

Phones Work but No Caller ID — Common Wiring Problems

I wrote this for the call we get all the time: the customer swears their phones are working perfectly — they ring, they dial, calls are clear — but no caller ID is showing up on the POS, and the Vertex (or Whozz Calling VoIP) unit looks like it is seeing nothing. Nine times out of ten this is not a broken unit and not an encrypted carrier. It is **how the unit is wired into the network**. Here is what I want you to check, in order.

Start here — the one thing to understand about the unit

The Vertex is an inline, passive listener — not a phone. It has two VoIP bridge ports (an IN and an OUT) and it sits *in the wire*. It only reads the SIP signaling that physically passes **between those two ports**. It does not "ask" the phone system for anything; it just watches what crosses it and pulls the caller name and number out of the call setup. (The management port then sends that to the POS over UDP 3520.)

So the symptom makes sense once you see it this way: the **phones have their own path** through the switch and out to the internet. They will ring and work fine even when the Vertex is sitting off to the side seeing none of their traffic. **If the SIP never travels through the Vertex, you get exactly this — good phones, no caller ID.**

This is hosted VoIP — there is no phone system in the building to tap. The phones register straight out to the cloud provider (RingCentral and the like). Do **not** give the old analog advice about tapping the trunk ahead of a phone switch — there is no in-house switch and no trunk side to move to. The only thing that matters is that the unit physically sits in the cable the SIP rides on.

How a hosted-VoIP call actually flows

Every phone's call setup takes the same trip out of the building:

Desk phone → desk/office switch → **router / internet gateway** → internet → cloud phone system
The one place **every** phone's SIP crosses is the link between the switch and the router. That is where the Vertex belongs — inline on that uplink (the **chokepoint**).

If you put the unit anywhere that does not carry that traffic, it only sees a sliver — or nothing. Keep that picture in your head as you work through the four wiring mistakes below.

The four common wiring mistakes

1 The phones aren't actually wired through the bridge

What I see: someone patched a phone — or the whole voice run — straight into the switch or the wall jack instead of running it through the unit's IN and OUT ports. The phone works, but its call setup never touches the Vertex.

Why it happens: a phone gets moved, a desk gets added, or a "cleanup" re-patches a cable, and the bridge quietly gets bypassed.

What I want you to do: make sure the traffic for the watched phones actually passes through the unit. The clean way is to put the bridge on the **one uplink between the phone switch and the router**, so every phone's SIP has to cross it. Do not hang it off a single phone's drop and expect to see the rest of the phones.

2 The bridge is on the wrong segment (not at the chokepoint)

What I see: the unit is inline somewhere — but not on the part of the network that carries the calls out to the internet. It might be between the switch and one back-office PC, or on a segment that never sees the WAN-bound traffic.

Why it happens: whatever cable was closest got used. On hosted VoIP, the calls go *out of the building* to the cloud, so the only spot that sees all of them is the switch-to-router link.

What I want you to do: move the unit so it sits **in series on the LAN-to-router uplink**. That is the chokepoint every call passes through on its way to the cloud. If you cannot get it there, you will only ever capture the phones on that one segment.

3 The IN and OUT legs aren't truly in series

What I see: both bridge ports end up plugged into the **same switch** (so traffic just loops around through the switch and never has to cross the unit), or the OUT leg goes somewhere that isn't actually downstream of the phones. The cabling *looks* connected, but the unit is not really "in the middle" of anything.

Why it happens: the two ports look identical, so it is easy to land them in the wrong jacks.

What I want you to do: wire it so one bridge port faces the **network/router side** and the other faces the **phones side**. The traffic must be forced to go *in one port and out the other*. If both legs come off the same switch, the switch will deliver the calls without ever sending them through the unit.

4 The phones are daisy-chained through their PC pass-through ports

What I see: VoIP desk phones have a built-in two-port switch (one jack to the wall, one "PC" jack for the computer). On a lot of installs the cabling is chained — wall → phone → phone's PC port → next phone or PC. When that happens, some phones' traffic takes a path that bypasses where the Vertex is sitting.

Why it happens: it saves cable drops, so installers do it constantly — but it scatters where each phone's call setup actually travels.

What I want you to do: get the Vertex **upstream of the whole chain**, on the uplink to the router, so it does not matter how the phones are daisy-chained behind it. Everything still has to funnel through that one link to reach the cloud.

If the wiring is right and you STILL get nothing — check the transport

This one is not a wiring problem, but it looks exactly like one, so rule it out. RingCentral (and some other carriers) deliver the call setup over **SIP-over-TCP** — typically port 5090 plus a per-call high port — and they send it **cloud-direct to each individual phone**. They do **not** use the classic UDP 5060 that the unit looks for by default. So you can have perfect wiring and still see "no SIP," because the unit is watching for UDP and the carrier is shipping TCP.

What I want you to do: if you have confirmed the unit is truly inline on the uplink and there is still no caller ID, stop re-checking cables — capture about 30 seconds of live traffic and look at what transport the SIP is actually riding. If it is TCP-only, that is the carrier's delivery method, and the fix is on the carrier/configuration side, not the wiring. Call me before you start swapping hardware on these — I want to see the capture.

Quick checklist

#	CHECK	GOOD
1	Is the unit physically inline (one port to the network, one to the phones), not hanging off a single drop?	In series, in the wire
2	Is it on the switch-to-router uplink, where every call goes out to the cloud?	On the chokepoint
3	Are the IN and OUT legs on opposite sides (not both on the same switch)?	Truly in series

#	CHECK	GOOD
4	Are daisy-chained phones all behind the unit (unit upstream of the chain)?	Upstream of all phones
5	Still nothing? Capture the traffic — is the SIP on TCP (e.g. 5090) instead of UDP 5060?	Transport, not wiring

Bottom line: the phones working tells you nothing about whether the Vertex can see the calls — they ride different paths. Get the unit physically in series on the one link that carries every call out to the cloud, confirm the IN/OUT legs are on opposite sides, and you fix the large majority of "phones work, no caller ID" tickets. If it is wired right and still blank, it is the transport, and that is a phone call to me — not a hardware swap.

Need help? Email support@callerid.com | Phone **770-263-7111** Mon-Fri 9am-5pm ET