VOIP TROUBLESHOOTING 2026

How to Troubleshoot VoIP & SIP Problems in Australia

Choppy audio, one-way calls, dropped calls, and registration failures, the real causes and the practical fixes. Plus why a managed Australian cloud platform makes most of this pain disappear for good.

Almost every VoIP and SIP problem traces back to a short list of causes on your local network. **Choppy or robotic audio** is jitter and packet loss, fix it with a wired connection, QoS, and enough upload bandwidth. **One-way or no audio** is nearly always NAT, a firewall, or (very often) **SIP ALG** on the router, turn SIP ALG off first. **Dropped and failed calls plus registration errors** come from NAT timeouts, registration timing, or DNS. **Latency and echo** come from long network paths and hardware, and **poor Wi-Fi quality** is best solved by going wired. A managed platform removes most of this: with no self-run SIP trunks or PBX, a network the provider owns, Australian hosting, managed media and QoS, and automatic failover, whole categories of problem simply vanish.

The short version

Symptom	Likely Cause	Practical Fix
Choppy, robotic, or garbled audio	Jitter & packet loss; voice competing for bandwidth	Go wired; enable QoS; add upload headroom; raise jitter buffer
Calls drop after ~30. 60 seconds	SIP ALG or a short NAT / session timeout	Disable SIP ALG; lengthen the router's UDP timeout
Phone won't register / loses registration	Registration interval, DNS, credentials, or SIP ALG	Fix the interval & DNS; verify credentials; disable SIP ALG
Echo (you hear yourself)	Acoustic feedback or hardware; sometimes latency	Lower speaker volume; use a headset; update firmware
Long delay / talk-over	High latency, often an overseas media path	Use local Australian hosting; reduce hops; go wired
Fine on cable, bad on Wi-Fi	Wi-Fi interference, congestion, roaming	Use Ethernet; 5 GHz; enable WMM; move closer to the AP
Audio cuts in and out under load	Intermittent loss or bufferbloat on a saturated link	QoS with buffer control; check the line; cap heavy uploads
No audio only with certain callers	Codec mismatch between endpoints	Align codecs (e.g. G.711 / Opus); let the provider negotiate

Key points

- Stop Debugging. Start Calling.

Keep the two halves straight

SIP (signalling) sets up, rings, answers, and tears down the call. If SIP fails, the phone will not register or the call will not connect at all.

Questions the full guide answers

- [ ] Why does my VoIP audio sound choppy or robotic?
- [ ] Why can I only hear one side of a VoIP call (one-way audio)?
- [ ] What is SIP ALG and should I disable it?
- [ ] Why do my VoIP calls keep dropping or failing to register?
- [ ] How much internet speed do I need for good VoIP call quality?
- [ ] Why is my VoIP call quality worse on Wi-Fi?
- [ ] Can switching to a managed cloud phone system fix my VoIP problems for good?

This is the summary. The guide has the detail.

Every point above is worked through in full on our site, with the Australian context, the numbers and the exceptions. If you would rather talk it through, call **1300 663 222** or visit **vocphone.com/about/contact-us**.