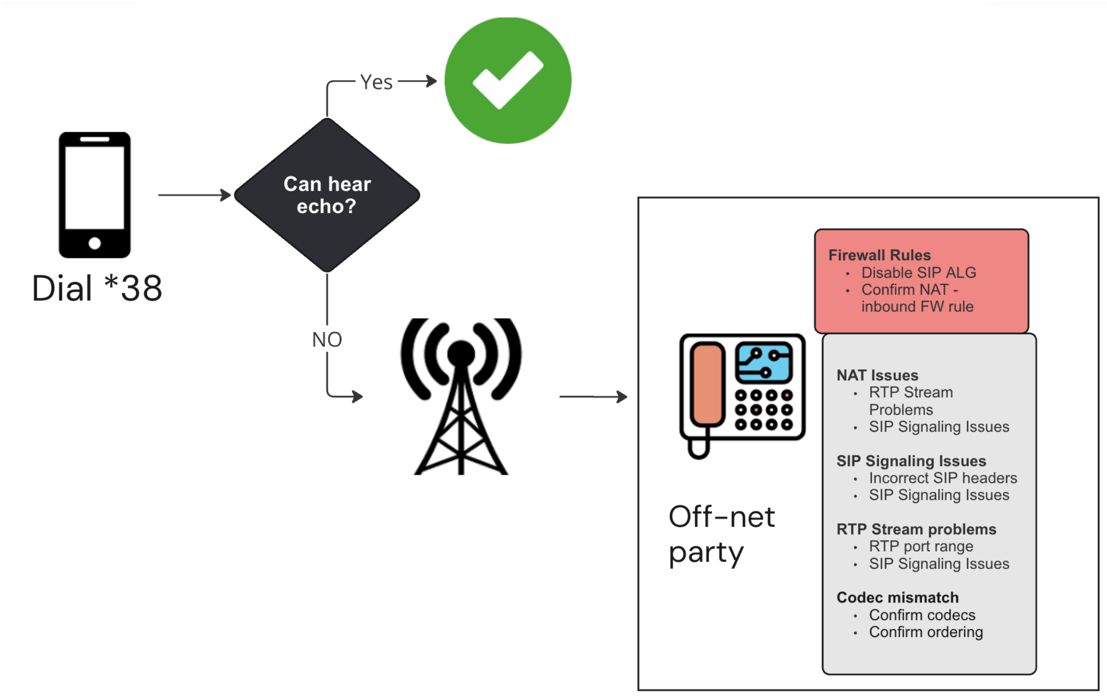
One way audio

Mike Johnstone - 2024-10-04 - Audio, Call Quality and Network Search

One-way Audio

Your browser does not support the audio element.

Our *38 echo test is a useful first step to help isolate the direction of the one-way audio, crucial for helping resolve any one-way speech issue.



*38 echo test

After dialing *38, if you can hear yourself speaking, there is no compatibility issue between your phone and our network.

Other

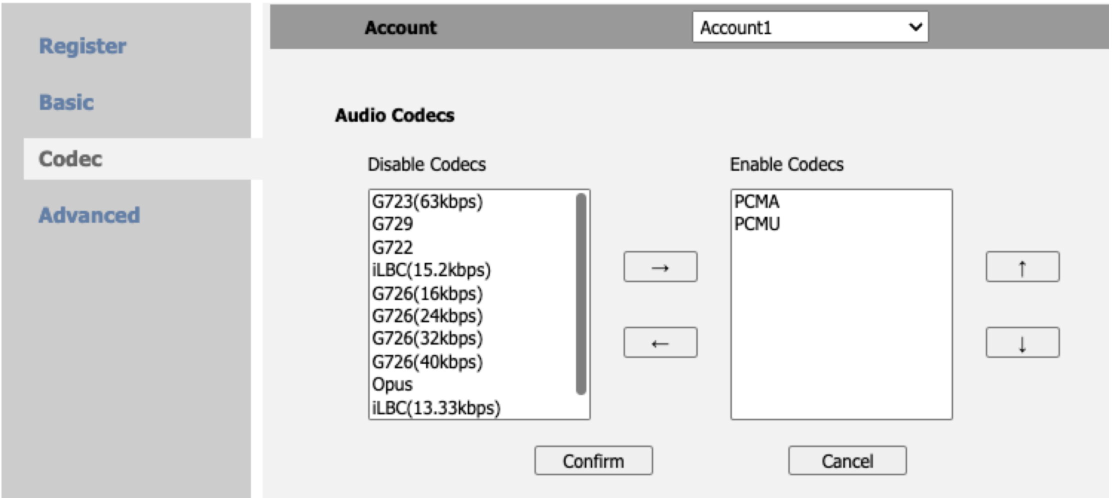
1. Incompatible codecs / Codec ordering

PCMA and PCMU are both audio codecs used for encoding VoIP (Voice over IP) systems. While they are both part of the G.711 standard, they differ in terms of the regions.

- PCMA (G.711 A-law): Primarily used in Europe, Australia most of the rest of the world, and international connections.
- PCMU (G.711 a-law): Primarily used in North America and Japan

In the example below we have disabled all other codecs and set the PCMA as priority for Australia. If in the US set PCMU as priority.

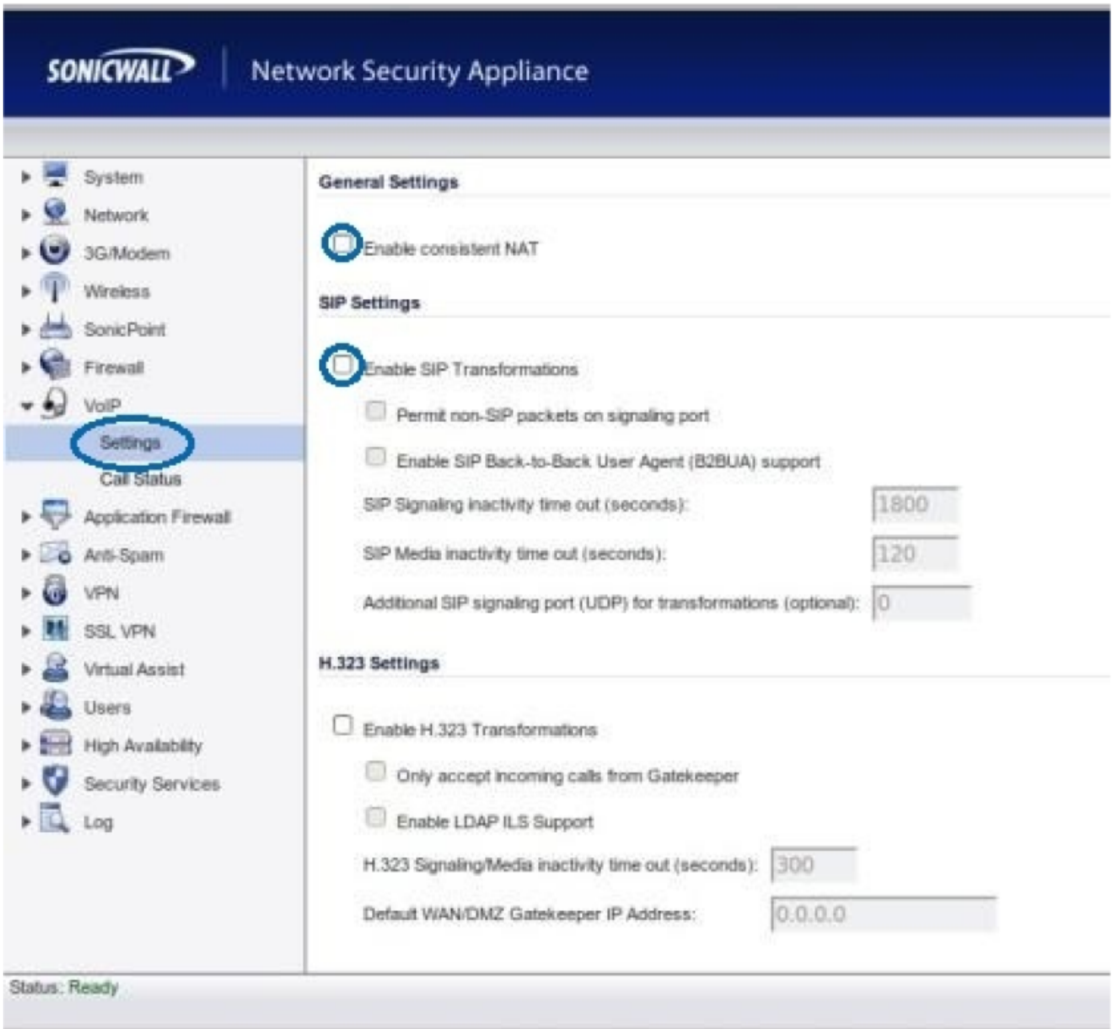
See also [Yealink - Change codec order]



2. SIP ALG issues

SIP ALG was designed to improve media and signaling but is a common cause of 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.

The following Sonicwall is a common example where the default is Enable SIP Transformations. In any one way speech issue we always recommend disabling any settings that implement **SIP ALG** or **SIP Transformations**.



3. Session Description Protocol mismatch

The example below, the SDP media mismatch will 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.

Offer SDP from Device A:

```
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
```

Answer SDP from Device B:

```
m=audio 5000 RTP/AVP 18 0
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
```

4. 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.