



The Intelligent Cloud Contact Center

VoiceStream Self Service

Developer's Guide

August 2026

Learn how the VoiceStream Self Service allows developers, systems integrators, or third parties to access and receive their domain's audio streams and call events in real time.

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About Five9

Five9 is the leading provider of cloud contact center software, bringing the power of the cloud to thousands of customers and facilitating more than three billion customer interactions annually. Since 2001, Five9 has led the cloud revolution in contact centers, delivering software to help organizations of every size transition from premise-based software to the cloud. With its extensive expertise, technology, and ecosystem of partners, Five9 delivers secure, reliable, scalable cloud contact center software to help businesses create exceptional customer experiences, increase agent productivity and deliver tangible results. For more information visit www.five9.com.

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Contents

What's New	4
VoiceStream Self Service Overview	5
Logging In	5
Permissions	6
Creating Self Service Subscriptions	7
Creating a Subscription	7
Subscription and Streaming Settings	9
Destination Configuration	9
Authentication	10
Audio Streaming: Enable Third-Party and Agent-Only Conference Calls	11
Streaming Filter	11
Save and Confirm the subscription	12
(Optional) Copy and save the API Key.	13
Save and Activate	13
Manage Subscription Status	14
Pause a Subscription	15
Resume a Paused Subscription	15
Validating the Call Event URL	16
Audio Streaming Protocols, Encryption, and Troubleshooting	17
Receiving Encrypted SIP Audio Streams	17
gRPC Streaming	20
Receiving Encrypted gRPC Audio Streams	21
Troubleshooting with Event Webhooks	23
CTI Call Events and Metadata	24
CTI Call Event Delivery and Authentication	24
Supported Call Events	25
Call Attached Variables (Metadata)	25
Distinguishing Inbound from Outbound Calls	26
Status Events and Metadata	27
Stream Status Events	27
Sample Events	28
Streaming Events	28
Ended Normally Events	29
Error Events	30

What's New

Release	Change	Topic
Aug 2026	Updated images for managing subs and renamed the MediaStream Destination to StreamStatus Event Destination.	Creating Self Service Subscriptions
July 2026	Added a new section - Manage Subscription Status.	Creating Self Service Subscriptions
Feb 2026	Expanded the source IP ranges for other regions.	gRPC Streaming
Jan 2026	Added new section.	Stream Status Events
Nov 2025	• Added redaction • Added Custom filters information • Added notes about BYOT • Updated logic for secondary streaming destination (SIP) • Added Audio Streaming	Creating Self Service Subscriptions
Oct 2025	Added a new section about pausing and resuming recordings.	Pause and Resume Recordings
Jan 2025	Updated topic.	Creating Self Service Subscriptions
Oct 2022	Clarified firewall and networking configuration.	Receiving Encrypted SIP Audio Streams Receiving Encrypted gRPC Audio Streams CTI Call Event Delivery and Authentication
Mar 2022	Initial release.	

VoiceStream Self Service Overview

Use VoiceStream Self Service allows developers, systems integrators, or third parties access and receive their domain's audio streams and call events in real time.

To use VoiceStream Self Service you must:

- Create a VoiceStream Self Service subscription in the VoiceStream/MediaStream Admin Console
- Be ready to receive the audio and CTI events information that VoiceStream will be sending to the destinations configured in the subscription.

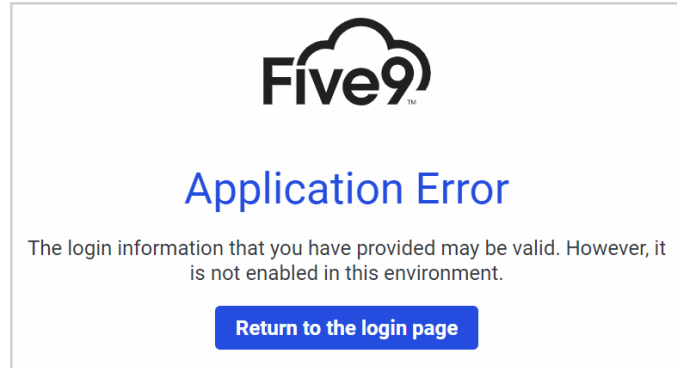
Logging In

You use the Five9 Administrator Console to log into your Five9 account and access VoiceStream or MediaStream tile.

- 1 Enter the URL that applies to your location.
 - US domains: <https://admin.us.five9.net/>
 - Canadian domains: <https://admin.ca.five9.net/>
 - UK domains: <https://admin.uk.five9.net/>
 - European domains: <https://admin.eu.five9.net/>

- 2 Enter your credentials (username and password).

This message indicates that you may have valid credentials to access Admin Console, but may not be enabled for the selected region. Ensure that the region you select represents your contact center locale.



Note: If you don't see the **VoiceStream** or **MediaStream** card on the Admin Console home page, contact your administrator and ask them to assign the required permissions (see [Permissions and Permission Sets](#)).

Permissions

Note: Only admins with permission to create and edit users can assign permissions on the Manage Users page. Contact your administrator to obtain these permissions.

This section presents the individual user permissions required to view the **VoiceStream** or **MediaStream** card on the Admin Console home page and access the described features. The permissions are:

Group	Subset	Permission	Description
voicestreamstream	configure	access	Required. View the MediaStream or VoiceStream card in Admin Console and access the Subscriptions page. You must enable this permission for the user to see the MediaStream or VoiceStream card and the Subscriptions page in Admin Console.

For more information on assigning permissions, see [Permissions and Permission Sets](#) in the [Administrator Console Administrator's Guide](#).

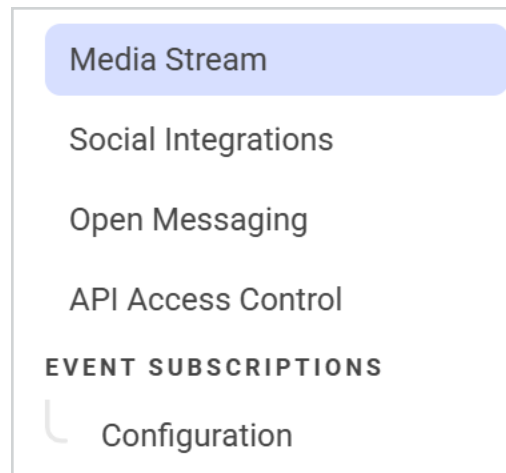
Creating Self Service Subscriptions

VoiceStream subscriptions allow streaming of call audio and events to external destinations. Each domain can support up to ten subscriptions.

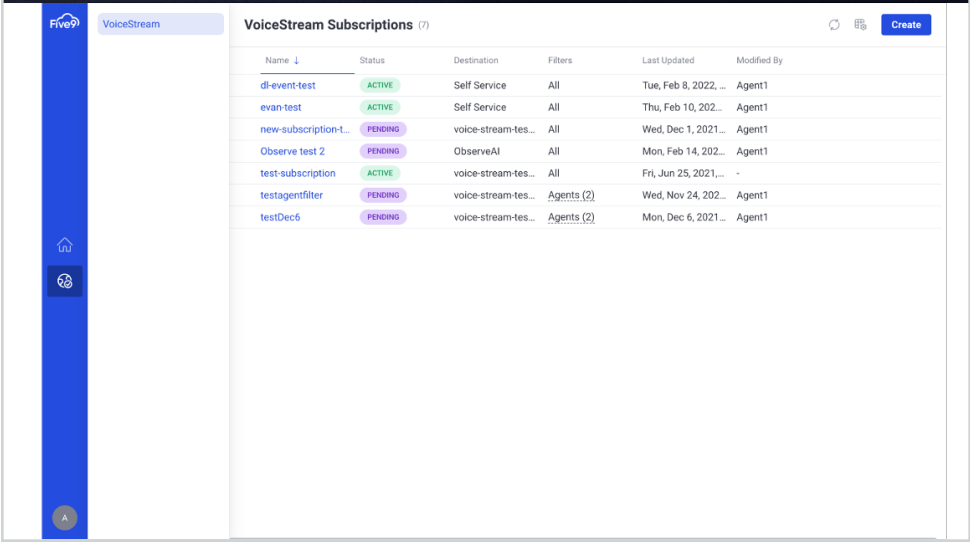
Important: Whether the UI says **VoiceStream** or **MediaStream** on the tile, side navigation, or subscriptions page, the product is the same. This name change can happen depending on your configuration. These instructions are valid for both.

Creating a Subscription

- 1 In the **Admin Console**, select the **VoiceStream** or **MediaStream** tile.
- 2 Select **MediaStream** or **VoiceStream** in the side navigation



The **MediaStream** or **VoiceStream** subscriptions page displays.

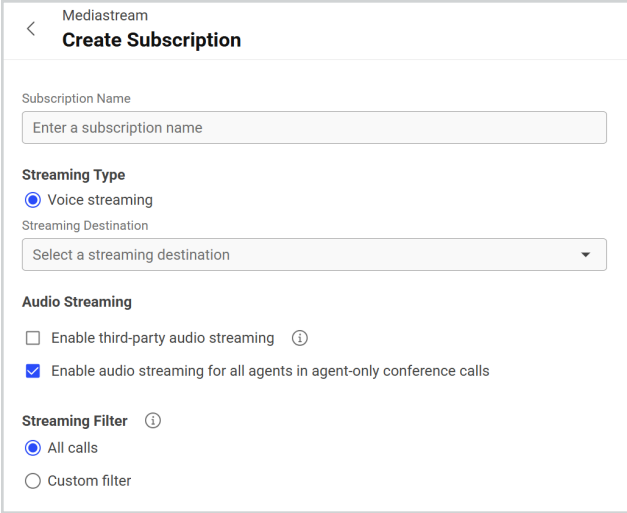


The screenshot shows the 'VoiceStream Subscriptions' table in the Five9 interface. The table has columns for Name, Status, Destination, Filters, Last Updated, and Modified By. There are 7 subscriptions listed. The first two are 'ACTIVE', and the others are 'PENDING'.

Name ↓	Status	Destination	Filters	Last Updated	Modified By
di-event-test	ACTIVE	Self Service	All	Tue, Feb 8, 2022, ...	Agent1
evan-test	ACTIVE	Self Service	All	Thu, Feb 10, 2022, ...	Agent1
new-subscription-t...	PENDING	voice-stream-tes...	All	Wed, Dec 1, 2021, ...	Agent1
Observe test 2	PENDING	ObserveAI	All	Mon, Feb 14, 2022, ...	Agent1
test-subscription	ACTIVE	voice-stream-tes...	All	Fri, Jun 25, 2021, ...	-
testagentfilter	PENDING	voice-stream-tes...	Agents (2)	Wed, Nov 24, 2022, ...	Agent1
testDec6	PENDING	voice-stream-tes...	Agents (2)	Mon, Dec 6, 2021, ...	Agent1

3 Click **Create Subscription**.

The Create Subscription page loads.



The screenshot shows the 'Create Subscription' form in the Mediastream interface. The form has the following fields and options:

- Subscription Name:** A text input field with the placeholder 'Enter a subscription name'.
- Streaming Type:** A radio button selection with 'Voice streaming' selected.
- Streaming Destination:** A dropdown menu with the placeholder 'Select a streaming destination'.
- Audio Streaming:** Two checkboxes: 'Enable third-party audio streaming' (unchecked) and 'Enable audio streaming for all agents in agent-only conference calls' (checked).
- Streaming Filter:** Two radio button options: 'All calls' (selected) and 'Custom filter'.

4 Enter a unique **SubscriptionName**.

5 **Streaming Type:** Select **Voice streaming**.

Important

Validate the call event URL before creating the VoiceStream Self Service subscription.

Subscription and Streaming Settings

- 1 **Name:** Enter a unique name for the subscription.
- 2 Under **Streaming Destination:** Select **Other (Self Service)**.
- 3 **Streaming Protocol:** Choose between **SIP** or **gRPC**.

Note

Recommendation - Use gRPC if your organization does not have Telephony/SIP experience.

Destination Configuration

Subscription Name

MW_grpc

Subscription ID: 97f

7b087

Streaming Type

☒ Voice streaming

Streaming Destination

Other (Self Service)

Streaming Protocol

gRPC

Primary Streaming Destination

grpc-sip-streaming-

:8090

CTI Call Event Destination

https://e05c827487f1715e7e85cb530f

Stream Status Event Destination

https://e05c827487f171

.pipedream.net

https://e05c827487f1715e7e85cb530f8526d0.m.pipedream.net/subscriptions/97f245df-d2bc-45bd-8f1c-ca44ce27b087

Trust Token

Enter New Token

API Key

- Enter a **Primary Streaming Destination** using the following formats:

- If you selected SIP: `sip:012:345:6789`
- If you selected gRPC: `grpc.example.com:port`

This is the destination where call audio is streamed to the configured external service.

- *(Optional SIP only)* Enter a **Secondary Streaming Destination**.

When a Secondary Streaming Destination is configured for SIP, the system initiates streaming by randomly selecting one of the available URLs. If the selected destination fails, the system automatically fails over to the next destination in the list to maintain streaming continuity.

- Enter the **CTI Call Event Destination** in the format **https://**.

The service will send events, such as Call on Hold, Call Transferred, and so on to this destination.

- Enter a **Stream Status Event Destination** in the format **https://**.

Five9 will post success/failure traffic status to this URL.

Important

Check stream status first when troubleshooting audio streaming issues.

Authentication

If you are using a **CTI Call Event Destination**, enter a trust token (up to 256 characters).

Five9 passes this token in the SIP or gRPC headers so your service can verify that traffic is coming from Five9. A trust token is required; enforcing token validation on your endpoint is **optional** but strongly recommended.

Audio Streaming: Enable Third-Party and Agent-Only Conference Calls

To stream transferred calls and other audio to external services, enable the third-party audio streaming option.

- **Enable third-party audio streaming:** *(Optional)* Select this to stream both transmit (Tx) and receive (Rx) audio when one agent speaks with one third party. This includes warm transfers or conferences where the customer leaves and only the agent and third party remain.
- **Enable audio streaming for all agents in agent-only conference calls:** **Default.** Select this option to stream both transmit (Tx) and receive (Rx) audio for agents in conferences that include only agents. Turn off this setting to stream only Tx audio for each agent. If a customer or third party joins the conference, all agents stream both Tx and Rx audio regardless of this setting.

Streaming Filter

Filter to control which calls are streamed.

- **All calls:** All inbound and outbound agent-customer calls will be streamed.
- **Custom Filters**
Filter by:
 - **Campaigns:** All inbound or outbound calls handled as part of the selected campaigns will be streamed.
To search for a specific campaign in the domain, enter the campaign name in the search box. Enter three or more characters to search for campaigns by name.
 - **Agents:** All inbound or outbound calls handled by the selected agents will be streamed.
To search for a specific agent in the domain, enter the agent's name or user name in the search box.

Note

Agent-to-agent calls or calls that agents have transferred to third party destinations will **not** be streamed **unless configured in the audio streaming options**.

- **Skills:** All inbound or outbound calls handled by agents assigned to specific skill groups.
 - 1 Click on the **Add Skill** button to open the **skill selection panel**.
 - 2 To search for a specific skill in the domain, enter the skill name in the **search box**.
 - 3 From the list of skills displayed, check the boxes next to desired skills (e.g., `billing_inquiries`, `order_tracking`).

Save and Confirm the subscription

- 1 Click **Create** to save the subscription.

If the Create button does not turn green, ensure that you have completed all required fields and are using the correct formats.

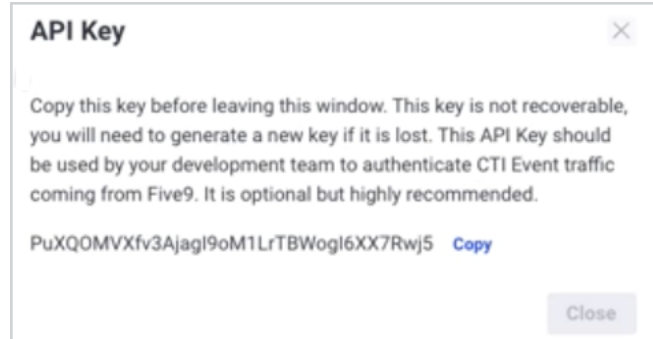
A confirmation screen displays.

Important

The subscription is not created until the CTI Call Event Destination and Trust Token pass URL validation, as specified in the VoiceStream Developer's Guide. This validation step is required for Five9 to verify that the configured CTI Event Destination is a valid https URL. An error is displayed if these cannot be validated.

- 2 Click **I approve** to confirm the beginning of your subscription.

After approving the consent form, the API Key dialog displays.



(Optional) Copy and save the API Key.

Use this `API Key` to authenticate that the CTI Event Traffic is coming from Five9. Using an `API Key` is optional but highly recommended.

If you lose your API Key, you can generate a new one by editing the subscription.

Once the subscription becomes **Active**, any call that matches the filter criteria will be streamed.

Save and Activate

- Select **Done** to save the subscription. The subscription displays in the TranscriptStream Subscriptions list as Pending until the partner app confirms that Five9 can begin streaming. Delays vary by partner.
- Click the **Refresh** button to refresh the page and determine if the subscription status is **Active**.

Once the subscription becomes **Active**, any transcript that matches the filter criteria streams to the Partner app.

Manage Subscription Status

Note

Pausing and resuming applies to both Voice and Text (TranscriptStream) subscription types

You can temporarily stop a subscription from streaming without deleting it. This is useful when you need to perform maintenance on your receiving endpoint or wish to halt streaming for business reasons.

MediaStream Subscriptions ⁷					
☐	Name ↓	Status	Streaming Type	Destination	Filters
☐	DebTest	 Pending	Voice	voicestream-test...	Agents (1)
☐	demo-transcript-su...	 Active	Text	Salesforce	All
☐	JsonArray	 Active	Text	ServiceNow	Campaigns (1)
☐	service-now	 Active	Text	ServiceNow	All
☐	TempTempTemp	 Active	Text	ServiceNow	All
☐	test-sub2	 Paused	Voice	Self Service	All
☐	test-voice	 Active	Voice	Self Service	Campaigns (557)

Subscriptions can be in one of three states:

Status	Toggle Appearance	Behavior
Active	Filled blue toggle	The subscription is live and streaming data for all new calls matching its filter criteria.
Paused	Grey outline toggle	The subscription is inactive. New calls are not streamed, but the subscription configuration (filters, destination, streaming type) is preserved.
Pending	Grey toggle (disabled)	The subscription has been created but has not yet been activated. The toggle cannot be changed until the subscription is fully provisioned.

Pause a Subscription

Any calls that are already in the process of streaming when the subscription is paused will continue to stream until they are completed. You will be billed for these in-progress calls.

To pause a subscription:

- 1 Navigate to the **MediaStream Subscriptions** page.
- 2 Locate the subscription you wish to pause in the list.
- 3 Click the toggle switch under the **Status** column to change it from **Active** to **Paused**.

The subscription status updates immediately. No new calls matching the filter criteria will be streamed while the subscription is paused.

Resume a Paused Subscription

Warning

The subscription begins streaming immediately for any new calls matching its filter criteria. Calls that occurred while the subscription was paused are not retroactively streamed.

When you are ready to restart streaming, you can reactivate a paused subscription at any time. The subscription resumes using its existing configuration — no reconfiguration is required.

To resume a subscription:

- 1 Navigate to the **MediaStream Subscriptions** page.
- 2 Locate the paused subscription in the list.
- 3 Click the toggle switch under the **Status** column to change it from **Paused** to **Active**.

Validating the Call Event URL

The **CTI Call Event Destination** and the **Trust Token** string configured during the subscription process must return valid for Five9 to verify that the destination is a valid HTTPS URL and managed by your organization.

Five9 issues an HTTP GET to the **CTI call event URL** specified during the subscription process. This HTTP GET must have a successful response that includes the SHA256 hash of the specified trust token. Unsuccessful responses void the subscription process.

For example, if you entered `"https://webhooks.example.com/five9"` as the CTI call event URL and `"trust"` as the trust token, an HTTP GET to `"https://webhooks.example.com/five9"` must respond with `"f796e2f28ae5811737ccb8233f34e09f8bb75d2511a135543d1ca37be0199a1d"`.

Note

The **Content-Type** in the response must be `application/JSON`. Since the response is a JSON string, the quotes in the above example are required.

You can derive the SHA-256 for the trust token using available libraries. For example:

```
String shaToken =  
org.apache.commons.codec.digest.DigestUtils.sha256Hex  
(trustToken)
```

On successful subscription creation, VoiceStream streams inbound or outbound calls that match the **Streaming Filter** specified in the subscription. The audio and CTI events are sent in real time to the specified **Streaming Destination**.

Audio Streaming Protocols, Encryption, and Troubleshooting

[Receiving Encrypted SIP Audio Streams](#)
[gRPC Streaming](#)
[Receiving Encrypted gRPC Audio Streams](#)
[Troubleshooting with Event Webhooks](#)

Receiving Encrypted SIP Audio Streams

Note

Five9 strongly recommends using gRPC streaming if you do not have experience with SIP telephony. See [Receiving Encrypted gRPC Audio Streams](#).

- 1 When you create a VoiceStream Self Service Subscription, you must select “SIP” as the **Streaming Protocol** to receive SIP audio streams.

The SIP streaming protocol supports the G711 codec, which uses an 8 kHz sampling rate. This protocol streams both inbound and outbound calls and sends two separate streams as the agent inbound audio and agent outbound audio. The SIP To header in the SIP Invite message provides a Five9 CallID unique to a customer domain, which allows the two separate streams to be associated. The Call Events passes the Call ID, allowing the streams and events to be mapped.

Note

Call IDs are only unique within a domain. They can be used across different domains.

- 2 Set the **Streaming Destination** option to accept traffic from the following IP address ranges:

147.124.160.0/19 (Five9-owned IP address space)

The RTP ports used are determined by the SDP offer/answer negotiation.

SIP INVITEs will include the following information:

- **To Header URI** - This header specifies the desired recipient of the request in the format {callId}@{sip.example.com}.
The URI, along with the X-F9-DomainId header (described below), determine the unique Call ID and associates this SIP call with the call event webhooks.

Note

The Call ID is unique within a domain. The VoiceStream directive defines the domain part of the URI.

- **From Header URI** - This header specifies the user of the request in the format {Agent|Customer}.

The URI allows the recipient to determine if this call contains the agent or customer streaming audio.

Note

In some agent consult transfers, a second agent's audio may be sent on a call where the From header indicates it is the customer. In this case the domain part of the **From Header URI** will be the Five9 public IP address system initiating the INVITE. A complete **From Header URI** example is: Agent@10.1.2.3.

- **X-Five9-SubscriptionId Header** - This header contains the identifier for the VoiceStream subscription, which matches the call and allows streaming.
- **X-Five9-DomainId Header** - This header contains the Five9 domain identifier for the call. The domain identifier indicates the Five9 customer to receive the call.
- **X-Five9-AgentId header** - This header contains the Five9 agent ID associated with the call. This is useful in conference scenarios where multiple agents are on the same call.
- **Trust Token** - This VoiceStream trust token string of up to 256 characters allows recipients to authenticate that the audio streams are coming from Five9. The token is passed in the SIP INVITE header X-F9-TRUST-TOKEN when streaming calls.

This example shows the **SIP INVITE** message for the agent stream:

```
INVITE sip:600000000004516@asterisks-cloud-gesit-
```

```

1.five9qe.cloud;transport=tls SIP/2.0
Via: SIP/2.0/TLS 34.133.77.94:5061;rport;branch=z9hG4bKeK3S12aQHBNXc
Max-Forwards: 70
From: "Agent" <sip:Agent@34.133.77.94>;tag=eeF4NpFQQ69Nm
To: <sip:600000000004516@asterisks-cloud-qesit-1.five9qe.cloud;transport=tls>
Call-ID: 57170b5e-1524-123b-3585-4201c0a81409
CSeq: 48578284 INVITE
Contact: <sip:mod_sofia@34.133.77.94:5061;transport=tls>
User-Agent: FreeSWITCH-mod_sofia/20.21.8-dev~64bit
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE,
REGISTER, REFER, NOTIFY
Supported: timer, path, replaces
Allow-Events: talk, hold, conference, refer
Content-Type: application/sdp
Content-Disposition: session
Content-Length: 687
X-F9-TRUST-TOKEN: trustMe
X-F9-DomainId: 8700000000000371
X-F9-SubscriptionId: d8dd71a9-82b3-4f2c-aec8-d2f2df52c2f5
X-Five9-Trust-Token: trustMe
X-Five9-DomainId: 8700000000000371
X-Five9-AgentId: 8700000000005326
X-Five9-SubscriptionId: d8dd71a9-82b3-4f2c-aec8-d2f2df52c2f5

```

Use these network and firewall configurations to enable secure SIP connections:

Traffic Type	Source IP Ranges	Notes
Audio	US: 147.124.160.0/19	You must have a valid publicly-signed certificate, especially when setting the directive to point to an IP address.
	EU and UK: 147.189.224.0/20 147.124.160.0/19	
CTI Events and other webhooks	0.0.0.0/0	There are no IP ranges to whitelist for any webhook traffic, and webhooks could originate from anywhere. Partners: Use the API Key provided during Self Service subscription creation to authenticate that this traffic is from Five9.

You must set standard SIP TLS with SRTP encryption over the public internet to receive streams.

For SIP over TLS, Five9 requires TLS v1.2 or higher, and the ability to negotiate a cipher in the `TLS_ECDHE_RSA` family. Five9 also requires a certificate signed by a trusted public Certificate Authority.

Note

The certificate must match the VoiceStream Subscription **Streaming Destination**.

For SRTP, Five9 supports the following ciphers:

- `AEAD_AES_256_GCM_8`
- `AEAD_AES_128_GCM_8`
- `AES_CM_256_HMAC_SHA1_80`
- `AES_CM_192_HMAC_SHA1_80`
- `AES_CM_128_HMAC_SHA1_80`

gRPC Streaming

gRPC is an alternative option to SIP streaming. When using gRPC, VoiceStream plays the role of gRPC client and always initiates a connection to the 3rd party application, which plays the role of gRPC server.

The gRPC method uses bidirectional streaming messages, which allow the remote server to, optionally, provide feedback or send call events (`START`, `PAUSE`, `STOP`, `RESTART`, and so on) to VoiceStream to control audio streaming. See the [Receiving Encrypted gRPC Audio Streams](#) for details.

Important

VoiceStream does not support multi-channel streaming. VoiceStream transmits each distinct audio stream over a separate gRPC session: one for audio from the agent, and one for audio to the agent.

A Five9 Call ID, unique for each customer domain, associates the two separate streams.

The Call ID is also passed as part of the Call Events so that the streams and events can be mapped together. When using gRPC as your streaming protocol, provide a valid gRPC hostname and port in the **Primary Streaming Destination**, such as `grpc.example.com:443`. A valid CA-signed certificate is required to establish a TLS session.

Use these network and firewall configurations to enable secure gRPC connections:

Traffic Type	Source IP ranges	Notes
Audio	US: 147.124.160.0/19	You must have a valid publicly-signed certificate, especially when setting the directive to point to an IP address.
	EU and UK: 147.189.224.0/20 147.124.160.0/19	
	JP / AU / IN: 202.92.198.0/23	
	BR: 34.39.142.160/31	
CTI Events and other webhooks	0.0.0.0/0	There are no IP ranges to whitelist for any webhook traffic, and webhooks could originate from anywhere. Partners: Use the API Key provided during Self Service subscription creation to authenticate that this traffic is from Five9.

For further information see [Five9 Internet Domains and IP Addresses](#).

Receiving Encrypted gRPC Audio Streams

Note

This capability requires familiarity with gRPC, protobuf, and application development using gRPC.

- 1 When you create a VoiceStream Self Service Subscription, select gRPC as the streaming protocol to receive gRPC audio streams.
- 2 Review the [Five9 voice streaming proto file](#), provided separately.
- 3 Generate source code out of the proto file in your voice.proto preferred language using the protoc compiler.
- 4 Compile the source code and integrate it to your application.

gRPC headers include:

- Call ID: `x-five9-call-id`
Trust Token: `x-five9-trust-token`

VoiceStream sends a message containing the following call information:

- `domainId` - The customer `domainId` associated with the call.
- `campaignId` - The `campaignId` associated with the call.
- `agentId` - The agent identifier associated with the call.
- `vccCallId` - The unique identifier used by Five9 to identify the call.
- `trustToken` - The partner supplied token used to validate the authenticity of the gRPC connection. The `trustToken` is specified as part of the directive.
- `subscriptionId` - The unique identifier of the VoiceStream Self Service Subscription.
- `callLeg` - An indicator of the source of the media being streamed.
 - `agent` - The media is from the agent.
 - `customer` - The media is from the customer.
 - `supervisor` (reserved for future use)
- `encoding` - An indicator of the encoding of the media being streamed.
 - `LINEAR16` - Uncompressed 16-bit signed little-endian samples (Linear PCM).
- `sample_per_second` - The sample rate in hertz (8000).

Important

In parallel to the gRPC channel, VoiceStream CTI Call Event Webhooks that provide CTI events and metadata will also be sent.

Troubleshooting with Event Webhooks

Note

These are different from VoiceStream CTI Call Event Webhooks. See [CTI Call Event Delivery and Authentication](#).

VoiceStream Event Webhooks provide feedback on the attempts Five9 makes to stream audio. Webhooks provide success/failure notifications for each call setup attempt, and when there is a failure, a category and details regarding the failure. If there is a failure to negotiate TLS, SRTP, and so on, for both SIP and gRPC flows, these webhooks contain that information.

Note

For customer support tickets, Five9 Support requires the information in these webhooks for troubleshooting.

The VoiceStream Event Webhook URL is constructed from the call event URL that you supply. For example, if your call event URL is `https://events.example.com/voicestream`, the VoiceStream Event Webhooks will be sent as an HTTP POST to `https://events.example.com/voicestream/subscriptions/{subscriptionId}` with a query parameter of `event=StreamStatusNotification`.

CTI Call Events and Metadata

Five9 offers an event Self Service subscription to allow delivery of server-side CTI events through webhooks.

[CTI Call Event Delivery and Authentication](#)

[Supported Call Events](#)

[Call Attached Variables \(Metadata\)](#)

[Distinguishing Inbound from Outbound Calls](#)

CTI Call Event Delivery and Authentication

Self-service customers must specify a destination for events during the VoiceStream Self Service subscription process. You can also specify a secondary destination. CTI events are then always sent to both destinations.

To authenticate that the webhooks are coming from Five9, a shared secret is established and is sent in all webhooks, in the header **X-F9-APIKEY**. Five9 generates the API Key and provides it to customers during the Self Service subscription process. There is a unique API Key for each VoiceStream Self Service Subscription.

Use these network and firewall configurations to enable secure connections:

Traffic Type	Source IP Ranges	Notes
Audio	US: 147.124.160.0/19 EU and UK: 147.189.224.0/20 147.124.160.0/19	You must have a valid publicly-signed certificate, especially when setting the directive to point to an IP address.
CTI Events and other webhooks	0.0.0.0/0	There are no IP ranges to whitelist for any webhook traffic, and webhooks could originate from anywhere.

Traffic Type	Source IP Ranges	Notes
		Partners: Use the API Key provided during Self Service subscription creation to authenticate that this traffic is from Five9.

Supported Call Events

These are the supported call events:

- Call Connected/ Accepted
- Put On hold/ Retrieved from Hold
- Parked / Retrieved from Park
- Consult Initiated/ Consultant Connected/ Consult Completed
- Transfer Initiated/ Transfer Completed
- Conferenced Initiated/ Conference Completed
- Call Ended/ Agent Ended

For more details, download the [call events JSON payload file](#). Please note that the file covers other events that are not applicable/available in the VoiceStream CTI Call Event webhooks.

Call Attached Variables (Metadata)

Call attached variables (CAVs) are additional call metadata passed alongside the call events as part of the payload.

You can configure CAVs that will be passed on as part of the call event webhooks. Call event webhooks may be delivered out of order and are not guaranteed to only be delivered once. You can use the timestamp and event ID in the header to assist with re-ordering and removing duplication if needed.

Distinguishing Inbound from Outbound Calls

`Call.type` and `Call.type_name` are sent using CAVs and you can configure those values in the VCC Admin Console. You can set these:

- OUTBOUND
- INBOUND
- TEST
- AGENT
- MANUAL
- PREVIEW
- AUTODIAL
- INTERNAL
- INBOUND_VM
- OUTBOUND_VM
- INTERNAL_VM

The values that distinguish outbound and inbound calls are:

- Outbound
 - MANUAL = Manual Outbound Call
 - OUTBOUND = Automatic Outbound Call
 - AUTODIAL = Automated Outbound Call for Autodial Campaign
- Inbound
 - INBOUND = Inbound Call

You can ignore other values as they refer to Test calls, Agent-to-Agent calls (not supported), Echo call (Preview), or Voicemail (VM).

Status Events and Metadata

VoiceStream sends status events to notify your system about the state of each media-streaming call leg. These events help you understand when streaming starts, ends, or fails. Your system can use these events to update dashboards, trigger workflows, or troubleshoot call issues.

For more information on the status updates using the PartnerStreamStatus, see [Stream Status Events](#).

Stream Status Events

VoiceStream sends `StreamStatusNotification` events to report the real time status of each media-streaming call leg. These events help your system understand when streaming begins, ends, or encounters an error. VoiceStream delivers these notifications to the webhook URL you configure as the MediaStream Event Destination in the directive.

These events are useful when you need to start AI processing when audio becomes available, handle errors quickly, or finalize downstream workflows when streaming completes.

Use case: A Five9 customer uses `StreamStatusNotification` events to start real time transcription as soon as VoiceStream begins streaming and to finalize summaries or analytics when streaming ends. This ensures the system processes complete and accurate audio for both the Agent and Customer call legs.

Event Name: `StreamStatusNotification`

Webhook URL Format: `https://<voicestreamEventUrl_From_Directive>/subscriptions/<voicestream-subscription-id>?event=StreamStatusNotification`

VoiceStream Self Service delivers each event to the webhook URL you configured as the **MediaStream Event Destination** in the directive.

In the UI, `voicestreamEventUrl_From_Directive` appears as **MediaStream Event Destination**.

Each `StreamStatusNotification` event includes the following fields:

- `callId` – Unique identifier for the call.
- `domainId` – Customer domain identifier.
- `callLeg` – Indicates whether the event applies to the Agent or Customer call leg.
- `status` – Current streaming status (e.g., `STREAMING`, `ENDED_NORMALLY`, `ERROR`).
- `failureType` – Specifies the error category when the event includes an error.
- `failureDetail` – Provides additional information for troubleshooting.
- `destinationUrl` – The MediaStream Event Destination URL used for the webhook.
- `occurredOn` – Timestamp (UTC) when the event occurred.

Sample Events

Streaming Events

Agent

```
{
  "callId": "600000000054602",
  "domainId": "domain-id",
  "callLeg": "Agent",
  "status": "STREAMING",
  "failureType": "",
  "failureDetail": "",
  "destinationUrl": "voicestreamEventUrl_From_Directive",
  "occurredOn": "2025-12-12T01:15:37.917Z"
}
```

Customer

```
{
  "callId": "600000000054602",
  "domainId": "domain-id",
  "callLeg": "Customer",
  "status": "STREAMING",
  "failureType": "",
  "failureDetail": "",
  "destinationUrl": "voicestreamEventUrl_From_Directive",
  "occurredOn": "2025-12-12T01:15:37.917Z"
}
```

Ended Normally Events

Agent

```
{
  "callId": "600000000054602",
  "domainId": "domain-id",
  "callLeg": "Agent",
  "status": "ENDED_NORMALLY",
  "failureType": "",
  "failureDetail": "",
  "destinationUrl": "voicestreamEventUrl_From_Directive",
  "occurredOn": "2025-12-12T01:15:37.917Z"
}
```

Customer

```
{
  "callId": "600000000054602",
  "domainId": "domain-id",
  "callLeg": "Customer",
  "status": "ENDED_NORMALLY",
  "failureType": "",
  "failureDetail": "",
  "destinationUrl": "voicestreamEventUrl_From_Directive",
  "occurredOn": "2025-12-12T01:15:37.917Z"
}
```

Error Events

Agent

```
{
  "callId": "600000000054602",
  "domainId": "domain-id",
  "callLeg": "Agent",
  "status": "ERROR",
  "failureType": "OTHER",
  "failureDetail": "null",
  "destinationUrl": "voicestreamEventUrl_From_Directive",
  "occurredOn": "2025-12-12T01:15:37.917Z"
},
```

Customer

```
{
  "callId": "600000000054602",
  "domainId": "domain-id",
  "callLeg": "Customer",
  "status": "ERROR",
  "failureType": "OTHER",
  "failureDetail": "null",
  "destinationUrl": "voicestreamEventUrl_From_Directive",
  "occurredOn": "2025-12-12T01:15:37.917Z"
}
```