VOIP TROUBLESHOOTING 2026

How to Troubleshoot VoIP & SIP Problems (2026 Guide)

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

Most VoIP and SIP problems trace back to a handful 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 almost always NAT, a firewall, or (very often) **SIP ALG** on your router, disable SIP ALG first. **Dropped or failed calls and registration failures** come from NAT timeouts, SIP 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 cloud platform removes most of this: with no self-managed SIP trunks or PBX, carrier-grade Australian hosting, managed media and QoS, automatic failover, and 24-hour local support that fixes issues for you, whole categories of problem simply disappear.

The short version

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

Key points

- Let Us Handle the Hard Part

Remember the Two Halves

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 881 662** or visit **unidenvoice.com/get-started**.