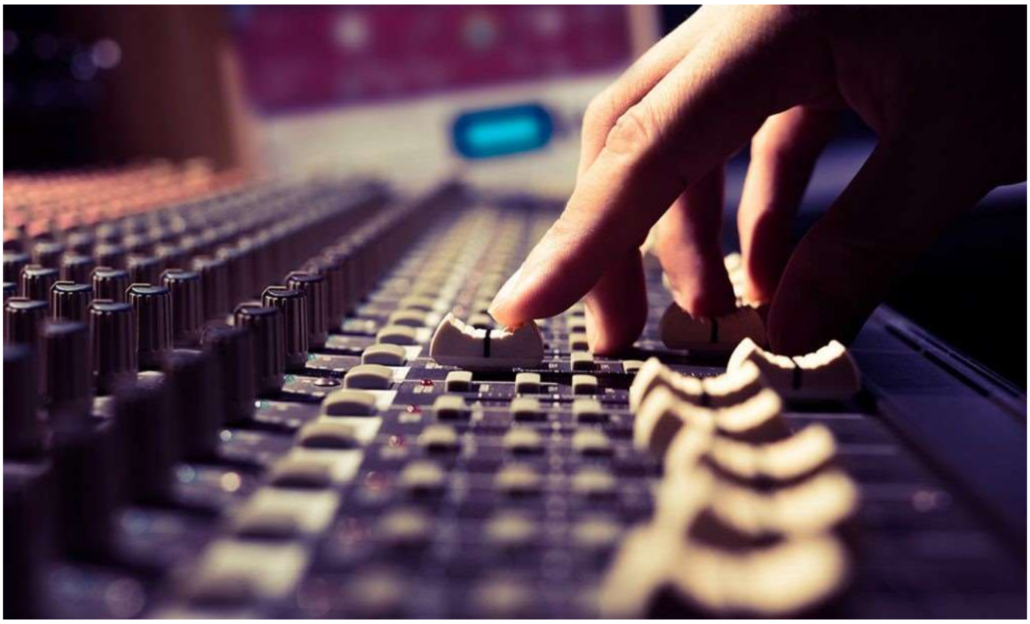
Case Study III - Dante to SIP Gateway



Dante audio networks can replace all of the classic analog and digital audio connections with a computer network, effortlessly sending hundreds of channels of audio over Ethernet cables with perfect digital fidelity. All connections can now be managed with software, making routing fast, readable and reliable. Because all devices share the same network, signals can be sent between all devices no matter where they are located.

Nonetheless, many studios face challenges when implementing this new approach into their networks. Dante is not always readily compatible with more specialized input and output methods, which often requires extra devices to act as converters or additional control elements. One of the more prominent examples of this challenge is handling SIP inputs in a Dante network.

Ferncast's aixtream helps the user avoid this issue entirely by offering software and hardware solutions that can serve any input and any output, including SIP to Dante.

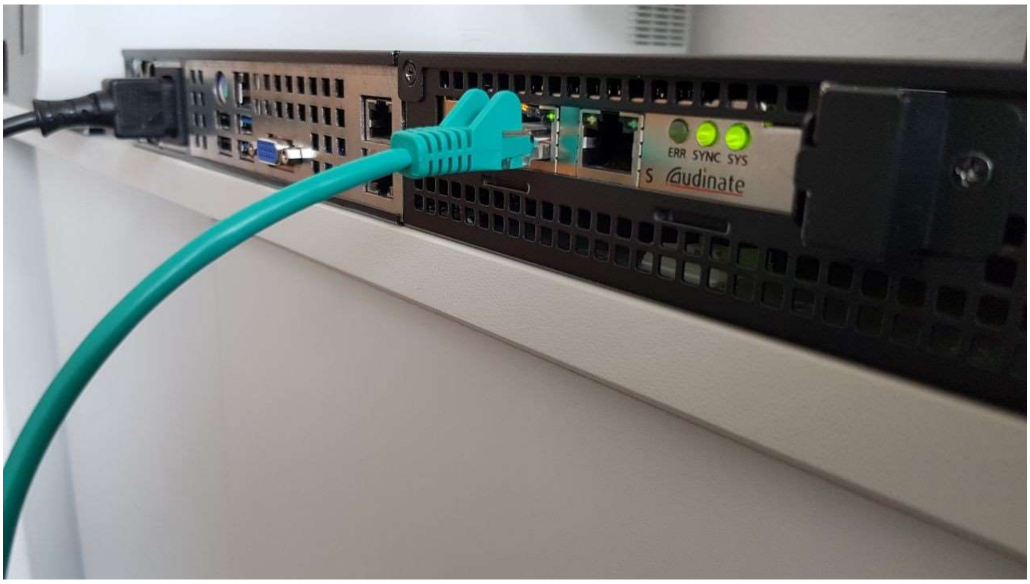
Dante™

The Challenge

How to receive a SIP call in a Dante network without using multiple converters?

Studios with Dante may want to easily access the remote studios and voice-over artist with the same management of audio as the rest of their Dante system. Typically, there are only audio codecs with analog or digital inputs available while some may now support AES67 as well, but even those will often feature a strongly broadcaster-driven functionality.

Studio operators have asked Ferncast to expand the software audio streaming solution aixtream™ towards a direct Dante to SIP conversion. They wanted to be more flexible and also have the complete SIP audio operation under their direct control.

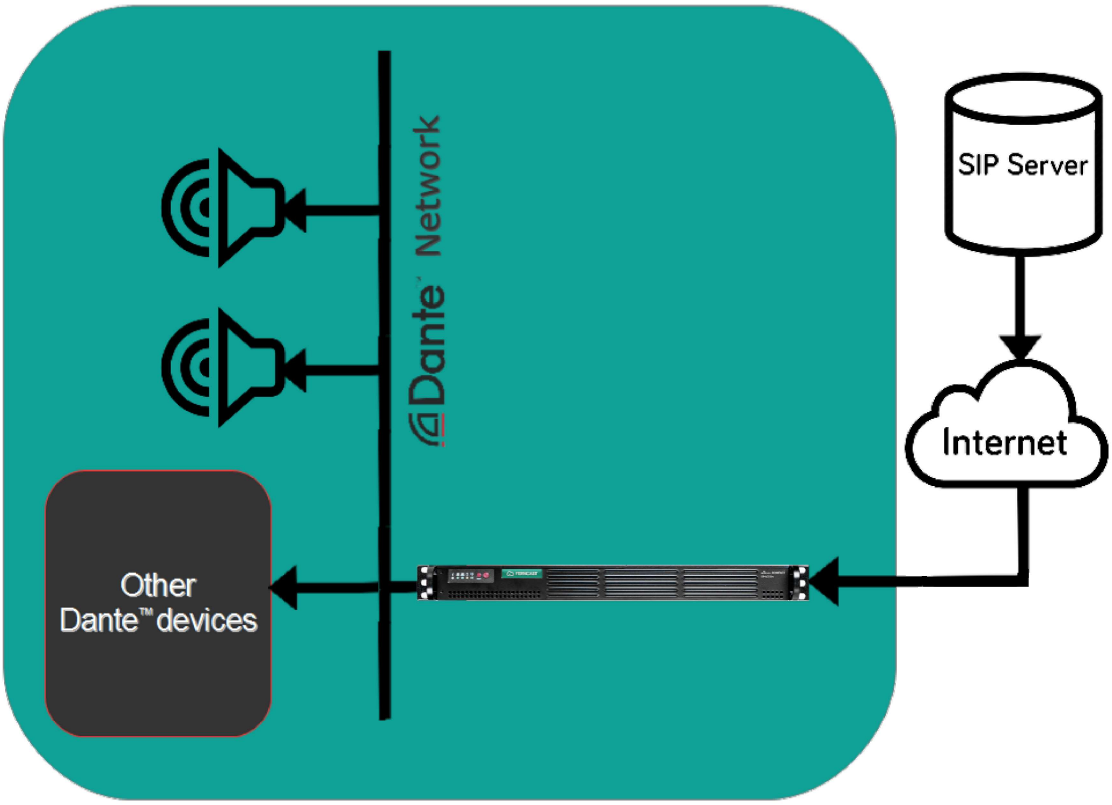


Hardware with Dante audio card

The Requirements

An existing Dante network and the right software

It is necessary that a server running audio codec software, such as **aixtream** or a product such as an **Audio Codec Server** already includes and supports Dante hardware. This allows a direct integration of the audio codec into a Dante environment. The type of Dante interfaces may vary depending on the number of channels which are required to communicate with other remote sites. Some audio codec products also support USB audio interfaces, which can also include Dante-to-USB interfaces.



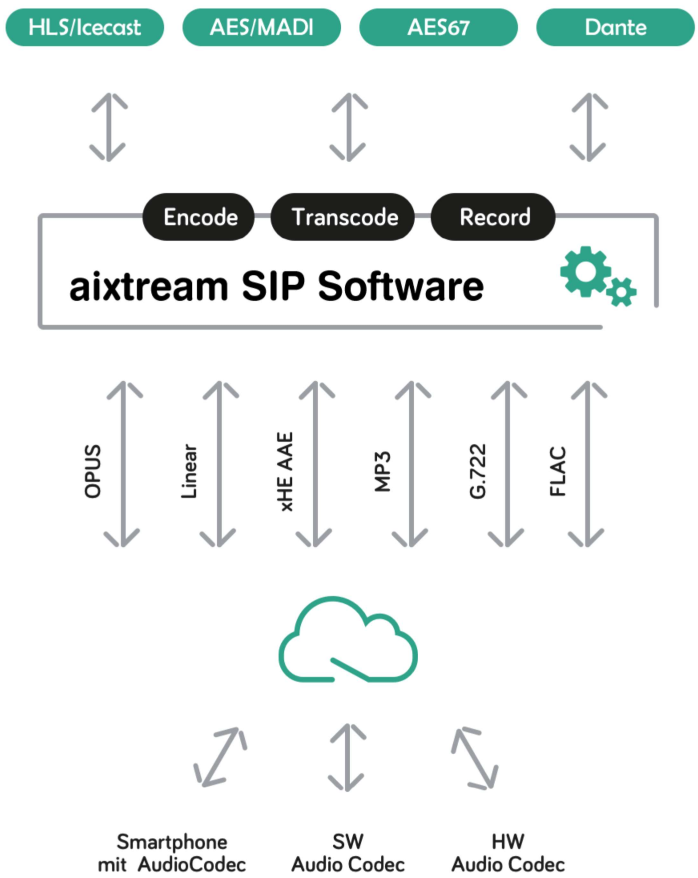
SIP to Dante network via Audio Codec Server

The Solution — aixtream

An all-in-one solution for SIP-to-Dante

aixtream™ from Ferncast is a software which can run on servers, either as full installations or virtualization and is supported by a large variety of different hardware setups, either generic or customized by Ferncast.

This software is entirely input-output-agnostic. No matter which audio input the user receives and which output they desire, aixtream can handle it. This makes it a perfect choice for any complicated input/output scenario, including SIP to Dante. In addition it offers many features to simplify the audio workflow of any network with additional scheduling and monitoring features.



Additional Benefits

aixtream as the premier solution

aixtream in a Dante studio environment has quite a number of interesting benefits for the operators:

- **Full gateway functionality** for Dante input to SIP output and vice versa
- **Highly available for 24/7 operation** including the option for an automatic fall back redundant system
- **Replaces multiple other devices** as an all-in-one solution
- Comprehensive monitoring of operations, including alarms and fast adaptation
- Regular quality-of-life updates offered by Ferncasts online customer service center
- A steadily increasing number of features offered by aixtream, e.g. network statistics and analysis, multi-channel encoding, e.g. AC-3