

MTG Advance Configuration Practice Guide

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Foreword

- This course is mainly:
 - Introduce MTG Advanced Features
 - Introduce MTG Advanced Configuration

Course Objective

Through this course
you will be able to



What are the Advanced Features



How To SET Advanced Features



How To Set IMS

Contents

- 1 Chapter One MTG Advanced Features
- 2 Chapter Two MTG Advanced Configuration
- 3 Chapter Three SNMP

Chapter One

MTG Advanced Features

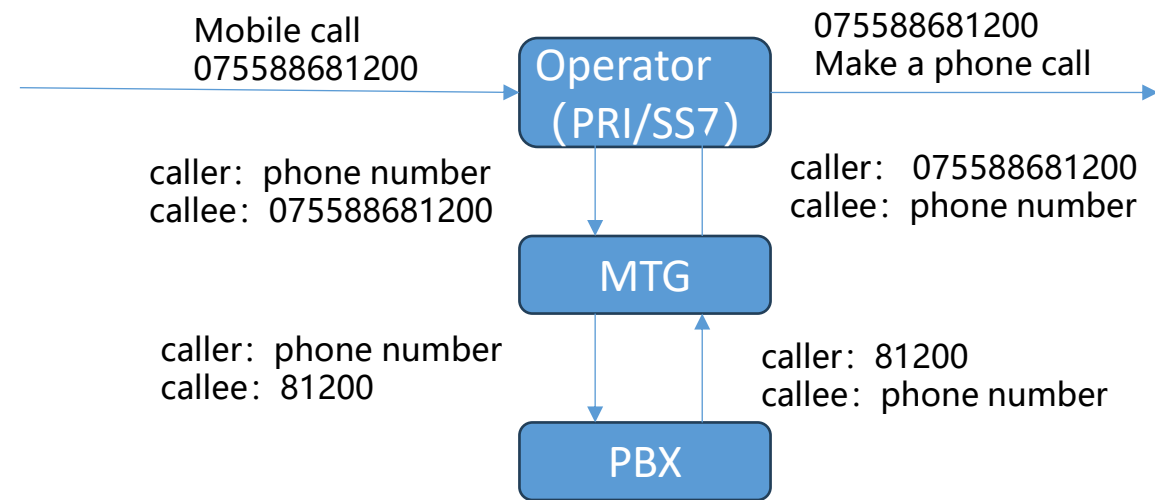
01

Number Manipulation

- Call Process

Incoming Call: operator-MTG-PBX, MTG needs to remove the 0755886 prefix from the called and send it to PBX

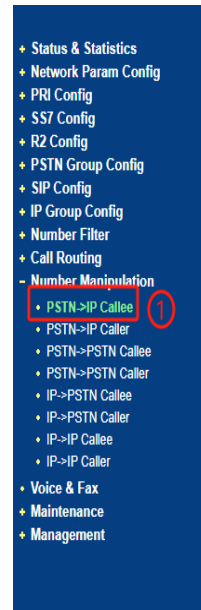
Outcoming Call: PBX-MTG-operator, extension number 81200 needs to be prefixed with 0755886 to be called out through the operator's line



Number Manipulation

• Incoming Call Number Manipulation

1. Click on **Number Manipulation - PSTN ->IP Callee**
2. Source type: PSTN trunk corresponding to the operator
3. The caller prefix Fill in . indicate any ,the callee prefix Fill in 07558864
4. Number of digits to strip from the left fill in 7



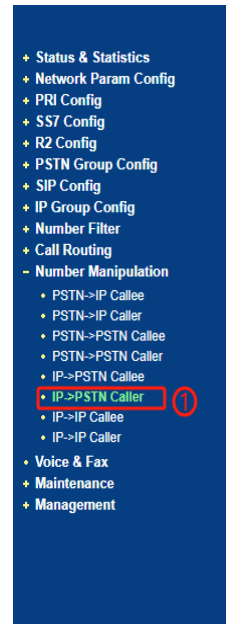
A screenshot of the 'PSTN->IP Callee Add' configuration window. The window contains several fields for configuration. The 'Index' field is set to 511. The 'Description' field is set to 'del'. The 'Source Type' field is set to 'PSTN Trunk'. The 'Callee Prefix' field is set to '07558864'. The 'Caller Prefix' field is set to '.'. The 'Number of Digits to Strip from Left' field is set to 7. The 'Number of Digits to Strip from Right' field is empty. The 'Prefix to Be Added' field is empty. The 'Suffix to Be Added' field is empty. There are buttons for 'OK', 'Reset', and 'Cancel' at the bottom. Red circles with numbers 2, 3, and 4 are placed next to the 'Source Type', 'Callee Prefix', and 'Number of Digits to Strip from Left' fields respectively.

NOTES: 1. Fields with "*" are MUST.
2. "." in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

Number Manipulation

• Outcoming Call Number Manipulation

1. Click on **Number Manipulation - IP ->PSTN Caller**
2. Source type: Select the SIP trunk corresponding to PBX
3. The caller prefix Fill in 81200, the callee prefix Fill in . indicate any
4. Prefix to Be Added: Fill in 0755886



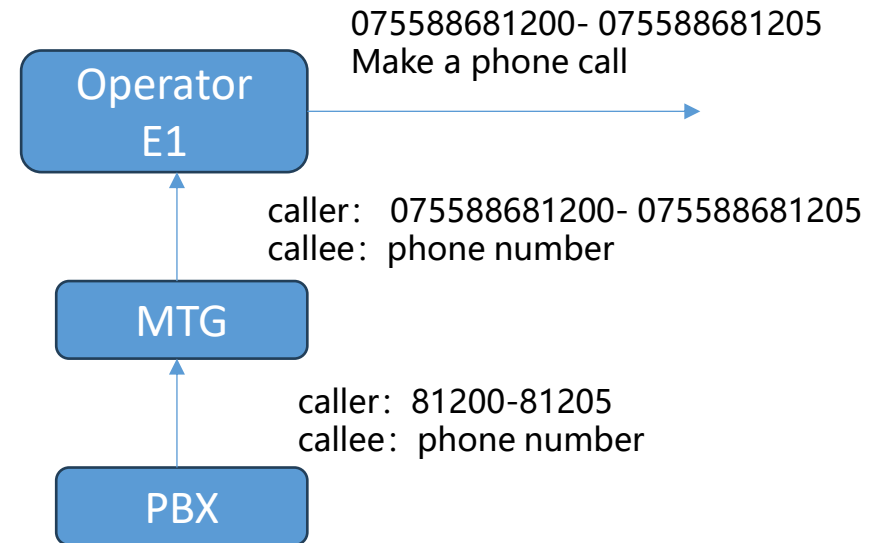
A screenshot of the 'IP->PSTN Caller Add' configuration form. The form has a light blue header and a white body. It contains several fields with red annotations: 1. 'Index' is set to '511'. 2. 'Trunk Type' is set to 'Trunk' (highlighted with a red box and a circled '2'). 3. 'Trunk Number' is set to '5 <pbx>' (highlighted with a red box and a circled '3'). 4. 'Callee Prefix' is set to '.' (highlighted with a red box and a circled '3'). 5. 'Caller Prefix' is set to '81200' (highlighted with a red box and a circled '3'). 6. 'Prefix to Be Added' is set to '0755886' (highlighted with a red box and a circled '4'). 7. 'Suffix to Be Added' is empty (highlighted with a red box and a circled '4'). Other fields include 'Description' (add), 'Source Type' (SIP), 'Number of Digits to Strip from Left' (empty), 'Number of Digits to Strip from Right' (empty), 'Number of Digits to Reserve from Right' (empty), 'Number Type' (Not Configured), and 'Presentation Indicator' (Not Configured). At the bottom are 'OK', 'Reset', and 'Cancel' buttons.

NOTES: 1. Fields with '*' are MUST.
2. '.' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

Caller Pool

- Call Process

Call out: It appears to be any one of the numbers 075588681200-075588681205 issued by the operator

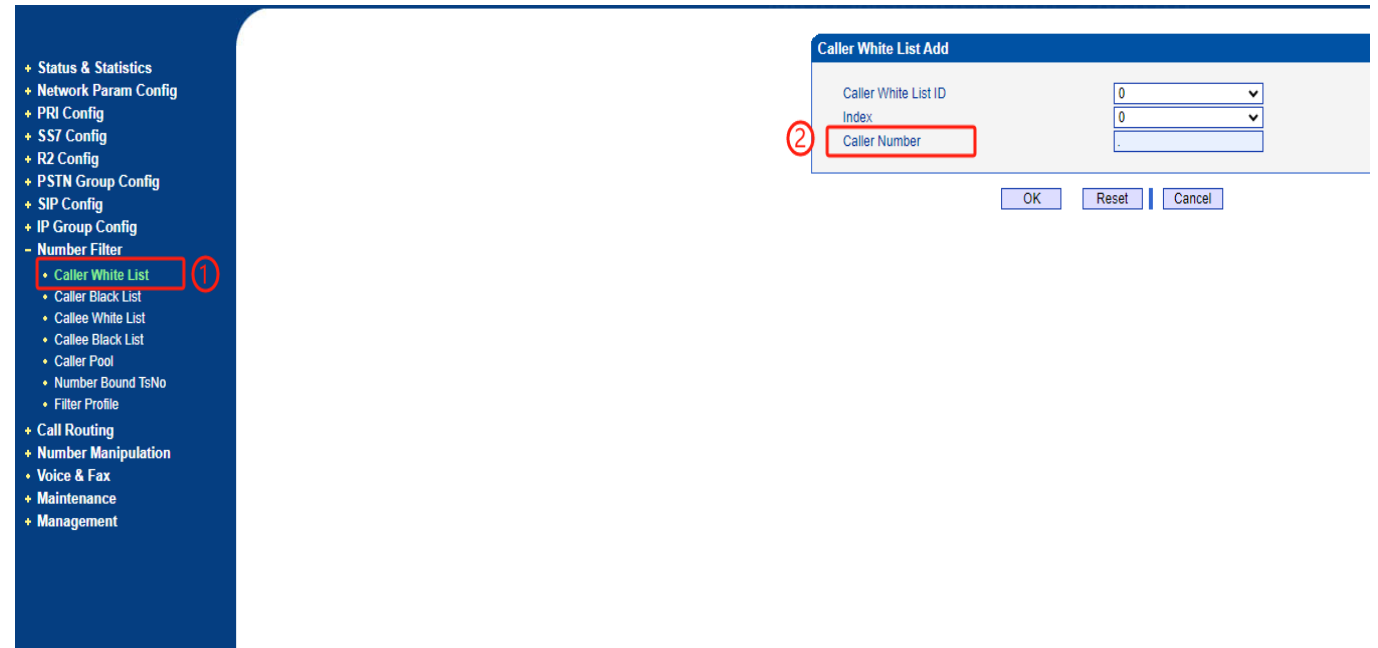


Caller Pool

- Caller Pool configuration



1. Click on **Number Filter - Caller White List**
2. Caller number fill in . indicate any

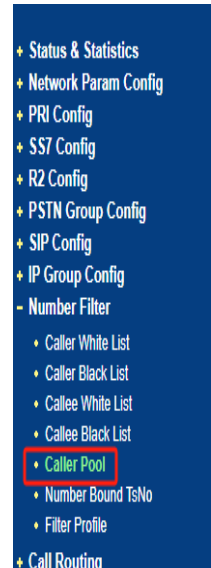


Caller Pool

- Caller Pool configuration



1. Click on **Number Filter - Caller Pool**
2. Fill in the starting number and count provided by the operator



NOTE: 1. eg.: option 'Starting Caller Number' is 80080000, and option 'Number Count' is 100, means that caller number range is 80080000-80080099.
2. Each group can contain at most 512 items, but the total items of all list can not more than 1024.
3. 'Number Count' can not be more than 4000.

Caller Pool

- Caller Pool configuration



1. Click on **Number Filter - Filter Profile**
2. Select Caller White List ID and Caller Pool for White List

Filter Profile Add

Filter Profile ID	0
Description	caller
Caller White List ID	0
Caller Black List ID	255 <None>
Callee White List ID	255 <None>
Callee Black List ID	255 <None>
Caller Pool for White List	0
Caller Pool for Black List	255 <None>
Caller Pool for Calling Transfer	255 <None>
Rcd Caller White List	255 <None>
Rcd Callee White List	255 <None>
Recog Caller White List	255 <None>
Recog Callee White List	255 <None>
Callee Bound TsNo	255 <None>
Presentation Indicator	Not Configured

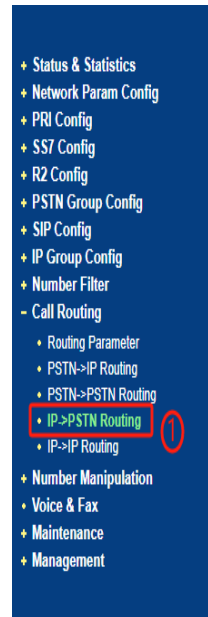
OK Reset Cancel

Caller Pool

- Caller Pool configuration



1. Click on **Call Routing- IP->PSTN Routing**
2. Select filter profile id



Index	511
Description	12312
Source Type	Trunk
Trunk Type	SIP
Trunk No.	5 <pbx>
Callee Prefix	.
Caller Prefix	.
Destination Type	Trunk
PSTN Trunk	0 <pri>
Filter Profile ID	0 <caller>

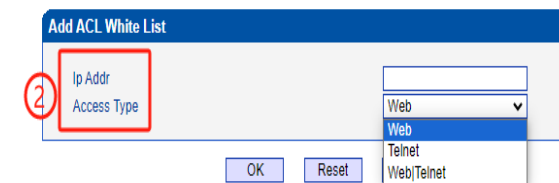
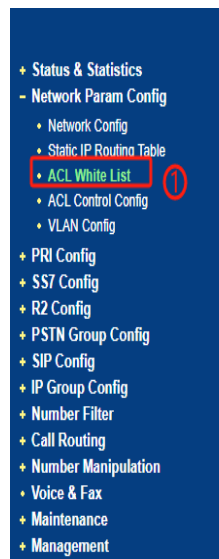
OK Reset Cancel

NOTE: '.' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

ACL

- ACL White List configuration

1. Click on **Network Param Config-ACL White List**
2. Fill in the IP address and select the access type



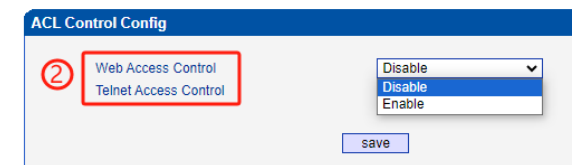
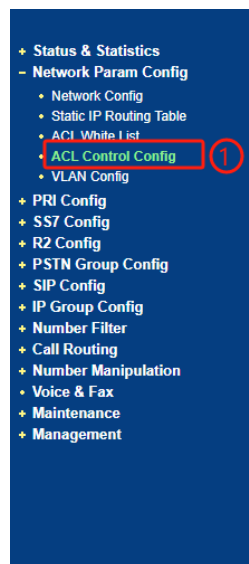
ACL

- ACL White List configuration

1. Click on **Network Param Config-**

ACL Control Config

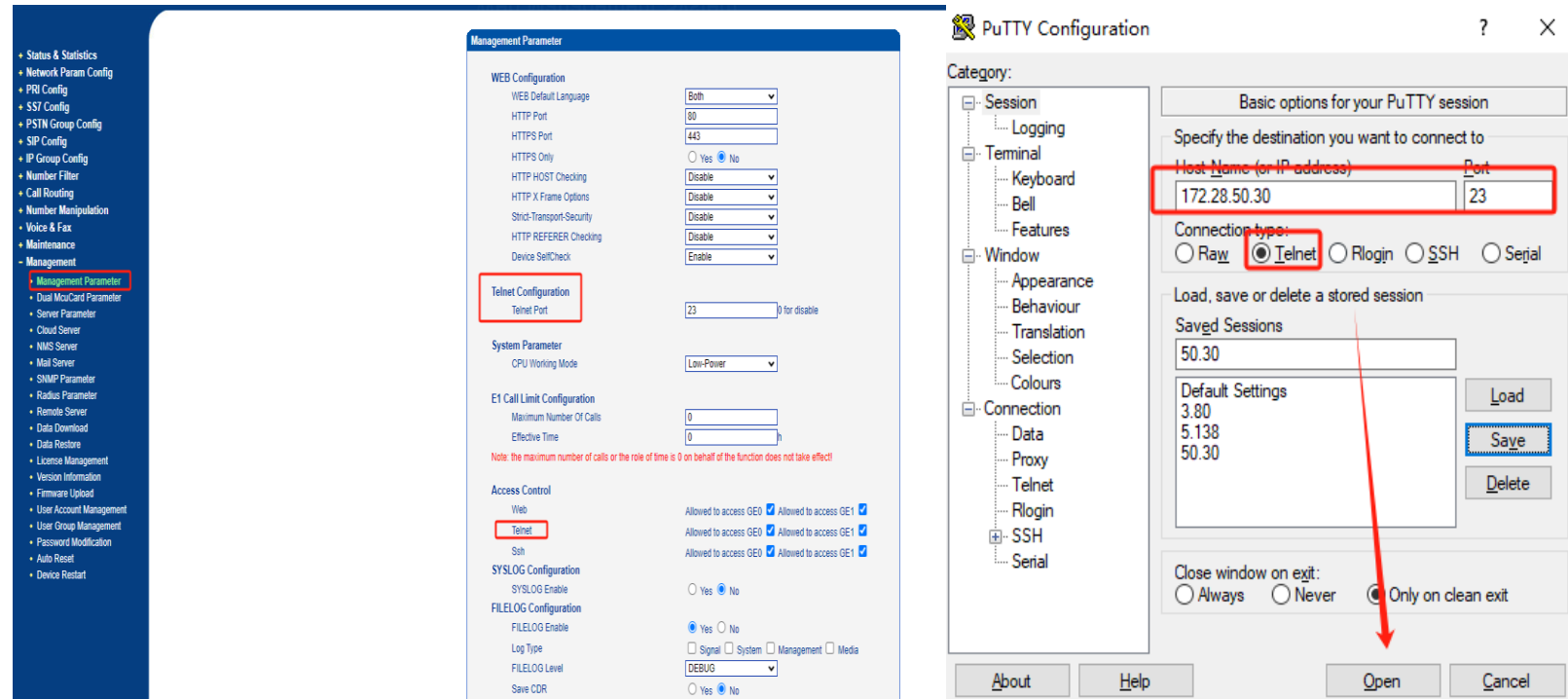
2. Choose whether to enable access control. Once enabled, IP addresses outside the ACL white list cannot access devices via Web/Telnet



Switch Mode

- Telnet MTG

1. Click on **Network Param Config-ACL Control Config**
2. Choose whether to enable access control. Once enabled, IP addresses outside the ACL white list cannot access devices via Web/Telnet



Switch Mode

- E1->Transcoding

The SIP-SIP call mode (such as PBX - MTG (GE1 public network) - MTG (GE0 intranet) - IMS), where both sides of the device are connected to the server through SIP trunk

- 1.Telnet MTG
- 2.Enter the command
- 3.Restart the MTG

```
username:admin ↵  
Password:***** ↵  
ROS>en↵  
ROS# ↵  
ROS#^i ↵  
Username: sa ↵  
Password:sa ↵  
ROS(sql)#update intparam (paramid=0) (paramval=3)  
ROS(sql)#update intparam (paramid=22) (paramval=1)  
↵  
ROS(sql)#db save↵
```

Switch Mode

- Transcoding->E1

1.Telnet MTG

2.Enter the command

3.Restart the MTG

```
Welcome to Command Shell!↵
```

```
username:admin ↵
```

```
Password:***** ↵
```

```
ROS>en↵
```

```
ROS# ↵
```

```
ROS#^i ↵
```

```
Username: sa ↵
```

```
Password:sa↵
```

```
ROS(sql)#update intparam (paramid=0) (paramval=0)
```

```
ROS(sql)#db save↵
```

IP Profile

1. Click on **SIP Group Config-IP Profile**

2. Ringback Tone to PSTN Originated from:

Local-- MTG playback

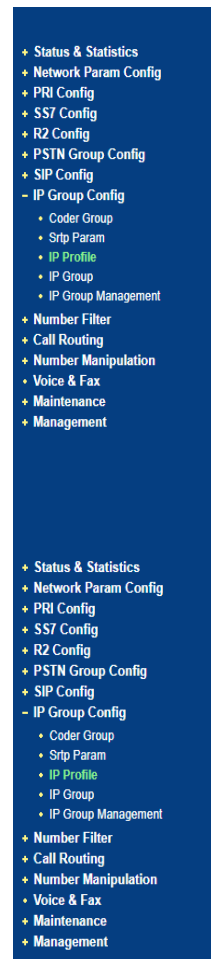
IP--SIP server side playback

Adaptive-- determine the playback side based on negotiation

3. Ringback Tone to IP Originated from:

Local-- MTG playback

PSTN-- PSTN side playback



A screenshot of the 'IP Profile Modify' dialog box. It contains several configuration fields: IP Profile ID (0), Description (Default), Declare RFC2833 in SDP (Yes), Support Early Media (Yes), Ringback Tone to PSTN Originated from (highlighted with a red box and set to IP), Ringback Tone to IP Originated from (Local), Wait for RTP Packet from Peer (Adaptive), and T.30 Expanded Type in SDP (X-Fax). At the bottom are OK, Reset, and Cancel buttons.

A screenshot of the 'IP Profile Modify' dialog box, similar to the one above but with different settings. The 'Ringback Tone to IP Originated from' field is highlighted with a red box and set to Local. The 'Ringback Tone to PSTN Originated from' field is set to IP. The 'Wait for RTP Packet from Peer' field is set to PSTN. The other fields and buttons are the same as in the previous screenshot.

User=phone



SIP config - SIP parameter can enable or disable user=phone

162 2024-09-30 18:58:54.542896 172.27.10.2 172.28.7.49 HTTP 466 HTTP/1.0 302 Redirect (text/html)

163 2024-09-30 18:58:54.544945 172.27.10.2 MTG 172.28.51.105 SIP/SIP 932 Request: INVITE sip:803@172.28.51.105;user=phone

164 2024-09-30 18:58:54.565755 172.28.7.49 172.27.10.2 TCP 66 24739 → http(80) [ACK] Seq=562 Ack=425 Win=262912 Len=0 TS

> Frame 163: 932 bytes on wire (7456 bits), 932 bytes captured (7456 bits)

> Ethernet II, Src: Dinstar_7e:9a:60 (f8:a0:3d:7e:9a:60), Dst: Dinstar_5a:63:59 (f8:a0:3d:5a:63:59)

> Internet Protocol Version 4, Src: 172.27.10.2 (172.27.10.2), Dst: 172.28.51.105 (172.28.51.105)

> User Datagram Protocol, Src Port: sip (5060), Dst Port: 20900 (20900)

> Session Initiation Protocol (INVITE)

> Request-Line: INVITE sip:803@172.28.51.105;user=phone SIP/2.0

> Message Header

> Message Body

• Status & Statistics

• Network Param Config

• PRI Config

• SS7 Config

• R2 Config

• PSTN Group Config

• SIP Config

SIP Parameter

• SIP Trunk

• SIP Account

• SIP DNS

• SIP RED Group

• IP Group Config

• Number Filter

• Call Routing

• Number Manipulation

• Voice & Fax

• Maintenance

• Management

Policy of overload Protection	Reject & Reply ErrCode
Error Code(Exceed Max Caps Limit)	486
Error Code(Lack of Resources)	486
Max Caps	100
Pre-Ringback	Disable
Same Number Forbidden	Disable
Diversion	Disable
To	Disable
PPI	Disable
PAI	Enable
Tel Format	Disable
HI	Disable
Account Select Mode	Cyclic Ascending
Account Match Mode	Original Calling Num
Register Speed	15
Expire Coefficient	0.8
Refresh Register with Auth	Disable
Precondition	Disable
PSTN->IP Match Diversion Number	Disable
OrigCallee from	PoolNumber
URI including "user=phone"	Enable
AMR Octet Align	Disable
PPbx Info	Disable
181 Forwarding	Disable
Invite with PEM Header	Disable
183 with PEM Header	Disable
GE1 Static Nat	Disable
GE0 Static Nat	Disable
User to User Header	Disable

NAT

- Static Nat

Implemented one-to-one mapping
between private and public addresses

1. **SIP config - SIP Parameter**, select the corresponding network port to enable static NAT, and fill in the public IP address
2. **SIP config - SIP Trunk**, select the corresponding SIP trunk, enable static NAT

- Status & Statistics
- Network Param Config
- PRI Config
- SS7 Config
- R2 Config
- PSTN Group Config
- SIP Config
 - SIP Parameter
 - SIP Trunk
 - SIP Account
 - SIP DNS
 - SIP RED Group
- IP Group Config
- Number Filter
- Call Routing
- Number Manipulation
- Voice & Fax
- Maintenance
- Management

- ✦ Status & Statistics
- ✦ Network Param Config
- ✦ PRI Config
- ✦ SS7 Config
- ✦ R2 Config
- ✦ PSTN Group Config
- SIP Config
 - ✦ SIP Parameter
 - ✦ SIP Trunk
 - ✦ SIP Account
 - ✦ SIP DNS
 - ✦ SIP RED Group
- ✦ IP Group Config
- ✦ Number Filter
- ✦ Call Routing
- ✦ Number Manipulation
- ✦ Voice & Fax
- ✦ Maintenance
- ✦ Management

Policy of overload Protection	Reject & Rely ErrCode	485
Error Code(Exceed Max Caps Limit)		486
Error Code(Lack of Resources)		100
Max Caps	Disable	▼
Pre-Ringback	Disable	▼
Same Number Forbidden	Disable	▼
Diversion	Disable	▼
To	Disable	▼
PPI	Disable	▼
PAI	Enable	▼
Tel Format	Disable	▼
HI	Disable	▼
Account Select Mode	Cyclic Ascending	▼
Account Match Mode	Original Calling Num	▼
Register Speed	15	▼
Expire Coefficient	0.8	▼
Refresh Register with Auth	Disable	▼
Precondition	Disable	▼
PSTN->IP Match Diversion Number	Disable	▼
Orig/Callas from	Disable	▼
URI including "user-phone"	Enable	▼
AMR Octet Align	Disable	▼
PPbx Info	Disable	▼
181 Forwarding	Disable	▼
Invite with PEM Header	Disable	▼
189 with PEM Header	Enable	▼
GE1 Static Nat	72s	▼
Nat ip		
GE0 Static Nat	Disable	▼
User to User Header	Disable	▼
User Agent Header	Disable	▼
Header Passthrough	Disable	▼
SIP Info Dtmf Mode	dtmf-relay	▼
SIP Default Error Code	500	▼
DNS Refresh Interval(0-60,0-disable)	0	min
SIP DNS Query Type	A Query	▼
SIP Header Param Escape	Disable	▼
Not Resolve Route Error Code		
ACK without Totag compatibility	Disable	▼
URL schemes	Sip URL	▼

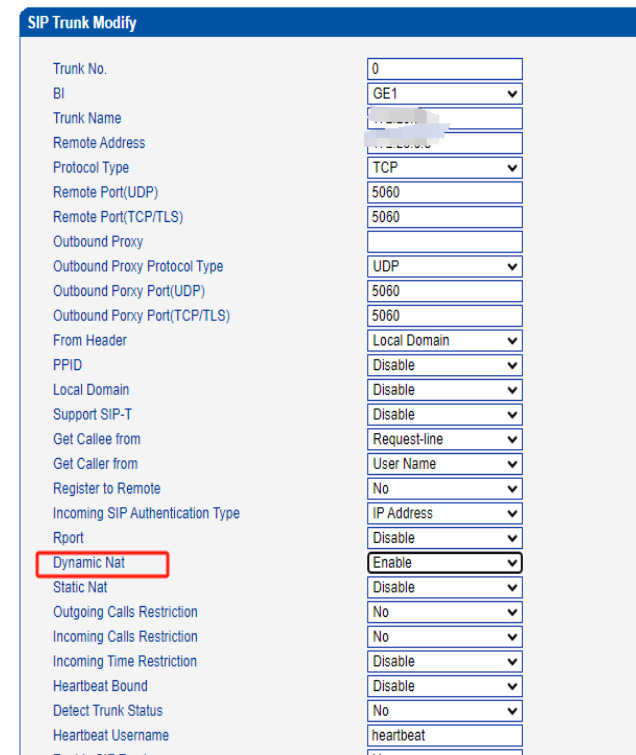
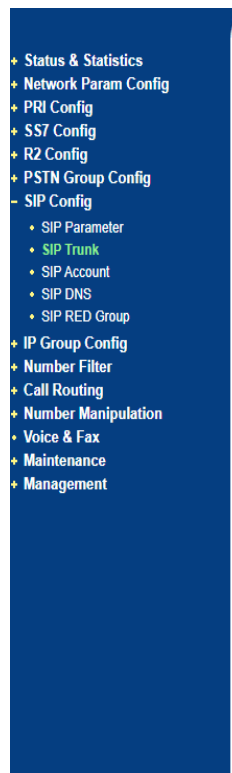
SIP Trunk Modify	
Trunk No.	0
BI	GE1
Trunk Name	
Remote Address	
Protocol Type	TCP
Remote Port(UDP)	5060
Remote Port(TCP/TLS)	5060
Outbound Proxy	
Outbound Proxy Protocol Type	UDP
Outbound Proxy Port(UDP)	5060
Outbound Proxy Port(TCP/TLS)	5060
From Header	Local Domain
PPID	Disable
Local Domain	Disable
Support SIP-T	Disable
Get Callee from	Request-line
Get Caller from	User Name
Register to Remote	No
Incoming SIP Authentication Type	IP Address
Rport	Disable
Dynamic Nat	Disable
Static Nat	Enable
Outgoing Calls Restriction	No
Incoming Calls Restriction	No
Incoming Time Restriction	Disable

NAT

- Dynamic Nat

When converting private IP addresses of an internal network to public IP addresses, the IP address pairs are uncertain and random

SIP config - SIP Trunk, select the corresponding SIP trunk, enable Dynamic NAT



Diversion

- Application scenarios

A calls B, and then B calls C. The number calling A should be displayed above C

- Configuration method

Click on SIP Config - SIP Parameter to enable Diversion



Diversion

- Example

A:70051

B:13986547

C:2000

ROS(ada)#
ROS(ada)#23 08:01:47.400 mpe_sip: < 76> [INFO] MSG :<--- from
23 08:01:47.400 mpe_sip: < 77> [INFO] INVITE sip:2000@172.23.88.2
Via: SIP/2.0/UDP 172.23.222.18;branch=z9hG4bK5f6a47a72aa2725d93ee261f
From: "70051" <sip:70051@172.23.222.18>;tag=b7163ec634f0b67a0a93b6149
To: <sip:2000@172.23.88.201>
Call-ID: ee110de94735829e57bf6e76dc4af9af@172.23.222.18
CSeq: 257027318 INVITE
Contact: <sip:70051@172.23.222.18;transport=UDP>
Supported: 100rel, replaces
Diversion: <sip:13986547@172.23.88.201:5060>;reason=unconditional;cou
Allow: INVIT
23 08:01:47.400 mpe_sip: < 78> [INFO] E, ACK, CANCEL, BYE, OPTION
Content-Type: application/sdp
Max-Forwards: 70
Content-Length: 340

v=0
o=sip-01.06.11.22 25874689 25874690 IN IP4 172.23.222.18
s=-
c=IN IP4 172.23.222.18
t=0 0
m=audio 6232 RTP/AVP 8 0 18 4 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:4 G723/8000
a=fmtp:4 annexa=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
23 08:01:47.400 mpe_sip: < 79> [INFO] MSG :---> to 172.23.222.
23 08:01:47.400 mpe_sip: < 80> [INFO] SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.23.222.18;branch=z9hG4bK5f6a47a72aa2725d93ee261f
From: "70051" <sip:70051@172.23.222.18>;tag=b7163ec634f0b67a0a93b6149
To: <sip:2000@172.23.88.201>
Call-ID: ee110de94735829e57bf6e76dc4af9af@172.23.222.18
CSeq: 257027318 INVITE
Content-Length: 0
23 08:01:47.400 mpe_st: < 81> [INFO] ST: <-1,Sip-t,3,70051,idle>
o=sip-01.06.11.22 25874689 25874690 IN IP4 172.23.222.18

q931_redirecting.pcap
文件(F) 编辑(E) 视图(V) 跳转(G) 捕获(C) 分析(A) 统计(S) 电话(V) 无线(W) 工具(T) 帮助(H)
应用显示过滤器 ... <Ctrl-/>
No. Time Source Destination P:
1 2021-04-23 08:00:44.000210 0.0.0.0 0.0.0.0 Q
2 2021-04-23 08:00:44.000260 0.0.0.0 0.0.0.0 Q
3 2021-04-23 08:00:44.000260 0.0.0.0 0.0.0.0 T
4 2021-04-23 08:00:44.000270 0.0.0.0 0.0.0.0 T
5 2021-04-23 08:00:44.000300 0.0.0.0 0.0.0.0 T
6 2021-04-23 08:00:44.000310 0.0.0.0 0.0.0.0 T
7 2021-04-23 08:00:46.000430 0.0.0.0 0.0.0.0 T
8 2021-04-23 08:00:46.000480 0.0.0.0 0.0.0.0 T
9 2021-04-23 08:00:46.000480 0.0.0.0 0.0.0.0 T
10 2021-04-23 08:00:46.000520 0.0.0.0 0.0.0.0 T
11 2021-04-23 08:00:51.000010 0.0.0.0 0.0.0.0 T
12 2021-04-23 08:00:51.000050 0.0.0.0 0.0.0.0 T
PKT, Version: 3, Length: 49
Q.931
Protocol discriminator: Q.931
Call reference value length: 2
Call reference flag: Message sent from originating side
Call reference value: 000b
Message type: SETUP (0x05)
Bearer capability
Channel identification
Calling party number: '70051'
Called party number: '2000'
Redirecting number: '13986547'
Sending complete
0000 ff ff ff ff ff ff ff ff ff ff ff 08 00 45 00E
0010 00 59 00 00 40 00 80 06 00 00 00 00 00 00 00Y-@
0020 00 00 00 00 00 00 00 00 00 00 00 00 00 50 18P

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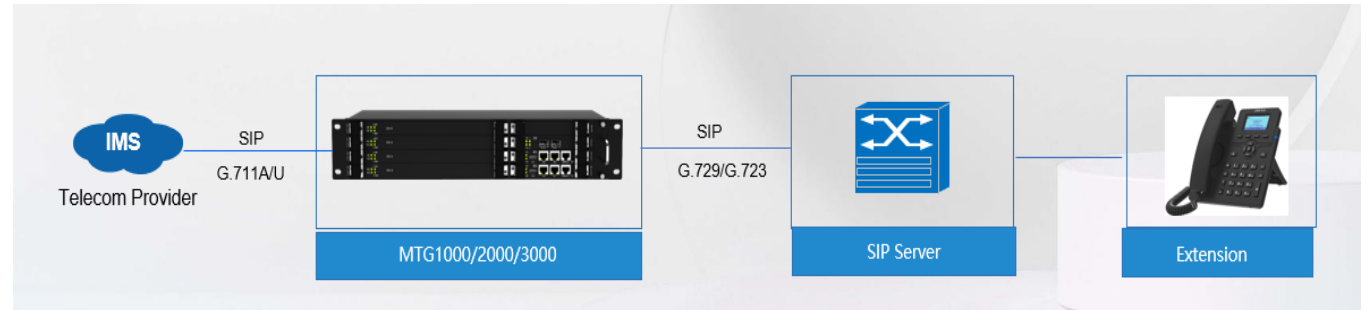
Chapter Two

MTG Advanced Configuration

02

- Application scenarios

1. Utilize MTG dual network ports, with one side connected to the IMS and the other side connected to the enterprise's local IP PBX, etc.
2. Connect the IMS dedicated line to the device network management port GE0, and add IMS accounts to register in the IMS core network.



- configuration



1. Click on **SIP Config – SIP Trunk**
2. Customize trunk name and select network port
3. Fill in the IP and port of PBX, and select the protocol

The screenshot shows the DINSTAR web interface. On the left is a navigation menu with the following items: Status & Statistics, Network Param Config, PRI Config, SS7 Config, R2 Config, PSTN Group Config, SIP Config, SIP Parameter, SIP Trunk (highlighted with a red box and a circled '1'), SIP Account, SIP DNS, SIP RED Group, IP Group Config, Number Filter, Call Routing, Number Manipulation, Voice & Fax, Maintenance, and Management. On the right is the 'SIP Trunk Modify' form. It contains the following fields: Trunk No. (0), BI (GE1), Trunk Name (pbx), Remote Address (172.27.10.23), Protocol Type (UDP), Remote Port(UDP) (5060), Remote Port(TCP/TLS) (5060), Outbound Proxy, Outbound Proxy Protocol Type (UDP), Outbound Proxy Port(UDP) (5060), Outbound Proxy Port(TCP/TLS) (5060), From Header (Local Domain), PPID (Disable), Local Domain (Disable), Support SIP-T (Disable), Get Callee from (Request-line), Get Caller from (User Name), Register to Remote (No), Incoming SIP Authentication Type (IP Address), Rport (Disable), Dynamic Nat (Disable), Static Nat (Disable), Outgoing Calls Restriction (No), and Incoming Calls Restriction (No). Red boxes and numbers highlight specific fields: a red box with a circled '2' around 'Trunk Name', a red box with a circled '3' around 'Protocol Type', and a red box around 'Remote Address'.

• configuration

Configure SIP Trunk (PBX)

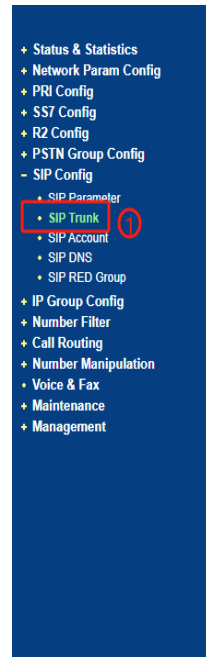
Configure SIP Trunk (IMS)

Configure SIP Account

Configure Static Routing

Route setting

1. Click on **SIP Config - SIP Trunk**
2. Customize trunk name and select network port
3. Fill in the domain, proxy IP and port of IMS, select the protocol
4. From Header: select peer domain
5. Register to Remote: select yes; Outgoing Call Mode: select access



A screenshot of the 'SIP Trunk Modify' form. Red boxes and numbers 2 through 5 highlight specific fields: 2. Trunk Name, 3. Remote Address, 4. From Header, and 5. Register to Remote and Outgoing Call Mode.

SIP Trunk Modify	
Trunk No.	1
2 Trunk Name	GE0
	ims
Remote Address	ims.gx.chinamobile.com
3 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
4 From Header	Peer Domain
PPID	Disable
Local Domain	Disable
Support SIP-T	Disable
Get Callee from	Request-line
Get Caller from	User Name
5 Register to Remote	Yes
Outgoing Call Mode	Access
Redundancy Mode	No
Account Select Mode	Default
Account Auth Mode	Username
Display Name by	Caller
Contact Username by	Default
Remote-Party-ID Username by	Default
Incoming SIP Authentication Type	IP Address
Report	Disable

- configuration

Configure SIP Trunk (PBX)

Configure SIP Trunk (IMS)

Configure SIP Account

Configure Static Routing

Route setting

1. Click on **SIP Config - SIP Account**
2. Customize name and select Trunk no.
3. Fill in username、Authenticate ID and password provided by IMS

SIP Account Add

SIP Account ID	0
Description	863983068898
Binding PSTN Group	None
SIP Trunk No.	1 <ims>
Username	+863983068898
Authenticate ID	+863983068898@ims.gx.chinamobile
Password	*****
Confirm Password	*****
Expire Time	1800 s
Max Calls	65535
Binding SIP Trunk	Disable
Enable Account	Yes

OK Reset Cancel

- configuration

Configure SIP Trunk (PBX)

Configure SIP Trunk (IMS)

Configure SIP Account

Configure Static Routing

Route setting

1. Click on **Network Param Config- Network Config**, View GE0 default gateway

2. Click on **Network Param Config- Static IP Route Table**:

Destination Network: IMS proxy IP address

Subnet Mask: Full mask or network segment mask

Gateway: GE0 default gateway

- + Status & Statistics
- Network Param Config
 - + Network Config
 - + Static IP Routing Table
 - + ACL White List
 - + ACL Control Config
 - + VLAN Config
- + PRI Config
- + SS7 Config
- + R2 Config
- + PSTN Group Config
- + SIP Config
- + IP Group Config
- + Number Filter
- + Call Routing
- + Number Manipulation
- + Voice & Fax
- + Maintenance
- + Management

- + Status & Statistics
- Network Param Config
 - + Network Config
 - + Static IP Routing Table
 - + ACL White List
 - + ACL Control Config
 - + VLAN Config
- + PRI Config
- + SS7 Config

Network Configuration

Service Ethernet Interface(GE1)

☐ Obtain IP address automatically

☒ Use the following IP address

Description: eth0

IP Address: 172.27.10.27

Subnet Mask: 255.255.0.0

Default Gateway: 172.27.1.1

Work Mode: Auto Negotiation

Ethernet Port Bond: Disable

Management Ethernet Interface(GE0)

Description: eth1

IP Address: 192.168.13.4

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.13.1

Work Mode: Auto Negotiation

DNS Server

☐ Obtain DNS server address automatically

☒ DNS Server

Master DNS Server: 114.114.114.114

Secondary DNS Server: 8.8.8.8

Default Gateway

Interface: GE1

System Parameter

Hostname: TG-20E1

Static IP Routing Table

Destination Network	Subnet Mask	Gateway
183.211.6.89	255.255.255.255	192.168.13.1

☐

Add Delete Modify

- configuration

Configure SIP Trunk (PBX)

Configure SIP Trunk (IMS)

Configure SIP Account

Configure Static Routing

Route setting

1. Click on **Call Routing- IP->IP Routing**
2. Customize name
3. Select source and destination SIP trunk

IP->IP Routing Modify

Index: 511
 Description: out
 Source Type: Trunk
 Trunk Type: SIP
 IP Trunk No.: 0 <pbx>
 Callee Prefix: .
 Caller Prefix: .
 Destination Type: Trunk
 Trunk Type: SIP
 IP Trunk No.: 1 <ims>
 Filter Profile ID: 255 <None>

NOTE: '.' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

IP->IP Routing

Index	Description	Trunk Type	Trunk No.	IP Group	Callee Prefix	Caller Prefix	Trunk Type	Destination Trunk No.	Destination IP Group	Filter Profile ID
☐ 510	in	SIP	1 <ims>	---	.	.	SIP	0 <pbx>	---	None
☐ 511	out	SIP	0 <pbx>	---	.	.	SIP	1 <ims>	---	None

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Contents

- 1 Chapter One MTG Advanced Features
- 2 Chapter Two MTG Advanced Configuration
- 3 Chapter Three SNMP

Chapter Three

SNMP

02

SNMP

• MTG Configuration

1. Click on **Management- SNMP Parameter**
2. Select SNMP Version
3. Configure Community
4. Configure trap



A screenshot of the 'SNMP Parameter' configuration page. The page has a blue header and a light gray background. Red circles with numbers 2, 3, and 4 are placed over specific sections. A red box highlights the 'SNMP Version' dropdown menu. Another red box highlights the 'Community Configuration' section. A third red box highlights the 'Trap Configuration' section. The 'SNMP Enable' checkbox is checked. The 'SNMP Listen Port' is set to 161. The 'Community' is set to 'public' and the 'Source' is set to 'default'. The 'TrapFlag' is set to 'v2c', 'TrapIP' is '172.28.7.49', 'TrapPort' is '162', and 'TrapCommunity' is 'public'. The 'Login' and 'Configuration' options are set to 'Disable'. A 'Save' button is at the bottom right.

SNMP Parameter

SNMP Enable ☒ Yes ☐ No

2 **SNMP Version** v2c

Config Mode Typical

SNMP Listen Port 161
Notice: SNMP default listen port is 161. The device must restart to take effect after changing port!

3 **Community Configuration**

Community Source

1 public default
Notice: default value of source is default. If other value, please input IP (eg: 192.168.1.1)

4 **Trap Configuration**

TrapFlag TrapIP TrapPort TrapCommunity

1st v2c 172.28.7.49 162 public

Login Disable

Configuration Disable

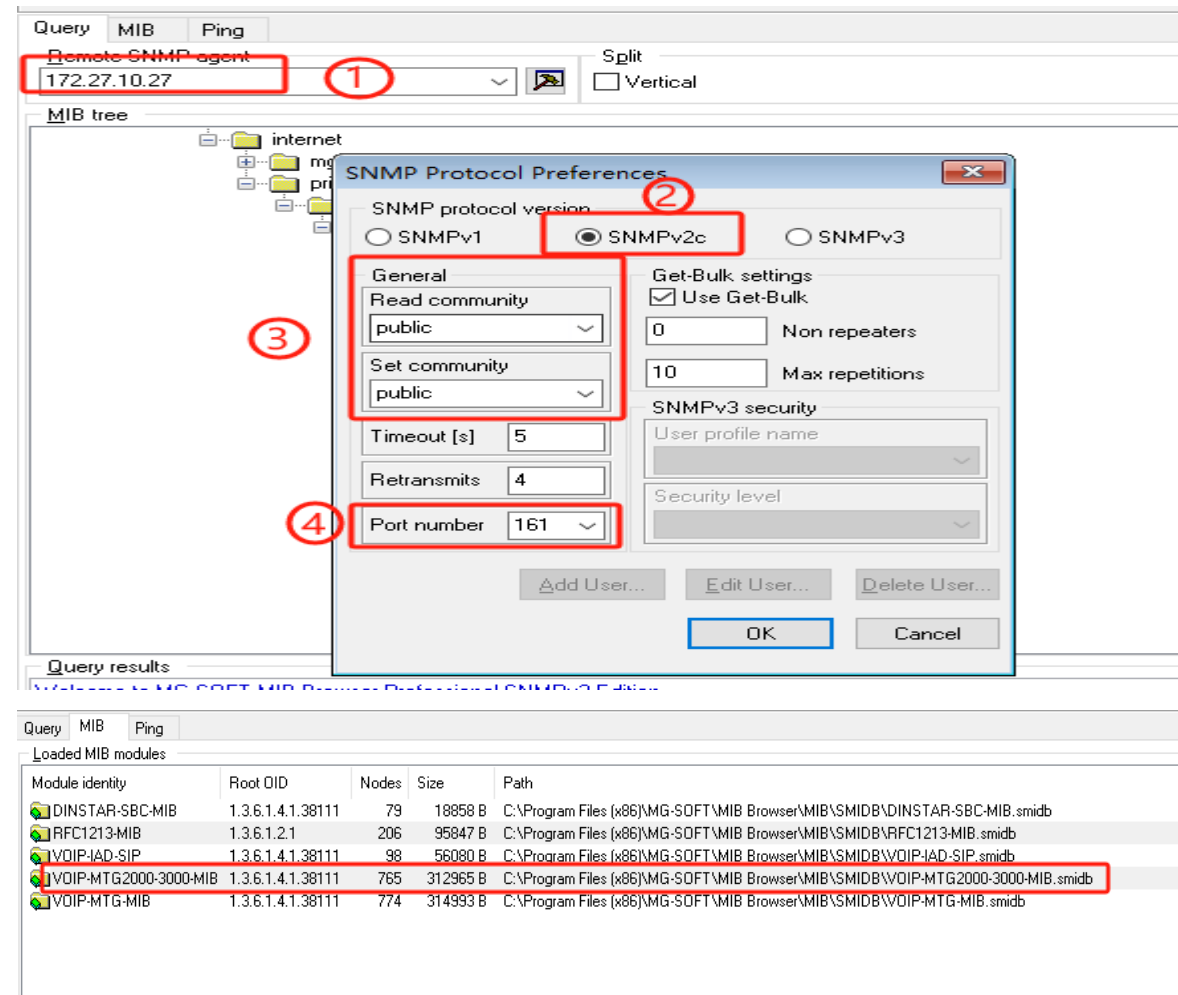
Save

1. The only one is effective between v1 and v2c.

SNMP

- SNMP Server Configuration

1. Configure the IP address of MTG
2. Select SNMP Version
3. Configure Community
4. Configure SNMP listen port of MTG
5. Import the mib file of MTG

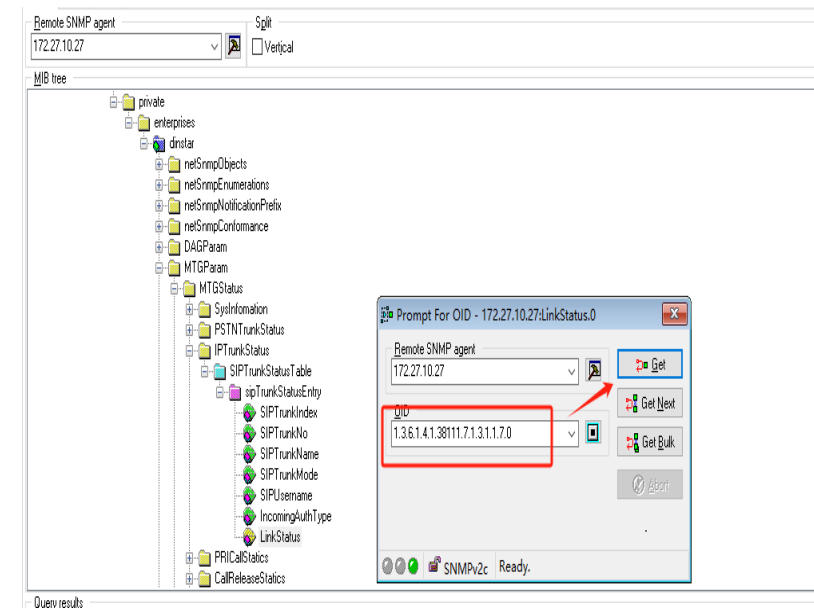
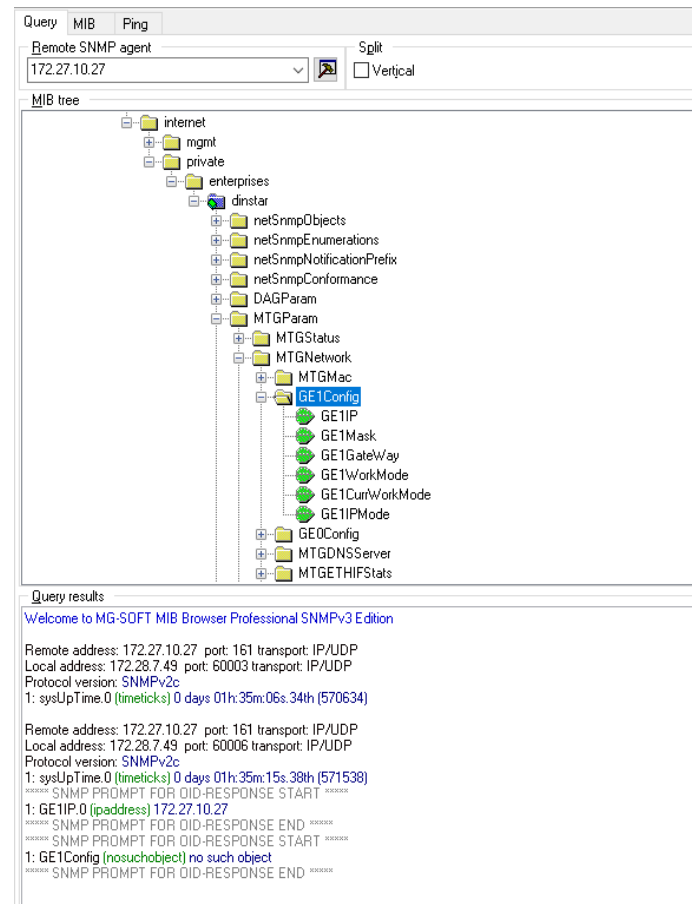


SNMP

- SNMP Server Retrieves MTG Information

Method 1 :Find the information you need to obtain on the MIB tree, right-click on GET, and results will display the obtained information

Method 2:Knowing the OID of the corresponding information, one can directly query it





THANKS



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