

SIP
Quick Installation Guide

TABLE OF CONTENTS

1. ABOUT THIS GUIDE.....	3
2. SIP COMMAND REFERENCE.....	3
3. DIAL PLAN SETUP.....	5
4. CALL TRANSFER.....	5
5. CONFIGURATION EXAMPLE.....	6

ABOUT THIS GUIDE

This SIP (Session Initiation Protocol) Reference Guide provides the user, who uses the Elite GW with the SIP protocol, the related configuration commands for SIP. Please refer to the Quick Installation Guide for other basic setting commands.

This Guide contains the following information:

SIP Command Reference: This chapter illustrates how to use the sip command for the Internet Telephony Gateway (ITG), registered to 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 reg

show sip cfg

- displays the configuration of registration

show sip cfg

show sip proxy

- displays the configuration of proxy

show sip proxy

show sip dns_ip

- displays the configuration of proxy

show sip dns_ip

show sip info_sw

- displays the information switch

show sip info_sw

show sip auto_reg

- displays auto-registration on/off when reboot

show sip auto_reg

show sip nat_call

- displays nat_call switch

show sip nat_call

show sip outboundproxy

- displays the outbound proxy

show sip outboundproxy

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 passwd 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 SET UP

Here, we list the differences between the H.323 and SIP setups. If you don't know how to setup the dial plan, you should read "QUICK INSTALLATION GUIDE" first.

For H.323 protocol, we use

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

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

For SIP protocol, we use

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

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

3. CALL TRANSFER

Support unattended Transfer.

4-1 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.

4-2 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.

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

Then you need set up the server.

```
Console>set sip cfg 0 0
```

Then you need set up the server.

```
Console>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.

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

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

```
Console>config activate
```

```
Console>config store
```

```
Console>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”**.