

MTG Troubleshooting

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Foreword

- This course is mainly:
 - Guide Ringback Tone Problems & Solutions
 - Guide Problems & Solutions for Calling
 - Guide Problems & Solutions for Login

Course Objective

Through this course
you will be able to



How To Solve The Problem Of No Ringing Tone



How To Solve The Problem Of Not Being Able To
Call



How To Solve Login Failure

Contents

- 1 Chapter One Ringback Tone Problems & Solutions
- 2 Chapter Two Problems & Solutions for Calling
- 3 Chapter Three Problems & Solutions for Login

Chapter One

Ringback Tone Problems & Solutions 01

Ringback Tone Problems & Solutions

- View Configuration

IP Grouping Config - IP Profile View Ringback tone Source and Early Media Configuration

Ringback Tone to PSTN Originated from:

Local-- MTG playback

IP--SIP server side playback

Adaptive-- determine the playback side based on negotiation

Ringback Tone to IP Originated from:

Local-- MTG playback

PSTN-- PSTN side playback

IP Profile Modify

IP Profile ID	0
Description	Default
Declare RFC2833 in SDP	Yes
Support Early Media	Yes
Ringback Tone to PSTN Originated from	Adaptive
Ringback Tone to IP Originated from	PSTN
Wait for RTP Packet from Peer	No
T.30 Expanded Type in SDP	X-Fax

OK Reset Cancel

Ringback Tone Problems & Solutions

- View Log

1. Verify whether the device has received a ringback signal from the E1 line
2. whether the device has played a ringback tone or transmitted a ringback tone

```
: <924,Sip-t,0,31786201,in_proc> <== SIP_CALL_RING, Local:018748807233@ims.sh.chinamobile.com, Peer:+862131786201@ims.sh.chinamobile.com
: <924,Sip-t,0,31786201,in_proc> called dev no:0, called term type:4, call type:2, IpProfileId:0, pem:none.
: <924,Sip-t,0,31786201,in_proc> Is need send local ringback tone to caller(ip->ip):yes, call type:2(ims)
: <924,Sip-t,0,31786201,in_proc> st_set_conn_param, local:255__, remote:255__
: <924,Sip-t,0,31786201,in_proc> transcode crcx_called, connid:33, chan:97, localIp:20.20.75.46, peerIp:20.20.75.46, port:6339, AlgoType
: <924,Sip-t,0,31786201,in_proc> transcode crcx_calling, connid:33, chan:33, localIp:10.10.60.122, peerIp:10.10.60.103, port:17900, Algo
: <924,Sip-t,0,31786201,in_proc> transcode_crcx success! CCBNo:924, connectId:33
: <924,Sip-t,0,31786201,in_proc> sst send ringback tone to net(5)!
: <924,Sip-t,0,31786201,in_proc> ==>> CC_ST_ALERTING, ccb:924, calling:31786201
: <924,Sip-t,0,31786201,in_proc> ==>> CC_ST_ALERTING, std sdp:, priv sdp:
: <924,Sip-t,7,65535,,wait ack> <== CC_ST_ALERTING, std sdp:, priv sdp:
: <924,Sip-t,7,65535,,wait ack> get bill alter time:15-26-46
: <924,Sip-t,7,65535,,wait ack> cc (ccb:924) alerting crypt cfg 4 Method:0,Type:0,Key:0,uIP:255.255.255.255,usPort:65535
: <924,Sip-t,7,65535,,wait ack> ==>> CC_ST_ALERTING
: <924,Sip-t,7,65535,,wait ack> ==>> CC_ST_ALERTING, std sdp:v=0
```

Ringback Tone Problems & Solutions

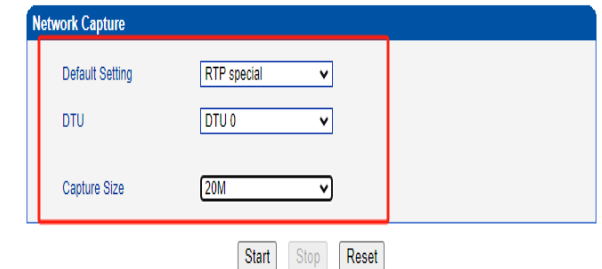
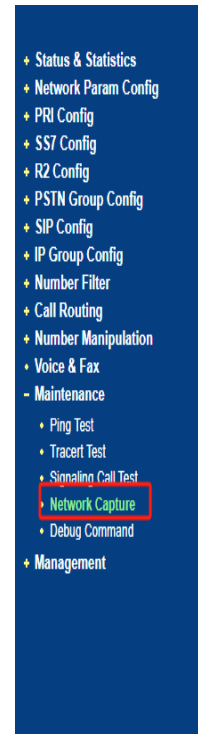
- RTP Capture

Retrieve RTP packets between SIP server and MTG

Call direction: IP-PSTN, capture to check if a ringback tone is sent to the SIP server

Call direction: PSTN-IP, capture to check if SIP server's ringback is received

Call direction IP-IP, capture packets to check for transparent transmission or local ringback tone



NOTE: All the items can be left it empty, it means get all the packets on the available interfaces.
Use ',' division multiple IP
If you want get the syslog packets, please make sure syslog is enabled.
If you want get RTP or RTCP packets, please make sure select UDP concurrently.
If you want get RTP or RTCP packets, please pick a DTU.
If Current Call larger than 15 and get RTP or RTCP packets, may cause SIP Trunk fault.

Ringback Tone Problems & Solutions

- PCM Capture

Obtain PCM Capture between MTG and E1 lines

Call direction: IP-PSTN, capture packets to check if line ringback is received

Call direction: PSTN-IP, capture packets to check if a ringback tone is sent to the line

Network Capture

Default Setting: PCM only

E1: E1 0

Ts: Ts all

Capture Size: 20M

Start Stop Reset

NOTE: All the items can be left empty, it means get all the packets on the available interfaces.
Use ',' division multiple IP
If you want get the syslog packets, please make sure syslog is enabled.
If you want get RTP or RTCP packets, please make sure select UDP concurrently.
If you want get RTP or RTCP packets, please pick a DTU.
If Current Call larger than 15 and get RTP or RTCP packets, may cause SIP Trunk fault.

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Chapter Two

Problems & Solutions for Calling

02

Failed to register IMS account

• View Configuration

1. **SIP Trunk:** Check if the configuration of the remote address, outbound proxy address, from header, outgoing call mode, and account auth mode is correct
2. **SIP account:** check if the username, authentication ID, and password are correct, with expire time of 1800s or 3600s
3. **Ping& Tracert:** check if the outbound Proxy IP can be pinged and static IP routing is correct

The image shows two screenshots of the Dinstar web interface. The top screenshot displays the 'SIP Trunk Add' configuration page. The left sidebar shows a navigation menu with 'SIP Trunk' highlighted. The main content area contains various configuration fields, many of which are highlighted with red boxes: 'Remote Address' (ims.gx.chinamobile.com), 'Remote Port(UDP)' (5060), 'Outbound Proxy' (183.211.6.89), 'Outbound Proxy Protocol Type' (UDP), 'Outbound Proxy Port(UDP)' (5060), 'Outbound Proxy Port(TCP/TLS)' (5060), 'From Header', 'Register to Remote', 'Outgoing Call Mode', and 'Account Auth Mode'. The bottom screenshot displays the 'SIP Account Modify' configuration page. The left sidebar shows 'SIP Account' highlighted. The main content area contains fields for 'SIP Account ID' (0), 'Description' (+867713480000), 'Binding PSTN Group' (None), 'SIP Trunk No.' (1), 'Username' (+867713480000), 'Authenticate ID' (+867713480000@ims.gx.chinamobile), 'Password' (*****), 'Confirm Password' (*****), 'Expire Time' (1800s), 'Max Calls' (65535), and 'Enable Account' (Yes). At the bottom of the page are 'OK', 'Reset', and 'Cancel' buttons.

SIP Trunk Add

Trunk No.	1
BI	GE1
Trunk Name	ims
Remote Address	ims.gx.chinamobile.com
Protocol Type	UDP
Remote Port(UDP)	5060
Remote Port(TCP/TLS)	5060
Outbound Proxy	183.211.6.89
Outbound Proxy Protocol Type	UDP
Outbound Proxy Port(UDP)	5060
Outbound Proxy Port(TCP/TLS)	5060
From Header	Peer Domain
PPID	Disable
Local Domain	Disable
Support SIP-T	Disable
Get Callee from	Request-line
Get Caller from	User Name
Register to Remote	Yes
Outgoing Call Mode	Access
Redundancy Mode	No
Account Select Mode	Default
Account Auth Mode	Auth ID
Display Name by	Caller
Contact Username by	Default
Incoming SIP Authentication Type	IP Address
Rport	Disable
Dynamic Nat	Disable
Static Nat	Disable
Outgoing Calls Restriction	No
Incoming Calls Restriction	No
Incoming Time Restriction	Disable
Heartbeat Bound	Disable
Detect Trunk Status	No
Heartbeat Username	heartbeat
Enable SIP Trunk	Yes
Early Alerting	Disable
No Prack for Incoming Call	Disable
User to User(callee/caller)	Disable
Request Add Port	Disable
OPTION Only Detects 200OK	Disable

SIP Account Modify

SIP Account ID	0
Description	+867713480000
Binding PSTN Group	None
SIP Trunk No.	1 <ims>
Username	+867713480000
Authenticate ID	+867713480000@ims.gx.chinamobile
Password	*****
Confirm Password	*****
Expire Time	1800s
Max Calls	65535
Enable Account	Yes

OK Reset Cancel

Failed to register IMS account

- Packet capture

1. Click on **maintenance - network capture**
2. Default configuration, click start
3. Reproduce the problem, the expire time is relatively long, you can disable the account first and then enable it again
4. Click to stop

- Status & Statistics
- Network Param Config
- PRI Config
- SS7 Config
- PSTN Group Config
- SIP Config
- IP Group Config
- Number Filter
- Call Routing
- Number Manipulation
- Voice & Fax
- Maintenance
 - Ping Test
 - Tracert Test
 - Signaling Call Test
 - **Network Capture**
 - Debug Command
 - SelfCheck MCU
 - SelfCheck DTU
 - Authorization Config
- Management

Network Capture

Default Setting: Custom

Network Interface: ☐ GE1 ☐ GE0

Source Host:

Destination Host:

Protocol(s): ☐ TCP ☐ UDP ☐ RTP ☐ RTCP ☐ ICMP ☐ ARP

DTU: DTU None

Capture Size: 4M

Start Stop Reset

NOTE: All the items can be left it empty, it means get all the packets on the available interfaces.

Use '*' division multiple IP

If you want get the syslog packets, please make sure syslog is enabled.

If you want get RTP or RTCP packets, please make sure select UDP concurrently.

If you want get RTP or RTCP packets, please pick a DTU.

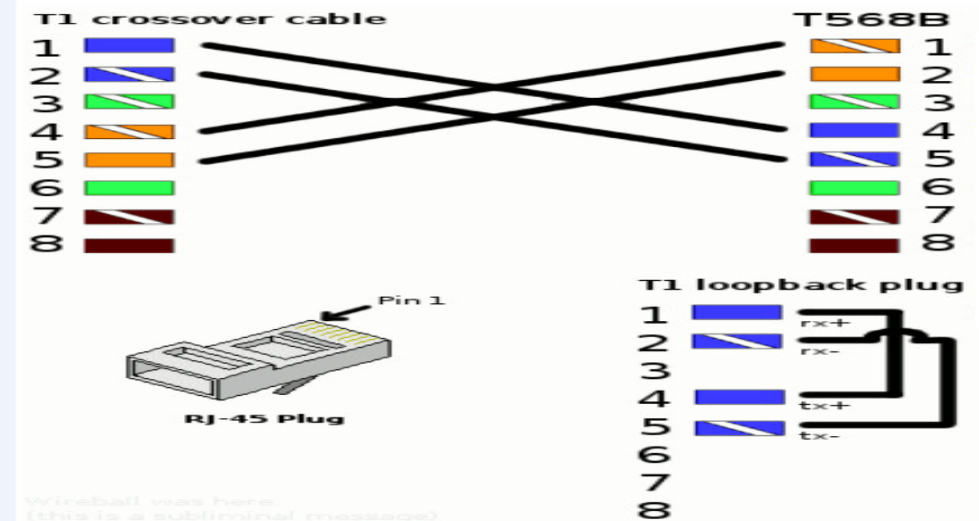
If Current Call larger than 15 and get RTP or RTCP packets, may cause SIP Trunk fault.

E1 port light is not on or flashing

- Troubleshooting steps

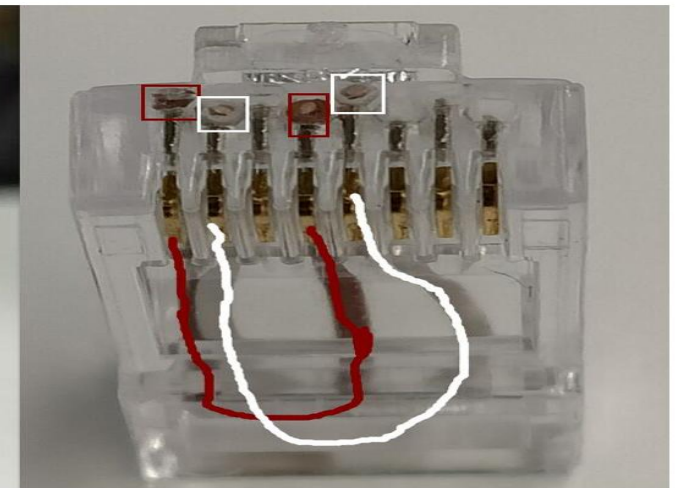
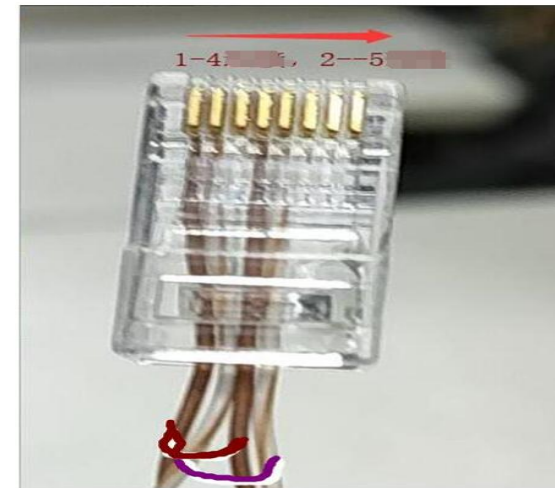
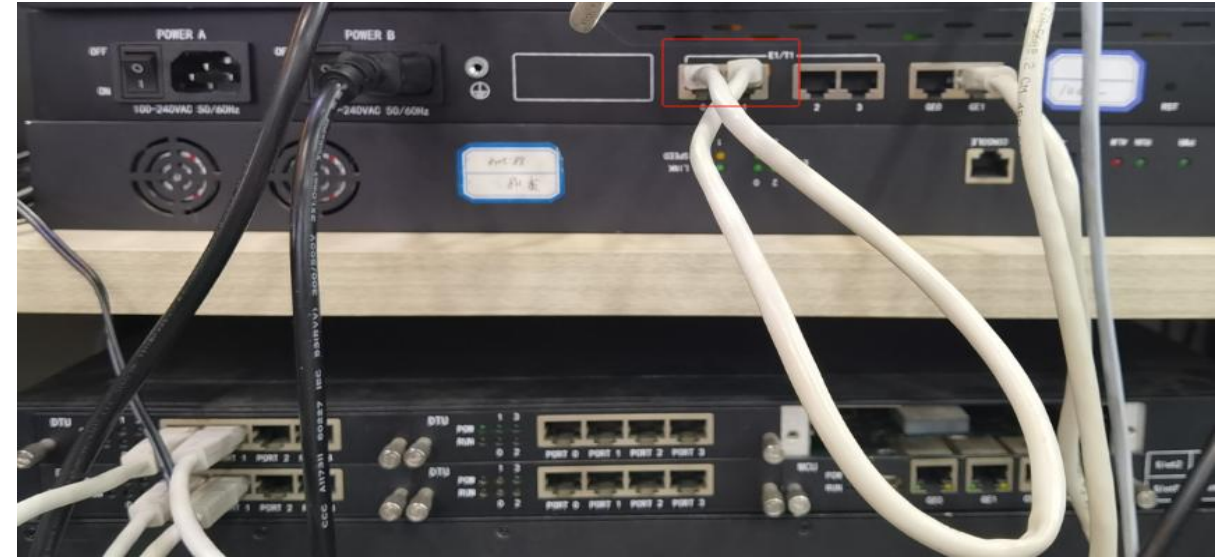
If the E1 indicator light on the MTG is not on or flashing, it indicates that there is a fault in the connection between the MTG and the switch. You can follow the following steps to troubleshoot

1. Check if the E1 connector is in good contact
2. Exchange the receiving and sending ends of the MTG side E1 interface
3. Check the E1 line sequence
4. Self loop test



E1 port light is not on or flashing

- Self loop test
 1. **The number of e1 ports>1:** use the E1 cable in the accessory to connect the two E1 ports
 2. **The number of e1 ports=1:** Use a crystal head to connect 1-4 and 2-5, create an E1 self ring head, and insert it into the corresponding E1 port





4. **"ISDN/SS7 signal alarm"** : indicates that the PRI trunk link has not been successfully established. Check if the E1 line sequence, BNC adapter box, and PSTN trunk status are receiving and transmitting, and attempt to modify frame format and clock source

Modify

SS7 link failure

- SS7 Config-SS7 Circuit Maintain

1. **"Fault"** : indicates that the E1 line is physically disconnected. It could be an issue with the E1 line, BNC adapter box, or a reverse connection of the BNC transmission and reception lines





2. **"RAI Alarm"** : indicates that the underlying E1 port of the device cannot receive data from the other party. It may be an issue with the E1 line, BNC adapter box, or the other party's interface

3. **"AIS Alarm"** : This type of alarm is usually caused by the other party not activating business data

















4. **"ISDN/SS7 signal alarm"** : indicates that the SS7 trunk link has not been successfully established. Check PSTN trunk status, SS7 trunk, SS7 MTP link, and ss7 cic configurations are consistent with the data provided by the other end, and attempt to modify the frame format and clock source

SS7 Circuit Maintain

Operation Mode: E1/T1

Master TG	0	1	2	3
Protocol Type			ISUP	ISUP
DTU 0				
	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Select All Invert Clear Block Unblock Reset Cancel

Activated	Disable	Fault	RAI Alarm	AIS Alarm	ISDN/SS7 Signal Ala...				
									
Frame-Sync	Idle	Signal	Busy	L-blocked	R-blocked	B-blocked	Blocking	Unblocking	Resetting
									

Notes: L-Blocked -- Local Blocked, R-Blocked -- Remote Blocked, B-Blocked -- Both Sides Blocked

Call failed

- Signaling Call Test—IP Trunk

Test purpose:

To test IP-PSTN directional calls and verify whether the connected SS7/PRI trunk lines can make normal calls

Testing method:

1. Click **maintenance- signaling call testing**
2. Source type: select "IP trunk" 、 Trunk type: select "SIP"
3. calling number: the line number assigned by the operator
4. called number :the test phone number

Test success phenomenon:

Click 'Save' and select 'Start'. The dialed phone will ring and you can pick it up

Signaling Call Test

Source Trunk

Source Type: IP Trunk

Trunk Type: SIP

IP Trunk No.: 0 <172.27.10.37>

Calling Number: 801

Called Number: 10086

Signaling Trace

```
CC(ccb:3) <=> CC_ST_SETUP, cr:5, calling:801, dial:10086, num_ok:1, trunkGrpId:255, profileId:255 [IP2Tel]Match Route Succ! Index:511(OUTSIDE)
CC(ccb:3) ==> CC_ST_SETUP
PRA(ccb:3) <=> CC_ST_SETUP, calling:801, dial:10086, send_ok:1
PRA(ccb:3) ==> CC_SETUP_REQ, index:6, if:65535, trunkGrp:0, calling:801, called:10086, presentId:0, trans:
PRA Send Msg: MT_SETUP
PRA(ccb:3) <=> CC_PROCEEDING_IND,, cause:0(OX)
PRA Send Msg: MT_SETUP
PRA(ccb:3) <=> CC_PROCEEDING_IND,, cause:0(OX)
```

Save Start Stop Clear

Call failed

- SS7 call process

SS7 Send Msg: ISUP_IAM. /* SS7 call requests sent by MTG*/

SS7 Got Msg: ISUP_ACM. /* The MTG receives a ringing signal from the line back */

SS7 Got Msg: ISUP_ANM. /* The MTG has received a communication command to connect the line back */

SS7 Got Msg: ISUP_REL. /* The MTG receives a signal to disconnect the line */

```
CC(ccb:20) <== CC_ST_SETUP, cr:38, calling:888888, longNum:888888, dial:15112272039,
num_ok:1, trunkGrpId:255, profileId:255
[IP2Tel]Match Route Succ! Index:511(888888)
CC(ccb:20) ==> CC_ST_SETUP
SS7(ccb:20) <== CC_ST_SETUP, calling: 888888, long: 888888, dial:15112272039, send_ok:1
SS7(ccb:20) ==> CC_SETUP_REQ, index:39, if:65535, trunkGrp:0, calling:888888, called:15112272039,
presentId:0, trans:
SS7 Send Msg: ISUP_IAM
SS7(ccb:20) <== CC_PROCEEDING_IND, cause:0(OK)
SS7 Got Msg: ISUP_ACM.
SS7(ccb:20) <== CC_ALERTING_IND, cause:0(OK)
CC(ccb:20) <== CC_ST_SETUP_ACK
CC(ccb:20) <== CC_ST_ALERTING
SS7 Got Msg: ISUP_ANM.
SS7(ccb:20) <== CC_SETUP_CFM, cause:0(OK)
CC(ccb:20) <== CC_ST_CONNECT
CC(ccb:20) <== CC_ST_CONNECT_ACK
SS7 Got Msg: ISUP_REL.
SS7 Send Msg: ISUP_RLC
SS7(ccb:20) <== CC_DISCONNECT_IND, cause:16(Normal call clearing)
SS7(ccb:20) <== CC_RELEASE_CFM, cause:16(Normal call clearing)
CC(ccb:20) <== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR), state:7
CC(ccb:20) <== CC_ST_REL_COMP, cause:0(CCS_NONE), state:9
```

Call failed

- PRI call process

PRA Send Msg: MT_SETUP /* PRI call requests sent by MTG */

PRA Got Msg: MT_CALL_PROCEEDING /* The other end has received the call request and is currently processing it */

PRA Got Msg: MT_ALERTING /* Received ringing signal returned by the line */

PRA Got Msg: MT_CONNECT /* Received communication command for line return connection */

PRA Got Msg: MT_DISCONNECT /* Received the disconnection of the line, the reason value is below */

```
CC(ccb:29) <== CC_ST_SETUP, cr:53, calling:888888, longNum:888888, dial:15112272039,
num_ok:1, trunkGrpId:255, profileId:255
[IP2Tel]Match Route Succ! Index:511(888888-888888) /* Call direction, index routing number */
CC(ccb:29) ==> CC_ST_SETUP
PRA(ccb:29) <== CC_ST_SETUP, calling: 888888, long: 888888, dial:15112272039, send_ok:1
PRA(ccb:29) ==> CC_SETUP_REQ, index:54, if:65535, trunkGrp:0, calling: 888888, called:15112272039,
presentId:0, trans: /* The final calling and called numbers sent by the device */
PRA Send Msg: MT_SETUP
PRA(ccb:29) <== CC_PROCEEDING_IND, cause:0(OK)
PRA Got Msg: MT_CALL_PROCEEDING
PRA(ccb:29) <== CC_PROCEEDING_IND, cause:0(OK)
CC(ccb:29) <== CC_ST_SETUP_ACK
PRA Got Msg: MT_ALERTING
PRA(ccb:29) <== CC_ALERTING_IND, cause:0(OK)
CC(ccb:29) <== CC_ST_ALERTING
PRA Got Msg: MT_CONNECT
PRA Send Msg: MT_CONNECT_ACKNOWLEDGE
PRA(ccb:29) <== CC_SETUP_CFM, cause:0(OK)
CC(ccb:29) <== CC_ST_CONNECT
CC(ccb:29) <== CC_ST_CONNECT_ACK
PRA Got Msg: MT_DISCONNECT
PRA(ccb:29) <== CC_DISCONNECT_IND, cause:144(Normal call clearing)
PRA Send Msg: MT_RELEASE
PRA Got Msg: MT_RELEASE_COMPLETE
PRA(ccb:29) <== CC_RELEASE_CFM, cause:0(OK)
CC(ccb:29) <== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR)), state:7
CC(ccb:29) <== CC_ST_REL_COMP, cause:0(CCS_NONE)), state:9
```

Call failed

- Signaling Call Test—PSTN Trunk

Test purpose:

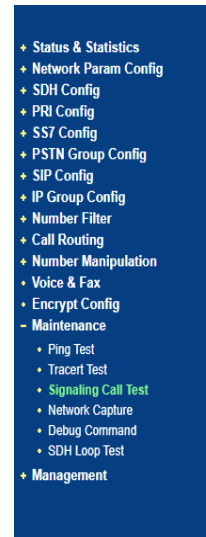
Test PSTN-IP directional calls and verify if the connected SIP trunk line can make normal calls

Testing method:

1. Click **maintenance- signaling call testing**
2. Source type: select “PSTN trunk”
3. calling number: any number, and if the route has requirements for the calling, fill in according to the requirements
4. called number :the test line number

Test success phenomenon:

Click 'Save' and select 'Start'. The corresponding extension is ringing and can be connected



A screenshot of the 'Signaling Call Test' configuration page. The page has a blue header with the title 'Signaling Call Test'. Below the header, there are several input fields and a text area. The 'Source Trunk' section has a dropdown menu set to 'PSTN Trunk'. The 'Source Type' section has a dropdown menu set to 'PSTN Trunk No.'. The 'Calling Number' field contains '10086'. The 'Called Number' field contains '801'. Below these fields is a 'Signaling Trace' section with a text area containing a log of SIP messages. At the bottom of the page, there are four buttons: 'Save', 'Start', 'Stop', and 'Clear'.

```
CC(ecb:1) <== CC_ST_SETUP, cr:1, calling:10086, dial:801,
num_ok:1, trunkGrpId:255, profileId:255
[Tel2IP]Match Route Succ! Index:511(123123)
CC(ecb:1) ==> CC_ST_SETUP
SIP(ecb:1) <== CC_ST_SETUP
SIP(ecb:1) ==> SIP_CALL_INVITE, local:sip:10086@172.29.50.42,
peer:sip:801@172.28.29.178
CC(ecb:1) <== CC_ST_SETUP_ACK
```

Call failed

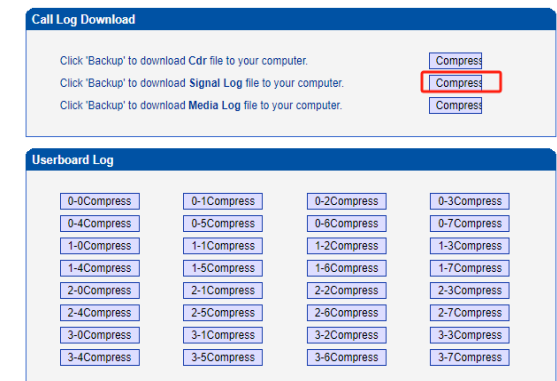
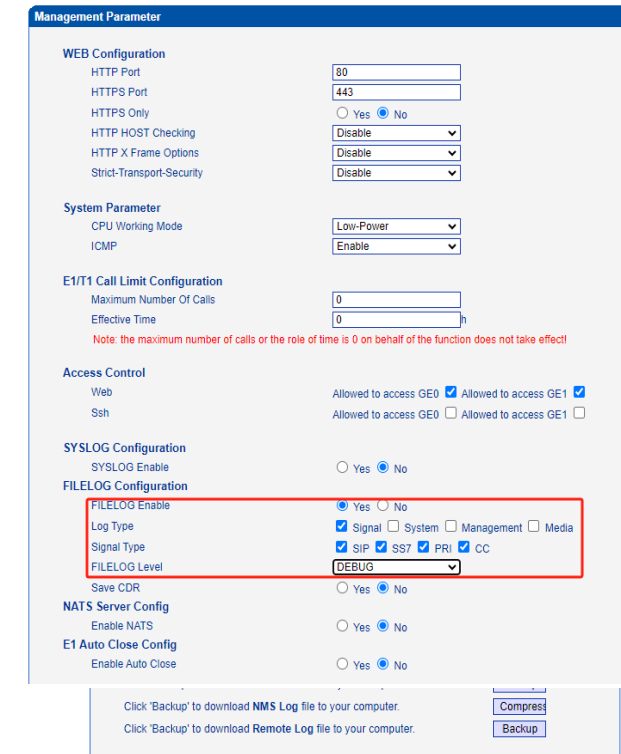
- FILELOG

Test purpose:

When combining SIP and PRI/SS7 messages for analysis, filelog can be enabled

Testing method:

1. Click **management- management parameter**
2. Enable Filelog and check the box as shown in the diagram
3. Reproduce the problem
4. Click **management- Date Download**, download signaling logs



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Chapter Three

problems & solutions for login

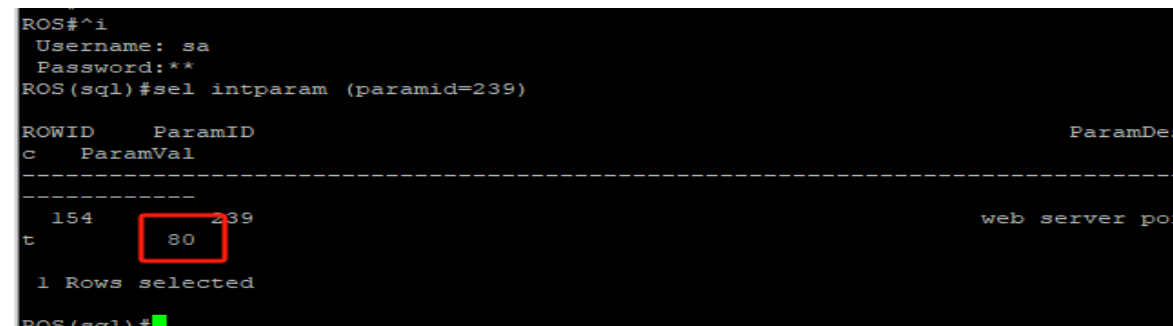
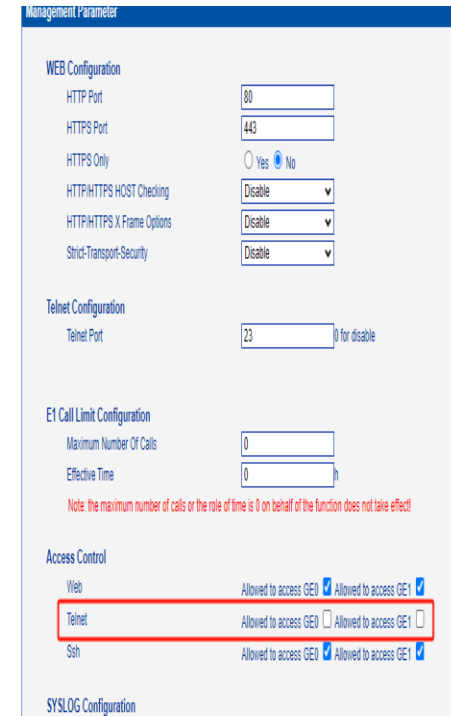
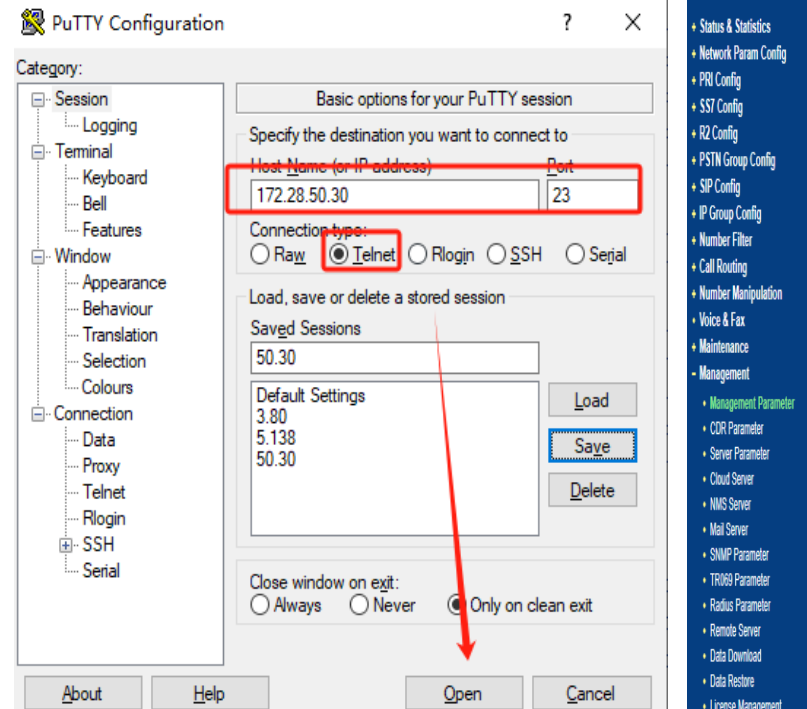
03

Unable to log in with correct IP and password



- Check if the web port is correct

1. Click **Management-Management Parameter**, Enable Telnet permission
2. Enter the MTG device IP and log in via telnet
3. Enter the MTG login username and password, enter the commands in sequence according to the picture and query the web port
4. http://ip: Port attempts to log in to device



Unable to log in with correct IP and password



- Telnet View ACL

Enter the MTG device IP and log in via telnet

- acl set**: check if the corresponding network port web access permission is enabled
- acl show** : check if the device has restrictions on IP addresses for web login and telnet access
- acl black show** :View Access Blacklist

```
Welcome to Command Shell!
Username:admin
Password:*****
ROS>en
ROS#^config
ROS(config)#acl set
Enable GE1 Web Access(y/n): y
Enable GE0 Web Access(y/n): y
Enable GE1 Telnet Access(y/n): y
Enable GE0 Telnet Access(y/n): y
Enable GE1 SSH Access(y/n): y
Enable GE0 SSH Access(y/n): y
Enable GE1 SNMP Access(y/n): y
Enable GE0 SNMP Access(y/n): y
Enable GE1 radius Access(y/n): y
Enable GE0 radius Access(y/n): y
Enable GE1 smtp Access(y/n): y
Enable GE0 smtp Access(y/n): y
Enable GE1 drp Access(y/n): y
Enable GE0 drp Access(y/n): y
Enable GE1 cloud Access(y/n): y
Enable GE0 cloud Access(y/n): y
Enable GE1 syslog Access(y/n): y
```

```
ROS(config)#
ROS(config)#acl black show
All Login Fobiden is Off!
Index    ip
```

```
ROS(config)#acl show
Type      status
GE0 Web Access :Allowed
GE1 Web Access :Allowed
GE0 Telnet Access :Allowed
GE1 Telnet Access :Allowed
GE0 Ssh Access :Allowed
GE1 Ssh Access :Allowed
```

```
GE0 Snmp Access :Allowed
GE1 Snmp Access :Allowed
```

```
GE0 Radius Access :Allowed
GE1 Radius Access :Allowed
```

```
GE0 Sntp Access :Allowed
GE1 Sntp Access :Allowed
```

```
GE0 Drp Access :Allowed
GE1 Drp Access :Allowed
```

```
GE0 Cloud Access :Allowed
GE1 Cloud Access :Allowed
```

```
GE0 Syslog Access :Allowed
GE1 Syslog Access :Allowed
```

```
GE0 Nats Access :Allowed
GE1 Nats Access :Allowed
```

```
White list      status
```

```
SIP Trunk       :Disable
H323 Trunk      :Disable
Telnet Login    :Disable
Web Login       :Disable
```

```
RowID  Address      AccessType
0      172.28.7.49  | Telnet Login |
```



THANKS



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www.dinstar.com



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