



One-way Audio

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One-way Audio

Identifying the cause of one way speech is notoriously difficult. While the normal culprit is SIP ALG, incompatible codecs, firewall blocking will all contribute to one-way audio. Our *38 test is a simple way of quickly identifying whether the issue is on or external to our service.

*38 echo test

To confirm one-way audio dial *38 from any phone on your account. If you can hear yourself speak (the echo) you DON'T have one-way audio issues caused by a codec incompatibility, SIP ALG or firewall issues.

SIP ALG issues

SIP ALG was a gallant attempt designed to improve media and signaling but more often that not breaks the SIP messaging causing one-way audio. If you can access your router, disable SIP ALG. Alternately, changing the transport from UDP to TLS encrypts the SIP message and will prevent SIP ALG one way audio.

Incompatible codecs

The example below, the SDP media mismatch in the SIP signaling could cause one-way audio. Device A offered 4 codecs - PCMU, PCMA, G729 and telephone-event. However Device B only accepted G729 and PCMU. The media port numbers also do not match - A offered 6000 but B answered 5000.

This type of mismatch means the call may connect, but only G729 voice path will work properly between both ends. The RTP packets containing other media types would not reach the desired destination ports. To avoid this, all participating devices must be configured consistently and support compatible codecs. The offer/answer process must result in matching codec and port selections on both sides. Any mismatch can lead to one-way audio problems. We accept, G711 a/u-law, G.729, G722, iLBC.

```
m=audio 6000 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

Offer SDP from Device A:

m=audio 5000 RTP/AVP 18 0

a=rtpmap:18 G729/8000

a=rtpmap:0 PCMU/8000

Answer SDP from Device B

Firewall or NAT

Strict outbound firewall rules blocking the RTP media stream from entering back into the internal network from the outside will be an issue. Outbound audio works, but return audio gets blocked. Firewalls and NAT require appropriate visibility and configuration tailored to handle the ports, IPs and bidirectional media flows involved in VoIP/SIP calls.