Strategies for Live Ambisonics Ensembles

Spatial Audio Workshop, Oulu, Finland

Roberto Becerra, Matas Samulionis

December 6, 2024

Contents

1 Introduction

Figure 1: Dante's inferno.

1.1 Problem Description

The integration of multiple performers in a live Ambisonics ensemble presents challenges in spatial audio rendering. These challenges have to do with having to aggregate all the performers' multichannel signals in order to route them out of one single hardware system. One more difficulty arises from the monitoring oneself during the performance itself. Things can get pretty messy, the sweet spot maybe taken and is hard to understand what is happening in the listening space. This workshop explore strategies for effectively managing digital audio formats, network configurations, and software tools to facilitate live performances in an Ambisonics context.

1.2 Network Audio

Network audio, also known as audio-over-IP (AoIP), refers to the transmission of digital audio signals over Ethernet-based networks. This approach leverages existing IT infrastructure to enable scalable, flexible, and high-quality audio distribution. Technologies like Dante, AES67, Ravenna, and AVB (Audio Video Bridging) are commonly used in modern network audio systems, each providing unique features and levels of compatibility with established standards.

The primary advantages of network audio include reduced cabling complexity, high scalability (support for hundreds of channels), and the ability to integrate audio, video, control signals over the same network, remote control and monitoring of devices.

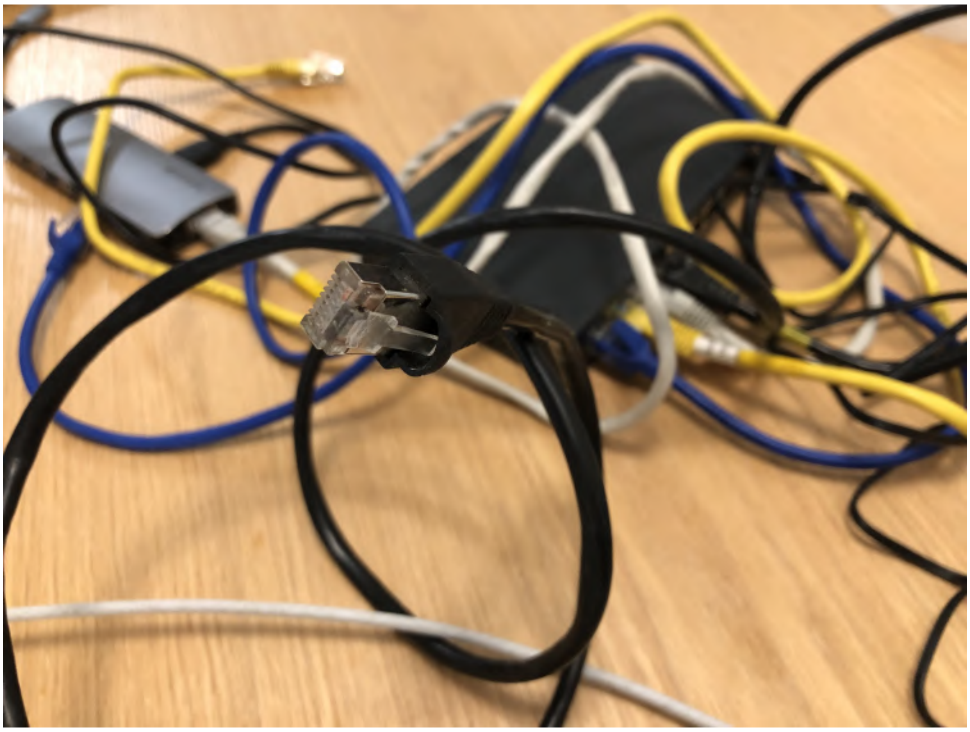

Figure 2: Network audio.

However, difficulties include: network configuration, knowledge of networking, etc.

For this workshop, we focus on a simplified implementation of a Dante network, that can be useful in the case of multiple performers or a number of other scenarios.

1.3 PCM (Pulse Code Modulation)

Before we begin, it is worth going throug some foundational concepts that take place in digital and network audio. Such is the case of the audio format that is actually transmitted over this medium: PCM.

PCM is the most widely used format for digital audio. It represents audio as discrete samples of amplitude at regular intervals. Typically found in formats like WAV, AIFF, and used in audio transport systems like Dante or MADI. When exporting a .wav or .aiff file from your DAW, this is what you get most of time.

Characteristics:

- High fidelity: PCM offers high-resolution audio, with formats ranging from 16-bit/44.1 kHz (CD quality) 16 bit-depth
- Other qualities are 48, 96, 192kHz and 24, 32 bit-depth.
- Low latency: PCM is widely supported in live sound systems, including Dante audio networks, enabling real-time, low-latency audio transport.
- Editability: PCM audio is straightforward to process and edit. Althought in modern environments also other [compressed] formats can be easily imported and manipulated

- High bandwidth: High sample rates and bit depths require significant bandwidth, especially in multichannel configurations common in Ambisonics systems.
- Storage and processing: Higher-resolution PCM formats demand more storage and computational resources.

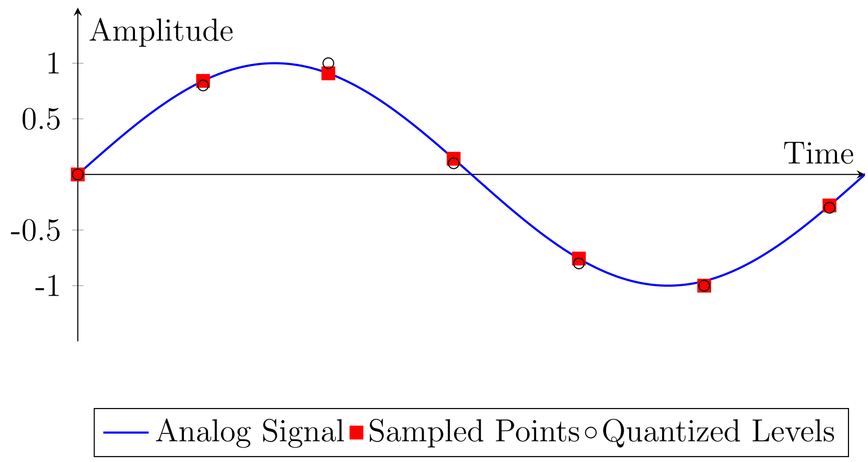

Figure 3: Illustration of Pulse Code Modulation (PCM). The analog signal (blue curve) is sampled at discrete intervals (red squares) and quantized to discrete levels (black circles). Quantized levels show that there can be quantization error.

1.4 Digital audio transport formats

We have chosen Dante for this solution, but may as well have chosen [or not] some other formats. Here is a brief overview of other available options

1.4.1 MADI

The Multichannel Audio Digital Interface (MADI) is a digital audio standard designed for up to 64 channels audio transmission. It is used in live sound, broadcast, and studios.

Key features of MADI:

- **Transmission Medium:** Typically uses coaxial cables with BNC connectors or optical fiber cables.
- **Channel Support:** Supports up to 64 channels of audio at 48 kHz and 24-bit resolution. Higher sample rates (e.g., 96 kHz) reduce the number of channels.
- **Applications:** Professional audio systems for connecting mixing consoles, audio routers, and multichannel recording systems.
- **Advantages:** Low latency. Large-scale setups where many audio channels need to be used.
- **Additional Features:** Long cable runs are possible, up to 100 meters with coaxial cables or several kilometers with optical cables.

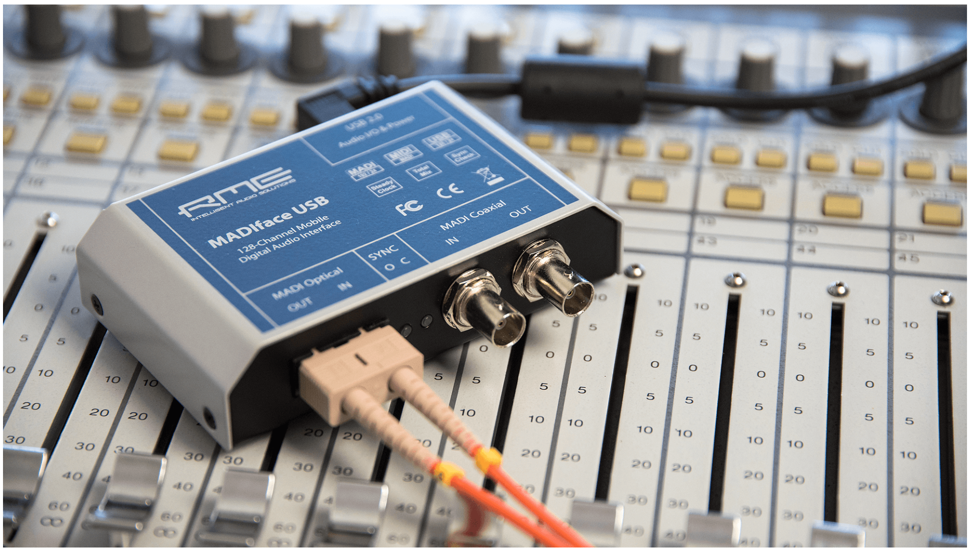

Figure 4: MADI interface and cable. (rme-usa.com)

1.4.2 S/PDIF

The Sony/Philips Digital Interface Format (S/PDIF) is a widely used protocol for transmitting digital audio between consumer and professional audio equipment. It is designed for stereo audio and supports both compressed and uncompressed data formats.

Key features of S/PDIF:

- **Transmission Medium:** Uses coaxial cables with RCA connectors or optical cables (TOSLINK).
- **Channel Support:** Transmits two channels of audio (stereo).
- **Sample Rates and Bit Depths:** Supports sample rates up to 96 kHz and bit depths up to 24-bit.
- **Applications:** Used in home theater systems, audio interfaces, and for interconnection of digital devices like CD players and DACs.

1.4.3 ADAT

The Alesis Digital Audio Tape (ADAT) protocol was developed by Alesis for transmitting multichannel digital audio. Used in studio setups for interconnecting audio gear.

Key features of ADAT:

- **Transmission Medium:** Uses optical cables with TOSLINK connectors.
- **Channel Support:** 8 channels of audio at 48 kHz and 24-bit, or 4 channels at 96 kHz.
- **Applications:** Studio setups for linking audio interfaces, digital mixers, and multichannel preamps.
- **Advantages:** Compact, affordable, and easy to implement for multichannel audio.

Figure 5: S/PDIF cables. (techterms.com)

Figure 6: ADAT audio. (futuremusic-es.com)

1.4.4 AVB

Audio Video Bridging (AVB) is a network protocol suite based on the IEEE 802.1 standards, designed for real-time, low-latency transmission of audio and video over Ethernet networks. AVB is widely used in professional audio environments, broadcast, and multimedia installations.

Key features of AVB:

- **Transmission Medium:** Uses standard Ethernet cables (Cat5e, Cat6, or fiber-optic) for audio and video transmission.
- **Channel Support:** Supports high channel counts, scalable based on network bandwidth and device capabilities.
- **Applications:** Live sound, recording studios, and multimedia installations.
- **Advantages:** Precise synchronization using IEEE 802.1AS (Precision Time Pro-

tocol, PTP), reduced network congestion through bandwidth reservation, and interoperability with other AVB-compliant devices. Apple support by default.

- **Disadvantages:** Cost as dedicated hardware switches are required. .

Figure 7: AVB interface

1.4.5 A mention of Clock in Digital Audio Formats

In digital audio systems, the concept of a "clock" is fundamental for maintaining synchronization across devices. A clock ensures that all devices in a network operate at the same sample rate. Problems with clocking can cause clicks, pops, and other artifacts caused by mismatched timing.

In MADI systems, word clock is often distributed via dedicated BNC cables, with one device designated as the clock master. ADAT uses embedded clocking signals within its optical connections, which are sufficient for small systems but may require external word clock for complex setups. AVB employs the IEEE 802.1AS standard, which provides precision time protocol (PTP). Similarly, Dante uses PTP but employs a simplified, robust implementation tailored for audio networks, designating a master clock through an automated election process.

Dedicated hardware clock generators are often used to provide a stable timing reference, for minimal jitter and maximum audio fidelity.

Figure 8: Antelope clock device.

2 DANTE

DANTE (Digital Audio Network Through Ethernet) is an audio-over-IP (AoIP) protocol developed by Audinate. It is scalable and designed for high-channel-count, low-latency audio networking.

Key features of DANTE:

- **Transmission Medium:** Uses standard Ethernet networks with Cat5e, Cat6, or fiber-optic cables.
- **Channel Support:** Can support hundreds of channels at sample rates up to 192 kHz and 32-bit resolution. Though more frequently setup is limited to 64 channels. Up to 512 channels is achieved for instance with PCI interfaces.
- **Applications:** Live sound, broadcast, recording studios, and large-scale audio installations.
- **Advantages:** Flexible, supports complex routing, and integrates with existing IT infrastructure.

Here we cover the installation and basic configuration for 3 elements of the Dante system:

- Dante Virtual Soundcard (DVS)
- Dante Controller
- Dante Via

We will also cover basic of network requirements, device configuration, and typical layouts for Dante-enabled systems.

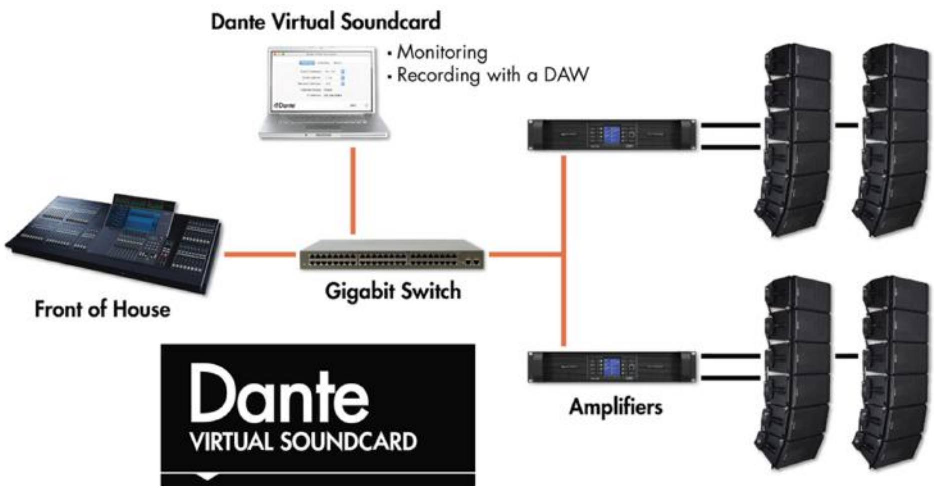

Figure 9: Dante Network example.

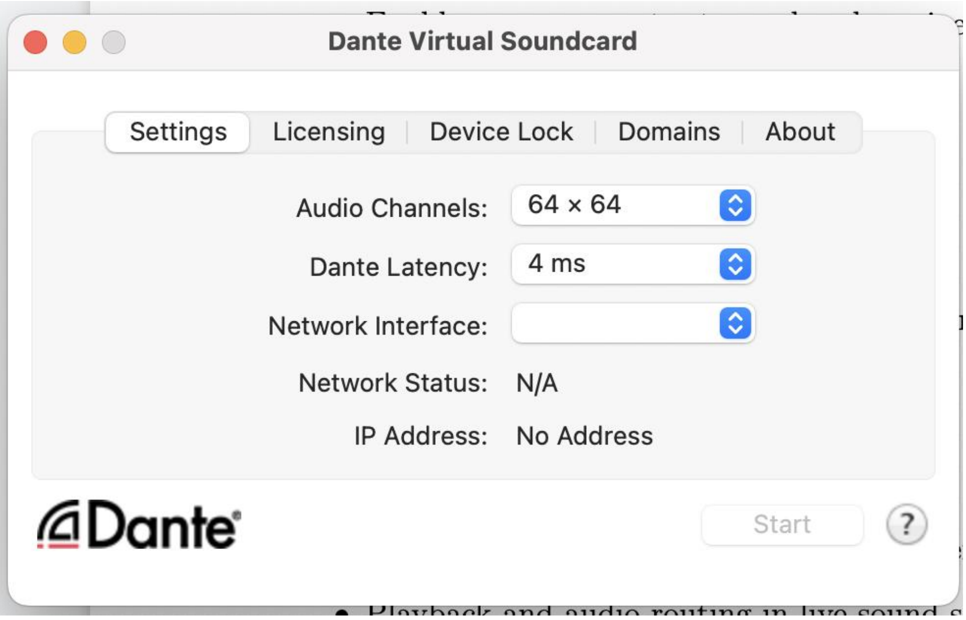

Figure 10: Dante Virtual Soundcard.

2.1 Installing Dante Software

2.2 Dante Virtual Soundcard (DVS)

Dante Virtual Soundcard (DVS) is a software application that turns your computer into a Dante-enabled audio device.

What DVS Does:

- Emulates a hardware soundcard but operates over the Dante protocol.
- Enables your computer to send and receive audio channels via a Dante network.

How DVS Works:

- DVS uses the computer's Ethernet port to connect to the Dante network.
- It supports standard audio driver formats like ASIO (Windows) and Core Audio (Mac).
- Audio software (e.g., DAWs, media players) interfaces with DVS as if it were a physical soundcard.

Install DVS:

1. Download Dante Virtual Soundcard from the official Audinate website: Dante Virtual Soundcard.
2. Install the software by following the on-screen instructions.
3. Activate the license using the activation key if you purchase, or the trial key.

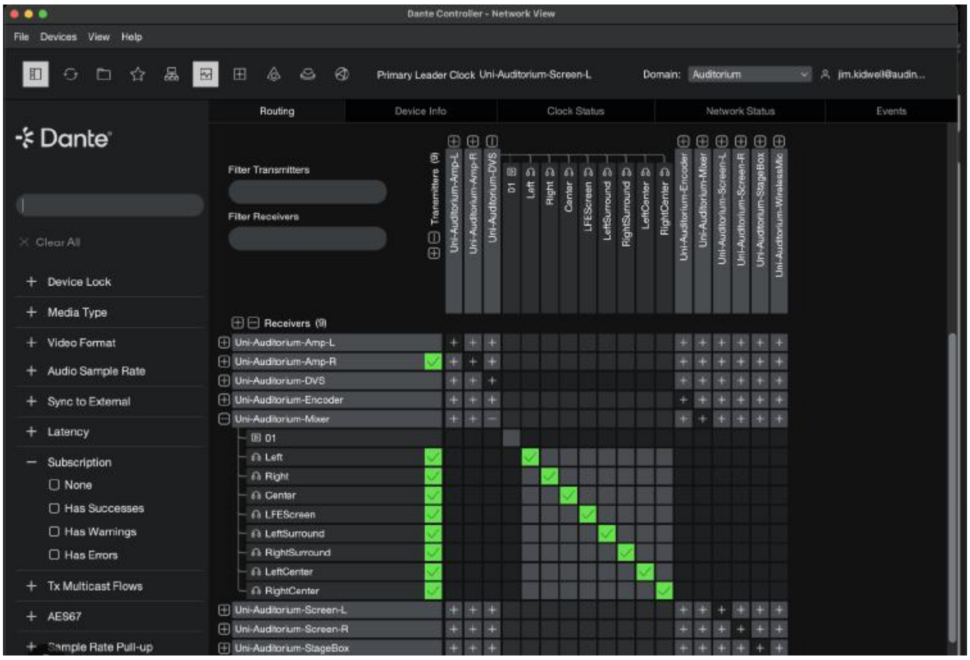

Figure 11: Dante Controller.

2.3 Dante Controller

Dante Controller is a software application used for managing and configuring Dante-enabled devices in your network.

What Dante Controller Does:

- Detects all Dante-enabled devices in a network.
- Allows users to route audio channels between devices using a simple matrix.
- Provides tools for clock synchronization and monitoring device status.

How Dante Controller Works:

- It communicates with devices over the Dante network via the Ethernet protocol.
- Displays a routing grid where users can click to map source channels to destination channels.
- Monitors latency, device health, and clock synchronization across the network.

Install Dante Controller:

1. Download Dante Controller from the Audinate website: Download link
2. Install and run the software.

2.4 Dante Via

Dante Via is software that allows users to integrate non-Dante devices and applications into a Dante network.

What Dante Via Does:

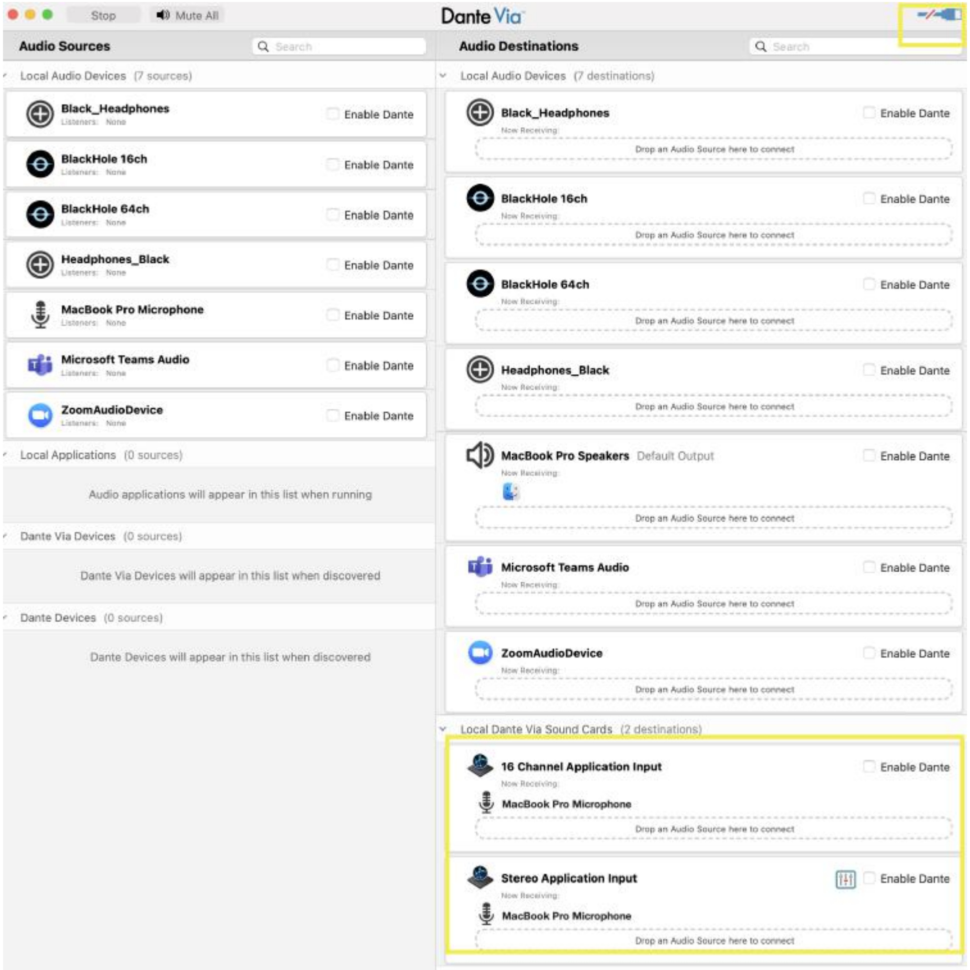

Figure 12: Dante Via.

- Captures audio from non-Dante hardware (e.g., USB microphones, built-in soundcards).
- Routes audio between software applications (e.g., streaming tools, DAWs) and the Dante network.

How Dante Via Works:

- Detects audio sources and destinations on the computer (e.g., local soundcards, applications).
- Creates virtual devices to make non-Dante sources appear as Dante devices in the network.

Install Dante Via:

1. Download Dante Via from the official website: Dante Via.
2. Install the software and follow the activation process.

2.5 How They Work Together

1. **DVS and Dante Controller**: DVS acts as a virtual soundcard for your computer. Dante Controller detects and configures DVS as a Dante device, allowing you to route its audio streams. You select DVS as output on your DAW and send channels as you would do with any other interface.

2. **Dante Via and Dante Controller**: Dante Via integrates non-Dante devices and software into the network. Dante Controller can then route audio from these sources just like any other Dante device. In order to send the DAW's autio into the Dante network you need to check "Enable Dante" in the Via window. Then from the DAW you will select the Dante Via 16 ch as output.

3. **DVS and Dante Via**: No Go.

In general, to emphasize, when there isn't any Dante Enabled hardware device, a computer running Dante Via will serve as the *network provider*, which means it will create the Dante network for everyone to use. The limitation of this is that Dante Via only support up to 16 channels.

3 Network Configuration and Layout

3.1 Basic Network Requirements

- **Gigabit Ethernet**: Ensure all switches and cables support Gigabit speeds.
- **Quality of Service (QoS)**: Configure switches to prioritize Dante traffic.
- **Redundancy**: Use Dante-enabled devices with redundant Ethernet ports for failover support.

3.2 Configuring Dante Virtual Soundcard

1. Open Dante Virtual Soundcard and configure:
   - **Audio Driver**: ASIO (Windows) or Core Audio (Mac).
   - **Channel Count**: Set to match your needs (e.g., 16x16, 32x32).
   - **Network Interface**: Select the active Ethernet adapter.
2. Start the Virtual Soundcard and verify in Dante Controller.

3.3 Using Dante Controller

1. Select the same interface as used in DVS.
2. It will automatically detect all Dante devices on the network.
3. Route audio by matching source and destination channels in the routing matrix.
4. Monitor latency, clocking, and device health.

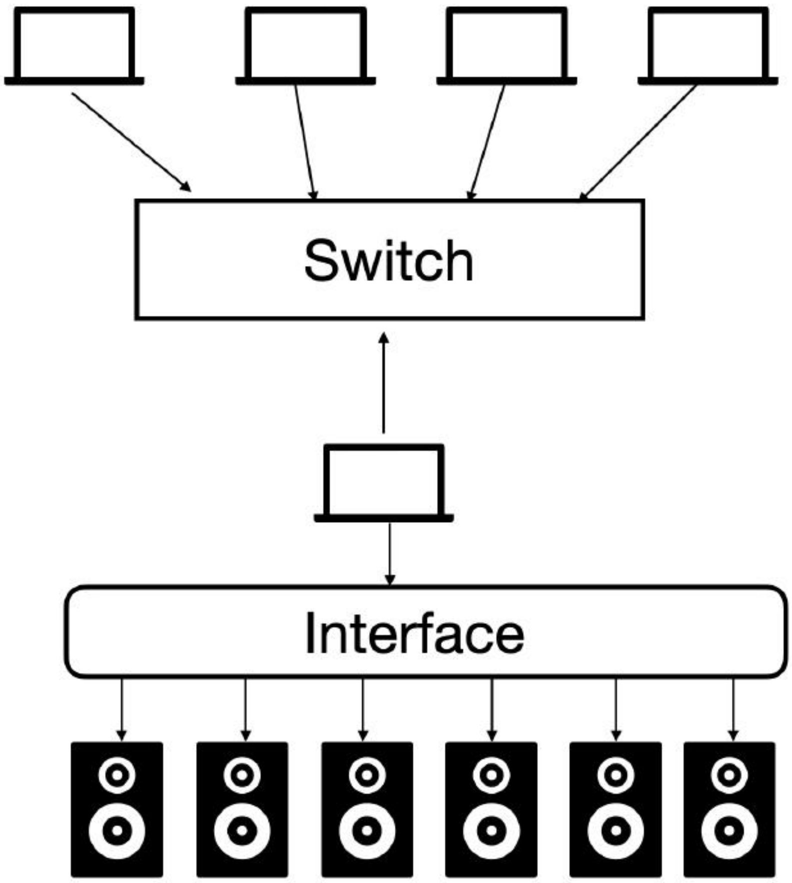

Figure 13: Dante network.

3.4 Using Dante Via

If there isn't a Dante enabled device to *provide* the Dante network, you can use Dante Via in on of the ensemble's computers. This will create a stable Dante network.

1. Select the same interface as used in Dante Controller.
2. Dante Via and DVS cannot be used simultaneously.
3. Possible sound sources and destinations will be shown in the Via window.
4. Click on "Enable Dante" for those sources you want to send over the Dante network (i.e. your DAW).
5. You can also drag and drop sources to destinations in order to send audio in between applications inside the same computer.

3.4.1 Possible Issues

- **Security:** Firewall settings may block Dante traffic.
- **Licensing:** Ensure valid licenses are obtained and activated.

- **Operating Systems:** Compatibility issues between different OS versions.
- **Cock issues**: Two Dante Via working at the same time. Saturated network. Bad Switch.

3.4.2 Useful Links

- Dante Virtual Soundcard
- Dante Controller
- Dante Via

4 Ambisonics Integration with Reaper

Now the time has come to finally route the audio to Reaper and use Ambisonics

4.1 Multichannel, Multiuser, Aggregated Ambisonics in Reaper

The approach to aggregating multiple audio sources into a unified Ambisonics output is to send send audio over Dante to a "main" computer. To this end, any of the cother computers will have to take the role of Dante Via in order to create a stable Dante Network.

IMPORTANT: Then clicking on the routing matrix in Dante Controller to send audio from Dante Via to another computer's DVS, a pop-up dialogue will appear in the Via's computer asking to allow the DVS computer to access the audio "endpoint", which means the signal being sent. If you don't accept this audio will not reach the DVS computer.

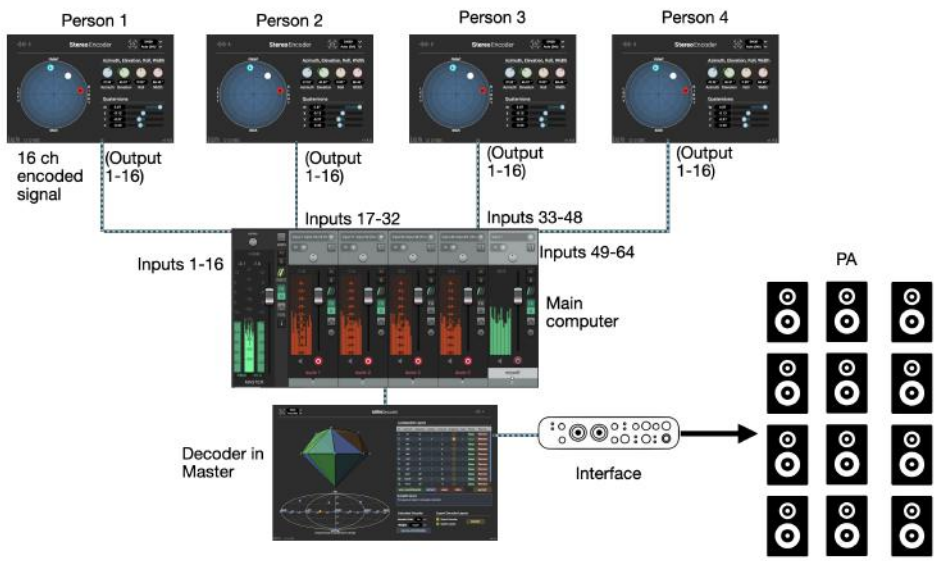

Figure 14: Ensemble Dante layout

Combining all elements into a Reaper project for live performance means to send over DANTE the encoded multichannel audio signals to a main computer that will aggregate all performer's audio into a single Decoding Bus.

Given that for the configuration at hand we have a maximum of 64 channels available in the Dante network, we can group of to 4 people sending 3rd order encode Ambisonics audio. And a fifth performer will take the role of the main computer that does not put audio out to the Dante network, but instead receives all the audio from the network, decodes it and sends it to the Hardware system.

The main computer then will have to create an **aggregate device** that includes the Dante Virtual Soundcard and whichever interface device is used to connect to the hardware output.

So for instance, person 1 will send his/hers encoded signal over outputs 1-16, and the main computer will receive that signal into inputs 1-16. Then, person 2 will send his/hers encoded signal over outputs 1-16, and the main computer will receive that signal into inputs 17-32. Person 3 will send his/hers encoded signal over outputs 1-16, and the main computer will receive that signal into inputs 33-48. Finally, person 4 will send his/hers encoded signal over outputs 1-16, and the main computer will receive that signal into inputs 49-64.

All 4 performers' audio will be then sent to a master track bus where the decoder, sending all G-format audio the hardware. The 5th performer, or main computer is also able to encode audio and send to the same master bus. But because this performer does not have to go into the Dante network, his/her audio can be of higher order.

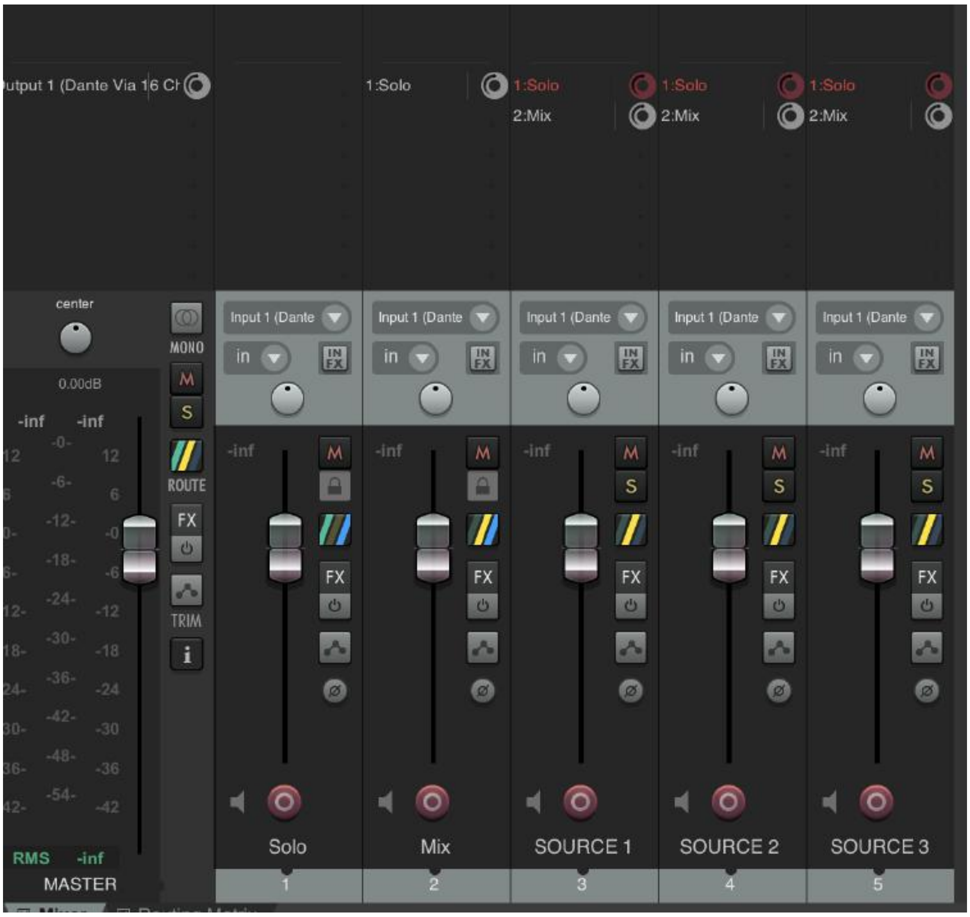

Figure 15: Reaper project configuration for live Ambisonics ensemble using ReaSolotus

4.2 Using ReaSolotus in Reaper

Employing ReaSolotus is a way to solo individual tracks and monitor binaurally during live performances.

During a performance of this kind things can be hard to monitor, and it may be confusing for performers to know how and where is their sound source being spatialised. Tools like Solotus can help by enabling a sort of PFL, or monitoring bus.

By installing ReaSolotus, Reaper is able to create a MIX and a SOLO bus. The MIX bus will sever as the new Master track, from which audio is sent to the hardware system. The SOLO bus starts as a copy of MIX, but when pressing Solo in any of the source tracks, it will contain that solo'ed track only. Much like what happens in concert mixers with monitoring and PFL. In the Spatial Audio performance context, this means that we can use a Binaural decoder and send that stereo signal to an alternative output where we can monitor with headphones.

Download solotus Here: .dylib file if you use mac, or .dll if windows.

4.2.1 Install and use

- In Reaper, go to Options - Show REAPER resource path in explorer/finder.
- Navigate to the Extensions Folder:
- Look for the UserPlugins or Plugins folder within the resource path, and put the downloaded file there.
- If you don't see the folder, create a new one named UserPlugins.
- Restart Reaper and go to Actions, filter by "solotus", select ReaSolotus init on a fresh Reaper project. This will create the Solo and Mix buses.
- For every new track, before configuring it, select it, and again select ReaSolotus init from actions. This will configure the sends from such track.
- After, that, configure all the channel counts and sends and receive from source tracks, solo and master.

This soloing, of PFL method is a good option to monitor oneslef in the case of growing cacophony when artist get together and they have brilliant ideas on making Ambisonics live music.

5 Dongle Music!

We finally get to make dongle music!

6 Troubleshooting

- Glitches - There may be clocking issues. Check your switch and or combination of devices. .

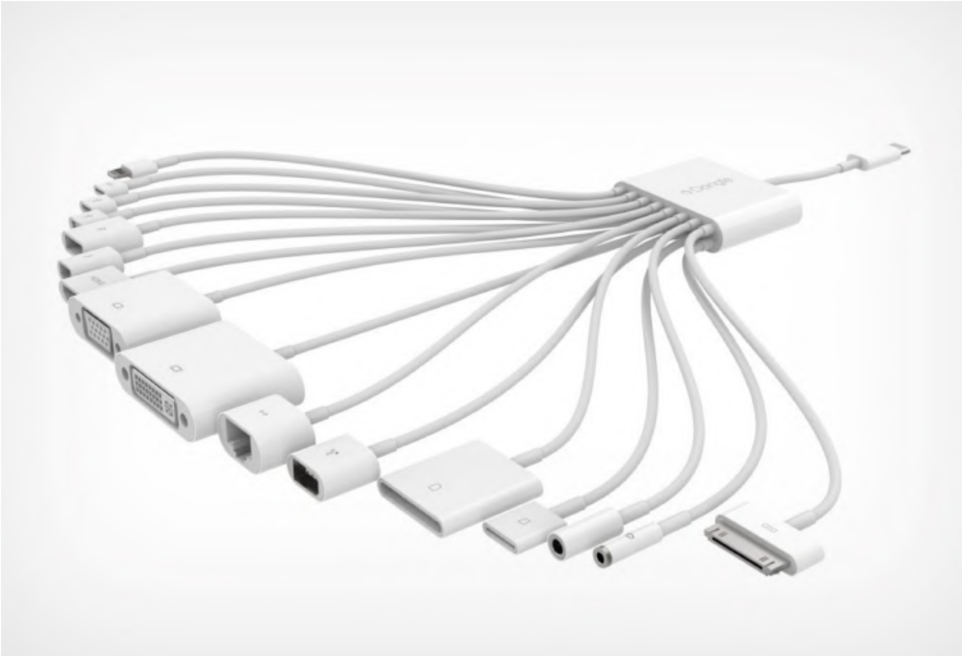

Figure 16: Dongle Music

- Via is crashing- Re-install Via.
- Dante Virtual Soundcard does not start. Error 512 - Restart computer.
- Connection is unstable, clock issues - Chech that your Ethernet adapter supports GB and not only MB speeds.
- Unable to receive performers audio. Unable to mix with the own main computer's audio - Receive performers audio in a seperate track than the one you use for the main computer's generated audio. Send everything to a separate master bus.
- Unable to receive audio from Dante Via computer - Make sure you accepted the pop-up that asks permission so the DVS computer can access the audio "endpoint" of the Via computer.

References

- Reaper Digital Audio Workstation
- IEM Plug-in Suite
- Audinate (Dante Technology)
- Dante Downloads
- ReaSolotus