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VoIP Requirements and Troubleshooting

Purpose

ringDNA Connectivity

ringDNA supports connectivity in a number of methods. You can connect with your desk phone, a SIP phone, or your mobile phone, but we also support a VoIP connection via the dialer alone.

ringDNA offers the ability to call via:

1. Browser (WebRTC)
2. Call Forwarding to a DID (phone or mobile)
3. SIP-based hard desk phone

Browser:

Browser based calling is convenient since no phone hardware or infrastructure is required on your part. All you need is Internet access and a USB headset to use with your computer. We assign a phone number to your ringDNA application and away you go.

This has worked well for some of our customers. That said, sometimes network and environment variables such as computer CPU and memory, Internet bandwidth/latency, using LAN cable vs. WiFi (always use cable where possible) can impact call quality.

Therefore we recommend testing Browser based connectivity first.

Call Forwarding to Desk and Mobile Phones:

Most customers use a ringDNA feature called Call Forwarding to make outbound and inbound calls to a landline or mobile phone. This bypasses the aforementioned Browser variables that can introduce call quality in certain situations. The only marginal downside of Call Forwarding is that the number of ringDNA minutes consumed doubles since there are two legs for each call (though this can be covered later by the AE).

SIP:

Some customers also use a SIP based hard desk phones with RingDNA to eliminate call quality issues. The advantage here is the customer only has to incur a one time hardware investment which includes a phone, headset and headset connector. There are no incremental recurring charges as would be incurred with connecting to a traditional desk or mobile phone. And SIP phones can be plugged into an Ethernet LAN cable (so yes Ethernet is required) i.e., you obviously do not need a phone line.

VoIP

VoIP exists at the intersection of internet and telecom, meaning that troubleshooting audio quality issues over it can sometimes be an intricate process. This guide is meant to help users and their IT/network admins troubleshoot common problems.

Why do audio issues sometimes occur?

Sometimes audio quality issues exist on the other end of the line, over the PSTN (publicly telephone network) connection the other party is using or as a result of their network. But many issues with VoIP are the result of problems with your computer or local network, and this guide can help account for and correct those problems.

User Hardware, Headset, and Browser Requirements

The exact device hardware requirements are dependent on a combination of multiple factors, but generally we recommend at a minimum:

1. Use the most recent version of Google Chrome:
   a. The latest version of the Chrome browser provides the best performance when using ringDNA voice.
2. 8 GB of memory for Windows, 4 GB for Mac.
3. Close any CPU/network intensive web pages such as video/audio streaming services.
4. Use an ethernet cable if available to reduce the latency caused by WiFi.

Headsets

Good-quality headsets can improve audio quality by minimizing echo. We recommend using a headset when calling from RingDNA to provide acoustic isolation and to minimize echo.

Headset for Lower-end PC Hardware:

For lower-end PC hardware, we recommend USB headsets, which bypass the native sound board, rather than 3.5mm jack headsets. For computers with a higher-end integrated sound board the 3.5mm connection is fine.

When Selecting Bluetooth Hardware:

Bluetooth headsets can present unique challenges, as each headset operates differently. If your headset came with a USB bluetooth adapter, we recommend you use the USB adapter, rather than bluetooth, to avoid bluetooth interoperability issues.

Note that static in your client audio is often due to a misbehaving or misconfigured headset. If you are experiencing static, try with different headset hardware or no headset hardware to narrow down potential causes of static.

Addressing Dropped Calls

In the unlikely scenario that ringDNA calls drop, many intermittent issues in ringDNA Voice can be resolved by clearing the browser cache and ensuring your on a strong and stable internet connection. You can clear the cache on your Chrome
browser via the following steps:

1. Open the Chrome history page: Ctrl-Shift-Delete (Windows) or Command-Shift-Delete (Mac).
2. In the dropdown select "the beginning of time."
3. Select the "Cached images and files," "Cookies..." and "Hosted app data" checkboxes.
4. Click "Clear browsing data."

Submitting Call Reports:

If you are experiencing issues with call quality while using RingDNA, please do utilize the "Report this Call" feature in ringDNA.

To report a call, go to ringDNA > Call History > Click into the failed call and on the bottom of the dialer, select "Report this call". You can also click "Report this call" at the bottom of the dialer from a live call, or a call that just ended.

Please enter a brief description on how the call failed and as soon as we receive the report, we'll investigate what the root cause was. The data we capture through call reports allow us to analyze server logs when the call specifically failed, it's our best tool to resolving issues of this nature.

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Voice Trace:

Oftentimes, issues with call quality require a special troubleshooting tool, known as Voice Trace, to be enabled with our carrier.

Voice Trace works at an Account level, so all calls reported by all reps during the Voice Trace will have this tool available.

Voice Trace captures calls at the level of individual packets for closer analysis. The data is only stored for call quality analysis purposes and the feature can only be activated for a maximum of two weeks at a time, after which the data is erased.

We do need further call reports after it is activated, since it is only able to capture data about calls that are placed after it is turned on.

Best Practice for IT Admin/Network Engineers

ringDNA Voice uses WebRTC, making performance largely dependent on the connection established between your network and our carrier's servers. This section contains some network optimization that can improve the audio quality of ringDNA service.

Bandwidth Requirements

Fast internet isn't the same as consistent, reliable internet. Even if your connection is very high-bandwidth, there could still be issues with jitter, packet loss, bufferbloat, or latency.

- The following table lists the network requirements to deliver reasonable audio quality:
    - Latency (RTT) of less than 200ms.
    - Jitter of less than 30ms.
    - Packet Loss of less than < 3%
        - [Packet Loss Tester](#)
- Minimum bandwidth: our carrier suggests at minimum a 100 kbps symmetrical connection for each concurrent call. You can use [this link](#) to test your existing network bandwidth: https://networktest.twilio.com/
- Recommended bandwidth: Typically we see reps engaging with other network activities while on the call (such as updating CRM records, navigating between different websites for research, etc.). If you engage in these network activities while calling prospects, we recommend at least 300-500kbps bandwidth for each concurrent call.

Firewall Configuration

Your firewall should allow outgoing UDP to the public internet from the browsers that will be using ringDNA Voice, and allow return traffic in response. This article includes a list of server/client ports used by WebRTC, which is the underlying protocol used by ringDNA.

IP Whitelisting

You can find a fixed range of IP addresses to whitelist in this [documentation](#).

IPs and Ports:
https://www.twilio.com/docs/voice/client/javascript/voice-client-js-and-mobile-sdks-network-connectivity-requirements#voice-media-servers-connectivity-requirements

Reducing Jitter and Packet Loss

Jitter, latency and packet loss are the biggest contributors to voice quality issues in any voice network.

- Jitter: When packets arrive in a different order compared to when they were sent. The main symptom is choppy audio quality.
- Packet Loss: Certain network connections such as WiFi are prone to packet loss, which leads to "robotic sounding" audio.

ringDNA recommends the following best practices to avoid jitter and packet loss:

1. Equip your reps with ethernet connections when possible.
2. Reduce packet conflicts on WiFi by reducing the number of devices operating on the same channel. Avoid large data file transfers going over the same WiFi environment concurrently with voice.
3. Configure your router with QoS rule to prioritize the traffic based on the [IP ranges here](#).
4. Avoid bufferbloat. Bufferbloat builds up large queues that cause noticeable latency and bursts of jitter in a voice call.
5. We recommend ensuring your router is configured with a low buffer size. The perceived buffer size should be around 100ms or less.

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Reducing Latency

Typical symptoms of latency are audio delay or people talking over each other. Callers typically start to notice the effects of latency once it breaches 250ms "mouth-to-ear" or roundtrip, and above ~600ms the experience is unusable.

Note that there will always be some latency – the time it takes the codec to encode the audio, the jitter buffer and the traversal time between our carrier's servers and your network will always introduce some level of latency. The object is to minimize it and keep the total trip time below 300-400ms for voice calls.

Strategies to minimize latency:
- Some lower-bandwidth fixed internet connections can often have a higher latency. If possible, upgrade your internet connectivity.
- LTE (mobile 4G data) can often have high latency.

Other

If your router includes SIP Application Level Gateway (ALG) functionality or Stateful Packet Inspection (SPI), disable both of these functions.