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

VIP-2100

CONTENTS

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/Outbounbd Proxy Server Support (Outbond 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

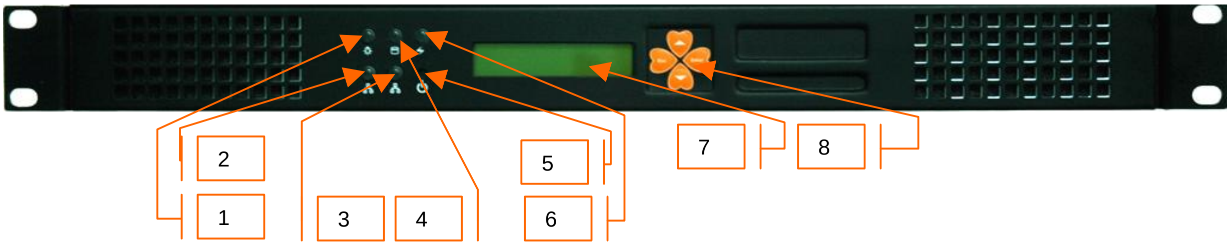
SIP Protocol Support		
1	Cause Code Mapping	Yes
2	HTTP Digest Authentication	Yes
3	SIP Call on Hold	Yes
4	SIP Early Media	Yes
5	SIP Overload Redirect	Yes
6	SIP Transfer (unattend)	Yes
7	SIP Transfer (attend)	Yes
8	SIP/TCP	Yes
9	SIP/UDP	Yes
10	SIP-180/SDP	Yes
11	SIP-183/SDP	Yes
12	SIP-PRACK	Yes
13	SIP-RFC 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 Registar 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

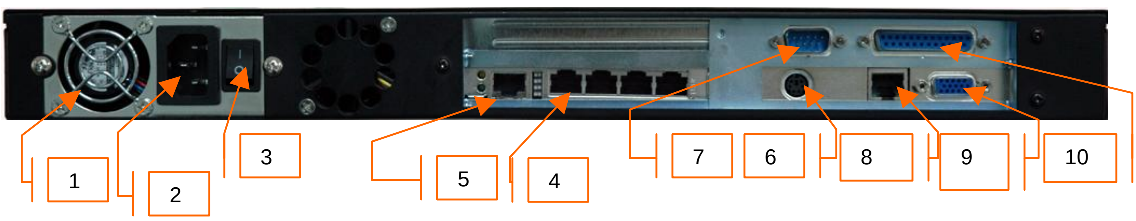
VIP-2100 Front Panel:



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.

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.

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.

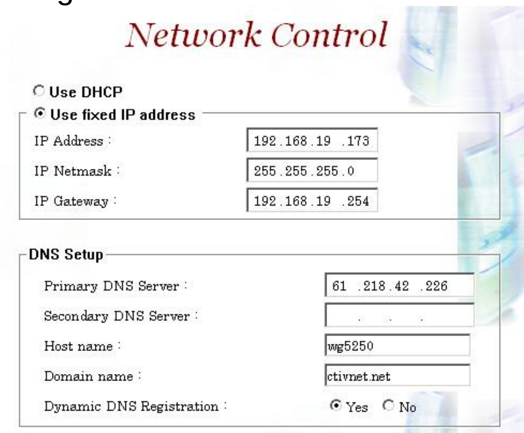


Figure 2.2-1

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

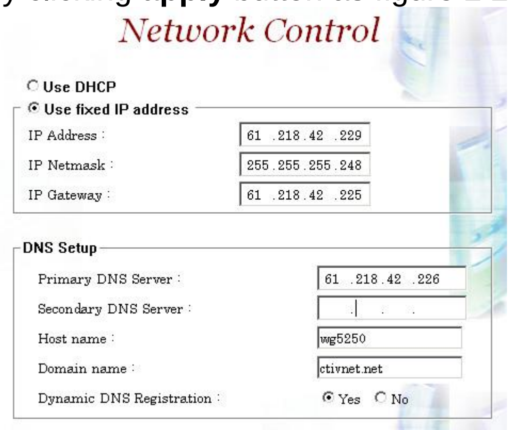


Figure 2.2-2

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.

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.

Figure 2.3-2

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

Figure 2.3-3

Step 4: The screen will shows ***"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.

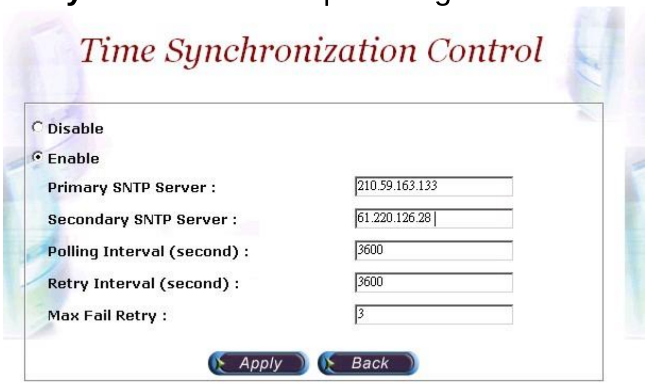


Figure 2.3-5

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.

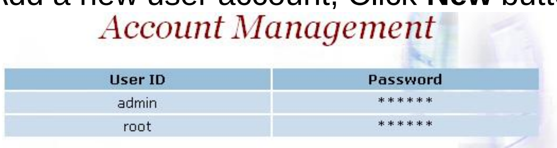


Figure 2.4-1

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 relogon by another user account as figure 2.5-1.

Figure 2.5-1

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

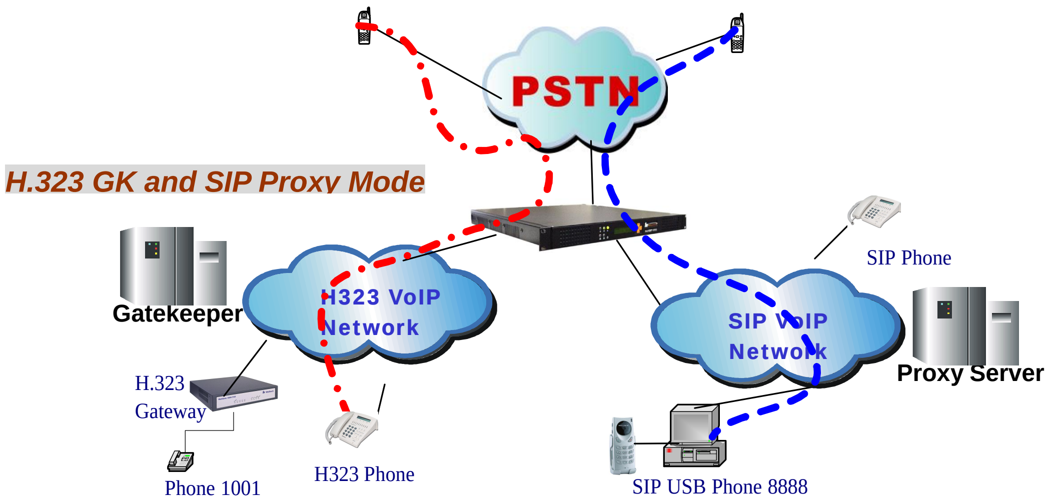
Figure 2.5-2

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.

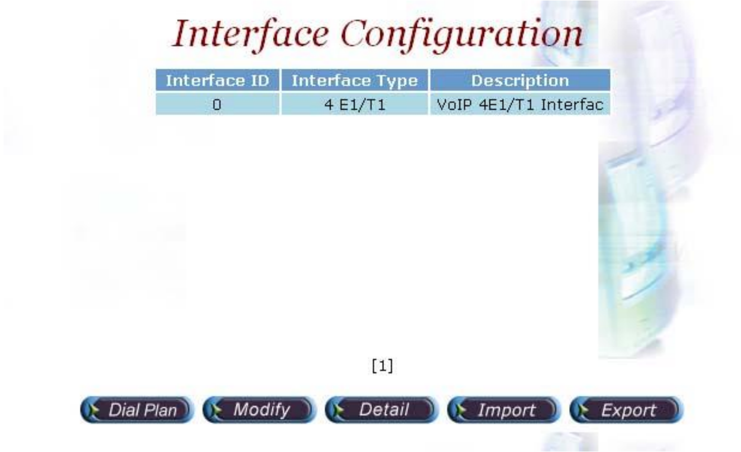


Figure 3.1-1

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

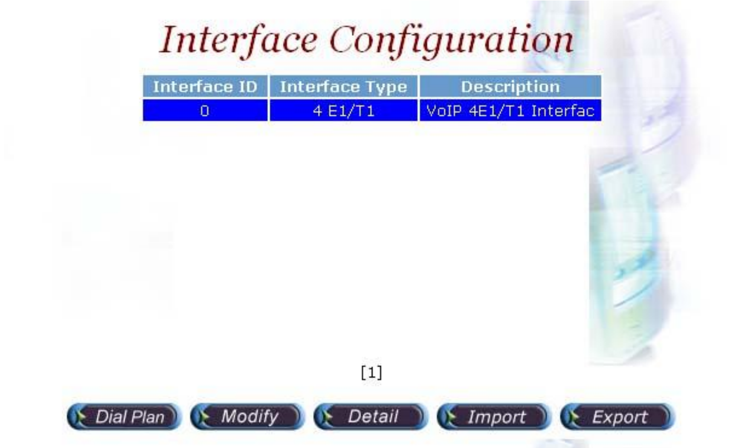


Figure 3.1-2

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.

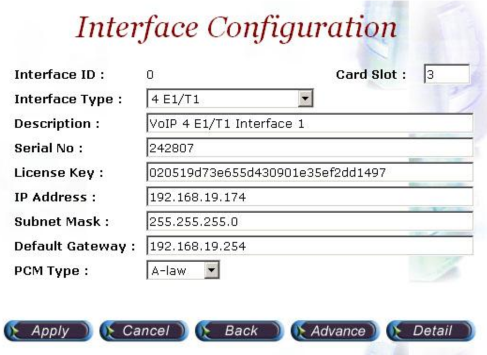


Figure 3.1-3

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.

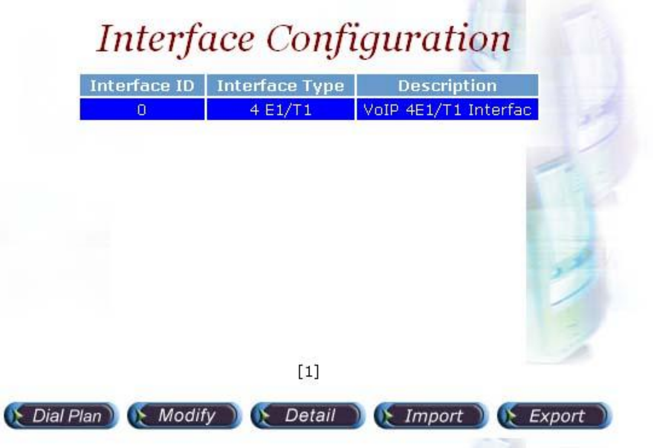


Figure 3.1-4

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.

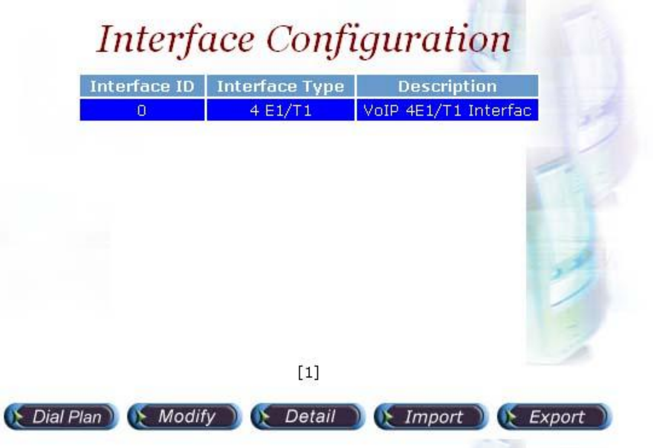


Figure 3.2-1

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

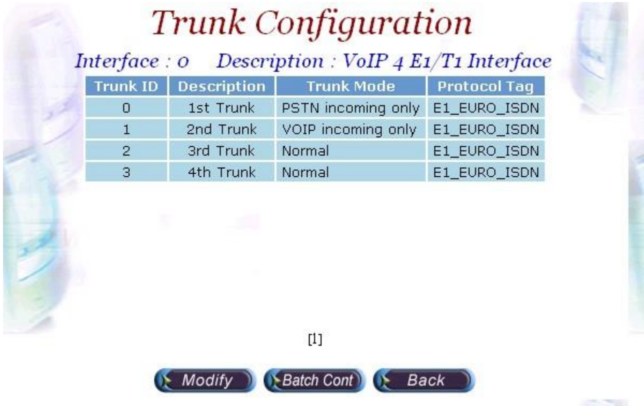


Figure 3.2-2

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.

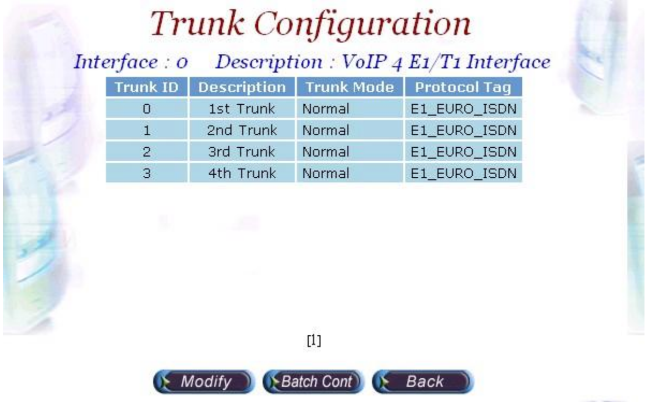


Figure 3.2-4

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.

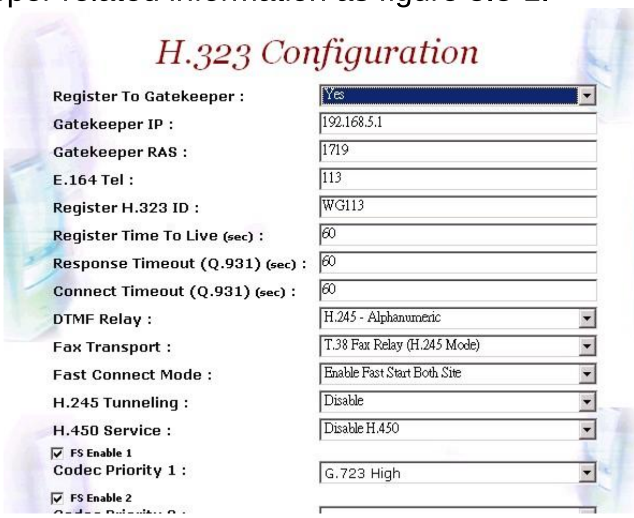


Figure 3.3-1

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 Registar Server: 192.168.19.150
- Primary Registar Port: 5060
- Primary Registar User: 173
- Primary Registar 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:

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

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)

H.323 incoming call

↓

Dialed number: 582265699

↓

Match the pattern 5

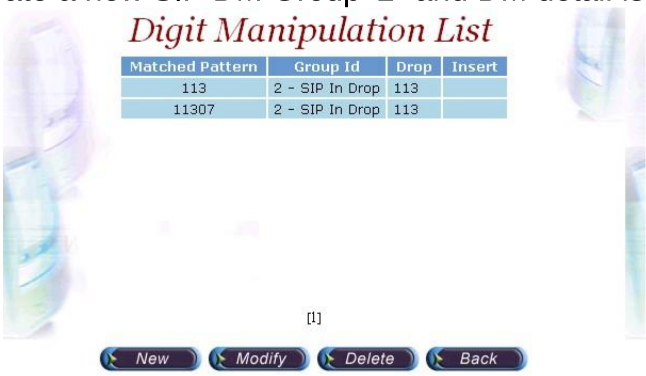
↓

Delete 5 (Drop)

↓

New dialed number becomes 82265699

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



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

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)

SIP incoming call
↓
Dialed number: 11307688222
↓
Match the pattern 11307
↓
Delete 113 (Drop)
↓
New dialed number becomes 07688222

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)

PSTN incoming call (DNIS mode)
↓
Dialed number: 02822656991001
↓
Match the pattern 0282265699
↓
Delete 0282265699 (Drop)
↓
New dialed number becomes 1001

☻ ***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

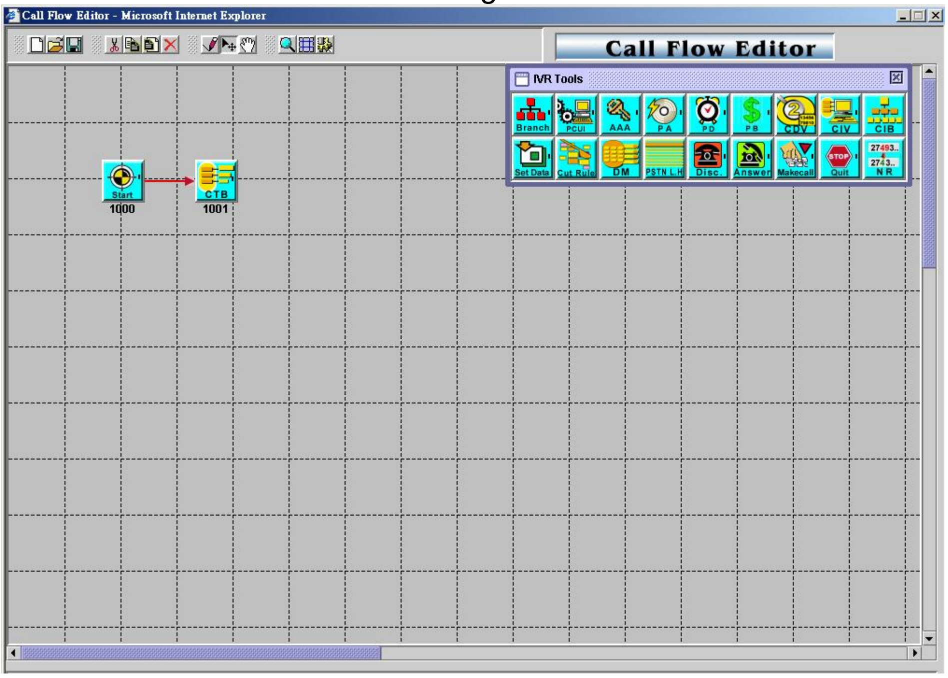


Figure 4-1

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

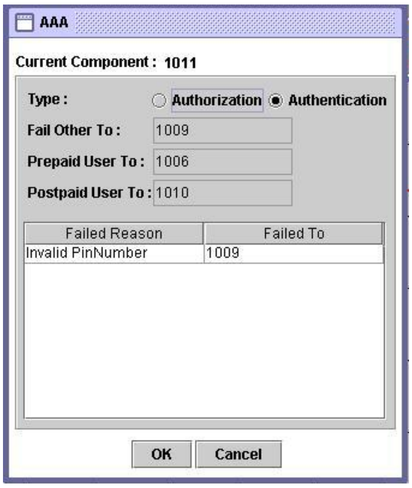
Step 2: Drag and prop the required component icon into the workspace as figure 4-2.

Figure 4-2

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

- 

AAA: Send Authorization or Authentication for validation



  - o Type: AAA type selection
    - Authorization: Send RADIUS Authorization packet out
    - Authentication: Send RADIUS Authentication packet out

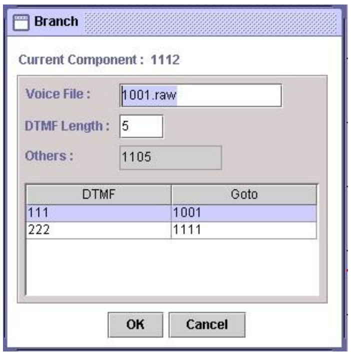
    Success To: Success to component

    Failed other to: Failed to component

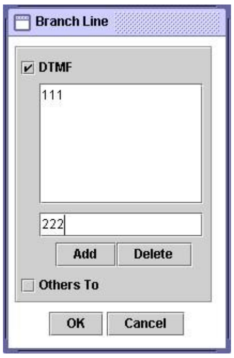
- o Failed Reason: Return code from RADIUS server
- o **Line Propriety:**
  - Invalid Account
  - Account In Use
  - Zero Balance
  - Account Expired
  - Over Credit Limit
  - Number of Retries Exceeded
  - Insufficient Balance

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

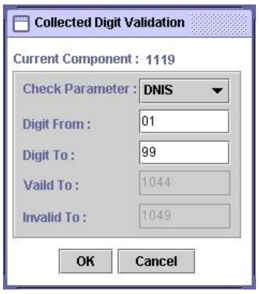
- Answer: Answer incoming call (PSTN only)
- Branch: Play an announcement and branch into different route



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

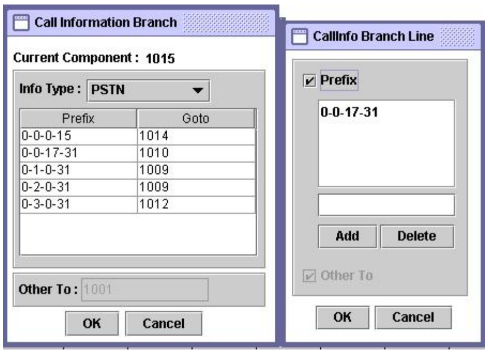


- CDV: Collected Digit Validation



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

- CIB: Call Information Branch



- o 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:

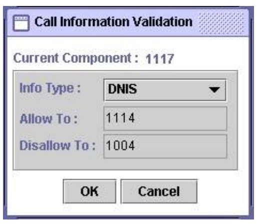
    0: Interface ID (Always 0)

    1: Trunk ID: 1

    17: Start from B Channel 17

    31: Stop from B Channel 31
- o Goto: The component to run next
- o 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



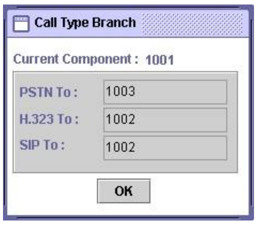
- o Info Type: The infor type to be validation

-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

- 

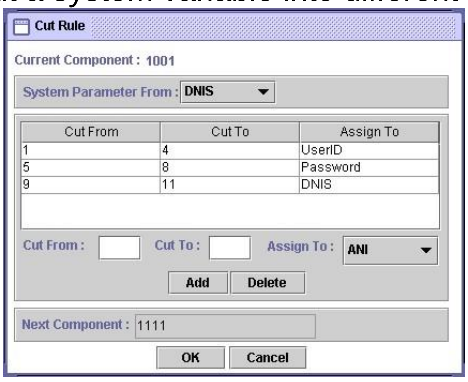
CTB: Call Type Branch



  - 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



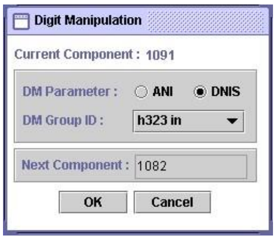
  - Cut From: Cut start digit from (start from 1)
  - Cut To: Cut end digit to
  - Assign To: Store the cutted result into

- 

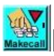
Disconnect: Disconnect the call

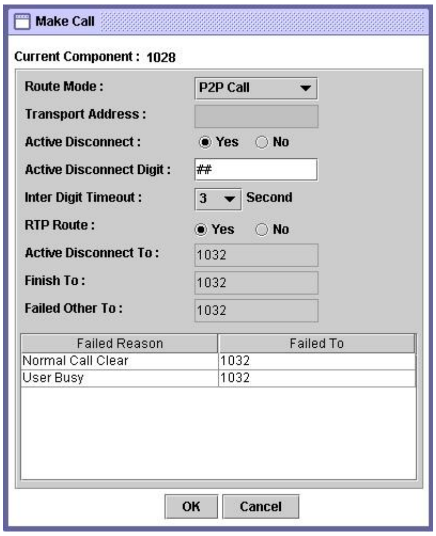
- 

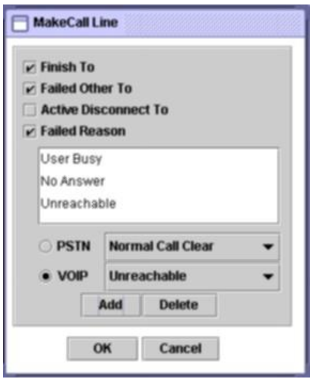
DM: Digit Manipulation



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

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

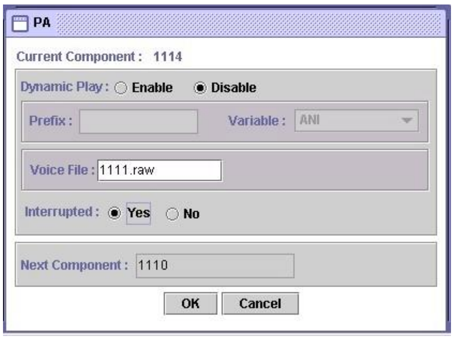




- o Route Mode: Gatekeeper Call or P2P Call or PSTN…etc. (for PSTN incoming call, please select the Gatekeeper, P2P Call, or SIP Proxy call TA; for H.323/SIP incoming call, please select the PSTN call)
- o Transport Address: When used for “H.323 TA” routing mode, the format used is “Ipaddr:port” (e.g.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

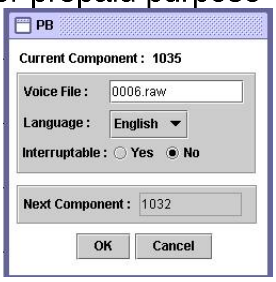
- o **Line Propriety:**
  -PSTN: PSTN disconnect reason code:
  -Normal Call Clear
  -User Busy
  -No User Response
  -No Answer
  -Call Reject
  - VoIP: VoIP disconnect reason code:
  -User Busy
  -No Answer
  -Unreachable
  -Other

- PA: Play Announcement



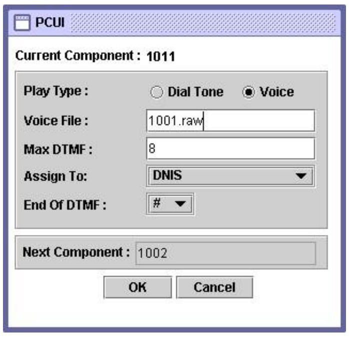
- o Dynamic Play: Dynamic play voice file by combine prefix and variable as the file name
- o Enable: Combine prefix to variable as the voice file to play
  -Prefix: Voice file prefix **(e.g. prefix: WT, variable: user1 (contact 201, played voice file is “WT201.raw”)**
  -Variable: Variable to be appending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

- PB: Play Balance for prepaid purpose



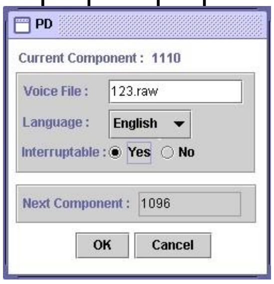
- o Voice File: Voice prompt file
- o Language: Play balance language section
  -English
  -Chinese.
- o Interrupted: Voice can be interrupted or not

- PCUI: Prompt and Connect User Information



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

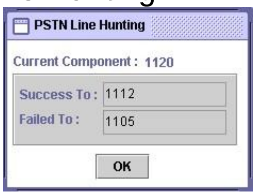
- PD: Play Duration for prepaid purpose



  - o Voice File: Leading voice prompt file
  - o Language: Play duration language section
    -English
    -Chinese
  - o 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.***

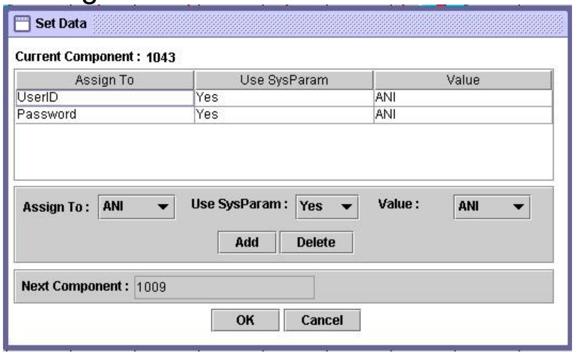
- PSTN L.H: PSTN Line Hunting



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

- 

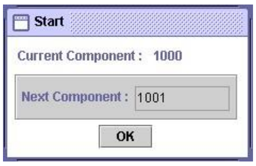
Set Data: Assign value to a variable



  - 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

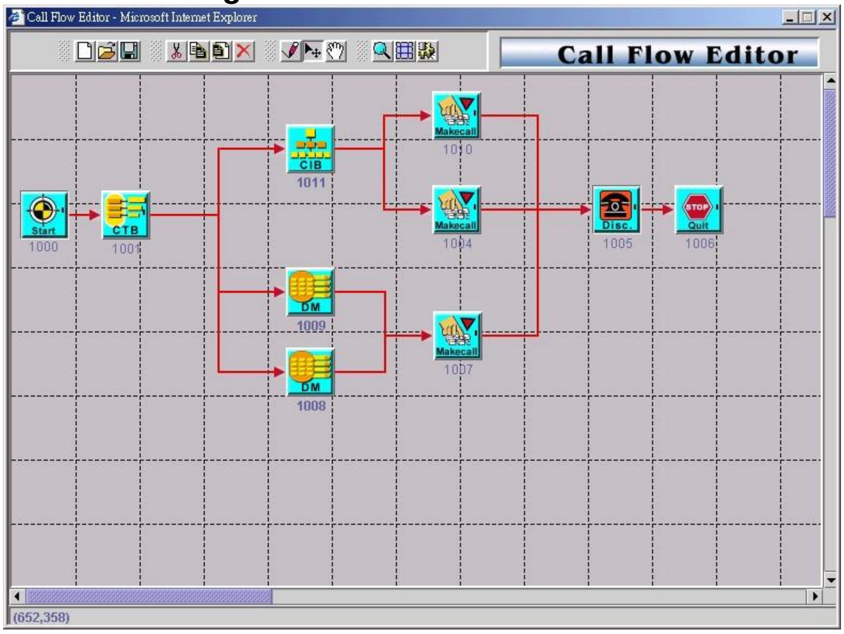


  - Next Component

- 

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	

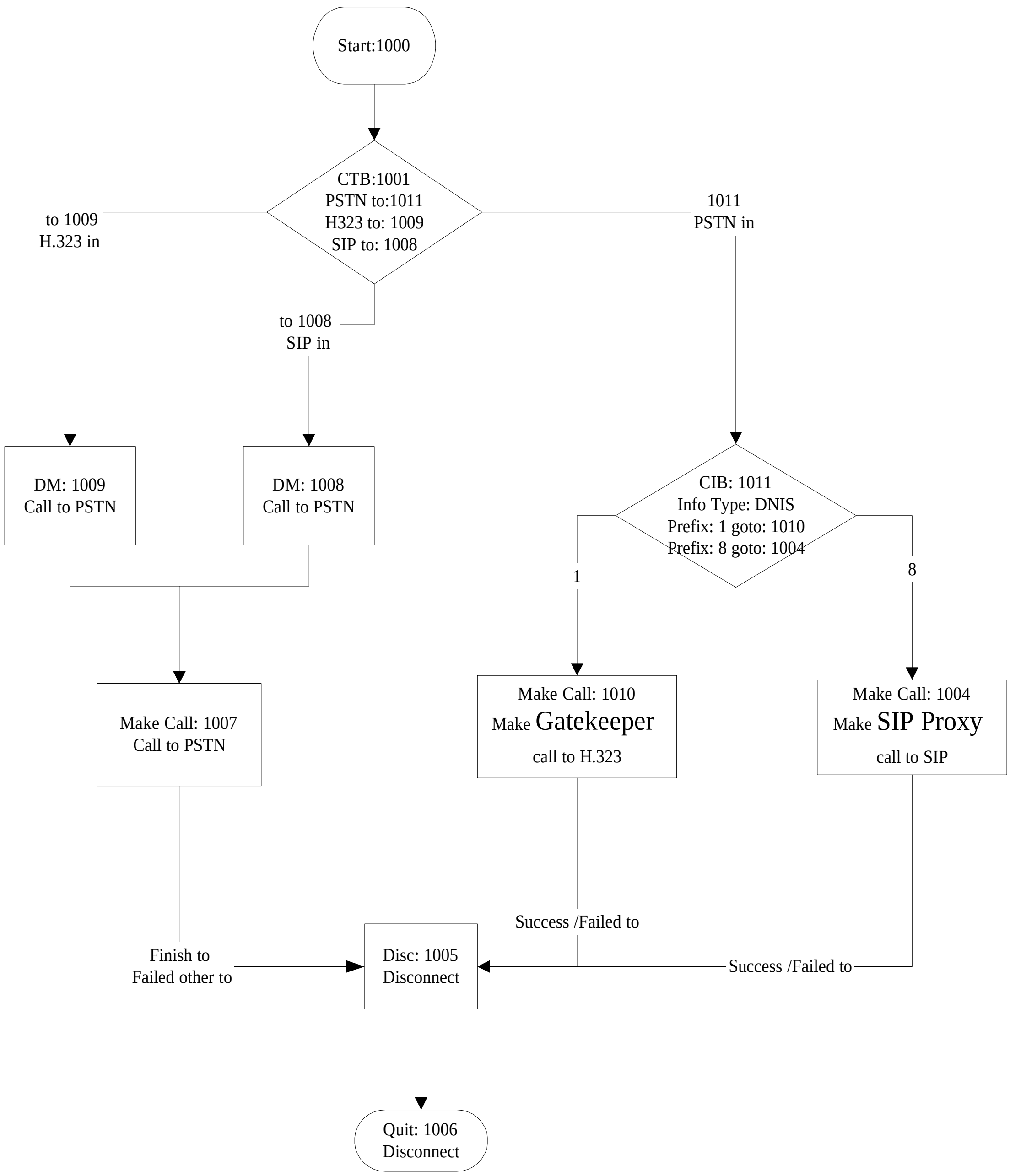
1001 Route for H.323Gatekeeper call

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.

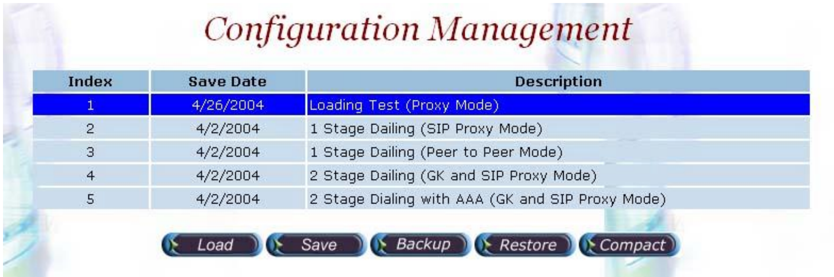


Figure 3.7-1

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.

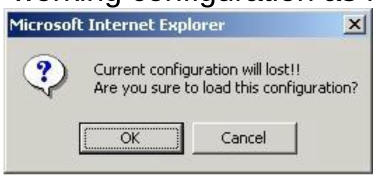


Figure 3.8-2

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.

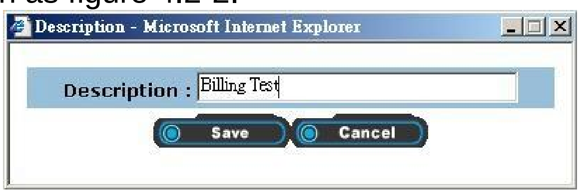


Figure 3.8-3

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

Index	Save Date	Description
1	4/26/2004	Loading Test (Proxy Mode)
2	4/26/2004	Billing Test
3	4/2/2004	1 Stage Dailing (Peer to Peer Mode)
4	4/2/2004	2 Stage Dailing (GK and SIP Proxy Mode)
5	4/2/2004	2 Stage Dialing with AAA (GK and SIP Proxy Mode)

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.

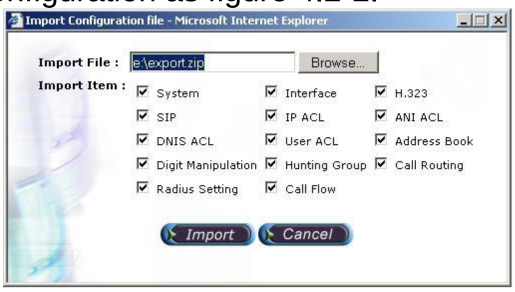


Figure 3.8-6

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 mode 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.

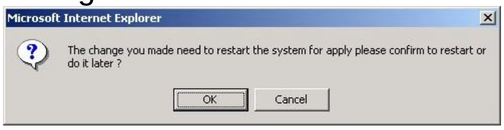
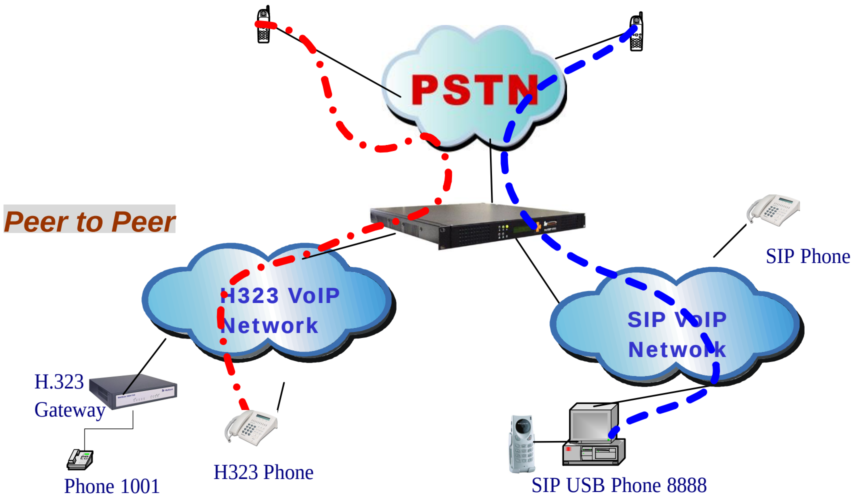


Figure 4.3-1

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 has 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.

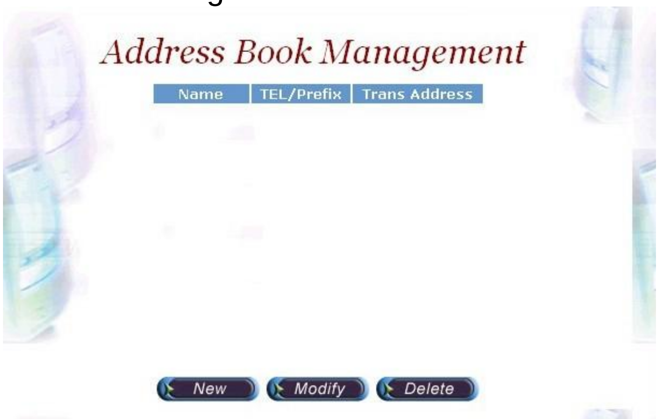


Figure 5.3-1

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

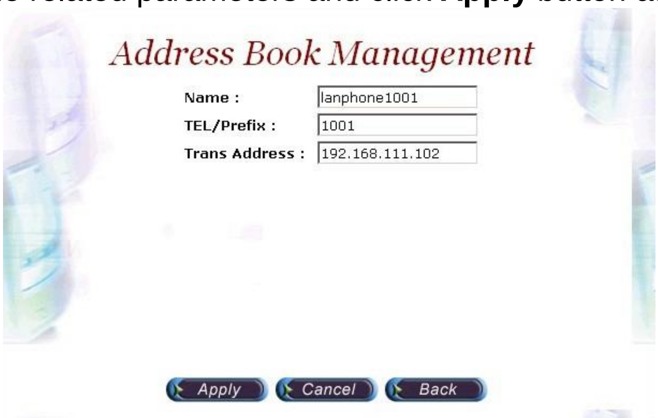
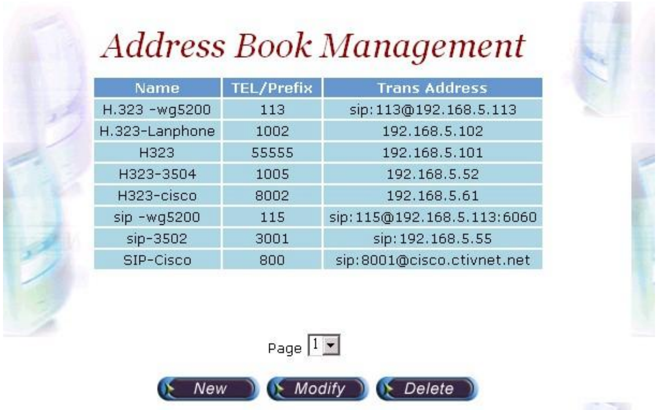


Figure 5.3-2

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.



Address Book Management

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.***

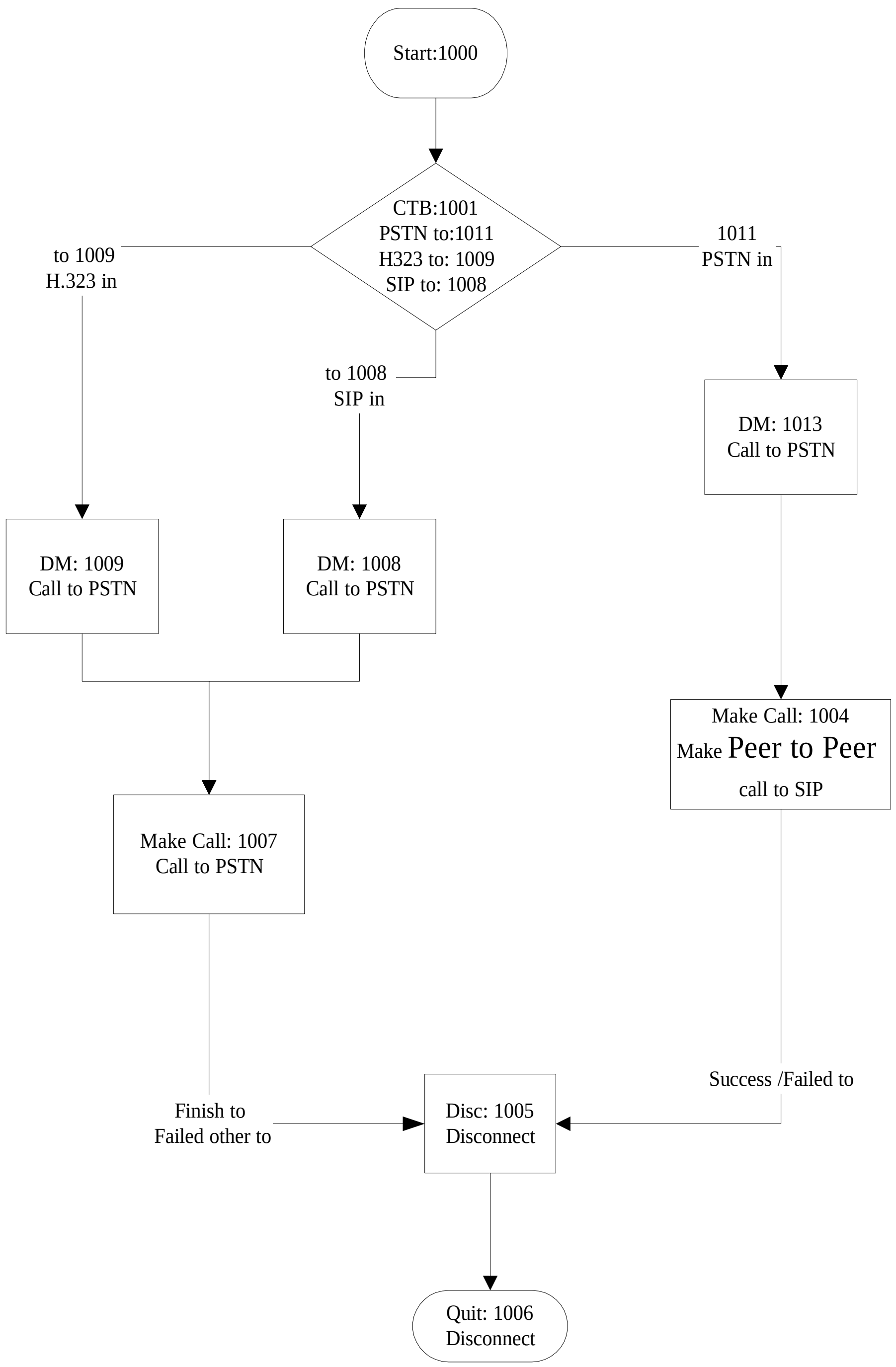
Digit Manipulation

Please refer to section "Digit Manipulation"

Call Flow Editor

Please refer to section "Call Flow Editor"

Call Flow (P2P Mode):



Configuration Manager

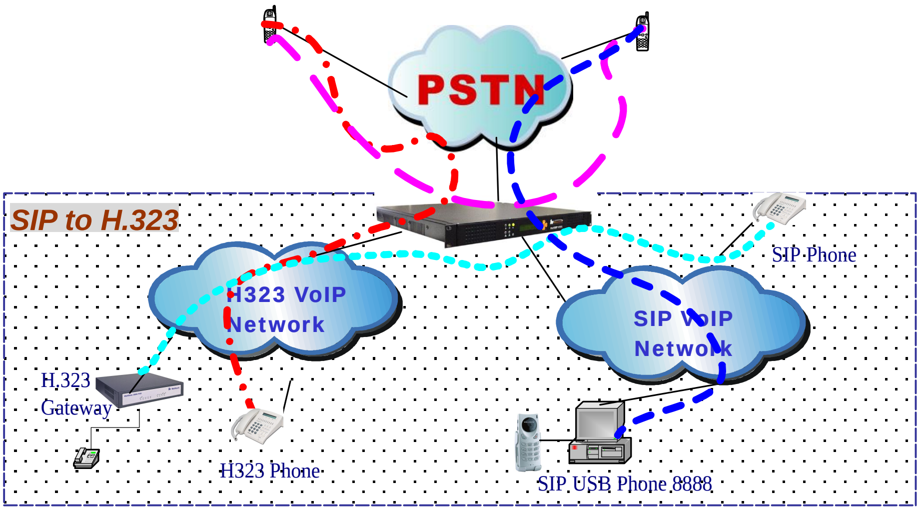
Please refer to section "Configuration Manger"

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"

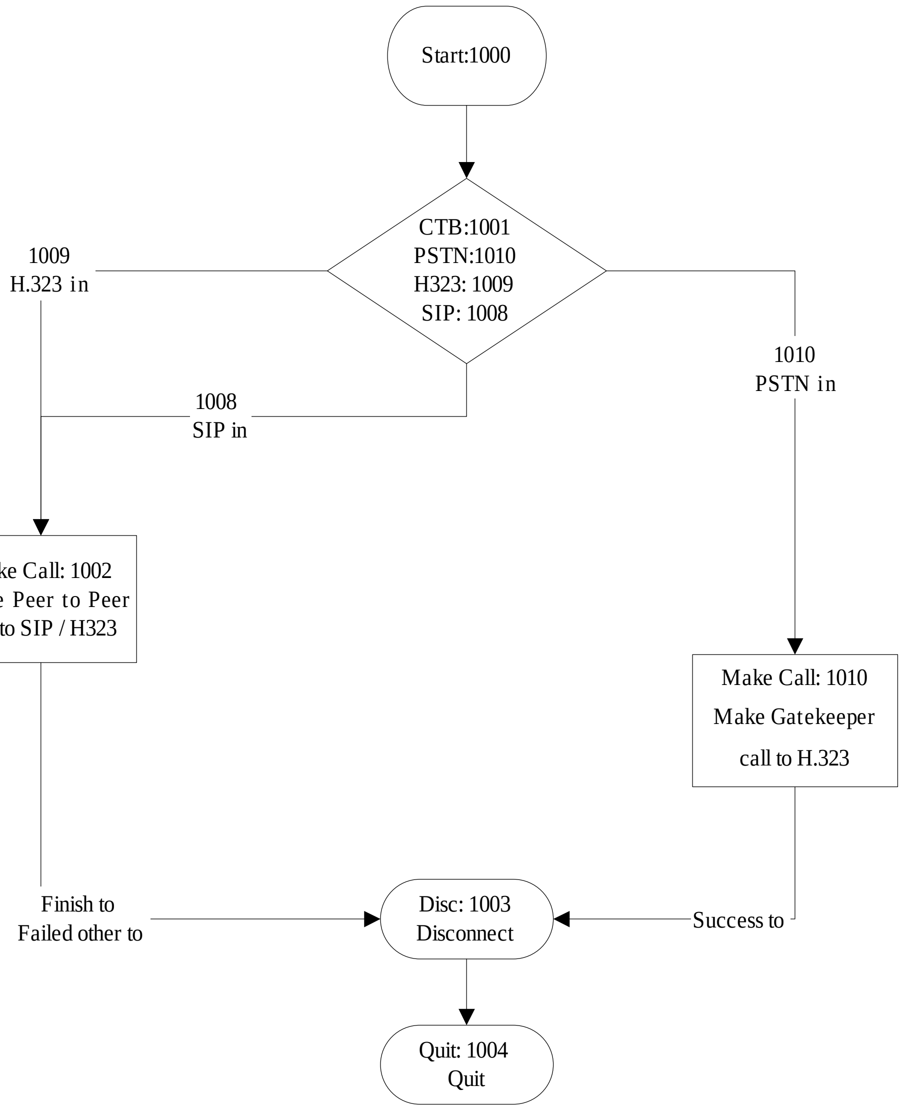
Digit Manipulation

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)
  - o File Only: Log CDR into the file only. It can be retrieved by ftp (directory c:\cd cdr).
  - o 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.
  - o 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:
  - o Critical: Show critical error messages only
  - o Warring: Show warring and critical error message only
  - o Information: Show information, warring and critical message only
  - o Debug: Show all debug messages
  - o 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

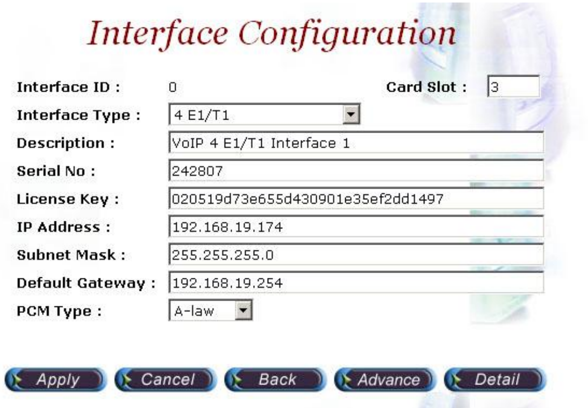


Figure 7.2-1

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

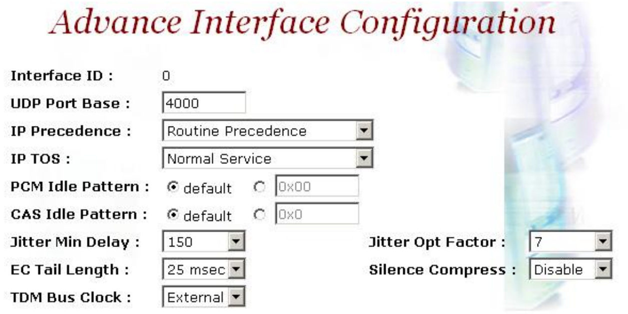


Figure 7.2-2

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
  - o Routine Precedence
  - o Priority Precedence
  - o Immediate Precedence
  - o Flash Precedence
  - o Flash Override Precedence
  - o Critical Precedence
  - o Internetwork Precedence
  - o Network Precedence
- IP TOS: Top of Service with the following priority selection
  - o Normal Service
  - o Minimize Monetary
  - o Maximize Reliability
  - o Maximize Thought
  - o 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

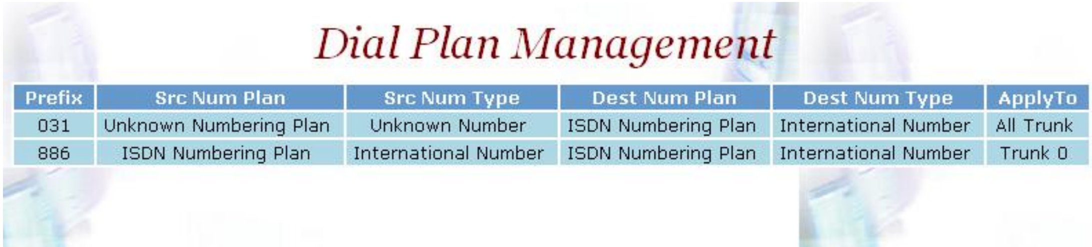
voice quality. Set to 12 will have long voice delay but with better voice quality

- EC Tail Length: Echo Cancellation Length, default value is 25ms
- Silence Compress: Enable silence compress or not
- TDM Bus Clock: TDM Bus clock source
  - o Internal: derived from internal oscillator
  - o 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 (selected)
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: T1or 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)
  - o User Side
  - o Network Side
- Trunk Mode: Trunk operation mode
  - o Disable: Disable the trunk
  - o Normal: Accept PSTN and VoIP calls
  - o PSTN incoming only: Allow the PSTN incoming calls only
  - o H.323 incoming only: Allow the H.323 incoming calls only
- Hunting Method: PSTN trunk hunting method for available channel
  - o Random: Hunt randomly
  - o Cyclic: Initial hunt (after power-up/reboot) begins with B channel 1; subsequent hunts begin with position following last successfully allocated resource
  - o Rotary: Hunt always begins with B channel 1
  - o Reverser Rotary: Hunt always begins with B channel 31
  - o 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:
  - o **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
  - o **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
  - o **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

o **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

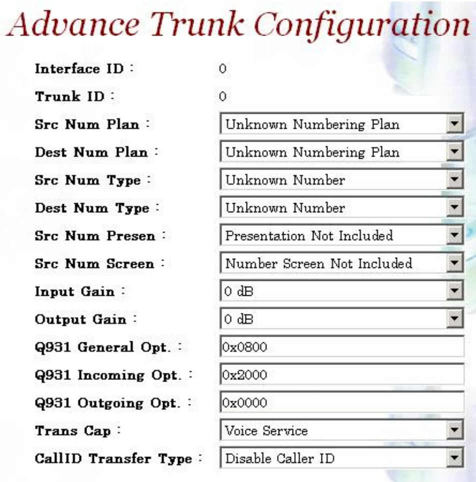


Figure 7.3-2

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

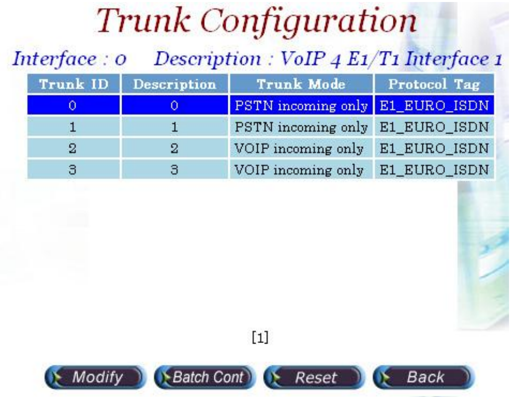


Figure 7.4-1

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

- o Disable: Don’t use Fast Start.
  - o Enable Fast Start Both Site: Use Fast Start for incoming call and outgoing H.323 calls
  - o Fast Start-H.323 incoming only: Enable Fast Start for H.323 incoming calls only
  - o Fast Start-H.323 outgoing only: Enable Fast Start for H.323 outgoing calls only.
  - o 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
  - o T.38 ECM Interoperable mode
  - o 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

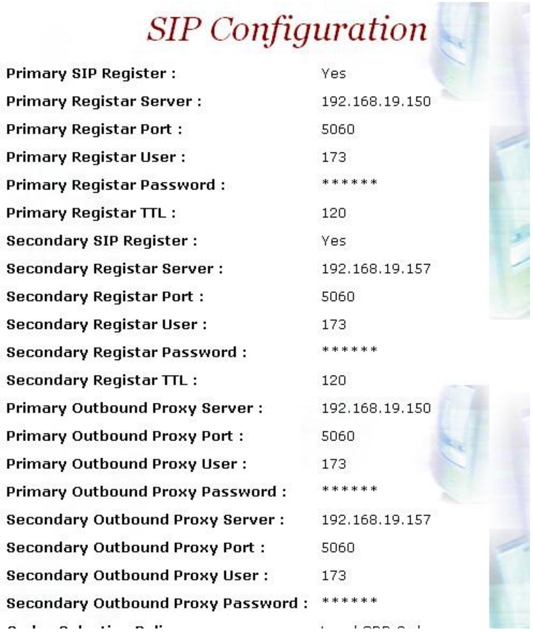


Figure 7.6-1

Basic Parameter Description:

- Primary SIP Register: Register to SIP proxy server or not
  - o Yes: Register to proxy server
  - o 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
  - o Yes: Register to proxy server
  - o 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

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 cancelInviteNoResponseTimer 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 caculate 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:
  - o Allow: the inputs IP Network are allowed for calls.
  - o 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

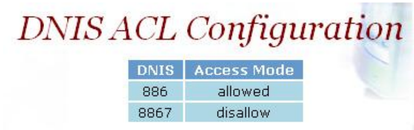
Parameters:

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

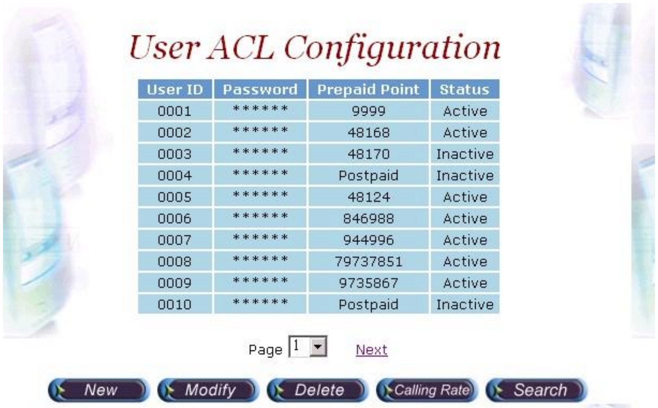
- DNIS: Called party number used for filter
- Access Mode:
  - o Allow: The called numbers are allowed for calls
  - o Disallow: The called numbers are disallowed for calls

☻***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

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.)
  - o Postpaid: postpaid user
- Status:
  - o Active: User is activeled
  - o Inactive: User is inactived

☻***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 Radiu 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.

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 nubmer

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 Rabin
2	Chan 2 Trunk 00	Round Rabin
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
  - o Random: Random select a trunk within this hunting group
  - o Priority: Select a trunk by priority. (Priority 1 has lowest priority; 9 has highest priority)
  - o 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 availity 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

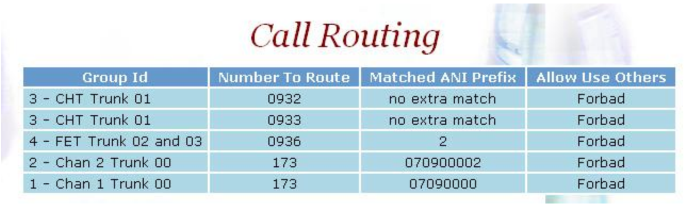
Route Plan channel mask: 0xffc00000

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


Call Routing

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	07090000	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

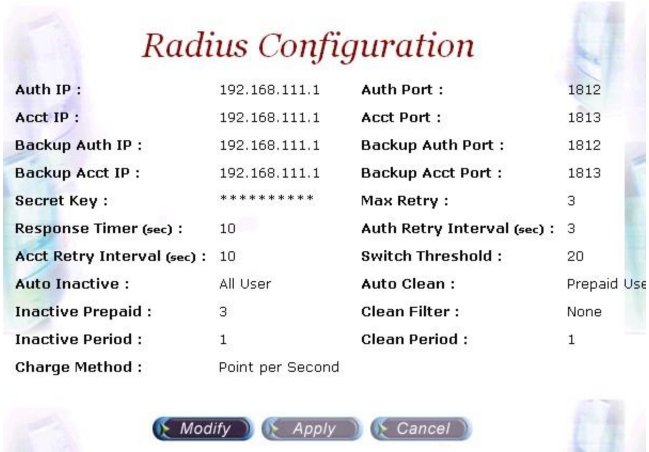


Figure 7.10-1

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 inactived
- Inactive Period: The max unaccess days for a postpaid user to be inactived
- Charge Method: Billing charge method selection
  - Point per Second: Point / calling rate = seconds
  - Second per Point: Point * calling rate / charge point = 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 inacived

Apply Change

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

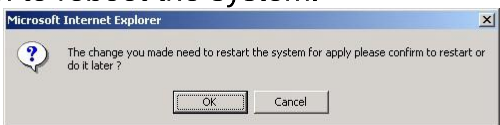


Figure 7.11-1

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

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

Stardand:

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:
  - o Standard: Use a predefined standard time zone **(Refer [Timezone to Country Mapping List](#))**
  - o Customize: Use a user defined time zone
- Auto Daylinght Saving: Auto adjust daylinght 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

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
  - o Month: 01 to 12
  - o Week Day: Sunday to Saturday
  - o 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)
  - o Hour: 00 to 23
- Standard Start: The date that a time zone enters daylight time
  - o Month: 01 to 12
  - o Week Day: Sunday to Saturday
  - o 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)
  - o 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.

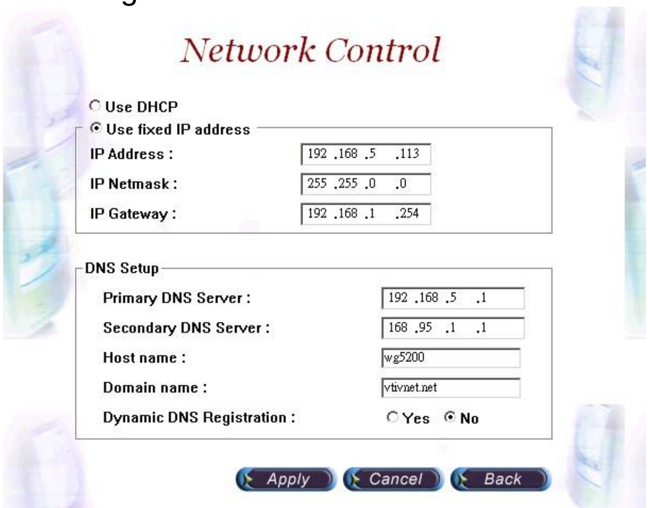


Figure 8.3-1

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**

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**

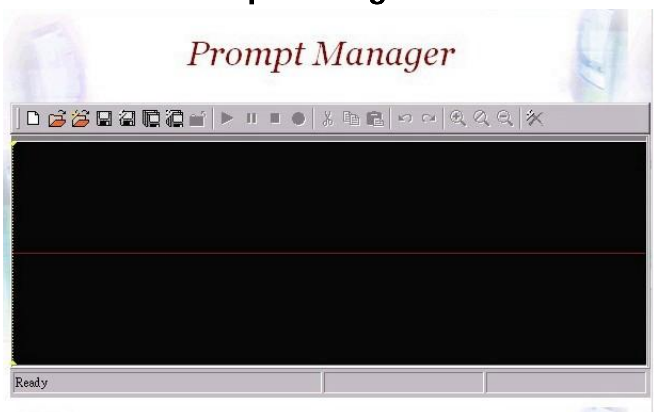
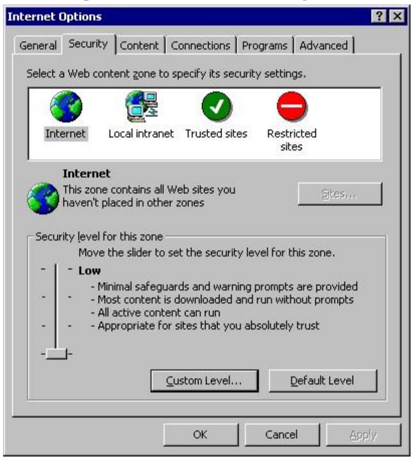


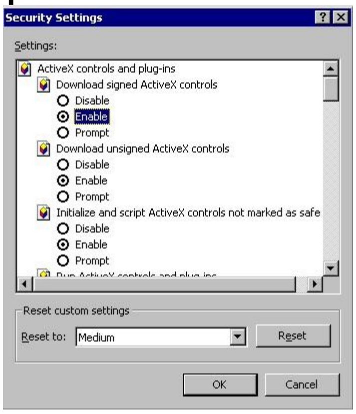
Figure 8.4-1

☻ **Note:**

1. **You mast has 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.

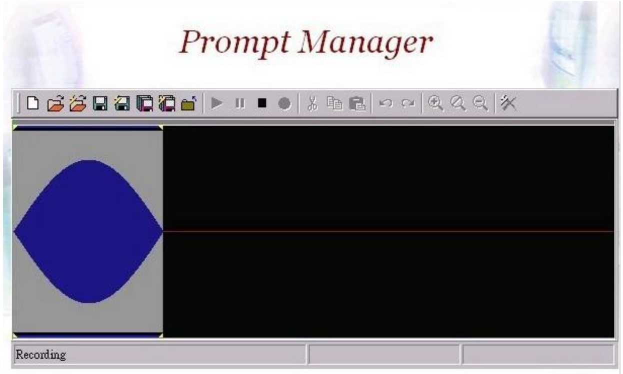


Figure 8.4-2

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.

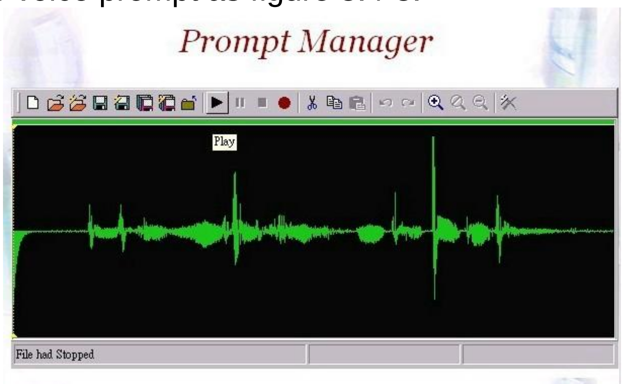


Figure 8.4-3

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.

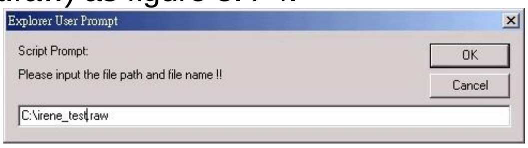


Figure 8.4-4

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

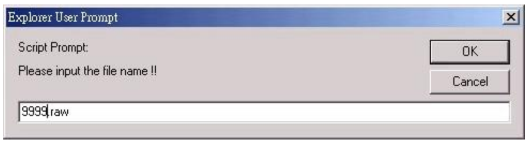


Figure 8.4-5

☻ ***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.

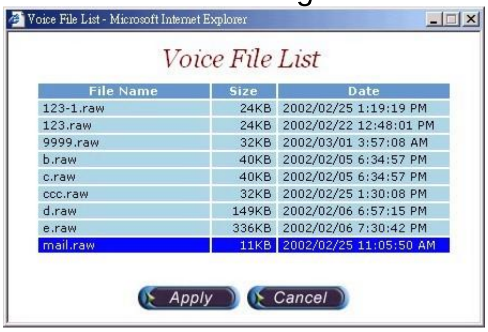


Figure 8.4-6

Open:

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

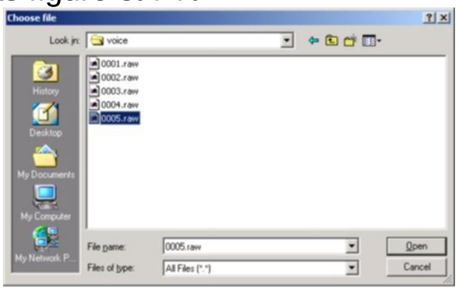


Figure 8.4-7

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.

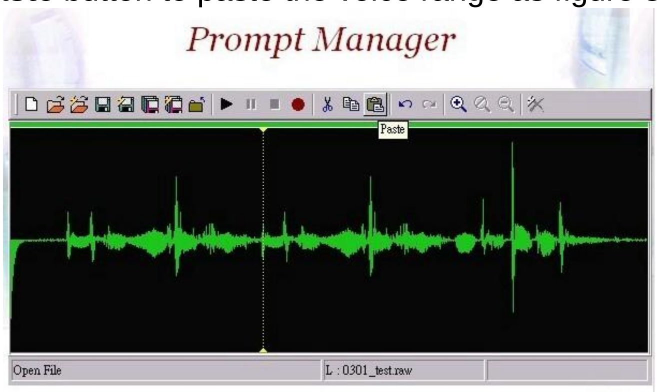


Figure 8.4-10

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.

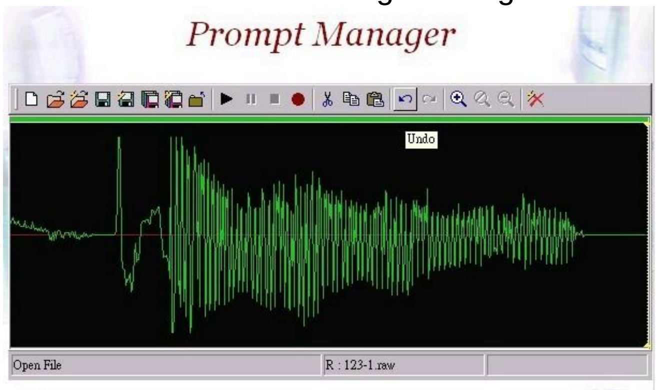


Figure 8.4-12

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.

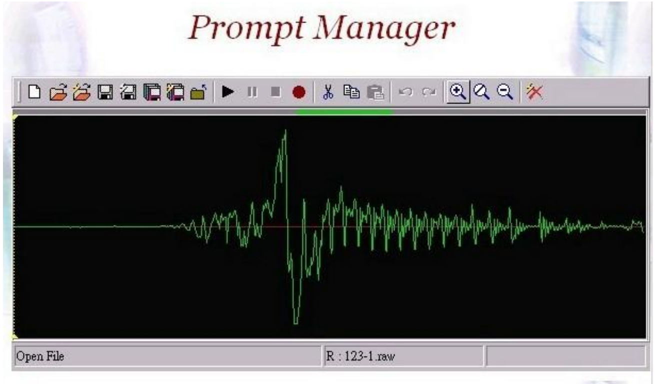


Figure 8.4-14

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.

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"

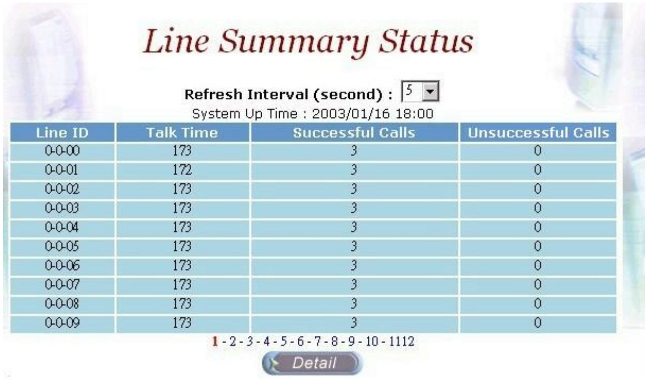
Chapter 9 System Monitor

It provides a way to monitor the system status.

Line Summary Status

Show channel summary status.

Start Path: Monitor→Line Summary Status


Line Summary Status

Refresh Interval (second) : 5

System Up Time : 2003/01/16 18:00

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

1 - 2 - 3 - 4 - 5 - 6 - 7 - 8 - 9 - 10 - 1112

Detail

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.

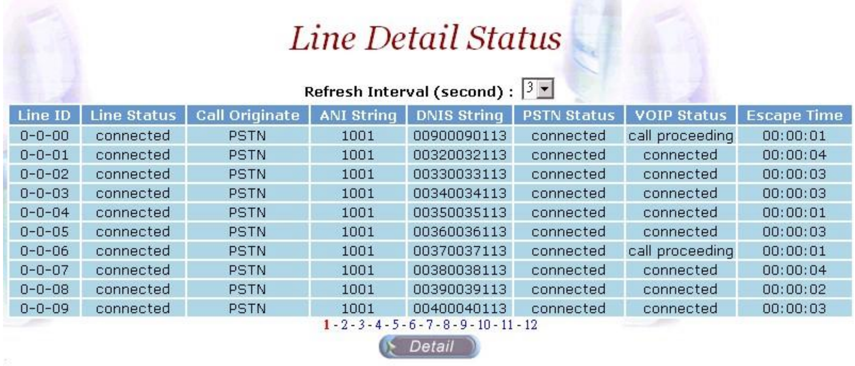
Figure 9.1-2

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

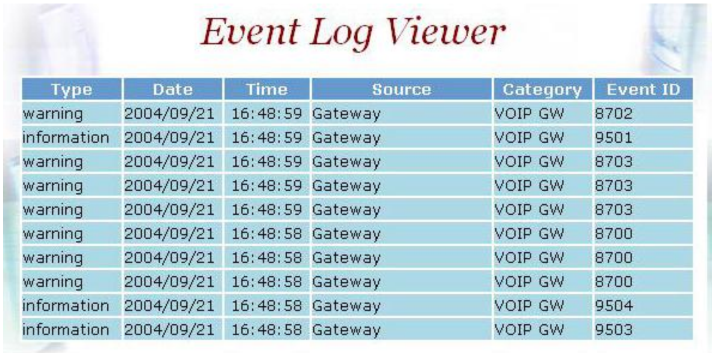
Field Description:

- 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
  - o Information
  - o Warring
  - o 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 Registratar Server
8700	VoIP Gateway application on the fly change	On the fly change (system change)
8703	[0]: evt: D CHANNEL STATUS: runkId=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	TelnSvr 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 Registratar 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"**

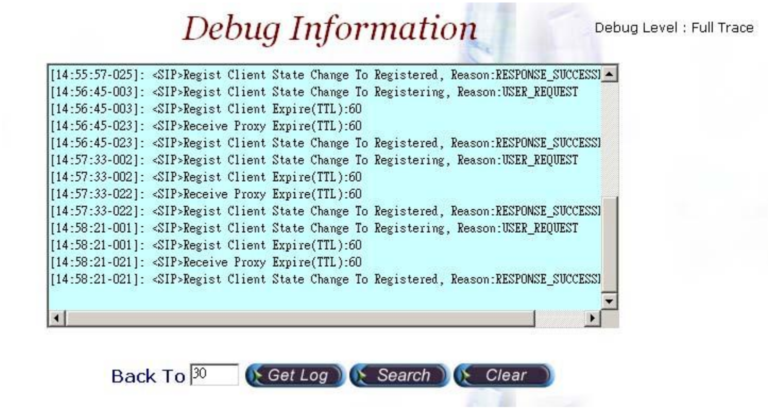


Figure 9.4-1

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 ralue)

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 –dchp	***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
[root#]ipconfig –ip 61.220.126 28 –mask 255.255.0.224 –gateway 61.220.126.1	***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 :
[root#]ipconfig –ip 61.220.126.115	***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 :
[root#]ipconfig –dns 210.59.126.53	***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
[root#] ping ?	Usage: ping IP. Example: ping 127.0.0.1
[root#]ping 61.220.126.1	***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
[root#] reboot ?	***Reboot System*** Are You Sure? (Y/N)
[root#]reboot Are You Sure?(Y/N)y	***VIP-2100 are rebooting***

Shutdown:

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

Reset:

Command	Purpose
[root#] reset ?	***Soft reset System*** Are You Sure? (Y/N)
[root#]reset Are You Sure?(Y/N)y	

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

	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
[root#]timezone	***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
[root#]timezone 40 n	***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
[root#]timezone -custom +08:00 -01:00 05-00-01-03 09-00-05-03	***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

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 **Ready |** **04-03-03 16:40**
Enter	Press Enter to select command **Event Log** **IP Config**
ESC	Quit the current command
▲	Up or previous edit mode
▼	Next or previous edit mode

Command Tree:
Main Menu

- Event Log
  - Show system log message
- IP Config
  - Show IP Info
  - Use DHCP
  - Use Fixed IP
- Reboot
  - Yes
  - No
- Reset PWD
  - Yes
  - No
- Soft Reset
  - Yes
  - No
- Shut Down
  - Yes
  - No

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

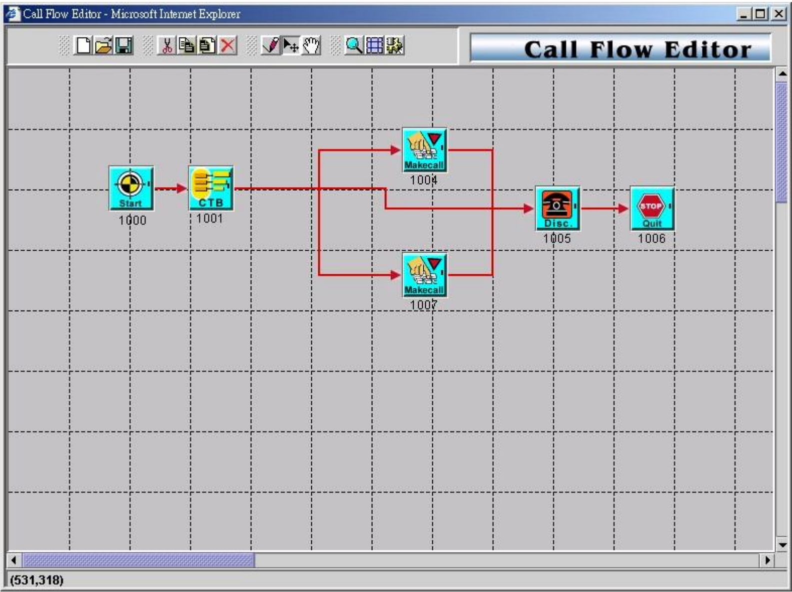
Enter	Reset or not
ESC	Quit Reset configure
ESC	Quit to main menu

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.323Gatekeeper call

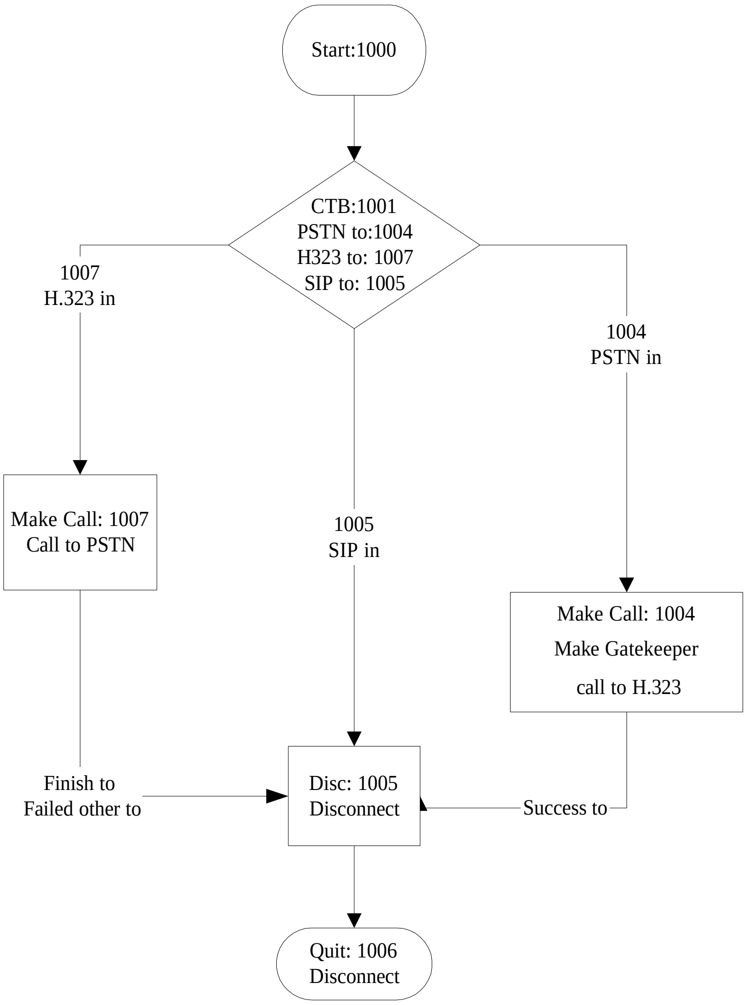
MakeCall Component ID: 1007	Route Mode: **PSTN** Finish To: 1005 Failed Other To: 1005
Disc Component ID: 1005	Reason: PSTN normal call clear Next Component: 1006
Quit	

Component: 1006	

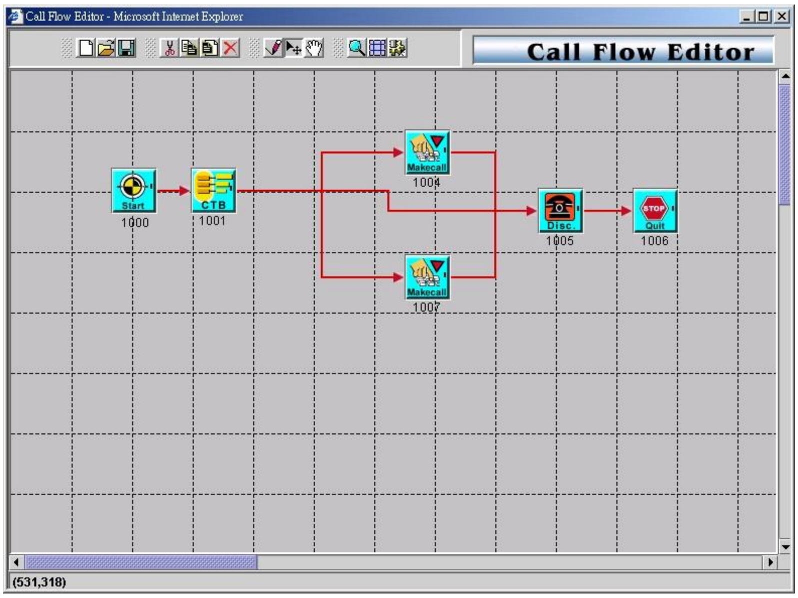
1004 Route for PSTN call

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

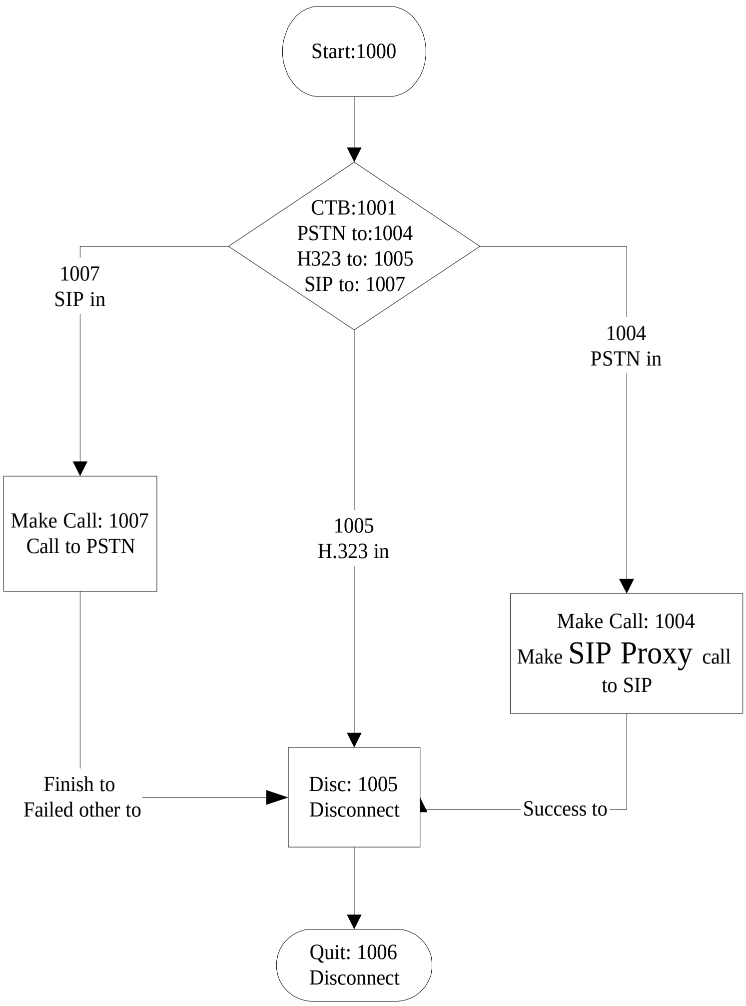
1007 Route for SIP Proxy call

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	

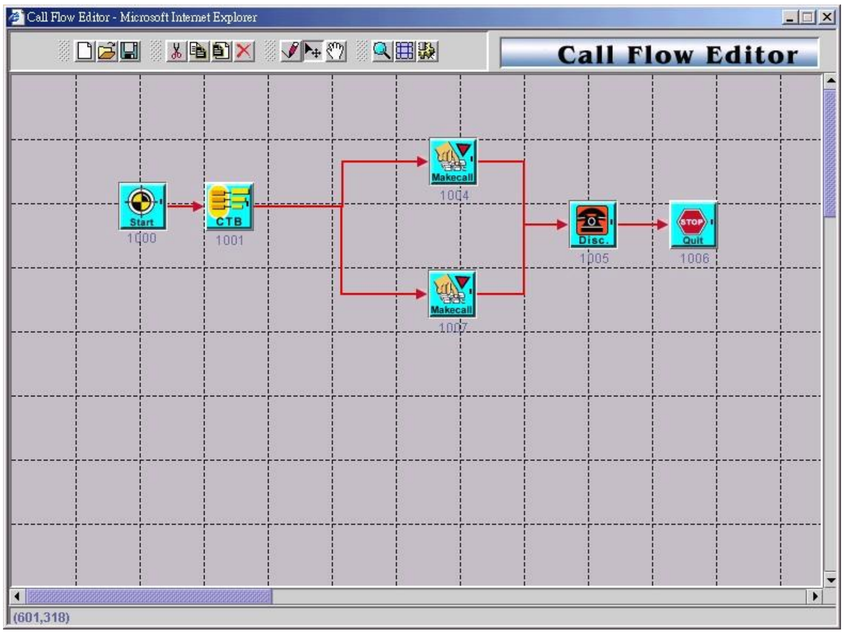
1004 Route for PSTN call

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

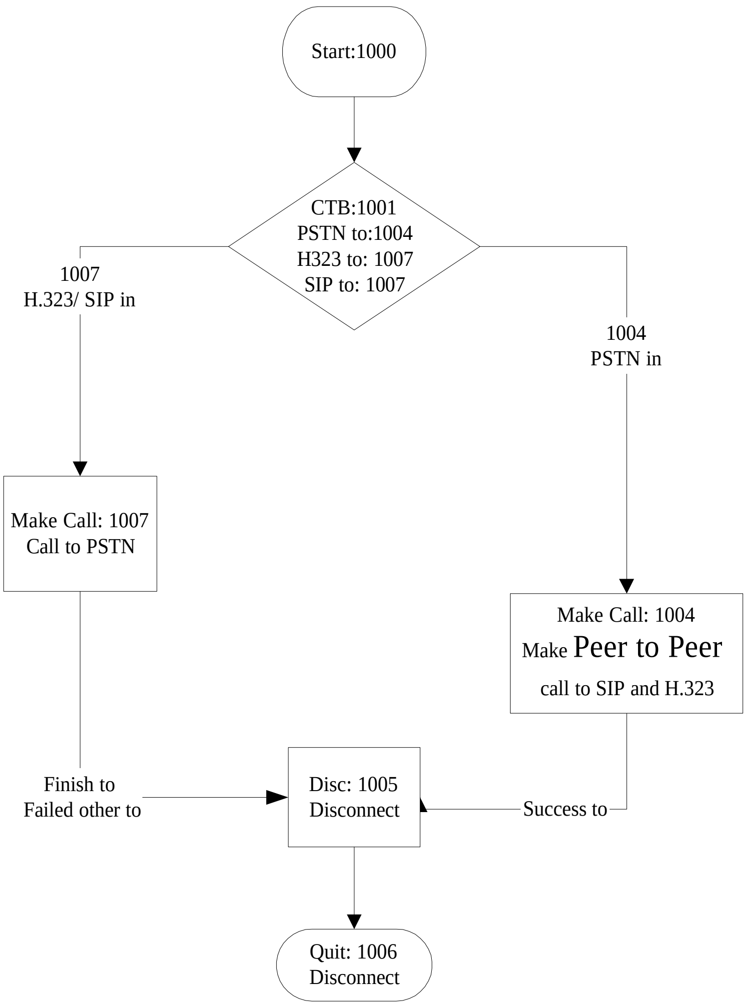
1007 Route for SIP Proxy or H.323 Gatekeeper call

 **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	

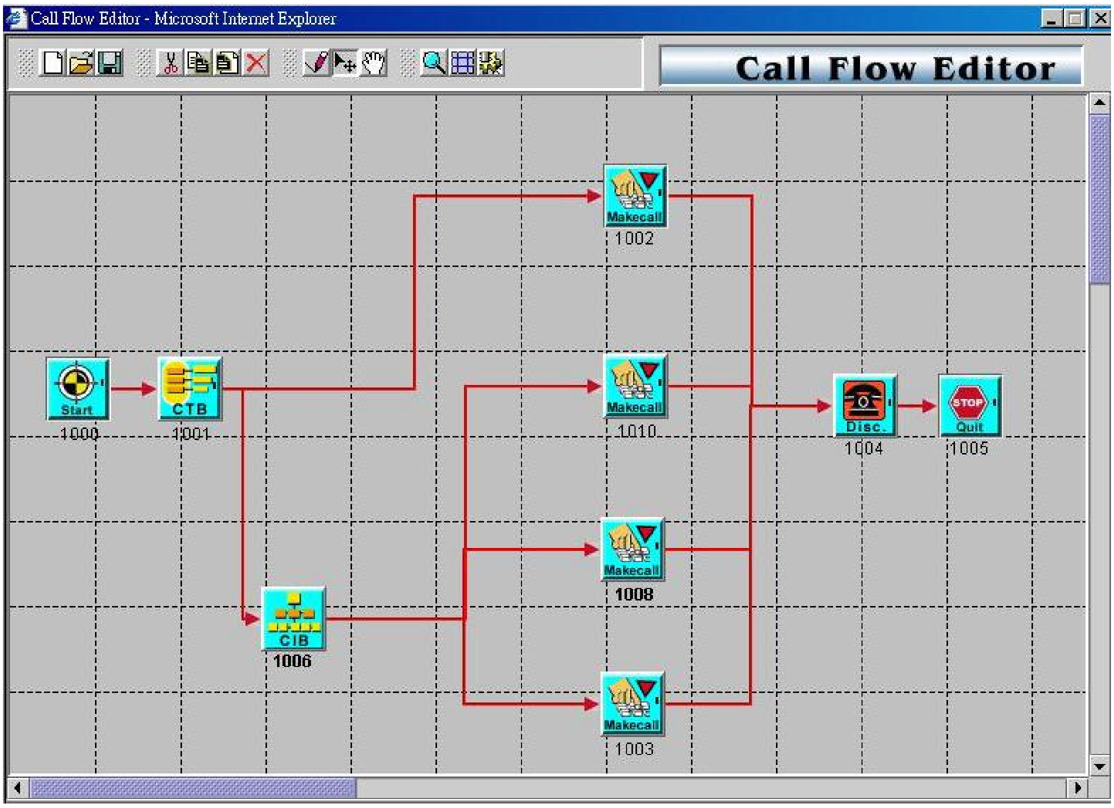
1004 Route for PSTN call

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

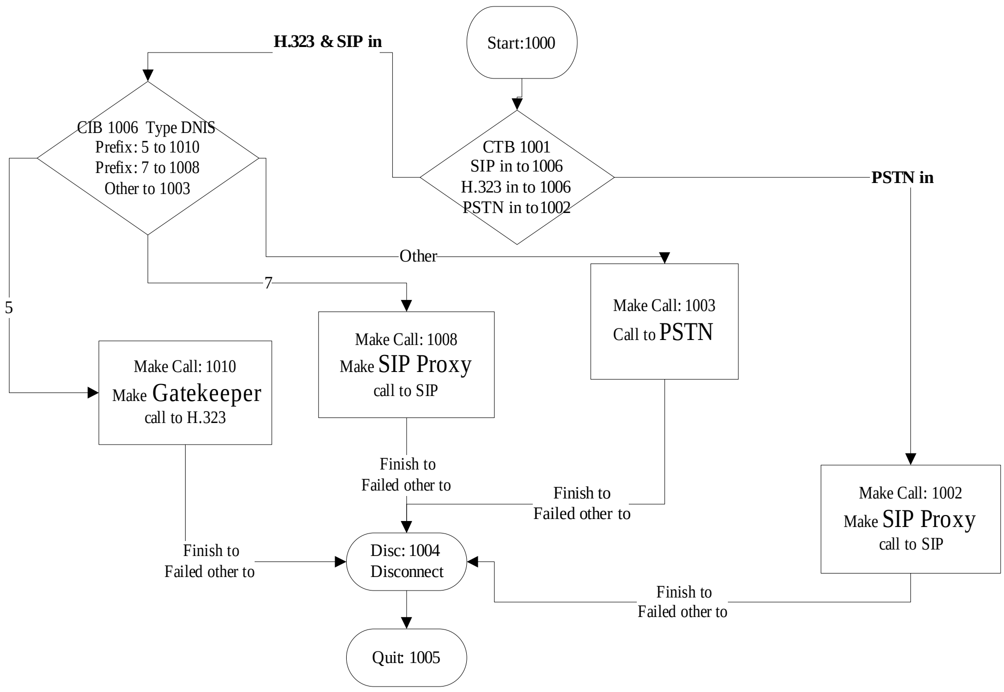
 **CIB** Component ID: 1006	Info Type: DNIS Prefix: 5 goto: 1010 Prefix: 7 goto: 1008 Other goto: 1003
 **MakeCall** Component ID: 1010	Route Mode: **Gatekeeper Call** Finish To: 1004 Failed Other To: 1004

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	

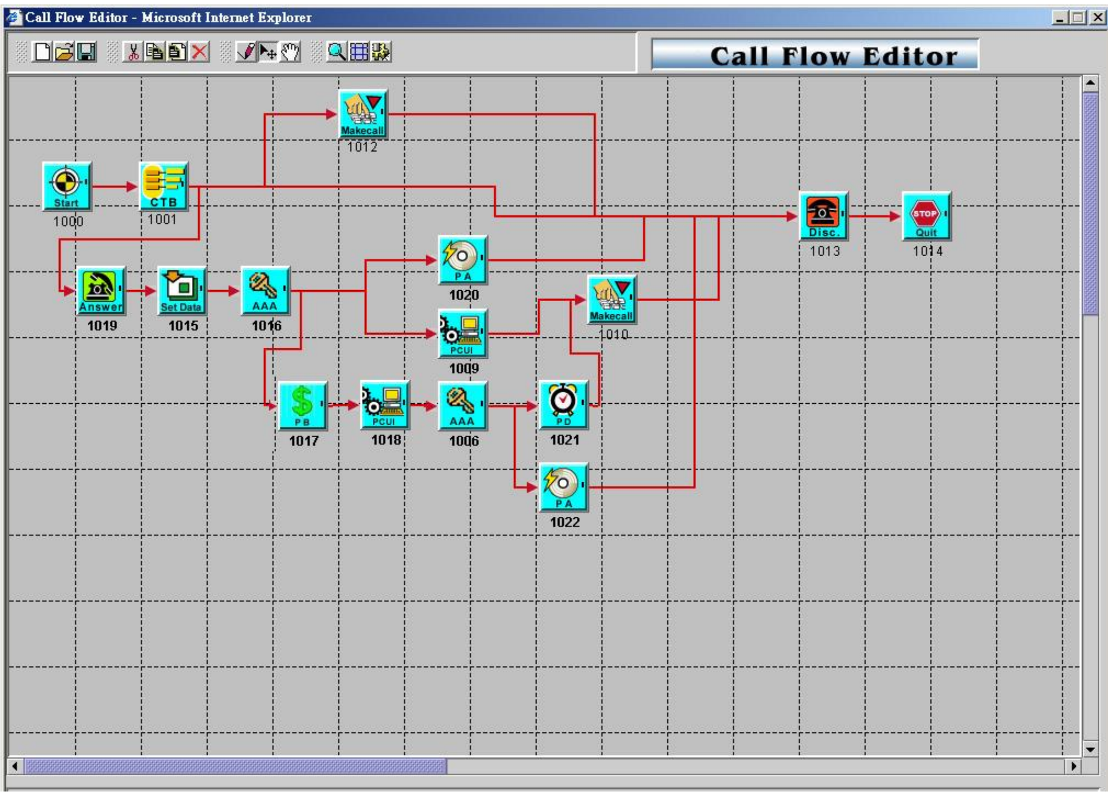
1001 Route for PSTN call

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	

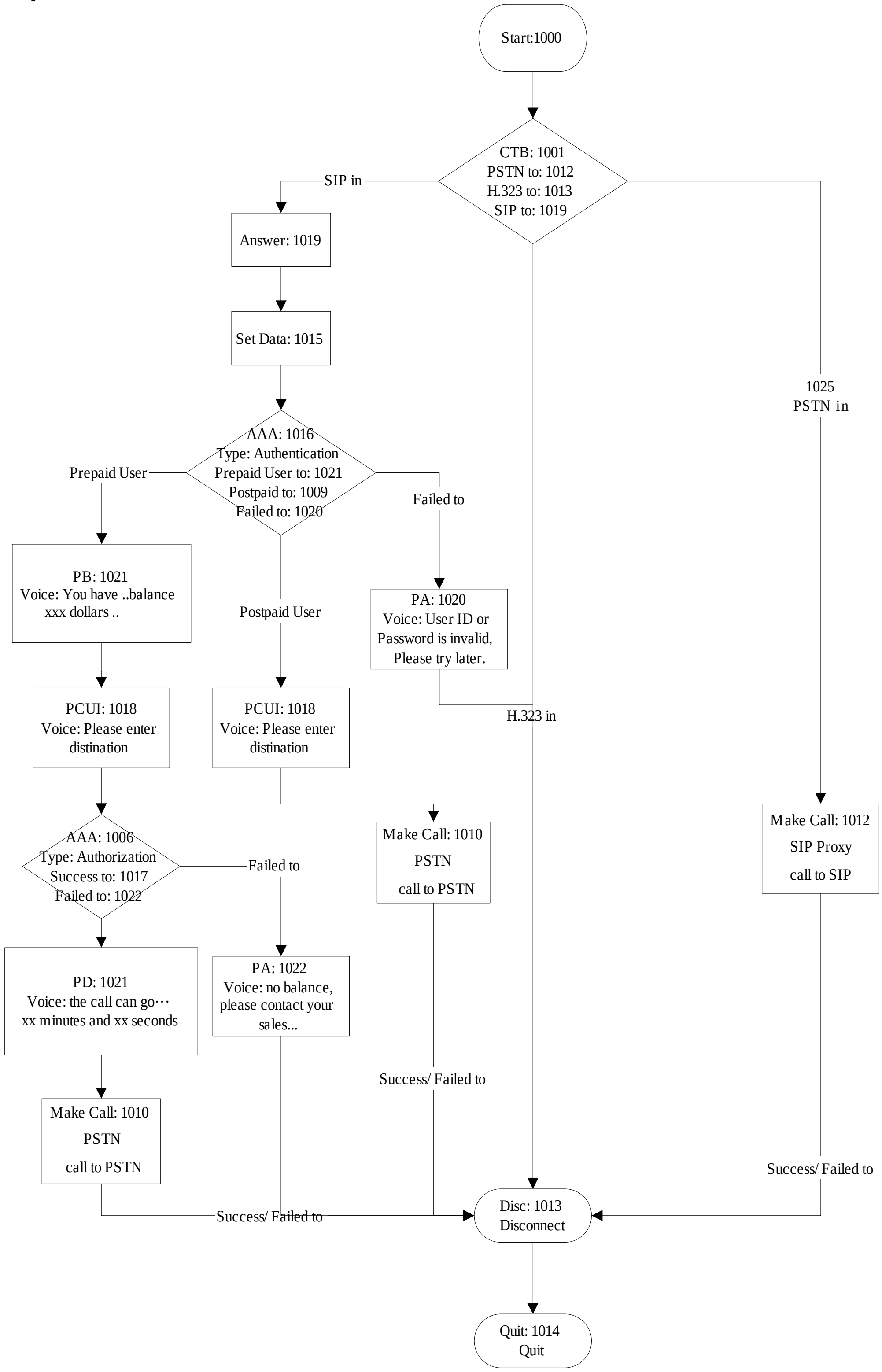
1012 Route for H.323 call	
Disc Component ID: 1013	Next Component: 1014
Quit Component ID: 1014	

1012 Route for SIP call	
Anser 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: - 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	

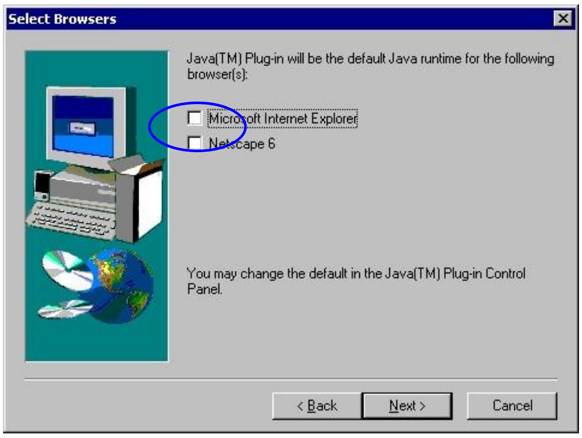
Example Used Call Flow:



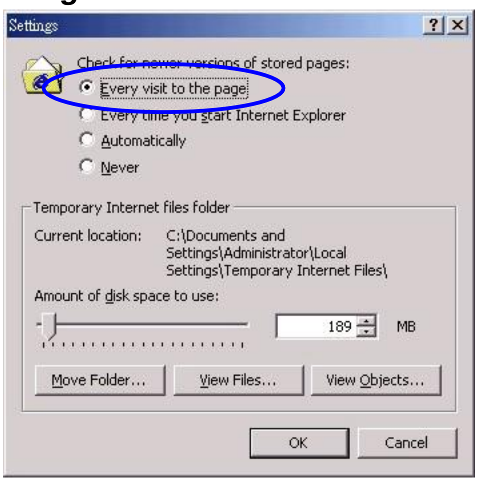
Appendix 2 Java plug-in Installation

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 planet\cdr
250 CWD command successful. "D:/planet/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: STRTyyyymmdd.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

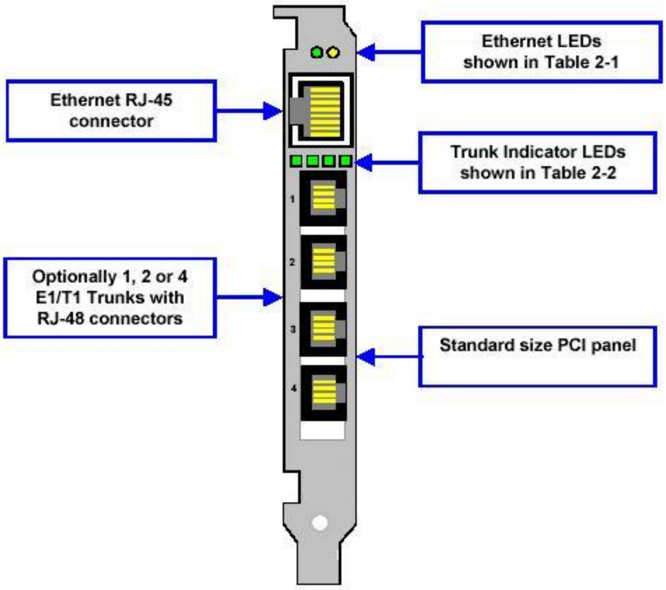
Billing Stop CDR:

- **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:



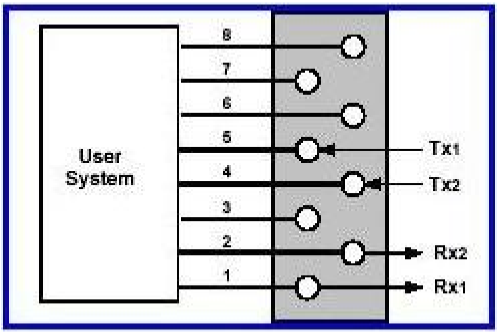
Ethernet LED:

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 Tim
(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
invaliRevision	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=192.168.19.150
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	Sting	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:

p Register to Gatekeeper
GK IP Address: _____. _____. _____. _____
GK RAS Port: ______

p Peer To Peer
Refer to User Guide-Phone Book setting

H.245 tunneling: p Enable p Disable
Fast Connect: p Enable Fast Start p Early H.245 p Disable
Separate H.245 after Fast Start: p Yes p No
Fast Start Enabled Codec:

p G.711 a-law
p G.711 u-law
p G.729
p G.729 A/B
p G.723.1 (5.3K)
p 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: p Peer To Peer

Refer to User Guide-Phone Book setting

p Register to SIP Proxy Server
Registar Proxy Server: _____. _____. _____. _____
Registar Proxy Port: ______
Registar User ID: _________
Registar Password: _________
Outbound Proxy Server: _____. _____. _____. _____
Outbound Proxy Port: ______
Outbound User: ________
Outbound Port: ________

180 SDP: p Yes p No
183 SDP: p Yes p No
Local Codec Codec:

- p G.711 a-law
- p G.711 u-law
- p G.729
- p None
- p G.723.1 (5.3K)
- p G.723.1 (6.3A)

Accept Proxy Call Only: p Yes p No

PSTN Interface:

PCM encoding: p A-law p Mu-law
PCM Idle Pattern:
Default (-1): p 0x55 A-law p 0xff u-law p specified: ________
Clock Source: p External p Internal

p E1
Framing Method:

- p Automatic CRC4 or Double Frame selection
- p Double Frame Format
- p CRC4 multi-frame
- p CRC4 extend multi-frame

Line Code: p HDB3 p AMI
ISDN/PRI:
Termination Site: p Network p User site
Variance:

- p Euro ISDN
- p Australia ISDN
- p Hong Kong ISDN
- p Korea ISDN

CAS:
CAS Idle ABCD signal: Default (-1): specified: ________
p E1 MFC R2
p E1 CAS R2
Variance:

- p E1 R2 MF Aregntina ANI
- p E1 R2 MF Aregntina ANI 7digits
- p E1 R2 MF Aregntina no ANI
- p E1 R2 MF Aregntina no ANI 7 digits
- p E1 R2 MF Bolivia ANI
- p E1 R2 MF Bolivia ANI 7digits
- p E1 R2 MF Bolivia no ANI
- p E1 R2 MF Bolivia no ANI 7 digits
- p E1 R2 MF Brazil ANI
- p E1 R2 MF Brazil ANI 7digits
- p E1 R2 MF Brazil no ANI
- p E1 R2 MF Brazil no ANI 7 digits
- p E1 R2 MF Chile ANI
- p E1 R2 MF Chile ANI 7digits
- p E1 R2 MF Chile no ANI
- p E1 R2 MF Chile no ANI 7 digits
- p E1 R2 MF China ANI

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

p 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: p AMI p B8ZS

ISDN/PRI:

Termination Site: p Network p User site

Variance:

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

CAS:

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

p 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 FGAImmediate
- 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 registar 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 weather 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.