CLOUDSWITCHED

FREE RESOURCE — CHECKLIST

VoIP Call Quality Troubleshooting Checklist

Diagnose and resolve VoIP call quality issues systematically — audio symptoms, network diagnostics, QoS, hardware checks, platform configuration and escalation. Work top to bottom, score each diagnostic area out of 10, and target anything scoring below 6.

6 DIAGNOSTIC AREAS	**49** CHECKS TO PERFORM	**/10** SCORE EACH SECTION	**Fillable** TICK & TYPE IN ANY VIEWER

AUDIO SYMPTOMS · NETWORK · QOS · HARDWARE · ESCALATION

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How to use this checklist

When call quality issues are reported, work through this checklist **systematically from top to bottom**. Start with the symptoms to narrow down the cause, then investigate the most likely areas. This is an **interactive PDF** — tick the boxes and type your scores, owners and notes directly in any modern viewer. Score each section to identify where your weaknesses lie.

WHAT THIS CHECKLIST IS

A practical, repeatable troubleshooting runbook for any business running a hosted VoIP / SIP telephony system. It walks an engineer from reported symptoms through network, QoS, hardware and platform diagnostics to a structured escalation process — so issues are resolved by evidence, not guesswork.

HOW TO USE IT

Begin with the reported **symptoms** to narrow the root cause, then investigate the most likely areas first. Network quality is the single biggest factor in VoIP call quality — if symptoms are widespread, start there. Score each section out of 10 and build an action plan for any area scoring below 6.

The six diagnostic areas

- **01 Audio Quality Issues** — symptoms, inbound vs outbound, intermittent vs constant, MOS scores.
- **02 Network Diagnostics** — ping, jitter, packet loss, bandwidth, switch ports, PoE, firewall logs.
- **03 QoS & Prioritisation** — DSCP markings, switch queues, end-to-end QoS, WAN preservation, load tests.
- **04 Hardware Checks** — firmware, handset swap, cabling, switch ports, headsets, speakerphones.
- **05 Platform & Configuration** — codecs, SIP registration, SIP ALG, NAT, port ranges, encryption.
- **06 Escalation & Resolution** — documentation, PCAPs, provider/ISP escalation, deadlines, knowledge base.

01

Audio Quality Issues

Identify the specific symptoms being reported to narrow down the root cause before investigating. The pattern of who, when and which calls are affected points you straight at the likely cause.

- ☐ **Document the specific audio symptoms** reported — choppy audio, one-way audio, echo, static, robotic voice, or complete silence.
- ☐ Determine if the issue affects **inbound calls, outbound calls, or both** — this indicates whether the problem is local or remote.
- ☐ Check if the issue is **intermittent or constant** — intermittent issues often point to network congestion at specific times.
- ☐ Identify if the problem affects **all users or specific users/extensions** — isolated issues suggest local hardware or cabling problems.
- ☐ Ask whether the issue occurs on **internal calls, external calls, or both** — internal-only suggests LAN issues; external suggests WAN or SIP trunk issues.
- ☐ Check if **softphone users** experience the same issues as desk phone users — this helps isolate handset hardware problems.
- ☐ Verify the **time of day** when issues occur most frequently — peak-hour problems often indicate bandwidth congestion.
- ☐ Review **MOS scores** in the VoIP platform dashboard for the affected calls to quantify the quality degradation.

NARROW BEFORE YOU DIG

Five minutes spent characterising the symptom — who, when, inbound vs outbound, internal vs external — saves hours of investigation. The pattern usually tells you whether to look at the LAN, the WAN, a handset, or the SIP trunk.

SECTION 01 SCORE ________ / 10 *Aim for 8+. Below 6 = symptom poorly characterised.*

REMEDIATION PRIORITY ☐ H ☐ M ☐ L **OWNER / NEXT ACTION** ____________________

02

Network Diagnostics

Network quality is the single biggest factor in VoIP call quality. Diagnose network issues thoroughly — latency, jitter and packet loss between you and the provider tell you most of the story.

- ☐ Run a **continuous ping test** to the VoIP provider's SBC/proxy server for at least 30 minutes to detect packet loss and latency spikes.
- ☐ **Measure jitter** using a VoIP-specific tool — jitter above 30ms causes audible quality degradation *(target: below 15ms)*.
- ☐ Check **internet bandwidth utilisation** at the time of reported issues — saturation above 80% commonly causes voice quality problems.
- ☐ Verify the connection speed **matches contracted levels** using a wired speed test during the affected periods.
- ☐ Check for **packet loss** on the path between your network and the VoIP provider using traceroute and MTR/WinMTR tools.
- ☐ Inspect **switch port statistics** for errors, CRC failures, and collisions on ports connected to affected phones.
- ☐ Verify **PoE power levels** on switch ports — insufficient power causes phones to restart or malfunction intermittently.
- ☐ Check for **network loops or broadcast storms** that could be flooding the voice VLAN with unnecessary traffic.
- ☐ Review **firewall and router logs** for any dropped VoIP packets, rate limiting, or SIP-related errors.

TARGETS THAT MATTER

Latency below 80ms, jitter under 15ms and packet loss at 0%. Even 1% packet loss is audible on voice. A 30-minute continuous ping to the SBC during the complaint window is the most useful single piece of evidence you can gather.

SECTION 02 SCORE ________ / **10** *Aim for 8+. Below 6 = network is the prime suspect.*

REMEDIATION PRIORITY ☐ H ☐ M ☐ L **OWNER / NEXT ACTION** ____________________

03

QoS & Prioritisation

Even with sufficient bandwidth, VoIP will suffer without proper Quality of Service configuration. QoS that stops at a single unconfigured device in the path negates the entire chain.

- ☐ Verify **QoS is configured on the firewall** to prioritise SIP (UDP 5060) and RTP (UDP 10000–20000) traffic above data.
- ☐ Confirm **DSCP markings** are set correctly: EF (46) for voice media (RTP) and CS3 (24) for SIP signalling.
- ☐ Check that **network switches honour DSCP markings** and prioritise tagged voice frames in their egress queues.
- ☐ Verify QoS is configured **end-to-end** — a single unconfigured switch or router in the path negates the entire QoS chain.
- ☐ Confirm the **ISP preserves DSCP markings** across the WAN — some providers strip QoS markings at the handover point.
- ☐ Check that **bandwidth reservation** for voice traffic is sufficient for the number of concurrent calls at peak times.
- ☐ Verify **traffic shaping policies** prevent non-essential traffic (streaming, social media) from consuming voice bandwidth.
- ☐ Test call quality during a deliberate **bandwidth load test** (e.g. large file transfer) to verify QoS is protecting voice traffic effectively.

WATCH OUT FOR

An ISP that strips DSCP markings at the WAN handover so your carefully tuned LAN QoS counts for nothing, and a single switch in the voice path that was never configured to honour markings — one weak link breaks end-to-end QoS.

SECTION 03 SCORE ________ / 10 *Aim for 8+. Below 6 = QoS gap under load.*

REMEDIATION PRIORITY ☐ H ☐ M ☐ L **OWNER / NEXT ACTION** ____________________

04

Hardware Checks

Faulty or misconfigured hardware causes call quality issues that no amount of network tuning will resolve. Rule out the physical layer — cables, handsets, headsets and switch ports — methodically.

- ☐ Check the **handset firmware version** and update to the latest stable version recommended by the VoIP provider.
- ☐ Test with a **known-good replacement handset** to rule out a faulty phone as the cause of the issue.
- ☐ Inspect **Ethernet cables** connecting phones to the network — damaged, kinked, or low-quality cables cause intermittent issues.
- ☐ Verify the phone is connected to a **Gigabit switch port** and not daisy-chained through another device at 100 Mbps.
- ☐ Check **headset compatibility** — non-certified headsets can cause echo, static, and audio level problems.
- ☐ Verify that **Bluetooth headsets** are not experiencing interference from nearby wireless devices or WiFi access points.
- ☐ Check conference room **speakerphones** for echo cancellation settings and ensure they are positioned away from reflective surfaces.
- ☐ Verify that **softphone audio settings** (input/output device, codec selection, echo cancellation) are configured correctly.

SWAP TO ISOLATE

The fastest way to clear or convict hardware is substitution: a known-good handset, a fresh cable, a different switch port. If the symptom moves with the device, you have found it; if it stays, the problem is upstream.

SECTION 04 SCORE ________ / **10** *Aim for 8+. Below 6 = hardware not yet ruled out.*

REMEDIATION PRIORITY ☐ H ☐ M ☐ L **OWNER / NEXT ACTION** ____________________

05

Platform & Configuration

VoIP platform settings and SIP trunk configuration can cause quality issues even on a perfect network. SIP ALG and stripped registrations are perennial culprits — check them early.

- ☐ Verify the **codec configuration** — G.711 provides best quality but uses more bandwidth; G.729 is compressed but lower quality.
- ☐ Check **SIP trunk registration status** — intermittent registration issues cause dropped calls and one-way audio.
- ☐ Verify **SIP ALG is disabled** on all routers and firewalls in the path — this remains the most common cause of VoIP issues.
- ☐ Check **NAT traversal settings** (STUN/TURN) are configured correctly if phones are behind NAT.
- ☐ Review **firewall rules** for SIP and RTP traffic — ensure the correct port ranges are open to the VoIP provider's IP addresses.
- ☐ Check for **SIP retransmissions** in the platform logs, which indicate packet loss or firewall issues on the signalling path.
- ☐ Verify the **RTP port range** is large enough for the number of concurrent calls and is correctly opened on the firewall.
- ☐ Check whether **TLS/SRTP encryption** is causing CPU overload on the router or firewall, especially on older hardware.

SIP ALG FIRST

SIP ALG (Application Layer Gateway) is the most common single cause of VoIP audio and registration problems. If you only check one platform setting, disable SIP ALG on every router and firewall in the path before going further.

SECTION 05 SCORE ________ / 10 *Aim for 8+. Below 6 = config / SIP trunk gap.*

REMEDIATION PRIORITY ☐ H ☐ M ☐ L **OWNER / NEXT ACTION** ____________________

06

Escalation & Resolution

When initial troubleshooting does not resolve the issue, follow a structured escalation process. Good evidence and a clear deadline prevent finger-pointing between vendors and keep the fix moving.

- ☐ **Document all troubleshooting steps** taken with results before escalating to the VoIP provider or ISP.
- ☐ Capture **packet captures (PCAP)** on the affected network segment to provide technical evidence to the provider.
- ☐ Request **call quality reports** from the VoIP provider for the specific calls experiencing issues (CDRs with MOS scores).
- ☐ If the ISP is suspected, request a **line test and circuit health check** with documented results.
- ☐ Escalate to the VoIP provider with **specific call examples** including timestamps, caller/callee numbers, and symptoms.
- ☐ If multiple providers are involved (ISP + VoIP), arrange a **three-way troubleshooting call** to avoid finger-pointing between vendors.
- ☐ Set a **resolution deadline** based on business impact — persistent call quality issues directly affect customer experience and revenue.
- ☐ **Document the root cause and resolution** for the knowledge base to accelerate future troubleshooting of similar issues.

EVIDENCE BEATS OPINION

A documented step log, PCAPs, and CDRs with MOS scores turn an escalation from “our calls are bad” into a specific, actionable case. Providers resolve evidence-backed tickets far faster — and the same record feeds your knowledge base.

SECTION 06 SCORE _________ / 10 *Aim for 8+. Below 6 = escalation lacks evidence.*

REMEDIATION PRIORITY ☐ H ☐ M ☐ L **OWNER / NEXT ACTION** ____________________

Σ

Audit score summary

Transfer the score for each diagnostic area into the table. Set a priority (H/M/L) based on the gap to target. Anything below 6 is a candidate for the top-three actions on the next page.

#	DIAGNOSTIC AREA	SCORE / 10	PRIORITY
01	Audio Quality Issues	______ / 10	☐ H ☐ M ☐ L
02	Network Diagnostics	______ / 10	☐ H ☐ M ☐ L
03	QoS & Prioritisation	______ / 10	☐ H ☐ M ☐ L
04	Hardware Checks	______ / 10	☐ H ☐ M ☐ L
05	Platform & Configuration	______ / 10	☐ H ☐ M ☐ L
06	Escalation & Resolution	______ / 10	☐ H ☐ M ☐ L
Σ	**TOTAL SCORE**	______ / 60	—

HOW TO READ YOUR TOTAL

The total out of 60 is a fast health indicator, but the single lowest section matters more than the average. A 50/60 with one section at 3 still points to a specific, fixable weakness — carry it forward to the priority actions.

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Interpretation & priority actions

Translate your total score into a risk band, then commit to a small number of next steps. The goal is one page of decisions, not a wish list.

Score interpretation

48–60	**Excellent.** Your VoIP troubleshooting and configuration are well-managed. Focus on continuous monitoring and emerging issues.
36–47	**Good foundation, gaps exist.** Prioritise any section scoring below 6 with named owners and deadlines.
Below 36	**Significant gaps** that put call quality and customer experience at risk. Consider an urgent review with a VoIP specialist.

Top 3 priority actions

01 ______________________

02 ______________________

03 ______________________

Additional notes

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AUDIT COMPLETED BY ______________ DATE ______________ NEXT REVIEW DUE ______________

Need help with your VoIP?

Cloudswitched diagnoses and resolves VoIP call quality issues for UK SMEs — from network and QoS tuning to provider escalation, with fixed-price remediation.

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