Wave iX Real Time Translation Configuration Guide

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Contents

Purpose of the Document

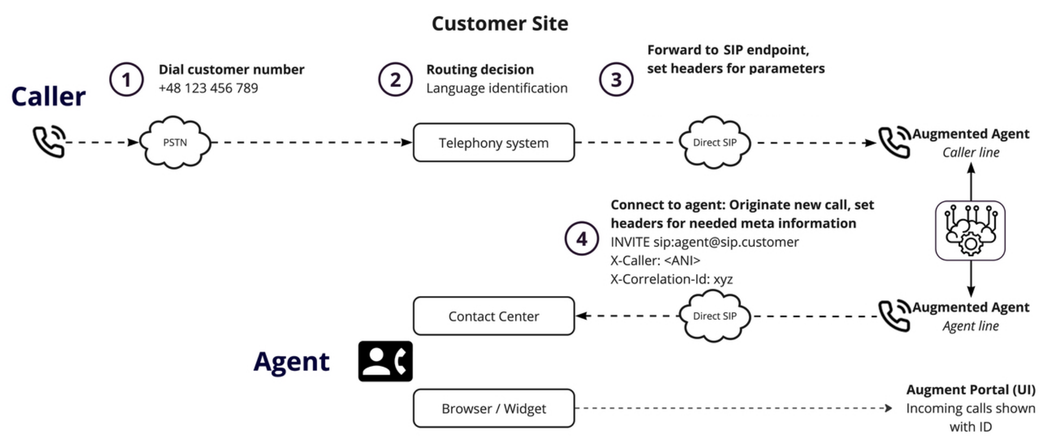
This document provides a high-level overview of Wave iX Real Time translation solution and steps involved in the telephony configurations.

1. High Level Call Flow:

Understanding the RTT (Real-Time Translation) call flow is crucial before delving into the details of the telephony integration. RTT bridges two separate calls to facilitate voice translation between customers and Customer Service Agents (CSAs). The two calls involved are:

1. **Incoming Call –** The original customer call, which is forwarded to Wave iX along with the customer's audio.
2. **Outbound Call –** A call generated by Wave iX, routed to the CCaaS (Contact Center as a Service) system. This setup enables the recognition and translation of the Agent's speech into the caller's language.

Below is a step-by-step description of the process to ensure a smooth end-to-end experience.



2. Call Flow Steps

2.1 Step 1: Customer Facing Number – Incoming call

- The caller dials the customer service number (for example, +1 123 456 789).

2.2 Step 2: Routing Decision

- The call reaches the telephony system at the customer site.
- The telephony system makes a routing decision based on language identification.

2.3 Step 3: Forwarding to Wave iX

- The call is forwarded to the Wave iX SIP endpoint with the necessary headers.
  - SIP INVITE example: INVITE sip:customer@sip.WaveiX.com
  - Header's example: X-WaveiX-Language: xx-XX

2.4 Step 4: Origination of Outbound call

- Wave iX receives the incoming call and originates a new outbound call to the CCaaS system.
- The outbound call contains the meta-information received in the headers of the incoming call.
    - SIP INVITE example: INVITE sip:agent@sip.customer
    - Header's example: X-Caller: <ANI>, X-Correlation-Id: xyz

2.5 Step 5: Handling by Augment Agent

- The outbound call is handled by the augmented agent at the contact center.
- The augmented agent line connects back to the original caller line.
- The augment portal (UI) displays incoming calls with ID information.

3. Configuration Instructions - SIP/PSTN Integration

3.1 What is SIP?

SIP, or Session Initiation Protocol, is a modern protocol for initiating, managing, and terminating voice and video sessions over the Internet. It offers flexibility, scalability, and the capability to transmit multimedia content. Wave iX uses SIP to efficiently connect with customer telephony infrastructures, providing enhanced communication capabilities.

3.2 What is PSTN?

The Public Switched Telephone Network (PSTN) is the traditional network for voice communication, consisting of telephone lines, cellular networks, and international cables. Although reliable, PSTN lacks the flexibility and multimedia support that SIP technology provides.

3.3 SIP vs. PSTN: The Key Differences

- **SIP** is preferred for its flexibility, multimedia support, and integration capabilities, making it ideal for modern businesses.
- **PSTN** is reliable for voice communications but is less scalable and lacks the multimedia capabilities of SIP.

Wave iX favors SIP for its efficiency, cost-effectiveness, and advanced features like direct call routing and data linkage capabilities.

3.4 Integrating with Your Phone System

Internet using SIP Connection – This is the preferred method for its direct, encrypted communication, reducing both costs and complexity.
Public Switched Telephony Network (PSTN) – Used when SIP is not feasible, involving third-party services for call routing.

3.5 SIP Integration Methods

The following describes the SIP methods commonly used with Wave iX: SIP INVITE and SIP REFER.

3.5.1 INVITE Method

- **Purpose –** To initiate or modify SIP sessions.
- **Functionality –** Starts a voice interaction or adds participants to a call.

- **Call Flow**
    1. Your PBX initiates the process by sending an INVITE to Wave iX.
    2. Wave iX responds with provisional responses, leading to a final response (such as 200 OK) to accept the call.
    3. The PBX sends an ACK request to Wave iX, acknowledging the final response, at which point the bot begins speaking.

3.5.2 REFER Method

- **Purpose –** Used for transferring an ongoing call to another endpoint.
- **Functionality –** Instructs the recipient to establish a call with a third-party endpoint.
- **Call Flow –**

1. Wave iX sends a REFER request to the current call party.
2. The party acknowledges and initiates an INVITE to the new target.
3. If accepted, the call is successfully redirected.

3.6 SIP Integration Requirements

Before initiating SIP integration, ensure your system:

- Has SIP Trunk Capability for establishing a SIP trunk via the Internet.
- Supports SIP Refers or provides an API for call control.
- Can manipulate SIP headers for accurate routing.

3.7 SIP Integration Steps

3.7.1 Step 1 – Establish SIP Trunk

- **Register with Wave iX**
    - Wave iX will provide you with a specific URI (Uniform Resource Identifier), such as rtt.voip.WaveiX.com, to connect your system with Wave ix's SIP server.
- **Configure DNS for FDQN resolution**
    - Set up your telephony system to resolve the FQDN (Fully Qualified Domain Name) provided by Wave iX. This step is essential for your system to locate Wave iX's SIP server over the Internet. In this case, rtt.voip.WaveiX.com is the FQDN.
- **Configure Port Number**
    - For unencrypted communication, use UDP port 5060.
    - For secure, encrypted communication, it's recommended to use TLS port 5061 to ensure the security of your SIP communications, especially when transmitting data over the public internet.
- **Whitelist our IPs**
    - You must whitelist our IPs on your network and open the necessary ports.
        - **HTTPS Traffic**
            - 20.119.83.207 (HTTP traffic)
            - 52.151.248.169 (HTTP traffic)
            - 20.62.165.147 (HTTP traffic)
        - **VoIP Traffic**
            - 74.235.108.116 (VoIP traffic)
            - 172.174.123.244 (VoIP traffic)
            - 20.115.81.243 (VoIP traffic)

3.7.2 Step 2 – Forward Calls to Wave iX

- **Identify the Call routing Feature**
  - Locate the call forwarding settings in your telephony system or PBX (Private Branch Exchange).
- **Setup Forwarding Rules**
  - Specify conditions under which calls should be forwarded to Wave iX, such as all incoming calls or calls during non-business hours.
- **Test Call Forwarding**
  - Perform a test call to verify that calls are being correctly routed to Wave iX.

3.7.3 Step 3 – Retrieve Calls from Wave iX

- **Configure your system for REFER and INVITE requests**
  - Enable it to accept and process SIP REFER/INIVTE requests. This allows for effective call redirection as per the above workflow mentioned in section 2 of this document.
- **Prepare for SIP REFER/INVITE Method**
  - Configure your system to receive and recognize SIP URIs from Wave iX. This step is essential for call redirection and sets the foundation for handling incoming requests.
- **Test Call Reception and Routing**
  - Work with Wave iX to perform test calls that include transferring calls back to your system using the REFER and INVITE methods.
  - During testing, ensure that calls are being correctly routed to the intended destination within your system.
- **Optional: Integrate Custom SIP Headers**
  - If needed, configure your system to recognize and use custom SIP headers, which Wave iX can add for advanced routing decisions and data exchange.

3.8 PSTN Integration

Wave iX offers PSTN integration by providing a dedicated phone number for call forwarding. Configure your PBX (Private Branch Exchange) or telephony system to forward and receive calls. Ensure to provide a return number for callbacks.

For further information or assistance, please contact your Wave iX representative.

4. Screen Pop up

The Wave iX CCaaS Screen Pop feature enhances workflow efficiency for agents by automatically opening the corresponding conversation in a new tab when an agent accepts a call in CCaaS.

4.1 Requirements

To use the CCaaS Screen Pop feature, ensure the following requirements are met:

- End-to-end SIP calls (both inbound and outbound)
- The User-to-User header must be set in the **Inbound** call
  - Wave iX will handle adding the header to the **Outbound** call
  - Example format: User-to-User: 0139393939393939;encoding=hex

4.2 Setup Instructions

Step 1: Configure CCaaS to open a URL in a new web browser tab when a call is accepted.

Step 2: Set the URL provided by Wave iX team. E.g.

https://augment.example.com/api/v1/conversations/{UUI}/redirect

Step 3: Replace {UUI} with the value of the User-to-User header.

5. Additional Considerations

5.1 Encryption and Security

Wave iX offers the option to uphold the highest security standards by enabling encryption of all SIP data using TLS (Transport Layer Security) and securing voice streams with sRTP (Secure Real-Time Transport Protocol) upon request.

To utilize these security features and ensure compliance with applicable data protection laws, please consult with your Wave iX representative or reach out to our support team for the necessary certificates.

5.2 Security and Privacy Standards

Wave iX adheres to internationally recognized security and privacy standards, ensuring that our SIP solutions meet the stringent requirements of various regulatory bodies.

5.3 Backend Integration

Wave iX offers a **generic HTTPS web service interface** for seamless integration with external data sources and APIs.

6. Revision History

Version	Date	Change By
V1.0 Draft 1	02/01/2025	Zeeshan Ahmed

Note: Where significant changes are made to this document, the version number will be incremented by 1.0. Where changes are made for clarity and reading ease only and no change is made to the meaning or intention of this document, the version number will be increased by 0.1.

7. Next Scheduled Review:

Date of Next Scheduled Review
1-Jul-2025