Relay Proxy

User Guide

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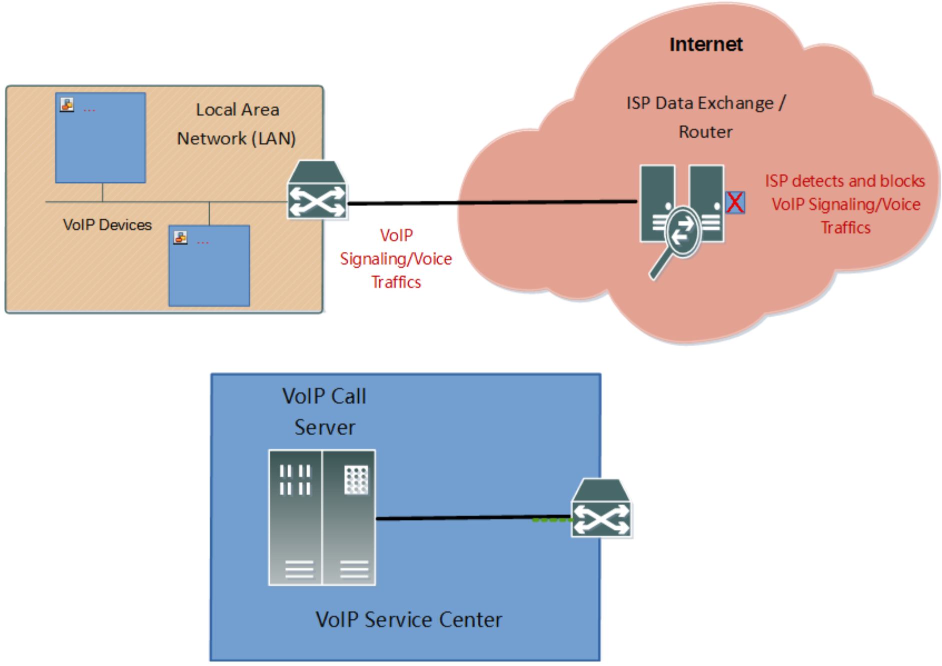
Relay Proxy

Relay Proxy is a server software developed by Hybertone Technology. Its main purposed is to facilitate the deployment of Hybertone's gateway in network environment that does not support NAT or blocks VoIP traffics.

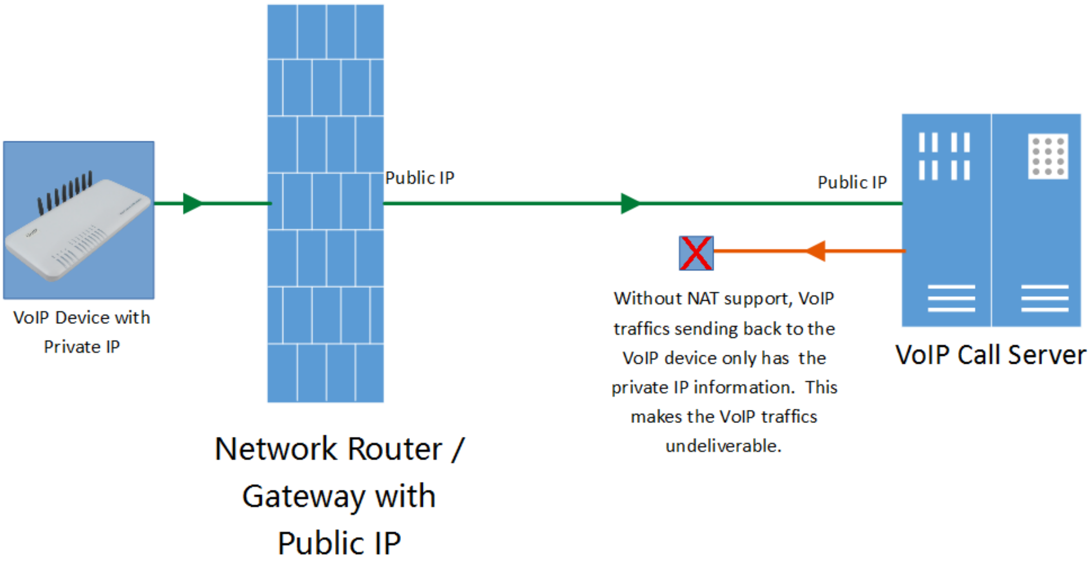
Two common issues encountered with VoIP deployment

Two major problems are encountered today with VoIP traffics.

1. ISP is blocking VoIP traffics in order to protect the traditional telephone services.
   The rapid increase in the internet and intranet data bandwidth in the last two decades has triggered the rapid development and deployment of VoIP services. This has greatly reduced the revenues of the traditional long distance services. As a result, ISPs in certain countries are blockings VoIP traffics as shown in the diagram below.

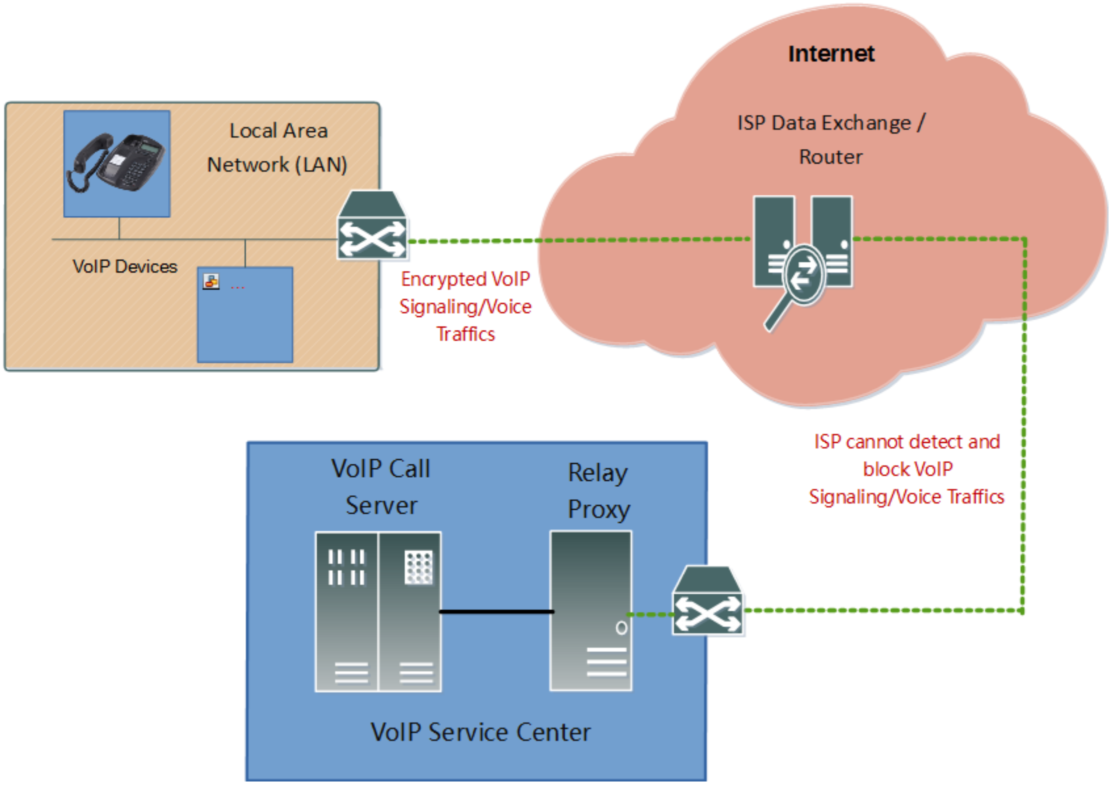


2. VoIP traffics are not getting through due to the inability to support NAT in the Call Server when VoIP devices are installed behind a network router as shown in the diagram below. Some VoIP Call Servers only support call signaling NAT. This means that the VoIP device installed behind NAT can registered and make / receive calls. However, when a call is established, only audio stream generated from the VoIP device can reach its destination. As a result, the VoIP device end cannot hear any audio at all.

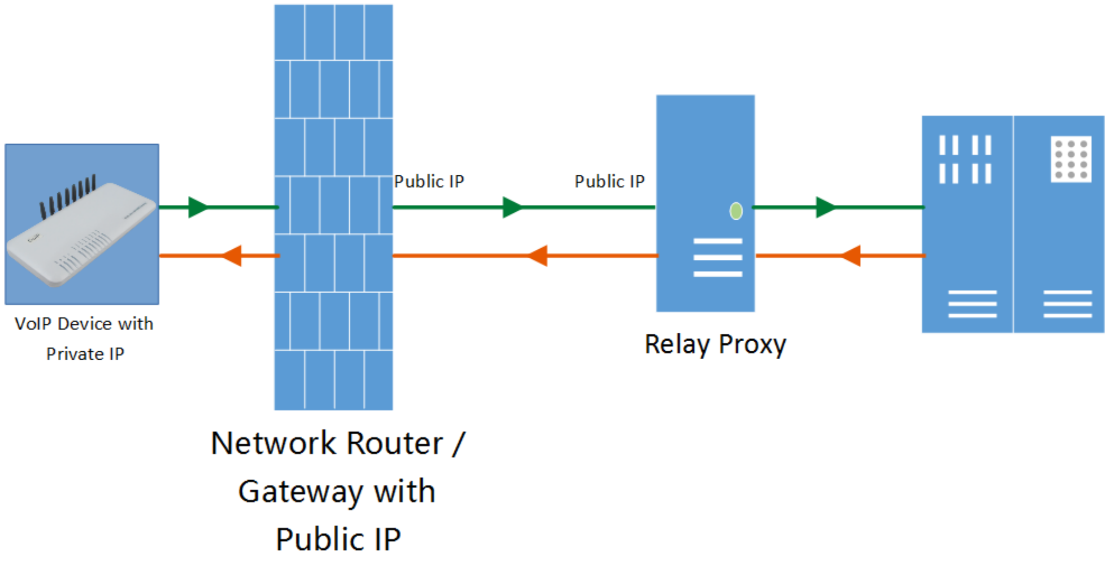


How can Relay Proxy solve both issues?

1. Relay Proxy establishes a virtual tunnel for both VoIP signaling and voice traffics with a very small increase in the data bandwidth. This makes it more difficult for ISPs to block such data traffics. To even protect the data traffics more, Relay Proxy also offers data encryption for both VoIP signaling and voice traffics. The data traffics between Relay Proxy and VoIP Call Server are resumed to the standard VoIP data format. Therefore, they must be installed in the same LAN network or an ISP network that does not block VoIP traffics.



2. Relay Proxy solves the NAT issue by re-routing data traffics from VoIP devices to the VoIP Call Server and vice versa as shown in the diagram below. All data packets now have the proper destination IP, so that they can be delivered. In general, if SIP registration is not successful or one way audio occurs when a call is established, you can try to solve the problems with Relay Proxy.



Installing Relay Proxy

1. PC hardware preparation

Relay Proxy only supports Linux OS and it should run on most PC hardware today with good performance. Relay Proxy has been tested for compatibility in the following Linux platforms.

1) RedHat
2) CentOS
3) Debian
4) Ubuntu

For 64-bit OS, the following extended packages should be installed by issuing the command listed below.

a) RedHat / CentOS
   yum install -y glibc.i686   zlib.i686   krb5-libs.i686

b) Debian / Ubuntu
   dpkg --add-architecture i386
   apt-get update
   apt-get install lib32z1-dev
   apt-get install libgssapi-krb5-2:i386

2. Software Installation and Execution

1) Login to the root directory.
2) Type wget http://www.hybervoice.com/update/relay_install-2.068.sh to download the Proxy Server Installation package
3) Type chmod 744   relay_install-2.068.sh to enable the installation property

4) Type ./relay_install-2.068.sh to execute the installation script

After the installation is completed

5) Type /root/relay/run_relaysrv to execute the Relay Proxy

6) Type /root/relay/run_sqlwebd to execute the Relay Proxy Web Interface

Please note that a Relay Proxy startup script is installed to execute both Relay Proxy and its web interface when the hardware boots up. For Dubian / Ubuntu platform, the startup script may not work properly. If this occurs, please delete the line “exit 0”in the document /etc/rc.local.

3. Other commands

1) Killall relaysrv –type this to terminate the Relay Proxy.

2) Killall sqlwebd –type this to terminate the Relay Proxy web interface.

4. Default Ports

Relay Proxy uses the followings ports for both TCP and UDP communications. Please configure the server firewall accordingly.

TCP21080，1701，8089

UDP1701，5000~60000

Note: Please contact technical Isupport at support@hybertone.com for assistance if needed.

Configuring Relay Proxy

1. To configure the Relay Proxy, you need to access its web interface shown below.
   - Type http://<server hardware IP>:8089
   - Enter “admin” as the login ID
   - Enter “admin” as the login password

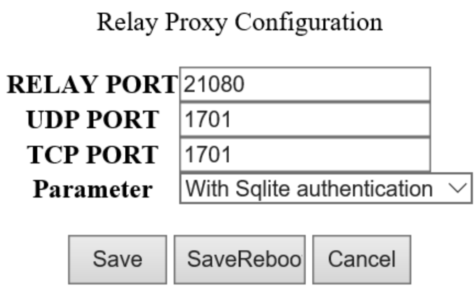
Relay Proxy configuration

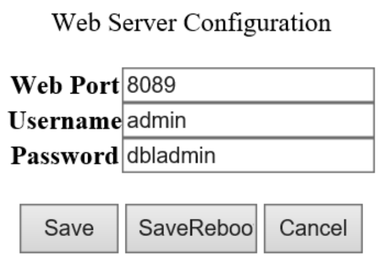
Relay Proxy Manage v1.0

Agent	Username		
dbl	dbl	Delete	Modify

Add

2. Click on Relay Proxy configuration to configure the Relay Proxy.





Relay Proxy Configuration

- Modify the Relay Port, UDP Port, TCP Port if needed.
- Parameters:

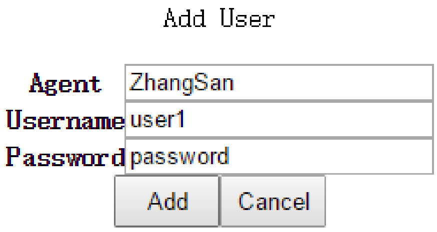
1) Default – No authentication
2) Sql Authentication–Reserved for testing.
3) Sqlite Authentication–Using SQLite database for authentication (choose this one by default)

4) Listen Localhost – Reserved for other use.

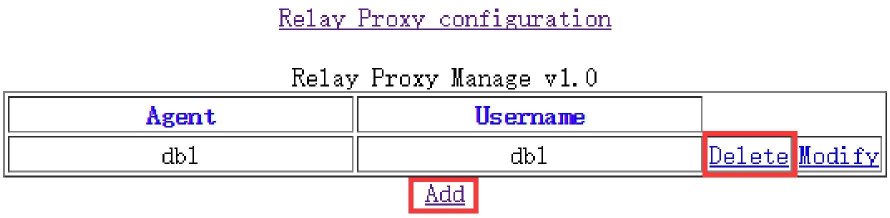
- Press Save to save the changes (not effective yet).
- Press Save to save and make the changes effective immediately

Browser displays a warning message of not be able to access the webpage.Please wait i10 seconds and then reload the webpage.

3. Click Add to add a new login account. Enter the fields as required and then click “Add” to complete. Agent is used to classify the user accounts only and it is not required for GoIP configuration. In addition, the same Username is allowed for multiple logins. This means that you can use the same account to configure all your devices to login to the same Relay



The default test account is “dbl”. To delete an unwanted account, please click Delete (on the right hand side of the account name) to delete the corresponding account.



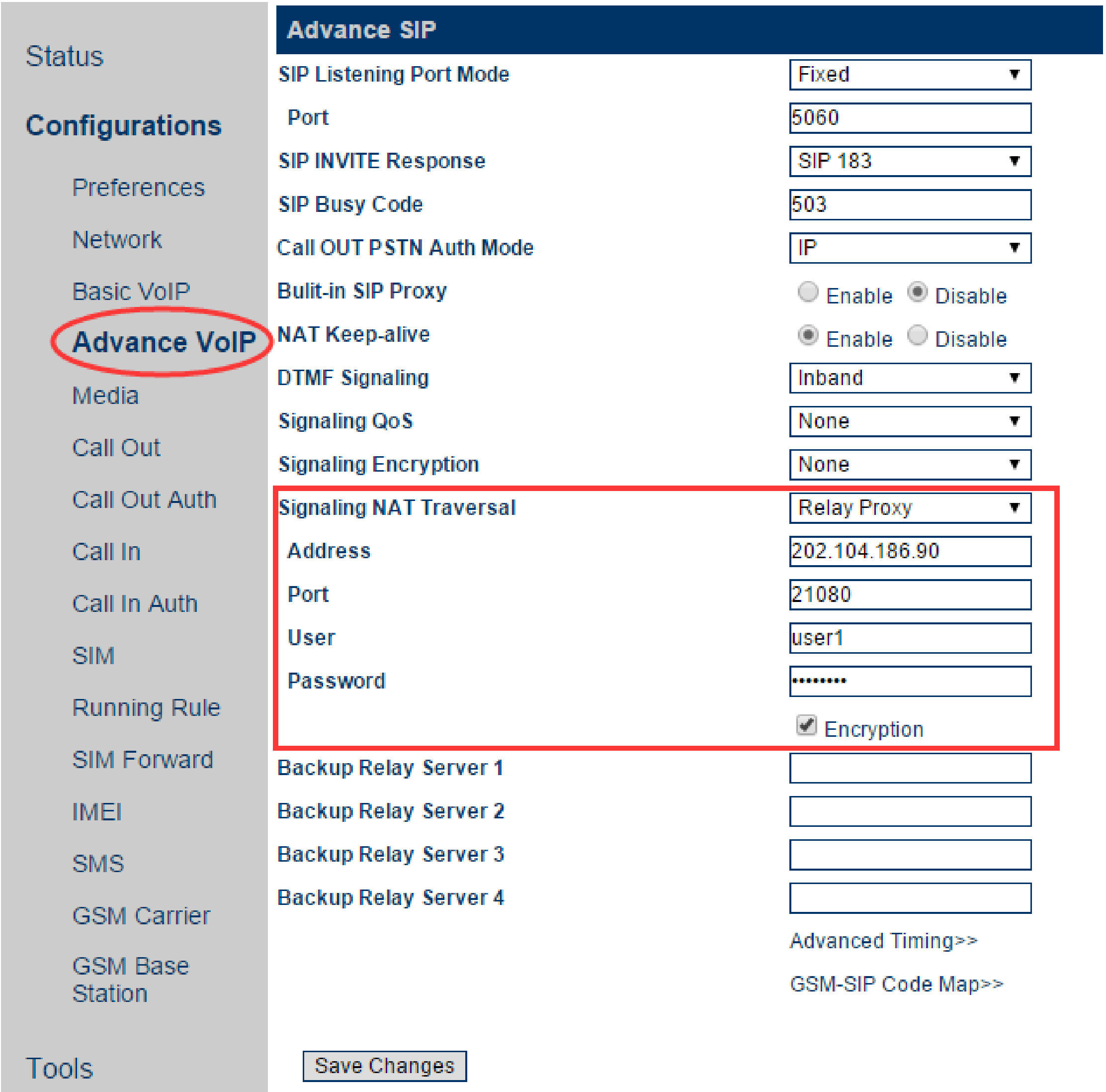
Configuring Hybertone’s VoIP devices

Please note that all VoIP devices developed by Hybertone supports Relay Proxy. Either or both VoIP Signaling data and VoIP voice stream can be configured to connect to Relay Proxy.

1. Configuring VoIP Signaling data to use Relay Proxy
   VoIP signaling is referring to the IP protocol used for VoIP Call establishment and management. Depending on your device, the location for its configuration varies. For GoIP, it is located under the “Advanced VoIP” as shown in the two figures below.

   Locate the parameter “Signaling NAT Transversal”andselect “Relay Proxy”. Then the following additional parameters are displayed.

- Address –enter the IP address or domain name of the Relay Proxy
- Port –enter the **Relay Port**as configured in the Relay Proxy
- User –enter the usernamethat iscreated for your device
- Password –enter the password that is defined for the username
- Click on the check box to enable encryption if it is required. Please note that additional network bandwidth is required if encryption is enabled.


Status
Configurations
Preferences
Network
Basic VoIP
Advance VoIP
Media
Call Out
Call Out Auth
Call In
Call In Auth
SIM
Running Rule
SIM Forward
IMEI
SMS
GSM Carrier
GSM Base Station
Tools
Advance SIP
SIP Listening Port Mode
Fixed
Port
5060
SIP INVITE Response
SIP 183
SIP Busy Code
503
Call OUT PSTN Auth Mode
IP
Bulit-in SIP Proxy
Enable Disable
NAT Keep-alive
Enable Disable
DTMF Signaling
Inband
Signaling QoS
None
Signaling Encryption
None
Signaling NAT Traversal
Relay Proxy
Address
202.104.186.90
Port
21080
User
user1
Password
Encryption
Backup Relay Server 1
Backup Relay Server 2
Backup Relay Server 3
Backup Relay Server 4
Advanced Timing>>
GSM-SIP Code Map>>
Save Changes

Call Settings

Endpoint Type	SIP Phone
Config Mode	Config by Line
(•) Line 1 ( ) Line 2 ( ) Line 3 ( ) Line 4	
Phone Number	149
Phone Number 2	
Display Name	
SIP Proxy	192.168.2.2
SIP Registrar Server	192.168.2.2
Register Expiry(s)	60
Outbound Proxy	
Home Domain	
Authentication ID	149
Password	•••
Dial Plan	18:+9
Call Forward Type	Not Forward
Call Forward Number	
Backup Server	( ) Enable (•) Disable
	Fax Line>>

Advanced Settings<<

SIP Local Port Mode	Fixed
Signaling Port	5060
Bulit-in SIP Proxy	( ) Enable (•) Disable
NAT Keep-alive	(•) Enable ( ) Disable
Virtual Ringback	( ) Enable (•) Disable
Reigster Mode	Mode 1
	Advanced Timing>>
DTMF Signaling	Outband
Outband DTMF type	RFC 2833
RTP Payload Type	101
Signaling QoS	None
Signaling Encryption	None
Signaling NAT Traversal	Relay Proxy
Address	202.104.186.90
Port	21080
User	user1
Password	••••••••
	[x] Encryption
Backup Relay Server 1	
Backup Relay Server 2	
Backup Relay Server 3	
Backup Relay Server 4	

Media Settings<<

RTP Port Range	16384 - 32768
PacketLength(ms)	30
Jitter Buffer	Fixed
Delay(ms)	60
Media QoS	None
Media Encryption	None
	[x] Symmetric RTP
Media NAT Traversal	Relay Proxy
Address	202.104.186.90
Port	21080
User Name	user1
Password	••••••••
	[x] Encryption
Relay Mode	1
Backup Relay Server 1	
Backup Relay Server 2	
Backup Relay Server 3	
Backup Relay Server 4	

Audio Codec Preference>>

2. Configuring VoIP voice stream to use Relay Proxy

When a VoIP call is established, voice stream are sent between the two VoIP devices. To configure the voice stream to use the Relay Proxy, please access your device webpage and then look for the “Media” as shown in the figure below or “Media Settings” as shown in the figure in the previous page.

Locate the parameter “Media NAT Transversal” and select “Relay Proxy”. Then the following additional parameters are displayed.

- Address –enter the IP address or domain name of the Relay Proxy
- Port –enter the **Relay Port** as configured in the Relay Proxy
- User –enter the username that is created for your device
- Password –enter the password that is defined for the username
- Click on the check box to enable encryption if it is required. Please note that additional network bandwidth is required if encryption is enabled.