VIP-450 SIP command Reference Guide

The VIP-450 is a SIP version 2.0 compliant voice gateway, which integrates a web-based graphical user interface, and command line administration that can cover most configurations and application demands.

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ABOUT THIS GUIDE

This VIP-450 SIP (Session Initiation Protocol) Reference Guide provides SIP configuration commands. Before starting configurations, it is suggested to check the SIP connection requirement carefully, then configure the required value in machine. If you're not sure which parameters should be inserted, please check this guide for corresponding parameters to establish the voice connection on your side. In the meantime, more VOIP concept and machine design algorithm can be referred in the user's manual CD for more understandings.

This Guide contains the following information:

SIP Command Reference: This chapter illustrates how to use the sip command for the Internet Telephony Gateway (ITG), to register with the SIP proxy server.

Dial Plan Set Up: Shows the difference between the H.323 and SIP protocol.

Call Transfer: This chapter illustrates the operations of call transfer.

Configuration Example: This chapter shows you how to set up the ITG to start up your call.

Before reading this guide, it is best that you are familiar with the **"QUICK INSTALLATION GUIDE"** and you know the network configurations, as well as the dial plan set up.

1. SIP COMMAND REFERENCE

show sip reg

- displays the information of registration

show sip cfg

- displays the configuration of registration

show sip proxy

- displays the configuration of proxy server

show sip dns_ip

- displays the configuration of DNS server

show sip info_sw

- displays the information switch

show sip auto_reg

- displays auto-registration on/off when reboot

show sip nat_call

- displays nat_call switch

show sip outboundproxy

- displays the outbound proxy

set sip reg add

- used to add registration information

set sip reg add [reg_num] [expires] [#registrar] [ip] [port] {name **|** 0} {passwd **|** 0} {ip port …}

Syntax description

reg_num	the phone number to be registered
expires	how long the reg_num is to re-register
#registrar	the number of registrar
ip	IP of the registrar
port	port of the registrar
name	[the name for authentication | 0 for no authentication]
passwd	[the password for authentication | 0 for no authentication]

set sip reg del

- used to delete registration information

set sip reg del [reg_num]

Syntax description

reg_num	the phone number to be deleted

set sip cfg

- used to set the configuration of sip

set sip cfg [udp_ctl_port] [rtp_data_port]

Syntax description

udp_ctl_port	<udp control port>**|**(0 : default port will be used)
rtp_data_port	<rtp data port>**|**(0 : default port will be used)

set sip proxy

- used to set the configuration of sip

set sip proxy [#proxy_svr] [proxy svr-1 nwa] { proxy svr-2 nwa }…

Syntax description

#proxy_svr	number of proxy server **|** 0 to remove existing server
proxy svr nwa	<ip> <port>

set sip actreg

- used to send out the registration

set sip actreg

set sip dns_ip

- used to set the domain name server. The IP domain should be in numerical format

set sip dns_ip [ip]

Syntax description

ip <x.x.x.x> | 0 to remove dns ip

set sip info_sw

- used to set the sip process information switch

set sip info_sw [on/off]

set sip auto_reg

- used to set registration auto start when reboot in the condition of registration server is set

set sip auto_reg [on/off]

set sip nat_call

- used to set nat_call switch on/off whether the gateway in public IP is to receive calls from the private IP or not. The default is on

set sip nat_call [on/off]

set sip outboundproxy add

- used to add outbound proxy

set sip outboundproxy add [ip/hostname] [port]

set sip outboundproxy del

- used to delete outbound proxy

set sip outboundproxy del

2. DIAL PLAN SETUP

Here, we list the differences between the H.323 and SIP dialplan configurations. If you are not familiar with machine dialplan setup, please refer to the VIP Quick Installation Guide and the User's manual for more understandings of the dialplan operations in machine.

Calling destination in PLANET H.323 voice gateways can be added via following commands:

atpm dadd [dest_id] **h323** [dest ip addr] [port]

atpm dadd [dest_id] **dns** [dest host name] [port]

In VIP-450 the "**h323**" part in the command is replaced by "**sip**":

atpm dadd [dest_id] **sip** [dest ip addr] [port]

atpm dadd [dest_id] **dns** [dest host name] [port]

3. CALL TRANSFER

VIP-450 supports unattended call transfer.

3-1 The Operations

When a remote session is established, both ends of the parties can transfer a call to another location by pressing the Flash/Transfer keypad on the phone set.

3-2 Operation Recovery

When the user is transferring a call to a destination that does not exist on the dial plan, an out-of-service tone will be generated. Pressing the Flash/Transfer keypad can bring users back to the original session.

4. CONFIGURATION EXAMPLE

Now, it is assumed that you are registering to a proxy server (referred to registrar). The information for registration is as follows:

ITG IP: 211.20.1.5

Registered ID: 2001

Expired Time: 500

Proxy Server IP: 211.20.1.1

Proxy Server Port: 5060

Set the information of registration.

```
ITG>set sip reg 2001 500 1 211.20.1.5 5060 0 0
```

Then you need set up the server.

```
ITG>set sip cfg 0 0
```

Then you need set up the server.

```
ITG>set sip proxy 1 211.20.1.1 5060
```

In order to support the complete function, you need to set a port cid number. We assume port 0 is used.

```
ITG>set port 0 cid number 2001
```

Next, you need to save the information and send out the registration.

```
ITG>config activate
ITG>config store
ITG>set sip actreg
```

***Note:** Before starting the first call, you need to set up the dial plan. Don't forget to use "**sip"** when you type "atpm dadd".