

Leveraging Amazon Chime Voice Connector for SIP Trunking

April 2020



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Abstract

This whitepaper outlines the features and benefits of using Amazon Chime Voice Connector. Amazon Chime Voice Connector is a service that carries your voice traffic over the internet and elastically scales to meet your capacity needs.

This whitepaper assumes that you are already familiar with Session Initiation Protocol (SIP) trunking.

Introduction

Amazon Chime Voice Connector is a pay-as-you-go service that enables companies to make and receive secure, inexpensive phone calls over the internet using their on-premises telephone system, such as a private branch exchange (PBX). The service has no upfront fees, elastically scales based on demand, and supports calling both landline and mobile phone numbers in over 100 countries.

Getting started with Amazon Chime Voice Connector is as easy as a few clicks on the AWS Management Console and then employees can place and receive calls on their desk phones in minutes.

About Amazon Chime Voice Connector

Amazon Chime Voice Connector uses standards-based Session Initiation Protocol (SIP) and calls are delivered over the internet using Voice over Internet Protocol (VoIP).

Amazon Chime Voice Connector does not require dedicated data circuits and can use a company's existing internet connection or use [AWS Direct Connect](#) public virtual interface for the SIP connection to AWS. The configuration of SIP trunks can be performed in minutes using the AWS Management Console or the [AWS SDK](#).

Amazon Chime Voice Connector offers cost-effective rates for outbound calls. In addition, calls to Amazon Chime audio conferences, as well as calls to other companies using Amazon Chime Voice Connector, are at no additional cost. With this service, companies can reduce their voice calling costs without having to replace their on-premises phone system.

Service Benefits

Amazon Chime Voice Connector provides the following benefits.

Low Cost and Reduced TCO

Amazon Chime Voice Connector provides an easy way to move telephony to the cloud without replacing on-premises phone systems. Using the service, you can reduce your voice calling costs by up to 50% by eliminating fixed telephone network costs and simplifying your voice network administration. To estimate the cost of using Amazon Chime Voice Connector, see the [Amazon Chime Pricing](#) page.

Amazon Chime Voice Connector allows you to use SIP trunking infrastructure on-demand with voice encryption available at no extra charge. The elastic scaling of the service eliminates the need to overprovision SIP and/or time-division multiplexing (TDM) trunks for peak capacity. You only pay for what you use and can track your telecom spending in your monthly AWS invoice. There is no charge for creating SIP trunks and no subscription or per-user license fees or concurrent conversation fees.

The following table shows a cost comparison of Amazon Chime Voice Connector with other service offerings.

Table 1: Cost Comparison of Amazon Chime Voice Connector and Other SIP Offerings

Monthly Cost	Offering 1	Offering 2	Offering 3	Amazon
Inbound call/minute	\$0.0000	\$0.0000	\$0.0045	\$0.0022
Outbound call/minute	\$0.0080	\$0.0120	\$0.0070	\$0.0049
Concurrent call charge per	\$0.8180	\$1.0907	\$0	\$0
Number rental	\$0.10	\$1.00	\$1.00	\$1.00
350 minutes/month	\$1.87	\$2.80	\$2.16	\$1.40
Normalized Pricing/month	\$2.78	\$4.89	\$3.16	\$2.40
Potential savings with Amazon Chime Voice Connector	14.67%	68.31%	27.34%	N/A

Flexible and On-Demand

Your telecom administrator uses the AWS Management Console to create the Amazon Chime Voice Connector and your organization can begin sending and receiving voice calls in minutes. You can route as much voice traffic to it as needed or desired, within the [AWS service quotas](#).

You can also choose to keep your inbound phone numbers, also known as Direct Inward Dialing (DID) numbers, with your current service provider or contact AWS Support to port the numbers to Amazon Chime Voice Connector and take advantage of the [Amazon Chime dial-in rates](#).

Use Case Scenarios

You can use Amazon Chime Voice Connector to send voice traffic from your on-premises PBX to AWS (outbound calls to public switched telephone network [PSTN] numbers), and to receive voice calls from your Voice Connector to your PBX (inbound calls from DID numbers), or both.

In both call flow scenarios (outbound and/or inbound calls), you can connect to Amazon Chime Voice Connector using your existing telephony devices. These devices can be a Session Border Controller (SBC), an IP PBX, or a media gateway.

In the following examples, an SBC is the network element that is used to connect the SIP trunks.

- [Outbound Calling Only](#)
- [Inbound and Outbound Calling](#)
- [Inbound and Outbound Calling Exclusively](#)
- [Inbound Calling Only](#)

Outbound Calling Only

In this deployment model, you benefit from the low-cost outbound calling to PSTN phone numbers. Calls from your PBX to Amazon Chime Voice Connector incur no outbound telephony charges. You can use Amazon Chime Voice Connector for outbound calling in conjunction with the existing connection to your current SIP trunking provider. Your inbound calling remains unchanged. In this use case, Amazon Chime Voice Connector is typically configured as a *route* for high availability in case the default route to the Existing SIP Trunking Provider is unavailable, as well as for least-cost-routing (LCR) within the IP PBX or SBC.

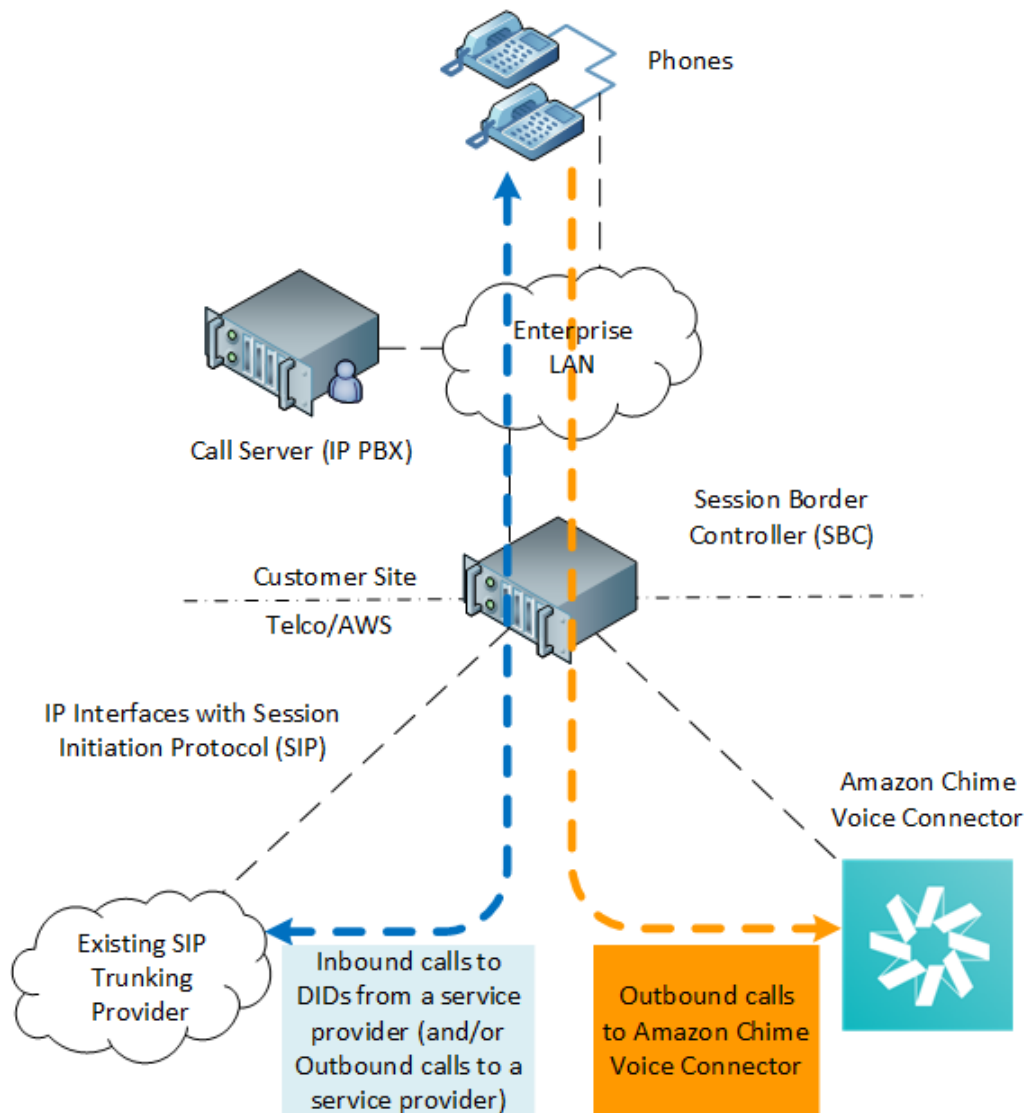


Figure 1: Outbound Calling Only

Inbound and Outbound Calling Exclusively

In this deployment model, you use Amazon Chime Voice Connector for both inbound and outbound voice calling. This eliminates the need for your existing SIP trunks and reduces network complexity. For inbound calling, you acquire new phone numbers from AWS, or port the existing phone numbers from your current service provider. For outbound calling, use Amazon Chime Voice Connector as the single route for all outbound voice calls from your PBX. Amazon Chime Voice Connector has built-in call failover, service resilience, and high availability features.

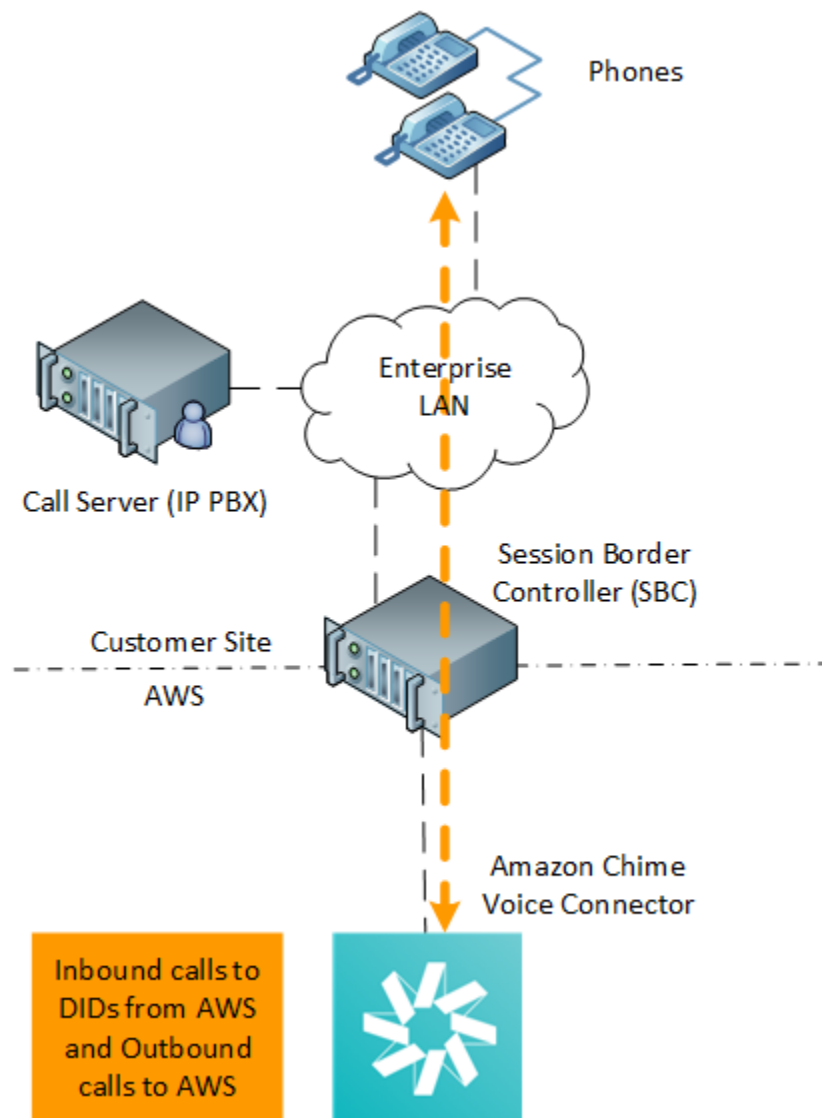


Figure 3: Inbound and Outbound Calling Exclusively

Inbound Calling Only

In this deployment model, you use Amazon Chime Voice Connector only for inbound voice calling. For inbound calling only, you acquire new phone numbers from AWS, or port existing phone numbers from your current service provider. For inbound calling only, you benefit from the routing features provided by Amazon Chime Voice Connector, such as load balancing, failure mitigation mechanisms, and easy phone number inventory management, using the AWS Management Console or the [AWS SDK](#). For more information on these features, see [Call Routing with Load Sharing](#) and [Phone Number Inventory Management](#).

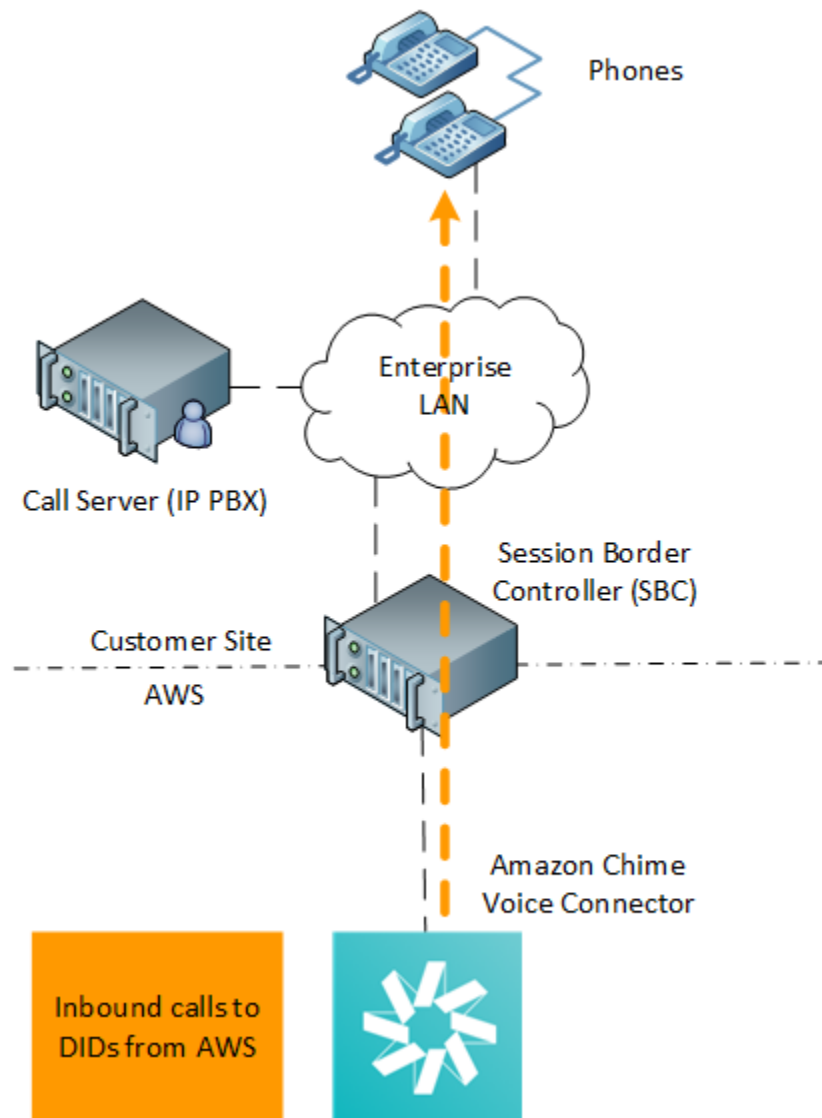


Figure 4: Inbound Calling Only

Service Features

Reliability and Elasticity

Amazon Chime Voice Connector delivers highly available and scalable telephone service for inbound calls to your on-premises telephone system, outbound calls to Amazon Chime Voice Connector, or both. Using Amazon Chime Voice Connector Groups, you can configure multi-region failover for inbound calls from PSTN calls to your Amazon Chime Voice Connectors. Additionally, Amazon Chime Voice Connector provides a load-sharing mechanism for inbound calls to your on-premises phone system using priority and weight.

AWS SDK

The [AWS SDK](#) allows you to perform and automate key administrative tasks, such as managing phone numbers, Amazon Chime Voice Connectors, and Amazon Chime Voice Connector Groups.

Security – Call Encryption

Call encryption is a configurable option for each Amazon Chime Voice Connector and is provided at no additional charge. If encryption is enabled, voice calls are encrypted between the service and your SIP infrastructure. Transport Layer Security (TLS) is used to encrypt the SIP signaling and Secure Real Time Protocol (SRTP) is used to encrypt the media streams. To learn about the SIP Signaling Specifications, see [Appendix B: SIP Signaling Specifications](#).

IP Whitelisting and Call Authentication

You can authenticate voice traffic to Amazon Chime Voice Connector by using the mandatory Allow List (IP whitelisting) and by using the optional Digest Authentication (as described in [RFC 3261, section 22](#)).

Call Detail Records (CDR)

Shortly after each call, Amazon Chime Voice Connector stores the Call Detail Record (CDR) as an object in your own Amazon Simple Storage Service ([Amazon S3](#)) bucket. You configure the S3 bucket in the AWS Management Console. You can retrieve the CDR records from Amazon S3 and import them into a VoIP billing system. To learn

about the CDR schema, see [Appendix A: Call Detail Record \(CDR\) Specifications](#). For the current CDR format, see the [Amazon Chime Voice Connector](#) documentation.

Phone Number Inventory Management

You can manage phone numbers using the AWS Management Console and the AWS SDK. You can manage your existing phone number inventory, order new numbers, review pending transactions, and manage deleted phone numbers. Contact AWS Support to port existing phone numbers.

Outbound Caller ID Name

Support for Outbound Caller ID Name (CNAM) is a component of caller ID that displays your name or company name on the Caller ID display of the party that you are calling. Amazon Chime Voice Connector makes it easy to set calling names for Amazon Chime Voice Connector phone numbers using the AWS Management Console. Amazon makes the necessary changes to the Line Information Database (LIDB), so that your configured name appears on outbound phone calls. There is no charge to use this feature.

You can set a default calling name for all the phone numbers in the Amazon Chime account once every 7 days using the AWS Management Console or AWS SDK. You can also set and update calling names for each phone number purchased or ported into Amazon Chime Voice Connector. The update can take up to 72 hours to propagate, during which time the previous setting is still active. You can track the status of the calling name updates in the AWS Management Console or the AWS SDK.

When you place a call using Amazon Chime Voice Connector, the call is routed through the public switched telephone network (PSTN) to a fixed or mobile telephone carrier of the called party. Note that not all landline and mobile telephone carriers support CNAM or use the same CNAM database as Amazon Chime Voice Connector, which can result in the called party either not seeing CNAM, or seeing a CNAM that is different from the value you set.

Call Routing with Load Sharing

Amazon Chime Voice Connector provides you with flexibility to configure how inbound calls from PSTN are routed to multiple offices, thus allowing you to improve the resiliency of your telephone network.

Inbound Calls

Inbound calls to your on-premises phone system are routed using user-defined priorities and weights to automatically route calls to multiple SIP hosts. Calls are routed in priority order first, with 1 being the highest priority. If hosts are equal in priority, calls are distributed among them based on their relative weight. This approach is useful for both load balancing and failure mitigation. If a particular host is unavailable, Amazon Chime Voice Connector automatically re-routes calls to the next SIP host based on priority and weight. This approach allows administrators to send all or a percentage of the calls to one site and to reroute the calls to another site in a disaster recovery scenario.

Outbound Calls

For outbound calls from your on-premises phone system, the hostname is a fully qualified domain name (FQDN) with dynamically assigned multiple IP addresses for load sharing.

Failover and Load Sharing

You can use Amazon Chime Voice Connector groups for fault-tolerant, cross-region routing for inbound calling to your on-premises phone system. By associating Amazon Chime Voice Connectors in different AWS Regions to an Amazon Chime Voice Connector group, you can create multiple independent routes for inbound calls to your on-premises phone system. In the event of loss of connectivity between an AWS Region and your phone system, or an Amazon Chime Voice Connector service unavailability in an AWS Region, incoming calls route to the next Amazon Chime Voice Connectors in an Amazon Chime Voice Connector group in priority order. For more information, see [Working with Amazon Chime Voice Connector Groups](#).

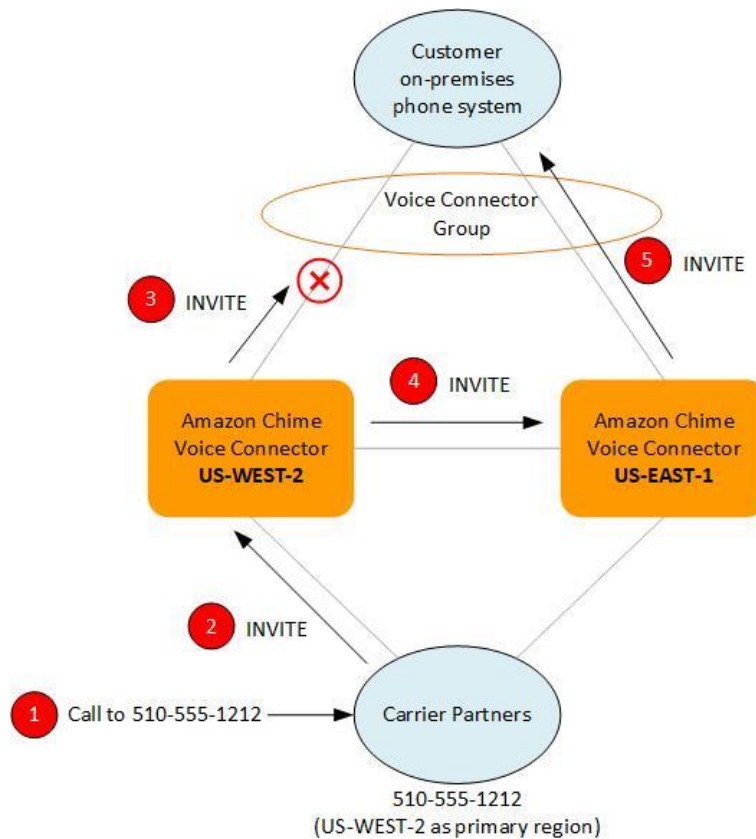


Figure 5: Voice Connector Groups, Failover

Fax

Amazon Chime Voice Connector supports faxing using SIP with either T.38 or G.711 μ -law. The SIP messaging when using T.38 should follow the format described in RFC 3362. In short, much of the SIP messaging stays the same as a voice call. One change is the “image/t38” MIME content type is added in the SDP to indicate a T.38 media stream will be present. Modern PBX and SBC systems will recognize T.38 and its messaging format.

Access

Access to the Amazon Chime Voice Connector can be provided through the internet or by using AWS Direct Connect.

Internet Access

You can connect to Amazon Chime Voice Connector using the internet. The bandwidth between Amazon Chime Voice Connector and your SIP infrastructure must be sufficient

to handle the number of simultaneous calls. For information about the bandwidth requirements, see [Network Configuration and Bandwidth Requirements](#).

AWS Direct Connect Access

You can connect using [AWS Direct Connect](#) public virtual interfaces which, in many cases, can reduce your network costs as it is more cost effective than Multiprotocol Label Switching (MPLS). AWS Direct Connect can also increase bandwidth throughput, and provide a more consistent network experience than internet-based connections. When you combine Amazon Chime Voice Connector with AWS Direct Connect, your voice call sessions use a single provider.

Real-time Audio Streaming to Amazon Kinesis Video Streams

Amazon Chime Voice Connector can stream audio from telephone calls to [Amazon Kinesis Video Streams](#) in real-time and gain insights from your business' conversations. Amazon Kinesis Video Streams is an AWS service that makes it easy to accept, durably store, and encrypt real-time media and connect it to other services for analytics, voice transcription, machine learning (ML), playback, and other processing. You can process audio streams with services like AWS Lambda, Amazon Transcribe, or Amazon Comprehend to build call recording, transcription, and analysis solutions.

For each audio call that is streamed to Kinesis Video Streams, two separate Kinesis streams are created for the caller and call recipient media streams. Each Kinesis stream within an audio call contains metadata, such as the TransactionId and the VoiceConnectorId, which can be used to easily filter the audio streams within the same phone call.

You can enable media streaming for all phone calls placed on the Amazon Chime Voice Connector using the Amazon Chime console, or you can enable real-time audio streaming on a per-call basis using SIPREC INVITE. For more information on streaming audio to Kinesis, see [Streaming Amazon Chime Voice Connector Media to Kinesis](#).

Audio Streaming using SIPREC

You can also send a SIPREC INVITE from your existing on-premises telephone system (Session Border Controller, or IP PBX) to Amazon Chime Voice Connector to initiate a real-time audio stream to [Amazon Kinesis Video Streams](#). You can use this feature to integrate your existing on-premises phone system with AWS services for analytics,

voice transcription, machine learning (ML), playback, and other real-time processing. After receiving the SIPREC INVITE from your on-premises phone system, Amazon Chime Voice Connector then sends the caller and call recipient media flows to your Amazon Kinesis Video Stream to connect the media streams to other AWS services for other processing. For more information on using SIPREC INVITE to stream media to Kinesis, see [Streaming Amazon Chime Voice Connector Media to Kinesis](#).

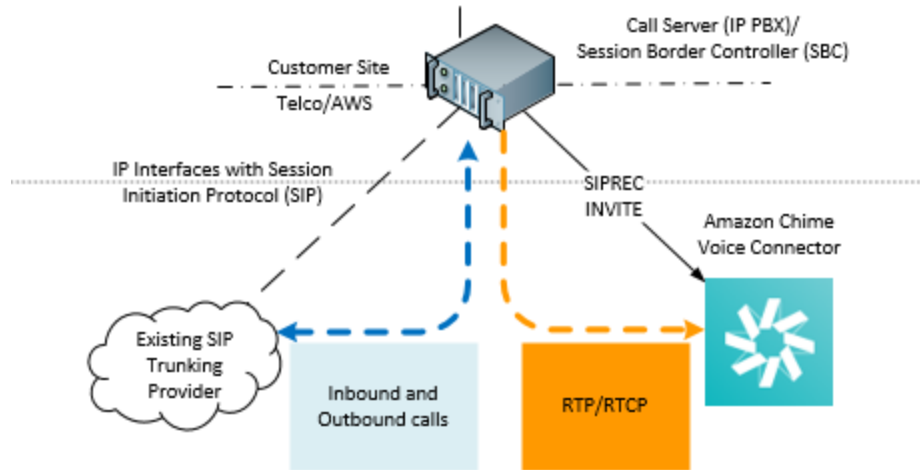


Figure 6: SIPREC Support

Monitoring Amazon Chime Voice Connectors

You can monitor Amazon Chime Voice Connector using Amazon CloudWatch, which collects raw data and processes it into readable, near real-time metrics. These metrics are kept for 15 months, so that you can access historical information and gain a better perspective on how your audio service is performing.

Amazon Chime Voice Connector sends metrics to Amazon CloudWatch Metrics that capture and process performance metrics across all Voice Connectors in your AWS Account. You can use Amazon CloudWatch Metrics to create dashboards and setup alarms to monitor the performance and availability of your calling solution. You can use Amazon CloudWatch Logs when configuring new Voice Connectors and troubleshooting issues. For more information, see [Monitoring Amazon Chime with Amazon CloudWatch](#).

CloudWatch Metrics

Amazon CloudWatch Metrics provides a near real-time stream of system events that describe metrics pertaining to the usage and performance of your Amazon Chime Voice Connectors. Using the Amazon CloudWatch Metrics, you can create dashboards, set up automated alarms, respond quickly to operational changes, and take corrective actions.

CloudWatch Logs

You can choose to send SIP Message Capture Logs from your Voice Connector to CloudWatch Logs. You can use **SIP Message Capture Logs** when setting up new Voice Connectors, or to troubleshoot issues with existing Voice Connectors. For more information, see [Monitoring Amazon Chime with Amazon CloudWatch](#).

Conclusion

Amazon Chime Voice Connector is simple to set up via the AWS Management Console or AWS SDK, and employees can place and receive calls on their desk phones in minutes. Calls are delivered to Amazon over an internet connection using industry-standard VoIP. With Amazon Chime Voice Connector, there are no upfront fees, commitments, or long-term contracts. You only pay for what you use.

Contributors

Contributors to this document include:

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Further Reading

For additional information, see:

- [Working with Amazon Chime Voice Connectors](#)
- [Amazon Chime - Pricing](#)
- [Amazon Chime Documentation](#)
- [RFC 3261](#)

Document Revisions

Date	Description
April 2020	Added fax support; updated dialed number requirements for outbound calls
November 2019	New features and content updates.
March 2019	First publication.

Appendix A: Call Detail Record (CDR) Specifications

Call Detail Record (CDR)

Storage Details

Call Detail Records (CDRs) are stored in your Amazon S3 bucket based on your bucket retention policy. CDR objects are stored using names in the following format:

`Amazon-Chime-Voice-Connector-CDRs/json/vconID/yyyy/mm/dd/HH.MM.SS.mmm-transactionID`

where:

- *vconID*– Amazon Chime Voice Connector ID
- *yyyy/mm/dd*– Year, month, and day that the call started
- *HH.MM.ss.mmm*– Start time of call
- *transactionID*– Amazon Chime Voice Connector transaction ID

For example:

`Amazon-Chime-Voice-Connector-CDRs/json/grdcp7r7fjejaautia8rvb/2019/03/01/17.10.00.020_123456789`

CDR Schema

CDR objects are stored with no whitespace or newline characters using the following format:

Value	Description
<code>{"AwsAccountId": "<i>AWS-account-ID</i>",</code>	AWS account ID
<code>"TransactionId": "<i>transaction-ID</i>",</code>	Amazon Chime Voice Connector transaction ID UUID
<code>"CallId": "<i>SIP-call-ID</i>",</code>	Customer facing SIP call ID
<code>"VoiceConnectorId": "<i>voice-connector-ID</i>",</code>	Amazon Chime Voice Connector ID UUID

Value	Description
"Status": " status ",	Status of the call
"StatusMessage": " status-message ",	Status message of the call
"SipAuthUser": " sip-auth-user ",	SIP authentication name
"BillableDurationSeconds": " billable-duration-in-seconds ",	Billable duration of the call in seconds
"BillableDurationMinutes": " billable-duration-in-minutes ",	Billable duration of the call in minutes
"SchemaVersion": " schema-version ",	The version of the CDR schema
"SourcePhoneNumber": " source-phone-number ",	E.164 origination phone number
"SourceCountry": " source-country ",	Country of origination phone number
"DestinationPhoneNumber": " destination-phone-number ",	E.164 destination phone number
"DestinationCountry": " destination-country ",	Country of destination phone number
"UsageType": " usage-type ",	Usage details of the line item in the Price List API
"ServiceCode": " service-code ",	The code of the service in the Price List API
"Direction": " direction ",	Direction of the call "Outbound" or "Inbound"
"StartTimeEpochSeconds": " start-time-epoch-seconds ",	Indicates the call start time in epoch/Unix timestamp format
"EndTimeEpochSeconds": " end-time-epoch-seconds ",	Indicates the call end time in epoch/Unix timestamp format
"Region": " AWS-region "}	AWS region for the Voice Connector
"Streaming": {" true false "}	Indicates whether the Streaming audio option was enables for this

Sample Call Detail Record (CDR):

```
{
  "AwsAccountId": "111122223333",
  "TransactionId": "879eee6e-eec7-4167-b634-a2519506d142",
  "CallId": "777a6b953100721d372188753f2059a8@203.0.113.9:8080",
  "VoiceConnectorId": "abcd11222223333334444",
  "Status": "Completed",
  "StatusMessage": "OK",
  "SipAuthUser": "5600",
  "BillableDurationSeconds": 6,
  "BillableDurationMinutes": 0.1,
  "SchemaVersion": "2.0",
  "SourcePhoneNumber": "+15105551212",
  "SourceCountry": "US",
  "DestinationPhoneNumber": "+16285551212",
  "DestinationCountry": "DE",
  "UsageType": "USE1-US-US-outbound-minutes",
  "ServiceCode": "AmazonChimeVoiceConnector",
  "Direction": "Outbound",
  "StartTimeEpochSeconds": 1565399625,
  "EndTimeEpochSeconds": 1565399629,
  "Region": "us-east-1",
  "Streaming": true
}
```

Streaming Detail Record (SDR)

Storage Details

Streaming Detail Record (SDR) objects are stored in your Amazon S3 bucket based on your bucket retention policy. SDR objects are stored using names in the following format:

`Amazon-Chime-Voice-Connector-SDRs/json/vconID/yyyy/mm/dd/HH.MM.SS.mmm-transactionID`

where:

- **vconID**– Amazon Chime Voice Connector ID
- **yyyy/mm/dd**– Year, month, and day that the call started
- **HH.MM.ss.mmm**– Start time of call

- **transactionID**– Amazon Chime Voice Connector transaction ID

Streaming Detail Records (SDRs) always correspond to a call detail record matching the object prefix, for example “**vconID/yyyy/mm/dd/HH.MM.SS.mmm-transactionID**”.

For example:

Amazon-Chime-Voice-Connector-SDRs/json/grdcp7r7fjejaautia8rvb/2019/03/01/17.10.00.020_123456789

SDR Schema

Value	Description
{ "SchemaVersion": " schema-version ",	The version of the CDR schema
"TransactionId": " transaction-id ",	Amazon Chime Voice Connector transaction ID UUID
"CallId": " SIP-call-id ",	Customer facing SIP call ID
"AwsAccountId": " AWS-account-ID ",	AWS account ID
"VoiceConnectorId": " voice-connector-id ",	Amazon Chime Voice Connector ID UUID
"StartTimeEpochSeconds": " start-time-epoch-second ",	Indicates the call start time in epoch/Unix timestamp format
"EndTimeEpochSeconds": " end-time-epoch-second ",	Indicates the call end time in epoch/Unix timestamp format
"Status": " status ",	Status of the call option (Completed, Failed, etc.)
"StatusMessage": " status-message ",	Details of the call option status
"ServiceCode": " service-code ",	The code of the service in the Price List API
"UsageType": " usage-type ",	Usage details of the line item in the Price List API
"BillableDurationSeconds": " billable-duration-seconds "	Billable duration of the call in seconds
"Region": " AWS-region " }	AWS region for the Voice Connector

Sample Streaming Detail Record (SDR)

```
{
  "SchemaVersion": "1.0",
  "AwsAccountId": "111122223333",
  "TransactionId": "879eee6e-eec7-4167-b634-a2519506d142",
  "CallId": "777a6b953100721d372188753f2059a8@203.0.113.9:8080",
  "VoiceConnectorId": "abcd112222223333334444",
  "StartTimeEpochSeconds": 1565399625,
  "EndTimeEpochSeconds": 1565399629,
  "Status": "Completed",
  "StatusMessage": "Streaming succeeded",
  "ServiceCode": "AmazonChime",
  "UsageType": "USE1-VC-kinesis-audio-streaming",
  "BillableDurationSeconds": 6,
  "Region": "us-east-1"
}
```

Appendix B: SIP Signaling Specifications

Ports and Protocols

Amazon Chime Voice Connector requires the following ports and protocols.

Signaling

AWS Region	Destination	Ports
US East (N. Virginia)	3.80.16.0/23	UDP/5060 TCP/5060 TCP/5061
US West (Oregon)	99.77.253.0/24	UDP/5060 TCP/5060 TCP/5061

Media

AWS Region	Destination	Ports
US East (N. Virginia)	3.80.16.0/23	UDP/5000:65000
US East (N. Virginia)	52.55.62.128/25	UDP/1024:65535
US East (N. Virginia)	52.55.63.0/25	UDP/1024:65535
US East (N. Virginia)	34.212.95.128/25	UDP/1024:65535
US East (N. Virginia)	34.223.21.0/25	UDP/1024:65535
US West (Oregon)	99.77.253.0/24	UDP/5000:65000

Supported SIP Methods

OPTIONS, INVITE, ACK, CANCEL, BYE

Unsupported SIP Methods

SUBSCRIBE, NOTIFY, PUBLISH, INFO, REFER, UPDATE, PRACK, MESSAGE

Required SIP Headers

In general, the service implements SIP as described in RFC 3261. The following SIP headers are required on all `OPTIONS`, `INVITE`, and `BYE` requests: `Call-ID`, `Contact`, `CSeq`, `From`, `Max-Forwards`, `To`, `Via`. `CANCEL` requests must also include these headers with the exception of `Contact`.

Further details about SIP headers can be found in [RFC 3261 § 20](#).

SIP OPTIONS Requirements

The Request-URI of the SIP OPTIONS requests that are sent to the service must identify the Voice Connector host name. For example:

```
OPTIONS sip:abcdefghijklmnop12345.voiceconnector.chime.aws SIP/2.0
```

SIPREC INVITE Requirements

The Request-URI must identify the Voice Connector host name. For example:

```
INVITE sip:+16285551212@abcd112222223333334444.g.voiceconnector.chime.aws:5060  
SIP/2.0
```

The user portion of the From: header must have a number in E.164 format. For example:

```
From: +16285551212 <sip:+16285551212@192.168.100.10>;tag=gK1005c68e
```

If you experience connectivity issues or dropped packets, the potential reason is that the UDP packets are dropped by the participating network elements, such as routers or receiving hosts on the internet because the UDP packets are larger than the maximum transmission unit (MTU). You can resolve this issue by either clearing the **Don't fragment (DF)** flag, or alternatively you can use TCP.

Dialed Number Requirements

- **Outbound calls:**

The dialed number must be valid and presented in E.164 format. Supported countries can be found under Calling Plan on the Termination Page in the Chime Console. Countries can be allowed or disallowed by the customer. If a call is placed from a customer PBX to a number that is not valid, the call will be rejected with a SIP 403 Forbidden response. The dialed number must be presented in E.164 format as the user portion of the Request-URI in the SIP INVITE, for example:

```
INVITE sip:+12125551212@abcdefghijklmnop12345.voiceconnector.chime.aws
```

The leading “+” is required.

- **Inbound calls:**

The called number is presented in E.164 format as the user portion of the Request-URI in the SIP INVITE. For example:

```
INVITE sip:+12065551212@abcdefghijklmnop12345.voiceconnector.chime.aws
```

Caller ID Number Requirements

- **Outbound Calls:**

The caller ID number is derived from the user portion of the `P-Asserted-Identity:` header, or the `From:` header, in that order. The caller ID must be a valid E.164 formatted phone number.

- **Inbound Calls:**

The caller ID number is presented in E.164 format as the user portion of the `P-Asserted-Identity:` and `From:` headers.

Caller ID Name

The delivery of Caller ID Name for inbound calls to your on-premises phone system is not supported. You can enable the delivery of Caller ID name for outbound calls from your on-premises phone system using the Outbound Calling Name (CNAM) feature.

Digest Authentication

Digest Authentication is an optional feature and it is implemented as described in RFC 3261, section 22.

Call Encryption

Enabling encryption in Amazon Chime Voice Connector to use TLS for SIP signaling and Secure RTP (SRTP) for media. Encryption is enabled using the **Secure Trunking** option in the console and the service uses port 5061.

When enabled, all inbound calls use TLS, and unencrypted outbound calls are blocked. You must import the Amazon Chime root certificate. Note that at this time the Amazon Chime Voice Connector service uses a wildcard certificate (`*.voiceconnector.chime.aws`). SRTP is implemented as described in RFC 4568.

For outbound calls, the service uses the SRTP default AWS counter cipher and HMAC-SHA1 message authentication. The following ciphers are supported for inbound and outbound calls:

- AES_CM_128_HMAC_SHA1_80
- AES_CM_128_HMAC_SHA1_32
- AES_CM_192_HMAC_SHA1_80
- AES_CM_192_HMAC_SHA1_32
- AES_CM_256_HMAC_SHA1_80
- AES_CM_256_HMAC_SHA1_32

At least one cipher is mandatory, but all can be included in preference order. There is no additional charge for voice encryption.

Session Description Protocol (SDP)

SDP is implemented as described in RFC 4566.

Supported Codecs

The service supports G.711 μ -law, and G.722 pass-through for Amazon Chime meeting dial-ins only.

DTMF

Dual-tone multifrequency (DTMF) is implemented as described in RFC 4733 (also known as RFC 2833 DTMF).

Appendix C: CloudWatch Metrics and Logs Examples

CloudWatch Metrics

Amazon Chime Voice Connector sends usage and performance metrics to Amazon CloudWatch. The namespace is `AWS/ChimeVoiceConnector`. To find a complete list of the CloudWatch Metrics sent by Amazon Chime Voice Connector, see [Monitoring Amazon Chime with Amazon CloudWatch](#).

CloudWatch Logs

SIP Capture Log Example

CloudWatch Logs log group name pattern

`/aws/ChimeVoiceConnectorSipMessages/[VoiceConnectorID]`

```
{
  "voice_connector_id": "abcdefg628ghsyzd8bwmh6",
  "event_timestamp": "2019-10-07T17:16:51Z",
  "call_id": "5bf5ecf1-27a1-4068-a7ee-6bd828a5f54a",
  "sip_message": "\nINVITE
sip:+15105551212@abcdefg628ghsyzd8bwmh6.g.voiceconnector.chime.aws:5061 SIP/2.0\nVia: SIP/2.0/TLS
192.168.100.10:8081;branch=z9hG4bK66a2d803;rport\nMax-Forwards:
69\nFrom: \"Testing Account\"
<sip:+16285551212@192.168.100.10:8081>;tag=as283a6f9b\nTo:
<sip:+15105551212@abcdefg628ghsyzd8bwmh6.g.voiceconnector.chime.aws:5061>\nContact:
<sip:+16285551212@192.168.100.10:8081;transport=TLS>\nCall-ID:
6347f9d4697c1539361a1d97727bd2c8@192.168.100.10:8081\nCSeq: 102
INVITE\nUser-Agent: Asterisk PBX 1.8.32.3\nDate: Mon, 07 Oct 2019
17:16:51 GMT\nAllow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
SUBSCRIBE, NOTIFY, INFO, PUBLISH, MESSAGE\nSupported: replaces,
timer\nContent-Type: application/sdp\nContent-Length:
322\n\nv=0\no=root 1248709283 1248709283 IN IP4
192.168.100.10\ns=Asterisk PBX 1.8.32.3\nnc=IN IP4
192.168.100.10\nnt=0 0\nnm=audio 15406 RTP/SAVP 0 101\na=rtpmap:0
PCMU/8000\na=rtpmap:101 telephone-event/8000\na=fmtp:101 0-
16\na=ptime:20\na=sendrecv\na=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:OkiaSoC0tQG15E7eG21+7DFprLZku9XkE8hl9Zlc\n"}

```