



Tech Bulletin: RingCentral SIP on Port 5090 — Vertex Requires UDP/5060

RingCentral delivers SIP on port 5090 (UDP and TCP) — Vertex only reads UDP/5060 — fix is a RingCentral account change

Internal Tech Bulletin — 21JUL26 | Source: packet captures 21-260610-41-21 (Outbound + Inbound, 21JUL26)

Root cause. RingCentral at this site delivers SIP on **port 5090** — some phones over UDP/5090, others over TCP/5090. The Vertex captures caller ID by reading SIP INVITE packets delivered as **UDP on port 5060**. Nothing at this site uses UDP/5060. The Vertex sees the SIP traffic crossing its bridge but does not process it because the port does not match. There is no setting in VCT to change this. The only resolution is to contact **RingCentral Tech Support** and request that SIP for this account be delivered as **UDP on port 5060**.

1. Packet Capture Analysis — 21JUL26 (Device 21-260610-41-21)

Two captures were taken simultaneously on the Vertex bridge ports during a live call session: outbound (8,789 packets) and inbound (4,113 packets). The captures are standard Ethernet pcap files and were parsed with no proprietary header offset.

What we looked for	Found?	Detail
UDP SIP on port 5060	NONE	Zero packets on UDP/5060 in either capture. This is the format the Vertex requires.
UDP SIP on port 5090	YES — 37 packets	Phone 192.168.20.59 ↔ RingCentral 104.245.57.211 on UDP/5090. Full call flow confirmed: REGISTER → INVITE → 180 Ringing → 200 OK → BYE. SIP is UDP at this phone — but port 5090, not 5060.
TCP SIP on port 5090	YES — 26 packets	Phones 192.168.20.60 and 192.168.20.61 use TCP/5090. Their INVITEs were cancelled (other phone answered first). Vertex cannot process TCP SIP regardless of port.
RTP voice media	YES	High-volume UDP exchange on ephemeral ports — 192.168.20.59 ↔ 103.44.71.149 (RingCentral media). Calls connecting and audio flowing normally.

Vertex CallerID
broadcast (UDP
3520)

PARTIAL

Vertex at **192.168.20.62** broadcasting on
255.255.255.255:3520 with status codes (WSIPD , WDMPG ,
NRD , X). No phone number in any broadcast — Vertex is running
but not processing the port-5090 SIP.

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# Confirmed SIP flows — device 21-260610-41-21 (correct parse, no header offset)
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UDP SIP (phone .59 ↔ RingCentral — 37 packets total)
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REGISTER sip:sip.ringcentral.com:5060 SIP/2.0      192.168.20.59:5090 →
104.245.57.211:5090
INVITE sip:14702100801@sip.ringcentral.com:5060    192.168.20.59:5090 →
104.245.57.211:5090
SIP/2.0 100 Trying                                  104.245.57.211:5090 →
192.168.20.59:5090
SIP/2.0 183 Session Progress                        104.245.57.211:5090 →
192.168.20.59:5090
INVITE sip:19106943129@192.168.20.59:5090          104.245.57.211:5090 →
192.168.20.59:5090 ← inbound
SIP/2.0 180 Ringing                                 192.168.20.59:5090 →
104.245.57.211:5090
SIP/2.0 200 OK                                       192.168.20.59:5090 →
104.245.57.211:5090
BYE / NOTIFY                                         both directions

TCP SIP (phones .60 and .61 — cancelled calls)
REGISTER sip:sip.ringcentral.com:5060;transport=tcp 192.168.20.60:36797 →
104.245.57.25:5090
INVITE sip:19804761178@192.168.20.60:5070          104.245.57.25:5090 →
192.168.20.60:36797
CANCEL / 487 Request Cancelled                      (call answered on .59 first)

UDP 5060 — zero packets in either capture —
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What the Vertex event codes mean. WSIPD = Vertex detected a SIP event (outbound call). WDMPG = dialing in progress. NRD = No Ring Detect — Vertex saw SIP traffic crossing the bridge but did not recognise an inbound ring because the source port (5090) does not match its expected SIP port (5060). X = call ended. The Vertex is powered, networked, and watching — but the port mismatch stops it from extracting caller ID.

2. Why the Vertex Cannot Capture Caller ID

The Vertex bridges inline between the phones and the router and inspects every SIP packet that crosses it. It identifies SIP specifically by looking for traffic on **UDP port 5060** — the original SIP standard (RFC 3261). RingCentral at this site uses **port 5090** for all SIP, on both UDP and TCP. The Vertex sees the packets but does not process them because they are on the wrong port.

Protocol & Port	RingCentral using?	Vertex can capture?	Result
UDP / 5060	NO	YES — designed for this	Would work — this is the target state after RingCentral change
UDP / 5090	YES — phone .59 confirmed	NO — wrong port	No caller ID captured
TCP / 5090	YES — phones .60, .61 confirmed	NO — wrong port and protocol	No caller ID captured

There is no setting in VCT to change this. The Vertex Configuration Tool does not have an option to change the SIP port the unit monitors. The Vertex is designed to listen for UDP/5060. The change must happen at the RingCentral account level — they need to deliver SIP to these phones as **UDP on port 5060**.

3. Resolution — Contact RingCentral Tech Support

Email RingCentral Tech Support and request that SIP signaling for this account be delivered as **UDP on port 5060**. RingCentral can configure the SIP transport at the account level. Provide the account number and the MAC addresses of all handsets so their team can locate and update the correct configuration.

Item	Detail
RingCentral Tech Support email	techsupport@ringcentral.com
What to include	RingCentral account number · MAC address of each handset · site name / address
What to request	Convert SIP delivery from port 5090 (UDP and TCP) to UDP port 5060 for all phones on this account
Reason to give RingCentral	Third-party caller ID hardware (CallerID.com Vertex) requires SIP over UDP/5060 and cannot process SIP on port 5090

Handset MAC Addresses (from capture — this site)

Phone IP	Ethernet MAC	SIP Transport seen in capture
192.168.20.59	f0:4e:a4:67:b7:10 (HP / Poly)	UDP / 5090
192.168.20.60	Confirm with customer	TCP / 5090
192.168.20.61	Confirm with customer	TCP / 5090

MAC addresses for .60 and .61. The Wireshark capture confirms the MAC for phone .59 (f0:4e:a4:67:b7:10 , HP/Poly brand). For phones .60 and .61, ask the customer to read the MAC label on the back of each handset, or pull it from the RingCentral admin portal under Phones & Devices.

Sample Email

Email Template — RingCentral SIP Port Change Request

To: techsupport@ringcentral.com

Subject: SIP Port Change Request — Convert to UDP/5060 — Account **[RingCentral Account Number]**

Hello RingCentral Tech Support,

We are requesting a SIP transport configuration change for our RingCentral account.

Account Number: *[RingCentral Account Number]*

Site / Location: *[Store Name and Address]*

Handset MAC Addresses:

f0:4e:a4:67:b7:10 (phone at 192.168.20.59)

[MAC of phone at 192.168.20.60]

[MAC of phone at 192.168.20.61]

Issue:

We have a CallerID.com Vertex VoIP caller ID unit installed inline on this account. Packet capture analysis shows that RingCentral is currently delivering SIP signaling to these phones on **port 5090** (one phone over UDP/5090, two phones over TCP/5090). The Vertex hardware is designed to capture SIP delivered as **UDP on port 5060** and cannot process SIP on port 5090. As a result, caller ID is not being received at our POS system.

Request:

Please convert SIP delivery for all handsets on this account from port 5090 to **UDP port 5060**. This is a standard SIP transport change. We do not need any change to voice quality, routing, or features — only the SIP port and transport protocol for this account.

Please confirm by email when the change has been applied so we can test caller ID reception.

Thank you,

[Your Name]

[Store Name]

[Contact Phone Number]

After RingCentral confirms the change. Make a test inbound call. In VCT Live View the caller's phone number should now appear with an **I** (inbound) flag within the first ring, and the POS caller ID pop-up should appear within 1-2 seconds. If VCT shows numbers but the POS does not, the issue is the Vertex Destination IP/MAC setting — see Section 4.

4. Destination IP / MAC — Where the Vertex Sends the Record

Once RingCentral delivers SIP as UDP/5060, the Vertex will generate caller ID records and broadcast them on UDP 3520. The Vertex at this site is already broadcasting correctly:

Setting	Current (from capture)	Status
Vertex IP	192.168.20.62	CONFIRMED — broadcasting on UDP 3520
Destination	255.255.255.255:3520 (broadcast)	CONFIRMED — broadcast packets seen in capture
POS / receiving PC	Must be on 192.168.20.x subnet to receive	Verify with customer after RingCentral change

Broadcast mode is correct for most installs. If the POS is on a different subnet or VLAN, switch to unicast in VCT — enter the POS IP and MAC (run `ipconfig /all` on the POS PC to get both values).

5. Troubleshooting Matrix

Symptom	Likely Cause	Action
Vertex broadcasts NRD or WSIPD on 3520 but no phone number	RingCentral still on port 5090 — Vertex not processing it	Confirm RingCentral has applied the change. Take a VCT capture during a call and open in Wireshark — filter <code>udp.port == 5060</code> . You should now see INVITE packets.
VCT Live View shows call numbers but POS sees nothing	Destination IP or MAC wrong, or firewall blocking UDP 3520	Set Vertex Destination to broadcast (255.255.255.255 / FF:FF:FF:FF:FF:FF). If POS then works, tighten to unicast. If still nothing, check that UDP 3520 is permitted on the switch port serving the POS.
VCT Live View shows nothing at all after RingCentral change	Vertex not wired inline, or SIP Gateway MAC still wrong	Verify Vertex is wired inline between phones and router. Confirm SIP Gateway MAC in VCT matches the router's MAC. Take a VCT capture to confirm UDP/5060 SIP is crossing the bridge.
RingCentral says they cannot change the port	Front-line support unfamiliar with SIP transport settings	Escalate the ticket and ask specifically for the “SIP transport port” to be changed to 5060 UDP . Reference RFC 3261. Ask for escalation to their SIP or VoIP engineering team if needed.

Caller ID works on one phone but not others	Mixed SIP transport — some phones still on TCP/5090 after change	Ask RingCentral to confirm the change was applied to ALL handsets on the account, not just one. Provide the MAC address of each handset that is still not working.
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6. Network Summary — This Site (21-260610-41-21)

Device	Address	SIP Transport	Status
Vertex VoIP unit	192.168.20.62	Reads UDP/5060 only	Broadcasting on UDP 3520 — hardware working, awaiting UDP/5060 SIP
Phone (HP/Poly)	192.168.20.59 MAC f0:4e:a4:67:b7:10	UDP/5090	Active — calls connect; caller ID not captured (port 5090)
Phone	192.168.20.60	TCP/5090	Active — INVITEs seen but cancelled; caller ID not captured
Phone	192.168.20.61	TCP/5090	Active — INVITEs seen but cancelled; caller ID not captured
RingCentral media server	103.44.71.149	RTP (UDP, ephemeral)	RTP audio confirmed — calls working normally
CallerID destination	255.255.255.255:3520	UDP broadcast	Any device on 192.168.20.x subnet receives records when caller ID is working