SIP ALG – is a parameter that is generally enabled on most commercial router because it helps to resolve NAT related problems. However, this parameter can be very harmful and can actually stop SIP Trunks from working correctly.

There are several different ways to counteract this type of failure such as STUN, TURN, ICE, and if there is a server you could use RptProxy or MediaProxy. However, since ALG is usually on the client side, these solutions may not work.

An Applications-Level Gateway (ALG) understands the protocol used by the specific applications that it supports (in this case SIP) and does a protocol packet-inspection of traffic through it. A NAT router with a built-in SIP ALG can re-write information within the SIP messages (SIP headers and SDP body) making signaling and audio traffic between the client behind NAT and the SIP endpoint possible.

SIP ALG example

- Caller behind NAT with private IP 192.168.10.25 (Master) 192.168.10.26 (Slave).
- Caller router public IP 192.0.2.200
- SIP proxy in Internet with domain "example.com".

INVITE from the LAN client (with private IP)

```
INVITE sip:ip1@192.168.10.26:6070 SIP/2.0
From: "Paul_Kansas"<sip:102@192.168.10.25>;tag=c0a80a19-13c4500ecfde
To: <sip:ip1@192.168.10.26>
Call-ID: 80da32c8-c0a80a19-13c4-500ecfde-2d4992d8-15a5@192.168.10.25
CSeq: 1 INVITE
Via: SIP/2.0/UDP 192.168.10.25:5060;branch=z9hG4bK-500ecfde-2d4992d8-2188
Max-Forwards: 70
Supported: 100rel,replaces
Trunk-Dial: line=ip1; cid=19132054979
Allow: INVITE,ACK,CANCEL,BYE,NOTIFY,REFER,OPTIONS,INFO,SUBSCRIBE,PRACK,UPDATE
User-Agent: NK-V1.6.8.1_Xblue
Contact: "Paul_Kansas" <sip:102@192.168.10.25:5060>
Content-Type: application/sdp
Content-Length: 298

v=0
o=Paul_Kansas 1343147998 1343147998 IN IP4 192.168.10.25
s=_
c=IN IP4 192.168.10.25
t=0 0
m=audio 10064 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,16
a=silenceSupp:off
a=ptime:20
a=sendrecv
```

Note that text in red needs to be fixed before it arrives to the proxy (in case our proxy doesn't provide us a NAT server solution). If not, the proxy reply will not arrive at the client (caller):

1. The caller couldn't receive in-dialog/sequential messages (ACK for the INVITE, BYE, REFER, re-INVITE...) since the address in "Contact" is not routable outside their network.
2. Unidirectional audio since the caller told the callee to send audio to a non-routable address and port (so the caller won't hear the callee).

The text in blue doesn't need to be fixed since SIP already handles it (the server adds the parameter "received=REAL_SOURCE_IP" to the "Via" header and sends the replies to that address). Anyway some ALG implementations also change this value.

The same INVITE modified by the ALG router:

SIP/2.0 183 Session Progress
From: "Paul_Kansas"<sip:102@192.168.10.25>;tag=c0a80a19-13c45010085c
To: <sip:ip1@192.168.10.26>;tag=c0a80a1a-17b65010085d
Call-ID: 80da3580-c0a80a19-13c4-5010085c-320e377e-3215@192.168.10.25
CSeq: 1 INVITE
Via: SIP/2.0/UDP 192.168.10.25:5060;branch=z9hG4bK-5010085c-320e377e-7cb8
Supported: 100rel,replaces
CID-Info: Trunk Call; line=0; actnum=19132054979; ringbackdisplayname=; privacy=0;
Contact: <sip:ip1@192.168.10.26:6070>
Content-Type: application/sdp
Content-Length: 222

v=0
o=ip1 2890844527 2890844527 IN IP4 208.93.224.231
s=SIP
c=IN IP4 208.93.224.231
t=0 0
m=audio 26704 RTP/AVP 0 101
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=sendrec

The ALG has fixed the NAT related problem by:

- Replacing IP in "Via" header with the public IP and port.
- Replacing "Contact" with mapped public IP and port.
- Replacing SDP media address with public IP and port.

SIP ALG problems

The main problem is the poor implementation at SIP protocol level of most commercial routers and the fact that this technology is just useful for outgoing calls, but not for incoming calls:

- Lack of incoming calls: When a SIP User Agent (SIP UA or UA), in our case the X-44 telephone is switched on it sends a REGISTER to the proxy in order to be localizable and receive incoming calls, which is modified by the ALG feature. If it is not the user wouldn't be reachable by the proxy, since it indicates a private IP in REGISTER "Contact" header and not a public one. A SIP ALG router rewrites the REGISTER request so the proxy doesn't detect the NAT and doesn't maintain the keep-alive (so incoming calls will be not possible).

- Breaking SIP signaling: Many commonly used routers come standard with SIP ALG enabled, which will modify the SIP headers and the SDP body incorrectly, or completely rewrite the headers, essentially breaking SIP and making communication impossible. Which means that many SIP ALG routers corrupt the SIP messages by modifying them (i.e. missed semi-colon ";" in header parameters) or rewriting incorrect port values greater than 65536.

List of routers with SIP ALG enabled

The following is a list containing SIP ALG router models, their issues and how to disable SIP ALG (enabled by default in most of the cases). Please add to the list as you find additional information.

Cisco

Models: 800 Series

To disable the NAT services for SIP in IOS, just run these commands:

no ip nat service sip tcp port 5060
no ip nat service sip udp port 5060

Dlink

Models: DIR-655

Fortinet

Models: All models come with SIP Helper enabled by default

To disable SIP helper:
~# telnet firewall
config system settings
set sip-helper disable
set sip-nat-trace disable
end

config system session-helper
delete 12
end

The preferred solution is to configure the SIP ALG. Policies that use the SIP ALG will not use SIP helper. Full documentation at http://docs.fortinet.com then pick FortiOS for the version on your device, then VoIP solutions: SIP.

Juniper / Netscreen

Models: SSG Series

To disable SIP ALG:
In the Web interface: Security -> ALG

Linksys

Models: WRV200, WRT610N
NAT type: Symmetrical
Issues:

The ALG replaces the private address in "Call-ID" header (not needed at all). Some phones (as Linksys with latest firmware) encode the "Call-ID" value in the "Refer-To" header (by escaping the dots) so the private IP appearing there is not replaced with the public IP. This causes that the call transfer fails since the proxy/PBX/endpoint will not recognize the dialog info.

To disable SIP ALG: ToDo no ALG related options found via web and telnet. No idea of how to disable it.
To disable SIP ALG on WRT610N: Web Interface: Administration, Management, under side heading 'Advanced Features' SIP ALG, can be disabled.

Motorola

Models: SBG6580

(SurfBoard Extreme Wireless Cable Modem Gateway)

No Registration possible behind NAT as the device changes Call-ID and causes the responses to be discarded by SIP clients/ATAs

No Solution at this time (SIP ALG, called SIP Pass Through, can not be disabled) .

Must disable NAT and put the device in bridge mode. (See this guide)

Models: 3300

Netopia has SIP Passthrough "on" by default and will not work with voice over ip services properly unless SIP ALG is disabled. (Or SIP ALG in newer models) In order to work with voice over ip, you must turn SIP Passthrough (SIP ALG) off.

The setting is not in the web gui. You have to telnet into the router, then type. Each line is a new command.

```
configure
set ip sip-passthrough off
save
exit
Restart
```

Models 4622xlt

Telnet into the device Use the arrow keys to highlight 'quick menu' and press "ctrl-n" That brings you to the console.

type:

```
ip nat alg sip enable no
save
exit
restart
```

Netgear

Models: WGR614v9 Wireless-G Router, DGN2000 Wireless-N ADSL2+ Modem Router
Firmware V1.0.18_8.0.9NA
To disable SIP ALG:
Click on the Advanced Tab → go to Wan Setup → uncheck the "Disable SIP ALG"

Peplink Multi-WAN Routers (Peplink Multi-WAN routers)

Models: All multi-WAN models
To disable SIP ALG, go to http://<router.LAN.IP>/cgi-bin/MANGA/support.cgi
Click the "Disable" button under "SIP ALG Support"
Issues:

- I'm not aware of any SIP ALG issues, but if you just want to turn it off, here you go.

SMC

Models: ToDo
NAT type: No symmetrical
Issues:

The ALG doesn't replace the private address in "Call-ID" header (that is correct) but it does replace the "call-id" value in "Refer-To" header so SIP transfer is broken.

To disable SIP ALG: ToDo no ALG related options found via web and telnet. No idea of how to dissable it.

SpeedTouch

Models: **ST530 v6** *(firmware >= 5.4.0.13)* comes with SIP ALG enabled by default.
NAT type: symmetrical
Issues:

- No incoming calls.
  It replaces the private IP appearing in SIP headers with the public IP using a dumb text replacement. If for example the private IP appears in the "Call-ID" it replaces it too (that it's completely unnecessary).

To disable SIP ALG:
~# telnet router
-> connection unbind application=SIP port=5060
-> saveall

Zyxel

Models: **660 family** comes with SIP ALG enabed by default.
NAT type: symmetrical
Issues:

- No incoming calls.
- SIP protocol broken making 50% of outgoing calls impossible because the wrong values are inserted into SIP headers.

To disable SIP ALG:
~# telnet router
Menu option "24. System Maintenance".
Menu option "8. Command Interpreter Mode".
ip nat service sip active 0