : T1 E1

Description : 1st Trunk

Termin Side : User Side

Trunk Mode : Normal

Hunting Method : rotary

Protocol Tag : E1 EURO ISDN

CAS Variance :

Framing Method : automatic CRC4 or Double Frame selection

Line Code : HDB3

PSTN Trace : No Trace

Inbound DM Group : None

Outbound DM Group : None

Local Ring Back : Yes No

Channel Mask : 0xffffffff

Clock Master : On Off

Figure 7.3-1

Basic Parameter Description:

- Interface ID: System parameter
- Trunk ID: System parameter
- Trunk Type: T1 or E1 selection
- Description: Description for this trunk ID
- Termin Side: Network site or User Site (normally, you set to “**user site**” when connect to switch)
 - User Side
 - Network Side
- Trunk Mode: Trunk operation mode
 - Disable: Disable the trunk
 - Normal: Accept PSTN and VoIP calls
 - PSTN incoming only: Allow the PSTN incoming calls only
 - H.323 incoming only: Allow the H.323 incoming calls only
- Hunting Method: PSTN trunk hunting method for available channel
 - Random: Hunt randomly
 - Cyclic: Initial hunt (after power-up/reboot) begins with B channel 1; subsequent hunts begin with position following last successfully allocated resource
 - Rotary: Hunt always begins with B channel 1
 - Reverser Rotary: Hunt always begins with B channel 31
 - Reverser Cyclic: Initial hunt (after power-up/reboot) begins with B channel 31, follows next available channel in reverser order
- CAS Variance: CAS counting variance
- Framing Method:
 - **For T1**
 - super frame
 - 4-frame multi-frame
 - 12 frame multi-frame (D4)
 - extend super frame without CRC6
 - extend super frame with CRC6
 - 72-Frame Multi-Frame
 - **For E1:**
 - Automatic CRC4 or Double Frame selection
 - Double Frame Format
 - CRC4 multi-frame
 - CRC4 extend multi-frame
- Protocol Tag: supported protocol on T1/E1 interface with PSTN switch
 - **For T1:**
 - T1 CAS
 - T1 RAW CAS
 - T1 NI2 ISDN
 - T1 4ESS ISDN
 - T1 5ESS 9 ISDN
 - T1 5ESS 10 ISDN

-
- T1 DMS100 ISDN
 - T1 NTT ISDN: used to connect NTT INS-1500 ISDN standard (Japan Only)
 - T1 HKT ISDN
 - T1 QSIG
 - T1 EURO ISDN
 - T1 DMS100 MERIDIAL ISDN
 - T1 NI1 ISDN
 - **For E1:**
 - E1 EURO ISDN: used for most of European ISDN standard
 - E1 MFCR2
 - E1 CAS
 - E1 RAW CAS
 - E1 AUSTEL ISDN: Australia E1 ISDN Variance
 - E1 HKT ISDN: Hong E1 ISDN Kong Variance
 - E1 KOR ISDN: Korea E1 ISDN Variance
 - QSIO
 - E1 TNZ ISDN
 - Line Code: T1: you can choose AMI, B8ZS; E1: you can choose AMI, HDB3
 - PSTN Trace: PSTN layer debug trace. It will generate a debug trace file for tracing purpose. Only enables it under Welltech technical supports instruction and disable it when complete the debug
 - Inbound DM Group: Digit Manipulation group used for incoming calls associated to this trunk
 - Outbound DM Group: Digit Manipulation group used for outgoing calls
 - Local Ring Back: Provide ring back tone for PSTN or not. It only works when VoIP outgoing Fast Start is disabled.
 - Channel Mask: Channel mask for incoming or outgoing calls (default: 0xffffffff)
Start from MSB each bit, indicate a time, slot a trunk (e.g. 0x0000ffff: 0~15 B channel mask, 17~31 B channel free)
 - Clock Master: PSTN trunk clock source

Advance Trunk Configuration:

Start Path: Configuration → Interface → Trunk → Advance

Advance Trunk Configuration

| | |
|------------------------|----------------------------|
| Interface ID : | 0 |
| Trunk ID : | 0 |
| Src Num Plan : | Unknown Numbering Plan |
| Dest Num Plan : | Unknown Numbering Plan |
| Src Num Type : | Unknown Number |
| Dest Num Type : | Unknown Number |
| Src Num Presen : | Presentation Not Included |
| Src Num Screen : | Number Screen Not Included |
| Input Gain : | 0 dB |
| Output Gain : | 0 dB |
| Q931 General Opt. : | 0x0800 |
| Q931 Incoming Opt. : | 0x2000 |
| Q931 Outgoing Opt. : | 0x0000 |
| Trans Cap : | Voice Service |
| CallID Transfer Type : | Disable Caller ID |

Figure 7.3-2

Advance Parameter Description:

- Interface ID: System parameter
- Trunk ID: System parameter
- Src Num Plan: ISDN source number plan
- Dest Num Plan: ISDN destination number plan
- Src Num Type: ISDN source number type
- Dest Num Type: ISDN destination number type
- Src Num Presen: ISDN source number presentation
- Src Num Screen: ISDN source number display
- Input Gain: Voice Gain from IP to PSTN side (default: 0 db)
- Output Gain: Voice Gain from PSTN to IP side (default: 0 db)
- Q.931 General Opt.: used for Q.931 general behavior.
 - 0x0001: No Status message send for unknown facility IE if it is set
 - 0x0002: No Status message send for invalid content of a valid facility IE if it is set
 - 0x0080: Send Connect Ack message when receive Connect message if it is set, you can OR the required option together
- Q.931 Incoming Opt.: used for Q.931 incoming call behavior
 - 0x0800: include Channel-ID IE in the first reply message (e.g. Call Proceeding or Alerting)
 - 0x2000: enable the system to include Channel-ID IE in the Call Proceeding message, you can OR the required option together
- Q.931 Outgoing Opt.: used for Q.931 outgoing behavior
 - 0x0010: use Mu-law if this bit is set, or A-law will be used. Apply only for Korea variance, you can OR the required option together
- Trans Cap: Transfer Capability
 - Voice Service
 - Data Service
 - Modem Service
- CallID Transfer Type: Call ID transfer type
 - Disable Caller ID: default parameter
 - Transparent Caller ID
 - Relay Caller ID
 - Bypass Caller ID

Rest Configuration

Reset a channel or a trunk idle state.

Start Path: **Configuration**→**Interface**→**Detail**→**Reset**

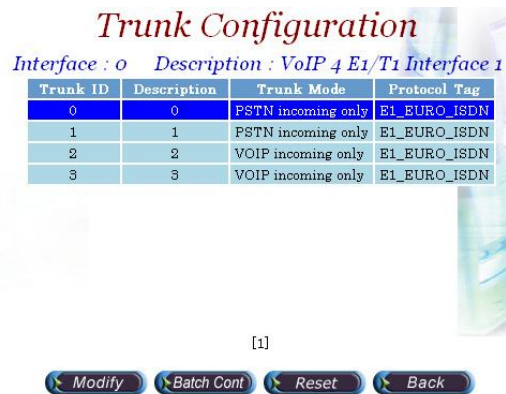


Figure 7.4-1

Start Path: **Configuration**→**Interface**→**Detail**→**Reset**



Figure 7.4-2

Basic Parameter Description:

- Trunk: Reset trunk ID
- Channel: Rest channel selection
 - All Channel: Reset all channel
 - 0~31: Reset 0~30 logical channel to reset

H.323 Configuration

Start Path: **Configuration**→**H.323**



Figure 7.5-1

Basic Parameter Description:

- Register To Gatekeeper: Register to Gatekeeper or not
 - Yes: Register to GK
 - No: Not register to GK
- Gatekeeper IP: Gatekeeper IP Address
- Gatekeeper RAS: UDP Port number listened on Gatekeeper (default: 1719)
- E.164 Tel: Telephone number to be registered to Gatekeeper
- Register H.323 ID: H.323 alias name to be registered to Gatekeeper
- Register Time To Live (sec): The registration maximum time to live setting when registered to the Gatekeeper
- Response Timeout (Q.931)(sec): The maximum time to wait for response from sending call setup signal out
- Connect Timeout (Q.931)(sec): The maximum time to wait for connection (answer) from dialing out the destination number
- DTMF Relay: DTMF transfer type selection
 - RTP relay (RFC 2833): DTMF relay via RTP packet (RFC2833 standard)
 - DTMF transparent: transmitter DTMF over voice channel
 - H.245 Signal input: DTMF relay via H.245 user signal input
 - H.245 Alphanumeric: DTMF relay via H.245 Alphanumeric signal
 - Q.931 User Information: DTMF relay via Q.931 User to user information
- Fax Transport: Fax transport type selection
 - Transparent mode: Transparent mode (by voice packet)
 - T.38 Fax Relay (H.245 mode): T.38 Fax relay (H.323 Annex D)
 - T.38 Fax Relay (Propriety mode): T.38 Fax Relay (propriety mode)
 - FRF11 Fax Relay (Propriety mode): FRF11 Fax Relay (propriety mode)
- Fast Connect Mode: Connection of H.323 call fast mode

- Disable: Don't use Fast Start.
- Enable Fast Start Both Site: Use Fast Start for incoming call and outgoing H.323 calls
- Fast Start-H.323 incoming only: Enable Fast Start for H.323 incoming calls only
- Fast Start-H.323 outgoing only: Enable Fast Start for H.323 outgoing calls only.
- Early H.245: Use Early H.245
- H.245 Tunneling: Transfer the H.245 message over the Q.931 channel
- H.450 Service: Enable the H.450 calls transfer service
- FS Enable 1-6 (Codec Priority 1-6): Enable Fast Start codec selection for each codec
- Inbound DM Group: Digit Manipulation Group for H.323 incoming calls
- Outbound DM Group: Digit Manipulation Group for H.323 outgoing calls

Advance H.323 Configuration:

Start Path: Configuration→H.323 →Advance

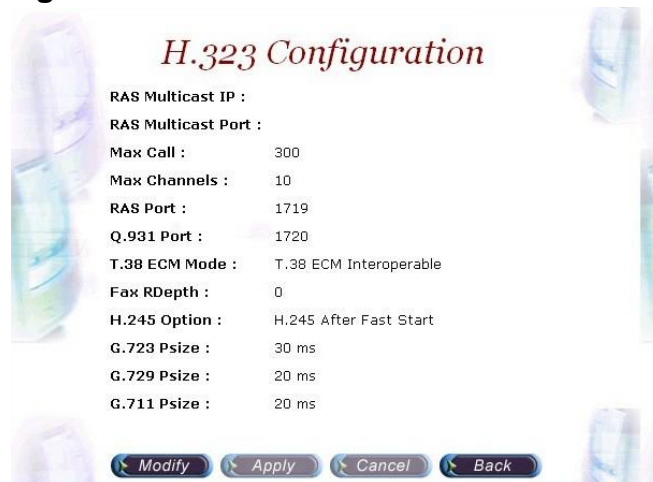


Figure 7.5-2

Advance Parameter Description:

- RAS Multicast IP: RAS multicast IP for Gatekeeper searching
- RAS Multicast Port: RAS multicast Port for Gatekeeper searching
- Max Call: The maximum H.323 calls
- Max Channel: The maximum channel of each H.323 call
- RAS Port: Local RAS port (default: 1719)
- Q.931 Port: Local TCP port number of Q.931
- T.38 ECM Mode: T.38 Error Correction Mode
 - T.38 ECM Interoperable mode
 - T.38 ECM Backward Compatible Mode
- FAX Rdepth: T.38 relay redundancy packet depth for high-speed mode.
- H.245 Option: Separate the H.245 channel in the call of the Fast Start mode or not.
- G.723 Psize: G.723 transmission packet size in ms (default: 30ms)
- G.729 Psize: G.729 transmission packet size in ms (default: 20ms)

- G.711 Psize: G.711 transmission packet size in ms (default: 20ms)

SIP Configuration

Start Path: Configuration→SIP

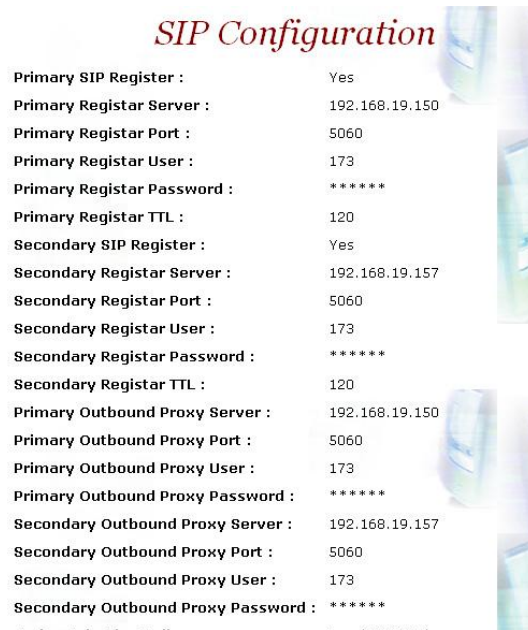


Figure 7.6-1

Basic Parameter Description:

- Primary SIP Register: Register to SIP proxy server or not
 - Yes: Register to proxy server
 - No: Not register to proxy server
- Primary Register Server: SIP register proxy server IP Address
- Primary Register Port: SIP register proxy server port number (default: 1719)
- Primary Register User: SIP register proxy server User ID
- Primary Register Password: SIP register proxy server User Password
- Primary Register TTL: The registration maximum time to live setting when registered to the SIP proxy server
- Secondary SIP Register: Register to SIP proxy server or not
 - Yes: Register to proxy server
 - No: Not register to proxy server
- Secondary Register Server: SIP register proxy server IP Address
- Secondary Register Port: SIP register proxy server port number (default: 1719)
- Secondary Register User: SIP register proxy server User ID
- Secondary Register Password: SIP register proxy server User Password
- Secondary Register TTL: The registration maximum time to live setting when registered to the SIP proxy server

-
- Primary Outbound Proxy Server: The IP address of an outbound Proxy the SIP Stack uses.
 - Primary Outbound Proxy Port: The port of an outbound Proxy the SIP Stack uses
 - Primary Outbound Proxy User: The User ID of an outbound Proxy the SIP Stack uses.
 - Primary Outbound Proxy Password: The password of an outbound Proxy the SIP Stack uses.
 - Secondary Outbound Proxy Server: The IP address of an outbound Proxy the SIP Stack uses.
 - Secondary Outbound Proxy Port: The port of an outbound Proxy the SIP Stack uses
 - Secondary Outbound Proxy User: The User ID of an outbound Proxy the SIP Stack uses.
 - Secondary Outbound Proxy Password: The password of an outbound Proxy the SIP Stack uses.
 - Codec Selection Policy: Selection order to match the remote SDP for codec selection.
 - Local SDP Order: Use local SDP order to match codec
 - Remote SDP Order: Use Remote SDP order to match codec
 - Local Codec 1~4: Codec selection priority (1 to 4) (1: highest, 4: lowest)
 - G.723 Bit Rate Used: G.723.1 high bits rate (6.3k) or low bit rate (5.3k) is used
 - 180 SDP: Set SDP for 180 ring message
 - 183 SDP: Set SDP for 183 call progress indication.
 - DTMF Relay Method: DTMF transport type selection
 - Transparent: transmit DTMF over audio channel
 - SIP INFO: Use SIP INFO Message to relay DTMF
 - RFC2833: Use RFC2833 for DTMF over RTP packet
 - RFC2800 Payload Type: RTP payload type used for RFC2833 DTMF relay
 - Fax Transmission: Fax transparent type selection
 - T.38 Fax Relay: T.38 fax relay
 - Transparent: Transparent mode (by voice packet)
 - Accept Proxy Call Only:
 - Yes: Only call from outbound proxy server is allowed
 - NO: Accept any SIP calls
 - Inbound DM Group: Digit Manipulation Group for SIP incoming calls
 - Outbound DM Group: Digit Manipulation Group for SIP outgoing calls

Advance SIP Configuration:

Start Path: Configuration→SIP →Advance

Advance SIP Configuration

| | |
|--------------------------------|--------|
| TCP Enable : | No |
| Max TCP Connection : | |
| Outbound Use TCP : | |
| Register Use TCP : | |
| TCP Port : | |
| UDP Port : | 5060 |
| Reliable Provision (100rel) : | No |
| Max Call Leg : | 300 |
| Max Transaction : | 1200 |
| Max Register Client : | 2 |
| Send Receive Buffer Size : | 8192 |
| Reject Unsupported Extension : | Yes |
| Message Pool Page Size : | 1024 |
| General Pool Page Size : | 1024 |
| Application Pool Page Size : | 1024 |
| Retransmission T1 : | 2000 |
| Retransmission T2 : | 4000 |
| Retransmission T4 : | 5000 |
| Invite Linger Timer : | 32000 |
| General Linger Timer : | 32000 |
| Registration Timeout : | 100000 |

Figure 7.6-2

Advance Parameter Description:

- TCP Enable: Receive SIP TCP call or not.
- Max TCP Connection: Max Call: The maximum SIP TCP calls.
- Outbound Use TCP: Use SIP TCP for outbound call or not. If it set to no, UDP is used.
- Register Use TCP: Use SIP/TCP to register to SIP register.
- TCP Port: The local TCP port on which the SIP Stack listens.
- UDP Port: The local UDP port on which the SIP Stack listens.
- Reliable Provision: Support PRACK or not (100rel)
- Max Call Leg: The maximum number of call-legs the SIP Stack allocates. You should set this value to the maximum number of call your expect the SIP Stack to handle simultaneously.
- Max Transaction: The maximum number of transactions the SIP Stack allocates. You should set this value to the maximum number of call your expect the SIP Stack to handle simultaneously.
- Max Register Client: The maximum number of Register-Clients the SIP Stack allocates. You should set this value to the maximum number of call your expect the SIP Stack to handle simultaneously.
- Send Receive Buffer Size: The buffer size used by SIP Stack for receiving and sending SIP messages.
- Reject Unsupported Extension: Yes or No
- Message Pool Page Size: Used to hold and process all incoming and outgoing message in the from of encoded messages or message objects. It is recommended that you configure the page size to the average message size your system is expected to message.
- General Pool Page Size: Used by SIP Stack objects, such as call-legs and transaction, to store the internal fields. For example, the call-legs object will store the To, From and Call-ID headers and the local and the remote contact addresses on the general pool pages. The general pool is also used from other activities that demand memory allocation.

-
- Application Pool Page Size: The size of page in the application pool
 - Retransmission T1: T1 determines several timer as defined in RFC3261. For example, When an unreliable transport protocol is used, a Client Invite transaction retransmits requests at an interval that start at T1 seconds and doubles after every retransmission. A Client General transaction retransmits requests at an interval that starts at T1 and doubles until it reaches T2. (Default Value: 500)
 - Retransmission T2: Determines the maximum retransmission interval as defined in RFC3261. For example, when an unreliable transport protocol is used, general requests are retransmitted at an interval which starts at T1 and doubles until reaches T2. If a provisional response is received, retransmission continue but at an interval of T2. (Default Value: 4000)
 - Retransmission T4: T4 represents the amount of time the network takes to clear message between client and server transactions as defined in RFC3261. For example, when working with an unreliable transport protocol, T4 determines the time that UAS waits after receiving an ACK message and before terminating the transaction. (Default Value: 5000)
 - Invite Linger Timer: After sending an ACK for an INVITE final response, a client cannot be sure that the server has received the ACK message; the client should be able to retransmit the ACK upon receiving retransmissions of the final response for inviteLingerTimer milliseconds.
 - General Linger Timer: After a server sends a final response, the server cannot be sure that the client has received the response message. The server should be able to retransmit the response upon receiving retransmissions of the request for generalLingerTimer milliseconds. (Default Value: 32000)
 - Provisional Timer: When a client receives a provisional response, it continues to retransmit the request, but with an interval of provisionalTimer milliseconds.
 - Cancel General No Response Timer: When sending a CANCEL request on a General transaction, the User Agent waits cancelGeneralNoResponseTimer milliseconds before timeout termination if there is no response for the cancelled transaction.
 - Cancel Invite No Response Timer: When sending a CANCEL request on a Invite transaction, the User Agent waits cancellInviteNoResponseTimer milliseconds before timeout termination if there is no response for the cancelled transaction.
 - General Request Timeout Timer: After sending a General request, the User Agent waits for a final response generalRequestTimeoutTimer milliseconds before timeout termination (in this time the User Agent retransmits the request every T1, 2*T1,...T2,...milliseconds)
 - 183 to Alerting: When receive a SIP 183 message from remote site, send Alerting in stead of Call Progress Indicator
 - AutoSend 183: VIP-2100 always send Call Progress Indicator (SIP 183) to VoIP party. It can be used for CAS protocol to enable early media.
 - Behind NAT: Does VIP-2100 is located behind NAT or not
 - Public Signal IP: The static mapped IP for SIP signal

-
- Public Signal Port: The static mapped Port for RTP stream
 - Public RTP IP: The static mapped RTP IP
 - Public RTP Port: The static mapped RTP starting port
 - Public RTP Port Interval: The VIP-2100 has at least 30 RTP channels. Each channel needs 3 ports mapping on NAT Server. The interval is used to calculate the right port for each channel.
 - Overload Redirect: SIP overload redirect when VIP-2100 is not able for service the call
 - Redirect Host: Redirect host URI (format: user@siphost, siphost)
 - Redirect Port: Redirect port number
 - Send 487 When Recv CANCEL: When receive CANCEL form remote site, send "487 Request canceled" or not
 - Caller ID Mode:
 - Local: use VIP-2100 proxy user id
 - Caller: use SIP calling party ANI
 - Receive Hold music source:
 - Auto: Auto determinate to play hold tone based on SIP signaling.
 - Local: Play hold tone locally.
 - On Hold music: Hold tone music file name

Behind NAT Example 1:

| | VIP-2100 | NAT Server Setting |
|------------------------------|----------------------|---------------------------|
| One-by-One Static IP Mapping | 192.168.111.112 | 210.59.163.11 |
| Static Port Mapping | 192.168.111.111:5060 | 210.59.163.10:10000 |

VIP-2100 NAT Enable Setting:

Public Signal IP: 210.59.163.10

Public Signal Port: 10000

Public RTP IP: 210.59.163.11

Public RTP base port: 4000 (same as “**Interface**→**Advance’s Config**”)Public RTP Port Interval: 10

Behind NAT Example 2:

| | VIP-2100 | NAT Server Setting |
|---------------------|--------------------------------------|---------------------------------------|
| Static Port Mapping | 192.168.111.111:5060 | 210.59.163.10:5060 |
| RTP Channel 01 | 192.168.111.112:4000 4001 4002 | 210.59.163.10:10000 10001 10002 |
| RTP Channel 02 | 192.168.111.112:4010 4011 4012 | 210.59.163.10:10003 10004 10005 |
| ⋮ | ⋮ | ⋮ |
| RTP Channel 30 | 192.168.111.112:4310 4311 4312 | 210.59.163.10:10357 10358 10359 |

VIP-2100 NAT Enable Setting:

Public Signal IP: 210.59.163.10

Public Signal Port: 5060

Public RTP IP: 210.59.163.10

Public RTP base port: 10000 (same as “**Interface**→**Advance’s Config**”)Public RTP Port Interval: 0

Access Control

Access Control list can be used to filter the calls forms the IP Network, DNIS, and ANI. ***It must be used in call flow edit to take effect.***

IP ACL

Start Path: Configuration→Access Control→IP ACL

IP ACL Configuration

| IP Network | Access Mode |
|----------------|-------------|
| 192.168.19.0 | allowed |
| 192.168.19.222 | disallow |
| 192.168.19.25 | disallow |

Figure 7.7-1

Parameters:

- IP Network: IP Address or prefix used to be filtered
- Access Mode:
 - Allow: the inputs IP Network are allowed for calls.
 - Disallow: The inputs IP Network are disallowed for calls.

● **Note: If in the system has both allowance and disallowance setup, the system will check allowance first and disallowance later. If only disallowance inputted all IP will allow to work except disallowed network. If only allowance inputted, only those IP from allowance list will work.**

ANI ACL

ANI ACL

Start Path: Configuration→Access Control→ANI ACL

ANI ACL Configuration

| ANI | Access Mode |
|-----------|-------------|
| 2 | allowed |
| 282265555 | allowed |
| 282265699 | disallow |

Figure 7.7-2

Parameters:

- ANI: Calling party number used to filter
- Access Mode:
 - Allow: the calling numbers are allowed for calls
 - Disallow: The calling numbers are disallowed for calls

● **Note: If in the system has both allowance and disallowance setup, the system will check allowance first and disallowance later. If only disallowance inputted all ANI will allow to work except disallowed ANI. If only allowance inputted, only those ANI from allowance list will work.**

DNIS ACL

Start Path: **Configuration**→**Access Control**→**DNIS ACL**

DNIS ACL Configuration

| DNIS | Access Mode |
|------|-------------|
| 886 | allowed |
| 8867 | disallow |

Figure 7.7-3

Parameters:

- DNIS: Called party number used for filter
- Access Mode:
 - Allow: The called numbers are allowed for calls
 - Disallow: The called numbers are disallowed for calls

● **Note:** *If in the system has both allowance and disallowance setup, the system will check allowance first and disallowance later. If only disallowance inputted all DNIS will allow to work except disallowed DNIS. If only allowance inputted, only those DNIS from allowance list will work.*

User ACL

User ACL is used to store subscriber information when internal AAA is enabled.

Start Path: **Configuration**→**Access Control**→**User ACL**

User ACL Configuration

| User ID | Password | Prepaid Point | Status |
|---------|----------|---------------|----------|
| 0001 | ***** | 9999 | Active |
| 0002 | ***** | 48168 | Active |
| 0003 | ***** | 48170 | Inactive |
| 0004 | ***** | Postpaid | Inactive |
| 0005 | ***** | 48124 | Active |
| 0006 | ***** | 846988 | Active |
| 0007 | ***** | 944996 | Active |
| 0008 | ***** | 79737851 | Active |
| 0009 | ***** | 9735867 | Active |
| 0010 | ***** | Postpaid | Inactive |

Page 1 Next

New Modify Delete Calling Rate Search

Figure 7.7-4

Parameters:

- User: User ID (0~9, *#)
- Password: Password (0~9, *#)
- Prepaid Point: Allowed prepaid point (When prepaid point is used, the system will deduct it automatically base on the rate setting.)
 - Postpaid: postpaid user
- Status:
 - Active: User is active
 - Inactive: User is inactive

● **Note:** *1. IP Authentication method must be set to “internal AAA” to talk effect.*

New a Calling Rate: The calling rate will have different appearance for different calling rate policy set in RADIUS configuration.

Click **Calling Rate** button to add a new calling rate as figure 7.7-5.

Calling Rate

| TEL Prefix | Calling Rate(Point) |
|------------|---------------------|
| * | 53 |
| 8862 | 96 |
| 8864 | 88 |
| 8867 | 60 |

Figure 7.7-5

Point per Second calling rate:

Calling rate (point per second) is used to convert prepaid point into prepaid time in second. For example, you can set calling rate to 5 for “100” prefix. When a caller, which has 200 prepaid point, calls “100xxxx”, the max talk time will be $200/5=40$ seconds. If a calling rate is set to “0”, it means free charge.

New a Calling Rate (Second per Point):

Click **Calling Rate** button to add a new calling rate as figure 7.7-6.

Calling Rate

| TEL Prefix | Calling Rate(Second) | Charge Point |
|------------|----------------------|--------------|
| * | 60 | 1.5 |
| 8862 | 60 | 3.5 |
| 8864 | 60 | 3.3 |
| 8867 | 60 | 3.9 |

Figure 7.7-6

Second per Point calling rate:

. Calling rate (Second per point) is used to convert prepaid point into prepaid second in time. For example, you can set calling rate (Second) to 6, charge point to 1 for “113” prefix. It means that every 6 seconds charge 1 point. When a caller, which has 200 prepaid point, calls “113xxxx”, the max talk time, will be $200*6/1=1200$ seconds.

J Note: Tel prefix * is used as a default rate, you need to create it to work.

Search Condition:

You can search a user by User ID, Prepaid or Postpaid condition as figure 7.7-7.

The screenshot shows a web browser window with the title "Search User ACL - Microsoft Internet Explorer". The main content area displays a form titled "Search Condition". The form includes a text input field for "User ID" with the value "65856". Below it is a checkbox labeled "Postpaid" which is currently unchecked. Underneath is a "Prepaid Point" field with a dropdown arrow and an empty input box. At the bottom of the form are two buttons: "Search" and "Cancel".

Figure 7.7-7

Number Replace

The purpose of “Number Replace” is to replace called number or calling number for PSTN or IP. **It must be used in call flow to take effect.**

Step 1: It is useful for real PSTN number to virtual VoIP number replacement. Click **Number Replace** to add a new Number Replace Group, add as figure 7.8-1.

Number Replace Configuration

| Group ID | Description |
|----------|----------------|
| 1 | SIP IN |
| 2 | SIP Proxy 2 in |
| 3 | H323 GK in |



Figure 7.8-1

Field Description:

- Group ID: 1 (Number Replace Group identify)
- Description: SIP in

Step 2: Click the New created NR and **Detail** button to add digits setting as figure 7.8-2.

Number Replace Detail

Group : 2 Description : SIP Proxy 2 in

| Original Number | Target Type | Target Number |
|-----------------|-------------|---------------|
| 070899 | ANI | 99998888 |
| 2333 | ANI | 99998888 |
| 6986 | ANI | 99998888 |



Figure 7.8-2

Field Description:

- Original Number: Original number filter
- Target Type: ANI or DNIS
- Target Number: The ANI or DNIS are change to target number

Routing Plan

The purpose of **Routing Plan** is to select T1/E1 trunk and channels by your preference when there is a call from IP side to PSTN side. **The PSTN must be used in call flow edit or line hunting component to take effect.**

Hunting Group

Start Path: Configuration→Routing Plan→Hunting Group

Hunting Group Configuration

| Group ID | Description | Hunting Method |
|----------|---------------------|----------------|
| 1 | Chan 1 Trunk 00 | Round Robin |
| 2 | Chan 2 Trunk 00 | Round Robin |
| 3 | CHT Trunk 01 | Priority |
| 4 | FET Trunk 02 and 03 | Random |

Figure 7.9-1

Parameters:

- Group ID: Hunting Group ID
- Description: Description of Hunting Group
- Hunting Method: Route selection
 - Random: Random select a trunk within this hunting group
 - Priority: Select a trunk by priority. (Priority 1 has lowest priority; 9 has highest priority)
 - Round Robin: Call is hunting rotationally

Start Path: Configuration→Routing Plan→Hunting Group→Detail

Hunting Group Detail

Group : 4 Description : FET Trunk 02 and 03

| Interface ID | Trunk ID | Priority | Channel Mask |
|--------------------------|----------|----------|--------------|
| 0 · VoIP 4 E1/T1 Interfa | 2 · 2 | 8 | 0x0000ffff |
| 0 · VoIP 4 E1/T1 Interfa | 3 · 3 | 9 | 0xffffffff |

Figure 7.9-2

Parameters:

Group: 4 Description: FET Trunk 02 and 03

- Interface ID: Interface ID
- Trunk ID: trunk id for group 4
- Priority: Trunk priority
- Channel Mask: Channel mask for incoming or outgoing calls (refer [T1/E1 Trunk Configuration](#))

J Note: When a Route Plan channels mask is cooperated to trunk channel mask to decide the channel availability 17~31 channels are available:

Example 1:

Trunk ID: 0 channel mask: 0xffffffff

Route Plan channel mask: 0x0000ffff

Available channel: 0x0000ffff (17~31) channels.

Example 2:

Trunk ID: 0 channel mask: 0xffff0000

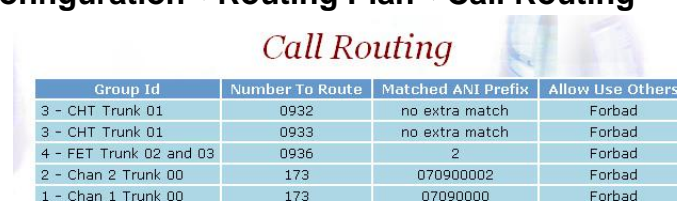
Route Plan channel mask: 0xffc00000

Available channel: 0xffc00000 (1~9) channels.

Call Routing

The call routing can be used for hunting a PSTN trunk by prefix.

Start Path: Configuration→Routing Plan→Call Routing



The screenshot shows a table titled "Call Routing" with the following data:

| Group Id | Number To Route | Matched ANI Prefix | Allow Use Others |
|-------------------------|-----------------|--------------------|------------------|
| 3 - CHT Trunk 01 | 0932 | no extra match | Forbad |
| 3 - CHT Trunk 01 | 0933 | no extra match | Forbad |
| 4 - FET Trunk 02 and 03 | 0936 | 2 | Forbad |
| 2 - Chan 2 Trunk 00 | 173 | 070900002 | Forbad |
| 1 - Chan 1 Trunk 00 | 173 | 070900000 | Forbad |

Figure 7.9-3

Parameters:

- Group ID: Select the T1/E1 according to the selection of Hunting Group ID when dialed number is matched
- Number To Route: The dialed telephone number to be matched
- Matched ANI Prefix: Calling party number used to filter
- Allow Use Others: To select other T1/E1 trunk when all trunk are busy at your desired Hunting Group.
 - Allowed: The call will use other T1/E1 trunks which is not belong to the selected hunting group
 - Forbad: The call will be disconnected immediately

Radius Setting

When you have an external RADIUS server to do the AAA (Authorization, Authentication and Accounting), enter the correct parameter to the Radius setting. **It must be used in call flow to take effect.**

Start Path: Configuration→Radius Setting

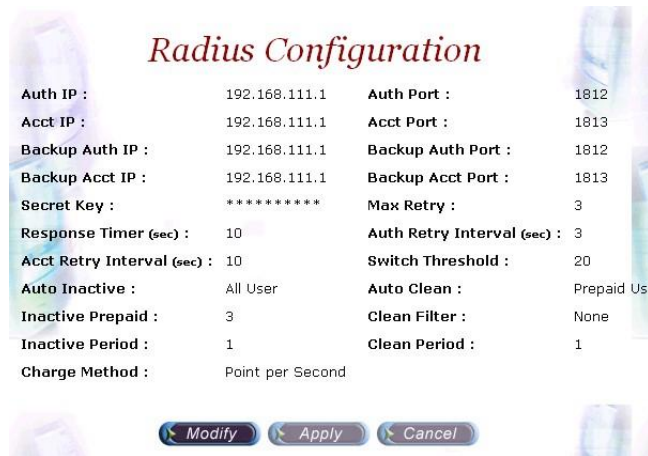


Figure 7.10-1

Parameters:

- Auth IP: Radius Authentication Server IP address (default)
- Auth Port: Radius Authentication Server Port
- Acct IP: Radius Account Server IP address
- Acct Port: Radius Account Server Port
- Backup Auth IP: Backup Radius Authentication Server IP address
- Backup Auth Port: Back Radius Authentication Server Port
- Backup Acct IP: Back Radius Account Server IP address
- Backup Acct Port: Back Radius Account Server Port
- Secret Key: The shared secret key with RADIUS Server
- Max Retry: The maximum retry times
- Response Time (sec): The maximum wait for response time from RADIUS Server
- Auth Retry Interval (sec): The internal to resend the Authentication packet to RADIUS Server.
- Acc Retry Interval (sec): The internal to resend the Account packet to RADIUS Server.
- Switch Threshold: Switch to alternate RADIUS Server when failures are occurred more than switch threshold.
- Auto Inactive: Auto inactive an unused or not
 - Disable: Don't auto inactive
 - Prepaid User: Auto inactive prepaid user only
 - Postpaid User: Auto inactive postpaid user only
 - All User: Auto inactive all unused user
- Inactive prepaid: The minimum credit point threshold for a prepaid user to be inactivated
- Inactive Period: The max unaccess days for a postpaid user to be inactivated
- Charge Method: Billing charge method selection
 - Point per Second: $\text{Point} / \text{calling rate} = \text{seconds}$
 - Second per Point: $\text{Point} * \text{calling rate} / \text{charge point} = \text{seconds}$
- Auto Clean: Auto clean the inactive user

-
- Disable: Don't auto clean inactive user
 - Prepaid User: Auto clean prepaid user only
 - Postpaid User: Auto clean postpaid user only
 - All User: Auto clean inactive user
 - Clean Filter: Auto clean filter
 - None: Auto clean users exceed clean period without access the network
 - Inactive: Auto clean only to inactive users
 - Clean Period: The maximum unaccess days to clean up. When the clean filter is set Inactive, the unaccess day is start counting when the user is inactivated

Apply Change

1. Some of modification needs to restart system before it is effective to system operation. "Apply the change" shows "**The change you made need to restart the system for apply please confirm to restart or do it later?**" Click on OK button to reboot the system.

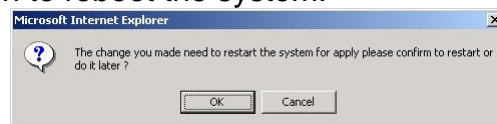


Figure 7.11-1

2. For the modification can be changed to fly, "Apply the Change" shows "**Are you sure to apply the running system?**" Click on OK button to take effecting.



Figure 7.11-2

Chapter 8 System Control

System

Start path: Click Control→System



Figure 8.1-1

Parameter:

- Soft Reset: Soft Reset at VIP-2100
- Restart: Restart the VIP-2100
- Shutdown: Shutdown the VIP-2100

System Time

Timezone Setting

Step 1: If you would like to use timezone, click **Timezone** button to setup the system timezone as figure 8.2-1.



Figure 8.2-1

Standard:

Step 2: Select the **Standard** option to setup the system predefined time zone as figure 8.2-2



Figure 8.2-2

Parameter:

- Time Zone:
 - Standard: Use a predefined standard time zone (**Refer [Timezone to Country Mapping List](#)**)
 - Customize: Use a user defined time zone
- Auto Daylight Saving: Auto adjust daylight saving time or not

User defined timezone :

Step 3: Select the **Customized** option and enter the time zone bias to set a user defined timezone as figure 8.2-3

Time Zone Control

Time Zone : Standard :
 Customize :

Daylight Bias :

Daylight Start : Month : Week Day :
Apply Week : Hour :

Standard Start : Month : Week Day :
Apply Week : Hour :

Figure 8.2-3

Parameter:

- Daylight Bias: The offset added to the Bias when the time zone is in daylight saving time
- Daylight Start: The date that a time zone enters daylight time
 - Month: 01 to 12
 - Week Day: Sunday to Saturday
 - Apply Week (Day:01 to 05, Specifies the occurrence of day in the month; 01 = First occurrence of day, 02 = Second occurrence of day, ...and 05 = Last occurrence of day)
 - Hour: 00 to 23
- Standard Start: The date that a time zone enters daylight time
 - Month: 01 to 12
 - Week Day: Sunday to Saturday
 - Apply Week (Day:01 to 05, Specifies the occurrence of day in the month; 01 = First occurrence of day, 02 = Second occurrence of day, ...and 05 = Last occurrence of day)
 - Hour: 00 to 23

Network

DNS Server Setting:

Step 1: After successfully logon to the system, we need to change the network configuration. Click **Control**→**Network** to setup the network parameters as figure 8.3-1.

Network Control

Use DHCP
 Use fixed IP address

IP Address : 192 .168 .5 .113
IP Netmask : 255 .255 .0 .0
IP Gateway : 192 .168 .1 .254

DNS Setup

Primary DNS Server : 192 .168 .5 .1
Secondary DNS Server : 168 .95 .1 .1
Host name : wg5200
Domain name : ybtvnet.net
Dynamic DNS Registration : Yes No

Figure 8.3-1

Parameter:

- Primary DNS Server: Primary DNS Server IP network
- Secondary DNS Server: Secondary DNS Server IP network
- Host Name: Host name used to register to DNS Server
- Domain Name: Domain name used to
- Dynamic DNS Registration: Enable Dynamic DNS registration or not

SNMP

Start path: Click **Control**→**SNMP**→**Community**

SNMP Community Management

| Community Name | Access Right |
|----------------|--------------|
| public | Read/Write |

Figure 8.4-1

Parameter:

- Community Name: Community name for network manager system accessing
- Access Rights: Giving access right to the community

Start path: Click Control→SNMP→Trap



Figure 8.4-2

Parameter:

- Trap Community: Trap community name for NMS
- Trap Host: Trap host IP address

J Note: It takes around 1-minute to update SNMP configuration and display successful message.

Prompt Manager

Start path: Click Control→Prompt Manager

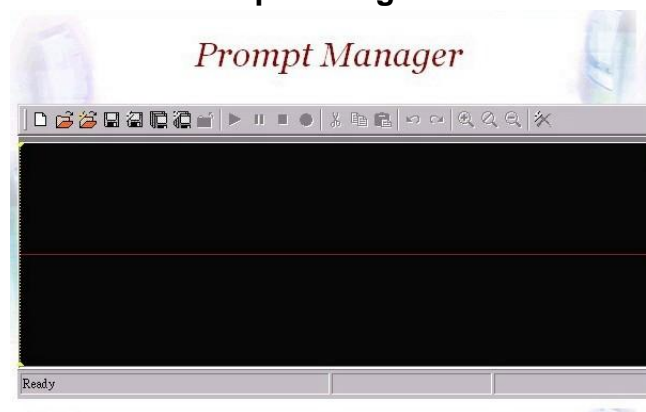


Figure 8.4-1

☉ **Note:**

1. You must have a sound card in your PC to record the voice. You need to set Network security in order to execute this recording. **Click Tool→Internet Option→Security→Custom Level.**



2. Enable the following security to active setting:

Voice prompt editor:

- **Download unsigned ActiveX control: Enable**
- **Initialize and script ActiveX control not marked as safe: Enable**



 **New**,  **Record:**

Step 1: Make sure you have installed microphone or other device when you want to record, Click **New** and **Record** buttons to record as figure 8.4-2.

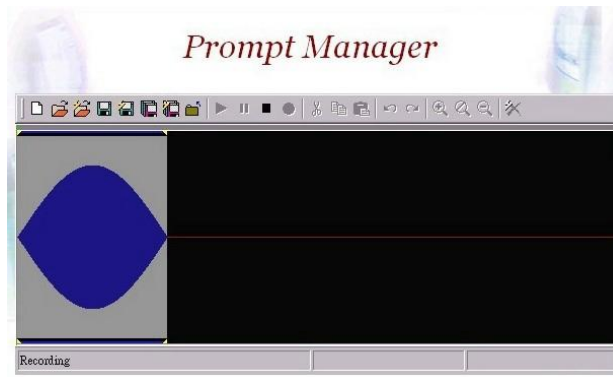


Figure 8.4-2

 **Stop**,  **Pause**,  **Play:**

Step 2: Click **Stop** or **Pause** button to stop record, and click **Play** button to listen the voice prompt as figure 8.4-3.

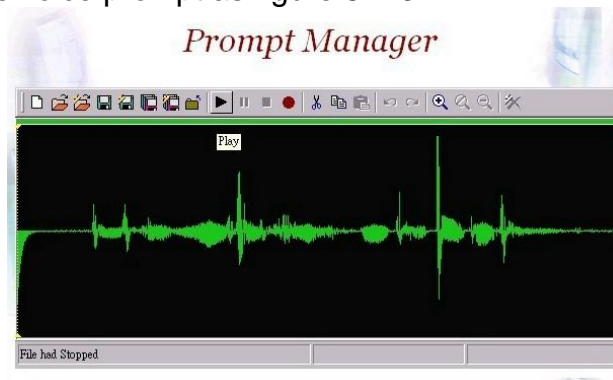


Figure 8.4-3

 **Save:**

Step 3: Click Save button to saving the voice file at local path, and the screen shows **Please input the file path and file name!!** (i.e. c:\irene_test.raw) as figure 8.4-4.

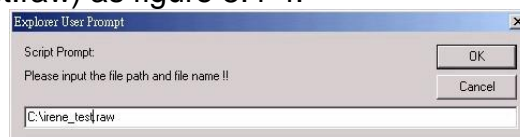


Figure 8.4-4

 **Save Remote File:**

Step 4: Click **Save Remote File** to saving the voice file at VIP-2100, and the screen shows **“please input the file path and file name!!”** (i.e. 9999.raw) as figure 8.4-5



Figure 8.4-5

● **Note:** The file name must be “.raw” file format.



Open Remote File:

Step 5: Click **Open Remote File** button to open voice file at VIP-2100 and screen shows **Voice File List** as figure 8.4-6.



Figure 8.4-6



Open:

Step 6: Click **Open** button to open local host voice file and screen shows **Choose File** as figure 8.4-7.

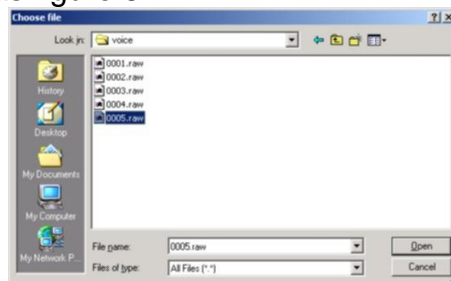


Figure 8.4-7



Close:

Step 7: Click **Close** button to close the voice file as figure 8.4-8.

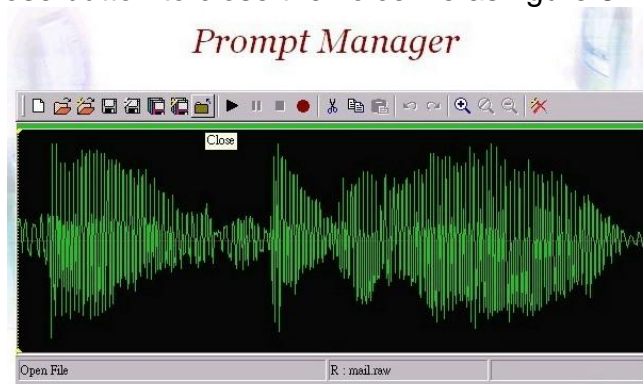


Figure 8.4-8

 **Copy:**

Step 8: Select the desired voice range and click **Copy** button as figure 8.4-9



8.4-9

 **Paste:**

Step 9: Click **Paste** button to paste the voice range as figure 8.4-10.

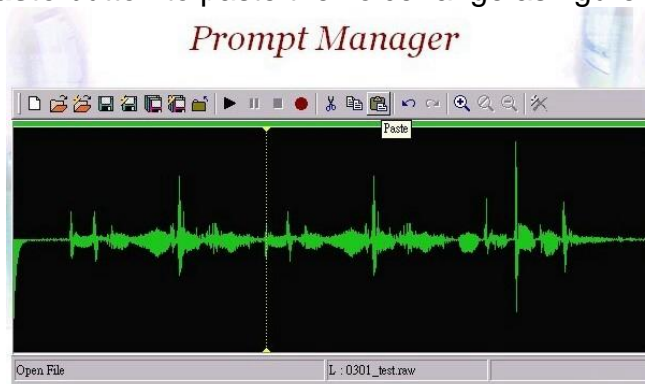


Figure 8.4-10

 **Cut:**

Step 10: Select the desired voice range and click **Cut** button as figure 8.4-11.



Figure 8.4-11

 **Save As:** Refer the Section "[Save](#)"

 **Save Remote As:** Refer the Section “[Save Remote File](#)”

 **Undo:**

Step 13: Click **Undo** button to return modification, you can see the configuration that haven't be changed as figure 8.4-12.

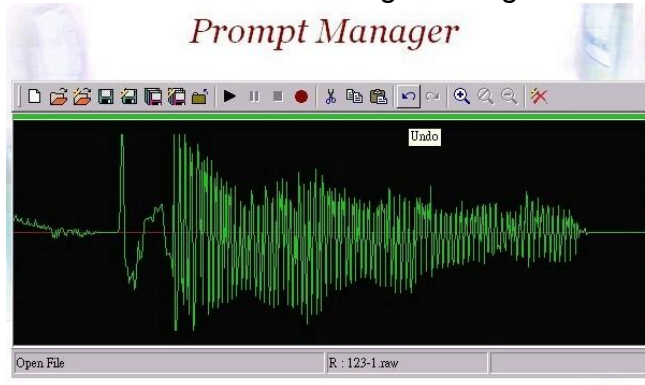


Figure 8.4-12

 **Redo:** Refer Section “[Undo](#)”

 **Zoom**  **Zoom In**  **Zoom Out:**

Step 14: Select the desired voice range click **Zoom** button as figure 8.4-13.



Figure 8.4-13

Step 15: The screen shows the zoom out voice file range as figure 8.4-14.



Figure 8.4-14



Delete Remote file:

Step 16: Click **Delete Remote file** button to delete remote voice file as figure 8.4-15.



Figure 8.4-15

Call Flow Editor

Please refer section "[Call Flow Editor](#)"

Account Manager

Please refer section "[Account Manager](#)"

Upgrade

Step 1: Click "Control→Upgrade" to upgrade the software as figure 7.5-1.

Upgrade Control



Figure 7.5-1

Field Description:

- File Name: Upload the software file name
- Upload: Remote Upload the software at VIP-2100
- Apply: Remote apply the upload at VIP-2100

Relogin

Please refer section "[Relogin](#)"

Chapter 9 System Monitor

It provides a way to monitor the system status.

Line Summary Status

Show channel summary status.

Start Path: Monitor→Line Summary Status

| Line ID | Talk Time | Successful Calls | Unsuccessful Calls |
|---------|-----------|------------------|--------------------|
| 0-0-00 | 173 | 3 | 0 |
| 0-0-01 | 172 | 3 | 0 |
| 0-0-02 | 173 | 3 | 0 |
| 0-0-03 | 173 | 3 | 0 |
| 0-0-04 | 173 | 3 | 0 |
| 0-0-05 | 173 | 3 | 0 |
| 0-0-06 | 173 | 3 | 0 |
| 0-0-07 | 173 | 3 | 0 |
| 0-0-08 | 173 | 3 | 0 |
| 0-0-09 | 173 | 3 | 0 |

Figure 9.1-1

Field Description:

- Refresh Interval (second): Refresh interval time (1, 5, 10 seconds)
- Line ID: Line ID (format: Interface: trunk: channel)
- Talk Time: Total conversation time
- Successfully calls: Total successfully calls (connected calls)
- Unsuccessfully calls: Total unsuccessfully calls (unconnected calls)

See the line detail:

Selection the line and click **Detail** button as figure 9.1-2.

| | | | |
|------------------|-----------------|---------------|-----------|
| Line ID : | 0-0-01 | Line Status : | connected |
| Call Originate : | PSTN | ANI : | 1001 |
| DNIS : | 00320032113 | PSTN Status : | connected |
| VOIP Status : | call proceeding | Escape Time : | 00:00:01 |

Figure 9.1-2

Refer to line detail for field description

Line Detail

Show detail channel status.

Start Path: **Monitor**→**Line Detail**

Line Detail Status

Refresh Interval (second) : 3

| Line ID | Line Status | Call Originate | ANI String | DNIS String | PSTN Status | VOIP Status | Escape Time |
|---------|-------------|----------------|------------|-------------|-------------|-----------------|-------------|
| 0-0-00 | connected | PSTN | 1001 | 00900090113 | connected | call proceeding | 00:00:01 |
| 0-0-01 | connected | PSTN | 1001 | 00320032113 | connected | connected | 00:00:04 |
| 0-0-02 | connected | PSTN | 1001 | 00330033113 | connected | connected | 00:00:03 |
| 0-0-03 | connected | PSTN | 1001 | 00340034113 | connected | connected | 00:00:03 |
| 0-0-04 | connected | PSTN | 1001 | 00350035113 | connected | connected | 00:00:01 |
| 0-0-05 | connected | PSTN | 1001 | 00360036113 | connected | connected | 00:00:03 |
| 0-0-06 | connected | PSTN | 1001 | 00370037113 | connected | call proceeding | 00:00:01 |
| 0-0-07 | connected | PSTN | 1001 | 00380038113 | connected | connected | 00:00:04 |
| 0-0-08 | connected | PSTN | 1001 | 00390039113 | connected | connected | 00:00:02 |
| 0-0-09 | connected | PSTN | 1001 | 00400040113 | connected | connected | 00:00:03 |

1 - 2 - 3 - 4 - 5 - 6 - 7 - 8 - 9 - 10 - 11 - 12

Detail

Figure 9.2-1

Field Description:

- Refresh Interval (second): Refresh interval time (1, 5, 10 seconds)
- Line ID: Line ID
- Line Status: Current time status
- Call Originate: Call originate site
- ANI String: Calling party number
- DNIS String: Called party number
- PSTN Status: PSTN site status
- VoIP Status: IP site status
- Escape Time: Talk time

Event Log

Show system log status.

Start Path: **Configuration**→**Event Log**

Event Log Viewer

| Type | Date | Time | Source | Category | Event ID |
|-------------|------------|----------|---------|----------|----------|
| warning | 2004/09/21 | 16:48:59 | Gateway | VOIP GW | 8702 |
| information | 2004/09/21 | 16:48:59 | Gateway | VOIP GW | 9501 |
| warning | 2004/09/21 | 16:48:59 | Gateway | VOIP GW | 8703 |
| warning | 2004/09/21 | 16:48:59 | Gateway | VOIP GW | 8703 |
| warning | 2004/09/21 | 16:48:59 | Gateway | VOIP GW | 8703 |
| warning | 2004/09/21 | 16:48:58 | Gateway | VOIP GW | 8700 |
| warning | 2004/09/21 | 16:48:58 | Gateway | VOIP GW | 8700 |
| warning | 2004/09/21 | 16:48:58 | Gateway | VOIP GW | 8700 |
| information | 2004/09/21 | 16:48:58 | Gateway | VOIP GW | 9504 |
| information | 2004/09/21 | 16:48:58 | Gateway | VOIP GW | 9503 |

Figure 9.3-1

Field Description:

- Type: Event Log type
 - Information
 - Warring
 - Error
- Date: Event created date

- Time: Event created time
- Source: Executable program
- Category: Event type (none, welltech Sys...)
- Event ID: Event Log

● **Note:** You can click **Clear** button to clear all event log.

See the detail event log:

Double click the log or select the log and click detail to see the log detail.



Figure 9.3-2

Event Description:

| Event ID | Event Description | Description |
|----------|--|--|
| 8003 | [GK]: [xxx.xxx.xxx.xxx:xxxx] not found or registered failure | Failed to register to H323 Gatekeeper |
| | [SIP Register]: [xxx.xxx.xxx.xxx:xxxx] not found or registered failure | Failed to registered to SIP Registrar Server |
| 8700 | VoIP Gateway application on the fly change | On the fly change (system change) |
| 8703 | [0]: evt: D CHANNEL STATUS: runkld=3, Status=1, Comment=", LOS=14, LOF=0, RAI=108, AIS=145, RAI_CRC=-1 | D Channel and Trunk ID (ID: 0) not available |
| 9500 | Gateway application started | VoIP Gateway program start |
| 9500 | AAA Mgr application started | AAA Manager program start |
| 9500 | TelSvr application started | Telnet Server program start |
| 9501 | VoIP Board (0) started | Interface (ID:0) start |
| 9502 | H323 stack started | H323 stack start |
| 9503 | H323 GK [xxx.xxx.xxx.xxx:xxxx] found & registered | Registered to H323 Gatekeeper |
| | [SIP Register]: [xxx.xxx.xxx:xxxx] Found & Registered. | Registered to SIP Registrar Server |
| 9504 | PSTN trunk (0) alarm clear | Connect to PSTN |
| 9505 | [0]: evt: D CHANNEL STATUS: TrunkId=3, Status=0, Comment=", LOS=29, LOF=67, RAI=31, AIS=1, RAI_CRC=-1 | D Channel and Trunk ID (ID:0) available |
| 9600 | SNTP client application started | Failed / Success to connect SNTP server |

Debug Info

Start Path: Click “**Monitor**→**Debug Info**”



Figure 9.4-1

Filed Description:

- Get Log: Get debug log (-1~999)
- Search: Search debug logs
- Clear: Clear log

Ping

You can use the “Ping” to check an IP is active or not.

Start Path: **Configuration**→**Ping**

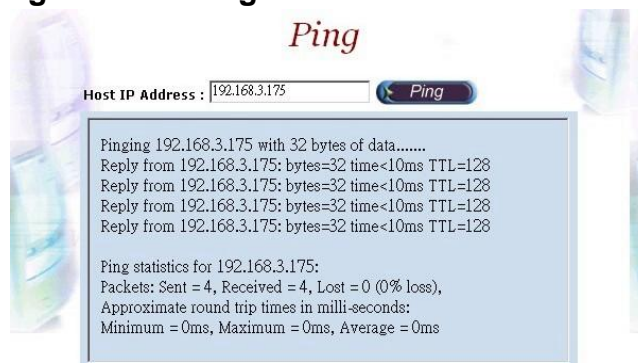


Figure 9.5-1

Field Description:

- Host IP Address: The IP address to ping

Chapter 10 Telnet & RS-232 Configuration

VIP-2100 also can support to be managed by Telnet or Console port (RS-232) for basic operations.

Interface:

- ✓ Network: TCP/IP Telnet
- ✓ RS232:
 - Connect using: COM1
 - Baud Rate: 9600
 - Data bits: 8
 - Parity: None
 - Stop bits: 1
 - Flow Control: None
 - Wire: Null modem line (crossed)

Logon VIP-2100 by Telnet

Use Windows build-in Hyper Terminal or other telnet terminal emulator to login (e.g. telnet 192.168.111.111:10086). User ID & password will be required for login (default login user id: admin, password: admin & user id: root, password: root).

Command List:

| Command | Description |
|-----------|---|
| echo | Auto echo on or off |
| eventlog | Clean or show system log message |
| exit | Quit the current session |
| ipconfig | Configure or show network information |
| ping | Check an IP address is available or not |
| reboot | Reboot |
| reset | Soft-reset |
| shutdown | Shutdown |
| time | Reset or show system time. |
| timezone | Setup or show system timezone |
| useradmin | Manage user account. |
| help & ? | View command list |

Echo: auto echo on or not

| Command | Purpose |
|-----------------|--|
| [root#]echo ? | Usage: echo on/off Example: echo on |
| [root#]echo on | Echo is on |
| [root#]echo off | Echo is off (default value) |

Eventlog: show system log message

| Command | Purpose |
|------------------------|---|
| [root#]eventlog ? | Usage: eventlog [-clear] Example: eventlog eventlog -clear |
| [root#]eventlog | Show system eventlog: Eventlog example: Time: 2003-06-19 20:15:17 Event ID: 8700 Type: Warning Source : wellgate5x00 Description: [0]: evt: TRUNK ALARM: TrunkId=3 Time: 2003-06-19 20:15:17 Event ID: 8700 Type: Warning Source : wellgate5x00 Description: [0]: evt: TRUNK ALARM: TrunkId=2 Time: 2003-06-19 20:15:14 Event ID: 9501 Type: Information Source : wellgate5x00 Description: [0]: evt: BOARD STARTED: SLOT:8 Press any key to continue or press 'Q' to quit |
| [root#]eventlog -clear | Clear all event log |

Exit: Quit the current session

| Command | Purpose |
|-------------|--------------------------|
| [root#]exit | Quit the current session |

Ipconfig: Configuration or show network information

| Command | Purpose |
|--------------------------------|--|
| [root#] ipconfig ? | Usage: ipconfig [-delete dns] [-dhcp] [-dns IPAddress1 IPAddress2] [-ip IPAddress -mask Mask -gateway Gateway] Example : ipconfig -ip 192.168.111.111 -mask 255.255.0.0 -gateway 192.168.1.254 : ipconfig -dhcp : ipconfig -dns 192.168.1.1 : ipconfig -delete dns |
| [root#]ipconfig | Show current network configuration USE FIXED IP (or DHCP) IP Address : 192.168.5.113 Subnet Mask : 255.255.0.0 Default Gateway : 192.168.1.254 DNS Servers : 192.168.5.1 168.95.1.1 |
| [root#]ipconfig -delete dns | Delete the DNS servers setting USE FIXED IP IP Address : 192.168.5.113 Subnet Mask : 255.255.0.0 Default Gateway : 192.168.1.254 DNS Servers : |
| [root#]ipconfig -dhcp | Enable DHCP USE DHCP |

| | |
|--|---|
| | IP Address : 192.168.5.10 Subnet Mask : 255.255.0.0 Default Gateway : 192.168.1.254 DNS Servers : 192.168.5.1 168.95.1.1 |
| <pre>[root#]ipconfig -ip 61.220.126 28 -mask 255.255.0.224 -gateway 61.220.126.1</pre> | Use fixed network configuration USE FIXED IP IP Address : 61.220.126.28 Subnet Mask : 255.255.255.1 Default Gateway : 61.220.126.254 DNS Servers : |
| <pre>[root#]ipconfig -ip 61.220.126.115</pre> | Changes IP address only. USE FIXED IP IP Address : 61.220.126.115 Subnet Mask : 255.255.255.1 Default Gateway : 61.220.126.254 DNS Servers : |
| <pre>[root#]ipconfig -dns 210.59.126.53</pre> | Changes DNS configuration only. USE FIXED IP IP Address : 61.220.126.115 Subnet Mask : 255.255.255.1 Default Gateway : 61.220.126.254 DNS Servers : 210.59.126.53 |

Ping: Check an IP address is available or not

| Command | Purpose |
|-------------------------------------|--|
| <pre>[root#] ping ?</pre> | Usage: ping IP. Example: ping 127.0.0.1 |
| <pre>[root#]ping 61.220.126.1</pre> | Ping result Reply from 61.220.126.1 bytes=64 time=1ms TTL=29 Reply from 61.220.126.1 bytes=64 time=1ms TTL=29 Reply from 61.220.126.1 bytes=64 time=1ms TTL=29 Reply from 61.220.126.1 bytes=64 time=1ms TTL=29 |

Reboot:

| Command | Purpose |
|--|---|
| <pre>[root#] reboot ?</pre> | Reboot System Are You Sure? (Y/N) |
| <pre>[root#]reboot Are You Sure?(Y/N)y</pre> | VIP-2100 are rebooting |

Shutdown:

| Command | Purpose |
|--|---|
| <pre>[root#] shutdown ?</pre> | Shutdown System Are You Sure? (Y/N) |
| <pre>[root#]shutdown Are You Sure?(Y/N)y</pre> | VIP-2100 are shutting down |

Reset:

| Command | Purpose |
|---|---|
| <pre>[root#] reset ?</pre> | Soft reset System Are You Sure? (Y/N) |
| <pre>[root#]reset Are You Sure?(Y/N)y</pre> | |

Time: Reset or show system time

| Command | Purpose |
|------------------------------------|--|
| [root#] time ? | Usage : time YYYY-MM-DD HH:NN:SS Example : Time 2002-01-01 12:00:00 |
| [root#]time | Show current time The current time is 2003-06-20 15:17:30 |
| [root#]time 2003-07-29 23:14:53 | Change system bios time |

Timezone: Setup or show system timzone

| Command | Purpose |
|--------------------|---|
| [root#] timezone ? | Fixed Zone List: 01. Afghanistan Standard Time 03. Arab Standard Time 05. Arabic Standard Time 07. AUS Central Standard Time 09. Azores Standard Time 11. Cape Verde Standard Time 13. Cen. Australia Standard Time 15. Central Asia Standard Time 17. Central European Standard Time 19. Central Standard Time 21. Dateline Standard Time 23. E. Australia Standard Time 25. E. South America Standard Time 27. Egypt Standard Time 29. Fiji Standard Time 31. GMT Standard Time 33. Greenwich Standard Time 35. Hawaiian Standard Time 37. Iran Standard Time 39. Korea Standard Time 41. Mexico Standard Time 2 43. Mountain Standard Time 45. N. Central Asia Standard Time 47. New Zealand Standard Time 49. North Asia East Standard Time 51. Pacific SA Standard Time 53. Romance Standard Time 55. SA Eastern Standard Time 57. SA Western Standard Time 59. SE Asia Standard Time 61. South Africa Standard Time 63. Taipei Standard Time 65. Tokyo Standard Time 67. US Eastern Standard Time 69. Vladivostok Standard Time 71. W. Central Africa Standard Time 73. West Asia Standard Time 75. Yakutsk Standard Time 02. Alaskan Standard Time 04. Arabian Standard Time 06. Atlantic Standard Time 08. AUS Eastern Standard Time 10. Canada Central Standard Time 12. Caucasus Standard Time 14. Central America Standard Time 16. Central Europe Standard Time 18. Central Pacific Standard Time 20. China Standard Time 22. E. Africa Standard Time 24. E. Europe Standard Time 26. Eastern Standard Time 28. Ekaterinburg Standard Time 30. FLE Standard Time 32. Greenland Standard Time 34. GTB Standard Time 36. India Standard Time 38. Israel Standard Time 40. Mexico Standard Time 42. Mid-Atlantic Standard Time 44. Myanmar Standard Time 46. Nepal Standard Time 48. Newfoundland Standard Time 50. North Asia Standard Time 52. Pacific Standard Time 54. Russian Standard Time 56. SA Pacific Standard Time 58. Samoa Standard Time 60. Singapore Standard Time 62. Sri Lanka Standard Time 64. Tasmania Standard Time 66. Tonga Standard Time 68. US Mountain Standard Time 70. W. Australia Standard Time 72. W. Europe Standard Time 74. West Pacific Standard Time |

| | |
|--|---|
| | <p>Usage1 : timezone Zone (1 to 75) AutoDaylight (Y or N) Example1 : timezone 1 Y Usage2 : timezone -custom Bias DaylightBias DaylightStart StandardStart Bias : -12:00 to +13:00 DaylightBias : -12:00 to +13:00 DaylightStart : MM (Month: 01 to 12) ; WD (Day of week: 00 to 06) DD (Day:01 to 05 ;Specifies the occurrence of day in the month; 01 = First occurrence of day, 02 = Second occurrence of day, ..., 05 = Last occurrence of day HH (Hour:00 to 23) StandardStart : MM (Month: 01 to 12) ; WD (Day of week: 00 to 06) DD (Day:01 to 05 ;Specifies the occurrence of day in the month; 01 = First occurrence of day, 02 = Second occurrence of day, ..., 05 = Last occurrence of day HH (Hour:00 to 23) Example2 : timezone -custom +08:00 -01:00 04-00-01-02 10-00- 05-02</p> |
|--|---|

| | |
|-----------------|---|
| [root#]timezone | <p>Show current timezone info Time Zone : (40) Mexico Standard Time (GMT -06:00) Daylight Bias : -01:00 Daylight Start : 05-00-01 02:00 Standard Start : 09-00-05 02:00 Auto Daylight : Y</p> |
|-----------------|---|

| | |
|----------------------|---|
| [root#]timezone 40 n | <p>Use pre-defined timezone Time Zone : (40) Mexico Standard Time (GMT -06:00) Daylight Bias : -01:00 Daylight Start : 05-00-01 02:00 Standard Start : 09-00-05 02:00 Auto Daylight : n</p> |
|----------------------|---|

| | |
|--|---|
| [root#]timezone - custom +08:00 - 01:00 05-00-01-03 09-00-05-03 | <p>Use customized timezone Time Zone : (99) Customized (GMT 08:00) Daylight Bias : -01:00 Daylight Start : 05-00-01 03:00 Standard Start : 09-00-05 03:00 Auto Daylight : Y</p> |
|--|---|

Please refer [Timezone to Country Mapping List](#)

Useradmin: Manager user account

| Command | Purpose |
|------------------------|---|
| [root#] useradmin ? | Usage: useradmin [-add User] [-delete User] [-password User] Example: useradmin -add irene |
| [root#]useradmin | Show the current login user account root |
| [root#]useradmin -list | Show the current user account list rdmin |

| | |
|--|--|
| | root irene |
| [root#] useradmin -add irene Password : irene Confirm : irene Add user Success. | Add the new user account: irene |
| [root#] useradmin -delete 1111 Are You Sure?(Y/N)y | Delete the user: 1111 |
| [root#] useradmin -password root New Password : 1234 Confirm : 1234 | Change the user: root's password. |

Chapter 11 LCD Display Configuration

VIP-2100 provides a front panel LCD for basic operations.

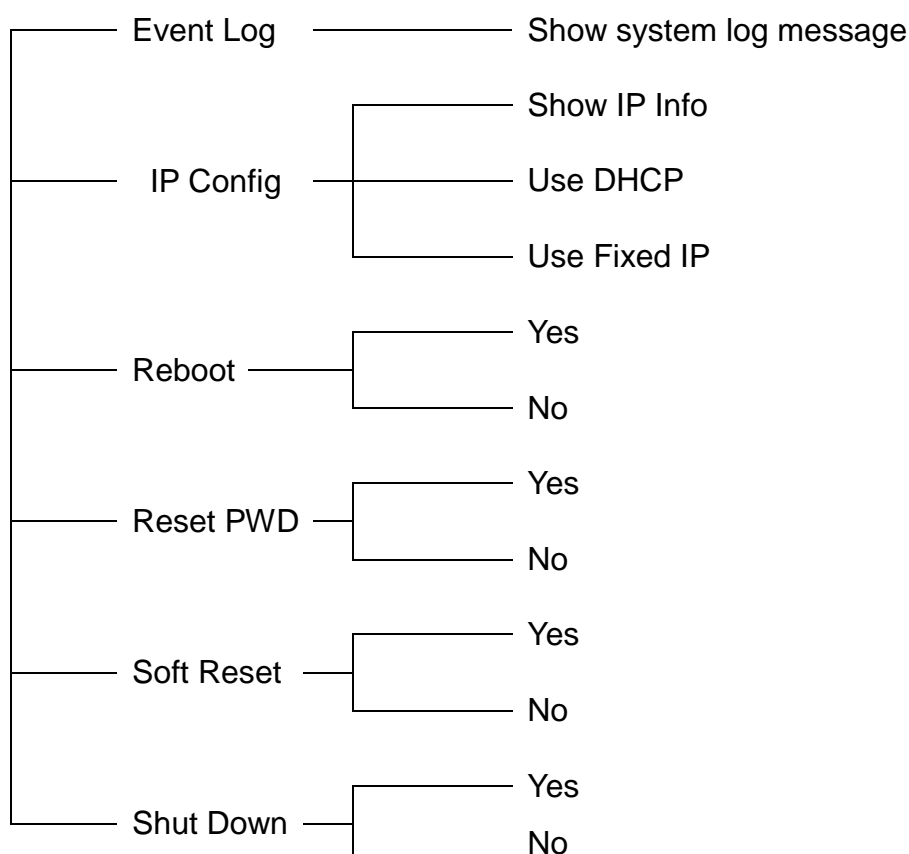


Button List:

| Button List | Description |
|-------------|--|
| | When the VIP-2100 is ready, the LCD screen shows as blow <div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 5px auto;"> Ready 04-03-03 16:40 </div> |
| Enter | Press Enter to select command <div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 5px auto;"> Event Log IP Config </div> |
| ESC | Quit the current command |
| ▲ | Up or previous edit mode |
| ▼ | Next or previous edit mode |

Command Tree:

Main Menu



Event Log:

| Configure | LCD Display |
|-----------|-------------------------------|
| ▲ | Previous event log |
| ▼ | Next event log |
| Enter | Show detail event log |
| ▲ | Previous line |
| ▼ | Next line |
| ESC | Quit detail event log viewing |
| ESC | Quit to main menu |

IP Config:

| Configure | LCD Display |
|-----------|---|
| ▲ | Select Network configuration |
| ▼ | Select Network configuration |
| Enter | Configure Network |
| ▲ | Increase the digit apply to network setting |
| ▼ | Decrease the digit apply to network setting |
| Enter | Apply change to network information |
| ESC | Quit network setting |
| ESC | Quit to main menu |

Reboot:

| Configure | LCD Display |
|-----------|---|
| ▲ | Select Reboot or not |
| ▼ | Select Reboot or not |
| Enter | Reset user: root's (or admin) user password |
| ESC | Quit Reboot configure |
| ESC | Quit to main menu |

Reset:

| Configure | LCD Display |
|-----------|--|
| ▲ | Select user to change password |
| ▼ | Select user to change password |
| Enter | Change user password |
| ▲ | Increase the alphabet apply to user password setting |
| ▼ | Decrease the alphabet apply to user password setting |
| ESC | Quit Reset configure |
| ESC | Quit to main menu |

Soft Reset:

| Configure | LCD Display |
|-----------|---------------------|
| ▲ | Select Reset or not |
| ▼ | Select Reset or not |

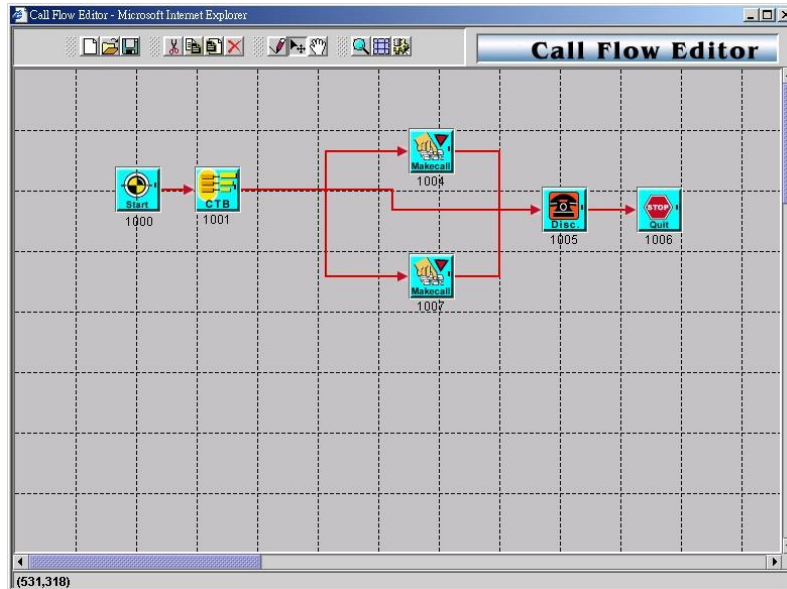
| | |
|-------|----------------------|
| Enter | Reset or not |
| ESC | Quit Reset configure |
| ESC | Quit to main menu |

Shutdown:



| Configure | LCD Display |
|------------------|-------------------------|
| ▲ | Select Shutdown or not |
| ▼ | Select Shutdown or not |
| Enter | Shutdown or not |
| ESC | Quit Shutdown configure |
| ESC | Quit to main menu |

Appendix 1 Call Flow Example




One Stage Dialing (Gatekeeper Mode)



Example Description:




| Components | Contents |
|---|---|
|  Start Component ID: 1000 | Next Component: 1001 |
|  CTB Component ID: 1001 | PSTN To: 1004 H.323 To: 1007 SIP To: 1005 |

1007 Route for H.323 Gatekeeper call

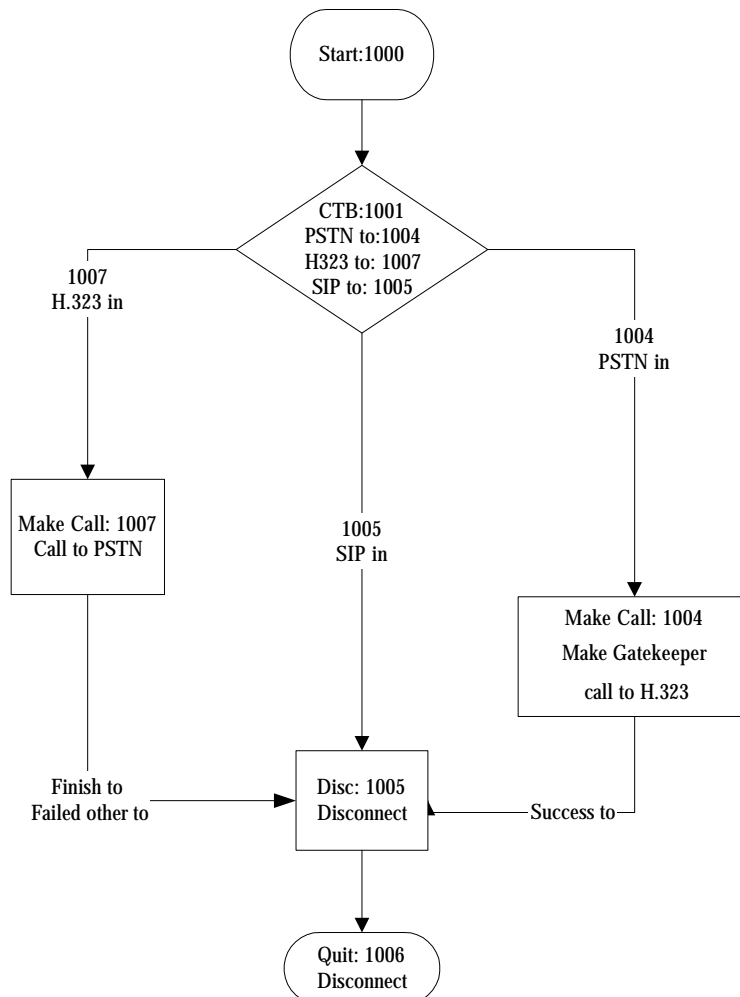
| | |
|--|---|
|  MakeCall Component ID: 1007 | Route Mode: PSTN Finish To: 1005 Failed Other To: 1005 |
|  Disc Component ID: 1005 | Reason: PSTN normal call clear Next Component: 1006 |
|  Quit | |

Component: 1006

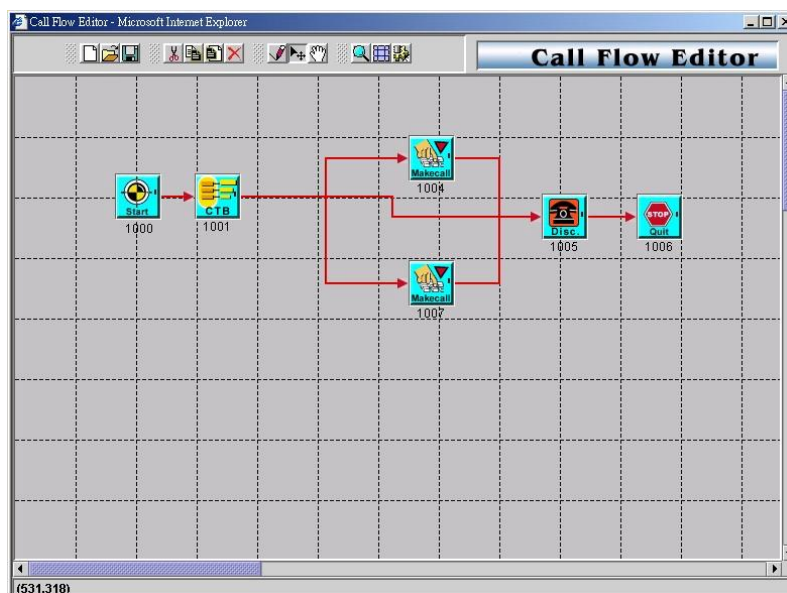
1004 Route for PSTN call

| | |
|--|---|
|  MakeCall Component ID: 1004 | Route Mode: Gatekeeper Finish To: 1005 Failed Other To: 1005 |
|  Disc Component ID: 1005 | Next Component: 1006 |
|  Quit Component: 1006 | |



Example Used Call Flow:






One Stage Dialing (SIP Proxy Mode)






Example Description:

| Components | Contents |
|---|---|
|  Start Component ID: 1000 | Next Component: 1001 |
|  CTB Component ID: 1001 | PSTN To: 1004 H.323 To: 1005 SIP To: 1007 |

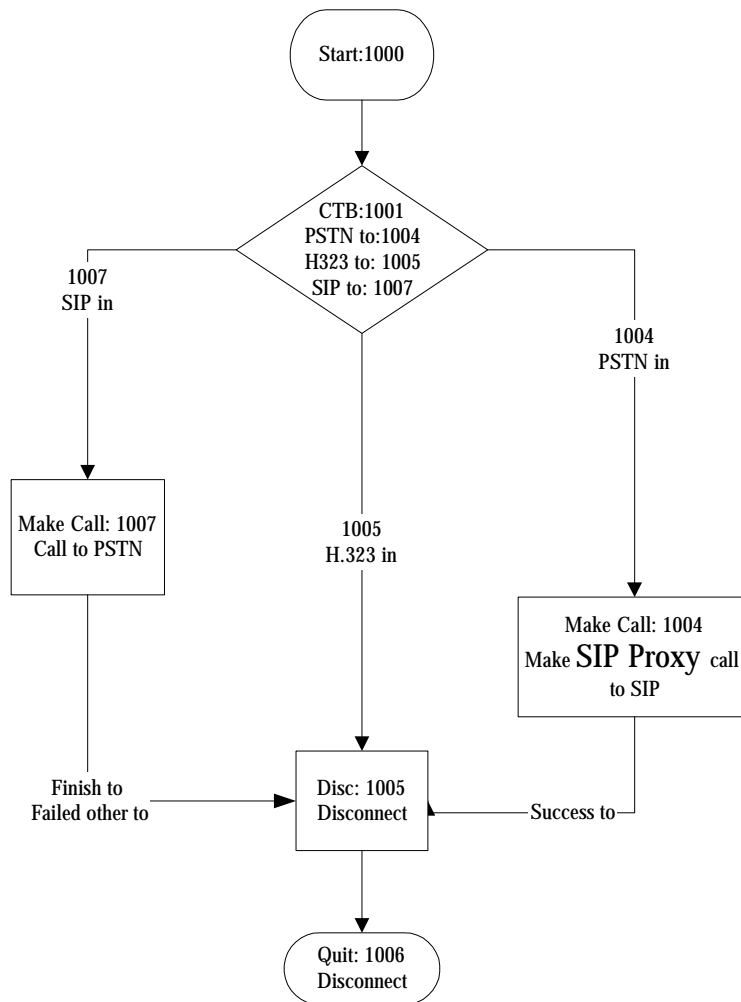
1007 Route for SIP Proxy call

| | |
|--|---|
|  MakeCall Component ID: 1007 | Route Mode: PSTN Finish To: 1005 Failed Other To: 1005 |
|  Disc Component ID: 1005 | Reason: PSTN normal call clear Next Component: 1006 |
|  Quit Component ID: 1006 | |

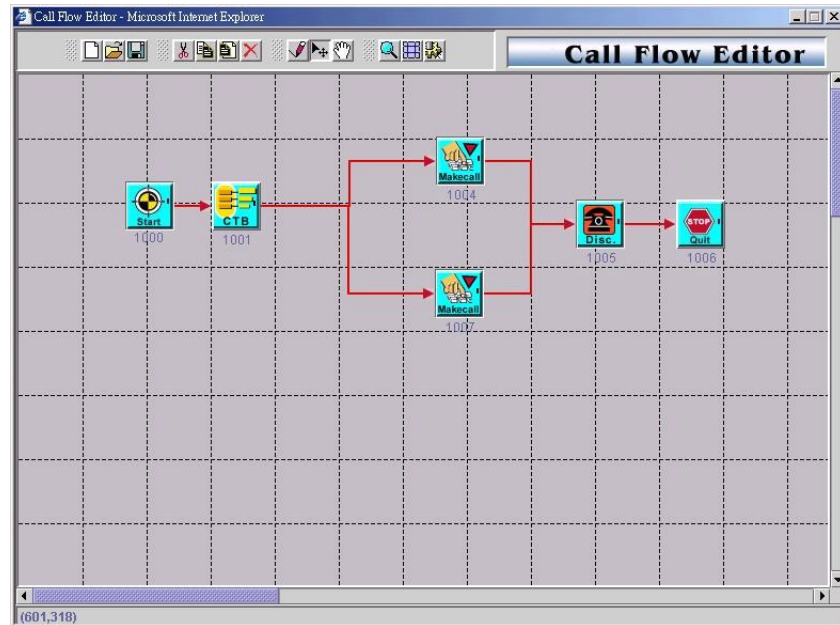
1004 Route for PSTN call

| | |
|--|---|
|  MakeCall Component ID: 1004 | Route Mode: SIP Proxy Call Finish To: 1005 Failed Other To: 1005 |
|  Disconnect Component ID: 1005 | Next Component: 1006 |
|  Quit Component ID: 1006 | |



Example Used Call Flow:






One Stage Dialing (Peer to Peer Mode)






Example Description:

| Components | Contents |
|---|---|
|  Start Component ID: 1000 | Next Component: 1001 |
|  CTB Component ID: 1001 | PSTN To: 1004 H.323 To: 1007 SIP To: 1007 |

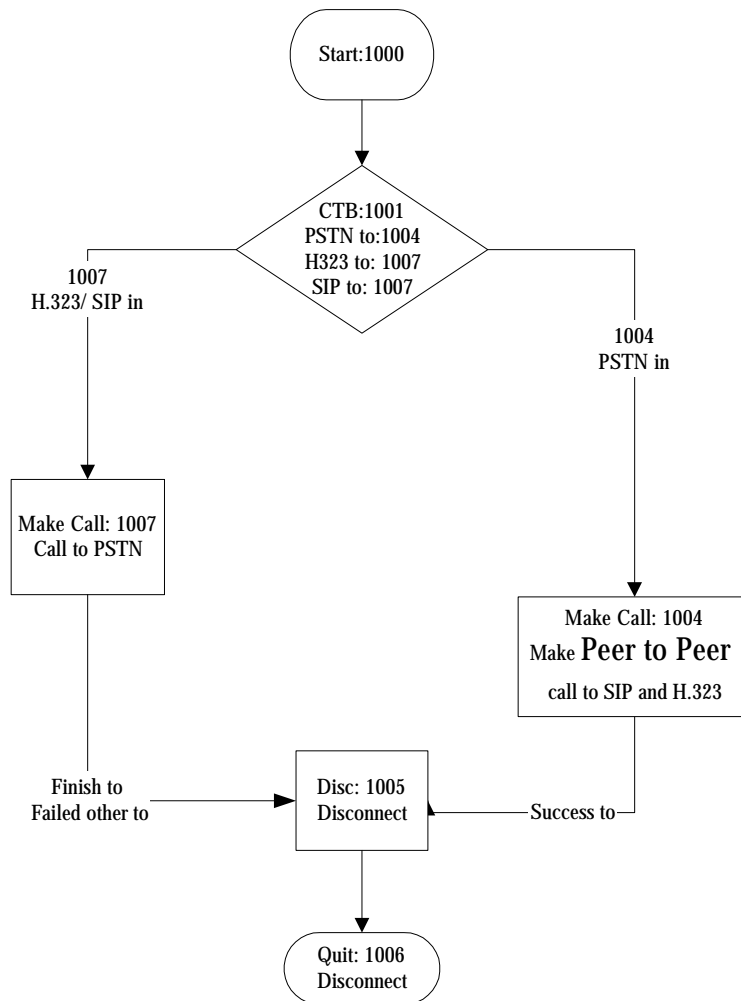
1007 Route for SIP Proxy or H.323 Gatekeeper call

| | |
|--|---|
|  MakeCall Component ID: 1007 | Route Mode: PSTN Finish To: 1005 Failed Other To: 1005 |
|  Disc Component ID: 1005 | Next Component: 1006 |
|  Quit Component ID: 1006 | |

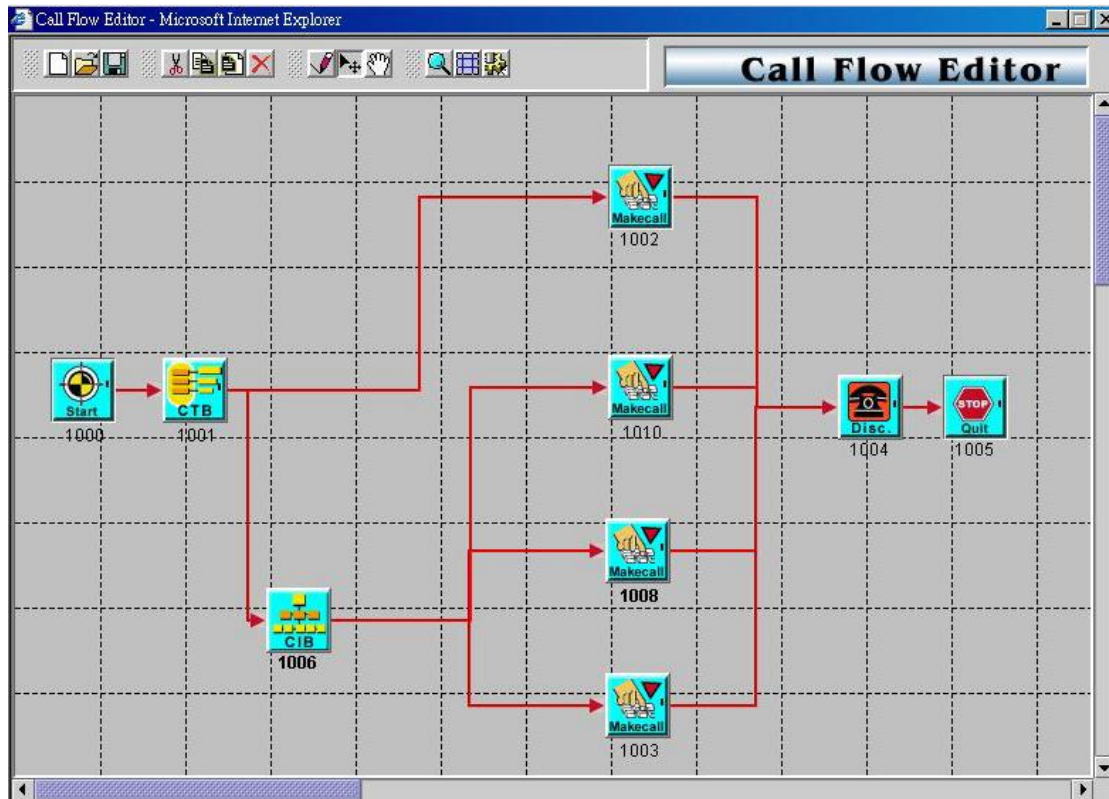
1004 Route for PSTN call

| | |
|--|---|
|  MakeCall Component ID: 1004 | Route Mode: P2P Call Finish To: 1005 Failed Other To: 1005 |
|  Disc Component ID: 1005 | Next Component: 1006 |
|  Quit Component ID: 1006 | |





Example Used Call Flow:







Two Stage Dialing (VoIP, PSTN mixed call)






Example Description:

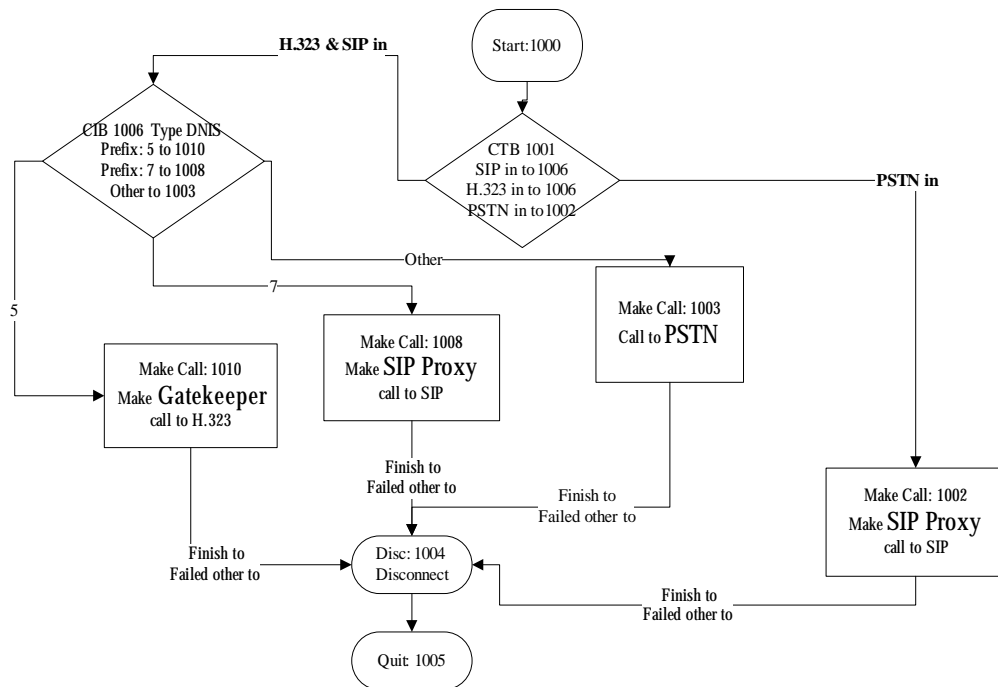
| Components | Contents |
|--|---|
| Call route from PSTN to IP Side | |
|  Start Component ID: 1000 | Next Component: 1001 |
|  CTB Component ID: 1001 | PSTN To: 1002 H.323 To: 1006 SIP To: 1006 |
| 1001 route for SIP Proxy and H.323 Gatekeeper call | |
|  CIB Component ID: 1006 | Info Type: DNIS Prefix: 5 goto: 1010 Prefix: 7 goto: 1008 Other goto: 1003 |
|  MakeCall Component ID: 1010 | Route Mode: Gatekeeper Call Finish To: 1004 Failed Other To: 1004 |

| | |
|--|---|
|  MakeCall Component ID: 1008 | Route Mode: SIP Proxy Call Finish To: 1004 Failed Other To: 1004 |
|  MakeCall Component ID: 1003 | Route Mode: PSTN Finish To: 1004 Failed Other To: 1004 |
|  Disc Component ID: 1004 | Next Component: 1005 |
|  Quit Component ID: 1005 | |

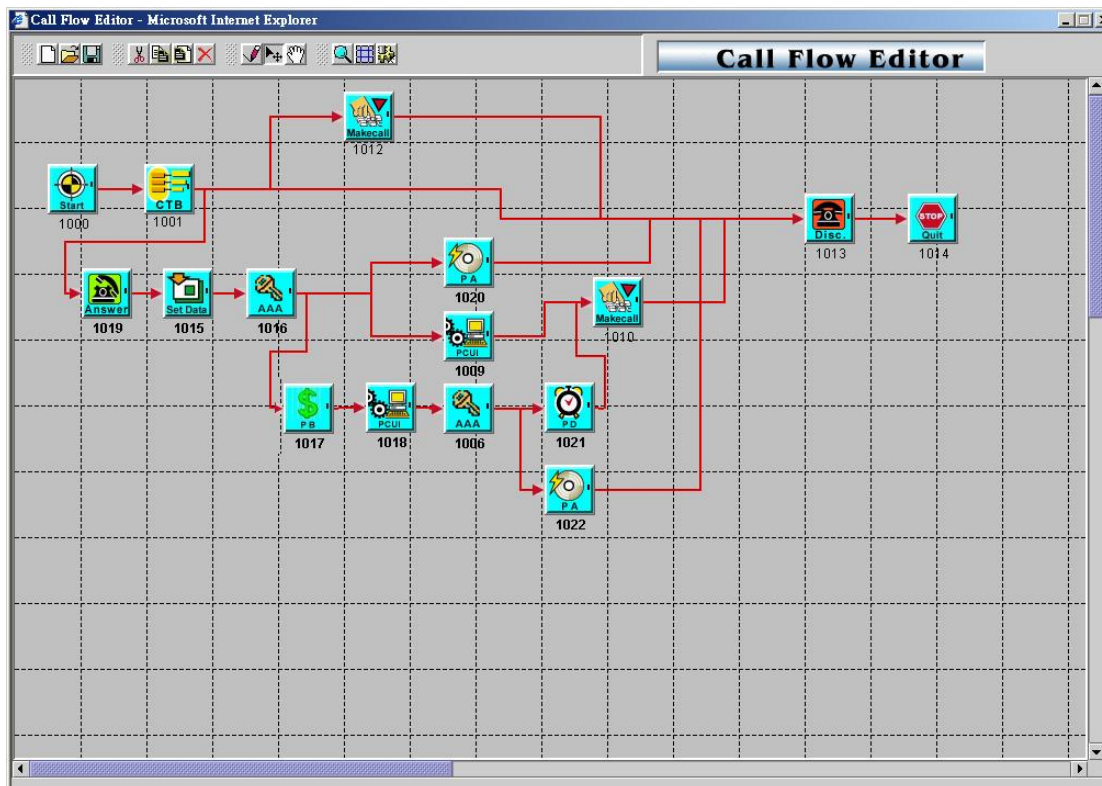
1001 Route for PSTN call

| | |
|--|---|
|  MakeCall Component ID: 1002 | Route Mode: SIP Proxy Call Finish To: 1004 Failed Other To: 1004 |
|  Disc Component ID: 1004 | Next Component: 1005 |
|  Quit Component ID: 1005 | |



Example Used Call Flow:





Two Stage Dialing with AAA (IP Side AAA)



Example Description:


| Components | Contents |
|---|---|
|  Start Component ID: 1000 | Next Component: 1001 |
|  CTB Component ID: 1001 | PSTN To: 1012 H.323 To: 1013 SIP To: 1019 |

1012 Route for PSTN call

| | |
|--|--|
|  MakeCall Component ID: 1012 | Route Mode: SIP Proxy Finish To: 1013 Failed Other To: 1013 |
|  Disc Component ID: 1013 | Next Component: 1014 |


| | |
|--|--|
|  Quit Component ID: 1014 | |
|--|--|


1012 Route for H.323 call


| | |
|--|----------------------|
|  Disc Component ID: 1013 | Next Component: 1014 |
|--|----------------------|

| | |
|--|--|
|  Quit Component ID: 1014 | |
|--|--|


1012 Route for SIP call


| | |
|--|----------------------|
|  Answer Component ID: 1019 | Next Component: 1015 |
|--|----------------------|










| | |
|--|--|
|  Set Data Component ID: 1015 | Assign to: User ID / Password Use SysParam: Yes Value: ANI Next Component: 1016 |
|--|--|

| | |
|---|--|
|  AAA Component ID: 1016 | Type: Authentication Prepaid User to: 1017 Postpaid User to: 1009 Failed to: 1020 Failed Reason: <ul style="list-style-type: none"> - Invalid Account - Account InUse - Zero Balance - Account Expired - Over Credit Limit - Number of Retries Exceeded - Insufficient Balance |
|---|--|

- Route for prepaid user call

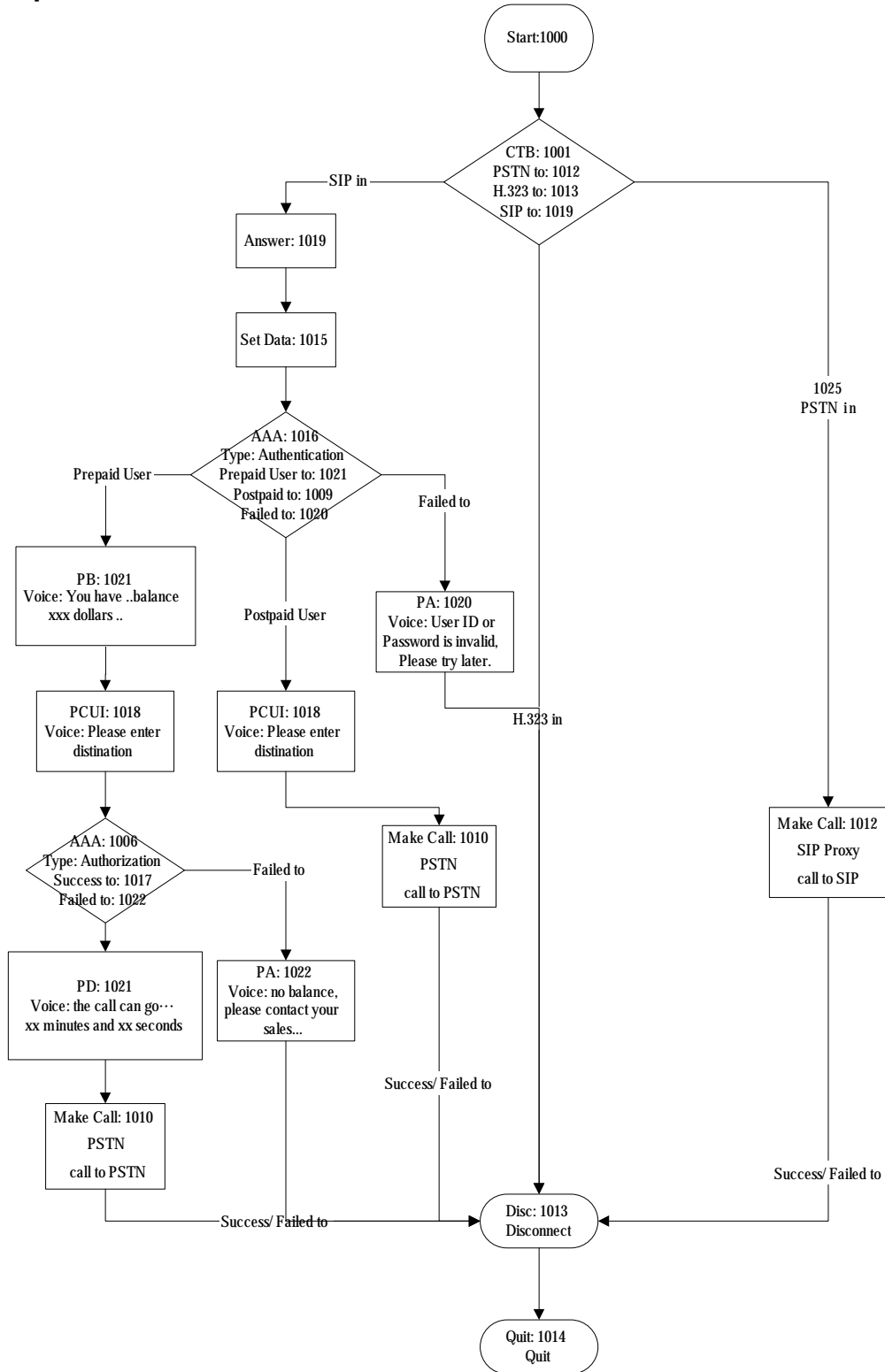
| | |
|--|--|
|  PB Component ID: 1017 | Voice File: 0004.raw Language: English Interrupted: No Next Component: 1018 |
|--|--|

| | |
|--|---|
|  PCUI Component ID: 1018 | Play Type: Voice or dial tone Voice File: 0001.raw Max DTMF: 30 Result Append To: DNIS End of DTMF: # |
|--|---|

| | |
|--|--|
| | Next Component: 1006 |
|  AAA Component ID: 1006 | Type: Authorization Success to: 1021 Failed to: 1022 Failed Reason: - Invalid Account - Account InUse - Zero Balance - Account Expired - Over Credit Limit - Number of Retries Exceeded - Insufficient Balance |
|  PD Component: 1021 | Voice File: 0004.raw Language: English Interrupted: No Next Component: 1010 |
|  MakeCall Component ID: 1010 | Route Mode: PSTN Call Finish To: 1013 Failed Other To: 1013 |
| - Route for failed user call | |
|  PA Component: 1022 | Dynamic Play: Disable Voice File: 0005.raw Language: English Interrupted: No Next Component: 1013 |
|  Disc Component ID: 1013 | Next Component: 1014 |
|  Quit Component ID: 1014 | |
| - Route for postpaid user call | |
|  MakeCall Component ID: 1012 | Route Mode: SIP Proxy Call Finish To: 1013 Failed Other To: 1013 |
|  Disc Component ID: 1013 | Next Component: 1014 |
|  Quit | |

| | |
|-----------------------|--|
| Component ID: 1014 | |
| | |

Example Used Call Flow:

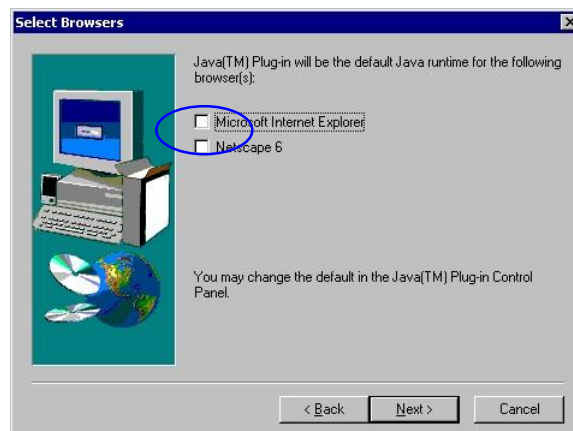


Appendix 2 Java plug-in Installation

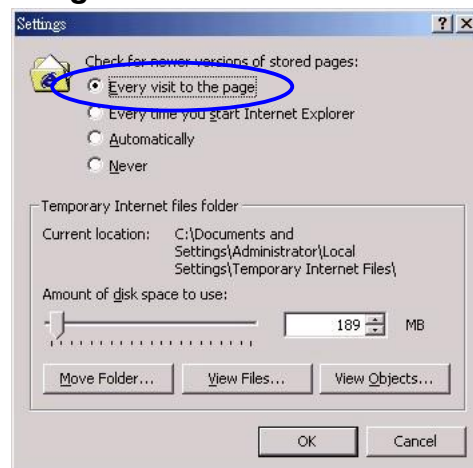
You need to install Java Plug-in before using call flow editor, prompt manager and upgrade. Please confirm you JRE version is 1.4.1_02 or above if your PC has already installed Java.



After downloaded the java runtime version (1.3.1 or later) from Sun, you just follow the wizard to install the Java runtime. When you see the display shows “Select Browsers”, **do not select any option item**, press **Next** button to continue.



You also need to set newer versions of stored pages. **Click Tool→Internet Option→General→Setting.**



After success, restart your browser to take effect.

Appendix 3 Retrieve CDR Information

J Retrieve method example (stop20040305.log) by ftp:

```
C:\>ftp 192.168.19.117
Connected to 192.168.19.117.
220 Server ready
User (192.168.19.117:(none)): root
331 Password required for root.
Password:
230 User root logged in.
ftp> cd welltech\cdr
250 CWD command successful. "D:/welltech/cdr/" is current directory.
ftp> dir
200 Port command successful.
150 Opening data connection for directory list.
drw-rw-rw-  1 ftp  ftp      0 Mar 06 00:02 .
drw-rw-rw-  1 ftp  ftp      0 Mar 06 00:02 ..
-rw-rw-rw-  1 ftp  ftp  53998192 Mar 05 23:57 STOP20040305.log
-rw-rw-rw-  1 ftp  ftp  20222855 Mar 05 23:50 STRT20040305.log
226 File sent ok
ftp: 403 bytes received in 0.25Seconds 1.61Kbytes/sec
ftp> bin
200 Type set to I.
ftp> lcd
Local directory now C:\.
ftp> get stop20040305.log
200 Port command successful.
150 Opening data connection for stop20040305.log.
226 File sent ok
ftp: 20222855 bytes received in 4.43Seconds 4569.10Kbytes/sec.
ftp>bye
221 Goodbye
```

Billing Start CDR:

- **File name:** STRTyymmdd.log
- **Field delimit:** ,
- **Field description:**
 - NAS-IP-Address : VoIP gateway IP address
 - NAS-Port-Type : (Network Access Server Port Type)

Asynchronous

| | |
|--------------------|---------------------------|
| User-Name | : User ID |
| Calling-Station-Id | : Calling station number |
| Acct-Status-Type | : Message type (1: start) |
| Service-Type | : 1: login |
| Gateway-Name | : VoIP gateway aliases |
| Conf-ID | : GUID |
| Call-Type | : Telephony or VOIP |

Call-Originate : originate or answer
Setup-Time : Call initiate time (UTC time)
Acct-Session-Id : N/A
Acct-Delay-Time : N/A

Billing Stop CDR:

- **File name:** STOPyyyymmdd.log
- **Field delimit:** ,
- **Field description:**
 - NAS-IP-Address : VoIP gateway IP address
 - NAS-Port-Type : (Network Access Server Port Type)

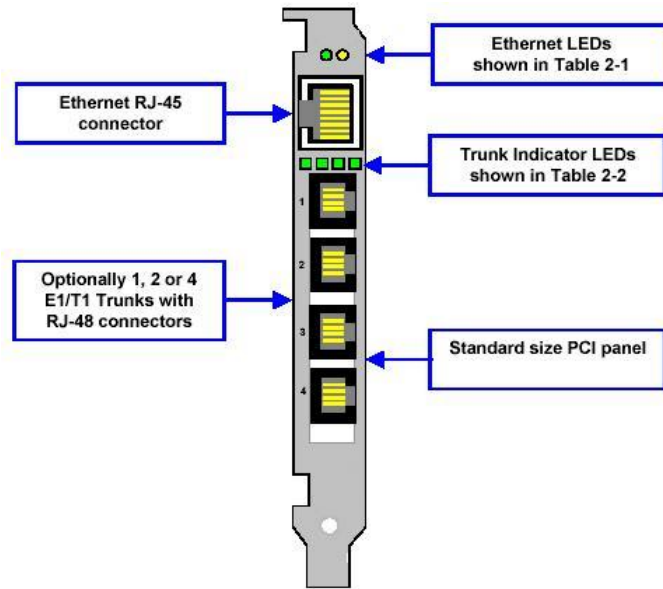
Asynchronous

User-Name : User ID
Called-Station-Id : Called station number
Calling-Station-Id : Calling station number
Acct-Status-Type : Message type (1: Start , 2: Stop)
Service-Type : 1: login
Gateway-Name : VoIP gateway aliases
Conf-ID : GUID
Call-Type : Telephony or VOIP
Call-Originate : originate or answer
Setup-Time : Setup Time (UTC time)
Connect-Time : Connect Time (UTC time)
Disconnect-Time : Disconnect Time (UTC time)
Disconnect-Cause : Disconnect cause code
Voice-Quality : Voice Quality
Gateway-ID : Remote gateway IP address
Acct-Session-Id : N/A
Acct-Input-Octets : N/A
Acct-Output-Octets : N/A
Acct-Input-Packets : N/A
Acct-Output-Packets : N/A
Acct-Session-Time : Talk time
Acct-Delay-Time : N/A
Charge rate : Internal AAA prepaid user charge rate
Available Balance : Internal AAA prepaid user available

balance

Appendix 4 Interface LED Description

Interface Real Panel:



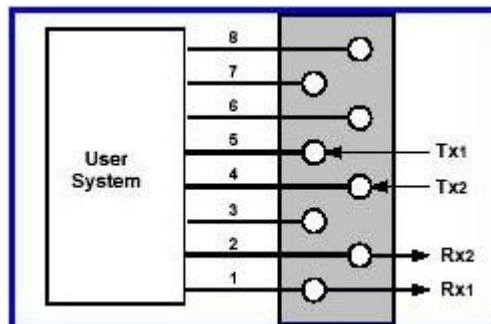
Ethernet LED:

| LED Color | LED Function |
|-----------|----------------------------------|
| Yellow | Receive |
| Green | Ethernet connection is ON (Link) |

Trunk LED:

| LED Color | LED Function |
|-----------|--|
| Green | Normal Operation Trunk is synchronized (No Alarms) |
| Red | LOS - Indicates Loss of Signal |
| Red | LFA - Indicates Loss of Frame Alignment |
| Red | AIS - Alarm Indication Signal (The Blue Alarm) |
| Red | RAI - Remote Alarm Indication (The Yellow Alarm) |

Trunk RJ48 Wiring:



Appendix 5 Build-in Voice Prompt Index

| File Name | Description |
|------------------|---|
| 0001.raw | Please enter the destination |
| 0002.raw | Please enter your user ID |
| 0003.raw | Please enter your password |
| 0004.raw | You have |
| 0005.raw | User ID or password is invalid. Please try later. |

Appendix 6 Timezone to Country Mapping List

| Greenwich Mean Time & Country List | Time Zone |
|--|-------------------------------------|
| (GMT-12:00) International Date Line West | 21. Dateline Standard Time |
| (GMT-11:00) Midway Island, Samoa | 58. Samoa Standard Time |
| (GMT-10:00) Hawaii | 35. Hawaiian Standard Time |
| (GMT-09:00) Alaska | 02. Alaskan Standard Time |
| (GMT-08:00) Pacific Time (US & Canada); Tijuana | 52. Pacific Standard Time |
| (GMT-07:00) Mountain Time (US & Canada) | 43. Mountain Standard Time |
| (GMT-07:00) Chihuahua, La Paz, Mazatlan | 41. Mexico Standard Time 2 |
| (GMT-07:00) Arizona | 68. US Mountain Standard Time |
| (GMT-06:00) Saskatchewan | 10. Canada Central Standard Time |
| (GMT-06:00) Guadalajara, Mexico City, Monterrey | 40. Mexico Standard Time |
| (GMT-06:00) Central Time (US & Canada) | 19. Central Standard Time |
| (GMT-06:00) Central America | 14. Central America Standard Time |
| (GMT-05:00) Indiana (East) | 67. US Eastern Standard Time |
| (GMT-05:00) Eastern Time (US & Canada) | 26. Eastern Standard Time |
| (GMT-05:00) Bogota, Lima, Quito | 56. SA Pacific Standard Time |
| (GMT-04:00) Santiago | 51. Pacific SA Standard Time |
| (GMT-04:00) Caracas, La Paz | 57. SA Western Standard Time |
| (GMT-04:00) Atlantic Time (Canada) | 06. Atlantic Standard Time |
| (GMT-03:30) Newfoundland | 48. Newfoundland Standard Time |
| (GMT-03:00) Greenland | 32. Greenland Standard Time |
| (GMT-03:00) Buenos Aires, Georgetown | 55. SA Eastern Standard Time |
| (GMT-03:00) Brasilia | 25. E. South America Standard Time |
| (GMT-02:00) Mid-Atlantic | 42. Mid-Atlantic Standard Time |
| (GMT-01:00) Cape Verde Is. | 11. Cape Verde Standard Time |
| (GMT-01:00) Azores | 09. Azores Standard Time |
| (GMT) Greenwich Mean Time: Dublin, Edinburgh, Lisbon, London | 31. GMT Standard Time |
| (GMT) Casablanca, Monrovia | 33. Greenwich Standard Time |
| (GMT+01:00) West Central Africa | 71. W. Central Africa Standard Time |
| (GMT+01:00) Sarajevo, Skopje, Warsaw, Zagreb | 17. Central European Standard Time |
| (GMT+01:00) Brussels, Copenhagen, Madrid, Paris | 53. Romance Standard Time |
| (GMT+01:00) Belgrade, Bratislava, Budapest, Ljubljana, Prague | 16. Central Europe Standard Time |
| (GMT+01:00) Amsterdam, Berlin, Bern, Rome, Stockholm, Vienna | 72. W. Europe Standard Time |
| (GMT+02:00) Jerusalem | 38. Israel Standard Time |
| (GMT+02:00) Helsinki, Kyiv, Riga, Sofia, Tallinn, Vilnius | 30. FLE Standard Time |
| (GMT+02:00) Harare, Pretoria | 61. South Africa Standard Time |
| (GMT+02:00) Cairo | 27. Egypt Standard Time |
| (GMT+02:00) Bucharest | 24. E. Europe Standard Time |

| | |
|---|-----------------------------------|
| (GMT+02:00) Athens, Istanbul, Minsk | 34. GTB Standard Time |
| (GMT+03:00) Nairobi | 22. E. Africa Standard Time |
| (GMT+03:00) Moscow, St. Petersburg, Volgograd | 54. Russian Standard Time |
| (GMT+03:00) Kuwait, Riyadh | 03. Arab Standard Time |
| (GMT+03:00) Baghdad | 05. Arabic Standard Time |
| (GMT+03:30) Tehran | 37. Iran Standard Time |
| (GMT+04:00) Baku, Tbilisi, Yerevan | 12. Caucasus Standard Time |
| (GMT+04:00) Abu Dhabi, Muscat | 04. Arabian Standard Time |
| (GMT+04:30) Kabul | 01. Afghanistan Standard Time |
| (GMT+05:00) Islamabad, Karachi, Tashkent | 73. West Asia Standard Time |
| (GMT+05:00) Ekaterinburg | 28. Ekaterinburg Standard Time |
| (GMT+05:30) Chennai, Kolkata, Mumbai, New Delhi | 36. India Standard Time |
| (GMT+05:45) Kathmandu | 46. Nepal Standard Time |
| (GMT+06:00) Sri Jayawardenepura | 62. Sri Lanka Standard Time |
| (GMT+06:00) Astana, Dhaka | 15. Central Asia Standard Time |
| (GMT+06:00) Almaty, Novosibirsk | 45. N. Central Asia Standard Time |
| (GMT+06:30) Rangoon | 44. Myanmar Standard Time |
| (GMT+07:00) Krasnoyarsk | 50. North Asia Standard Time |
| (GMT+07:00) Bangkok, Hanoi, Jakarta | 59. SE Asia Standard Time |
| (GMT+08:00) Taipei | 63. Taipei Standard Time |
| (GMT+08:00) Perth | 70. W. Australia Standard Time |
| (GMT+08:00) Kuala Lumpur, Singapore | 60. Singapore Standard Time |
| (GMT+08:00) Irkutsk, Ulaan Bataar | 49. North Asia East Standard Time |
| (GMT+08:00) Beijing, Chongqing, Hong Kong, Urumqi | 20. China Standard Time |
| (GMT+09:00) Yakutsk | 75. Yakutsk Standard Time |
| (GMT+09:00) Seoul | 39. Korea Standard Time |
| (GMT+09:00) Osaka, Sapporo, Tokyo | 65. Tokyo Standard Time |
| (GMT+09:30) Darwin | 07. AUS Central Standard Time |
| (GMT+09:30) Adelaide | 13. Cen. Australia Standard Time |
| (GMT+10:00) Vladivostok | 69. Vladivostok Standard Time |
| (GMT+10:00) Hobart | 64. Tasmania Standard Time |
| (GMT+10:00) Guam, Port Moresby | 74. West Pacific Standard Time |
| (GMT+10:00) Canberra, Melbourne, Sydney | 08. AUS Eastern Standard Time |
| (GMT+10:00) Brisbane | 23. E. Australia Standard Time |
| (GMT+11:00) Magadan, Solomon Is., New Caledonia | 18. Central Pacific Standard Time |
| (GMT+12:00) Fiji, Kamchatka, Marshall Is. | 29. Fiji Standard Time |
| (GMT+12:00) Auckland, Wellington | 47. New Zealand Standard Time |
| (GMT+13:00) Nuku'alofa | 66. Tonga Standard Time |

Appendix 7 IP Bandwidth Requirement

| Compression | Packet duration | 1 voice paths Bandwidth (kbps) | 30 voice paths Bandwidth (kbps) | 60 voice paths Bandwidth (kbps) | 120 voice paths Bandwidth (kbps) |
|-------------------------|------------------------|---------------------------------------|--|--|---|
| 7.231.1 (5.3kbps) | 30 ms | 32 | 960 | 1920 | 3840 |
| | 60 ms | 21.2 | 640 | 1280 | 2560 |
| | 90 ms | 17.8 | 534 | 1068 | 2134 |
| 7.231.1 (6.4kbps) | 30 ms | 34 | 1024 | 2048 | 4096 |
| | 60 ms | 23.4 | 704 | 1408 | 2816 |
| | 90 ms | 19.8 | 598 | 1196 | 2390 |
| G.729A (8kbps) | 20 ms | 48 | 1440 | 2880 | 5760 |
| | 40 ms | 32 | 960 | 1920 | 3840 |
| | 60 ms | 26.6 | 800 | 1600 | 3200 |
| G.711 (PCM) (64kbps) | 20 ms | 160 | 4800 | 9600 | 19200 |
| | 40 ms | 144 | 4320 | 8640 | 17280 |
| | 60 ms | 138.6 | 4160 | 8320 | 16640 |

Appendix 8 Release Complete Cause Code

H.225 Release Complete Reason to cause IE mapping

| RelaseCompleteReason code | Corresponding Q.931/Q.850 cause vale |
|----------------------------------|---|
| noBandwidth | 34 - No circuit/channel available |
| gatekeeperResources | 47 – Resource Unavailable |
| unreachableDestination | 3 – No route to destination |
| destinationRejection | 16 – Normal call clearing |
| invalidRevision | 88 – Incompatible destination |
| noPermission | 111 – Interworking, unspecified |
| unreachableGatekeeper | 38 – Network out of order |
| Gateway Resources | 42 – Switching equipment comgestion |
| badFormatAddress | 28 – Invalid number format |
| adaptiveBusy | 41 – Temporary Failure |
| inConf | 17 – User busy |
| undefineReason | 31 – Normal, unspecified |
| FacilityCallDeflection | 16 – Normal call clearing |
| securityDenied | 31 – Normal, unspecified |
| calledPartyNotRegistered | 20 – Subscriber absent |
| callerNotRegistered | 31 – Normal, unspecified |

PSTN to SIP Cause Code Mapping

| PSTN Cause Code | Description | SIP Event |
|---|---|-----------------------------|
| 1 | Unallocated number | 404 Not found |
| 2 | No route to specified transit network | 404 Not found |
| 3 | No route to destination | 404 Not found |
| 17 | User busy | 486 User here |
| 18 | No user response | 480 Temporarily unavailable |
| 19 | No answer from the user | |
| 20 | Subscriber absent | |
| 21 | Call Rejected | 403 Forbidden |
| 22 | Number changed | 410 Gone |
| 26 | Non-selected user clearing | 404 Not found |
| 27 | Destination out of order | 404 Not found |
| 28 | Address incomplete | 484 Address incomplete |
| 29 | Facility rejected | 501 Not implemented |
| 31 | Normal, unspecified | 404 Not found |
| 34 | No, circuit available | 503 Service unavailable |
| 38 | Network out of order | 503 Service unavailable |
| 41 | Temporary failure | 503 Service unavailable |
| 42 | Switching equipment congestion | 503 Service unavailable |
| 47 | Resource unavailable | 503 Service unavailable |
| 55 | Incoming calls barred within Closed User Group(CUG) | 403 Forbidden |
| 58 | Bearer capability not presently available | 403 Forbidden |
| 65 | Bearer capability not implemented | 501 Not implemented |
| 79 | Service or option not implemented | 501 Not implemented |
| 87 | User not member of Closed User Group (CUG) | 503 Service Unavailable |
| 88 | Incompatible destination | 400 Bad request |
| 95 | Invalid message | 400 Bad request |
| 102 | Recovery on Expires timeout | 408 Request timeout |
| 111 | Protocol error | 400 Bad request |
| Any code other than those listed above: | | 500 Internal server error |

SIP to PSTN Cause Code Mapping

| IP Event | Description | PSTN Cause Code |
|-----------------------------|---|-----------------|
| 404 Not found | No route to destination | 3 |
| 486 User here | User busy | 17 |
| 480 Temporarily unavailable | No user response | 18 |
| | | 18 |
| | | 20 |
| 403 Forbidden | Call Rejected | 21 |
| 410 Gone | Number changed | 22 |
| 404 Not found | Unallocated number | 3 |
| 404 Not found | Unallocated number | 3 |
| 484 Address incomplete | Address incomplete | 28 |
| 501 Not implemented | Service or option not implemented | 79 |
| 404 Not found | Unallocated number | 3 |
| 503 Service unavailable | Service or option unavailable | 63 |
| 503 Service unavailable | Service or option unavailable | 63 |
| 503 Service unavailable | Service or option unavailable | 63 |
| 503 Service unavailable | Service or option unavailable | 63 |
| 503 Service unavailable | Service or option unavailable | 63 |
| 403 Forbidden | Bearer Capability not authorized | 21 |
| 403 Forbidden | Service or option not implemented | 79 |
| 501 Not implemented | Service or option not implemented | 79 |
| 501 Not implemented | Service or option not implemented | 79 |
| 503 Service Unavailable | Service or option unavailable | 63 |
| 400 Bad request | Interworking, unspecified | 95 |
| 400 Bad request | Interworking, unspecified | 95 |
| 408 Request timeout | Recovery on Expires timeout | 102 |
| 400 Bad request | Protocol error | 111 |
| 500 Internal server error | Any code other than those listed above: | 127 |

Appendix 9 RADIUS Format Attributes

RADIUS Format V2.0

Start Accounting Request Attributes

| ttri-bute | Name A | Description | Format | Sample |
|------------------|-----------------------|--|---------------|---|
| 4 | NAS-IP-Address | IP Address of the In-Bound gateway | Numeric | 4 bytes unsigned long |
| 61 | NAS-Port-Type | Physical port type | Numeric | 0: Asynchronous |
| 1 | User-Name | Account number(with 4 digit pin number on postfix) | String | 5500033440 |
| 31 | Calling-Station-Id | Calling Party Number (ANI) | String | 886282265699 |
| 30 | Called-Station-Id | Destination phone number | String | 86258765432 |
| 40 | Acct-Status-Type | Accounting Request Type | Numeric | 1: Start Accounting 2: Stop Accounting |
| 6 | Service-Type | Type of service requested | Numeric | 5: Outbound |
| 26 | h323-gw-id -33 | Name of the Gateway (IP address) | String | h323-gw-id =VIP2100 |
| 26 | h323-conf-id -24 | GUID | String | h323-conf-id=xxxx |
| 26 | h323-call-type -27 | Protocol type or family used on this leg of the call (Telephony or VOIP) | String | h323-call-type=VOIP |
| 26 | h323-call-origin - 26 | 'Originate' or 'Answer' | String | h323-call-origin =Originate |
| 26 | h323-setup-time -25 | Setup time in NTP format | String | h323-setup-time= 23:24:19.810 UTC Sun Sep 26 2001 |
| 44 | Acct-Session-Id | A unique accounting identifier - match start & stop | String | 8 bytes, like 00012345 |
| 41 | Acct-Delay-Time | No of seconds tried in sending a particular record | Numeric | 5 |

Stop Accounting Request Attributes

| Attribute | NAME | Description | Format | Sample |
|-----------|---------------------------|--|---------|--|
| 4 | NAS-IP-Address | IP Address of the In-Bound gateway | Numeric | 4 bytes unsigned long |
| 61 | NAS-Port-Type | Physical port type | Numeric | 0: Asynchronous |
| 1 | User-Name | Account number (with 4 digit pin number on postfix) | String | 5500033440 |
| 30 | Called-Station-Id | Destination phone number | String | 862587654321 |
| 31 | Calling-Station-Id | Calling Party Number (ANI) | String | 886282265699 |
| 40 | Acct-Status-Type | Account Request Type | Numeric | 1: Start Accounting 2: Stop Accounting |
| 6 | Service-Type | Type of service requested | Numeric | 5: Outbound |
| 26 | h323-gw-id -33 | Gateway IP address | String | h323-gw-id =VIP2100 |
| 26 | h323-conf-id -24 | GUID | String | h323-conf-id =xxxx |
| 26 | h323-call-type -27 | Protocol type used on this leg of the call - Telephony or VOIP | String | h323-call-type=VOIP |
| 26 | h323-setup-time -25 | Setup time in NTP format | String | h323-setup-time=23:24:19.810 UTC Sun Sep 26 2001 |
| 26 | h323-connect-time -28 | Connect time in NTP format | String | h323-connect-time=23:24:19.810 UTC Sun Sep 26 2001 |
| 26 | h323-disconnect-time -29 | Disconnect time in NTP format | String | h323-disconnect-time=23:24:19.810 UTC Sun Sep 26 2001 |
| 26 | h323-disconnect-cause -30 | Q.931 disconnect cause code | String | h323-disconnect-cause=16 |
| 26 | h323-call-origin - 26 | 'Originate' or 'Answer' | String | h323-call-origin =Originate |
| 26 | h323-remote-address-23 | IP address of the Out-Bound gateway | String | h323-remote-address=192.168.19.15 0 |
| 44 | Acct-Session-Id | A unique accounting identifier-match start & stop | String | 8 bytes, like 00012345 |
| 46 | Acct-Session-Time | For how many second the user receive the service | Numeric | |
| 41 | Acct-Delay-Time | No of seconds tried in sending a particular record | Numeric | 5 |

Authentication Request Attributes

| Attribute | NAME | Description | Format | Sample |
|-----------|--------------------|---|---------|---------------------------|
| 4 | NAS-IP-Address | IP Address of the In-Bound gateway | Numeric | 4 bytes unsigned long |
| 61 | NAS-Port-Type | Physical port type | Numeric | 0: Asynchronous |
| 6 | Service-Type | Type of service requested | Numeric | 8: Authentication Only |
| 1 | User-Name | Account number (with 4 digit pin number on postfix) | String | 5500033440 |
| 31 | Calling-Station-Id | Calling Party Number (ANI) | String | 886282265699 |
| 26 | h323-conf-id - 24 | GUID | String | h323-conf-id=xxx |
| 2 | User-Password | 16 octets user password | String | |

Authentication Response Attribute

| Attribute | NAME | Description | Format | Sample |
|-----------|-------------------------|--|--------|--|
| 26 | h323-return-code -103 | The reason for failing authentication | String | h323-return-code=0 0: Authenticated 1: Invalid Account 2: Invalid pin number 3: Account in use 5: Account Expired 6. Over Credit Limit 7: Denied User 10: Number of Retries Exceeded 11: Insufficient Balance |
| 26 | h323-credit-amount -101 | Amount of credit (currency) remaining in the account | String | h323-credit-amount=13.25 |
| 26 | h323-billing-model -109 | Type of billing service for a specific call. | String | h323-billing-model=1 0:Credit (Post Paid) 1:Debit (Prepaid) |
| 26 | h323-currency-type -110 | Currency for use with h323-credit-amount | String | h323-currency-type=USD ISO 4217 USD America, Dollars EUR Euro GBP U.K., Pounds |

Authorization Request Attributes

| Attribute | NAME | Description | Format | Sample |
|-----------|--------------------|---|---------|-----------------------|
| 4 | NAS-IP-Address | IP Address of the In-Bound gateway | Numeric | 4 bytes unsigned long |
| 61 | NAS-Port-Type | Physical port type | Numeric | 0: Asynchronous |
| 6 | Service-Type | Type of service requested | Numeric | 5: Outbound |
| 1 | User-Name | Account number (with 4 digit pin number on postfix) | String | 5500033440 |
| 30 | Called-Station-Id | Destination phone number | String | 862587654321 |
| 31 | Calling-Station-Id | Calling Party Number (ANI) | String | 886282265699 |
| 26 | h323-conf-id - 24 | GUID | String | h323-conf-id=xxxx |
| 2 | User-Password | 16 octets user password | String | |

Authorization Response Attributes

| Attribute | NAME | Description | Format | Sample |
|-----------|-----------------------|--|--------|---|
| 26 | h323-return-code -103 | The reason for failing authentication | String | h323-return-code=0 0: Authenticated 1: Invalid Account 2: Invalid pin number 3: Account in use 4: Zero Balance 5: Account Expired 6: Over Credit Limit 7: Denied User 9: Called Number Blocked 10: Number of Retries Exceeded 11: Insufficient Balance |
| 26 | h323-credit-time -102 | Number of seconds for which the call is authorized | String | h323-credit-time=360 |

Appendix 10 Quick Start Check List

Host Network:

IP Address: _____ . _____ . _____ . _____
Sub-Mask: _____ . _____ . _____ . _____
Default-Gateway: _____ . _____ . _____ . _____

Interface Network:

IP Address: _____ . _____ . _____ . _____
Sub-Mask: _____ . _____ . _____ . _____
Default-Gateway: _____ . _____ . _____ . _____

► H.323 Call:

VoIP Configuration:

Register to Gatekeeper
GK IP Address: _____ . _____ . _____ . _____
Phone
GK RAS Port: _____

Peer To Peer
Refer to User Guide-

Book setting

H.245 tunneling: Enable Disable

Fast Connect: Enable Fast Start Early H.245 Disable

Separate H.245 after Fast Start: Yes No

Fast Start Enabled Codec:

- G.711 a-law
- G.711 u-law
- G.729
- G.729 A/B
- G.723.1 (5.3K)
- G.723.1 (6.3A)

Codec Select Priority:

- ___ G.711 a-law
- ___ G.711 u-law
- ___ G.729
- ___ G.729 A/B
- ___ G.723.1 (5.3K)
- ___ G.723.1 (6.3A)

► SIP Call:

VoIP Configuration: Peer To Peer

Register to SIP Proxy Server
Phone

Refer to User Guide-

Registrar Proxy Server: _____ . _____ . _____ . _____
Registrar Proxy Port: _____
Registrar User ID: _____
Registrar Password: _____
Outbound Proxy Server: _____ . _____ . _____ . _____
Outbound Proxy Port: _____
Outbound User: _____
Outbound Port: _____

Book setting

180 SDP: Yes No

183 SDP: Yes No

Local Codec Codec:

G.711 a-law

G.711 u-law

G.729

None

G.723.1 (5.3K)

G.723.1 (6.3A)

Accept Proxy Call Only: Yes No

PSTN Interface:

PCM encoding: A-law Mu-law

PCM Idle Pattern:

Default (-1): 0x55 A-law 0xff u-law specified: _____

Clock Source: External Internal

E1

Framing Method:

Automatic CRC4 or Double Frame selection

Double Frame Format

CRC4 multi-frame

CRC4 extend multi-frame

Line Code: HDB3 AMI

ISDN/PRI:

Termination Site: Network User site

Variance:

Euro ISDN

Australia ISDN

Hong Kong ISDN

Korea ISDN

CAS:

CAS Idle ABCD signal: Default (-1): specified: _____

E1 MFC R2

E1 CAS R2

Variance:

E1 R2 MF Aregntina ANI

E1 R2 MF Aregntina ANI 7digits

E1 R2 MF Aregntina no ANI

E1 R2 MF Aregntina no ANI 7 digits

E1 R2 MF Bolivia ANI

E1 R2 MF Bolivia ANI 7digits

E1 R2 MF Bolivia no ANI

E1 R2 MF Bolivia no ANI 7 digits

E1 R2 MF Brazil ANI

E1 R2 MF Brazil ANI 7digits

E1 R2 MF Brazil no ANI

E1 R2 MF Brazil no ANI 7 digits

E1 R2 MF Chile ANI

E1 R2 MF Chile ANI 7digits

E1 R2 MF Chile no ANI

E1 R2 MF Chile no ANI 7 digits

E1 R2 MF China ANI

p E1 R2 MF China ANI 7digits
p E1 R2 MF China no ANI
p E1 R2 MF China no ANI 7 digits
p E1 R2 MF Czech-Republic ANI
p E1 R2 MF Czech-Republic ANI 7digits
p E1 R2 MF Czech-Republic no ANI
p E1 R2 MF Czech-Republic no ANI 7 digits
p E1 R2 MF Egypt -ANI
p E1 R2 MF Egypt -ANI 7digits
p E1 R2 MF Egypt - no ANI
p E1 R2 MF Egypt - no ANI 7 digits
p E1 R2 MF India – 10 Digits no ANI
p E1 R2 MF India – 10 Digits with ANI
p E1 R2 MF India – Type 1 No ANI 10
p E1 R2 MF India – Type 2 Orig ANI 10
p E1 R2 MF India – Type 2 Term ANI 10
p E1 R2 MF India – Type 2 Term No ANI 10
p E1 R2 MF India – Type 2 Orig ANI 10
p E1 R2 MF India – Type 3 ANI 10
p E1 R2 MF India – Type 3 NoANI 10
p E1 R2 MF Indonesia - ANI
p E1 R2 MF Indonesia - ANI 7digits
p E1 R2 MF Indonesia - no ANI
p E1 R2 MF Indonesia - no ANI 7 digits
p E1 R2 MF Israel(Bezeq) - ANI
p E1 R2 MF Israel(Bezeq) - ANI 7digits
p E1 R2 MF Israel(Bezeq) -c no ANI
p E1 R2 MF Israel(Bezeq) - no ANI 7 digits
p E1 R2 MF ITU - ANI
p E1 R2 MF ITU - ANI 7digits
p E1 R2 MF ITU - no ANI
p E1 R2 MF ITU - no ANI 7 digits
p E1 R2 MF KOREA - ANI
p E1 R2 MF KOREA - ANI 7digits
p E1 R2 MF KOREA - no ANI
p E1 R2 MF KOREA - no ANI 7 digits
p E1 R2 MF Malaysia - ANI
p E1 R2 MF Malaysia - ANI 7digits
p E1 R2 MF Malaysia - no ANI
p E1 R2 MF Malaysia - no ANI 7 digits
p E1 R2 MF Mexico - ANI
p E1 R2 MF Mexico - ANI 7digits
p E1 R2 MF Mexico - no ANI
p E1 R2 MF Mexico - no ANI 7 digits
p E1 R2 MF Philippines - ANI
p E1 R2 MF Philippines - ANI 7digits
p E1 R2 MF Philippines - no ANI
p E1 R2 MF Philippines - no ANI 7 digits
p E1 R2 MF Thailand -Republic ANI
p E1 R2 MF Thailand - ANI 7digits
p E1 R2 MF Thailand - no ANI
p E1 R2 MF Thailand - no ANI 7 digits
p E1 R2 MF Uruguay - ANI
p E1 R2 MF Uruguay - ANI 7digits
p E1 R2 MF Uruguay - no ANI
p E1 R2 MF Uruguay - no ANI 7 digits

-
- E1 R2 MF Venezuela - ANI
 - E1 R2 MF Venezuela - ANI 7digits
 - E1 R2 MF Venezuela - no ANI
 - E1 R2 MF Venezuela - no ANI 7 digits

T1

Framing Method:

- super frame
- 4-frame multi-frame
- 12 frame multi-frame (D4)
- extend super frame without CRC6
- extend super frame with CRC6
- 72-Frame Multi-Frame

Line Code: AMI B8ZS

ISDN/PRI:

Termination Site: Network User site

Variance:

- NI2 ISDN
- 5ESS 9 ISDN
- 5ESS 10 ISDN
- DMS100 ISDN
- NTT ISDN (INS1500)

CAS:

CAS Idle ABCD signal: Default (-1): specified: _____

T1 CAS

Variance:

- T1 E&M BellCore Feature Group D Wink Start
- T1 E&M Delay Start
- T1 E&M Feature Group A Immediate Start
- T1 E&M Feature Group B Wink Start
- T1 E&M Feature Group D Wink Start(ANI B4 ADDR)
- T1 E&M Feature Group D Wink Start
- T1 E&M FGAImediate
- T1 E&M FGB Wink
- T1 E&M FGB Wink(ANI B4 ADDRESS)
- T1 E&M FGD Wink
- T1 E&M Immediate
- T1 E&M Immediate Start
- T1 E&M Wink
- T1 E&M WinkStart A-Bit Only FXO
- T1 E&M WinkStart A-Bit Only FXS
- T1 E&M Wink Start
- T1 GroundStart FXO
- T1 GroundStart FXS
- T1 LoopStart FXO
- T1 LoopStart FXS

VIP-2100 FAQ

Q1. Forgotten user password to logon VIP-2100.

Answer:

- a. Logon by a user has Administrator right to reset the user's password
- b. Use the LCD control panel to change the user id: admin or root's password.

Q2. In H.323 Mode: Cannot hear ring back tone for PSTN caller.

Answer:

Normally, the ring back tone is generated by the nearest PABX connected to VIP-2100. If a caller from PSTN site cannot hear the ring back tone, please check:

- a. Consult to PABX/PSTN vender to clarify the PABX/PSTN will generate ring back tone.
- b. For Fast Start mode, make sure the far end VoIP end point will have ring tone generated. For example, a PSTN subscriber calls a VoIP H.323 IP Phone. When it is on Fast Start mode, VIP-2100 will cut through the voice path after receive Fast Start Ack. Please make sure the Far End VoIP Endpoint will generate ring back tone over RTP media path to VIP-2100.
- c. If you really need VIP-2100 to generate PSNT ring back tone, please do the following setting:
 - Turn on "local ring back" from "Interface -> Trunk" for each trunk required local ring back tine generation.
 - Disable Fast Start for H.323 outgoing call (set to disable or H.323 incoming call only.)

Q3. In H.323 Mode: VIP-2100 cannot keep registering to Gatekeeper after Gatekeeper restarted.

Answer:

- a. Check whether VIP2100's register time to live is too long or not. If yes, make it shorter from "H.323 -> Register Time to Live". If we make it longer, it means it might need take long time to re-register to Gatekeeper after Gatekeeper failed or restart. If it is very short, will cause more IP traffic.
- b. Check whether Gatekeeper has a preset TTL setup or not. If so, the GK TTL will overwrite the VIP-2100's TTL request by using the default value.

Q4. In SIP Mode: VIP-2100 cannot keep registering to SIP Register Server after Register Server restarted.

Answer:

- a. Check whether VIP-2100's registrar IP address, port, user id and password are correct.
- b. Check whether Register Server has a preset TTL setup or not. If so, the Register Server TTL will overwrite the VIP-2100's TTL request by using the default value.

Q5. VIP-2100 cannot make a success call.

Answer:

- a. Check PSTN trunk ready to work or not. You need to have the following event generated - "9504: trunk alarm clear (trunk #)"
- b. Check VIP-2100 is registered to H.323: Gatekeeper /SIP: Register Server or not. You need to have the following event generated - "9503: H323 GK/ SIP Register [xxx.xxx.xxx.xxx] found & registered"
- c. Check the digit manipulation setting is correct or not. Make sure you have DM put into call flow editor, interface or VoIP.
- d. For P2P call, make sure you have the address book setting for dialed number.

Q6. Cannot hear voice after the calls connect.

Answer:

- a. Make sure the interface and host Ethernet are well connected.
- b. Ping each related IP to see network is working or not.
- c. The voice codec priority should be matched both side.

Q7. In H.323 Mode: Failed to setup a fast start call.

Answer:

- a. Make sure the far end "Fast Start" is enabled.
- b. Check whether Gatekeeper can support Fast Start or not. (Some Gatekeepers are not.)
- c. If you cannot hear early announcement, make sure the far and H.323 end point can listen RTP port before connect.

Q8. In SIP Mode: Failed to setup a normal call.

Answer:

- a. Make sure the voice codec priority should be matched both side.
- b. Make sure the VIP-2100 accept proxy call only or not.

Q9. In SIP Mode: Failed to hear the early media before call connected.

Answer:

- a. Make sure the 180 SDP or 183 SDP are enabled.
- b. Make sure the remote SIP end point can cut through voice before call connected.

Q10. Cannot send or receive the DTMF to/from far end VoIP end point.

Answer:

- a. *In H.323 Mode:* Make sure VIP-2100 and other endpoint use same DTMF relay mode (e.g. H.245 Alphanumeric.)
- b. *In SIP Mode:* Make sure VIP-2100 and other endpoint use same DTMF relay mode (e.g. SIP Info or RFC2833-payload type.)
- c. *In SIP Mode:* Make sure the remote SIP end point's RTP payload type is supported or not.
- d. If use Q.931 UUI DTMF Relay mode, make sure Gatekeeper can correctly forward Q.931 UUI when registering to Gatekeeper is set to true.

Q8. Does VIP-2100 can cooperate with the Cisco VoIP products?

Answer:

The short answer is “yes”. The following configuration example can be used for normal and fax call.

Example for H.323 Mode

```
voice service voip
  fax protocol t38 ls-redundancy 1 hs-redundancy 1

voice class codec 100
  codec preference 1 g723r63
  codec preference 2 g729r8
  codec preference 3 g711ulaw
  codec preference 4 g711alaw

dial-peer voice 100 pots
  application session
  destination-pattern 8001
  progress_ind progress enable 8
  port 1/1/0

dial-peer voice 200 voip
  destination-pattern 2T
  voice-class codec 100
  session target ras
  dtmf-relay h245-signal h245-alphanumeric
  fax rate 14400
  fax-relay ecm disable
  fax protocol t38 ls-redundancy 1 hs-redundancy 1
  no vad
```

Example for SIP Mode

```
voice service voip
  fax protocol t38 ls-redundancy 1 hs-redundancy 1
  dial-peer voice 300 voip
  destination-pattern 20T
  rtp payload-type nte 110
  voice-class codec 88
  session protocol sipv2
  session target ipv4:192.168.5.205
  dtmf-relay rtp-nte
  fax-relay ecm disable
  fax rate 14400
  fax protocol t38 ls-redundancy 2 hs-redundancy 2 fallback cisco
  no vad

dial-peer voice 250 voip
  application session
  destination-pattern 2T
  voice-class codec 2
  session protocol sipv2
```

```
session target sip-server
fax rate 14400
fax protocol t38 ls-redundancy 1 hs-redundancy 1 fallback cisco
```

```
sip-ua
```

```
line con 0
speed 115200
line aux 0
line vty 0 4
```

.....

Q9. VIP-2100 cannot register to Cisco gatekeeper.

Answer:

- a. Make sure GK IP and port number is correct.
- b. If the gatekeeper can only allow predefined endpoint, make sure VIP-2100 has it defined.
- c. If you need prefix support, set it on GK.

Q10. External Radius server does not work.

Answer:

- a. Make sure "VoIP Authentication method" is set to " external AAA".
- b. Make sure "AAA" component is used in the call flow editor to take effect.
- c. Make sure Radius server IP and port for authentication & billing are correct.

Q11. Internal Radius server does not work.

Answer:

- a. Make sure "VoIP Authentication method" is set to " internal AAA".
- b. Make sure "AAA" component is used in the call flow editor to take effect.
- c. Make sure only debit user is used for VoIP caller.

Q12. PSTN hunting Group does not work.

Answer:

- a. Make sure "PSTN hunting Group" component is used in the call flow editor to take effect.
- b. Make sure "prefix" is met your target dialed number.