LiveVoice

Dante RTP streaming to LiveVoice

If you want to use hardware to stream to LiveVoice via Dante, here is how it works.

Advanced | Live Translation | Operator | Admin

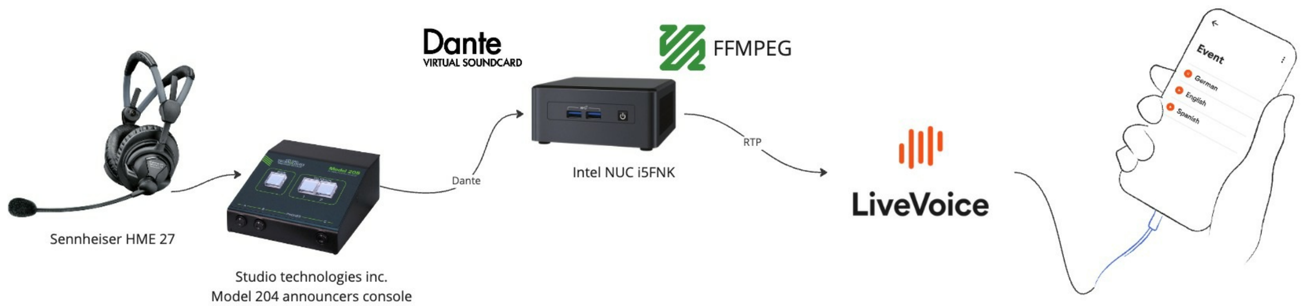
Overview

This article explains how to stream an interpreters audio feed via Dante Virtual Soundcard (DVS) to LiveVoice. The goal is to make the translation accessible to listeners through the LiveVoice app while having a permanent hardware setup ready that is started the same way every time very quickly.

Thanks to ICF Zürich that provided this practical guide to share how they have done their setup!

Hardware

- **Headset**: Sennheiser HME 27
- **Console**: Studio Technologies Inc. Model 204 Announcers Console
- **Computer**: Intel NUC running Windows
- **Dante Virtual Soundcard (DVS)**: To receive audio
- **FFmpeg**: To stream the audio feed via RTP
- **LiveVoice**: Online platform for audio streaming



Connect Hardware

- Connect the headset to the announcers console.
- Link the console to the Intel NUC or macOS device via the Dante network.**Install Software**:
- Install **Dante Virtual Soundcard (DVS)** and configure the appropriate channels (e.g., Receive 1-2 for English).
- Download **FFmpeg** from the official website and extract it to your device.**Set Up the LiveVoice Channel**:
- Log in to LiveVoice and enable the “Import Audio” option on the desired channel.
- Select **RTP** as the protocol and **OPUS** as the audio format.
- Copy the stream URL and port provided by LiveVoice into the FFmpeg script.

Streaming Script for Windows

For **Windows**, use the following commands:

- **List Available Audio Devices**:

```
ffmpeg -list_devices true -f dshow -i dummy
```

- **Streaming Script**: Save the following script as a .bat file, e.g., EnglishStream.bat :

```
@echo off
TITLE Livevoice English Stream from DVS Receive 1-2
ffmpeg -f dshow -i audio="DVS Receive  1-2 (Dante Virtual Soundcard)" -acodec libopus -bufsize 500k -ab 192k -ac 2 -f rtp rtp://<LiveVoice-URL>:<Port>
exit
```

Replace and with the values provided by LiveVoice.

Streaming Script for Mac

For **Mac**, use the following commands:

- **List Available Audio Devices**:

```
./ffmpeg -f avfoundation -list_devices true -i ""
```

- **Streaming Script**: Save the following as a shell script file (e.g., EnglishStream.sh):

```
#!/bin/bash
ffmpeg -f avfoundation -i ":0" -acodec libopus -bufsize 500k -ab 192k -ac 2 -f rtp rtp://<LiveVoice-URL>:<Port>
```

- Replace and with the values provided by LiveVoice. Make the script executable by running:

```
chmod +x EnglishStream.sh
```

Configure Autostart

- **Windows**: Place the .bat file in the Windows startup folder:

```
C:\Users\<Username>\AppData\Roaming\Microsoft\Windows\Start Menu\Programs\Startup
```

- **Mac**: Add the shell script to a login item using macOS System Preferences or a third-party automation tool.

Perform a Soundcheck

Before the event, test the setup by streaming an MP3 file:

- **Windows**:

```
ffmpeg -re -i test.mp3 -acodec libopus -bufsize 500k -ab 192k -ac 2 -f rtp rtp://<LiveVoice-URL>:<Port>
```

- **Mac**:

```
./ffmpeg -re -i test.mp3 -acodec libopus -bufsize 500k -ab 192k -ac 2 -f rtp rtp://<LiveVoice-URL>:<Port>
```

Verify the connection and audio quality in the LiveVoice app.

Troubleshooting

- **No Audio Input Detected**: Ensure the correct channel (e.g., “DVS Receive 1-2”) is selected in FFmpeg.
- **RTP Stream Not Reachable**: Check network connectivity and firewall settings.
- **Low Audio Quality**: Adjust the bitrate (-ab) and buffer size (-bufsize) in the FFmpeg script.