AudioCodes Intuitive Human Communications for Chatbot Services

Voice.Al Gateway

Release 1.0



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Product Description Notices

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Related Documentation

Document Name

Voice.Al Gateway API Reference Guide

Voice.Al Gateway Integration Guide



General Notes, Warnings, and Safety Information



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30903	Features updated; deployment options updated.		
30904	Support for Google Dialogflow, Google Cloud Text-to-Speech and Speech-to-Text. Sections updated: Connect on Bot Prompt (default); Language (voice name); SSML for TTS; Punctuation of STT Transcriptions (default); Barge-In (seconds); TTS Caching (default); Call Transfer; Disconnect		

Documentation Feedback

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our website at https://online.audiocodes.com/documentation-feedback.

Product Description 1. Introduction

1 Introduction

AudioCodes Voice.Al Gateway enhances chatbot functionality by allowing human communication with chatbots through **voice** (voicebot), offering an audio-centric user experience. Employing AudioCodes' Voice.Al Gateway service provides you with a single-vendor solution, assisting you in migrating your text-based chatbot experience into a voice-based chatbot.

The Voice.Al Gateway also allows you to provide this enhanced chatbot service to a wider audience that uses various voice engagement channels from where users initiate calls with chatbots:

- SIP Trunk
- PSTN or cellular
- Enterprise Unified Communications (UC) and IP-PBXs
- Contact Centers
- WebRTC

AudioCodes' field-proven and sophisticated voice communications technology embedded in the Voice.Al Gateway allows seamless integration into any existing voice network. The Voice.Al Gateway can connect to and integrate with any third-party cognitive service – bot frameworks, speech-to-text (STT) engines, and text-to-speech (TTS) engines. It can also operate with TTS and STT engines for multiple languages.

Voice Engagement Channels Cognitive Services Text-to-speech services SIP/RTP Text-to-speech APIs **PSTN Enterprise UC Bot frameworks** Voice.Al **Bot frameworks APIs Contact Center** Service G Dialogflow Speech-to-text services SIP over WebSocket/SRTP Speech-to-text APIs

Figure 1-1: Voice Engagement Channel and Cognitive Services of Voice.Al Gateway



The Voice.Al Gateway provides advanced call management capabilities, for example, call disconnect, call transfer to a human agent, and call recording.

As such, the Voice.Al Gateway provides and uses various APIs for different purposes:

- SIP, RTP and WebRTC APIs for communicating with voice engagement channels
- HTTP-based APIs for interfacing with third-party voice cognitive services that convert voice to text (STT) and text to voice (TTS)
- HTTP-based APIs for connecting to third-party bot frameworks such as Azure, Google, and AWS

Product Description 2. Conversation Flow

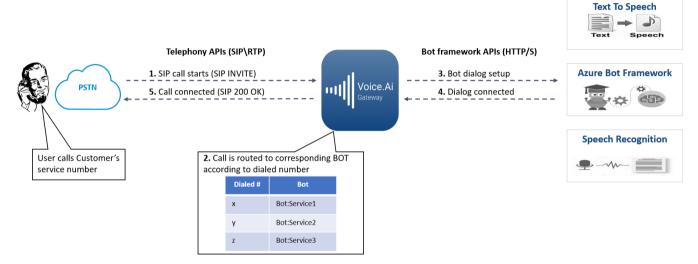
2 Conversation Flow

This section describes the call flow between the voice engagement channel (telephony) and the cognitive chatbot services, through AudioCodes Voice.Al Gateway.

2.1 Conversation Setup

The call setup flow between the voice engagement channel (telephony) and the cognitive chatbot services through AudioCodes Voice.Al Gateway is shown and explained below:

Figure 2-1: Dialog Flow Between Voice Channel and Chatbot Cognitive Services



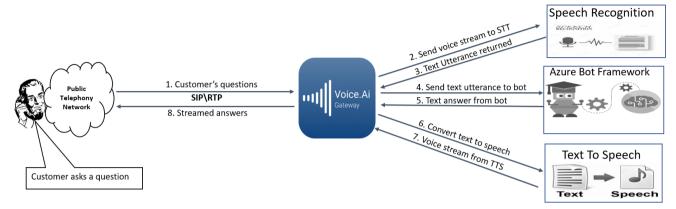
- 1. The user initiates a call with the chatbot, sending a SIP INVITE message to the Voice.Al Gateway.
- 2. The Voice.Al Gateway processes the call by employing its inherent session border controller (SBC) functionality. This includes basic call handling stages (call classification, message manipulation and routing) and optionally, other feature-rich SBC capabilities. The device searches its IP-to-IP Routing table for a matching routing rule (typically based on dialed number), which determines the route to a specific bot framework (destination).
- **3.** The device initiates an HTTP-based dialog setup with a specific bot framework, according to the destination of the matched routing rule.
- An HTTP-based connection is established between the Voice.Al Gateway and the bot framework.
- 5. The Voice.Al Gateway sends a SIP 200 OK response to the user, establishing the media session between it and the user.



2.2 Speech Flow

The speech flow between the voice engagement channel (telephony) and the cognitive chatbot services via AudioCodes Voice.Al Gateway is shown and explained below:

Figure 2-2: Speech Flow Between Voice Channel and Chatbot Cognitive Services



- 1. The user asks the bot a question, which is sent to the Voice.Al Gateway over RTP.
- 2. The Voice.Al Gateway sends the voice stream to the third-party speech-to-text (STT) engine.
- **3.** The STT engine converts the spoken utterance into text, and then sends it to the Voice.Al Gateway.
- 4. The Voice.Al Gateway sends the text transcription to the bot.
- **5.** The bot employs artificial intelligence and searches its database for the most appropriate answer, and then sends it in text format to the Voice.Al Gateway.
- 6. The Voice.Al Gateway sends the text-based answer to a text-to-speech (TTS) engine.
- The TTS engine converts the text into speech and then sends the voice stream to the Voice.Al Gateway.
- 8. The Voice.Al Gateway plays the audio-based answer to the user.

3 Main Components of Voice.Al Gateway

The Voice.Al Gateway solution is comprised of the following main components:

- Voice Engagement Channel Component
- Cognitive Services Component

3.1 Voice Engagement Channel Component

The Voice Engagement Channel component is based on the Voice.Al Gateway's embedded and feature-rich session border controller (SBC) module. It is responsible for interfacing with almost any voice engagement channel, as shown in the figure below. This component inherits all the SBC capabilities, for example, SIP interoperability, media handling (including translation), security, high availability, and scalability.

The Voice Engagement Channel component processes the SIP signaling and media (RTP) traffic and then converts the traffic into HTTP, which it routes to specific bots residing on specific bot frameworks (discussed in the next section).

Voice Engagement Channels

SIP/RTP

Enterprise UC

Contact Center

SIP over WebSocket/SRTP

WebRTC

Voice.Al Gateway Service

Figure 3-1: Voice Engagement Channel Component of Voice.Al Gateway



3.2 Cognitive Services Component

The Voice.Al Gateway's Cognitive Service component interfaces with third-party, cloud-based chatbot services, which include bot frameworks, speech-to-text (STT) engines, and text-to-speech (TTS) engines. The Voice.Al Gateway implements a multi-chatbot platform approach, enabling the use of different chatbot services based on dialed (called) number. Integration and support for these multi-bot frameworks and speech services is done through HTTP-based APIs, which convert any SIP event into the bot framework API, and vice versa. The Voice.Al Gateway's Cognitive Service component thus orchestrates the stream flow between the bot framework, STT engine and TTS engine.

Text-to-speech APIs

Text-to-speech Services

Text-to-speech Services

Amazon Polly

Bot frameworks

Amazon Lex

Service

Speech-to-text Services

Azure

Azure

Speech-to-text Services

Azure

Speech-to-text Services

Azure

Azure

Speech-to-text Services

Azure

Azure

Azure

Speech-to-text Services

Azure

Figure 3-2: Cognitive Services Component of Voice.Al Gateway

3.2.1 Text-to-Speech Services

The Voice.Al Gateway uses third-party text-to-speech (TTS) engines to convert text, sent from the bot into speech, which it then plays to the user.

The Voice.Al Gateway supports the following third-party TTS services:

- Azure Speech Services
- Amazon Web Services (AWS) Amazon Polly ¹
- Google Cloud Text-to-Speech
- Almagu¹
- Yandex¹



Note: The Voice.Al Gateway interfaces with the TTS engine, using the Customer's account of the TTS provider.

¹ Currently, this TTS service with Voice.Al Gateway has not been released, but can be offered as a trial.

3.2.2 Speech-to-Text Services

The Voice.Al Gateway uses third-party speech-to-text (STT) engines to convert audio (speech) spoken by the chatbot user, into text, which it sends to the bot.

The Voice.Al Gateway supports the following third-party STT service providers:

- Azure Speech Services
- Google Cloud Speech-to-Text
- Yandex²

The Voice.Al Gateway uses the streaming APIs of the speech services to reduce delays, resulting in a better user experience.



Note: The Voice.Al Gateway interfaces with the STT engine, using the Customer's account of the STT provider.

² Currently, this STT service with Voice.Al Gateway has not been released, but can be offered as a trial.



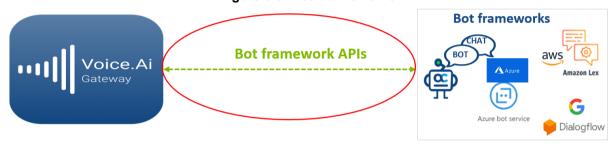
3.2.3 Bot Frameworks

The Voice.Al Gateway connects to bots through one of the following third-party chatbot frameworks:

- Microsoft Azure Bot Framework (using Direct Line 3.0 API)
- Google Dialogflow
- Amazon Lex³

The Voice.Al Gateway adapts to the above-mentioned bot framework's APIs. In addition, the Voice.Al Gateway provides a proprietary exposed HTTP-based APIs that allows **any bot frameworks or middlewares** to connect with it over HTTP/S. For a detailed description on the Voice.Al Gateway's APIs, contact AudioCodes.

Figure 3-3: Bot Frameworks



These APIs are used to initiate text-based dialogs to a specific bot implemented by the bot developer on a specific bot framework. Basic dialog initiation, authentication and text exchange is done according to the specifications of the chosen bot framework.

The Voice.Al Gateway also sends telephony metadata and notifications, as described in the document Voice.Al Gateway Integration Guide.

The bot developer can send specific actions and instructions to the Voice.Al Gateway, as described in the document <u>Voice.Al Gateway Integration Guide</u>.

Dialog initiation requires credentials to authenticate the specific bot according to the specific bot framework. These credentials must be configured on the Voice.Al Gateway per bot.

³ Currently, this bot framework has not been tested with Voice.Al Gateway, but can be offered as a trial.

4 Deployment Options

The Voice.Al Gateway offers multiple deployment options, which are all offered as a managed service by AudioCodes. This managed service includes installation, initial configuration, integration, as well as maintenance, active monitoring and fault management.



Note: For initial deployment, AudioCodes requires various information on the third-party STT, TTS, and bot frameworks used by the Customer, as well as bot-specific configuration. For more information, refer to the <u>Voice.Al Gateway Integration Guide</u>.

4.1 AudioCodes Cloud Account

The Voice.Al Gateway can be deployed and managed in AudioCodes cloud account. This deployment option uses AudioCodes infrastructure and resources required for the Voice.Al Gateway functionality. For more information on this managed service, please contact AudioCodes Support.

AudioCodes Azure account

Azure

Text-to-speech services

Azure

Text to speech APIs

Text to speech APIs

Text to speech APIs

Text to speech APIs

Speech to text APIs

Figure: Voice.Al Gateway Deployed in AudioCodes Cloud



4.2 **Customer's Data Center**

The Voice.Al Gateway can be deployed in the Customer's data center, but managed by AudioCodes. For more information on this managed service, please contact AudioCodes Support.

Text-to-speech services

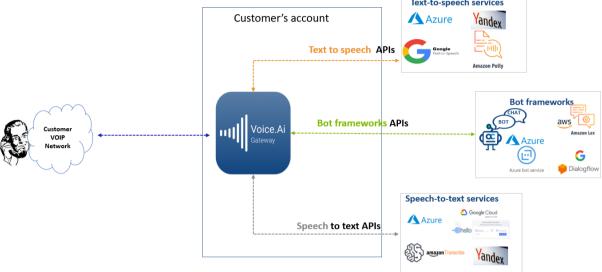


Figure: Voice.Al Gateway Deployed in Customers Data Center

4.2.1 Required Resources

The required resources when Voice.Al Gateway is deployed in the Customer's data center includes the following:

- Single Voice.Al Gateway components (for Azure):
 - VoIP-facing component: 4 vCPU, 4 Gigabyte RAM (Azure DS3_v2)
 - Connector: 2 vCPU, 4 Gigabyte RAM, 20 Giga disk (Azure D4s_v3)
 - Overall capacity: 300 sessions

Figure 4-1: Single Voice.Al Gateway



- For resiliency and scalability (thousands of sessions), more resources are required, by implementing 1+1 AudioCodes Session Border Controllers (SBC) that load-balances traffic to multiple Voice.Al Gateways:
 - 2 x SBC load balancer (1+1): 4 vCPU, 4 Gigabyte RAM (Azure DS3_v2)
 - N x Voice.Al Gateways, where each can scale up to 300 sessions and requires the following:
 - VoIP-facing component: 4 vCPU, 4 Gigabyte RAM (Azure DS3_v2)
 - Connector: 2 vCPU, 4 Gigabyte RAM, 20 Giga disk (Azure D4s_v3)

Figure 4-2: Voice.Al Gateways with Multiple SBCs





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5 CDRs and Call Transcripts

The Voice.Al Gateway can be enabled (default) to store call detail records (CDRs) of live and history chatbot calls on a local database. They can be viewed by connecting to the Voice.Al Gateway's Web-based management interface. Below shows an example of CDRs for historical chatbot call.

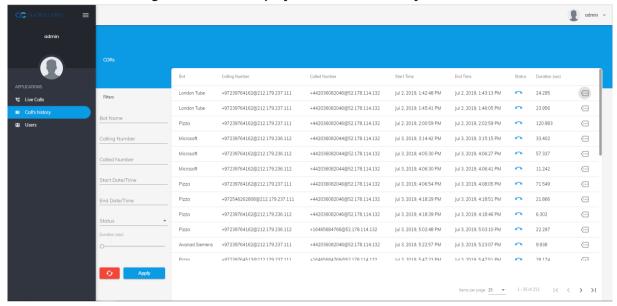


Figure 5-1: CDR Display in Voice.Al Gateway Web GUI

The Voice.Al Gateway can also be enabled (default) to store chatbot call transcripts (speech-to-text) on a local database. You can view the transcripts by connecting to the Voice.Al Gateway's Web-based management interface. Below shows an example of the transcript of a live chatbot call.

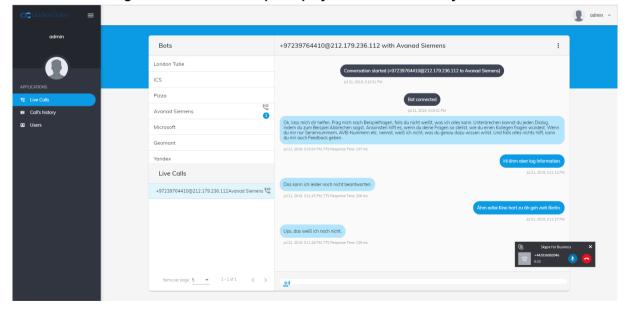


Figure 5-2: Call Transcript Display in Voice.Al Gateway Web GUI



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6 Voice.Al Gateway Features

This section describes features supported by the Voice.Al Gateway.

Some features affect the entire conversation and are configured by the Administrator. For example, audio caching of received text-to-speech results.

Some features can be applied dynamically during the conversation and are controlled by the bot using the specific bot framework's APIs.

In addition, the Voice.Al Gateway sends at various stages of the conversation events with their relevant metadata to the bot. For example, when a conversation starts, the Voice.Al Gateway sends an event message that includes the phone number of the caller and called party.

Features that can be configured by the Administrator is done by AudioCodes according to the Customer's requirements. Features that are controlled by the bot are activated by the Bot Developer using the specific bot framework's APIs (as described in the Voice.AI Gateway Integration Guide).

6.1 Bot Frameworks

The Voice.Al Gateway can be configured to operate with specific third-party bot frameworks per bot. For bot frameworks supported by the Voice.Al Gateway, see Section Bot Frameworks.

This feature is controlled by the Administrator.

6.2 Conversation Initiation Features

This section describes conversation initiation features.

6.2.1 Initial Activity to Bot

When the user starts a conversation, the Voice.Al Gateway sends an initial activity message to the bot, by default. This message includes metadata such as caller and callee number and host, and caller display name.

The Voice.Al Gateway can be configured to override the content of this activity message with a specific JSON object, or to disable this activity entirely.

This feature is controlled by the Administrator.

6.2.1.1 Passing SIP Headers to Bot

In some setups, it is important to forward SIP header values, extracted from the SIP INVITE message received by the Voice.Al Gateway, to the bot. For example, in contact centers, an ID is associated with the call which is usually sent in a SIP header, and may be required by the bot application.

The Voice.Al Gateway can be configured to extract values from the SIP INVITE message and then send them in the initial message to the bot. The values are sent as additional attributes in the detail of the initial message.

For more information, please contact AudioCodes support.



6.2.2 Connect on Bot Prompt

By default, when the user starts a conversation the Voice.Al Gateway waits for the first bot message from the bot framework before connecting the call. It is possible to configure the Voice.Al Gateway to connect immediately without waiting for the first bot message.

This feature is controlled by the Administrator.

6.2.3 Initial Message to Bot

The Voice.Al Gateway can be configured to send a textual message to the bot when the user starts a conversation. This is typically used for bots that require a specific "hello" message to start a conversation.

This feature is controlled by the Administrator.

6.2.4 Welcome Message

The Voice.Al Gateway can be configured to send an initial "welcome" text message to the user when conversation starts. This is typically required in scenarios where the bot doesn't send a welcome text prompt.

This feature is controlled by the Administrator.

6.3 Speech Features

This section describes speech-related features.

6.3.1 STT and TTS Providers

The Voice.Al Gateway can be configured to operate with specific third-party STT and TTS providers per bot. For TTS and STT providers supported by the Voice.Al Gateway, see sections Text-to-Speech Services and Speech-to-Text Services, respectively.

This feature is controlled by the Administrator.

6.3.2 Language

The language (e.g., English US, English UK, or German) is required by the TTS and STT engines for interfacing with the user. In addition, the TTS engine can be configured with a specific TTS voice (e.g., female or male).

By default, the language and voice are configured by the Administrator per bot. In addition, the bot can dynamically change the language and voice during conversation.

6.3.3 Custom Language and STT Context

Certain STT engines (e.g., Azure) allows the user to build and use its own customized language that adapts to the specific use case jargon. Other STT engines (e.g., Google) allow the user to dynamically associate STT contexts with each STT query. These methods can be used to optimize STT accuracy.

The Voice.Al Gateway can be configured to use a specific custom language (Azure). It also allows the Bot Developer to add dynamic context (Google) or change the custom speech at any stage of the conversation.

6.3.4 SSML for TTS

Speech Synthesis Markup Language (SSML) can be sent in Text-to-Speech requests to allow for more customization in the audio response, by providing details on pauses, and audio formatting for acronyms, dates, times, abbreviations, or prosody.

The bot can use SSML when sending text messages to the Voice.Al Gateway. The Voice.Al Gateway forwards the SSML-based requests to the TTS engine.



Note: SSML support depends on the specific TTS provider used in the chatbot environment.

6.3.5 Continuous ASR

By default, the STT engine recognizes the user's end of utterance according to the duration of detected audio silence (or by other means). Each recognized utterance is sent by the Voice.Al Gateway to the bot as a separate textual message.

Sometimes, the detection of end of utterance occurs too quickly and the user is cut off while speaking. For example, when the user replies with a long description that is comprised of several sentences. In such cases, all the utterances should be sent together to the bot as one long textual message.

Continuous automatic speech recognition enables the Voice.AI Gateway to collect all the user's utterances. When it detects silence for a specified duration or a configured DTMF key (e.g., # key) is pressed by the user, it concatenates the multiple STT detected utterances, and then sends them as a single textual message to the bot. In this way, the user can configure a longer silence timeout.

This feature is controlled by the Administrator. In addition, the bot can dynamically control this mode during the conversation.

6.3.6 Punctuation of STT Transcriptions

By default, the Voice.Al Gateway enables punctuation detection (e.g., periods, commas and question marks) for the STT engine's transcription and therefore, messages sent by Voice.Al Gateway to the bot includes punctuation. The Voice.Al Gateway can be configured to disable punctuation detection by the STT engine.

Note punctuation detection is supported only by specific STT providers.

This feature is controlled by the Administrator. In addition, the bot can dynamically control this feature during the conversation.



6.3.7 Barge-In

The Barge-In feature controls the Voice.Al Gateway's behavior in scenarios where the user starts speaking (interrupts or "barges-in") while the bot is saying something to the user.

When this feature is disabled (default), the Voice.Al Gateway ignores user speech input, from the detection of end of utterance until the bot has finished⁴ playing its response (or responses if the bot sends multiple consecutive response messages). Only then does the Voice.Al Gateway expect user speech input. However, if no bot response arrives within a user-defined timeout, triggered from the detection of end of utterance, speech-to-text recognition is re-activated and the user can speak again.

When the Barge-In is enabled, detection of user speech input is always active. If the user starts to speak while the bot is playing its response, the Voice.Al Gateway detects this speech and immediately stops the bot response playback and sends the detected user utterances to the bot. If there are additional queued text messages from the bot, they are purged from the queue.

This feature is controlled by the Administrator. In addition, the bot can dynamically control this feature during the conversation.

6.3.8 TTS Caching

By default, the Voice.Al Gateway caches TTS results from the TTS engine. This offers various benefits such as the following:

- Speeding up service by eliminating TTS round-trip delay
- Reducing TTS service costs by avoiding unneeded TTS transactions.

The default lifetime of the TTS cache is 24 hours (configurable). The TTS cache can also be cleared by the Administrator.

The Voice.Al Gateway can be configured to disable caching of TTS results.

This feature is controlled by the Administrator and can be disabled by the Bot Developer per text message.

⁴ The Voice.Al Gateway actually starts detecting speech a few tenths of a second before the bot finishes playing its response. This ensures that the first words of the user aren't skipped.

6.4 Call Control

This section describes call control-related features.

6.4.1 Call Transfer

A common scenario, especially when chatbots are used in contact center environments, is the need to transfer the call to a human agent. This typically occurs when the bot cannot provide an adequate response (answer) to the user's question, or if the user explicitly requests to speak to someone. In such cases, the Voice.Al Gateway provides the option to transfer the call to either a live agent or a different bot.

The transfer can be initiated by the Bot Developer, by issuing the specific transfer activity through the bot framework's APIs. The transfer activity has multiple attributes:

- Transfer target URI, which defines the new destination of the call (SIP-based live agent or a different bot). A default transfer target (URI) can be configured (by the Administrator), which is used when the bot requests a transfer without indicating a target.
- Textual description of the reason for the transfer. This is used for logging and CDRs.

There are two options to implement the transfer by the Voice. Al Gateway:

- Sending a new SIP INVITE message to the transfer target URI. (In this option, the Voice.AI Gateway remains part of the path for the call, even after the call is successfully transferred.)
- Sending a SIP REFER message to the SIP peer (e.g., contact center or SIP service provider) who initiated the original call. This REFER message causes the SIP peer to perform the transfer. In this option, the Voice.AI Gateway leaves the signaling path after the call is successfully transferred. This requires support for REFER messages by the SIP peer.

After the call is transferred, the Voice.Al Gateway disconnects the bot from the conversation.

This feature is controlled by the Bot Developer at any stage of the conversation and is done by sending the transfer activity, as described in the Voice.Al Gateway Integration Guide.

6.4.1.1 Adding SIP Headers on Call Transfer

When the bot performs a call transfer, the bot can optionally add data that is sent as SIP headers in the SIP message (REFER or INVITE) generated by the Voice.AI Gateway. This can be set by the Bot Developer as an attribute in the transfer activity, which lists names and their values that are converted into SIP headers (e.g., "My-Header: my-value") by the Voice.AI Gateway.

6.4.2 Disconnect

At any stage of the conversation, the bot can disconnect the conversation by initiating a disconnect activity (*hangup*). The conversation is disconnected by sending a SIP BYE message by the Voice.AI Gateway to the SIP peer.

The disconnect activity has an optional attribute that describes the reason for the disconnect.

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