

# Amazon Connect SIP Connector

## Data Sheet



# Disclaimer

**Amazon Connect does not support SIP Protocol, therefore you can not open a ticket to AWS for SIP.**

**Comstice provides the full support for the SIP scenarios mentioned in this document. Comstice WebRTC Gateway must be used for the solutions to work.**

**Comstice WebRTC Gateway can be hosted by your organization as an EC2 or Comstice can host them for you as well.**

# What is Comstice WebRTC Gateway?

Comstice WebRTC Gateway is created to accommodate certain SIP to WebRTC scenarios. It helps WebRTC endpoints to communicate with the SIP Networks and vice versa, It supports standard WebRTC as well as custom WebRTC formats such as Amazon Connect's LilyRTC.

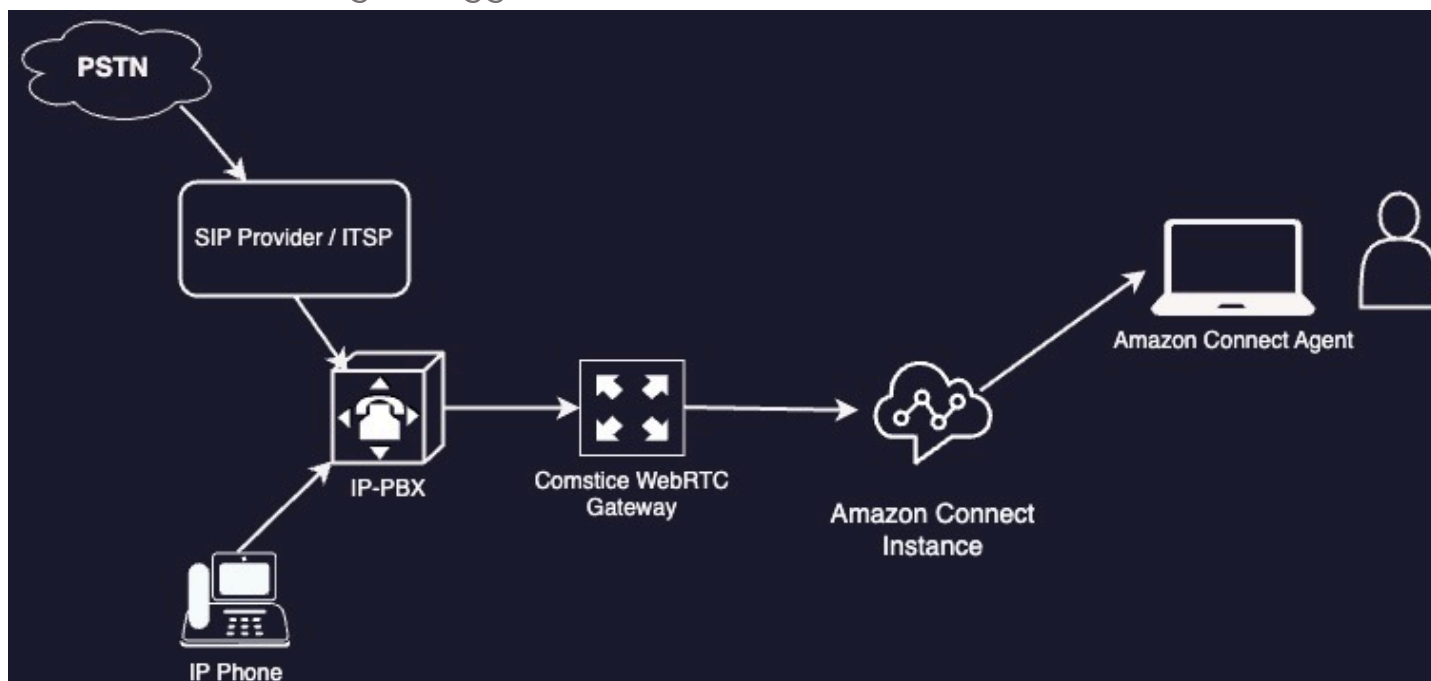
WebRTC Gateway includes multiple services running as Docker containers;

- SIP to WebRTC Converter (both ways)
- SIP Proxy
- SIP Session Border Controller
- Web-Based Comstice Dialplan Manager
- Web-Based Audio Quality Monitoring Module
- RTP Proxy Module (for certain NAT and PAT scenarios)
- Audio Recording Module. (You can still record sessions on Amazon Connect)

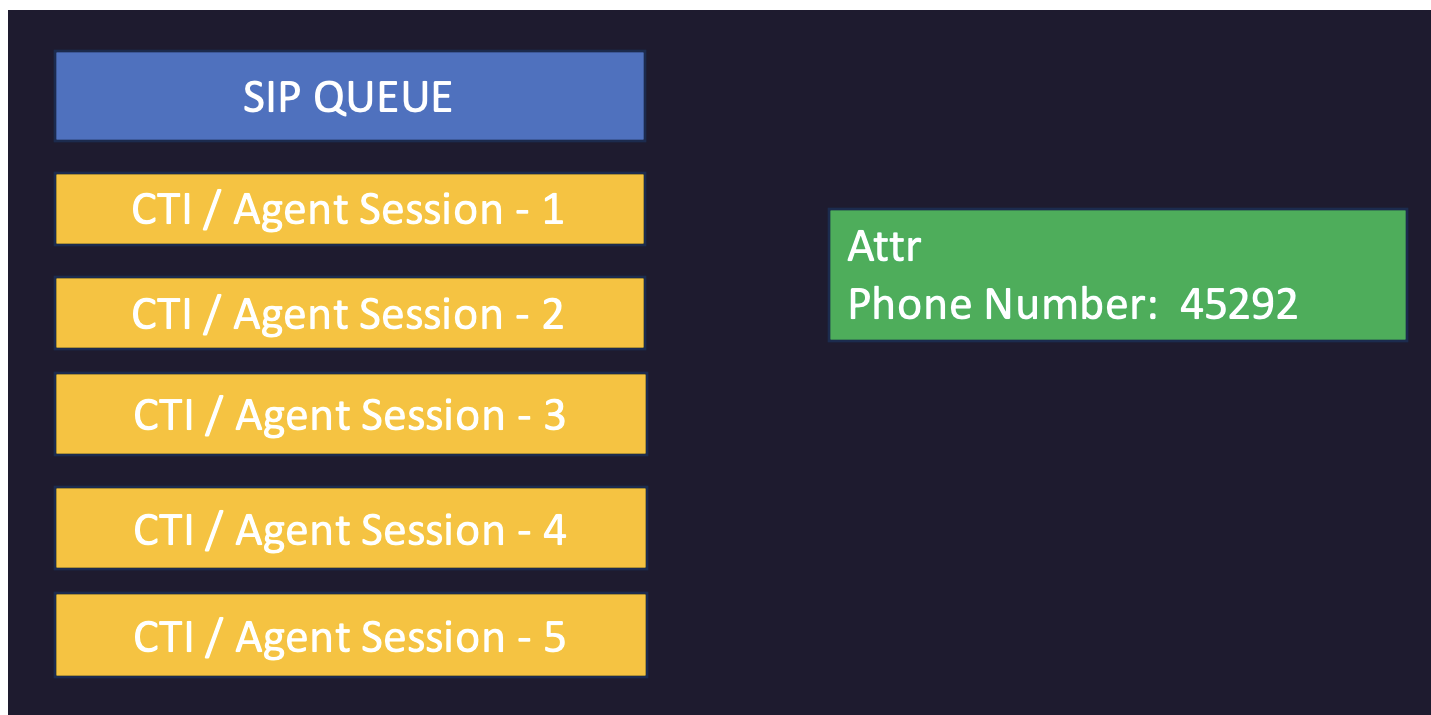
Comstice WebRTC Gateway runs on a LinuxOS. It can be run on your EC2 service or your VMWare ESXi solution. It can also be hosted by Comstice as well.

# How It Works

Comstice WebRTC Gateway uses Amazon Connect APIs to send SIP calls to Amazon Connect. WebRTC Gateway converts SIP calls to Amazon's WebRTC protocol and Communication Gadgets trigger Amazon Connect Contact Flows.



Calls from Amazon Connect to SIP are sent through some agent sessions. Comstice WebRTC Gateway converts certain agent sessions to SIP and extends the call to SIP Network by updating the SIP Header.



# Amazon Connect Pricing

For the calls coming from PSTN and terminated in Amazon Connect, there is no additional cost to extend the call to the SIP Networks.

There is additional cost when a SIP call is sent to Amazon Connect and an Amazon Connect Contact Flow is triggered. It is named as in-app, web calling and currently priced as 0.010 USD /min.

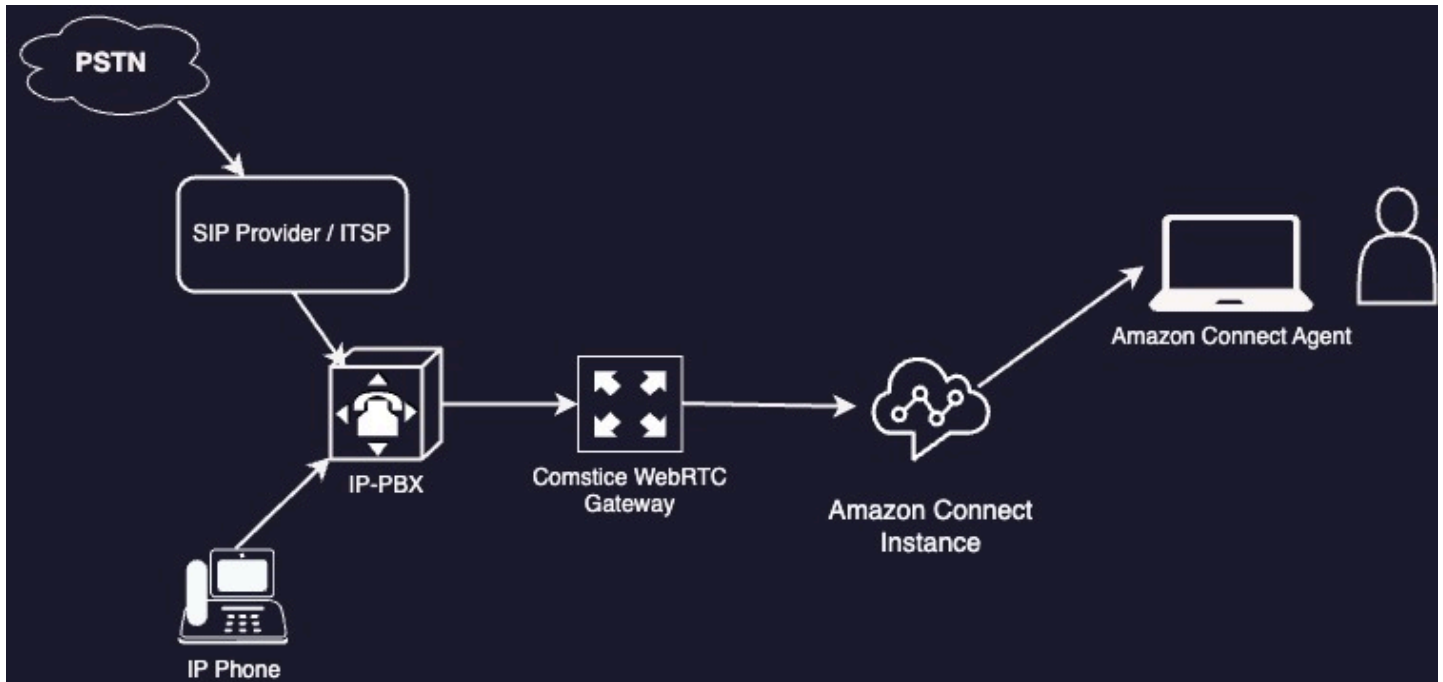
# SIP REFER Support

When there is a call transfer from SIP to Amazon Connect then back to SIP, using the SIP REFER signaling, the second leg of the call is terminated to avoid and call tromboning between SIP and Amazon Connect networks.

Comstice WebRTC Gateway supports SIP REFER so that, if the SIP call is transferred back from Amazon Connect to SIP, Amazon Connect section of the call will be terminated to avoid additional cost and idling the WebRTC Gateway capacity.

# Integrating an IP PBX to Amazon Connect

Comstice SIP Connector can help to integrate an IP PBX into Amazon Connect so that you can send calls from your IP PBX and existing inbound phone numbers into Amazon Connect contact flows. Your enterprise users can remain on the existing IP PBX and your contact center run on Amazon Connect .



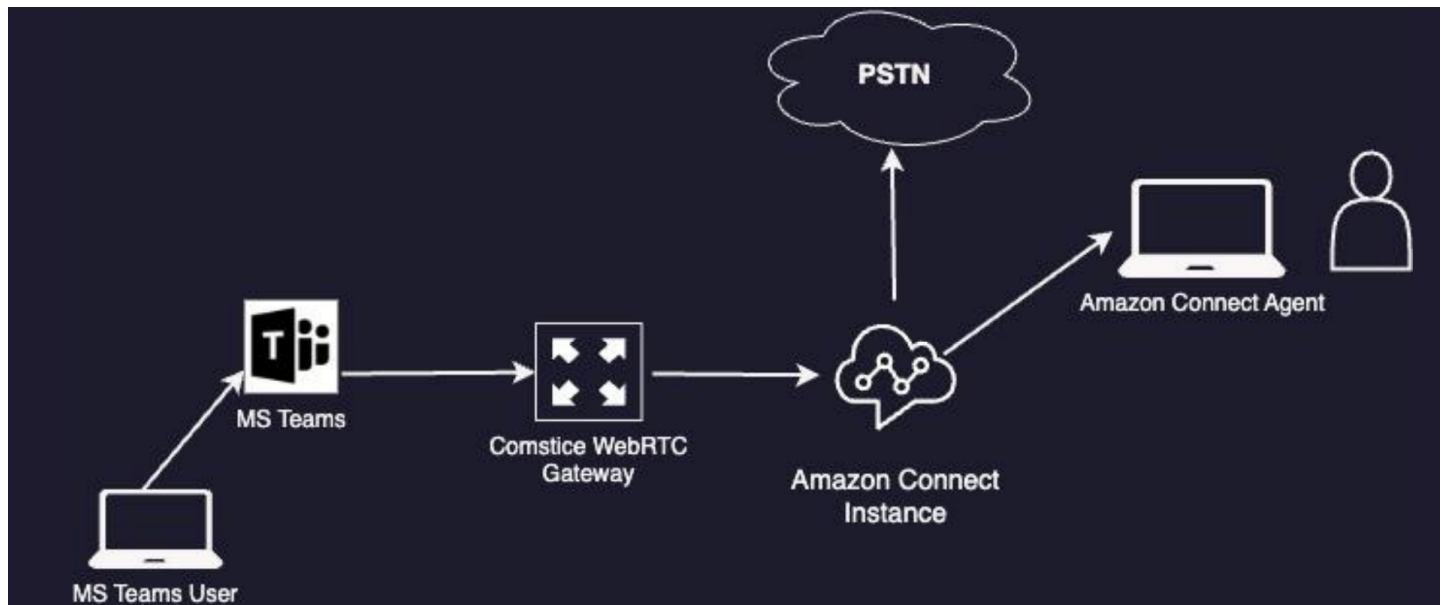
You can connect your IP PBX to Comstice WebRTC Gateway with a SIP Trunk. WebRTC Gateway also terminates agent sessions on the server level. These sessions act like CTI ports for the calls sent from Amazon Connect to SIP networks.

# Microsoft Teams Voice Integration

MS Teams users make and receive calls from Amazon Connect and can also use Amazon Connect as their PSTN Breakout. Comstice WebRTC Gateway includes a Session Border Controller Module and integrates with MS Teams for the audio and the video.

Amazon Connect agents can transfer calls to transfer queue, type the extension or say the name of the MS Teams user, then complete the transfer.

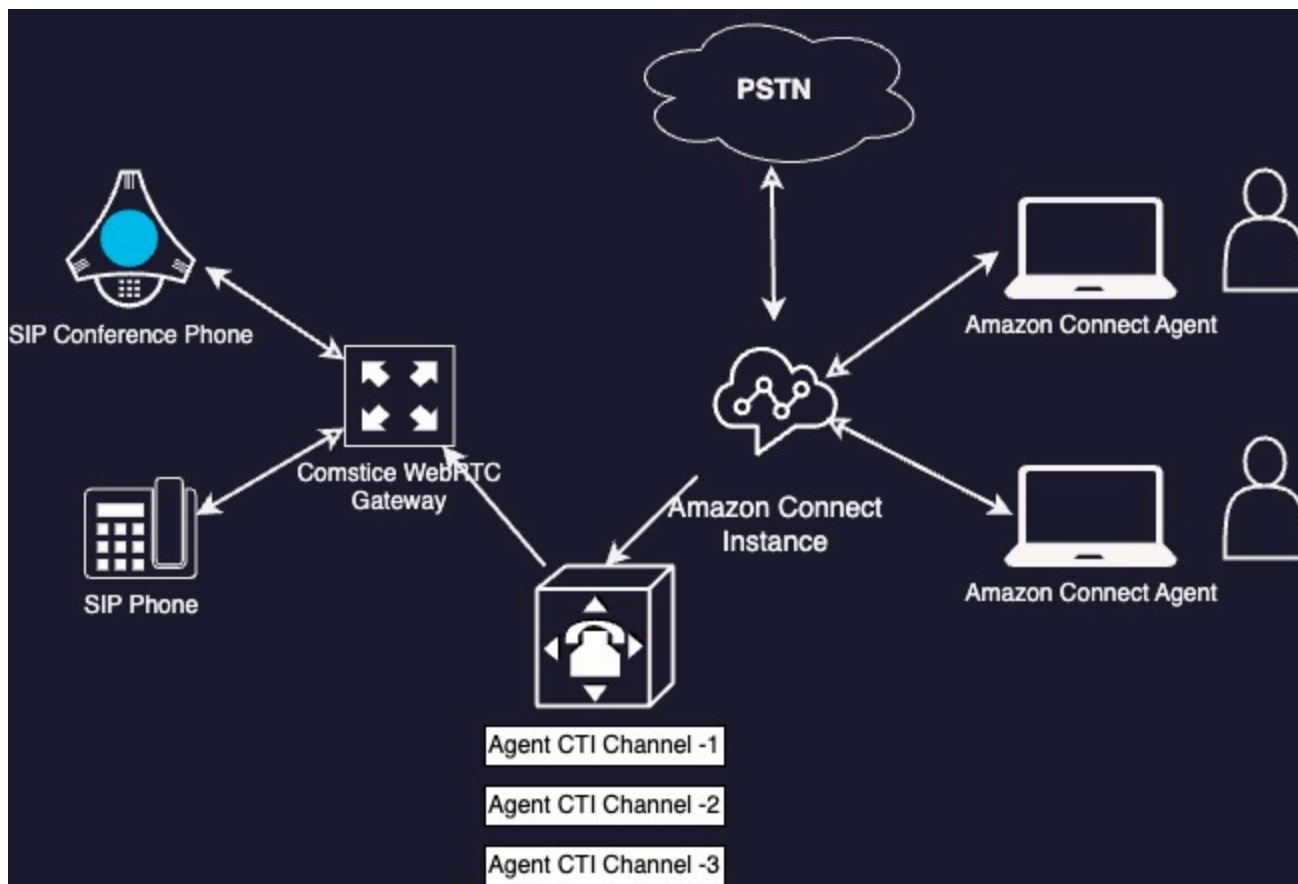
MS Teams users can type the extension of the Amazon Connect queues or agents and Comstice WebRTC Gateway can transfer the call to Amazon Connect destinations.





# Integrating SIP Phones with Amazon Connect

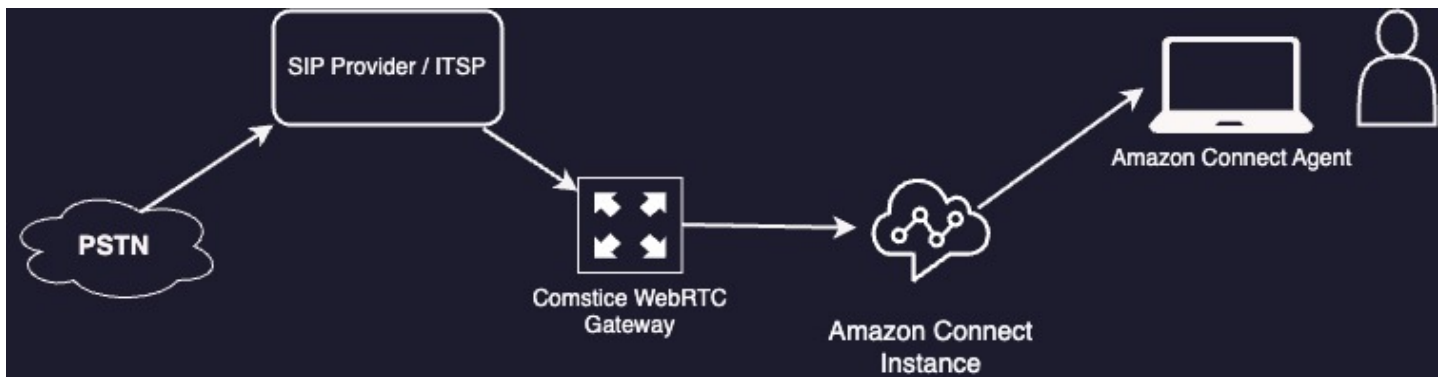
Comstice WebRTC Gateway includes a SIP Proxy where you can register your SIP Phones. These can be standard phones, conference phones and SIP softphones. Using Comstice DialPlan Manager, you can create a dial plan, assign the extensions to SIP phones, Amazon Connect agents and queues.



Amazon Connect agents can not use SIP Phones to work as a call center agent.

# Retain Old PSTN Numbers

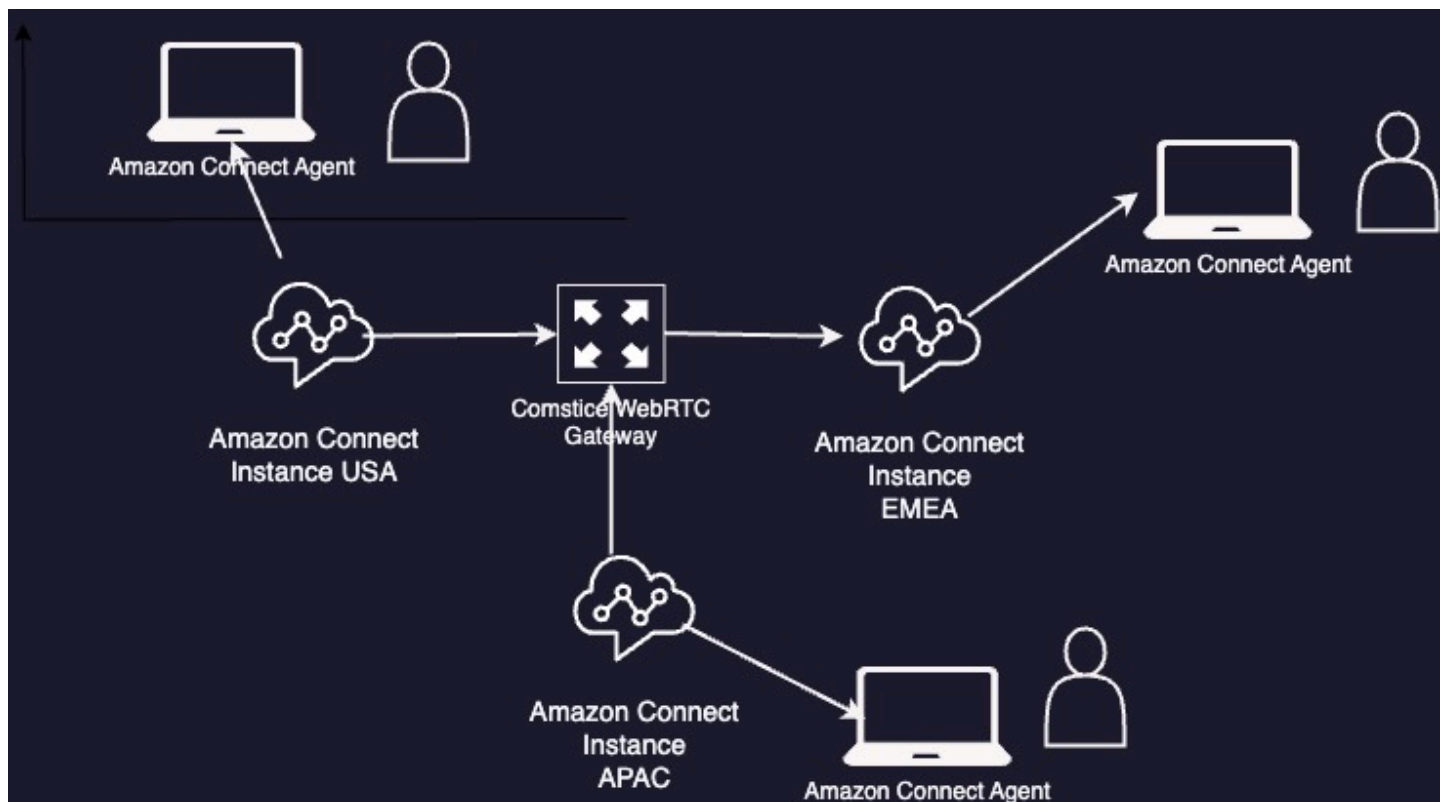
You can keep old PSTN numbers and integrate them with Amazon Connect. Comstice SIP Connector includes a Session Border Controller and it can terminate your SIP trunk accordingly. This helps to retire your old PBX system but still retain your existing public phone numbers without migrating them to Amazon Connect.



# Amazon Connect Federation

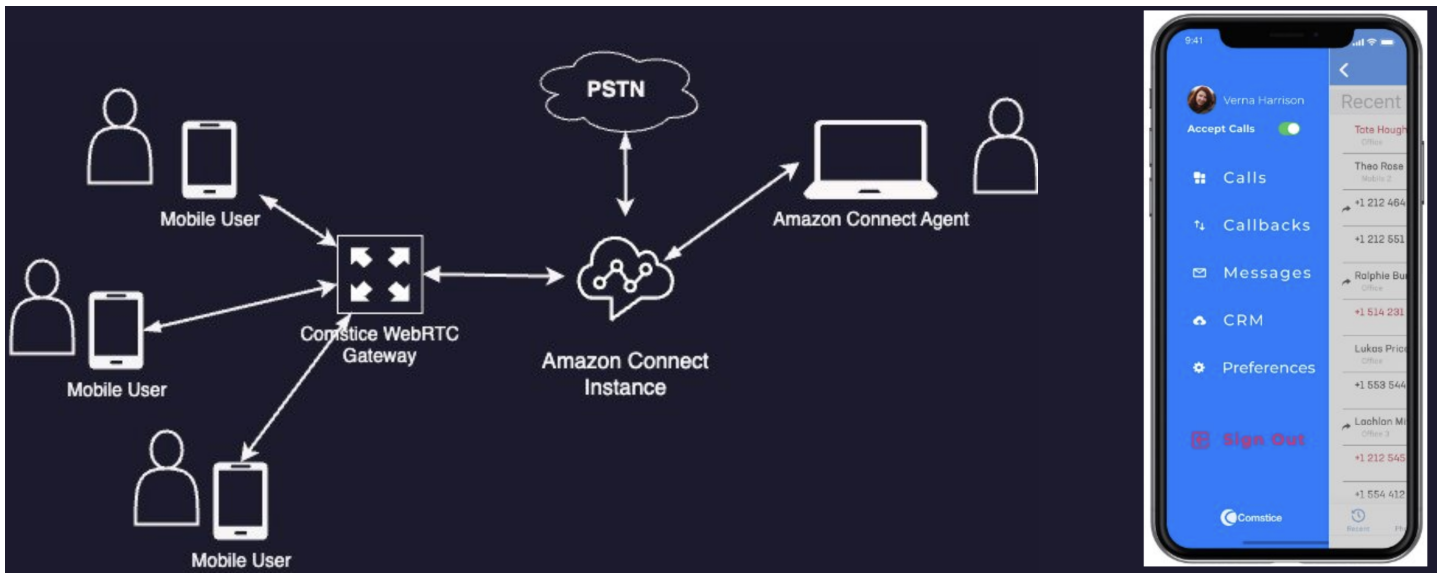
If you have multiple Amazon Connect instances, you can create a federation among them so that the calls can be sent from one instance to the other as VoIP. Users can also call each other directly from one instance to another instance as VoIP.

Using Comstice Dialplan Manager, you can assign extensions to each agent and contact flows. Users can call by name or extension.



# Comstice Mobile Softphone with Amazon Connect

Comstice SIP Connector also integrates with Comstice Mobile Connect app so that standard users can login, make and receive calls from Amazon Connect using their mobile devices and tablets. This helps remote users to become part of the customer journey, handle calls and make calls through Amazon Connect.



Download **Comstice Mobile Connect App** from **Apple AppStore** and **Google Play Store**. today or scan the barcode below



# Agent to Agent Direct Calls

One of the missing features in Amazon Connect is that, agents can not call each other directly as VoIP. Using Comstice Webphone gadget on Amazon Connect Workspaces, you can make and receive calls as an agent to other agents.

The screenshot displays the Amazon Connect interface with a dark header bar. On the left, a sidebar shows a call log with three entries: Sofia Martinez (06:43), Follow up with Nikki Wolf (06:43), and Follow up with Richard (08:12). Below the log, a blue bar indicates a connected call to +19145550199 at 00:10. The main area is divided into two tabs: 'Customer profile' and 'Cases'. The 'Customer profile' tab shows details for Ana Carolina Silva, including her full name, queue (New reservations), case ID (1234567), and IVR response (Make a new reservation). Below this, a section titled 'Ana may be contacting about...' lists five potential reasons for the call: 'Make new reservation', 'Car rental - New York' (upcoming Sep 17, 2022), 'Trip to Mexico' (upcoming Aug 15, 2022), 'Flight to France' (upcoming Dec 5, 2022), and 'Refund flight to Atlanta' (refunded July 10, 2022). A button labeled 'It's something else' is at the bottom of this list. The left sidebar also includes a 'Voice ID: Authenticated' status, a 'Fraud Risk: Low risk' indicator, and buttons for 'Hold', 'Mute', 'Number pad', 'Quick connects', 'Create a task', and 'End call'.

# Support

## What does it cover?

Comstice provides **break-fix support** for the Comstice SIP Connector software.

**Priority Support** is provided on up to 1-hr response and up to 24 hours fix basis due to all the dependencies around the solution. This is not guaranteed, however we aim to resolve 90% of all support queries within this time frame. **Priority support is available 24/7 and requires High Availability servers and secondary WebRTC Gateway server.**

Support service provided by qualified engineers via email or WebEx. Troubleshooting steps are performed via Webex.

## How to raise a support ticket?

End customer IT personnel or partners can raise a support tickets online from [www.comstice.com/support](http://www.comstice.com/support) or by calling Comstice Support Desks from +1-713-929-3714 or +44 203 051 7796. Alternatively, you can send an email to [support@comstice.com](mailto:support@comstice.com) or use webchat feature at [comstice.com](http://comstice.com)

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