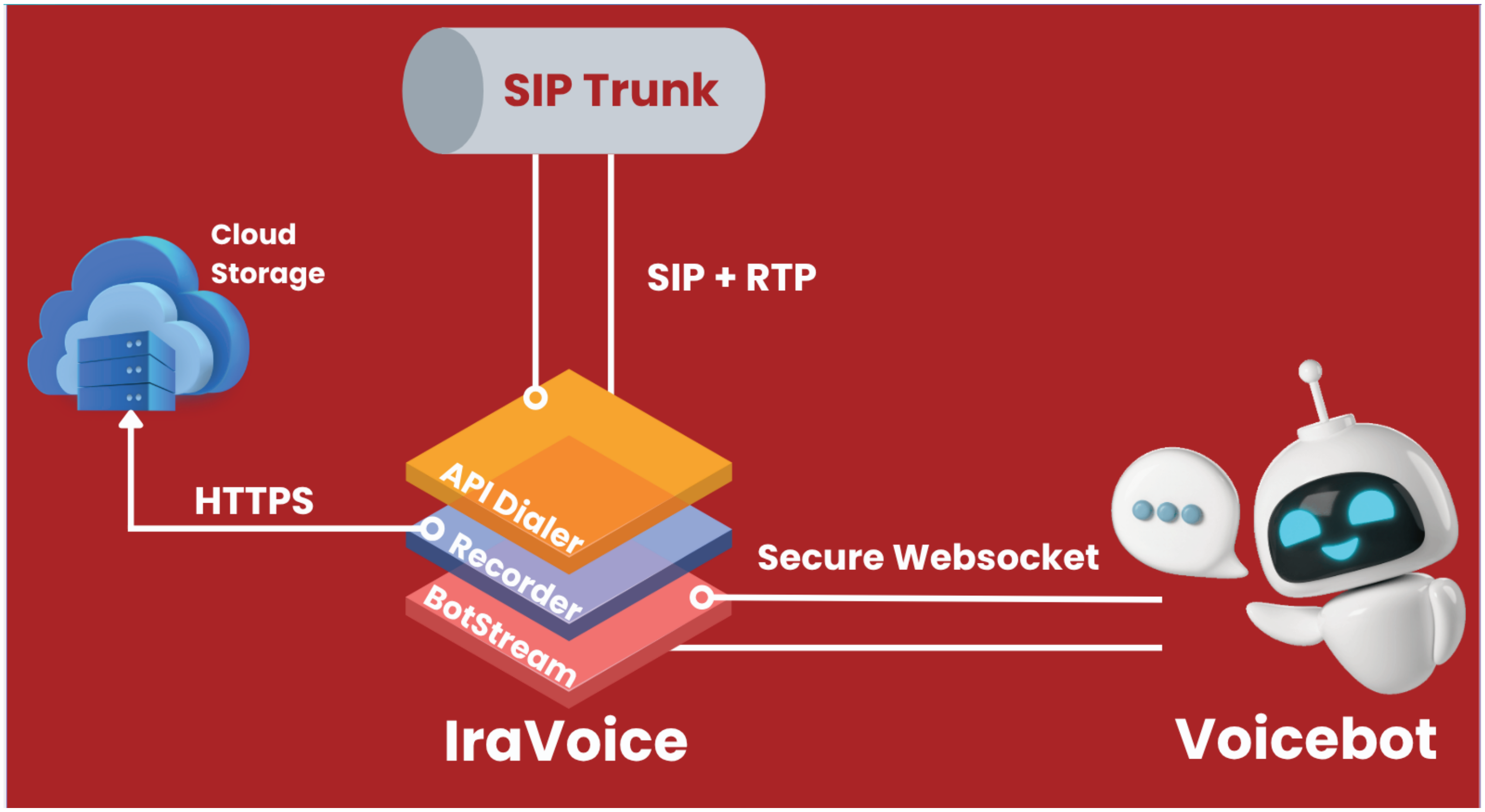
IraVoice FAQs



This FAQ sheet covers the most common questions about **IraVoice**, from deployment and integration to scalability, compliance, and performance. It serves as a quick reference for teams looking to understand how IraVoice simplifies telephony infrastructure for VoiceAI and enterprise communication environments.

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GENERAL

“ Do we have a dedicated SIP trunk? ”

*Yes, You will have a **dedicated SIP trunk**, generally hosted in the same **data center** as your deployment for optimal performance and minimal latency.*

“ Any concurrency limitations? ”

*IraVoice, when hosted on an 8-core, 8GB RAM server, can handle up to **500 simultaneous voicebot sessions with recording enabled**, and up to **1000 simultaneous sessisons without recording**.*
*Besides CPU resource constraints, the call capacity also depends on the SIP trunk configuration, including the number of **channels** allocated and the **CPS** (Calls Per Second) limit defined by the telecom provider.*

“ How many concurrent calls can we support during the POC phase? ”

You can make up to 5 concurrent calls in the POC Sandbox setup. If you need additional capacity, please reach out to Epicode Support.

“ For a given DID number, how many simultaneous inbound and outbound calls can be handled? ”

*Simultaneous call capacity is not limited by the DID number but by the number of **channels** configured on your SIP trunk. 1000 channels = 1000 simultaneous calls. A single DID number can be used to make multiple simultaneous calls as long as it does not cross the number of channels allocated.*

Do you support WhatsApp Business Calling API?

*yes, IraVoice supports supports **WhatsApp Business Calling.***

Will I be able to configure the number of channels based on our use case?

*Yes, Epicode team will recommend the ideal **channel** and **CPS** (Calls Per Second) capacity based on your specific **use case** and **expected call traffic**.*
If at any point call traffic surpasses the anticipated volume, you can scale on demand by increasing SIP trunk limits or provisioning additional CPU resources.

Where or how can we adjust dial limits?

*Dial limit can be managed via the **IraVoice Trunk Manager**, either through the **HTTP API** or directly in the **Trunk Manager interface**.*

Is there support for 16kHz audio streaming?

*IraVoice supports streaming in both **8khz** and **16khz.***

How are inbound calls configured on your platform?

*Inbound calls in **IraVoice** are configured through **dialplans** within the setup. These **dial plans** are typically implemented by the Epicode team based on your inbound routing requirements.*
You can refer to this article for more details on setting up a dialplan:
[Epicode Support – Dialplan Configuration]

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Are there any parameters available for VAD configuration : Speech threshold, silence threshold?

Parameters can be set within the call_params as follows:

- ***VAD Mode** ("enable_vad": true)*
  - *Audio is delivered as complete utterances whenever the user finishes speaking.*
- ***Non-VAD Mode** ("enable_vad": false)*
  - *Audio is streamed in chunks.*
  - *chunk_size can be configured under call_params.*
  - *Default: 3200 bytes (200 ms of audio).*
- ***Silence Threshold** ("silence_threshold")*
  - *Audio level threshold to consider a segment as silence.*
  - *Can range from **1 to 20**, with a **default of 5**.*
  - ***Silence Duration:** Calculated as threshold value × 250 ms to determine when a segment is considered silent.*
- ***Speech Threshold** ("speech_threshold")*
  - *Speech level threshold to consider a segment as speech*
  - *Defines the minimum **amplitude level** required to classify a segment as speech.*
  - *Works best when set between **500** and **600**. **Default: 800***

How can I set custom SIP headers for inbound/outbound calls?

*Custom SIP headers for IraVoice outbound calls can be configured using the **channel_vars** parameter in the **make_call** API:*

"channel_vars": { "<sip-header-name>": "value string" }

For Inbound Calls, custom SIP headers can be defined within the IraVoice dialplan configuration.

> *What are the possible causes of latency and Voice distortion in IraVoice and how can they be minimized or avoided?*

Possible causes of latency and voice distortion in IraVoice include:

- **Network issues**: *High jitter, packet loss, or unstable bandwidth between SIP trunks, media servers, and VoiceAI endpoints.*
- **Server resource constraints**: *CPU or memory saturation on the host machine.*
- **Inconsistent streaming configurations**: *Mismatched **sample rates** or **chunk sizes** between endpoints leading to distorted audio.*
- **Media routing complexity**: *Long network paths or multiple proxy hops introducing transmission delays.*
- **VoiceBot response time:** *Slow response from AI models or APIs used in VoiceBots adding to overall call latency.*

To minimize or avoid these issues:

- *Use **dedicated bandwidth** and maintain network jitter below **30 ms**.*
- *Allocate adequate **CPU and memory resources** based on expected concurrency.*
- *Ensure consistent **audio streaming configurations** across all endpoints.*
- *Optimize **VoiceBot applications** to minimize response delays between streaming chunks.*

> *What storage options are available to upload the recordings ?*

We support recording uploads to your AWS s3 buckets, Google cloud storage, or directly to your webhook endpoint.

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> *How are the recordings shared with us? Do we receive downloadable links for each call in a consolidated format (e.g., Excel or API)?*

To upload call recordings to your cloud, we will require the necessary credentials for your cloud storage. Recordings will be uploaded as calls are completed.

If you prefer to use non-cloud storage, please provide a server with adequate storage capacity. Epicode will upload the recordings to this server and share a secure HTTP endpoint for download access.

> *If we provide our own storage bucket, what naming convention will be used for the recording files (e.g., call_uid or additional metadata)?*

If you prefer to use your own cloud storage bucket, You can share the necessary credentials to us to upload the recordings. The recording files will follow a standardized naming convention that includes the call UUID.
Example: e88e852e-da28-4e1f-bc97-fc00469faebf.mp3

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