Dante Networking and System Design at MTSU

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MRAT 6650:

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Spring 2024

Special Thanks and Acknowledgements

Special thanks to my family, especially my father and mother who have patiently bore with me while I stressed out and rambled on about this project. They have encouraged me to continue experimenting and pushing forward with my education. Yakoke micha chi'hollo'li.

Special thanks to Michael Hanson, my primary project advisor who has graciously provided opportunity, equipment, and (most importantly) advice. I am truly grateful for his time and effort that he has invested in this project, and me. I am also grateful for each long conversation we have had about engineering or just life in general, especially the many times he has put his faith in me and scolded me to aim higher in what I do.

Special thanks to Frank Baird, my secondary advisor, who has also been a huge source of encouragement. At every meeting and conversation, he has been nothing but inspiring for me and has thrown out many ideas to chase and introduced me to many great connections.

Special thanks to Sadika Anderson, Mac Palao, Riley Jermier, Mark Smith, Christian Hannah, Aaron Walden, and Danny Maloney for their help in the project and Final Session

Special thanks to Lee Whitmore at Focusrite who provided an early release of their RedNet PCIeNX card to me for this project.

Abstract:

Dante’s networked audio solutions have been growing in popularity for many years. The study of how Audio-over-IP is an increasingly important topic for audio engineers, but few understand the principles behind how transmission is accomplished using computer networking. This Final Project delves into the foundational concepts of how networks are used for this purpose and what Dante provides. It also provides case studies through a journal-style approach of how Dante technology can be utilized in remote sessions, livestream concerts, and larger inter-studio sessions. This leads to the end-product of a system design for Middle Tennessee State University to be able to implement Dante into its’ recording facilities.

MIDDLE TENNESSEE STATE UNIVERSITY GRADUATE COLLEGE

DEFENCE COMMITTEE AND FINAL READING APPROVALS

Of the Final Project submitted by

Dale Shackleford

The following individuals read and discussed the final project submitted by student Dale E. Shackleford, and they evaluated the student's presentation and response to questions. They found that the student passed the final written examination.

Prof. Michael Hanson Chair, Primary Advisor of Committee

Prof. Frank Baird Member, Secondary Advisor of Committee

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PROJECT PROPOSAL

Executive Summary:

As our world continues to progress further and further forward into the digital era, computer networking becomes increasingly more important. The audio industry is no exception to this shift as needs become more prevalent to send transmissions faster and over farther distances. Analog technology has limitations that cannot address these particular needs. Instead, using networks to transmit audio and visual data has become the solution. Dante designed by Audinate is one such protocol created to interface professional audio equipment over a network. The goal of this Final Project is to connect MTSU studios within Bragg together through a Dante network. This will bring near latency free audio from one location to another, allow students and instructors innate access within these studios to learn and teach modern networking protocols, and allow for more flexible work flows wherein studios could be chained together for larger recording events and sessions.

Project Concept and Description:

This project will build a Dante network between studios and classrooms on MTSU's campus. This will begin a more publicly usable audio-over-IP network that teachers and students alike will be able to access for teaching and study. Once completed, this network will interface

Control Rooms A-E, Live Tracking Rooms A-E, Lab A (Mastering Lab), Mix Lab (Bragg 180), WMOT, WMTS, Post Lab, The Tennessee Room, Hinton Hall, and Studio 1/TV 1. Once this system is installed and operational, the designs of this production will foster easy expandability to additional rooms or other buildings as the university sees fit. This project is not meant to build an entirely fixed and comprehensive Dante set-up for MTSU, but rather a solid beginning that will help demonstrate its' potential.

In order to accomplish this project, I will be researching the history and best practices for Dante networking. Networked audio is not currently taught in depth in MTSU classes nor is required in traditional analog thus my independent research component will be vital to the success of this project. Such concepts as network topology, Power of Ethernet (PoE vs PoE+ vs PoE++), IGMP Snooping, and Quality of Service (QoS) will be of major concern to my research. The knowledge gained in this research will directly inform the decision-making process used to integrate this network at MTSU.

As this project aims to build a network at MTSU and not another location, how each room is configured will be completely dependent on the needs of each room. PoE may not be needed in some places but will be needed in others. Similarly, the number of ports one room may require might be more than needed in others. The designs for this Dante network will take considerations like these into account in creating a custom network for MTSU's needs. Part of the configuration will also be in utilizing Dante Domain Manager. This program allows for additional control of Dante on a network otherwise unavailable. MTSU currently already owns a licensed copy of Dante Domain Manager, and I will be spending time learning its ins-and-outs to streamline the workflow of Dante.

Since Dante is built on the backbone of regular computer networking, I will also be researching best practices for installation of networks both with and without Dante in mind. Among the functional concepts of research involved for this project is network security, especially in a university setting. I will be working closely with MTSU's I.T. Department to ensure that everything is done cleanly and properly. The degree with which we work together will greatly help or hinder the integration of other buildings other than Bragg on campus.

Another portion of research that this project will include is Dante's newer video abilities. While it will not be the primary focus of this project, it is important to recognize the potential impact on network design that video will require. As such, the network design will take into consideration the needs of Dante video for future expansion.

The practical application phase of this project will focus on implementing the designs developed during the research phase. An itemized budget of suggested gear for MTSU to completely hard-install equipment will be created and delivered to the RIM department for future recommended development. The rooms will be connected within the network and tests will be conducted to ensure its operability. Dante Domain Manager and managed switches on the network will be configured for the best functionality possible. By using Dante Domain Manager this project will create the network designed in a star where the Central Machine Room (CMR) will be the main hub for disbursement (see appendix A). Each location where Dante will be routed will also have an additional line installed for a future redundancy network. If accessible, Dante Domain Manager may move to being housed within ITD's main network infrastructure to streamline connections.

Finally, a test demonstration will be conducted to prove the viability of this Dante Network. This demonstration will include three studio control rooms recording a session or performance of a band in a singular location. By using the network, we will be able to have multiple engineering teams working on the same record simultaneously in separate locations. To support the event, I would like to recruit undergraduate classes to compose the engineering teams in two of these studios to help expose them to the potential of networked audio. The third team will be the “lead” team of the session (including the producer) which will be comprised of graduate students like myself. To aid this large session, integrating Dante’s video component into this event for communication and documentation purposes could be extremely valuable and is something to consider.

The concept for this project comes at a crossroad of three primary driving forces: the desire to challenge myself in learning, a directly applicable skill and product that will aid my career post-graduation, and a final product that will have relevance, application, and value to an audience greater than just myself. Designing and implementing a network for audio across the studios at MTSU satisfy all of these personal requirements. This project will prove challenging as I learn the ins-and-outs of Dante, especially considering the environment being a public university with its’ added potential red tape. Once completed, this will greatly aid me as an outgoing student have both a new set of skills and a physical living example of my work to share with potential future employers or industry members. As such, it is relevant to myself, to future business partners, as well as current MTSU staff and students. Creating such a network will allow teachers the ability to easily teach networked audio and its’ applications without having to set up temporary rigs or having to worry about students being able to access such equipment for study. This also has great implications for the ability for individuals to have a singular band and multiple control rooms tracking concurrently or for one control room to have access to multiple live rooms at once for larger projects.

Defined Audience for final product:

As mentioned before, the audience for this project is two-fold. In the immediate future its primary audience will be MTSU faculty and students. Farther in the future, this will serve as a calling card to business partners of the kind of installation work and projects I am able to deliver upon.

Goal/Learning Outcomes:

The goal and learning outcome of this project is to learn how to create a complex Dante system from the ground up. By embarking on this project, I will learn more about Dante and digital networking than a passing user is required to know while making something valuable for others at the same time. I will also learn more about modern studio and live workflows which utilize networked audio (both Dante and other protocols).

Deliverables:

The deliverable for this project will be a network integration upgrade for MTSU's Rim and MRAT departments. This will connect via Dante Control Rooms A-E, Live Tracking Rooms A-E, Lab A (Mastering Lab), Mix Lab (Bragg 180), WMOT, WMTS, Post Lab, The Tennessee Room, Hinton Hall, and Studio 1/TV 1. This will allow for teaching of modern audio workflows and network integration within these classrooms. Once the install is complete and tested, to demonstrate its' potential and operability, I will produce a recording session wherein a band will play in a singular room with three engineering teams tracking the band simultaneously in other available networked rooms (Control Room A-E, Mix Lab, Post Lab). Also, once this project is finished an itemized recommended equipment budget for complete installation will be provided to the RIM department.

Personnel Involved in Project:

Dale Shackleford – Student Producer

Michael Hanson – Primary Advisor

Frank Baird – Second Reader

Three student engineering teams

One Band for demonstration session

Timeline

Summer 2023 (non-enrolled)

- May-July
  - Research Dante Network Designs and best practices
  - Continue communications with MTSU’s ITD
  - Begin developing network designs
- August
  - Finalize network designs
  - Compile and finalize research analysis (literature review)

Fall 2023 (6 credit hours)

- August
  - Submit project research analysis (literature review)
- September
  - Begin installation and running cabling for network
  - Connect two locations and test connectivity
- October
  - Continue running cabling if needed
  - Continue small scale testing
  - Gain network access to additional buildings outside of Bragg
  - Configure primary network switch
  - Begin configuring Dante Domain Manager
- November
  - Continue configuration of DDM and primary switch
  - Continue small scale testing
- December
  - Finalize DDM and switch configuration
  - During break book multiple studios and test multicast operability between studios and/or external rooms

Spring 2024 (3 credit hours)

- January
  - Finish installation and testing of the Dante network
- February
  - Conduct multi-studio demonstration session
  - Begin writing post project analysis
- March
  - Post mix recordings and video from demonstration session
  - Continue writing project analysis
- April

    - Submit project reflection and analysis
- May
    - Celebrate
    - Sleep

INTRODUCTION

Professional audio recording and transmission has greatly changed since the times of wire recorders and wax cylinders. The era of phonographs transitioned to analog tape which gave way to digital recorders. The ability to fixate audio in a digital format allowed for access and flexibility that could never have been achieved with analog electrical signals. Modern computer networking has pushed the envelope even further and now instantaneous transmission of professional quality audio in real-time to multiple destinations is not only a reality but is easy to achieve with Audio-over-IP (AoIP) solutions like Dante.

The purpose and goal of this final project is to study Audinate's Dante (Digital Audio Network Through Ethernet) and how it works, and to create a practical installation design for Middle Tennessee State University's campus studios. This first portion of the project, the literature review, recounts the research discovered on how Dante operates both from a user and a system designer's perspective. Upon completion, testing of Dante equipment and campus infrastructure will be conducted after which the design recommendations will be submitted and a live demonstration of Dante's capabilities on MTSU's campus will be displayed through a multi-studio recording session.

SECTION 1: NETWORKING FOR DUMMIES

Significance of Audio Networking

Every home, office, internet café, and local community center uses computer networks. We constantly use them to check social media, look at YouTube, use printers, and send emails with sarcastic business rhetoric to our favorite co-workers, but how often do you use networks to transmit real-time sounds? Perhaps never, you think to yourself. You might be wrong. Even an act as simple as using Wi-Fi to make a phone call uses technology called Voice-Over-IP (VoIP) which is common audio networking. Even though this goes on every day, few understand the principles surrounding real time transmission of audio over a network. Browsing news articles and sending those emails, after all, do not require any kind of synchronization. The speed at which you view them is entirely based upon your time. How then do we get time sensitive information from one place to another quick enough and reliably enough to hold conversations... or record a band?

In professional environments there is a need to send quality audio quickly and efficiently whether you are engineering a concert or recording a podcast. For decades this has been done with analog isolated electrical signals traveling down cables. However, for every channel of audio you want to send you must have a separate cable, which quickly builds to copious quantities running every direction for microphones, amplifiers, and speakers. While this works and works well, what if there was a way to not only reduce the number of cables, but also increase the distance signal can be sent, reduce the potential effect of electrical magnetic distortion, and more importantly increase audio routing possibilities all at one time? Networked audio creates such opportunities to improve the efficiency of an engineer's workflow. Audio-over-IP solutions such as Dante, RAVENNA, and AVB make these things a reality in user friendly ways. While making this become a reality may sound complicated, in an Audio Engineering Society panel discussing networked audio, Ethan Wetzel pointed out that analog audio is not that much simpler; we must consider various gain staging signal levels, connector types, balanced and unbalanced signals, etc.[1] Networked audio is perhaps not astronomically more complicated, but rather just new and unfamiliar.

[1]Francis Rumsey, "Rolling out AES67" in Journal of the Audio Engineering Society (Audio Engineering Society, 2017), 148.

Overview of Ethernet and IP in Audio Industry

BRIEF HISTORY OF AOIP LEADING UP TO DANTE

When we think about Dante and networked audio, we typically think of it being a modern solution for the modern world. While this is true, it has a much longer history than one might expect. While it might be argued that long distance telegraph communication (which traces back to 1830s) or even commercial broadcast radio some 90 years later (early 1920s) is not audio networking, it can directly be seen in the more modern sense in 1960's.[2][3][4] In 1966, Max Neuhaus used public telephone networks to distribute a musical performance. While his work was a success, that method still lacked the capability to function in real time or have audio fidelity acceptable for professional use. It took experimentation over the next 20-30 years to accomplish better quality transmissions.

In the 1990s technological breakthroughs were happening across industries, and audio was no exception. Individuals figured out how to use ISDN (telephone networks) and ATM (Asynchronous Transfer Mode) systems to deliver audio over

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[2] "Invention of the Telegraph," The Library of Congress, https://www.loc.gov/collections

[3] "History of Commercial Radio," Federal Communications Commission, www.fcc.gov

[4] Nickolas Bouillot, "Best Practices in Network Audio," in Journal of the Audio Engineering Society (Audio Engineering Society, 2009), 730-739

distances. "True real-time audio networking was first introduced in installed sound reinforcement applications. By the mid-1990s, digital signal processing was in widespread use in this market segment"[5] Counter to the primary modern application, according to Nicolas Bouillot of McGill University, "Live sound applications were the last to adopt digital technology". While successful, these too were limited to one-way transactions. These one-way interactions were not satisfactory for an ever-growing industry craving more flexibility.

Two-way audio networking was finally achieved in 1996 with the Distributed Rehearsal Studio who used something called Asynchronous Transmission Mode (ATM) technology. This marks the beginning of how we utilize networks to transmit audio bi-directionally for concerts, remote recording, and performance alike. As computer networking migrated from ATM technology to Ethernet-based technology (our modern infrastructure) it is interesting to note that early Ethernet protocols were not intended to transport time-sensitive media, but rather for data transmission.[6] The ATM technology at that point in time successfully transmitted an approximate delay of 85ms one-way. Since then, technological advancements in our current network infrastructure can allow us to send and receive audio data as

[5] (Nicolas, 2009)

[6] Francis Rumsey, "Audio Networking! Now We're Talking!" in Journal of the Audio Engineering Society (Audio Engineering Society, 2013), 149

low as theoretical sub-1ms, which allows us to perform and record music in real time synchronously between remote locations. There are still multiple variables that will affect latency which we will discuss more in detail later. Figure 1.0 shows a variety of solutions available as of 2014 along with details on their transport protocols, synchronization methods, and release dates. "The vision of networked musical performance (NMP) has been a holy grail of telecommunications and videoconferencing technologies with respect to its demands on both signal quality and minimized latency."[7] After decades of development, this goal is now a commonplace reality with solutions like Audinate's Dante.

Technology	**Purveyor**	**Date Int.**	**Synchronization**	**Transport**
Ravenna	ALC Networkx	2010	IEEE 1588-2008	RTP
AVB	mIEEE, AVnu	2011	IEEE 1588-2008 advanced profile (IEEE 802.1AS)	Ethernet, RTP
Q-LAN	QSC Audio	2009	IEEE 1588-2002	UDP
N/ACIP	EBU	2007	Data packet arrival times	RTP

[7] (Nicolas, 2009)

Dante	Audinate	2006	IEEE 1588-2002	UDP
Wheatnet-TP	Wheatstone	2005	Proprietary	RTP
LiveWire	Telos/Axis	2004	Proprietary	RTP

Table 1 Technology and Transport Protocols

ALTERNATIVE PROTOCOLS

To understand the full scope of flexibility of Dante, we must also acknowledge that there are alternatives. These are also important to know about due to their place in the market along with potential interoperability between protocols. Two of the other more popular solutions are RAVENNA, and AVB (Audio Video Bridging). Both of these solutions offer similar practical uses as Dante but are achieved by differing methods. Some of these differences will be highlighted as we continue discussing Dante. However, one major difference to note now is that Dante and RAVENNA are both proprietary solutions while AVB is an open-source solution designed around industry standardizations. Various IEEE (Institute of Electronics and Electrical Engineers) standards are present in all these products, but AVB is built entirely upon them.

Whereas Dante and RAVENNA are intended for existing networks, AVB also differs in its approach as it is trying to influence network device manufacturers to be more friendly to real-time audio and visual data transmission. In discussing this on the panel, Will Hoult said, "...AVB is aiming to transform devices to be more friendly to time-sensitive media data... AVB is therefore considered to be more of a network-management and infrastructure facilitator than an audio transport solution."[8] During its development, an organization called AVnu Alliance was formed to certify interoperability between devices working with AVB. Francis Ramsey, staff writer for the Audio Engineering Society journal wrote "AVnu is an independent industry forum that aims to ensure the interoperability of devices running the AVB standard protocols, for professional audio and video, as well as automotive and consumer applications." At that point in time, AES3 was the primary method to interface format conversions, however, they were beginning to use a newly developing standard called AES X192.[9]

AES X192 was created by a team of over 100 companies working together over a three-year period, led by Kevin Gross. Now developed, AES X192 is a protocol available to users known under the name AES67, which is in itself an Audio-over-IP (AoIP) standard. Rumsey described it as, "The AES67 standard for network audio interoperability, launched in September last year (2013), stands as a beacon of cooperation between otherwise competing parties."[10] It comes into the conversation not as a competitor of other solutions, but as an open-source protocol

[8] (Francis, 2017)
[9] (Nicolas, 2009)
[10] (Francis, 2017)

facilitating interoperability. Francis Rumsey also said in an article titled *Rolling Out* that Dante and RAVENNA (among others) are complete AoIP solutions whereas AES67 is not. AES67 is primarily a method for those solutions to communicate across platforms, and is not intended to be an independent operating system.[11] Since its creation, AES67 has become the primary protocol to make solutions like Dante and RAVENNA talk to each other when they otherwise would not. Thus Dante, RAVENNA, and other solutions have adopted AES67 compatibility.[12]

Fundamentals of Networking

Since AoIP inherently utilizes computer network infrastructure, it is also good for users to have at least a basic understanding of how networks operate. To begin, it would be best to understand that in this paper one will find a distinction between three major groupings of networks: local area networks (LAN), wide area networks (WAN), and LAN networks that are more complex than a single subnet. This writing will mostly be focusing on LAN and complex LAN systems.

For the purpose of this essay, simple LAN networks are uncomplicated networks that all live in one subnet, which could

[11] (Francis, 2017)
[12] (Francis, 2013)

consist of one or multiple switches, but no routers or security variations between device areas are required. This is like the type of network you might have in your home, or a small office. It is simply a collection of devices connected via one (or more) switches that allow devices that would not normally talk to each other to communicate. Complex LANs take the next step and introduce multiple subnets via router connections which offer greater expandability and control to IT teams but create difficulties to AoIP protocols. These networks might be found in larger businesses which have multiple departments or buildings that need to communicate with each other via a network connection. Virtual Local Area Networks (VLAN) are also categorized in this, but instead of physically different networks, ports on a managed switch are set up in such a way that multiple LANs are present on a single switch. Wide area networks are the great, wide-open internet that each of us use every day for surfing the web via internet connectivity through a router. The WAN is the only one of these networks that *requires* internet connectivity, the other two can feature it, but can (and often do) operate without it.

Lastly, a virtual private network (VPN) can be created to allow remote access to a LAN. This creates a private end-to-end connection from a computer(s) to the LAN but travels through public internet. Once set up, the network can live in one physical place and the computer can travel abroad but still have direct access to the LAN through any internet connection. Some routers have VPN creation capabilities built into them; others must be created through forwarding ports manually from the router. VPNs are often encrypted for an extra layer of security. Because of its' remote nature and need to utilize public internet infrastructure rather than controlled private equipment, VPNs are not usually as stable of a connection nor as quick.

IP AND MAC ADDRESSING:

IP (Internet Protocol) addresses are given to each device on a network. Various ways in which they are given will be discussed more later, however, for now understanding that LANs often use the exact same set of addresses is important. IP addresses for local networks often look something like 192.168.X.XXX or 10.X.X.X where X can be any number from 0-255. These are reused often between networks, but each network only assigns each number to one device at a time on that specific network. In a 192.168.X.XXX network, the first three fields reveal what network and subnet a device is on. If two devices try to contact each other and the first three fields are identical then the switch will forward the data to the proper device directly. However, if one device has an address of

192.168.0.X and the other device has an address of 192.168.12.X then they live on separate subnets and the router’s gateway must be contacted to transfer the data. Once the gateway is contacted, IP addresses are no longer used, and the router will convert the IP addresses to MAC (Media Access Control) addresses. MAC addresses are device specific identification codes baked into each one’s chipset during manufacturing. Each device receives its’ own MAC address much like United States citizens receive Social Security Numbers. Switches passively take note of what MAC addresses are on a network as signals are sent through its’ subnet using a process called Address Resolution Protocol (ARP). Once IP addresses are converted to MAC addresses, the two devices on separate subnets can then contact each other successfully using MAC addresses through a router.[13] The method of data transfer on a single subnet is often called Ethernet while the process that bridges between subnets is often referred to as IP.

ETHERNET VS IP AND THE OSI MODEL:

An understanding of the differences between Ethernet vs IP and how devices contact each other differently depending on one's subnet is important in understanding how Dante operates in more complicated networks. To aid in understanding, there is a

[13] “Dante,” Dante, Audinate, 2021, audinate.talentlms.com

theoretical model called the OSI (Open Systems Interconnection) model that describes network operations through a 7-tiered system.[14] Each tier represents another layer of purpose, control, and design functions. Figure 2 shows each of the layers alongside the kind of operation they are associated with.[15] As users, one typically only manipulates layers 1-3, however, system designers such as Audinate use all 7. Layers 5-7 deal more with coding and program visual interfacing than transportation of data, thus it will not be discussed in detail.

Layers 1 represents the physical layers such as devices and cabling. Layer 2 represents how those devices know each other at the local area network level with MAC addresses (device serial codes). Layer 3 is how devices on separate subnets or networks are able to communicate via IP addresses. The ability for solutions such as Dante and RAVENNA being able to operate at layer 3 creates great opportunities for expansion and creative uses as they are no longer bound by local area network restrictions, but can bridge subnets and even WANs, unlike some older protocols. Layer 4 will be discussed more further on. "At its most basic level Ethernet describes a physical layer (Layer 1) and a data link layer (Layer 2). Some audio networking systems use no more than these two, using proprietary means to

[14] (Nicolas, 2009)
[15] Kieran Walsh "Dante and AVB Networking" in The Ins and Outs of Audio – AES 24th UK Conference (Audio Engineering Society, 2011), 09-1

deal with everything else. Above this there can be higher layers, such as the network layer (Layer 3), which is often organized using the Internet Protocol (IP) to deliver packets to and from specific addresses, and the transport layer (Layer 4), which is responsible for delivering data to and from specific applications running on the host computer."[16]

7	Data	Application: Network Process to Application
6	Data	Presentation: Data Presentation and Encryption
5	Data	Session: Interhost Communication
4	Segments	Transport: End-to-End Connections and Reliability
3	Packets	Network: Path Determination and IP (Logical Addressing)
2	Frames	Data Link: MAC and LLC (Physical Addressing)
1	Bits	Physical: Media, Signal and Binary Transmission

Table 2 OSI Model[17]

[16] Francis Rumsey, "Networking for the Pros," in Journal of the Audio Engineering Society (Audio Engineering Society, 2009) 274
[17] (Kieran, 2011)

As mentioned, system engineers for general-use networks or for AoIP networks primarily use layers 1-3. To be clear, switches are a layer 1 device, but belong in both layers 1 and 2 since they help facilitate the transmission of data in a local network. Typical switches cannot send signals into subnets that they do not live within. Routers then, are also multi-layered devices being both a physical piece of hardware (layer 1) that transmit data both in a local network (layer 2) and that can transfer signal to other subnets (layer 3). When data moves through a router to another network or subnet it becomes a layer 3 process.[18]

SYSTEM DESIGN TOPOLOGIES (STAR, DAISY-CHAIN, TREE, ETC.)

How devices are physically connected from place to place with cables on a network can be designed in a variety of ways. There is no true one-size-fits-all method for general networks. Here are a few of the common network topologies along with pros and cons of each[19] [20] [21]:

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[18] "Dante", 2021

[19] "E 115: Introduction to Computing Environments," North Carolina State University (NCSU), https://e115.engr.ncsu.edu

[20] Kevin Wilson, "*Exploring Computer Systems : The Illustrated Guide to Understanding Computer Systems, Hardware & Networks*," (Elluminet Press 2019)

[21] (Nicolas, 2009)

i) Bus: A limited series of clients or nodes connected together via a single line. Mostly obsolete today.

a. Pros: Limited equipment is needed for this set up as all devices operate on the same communication line; Inexpensive to set up. Good for temporary set ups.

b. Cons: If the line breaks, all devices become disconnected. Limited physical distance and number of devices allowed. "Problems occur when two devices want to communicate at the same time."

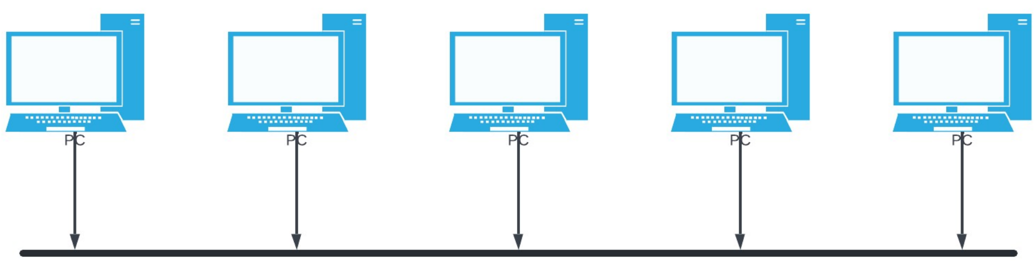

Figure 1: Bus Topology

ii) Daisy-chain: Devices are connected directly from one device to the next in a single direction.

a. Pros: Simple to set up; requires no additional network equipment.

b. Cons: If one device fails, all are likely to disconnect from the network, or two networks will form on either side of the faulty device.

iii) Ring: Devices are connected in a circular fashion where data can travel clockwise or counterclockwise. Similar to daisy-chain, but end devices are connected to form a circle. Like the Bus topology, this is mostly obsolete today.

a. Pros: All devices have equal access to the network, thus limited impact on expandability.

b. Cons: Data can only flow one direction at a time, devices must wait for the data to travel the entire circuit until their connection point in order to receive the data. If one device fails, all others are likely to be disconnected as well.

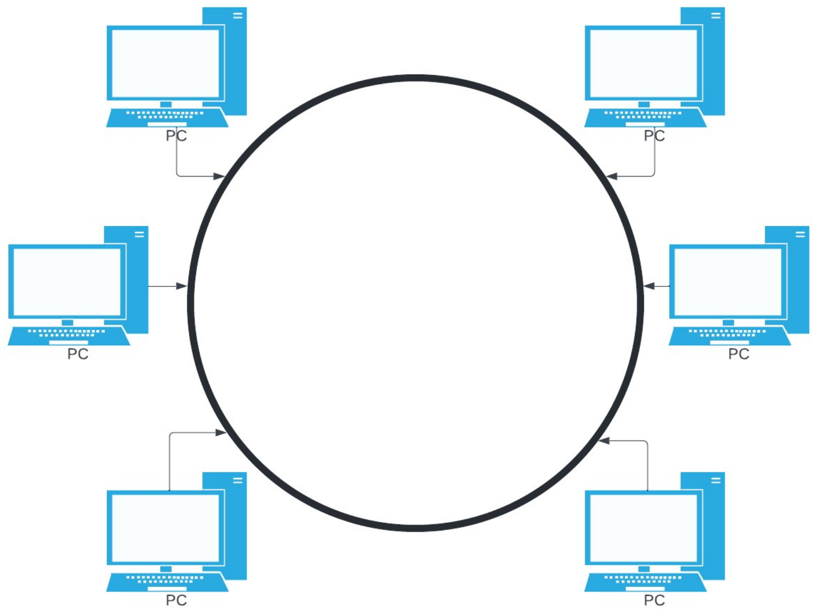

Figure 2: Ring topology

iv) Star: Each device is connected centrally through a central hub which is a network switch. Most modern networks are designed with some form of a star configuration today.

a. Pros: Great expandability, if one device fails it rarely affects other devices on the network.

b. Cons: If the switch fails, all devices become disconnected. Requires more cabling than most other topologies.

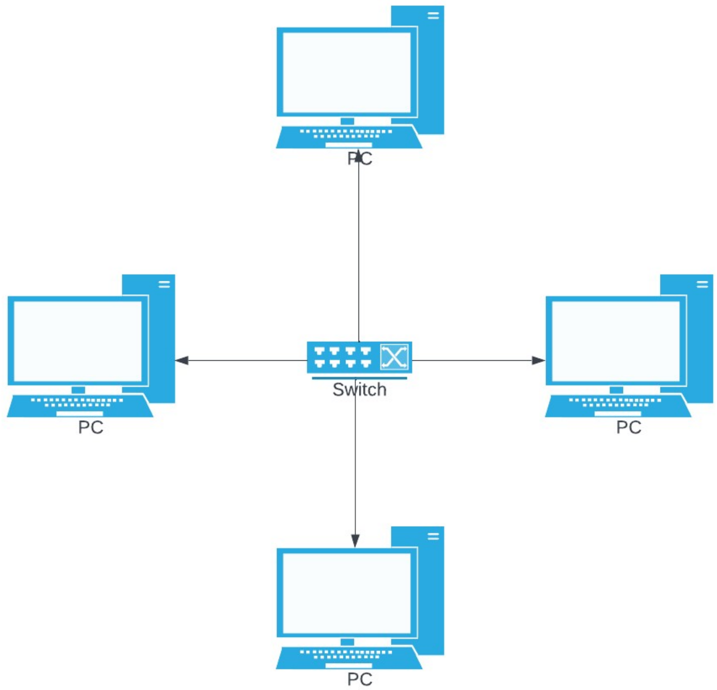

Figure 3: Star Topology

v) Tree: This topology is a collection of star systems connected together, arranged in a hierarchy. As a network grows new switches are connected to the existing switches (concentrators) which begin a new star with added ports.

a. Pros: Great expandability. If an end point device fails, it rarely affects other devices on the network. If concentrator fails, subsequent switches will remain active amongst themselves, though without greater network connectivity.

b. Cons: If a switch fails, all subsequently connected switches and devices will become disconnected from the greater network.

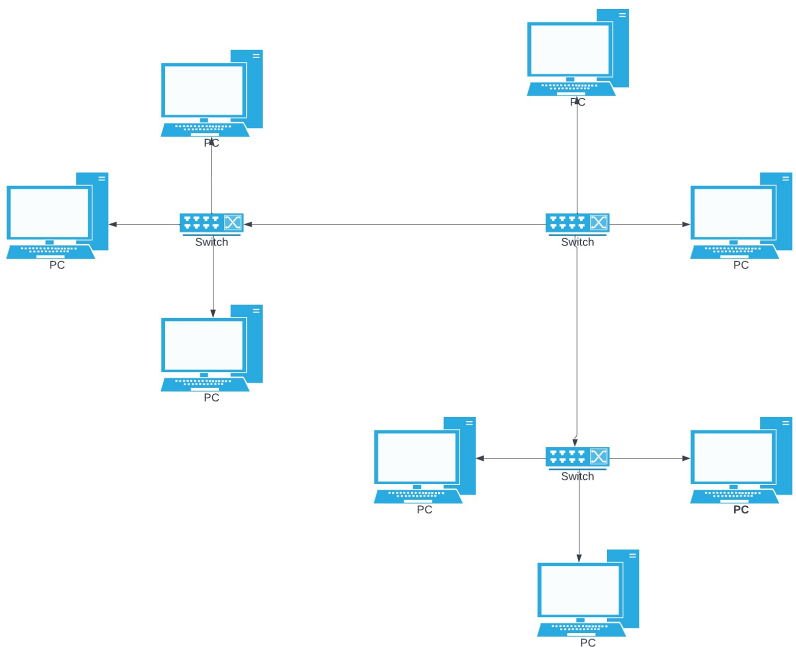

Figure 4: Tree Topology

In audio centric networks, star or tree topologies are preferred. One reason for that choice is that if one device falls off-line all other devices can remain operational. Another reason is that it decreases the potential number of devices a signal must go through to reach its' destination device, which follows the short wire theory.[22] Each device, switch, or

[22] Short Wire Theory: Take the shortest possible signal path from origin to destination, skipping processing and alternate routes in order to maintain simplicity and integrity while also discouraging points of failure from occurring.

connection point a signal must go through will increase latency of the signal which can be damaging to music synchronization. In an AES White Paper titled *Best Practices in Network Audio* the team of authors wrote about star configurations stating, "This provides flexibility in adding, removing, and troubleshooting individual connections, and offers the benefit that, at least for a simple star, traffic between any two devices does not interfere with others."[23] Interestingly, in the same paper it is mentioned that synchronization times can be increased by switching to a bus configuration, however, it is at the severe cost of flexibility and reliability.

UNICAST, MULTICAST, AND BROADCAST

There are three ways in which network devices send out signals to other devices on a network: unicast, multicast, and broadcast. In Dante, unicast and multicast are used most, but broadcast also plays a vital role. While Dante's specific uses of these types of signals will be discussed, first a foundation of their operating principles should be recognized.

Unicast is the simplest of the three. It is when one device sends signal directly to one other device (one-to-one). Communication is one way and is not transmitted to any other devices than the one intended. Broadcast is also simple but is

[23] (Nicolas, 2009)

the polar opposite of unicast. Instead of one signal going to *one* device, it is one signal going to *all* devices (one-to-all). Multicast then, is the outlier and lesser-known stepsibling of data transmission. It is a single signal going to a many, but limited number of devices (one-to-x). Multicast does not increase the number of signals being transmitted from the device itself, but rather it is multiplied at the switch. This means it increases bandwidth used on the system but decreases strain on the origin device allowing for lower sending latency.

Unicast is similar to a hard drive connected directly to a computer and stored data is transferred in one direction. Multicast is more like sending an office email and CC'ing multiple people on the thread. Broadcast is used exactly like broadcast tv (hence the name) where a television station sends out a single signal which can be intercepted by anyone's antenna to view. In common networks we might use unicast to connect various devices together at home or we might use broadcast to access public internet websites. However, multicast is not something we typically use.

One important factor about these methods is that all three are usable in a local network, but only unicast is able to transverse routers into other subnets. This becomes vitally important in AoIP when we discuss discovery, audio routing, and clocking.[24] Multicast and Broadcast signals are not allowed through a router so that endless loops of signals are not continuously being relayed to every other internet device in the world, causing mass bandwidth bottlenecking.

[24] "Dante", 2021

SECTION 2: CHALLENGES OF AUDIO NETWORKING

As we step out of general use networks and become focused on synchronous audio networks there are a handful of challenges that become prominent. Discovery of devices, bandwidth usage, clocking for synchronization, and latency or speed of transmission between endpoints.

Discovery

Discovery is the term used to describe how network devices know that there are other devices on a network and what those devices are. Without discovery there would be no communication, or communication would be limited to broadcast only signals. So how does discovery work?

In a single LAN network, discovery makes use of multicast signals. This is one way in which AoIP networks use multicast more actively than general use networks. According to Audinate's guide for IT Manager, "Multicast is necessary as a part of Dante and all other AV-over-IP solutions, as it is used to send clock sync, timing, and discovery messages to all devices at once within broadcast domains."[25]

[25] "Implementing Dante AV-over-IP from an IT Manager's Perspective," Audinate (Audinate, 2022)

When working with a larger, multi-subnet network and Dante Domain Manager, multicast discovery is replaced with unicast. This is due to multicast signals not being able to travel through routers to other subnets. Thus, instead of a multicast signal traveling directly to a device, if the device is outside the LAN the router/gateway is contacted to gain access. This happens in the network endpoint that is initiating the connection.[26] Remember, when only in simple LAN networks, IP addresses are used but once that router is contacted MAC addresses are used instead, using ARP as discussed before. Using these types of signals Dante Controller automatically discovers new devices on a network.

There are multiple protocols on how multicast signals are used to initiate discovery. With Dante, mDNS and DNS-SD are the two automatic methods. In single subnet architecture, using automatic discovery with mDNS works. That does not work on networks with more than one subnet, however. DNS-SD replaces mDNS as the discovery mechanism when in larger networks.[27] DNS—SD can be used in single and multi-subnet environments.[28] In cases where automatic discovery may not be functioning, non-automatic discovery can be used within Dante Domain Manager software to enroll everything. If a device is not automatically discovered,

[26] "Dante", 2021
[27] (Audinate, 2022)
[28] "Dante", 2021

you can add a device to a domain by manually entering the IP address.[29]

When working with a dedicated AoIP network Dante will assign addresses to devices using either DHCP (Dynamic Host Configuration Protocol), Link-Local, or Static Addressing. Static Addressing is manual assignment of IP addresses given to devices by the user. DHCP is a protocol emitted by a router that automatically assigns IP addresses to devices for a limited amount of time. Once that time has elapsed the IP address will be revoked and become available for a new device. This is commonly used for public Wi-Fi networks so that the same IP addresses can be reused endlessly as new patrons log-on to surf the web. Link-local is the automatic addressing protocol Dante uses when there is no router on the network.

Other AoIP systems may use other protocols, or none at all. With AES67, discovery protocols were intentionally left out of the standard so that it could play nicely with other AoIP solutions. "In AES67, there is no connection management on top of the multicast mode. In essence, audio streams are broadcast on the network and other devices pick them off as required. This is very simple to implement. Internet Group Management Protocol (IGMP) is usually implemented in a network switch, and it simply

[29] "Dante", 2021

requires that devices advertise themselves as potential sources, while receivers subscribe to a particular multicast stream.”[30]

Bandwidth

Available bandwidth in a network can also pose a problem to using AoIP solutions. This one is a much simpler and more hands-off solution but can require more finance. Bandwidth is the total data transmission capacity inside of a network. Just like we describe water pipes being able to distribute X number of gallons per minute, bandwidth is measured in megabits or gigabits per minute. Thankfully, most modern networks have decent enough bandwidth capabilities to run Dante without much change, but if there are any switches that throttle the signals those should be replaced with ones of a higher capacity.

The general recommendations by Audinate for Dante are to have 1Gb/s capacity switches in most places, but how do we know how much we need? We must calculate the total amount of data we will need to send on our network and leave extra space for headroom. Our variables in this calculation are sample rate, bit depth, compressed vs uncompressed signals, unicast vs multicast signals, and audio vs video signals.

30 (Francis, 2014)

In all forms of digital audio, not just networked audio, an engineer must choose the sample rate and bit depth when the signal is converted from the analog to the digital domain. These choices affect the accuracy of the digital representation of the recording along with the amount of data used to retain it. Digital audio works much like video; it takes a series of "pictures" extremely quickly which when played back sounds like a continuously moving sound. These momentary snapshots are called samples; thus, the sample rate is the number of impulse snapshots per second. The higher the sampling rate or the more samples are taken per second, the more data it creates. Common sampling rates are 44.1kHz, 48kHz, 96kHz, and 192kHz. As we have been describing it, 48kHz would be 48,000 snapshots per second. Most AoIP solutions make their data calculations based off of 48kHz as it is the standard sampling rate when audio may need to be used in sync with video.

Bit depth also plays a leading role in the data size of an audio signal. It works closely with sampling rate to convert analog audio into digital audio. Whereas our samples are snapshots in time of an analog signal, bit depth is the quantified "level" at which the analog audio is sampled. As computers operate using concrete numbers (usually some form of binary 0s and 1s), bit depth quantization assigns a fixed number to each amplitude step during conversion. Bit depth then is the digital representation and capacity of the dynamic range of an analog audio signal. Common bit depths for audio are 16bit, 24bit, and 32bit. 24bit is the most commonly found bit depth in networked audio and has a dynamic range of 144dB which is more than enough for most professional applications. One's bit depth and sample rate are the two of the biggest factors in the data size of an audio signal being streamed from one place to another.

Another factor that plays into data size is whether one uses a codec and compresses the audio signal or sends full fidelity, uncompressed audio. For professional use, most engineers will want nothing less than uncompressed and untouched linear Pulse Code Modulation (PCM) audio. This is audio in its' most raw state from the pre-amplifier and is the highest fidelity that signal will ever be in the digital world. However, using uncompressed audio will utilize the most bandwidth. Other options include using codecs to encode PCM audio into .MP3, .AAC, one of the Voice-over-IP (VoIP) codecs, or one of the Opus codecs.[31] These codecs will decrease the data size, but at the cost of signal degradation that cannot truly be recovered. Some AoIP solutions make use of alternate codecs like AVB which uses .AAF with 32bit 192kHz signals.[32] Encoding the audio will also

[31] (Francis, 2013)
[32] Francis Rumsey, "Making Audio Networking Easier," in the Journal of the Audio Engineering Society (Audio Engineering Society, 2019)

cause the time it takes for a signal to go from one location on the network to another as it adds the step of encoding to the signal flow. Thankfully, most modern networks have the capability to send full PCM audio without issue if there was mindful planning of what the network infrastructure was built with. Using 24bit 48kHz linear PCM a 1Gb/s network can send at least 256 individual unicast audio channels without issue, and even more if sent via multicast.

To add to our discussion of unicast vs multicast, one of the primary reasons we Dante users would want to use multicast is to reduce bandwidth being utilized in streams that need to go to many different destinations. A single unicast stream (or flow) at 24bit 48kHz uses approximately 6Mb/s and carries up to four channels of audio.[33] Multicast by contrast carries approximately 1.5Mb/s per audio channel and can carry up to 8 audio channels per flow. Thus, for signals that need to only go to a few destinations, unicast has an obvious advantage, but when large scale distribution of the same audio is needed multicast takes the edge. For example, in a live concert if you were to send a stage box of 16 channels of audio to three destinations such as a front of house, monitor, and broadcast console it would require 12 unicast flows ([# of channels/4] x

[33] "Dante", 2021

[number of destinations]). This would use approximately 72Mb/s. In comparison, the same number of channels would require only 2 multicast flows. Utilizing two multicast flows could stream to all three consoles (or more), and only be using approximately 24Mb/s. It is important to understand, however, that multicast flows often transmit with more latency than unicast flows.

Lastly, if video is a necessary implementation on a network, bandwidth can become more of an issue. Video signals require much more data than audio does and have several additional variables. The resolution, color chroma pattern, frame rate, and video bit depth all play roles in the bandwidth required. Because video streams are much larger, Dante video is only sent via unicast and cannot be sent via multicast. Below is a chart created by Audinate that shows some of the bandwidth requirements for different video stream settings.

Resolution	1080p	1080p	1080p	1080p	4K	4K	4K
Chroma	4:2:2	4:4:4	4:2:2	4:4:4	4:2:2	4:4:4	4:2:2
Frame Rate	30	30	60	60	30	30	60
Bit Depth	10	10	10	10	10	10	10
Bandwidth	85Mbps	126Mbps	135Mbps	253Mbps	337Mbps	506Mbps	675Mbps

Table 3 Dante Video Settings and Requirements[34]

[34] (Audinate, 2022)

So how much bandwidth does a network need? The answer still may be: "it depends." However, it should now be clear why recommendations start at 1Gb/s. While you can operate smaller systems with 100Mb/s (1/10th the size), it is not recommended as it will limit the system to a much smaller number of channels available for use at any given time. In an AES convention tutorial, Patrick Killaney stated, "When designing a network for audio it's highly advantageous to be using 1-Gbit/s network switches as opposed to 100 Mbit/s, as not only will the network speed be faster, but the timing accuracy of sync packets can be much better."[35]

1Gb/s systems will allow for most moderate sized networks to operate without issues. However, when working with bigger network needs the ability to scale up might be as simple as purchasing larger capacity switches. Killaney later said, "The network backbone needs to be capable of twice the speed of the switches combined... as data needs to flow in both directions and multiplied by the number of switches to be served by the backbone. So, 20 one-gigabit switches need a 40-gigabit backbone to create a nonblocking architecture where there are no bottlenecks." As with all computer-based technical needs, it is a wise idea to ensure that there is enough bandwidth for

[35] (Francis, 2019), 146

overhead processing in the system. Experts will provide differing answers as to when one should begin looking at increasing their network's capabilities, but some recommend shopping around at as low as 50% capacity being used.[36]

Clocking and Synchronization

"There is only one clock that matters... and that is real time," Greg Shay of Telos Alliance and Axia.[37] For audio, clocking is just as important as a drummer's kick drum is for a band. Everyone must keep in sync with each other or what should be music sounds like mayhem. Clocking to the professional audio world is no new issue, but with the introduction of networks as a new transportation method, new shared clocking methods must also be created.

Most of the world had little need in the past for real-time networks apart from telephone communication. Only recently, due to the rise of remote jobs and video conferencing becoming a much greater presence, did the general public begin to have need of such network capabilities. However, most general public internet and networks needs still do not require real-time synchronization. Kiaren Walsh from Audinate summed this up by saying "Much of the information sent over Local area networks

[36] (Francis, 2017)
[37] (Francis, 2013)

and inter-networks is considered “best-effort”. For example, sending an email has no specific time-critical requirement. The importance with a service like email is that the mail is delivered accurately. The speed of modern data networks means that email is delivered rapidly across the globe. There is no specific guarantee of the precise time that it will take for delivery. Audio networks in particular need to work in real time in a live environment, and therefore present a different set of challenges.” [38]

If two end points of a network need to work together in real time, how do they do it? Is it simply magic, and they just work? Unfortunately, no. Signals must be sent through the network alongside the audio signals to maintain synchronization. Walsh stated that in older days, IP networks would use super high frequency audio signals to act as a sync clock.[39] The problem with this method, though, is that it took up a lot of valuable bandwidth and channels that should be used for the actual audio data payload. As of 2008, the Institute of Electronics and Electrical Engineers (IEEE) introduced a new standard for clocking called Precision Time Protocol (PTP). Also known as IEEE 1588-2008, PTP (version 1) was not initially

[38] (Kieran, 2011)
[39] (Kieran, 2011)

developed for audio networks but for industrial automation.[40] Turns out, though, that it works well for real-time audio needs. As of 2023, PTP version 2 (PTPv2) has been adopted as the current standard for AoIP, though many devices still operate using PTPv1 or are at least legacy compatible.

Networks elect a device ’leader’ for clocking which all subsequent ’follower’ devices derive their clock from. PTPv2 generates and distributes from the leader ”absolute time data to devices on the network.”[41] This ”absolute time data” is not to be confused with an actual clock signal, though. Individual devices must take this absolute time data, convert it, and generate internally a usable clock signal that they keep synchronization with. As described in Audinate’s writings for IT Managers,”(PTP) generates a few small packets, a few times per second… (the leader) sends multicast sync and follow up messages to all follower devices. Follower devices send delay requests back to the leader to determine network delay. A Dante device with a 1 Gbps connection acting as clock leader can keep about 250-300 Dante devices in sync.”[42] Due to PTP using multicast, this solution only works for a single subnet. In Dante, for larger networks using Dante Domain Manager, an additional unicast PTP

[40] (Francis, 2013)
[41] (Francis, 2014)
[42] (Audinate, 2022), 3

signal will be sent to the router which can then forward the unicast signal to the next subnet where a ’boundary clock’ leader will be waiting to receive and redistribute the data as multicast to the devices in its’ LAN.[43]

Does physical distance between devices affect clocking? What if over a long distance the clocking data begins to shift and lose its’ precision? What if there is a desire to send audio over many miles to a new location in a different time zone? Over any kind of substantial distance, network signals will fluctuate and pieces of information transmitted in tandem will arrive at their destination at slightly different speeds. This is a common problem and is called network Jitter. It is defined in Dante’s Level III certification program as “… the variance in time it takes for signals to transmit over a network and how clocking is affected by this uncertain shift."[44] How then can one create an absolute clock that each device, no matter the physical location, can follow? The answer is in the skies.

Global Positioning Systems, better known as GPS, are a series of satellites that orbit the earth perfectly coordinated with each other which can be used to generate extremely accurate clocks on earth. GPS clocking (and positioning) is derived from

[43] Paul Ferguson, “Trans-Europe Express Audio,” in 148th Audio Engineering Society Convention (Audio Engineering Society, 2020)
[44] “Dante”, 2021

a signal beaming up into space where it reflects off multiple orbiting satellites. The time offset from transmission to reencounter is referenced against the same signal's return from alternate satellites and accurate timing and positioning can be calculated. On one's audio network, if a module with GPS clocking capabilities is installed on both ends of the long-distance network, then both locations can maintain precise sync with each other. Both ends of the network thus have physically different clock sources (the local GPS module), but both sources are generating their absolute time from the same absolute source above.[45] Dante users can utilize this through combining GPS hardware modules and Dante Domain Manager software.

Latency

What can be more important to real-time communication than the speed of time itself? Latency in AoIP networks is the time it takes for a signal to originate at one point of the network and completely travel to its' destination, round trip or one-way to another network endpoint. This could be anything from music streaming from a computer in an IT closet being played out in a corporate lobby, musicians hearing themselves in headphones as they track a song, or a broadcast truck feeding a mix of a

[45] (Paul, 2020)

football game to a TV station 700 miles away. Striving for a low latency value is key in almost every network, but especially if two ends of a network must work together in real-time. Beyond the simple annoyance of having to wait for signals, latent audio causes legitimate problems when trying to play music, record, or speak when needing to listen to the latent signal. Greg Shay of Telos Alliance also spoke to this point regarding radio saying, "The first thing people care about in radio... is low latency. Any delay that gets in the way of the signal's transfer is a problem... In a typical broadcast situation when listening to one's own voice over headphones, anything more than a few milliseconds delay leads to a very unpleasant sensation that feels like having your ears sucked out."[46]

As with other things we have discussed, latency is not an isolated issue with networked audio, but with new technologies come new problems and solutions. What causes latency in networks? The biggest offenders are: packetization time, transmit time, switch hops, and payout buffer.[47] Packetization time is the time it takes for a signal to be converted from its digital state to many small packages that are compatible with network technology. Transmit time is exactly what it sounds like; the time it takes for those packages to travel from one

[46] (Francis, 2009), 273
[47] (Kieran, 2011)

point on the network to another. During a package’s travels across a network, every switch or piece of hardware it encounters will have to process and forward the data, called switch hops which take additional time. Payout buffer is the amount of time a signal is allotted once it reaches its’ destination before it is processed by the endpoint. It is necessary to allow all data packets to be fully transmitted before reassembly and playback.

When it comes to acceptable latency times, there is at least one scientific principle that must be observed first: the Haas effect. The Haas Effect (Law of First Wavefront or the Precedence Effect) is a psychoacoustical phenomenon that describes how humans identify sounds as being independent incidents of creation versus being separate sources. If two identical (or near identical) sounds are created under a certain time frame of each other, we perceive them as being the same sound. However, if they occur outside of that timeframe, we perceive them as being from different origins. The specific threshold time is debated among the audio industry; however, it is generally considered to be somewhere between 20 and 40 milliseconds. This is extremely important when it comes to producing musical material in real-time as anything above that threshold can cause not only “ear sucking out” sensations, but also blatant timing misrepresentations that lead to inaccurate performances. Thus, for audio networks that intend to have multiple sources input simultaneously (e.g., musicians in multiple locations playing at the same time) they *must* be under the threshold for quality outcomes. However, for one-way transmission of audio through a network this is not as necessary to observe.

Acceptable latency times have changed greatly over the last 15 years. Telephone VoIP is considered acceptable at 100-150ms, but that simply does not work for professional audio. Professional audio systems also must work with analog to digital and digital to analog conversion which itself can take 15-20ms.[48] This leaves little room for networks to operate within. Because substantial amounts of data take more time to packetize, early innovators attempted to use MP3 or AAC codecs to reduce the data thus also reducing packetization time. This was unsuccessful, though, due to it adding back or even increasing the overall time it reduced because of its' encoding stage. There were a few codecs that promised some hope, namely AAC's Enhanced Low Delay (ELD) which reduced encoding time to 5ms over the previous 20ms, however, packetization time still added 10ms. Later, Fraunhofer's Ultra Low Delay Audio codec (ULD) achieved an inherent encoding delay of 2.6ms with only another 2.6ms

[48] (Nicolas, 2009)

required for audio packetization. It brought latency down, but at the cost of audio quality which was said to be comparable to MP3 in 2009.[49] Even just five years later, in 2014, network infrastructure technology had improved enough that AES67 required support of packets lasting 1ms and allowed for packets that were as low as 1/8ms for uncompressed PCM audio.[50]

While that makes latency seem an issue of the past, the truth is far from it. Though we have network equipment that can handle low latencies, system designers must appropriately organize system structure in such a way to facilitate it. This includes reducing the number of switches signals must travel through to reach their destination, prioritizing audio transmission on a network by using tools like Quality of Service, reducing the data strain on individual pieces of hardware that are transmitting signals by unicast versus multicast options, ensuring the network has enough available bandwidth for all transmission needs, etc. All these things must be considered carefully when designing a network, especially if it is a shared network with non-audio traffic.

With the rise in using networks for audio transmission also comes the desire for remote, long-distance operation. Regarding latency we must acknowledge the current limitations that come

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[49] (Nicolas, 2009)
[50] (Francis, 2014)

with this process. Optical fiber cables, which use light to transmit data, are the current standard high speed inter-networking connections. Due to the scientific principle which they are built upon, they transmit at the speed of light which is approximately 168,000 miles per second, or 168 miles per millisecond.[51] This means that theoretically to keep our latency below the Haas Effect threshold one's network endpoints must be within 3,360 miles, or 1,680 miles round trip. That is comparable to a direct flight from New York, NY to Austin, TX. However, it is extremely unlikely for one to have a single direct fiber cable connecting that kind of distance. Reality must also factor in how many junctions (switch hops) an internet provider might make your signal pass through, which is not readily available information. In 2020, there was a collaboration between Edinburgh Napier University (Scotland), Standford University (USA), and Technische Universität Berlin (Germany) to test long distance audio transmission using both Dante and Jacktrip (a free, open-source solution).[52] The "Trans-Europe Express" sent audio bi-directionally from Edinburgh to Berlin, approximately 700 miles by flight. In this real-world application they found that their ping between locations averaged 40ms with Jacktrip. Using Dante, they were able to

[51] "How 'Fast' is the Speed of Light" Glenn Learning Technologies Project (LTP) (NASA, 2021) https://www.grc.nasa.gov

[52] (Paul, 2020)

reach a peak latency of 22.7ms between trans-Atlantic locations. To accomplish this connection, they used cheap $80 GPS clocks soldered to a Behringer interface to keep sync, and online software to discover 16 different switch hops between locations. That latency time could potentially be shortened if direct fiber lines were purchased, and audio did not have to be transmitted through open networks provided by their internet provider. In either case, this demonstrates that our modern infrastructure is capable of impressive feats, but there are still physical limitations that must be considered.

SECTION 3: DANTE

Understanding Dante Technology

With an understanding of networks and how they operate, attention can now be focused more directly on how Dante handles these principles. First, Dante considers itself an AoIP solution rather than simply a protocol because it incorporates more than just computer coding to achieve its goal. Dante features a mixture of both software and hardware components that allow for transmission of both audio and video over networks. This includes products like microphones, interfaces, remote controlled pre-amps, loud-speakers, digital signal processors (DSPs), power amplifiers, video encoders/decoders, and software for control and interfacing.

Key Software Components of Dante

Audinate has created six separate programs that allow for various manipulation and application of signals on the network: Dante Controller, Dante Virtual Soundcard (DVS), Dante Domain Manager (DDM), Dante Studio, and Dante Video Viewer (DVV). Dante Studio and Dante Video Viewer deal specifically with Dante's video suite and will be discussed in more detail further on.

Dante Controller is the primary application that most audio engineers and daily system operators will use. Its' purpose is to route signals from one place on the network to another in a

LAN. It uses "label-based routing of signals to make configuration easy for users."[53] This means that it uses user assigned labels for each network device to direct data traffic. It is important to note that audio *does not* flow through Dante Controller, it merely is controlled by it. When needed, Dante Controller also has the ability to send out firmware updates to all network devices at one time. Beyond simple routing, Dante Controller works as the origin point for device discovery, operators can assign names to channels, create unicast and multicast flows, and even see real-time system analytics.

Dante Virtual Soundcard (DVS) is a purchasable program that turns your computer into a routable network device. This allows users to originate a signal on their computer and feed it to other devices across the network, like for corporate lobby music. It also allows signals to be routed *TO* a computer. DVS allows up to 64 channels of audio to flow to/from the host computer, making it an alternative connection type for multitracking a band for example. This adds an easy way for engineers to have a redundant recording device capture musicians' performance with ease by simply routing the source signals to two (or more) separate recording computers.

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[53] (Francis, 2009), 273

Dante Via is another purchasable software from Audinate that creates easy routing within one's computer to other places within the same computer. It allows the computer to loop up to 16 bi-directional channels of audio directly from one program to another. It also allows one to connect their existing audio interface into a Dante network cheaply and easily with no need for proprietary hardware.[54]

Dante Domain Manager (DDM) is the big dog among a small pack in Audinate's software suite. DDM is necessary for Dante to work in a level-3 (as described by the OSI model) environment. It expands the routing capabilities of audio signals to more than a single LAN. It is a permanent fixture on a network and is crucial to large Dante systems as it becomes the "server-based centralized monitoring and management application for Dante Networks that automatically configures Dante devices to use unicast traffic to cross routers."[55] [56] It also allows for the creation of custom domains and users that can restrict public access of control features in Dante Controller.[57]

With custom domains, one can group devices together in logical collections that need to consistently interact with each other but may not need to interact with other devices on the

[54] "Audinate - Dante Via," Audinate (Audinate), https://www.audinate.com
[55] (Audinate, 2022)
[56] "Dante", 2021
[57] (Paul, 2020)

network as often. Unlike a single LAN network, these domains do not have to clock together, however, you can collect devices on different subnets to be within the same domain (this is how the Trans-Europe Express was achieved). Other uses of domains might be found at a university where a systems manager might make independent lecture halls their own domain as those devices need to share info, but external classrooms may not need to interact with the halls.

Custom user profiles allow for passwords and security of who can access what devices on the network where you might have managers, engineers, and day workers all needing various levels of routing access. It is also important to note that like Dante Controller, audio data *does not* flow through DDM, it merely aids in controlling network traffic. Other things of importance about DDM include that it takes over the primary role of network device discovery and that it can send notifications of system activity, errors, and alert messages.[58]

Key Hardware Components of Dante

Audinate has a plethora of hardware physical devices that are licensed to work with Dante. Some devices are designed to work solely as Dante interfaces with existing equipment (such as

[58] "Dante", 2021

encoders/decoders or the AVIO adapter series), while for other equipment Dante compatibility is more of an additional feature in an already larger product (such as Yamaha mixing consoles). These hardware solutions require some form of physical chipset available from Audinate like the Brooklyn card or the Ultimo series.[59] For some equipment, after market replacement cards are available to retrofit popular pieces of gear to be Dante compatible, one of the most common being the replacement I/O card for the Behringer X32 mixing console.

Cables

Perhaps the most important aspect of the hardware side of Dante is the cabling that weaves together a network. Without the cable no system would be able to communicate. The Layer 2 portion of Dante uses cables which are commonly refer to as “network cables”, “Ethernet cables”, or “Cat5” cables but what are they truly? There is a style of cabling called the Category (or Cat) “X” cables that connect computers together via 8 strands of copper. In this, “X” is a number that denotes the design generation that cable belongs to. Cat5 has become a standard name for all these cables even though Cat5 is being

[59] “Solutions for Hardware Developers,” Audinate (Audinate), https://www.audinate.com

phased out for newer generations. For Dante, Cat5e is the oldest recommended generation, but Cat6 is preferred, and all of these utilize RJ45 connectors. Cat5's maximum transmission rate is 100Mb/s whereas Cat5e improves to 1Gb/s. Cat6 also operates at 1Gb/s, but it is preferred due to its improved durability. Cat5e is susceptible to damage that can limit its' transmission rate back down to 100mb/s. A newer generation than Cat6 can be chosen for a network that may increase the maximum potential data transmission capacity up to 40Gb/s with Cat8, if needed. It is also worth noting that for long distances (<1500ft), optical cables can be chosen, however, conversion devices will then be required.

Another set of choices system designers must face when selecting cables is whether to use solid core or stranded core cables and whether cable shielding will be necessary. Popular equipment manufacturer, Yamaha, broke down this question and explained that the answer is primarily in whether the use will be for installation or mobile applications.[60] Solid core cables with (or even without) shielding are less flexible, but signal continuously remains on a single cable, thus are great for installations. Stranded cables have some advantages when being handled more often such as that they are considerably more

[60] "Selecting Network Devices," Yamaha (Yamaha), usa.yamaha.com

flexible, less susceptible to damage from being bent, and if one strand breaks it is still maintains connectivity due to the presence of other touching strands. Shielding adds a layer of thin metal between the eight core wires and the exterior plastic or rubber jacket. This shielding helps provide protection against electromagnetic interference. Metal Ethercon connector shields can also be added for additional defense of the terminals if the intended device it will be paired with is compatible.

	Solid Core Cable	**Stranded Cable**
Characteristics	Excellent for long distance transmission	Flexible and easy to handle
Suited for	Installations	Touring and short patch links

Table 4 Solid Core vs Stranded Core Cables[61]

61 ("Selecting," Yamaha)

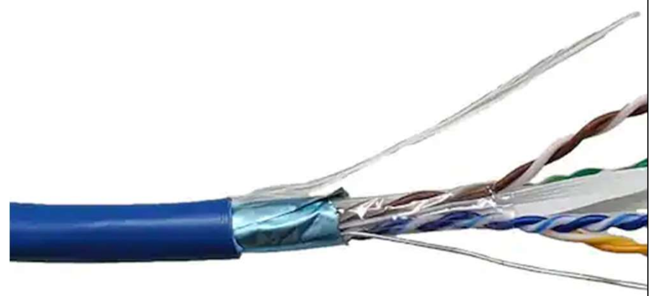
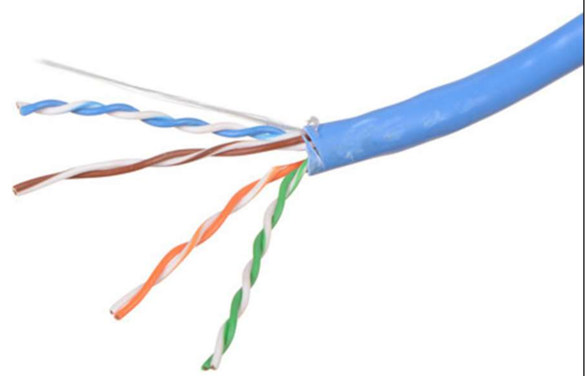
Shielded Twisted Pair (STP)	**Unshielded Twisted Pair (UTP)**
	

Figure 5: STP vs UTP Cables

Network Switch

Network switches also play a vital role in the success of a AoIP network. As with everything else discussed thus far, there are a variety of options and considerations that need to be made to choose the right switch for your network. Some of these include the data transmission capacity, managed vs unmanaged, number of ports, and Power Over Ethernet capabilities.

As mentioned previously, for Dante a minimum of 1Gb/s is recommended. Patrick Killaney said, "When designing a network for audio it's highly advantageous to be using 1-Gbit/s network switches as opposed to 100 Mbit/s, as not only will the network speed be faster, but the timing accuracy of sync packets can be much better."[62] 1Gb/s switches are great starting places, however, if designing a much larger network there is another

[62] (Francis, 2019)

rule of thumb that Killaney recommended following. “The network backbone needs to be capable of twice the speed of the switches combined... as data needs to flow in both directions and multiplied by the number of switches to be served by the backbone. So, 20 one-gigabit switches need a 40-gigabit backbone to create a nonblocking architecture where there are no bottlenecks.”[63]

For Dante it is also highly recommended to use managed switches, especially for important intersections on the network. Unmanaged switches are designed to be plug-n-play with anything connected to them, however, they usually come with settings (such as power saving modes) that cause issues with Dante. Managed switches allow you to customize how traffic travels through your switch, which can be especially important.

1. EEE

One of the biggest enemies to a Dante network is Energy Efficient Ethernet or EEE (IEEE 802.3az). It will automatically ‘turn off’ data transmission if it does not detect activity through a port. While this is great for office style work where computers do not need constant access, for Dante equipment this can cause significant issues, especially with clocking and sync,

[63] (Francis, 2019)

and can even lead to audio drop out.[64] When choosing a switch (managed or unmanaged) one *must* ensure that EEE can be disabled.

2. IGMP

Internet Group Management Protocol (IGMP) is designed to keep network traffic from becoming unnecessarily oversaturated by only forwarding multicast information to ports that are intending to receive it. Without it, multicast traffic will flood to every destination like a broadcast signal causing an overloading of the network. IGMP snooping is only required on Dante only networks when using 100Mb/s or mixed 1Gbps/100Mbps networks, though for large networks it is recommended. Dante needs multicast traffic to operate properly, thus one switch on a network should be set to be the IGMP Quarrier (leader) while all the others follow, allowing it to snuff out all the excess traffic. If setting custom Query settings, Query intervals should be short, and time out values long for best results.[65]

3. QoS and DSCP

Quality of Service (QoS) is a method to categorize and prioritize traffic on a network. QoS is important to engage and optimize when Dante is being used on a converged (Dante and non-Dante) network. In fact, Francis Rumsey wrote, "Quality of

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[64] ("Selecting," Yamaha)
[65] "Dante", 2021

Service support is especially required if other services are running on the same infrastructure. Without this it is possible for the timing of PTP (precision time protocol = IEEE 1588) packets to be affected… which can mess up the synchronization of devices on the system."[66] QoS creates a decimal scale with Differentiated Services Code Point (DSCP) tags, four-tiers of importance, and 64 levels of priority for packets.[67] Each packet is designated a DSCP label and decimal number that fits within those tiers which prioritize which packets get forwarded by a switch first upon arrival. Packets such as clocking should be of the highest priority for network stability. For Dante-only networks it is not as important to engage this feature, however, as a network grows it should become a consideration, especially once video is involved.[68]

If signals come through with no DSCP tags present, then QoS remains passive and does not affect packets' travel.[69] If two packets arrive at the switch at the same time, but one is tagged with some sort of priority level and the other comes in as "best effort" (meaning of lowest priority) then the packet with a DSCP tag will take priority. If the untagged packet arrives first and the tagged packet arrives second, then the switch will finish

[66] (Francis, 2017)
[67] (Kieran, 2011)
[68] "Dante", 2021
[69] (Kieran, 2011)

its initial task. “On a one gigabit network switch this will only take 13 microseconds, if the lower priority packet is completely full of data.”

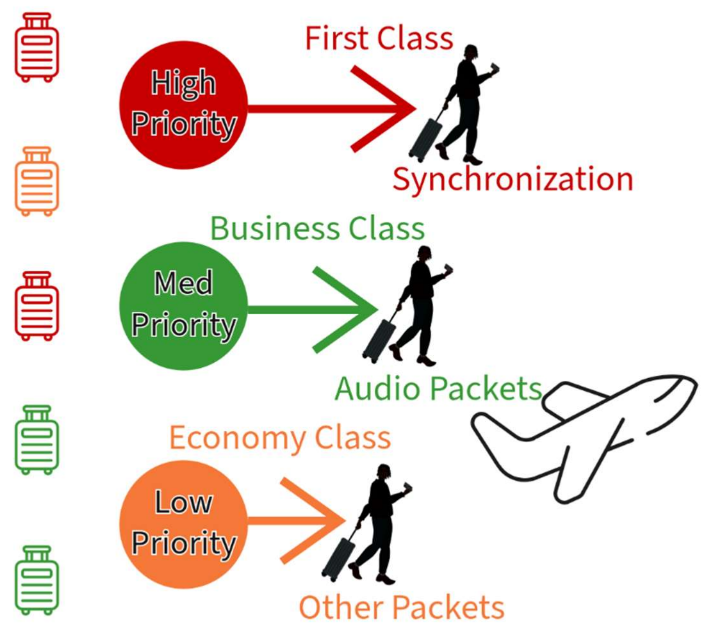

Figure 6: QoS HIERARCHY

4. PoE

Power over Ethernet (PoE) is an important capability that many network switches have. It allows the switch to power devices directly through the network cabling rather than needing external power supplies. This is primarily for smaller devices such as Pan-Tilt-Zoom (PTZ) cameras, headphone boxes, telephones, etc., however, in special situations larger devices like TVs can be powered through special PoE set ups. When choosing a switch, one must be aware of how much power is needed for the gear being used along with what class of PoE the switch is equipped with. The four primary classes are PoE, PoE+, PoE++ type 3, and PoE++ type 4. PoE and PoE+ both utilize two twisted pairs of wires for power delivery whereas both types of PoE++ use four twisted pairs. One must also consider the total amount of power output needed between all ports so that it does not exceed the maximum wattage of the switch.[70]

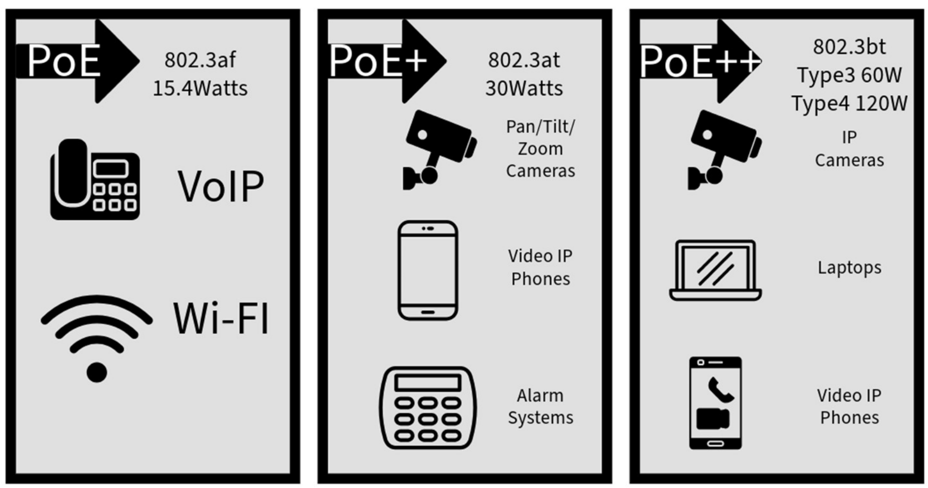

Figure 7: POE, POE+, POE++

Each of these options are important to consider before selecting a switch for a Dante network. There are switches being manufactured specifically for AoIP networks, such as those

[70] Charlene, "Understanding PoE Standards and Wattage," FS (FS, 2022), community.fs.com/blog/

designed for AVB which could be beneficial, however, for Dante, "AoIP aware" switches are not necessary. For example, in Dante, PTP aware switches are not a requirement, and in fact according to Audinate's training courses, "In most cases Dante does not benefit from enabling boundary clock or transparent clock on switches."[71] However, if AES67 is a part of your network, it may be beneficial to have a switch that is PTP aware to aid clocking accuracy.[72] What is most important is that the switch is a non-blocking layer 2 switch capable of at least 1Gb/s (no EEE or other parameters that may inhibit transmission, enough bandwidth for total network requirements).[73]

Network Ports

Generally, there is no need to make any adjustments to network port settings when using Dante. However, it is good to be aware of what ports are in brief, and which ones Dante uses to transverse the network. As discussed, when devices connect via a computer network, they do so using IP addresses, then use IP ports to identify what the connection is for. Using these ports, they grant or deny access to the other devices. Ports are identified by 16-bit or two-byte numbers (0-65,535). Many lower

[71] "Dante", 2021
[72] (Francis, 2014)
[73] ("Selecting," Yamaha)

ports are established for basic computer functions and networking protocols (DHCP, HTTP, FTP, etc.). For example, a computer attempting to access an HTTP web page might look like 192.168.1.20:80 where ":80" is the port it is attempting to access. A router can be used to block, allow, or forward these ports. Dante has specific ports it uses such as PTP or mDNS. Dante's primarily utilized ports are:

5353	Discovery
319/320	PTP
9998	PTP Clock Logging
14336-14600	Unicast A/V Streams
4321	Multicast A/V Streams

Table 5 Network Port Assignments

With unicast streams, every stream will have the full IP address on it. "… With multicast the IP address is more like a TV channel we tune into a stream based on its IP address and so we can sort based on that"[74]

Transport Protocol

Although Dante is considered a layer 3 solution, under the hood it operates at layer 4 as well. As a reminder, layer 4

[74] "Dante," 2021

describes the transportation method between devices. Dante uses a method called UDP (User Datagram Protocol). UDP is how Dante packetizes the audio data and sends it from one place to another.

UDP is a one-way transport protocol in which once sent there is no confirmation of delivery or return signal ensuring success. There is no data bank where it stores data already sent. Because of this, it is not usually used outside of real-time services as it is part of a category of transport protocols named “unreliable”.[75] However, because UDP does not store any data already transmitted nor does it have any return confirmation messages it has considerably less overhead and data space requirements than other transport protocols such as the Transmission Control Protocol (TCP) which is used for emails and other non-real-time applications.[76]

Realtime Transport Protocol (RTP) is another protocol (which is based on UDP) that other AoIP solutions such as RAVENNA, AES67, and AVB use. It, however, is categorized as ”reliable”.[77] This protocol has a ’handshake’ with each transmission ensuring that the packets were successfully delivered, thus its categorization. While RTP might be

[75] (Kieran, 2011)
[76] (Francis, 2009)
[77] (Francis, 2014)

considered more ”secure”, the pivotal deciding factor is time; UDP is a faster protocol as it does not have a return confirmation message like RTP. Both protocols are acceptable for real-time networks but could be part of the consideration when selecting the right solution for one‘s network.

Security

Security has been a topic of conversation in the AoIP community in the past. For Dante only networks it is not as big of a concern as the only access other than audio devices on the network are most likely one, perhaps two, instances of Dante Controller. The only security truly needed would be typical Wi-Fi passwords to keep unwanted users from accessing Controller routing. In converged networks, however, it is something to be more mindful of.

When discussing AoIP networks designed for archival work, Nicolos Bouillot wrote, “physical security of media is not a primary concern... but there are a new host of security issues... unauthorized access and distribution, file integrity, and malicious code. The network’s architect must provide security permissions that will allow access to a network audio objects strictly on a per-user or group basis.”[78] Dante does that

[78] (Nicolas, 2009)

exact thing within DDM where administrators can create password protected user profiles which have custom access given by the administrator.[79] Certain users can access select sets of domains or devices on a network without being able to access others. This helps prevent accidental mix-ups of standard device routing and helps keep unwanted guests from accessing controls to an important network.

Other important steps to take for security of a network include the use of firewalls on the router along with professional anti-virus software. These can provide security systems such as "stateful packet inspection, tarpits, blacklisting, port scan detection, MAC address filtering, SYN flood protection, network address translation (NAT), and typically a secure channel via VPN for remote users."[80]

Dante AV: Integrating Audio and Video

Dante also features integrated video solutions for those needing to transmit visuals over the same network. Like audio, video uses the same Controller and must follow the same rules of the road when it comes to using network infrastructure for transmission. However, where audio might be considered a small

[79] "Dante", 2021
[80] (Nicolas, 2009)

car, video is more like an 18-wheel truck going down Cat6 highways. Video has both a larger payload and a company enforced timeframe to deliver by, meaning it uses considerably more bandwidth and adds a new challenge of audio to video lip-synchronization.

Transmission of video can either work via unicast to an individual location, or multicast to multiple locations, but cannot travel to multiple locations simultaneously via unicast like audio can. The video and audio elements of the source can be sent independently to differing locations. This ability could be useful in many situations such as if a venue lobby has audio monitoring for a concert, but not television units for visual monitoring or vice versa.

There are 3 Dante Video implementation levels, Dante AV-Ultra, Dante AV-A, and Dante AV-H. Dante AV-Ultra and AV-H are the most popular categories. AV-Ultra is featured on native (out of the box compatible) Dante equipment and allows for 1 video stream along with 8 audio streams and various control signals (USB mouse/keyboard, inferred remote signals, Consumer Electronic Commands [CEC], etc.). It allows for sub-frame latency which is boasted at approximately 8ms.[81] AV-H is Dante's software solution which can be introduced into camera outputs

[81] "Dante AV," Audinate (Audinate), https://www.audinate.com

post-factory on compatible equipment that output H.264 or H.265 compressed signals. It carries 1 video stream and up to 2 audio streams and has multiple frames of latency, approximately 100ms.

First, the issue of bandwidth should be addressed. If there is not enough bandwidth, then transmission becomes impossible. Dante Video can still be operated on a 1Gb/s network, and is optimized to do so, but it has a bandwidth cap of 700Mb/s per stream.[82] To meet this demand, raw video cannot be transported. Instead, Dante uses JPEG2000 ("visually lossless") encoding on video streams which can accommodate up to 4k and 60 frames per second.[83]

How much of the 700Mb/s per stream a video takes varies based upon user settings such as frame rate, resolution, color chroma, and pixel bit depth. JPEG2000 typical bandwidth for 1080p at 60fps with a color chroma of 4:2:2 will use approximately 135Mb/s. 4k resolution with 60 frames per second and a color chroma of 4:2:2 will accrue approximately 675Mb/s. This should all be considered in contrast to audio's 1.5Mb/s per stream. Quickly one can see how this can fill the majority of a 1Gb/s network with only one stream. If more than one high quality video stream is required, upgrades may be nearly inevitable.

[82] "Dante", 2021
[83] "Dante", 2021

Resolution	1080p	1080p	1080p	1080p	4K	4K	4K
Chroma	4:2:2	4:4:4	4:2:2	4:4:4	4:2:2	4:4:4	4:2:2
Frame Rate	30	30	60	60	30	30	60
Bit Depth	10	10	10	10	10	10	10
Bandwidth	85Mbps	126Mbps	135Mbps	253Mbps	337Mbps	506Mbps	675Mbps

Table 6 Dante Video Settings and Requirements[84]

The bandwidth needed to transmit video along with the inherent time it takes to capture an image into a frame creates substantially more latency than is present in audio. For reference, each individual frame's duration at 60fps is 16.7ms. At 24fps the duration increases to 41.7ms. Encoding raw video signals to JPEG2000 or ingesting H.26x streams will once again increase the total time from camera to visual monitor. That is the reason that Dante AV-Ultra chooses to packetize video into sub-frames. In ultra-low latency mode, a packet is delivered to the decoder after 8ms. With consideration that each frame takes 16.7ms to capture (at 60fps), Dante can transmit the frame at less than half a frame of latency. Ultra-low latency mode transmits the video in an inter-frame manner, it breaks the image up into 16 regions and encodes them separately from top left to bottom right. This allows for faster encoding and

[84] (Audinate, 2022), 4

transmission time. In broadcast mode, the encoder will wait the duration of an entire frame before encoding and sending it. This is slower but keeps the entire image together. At 60fps this process would take 64ms (around 4 frames) from glass to glass. These numbers do change upon individual devices.

To ensure that audio and video are both in sync, Dante utilizes time stamps on both the audio and video channels. This allows for both signals to be sent separately then realigned upon arrival at the decoder just prior to play back. If further adjustments need to be made, Dante also offers a manual delay slider to compensate for encoding and decoding latency.

In line with the audio side of Dante, there is also video monitoring software for end-users to employ on computers. Dante Studio serves much like Dante Virtual Soundcard, it allows the user to receive RX (reception) feeds to monitor locally while also allowing for the computer to capture the screen’s output and send TX (transmission) signal to the Dante network. Dante Video Viewer is another software application offered which serves purely as a monitoring and control station for video signals on the network.

Lastly in dealing with security, Dante does support HDCP (High-bandwidth Digital Content Protection). HDCP locks copyrighted material transport to prevent piracy. Dante AV-Ultra allows unicast and multicast transmission of HDCP content in up to 16 devices in the video path in accordance with HDCP v2.3. Non-HDCP content can be multicast without limit, however.

SECTION 5: CREATIVE PROJECT AND CASE STUDIES

Equipment Setup

The equipment available for this project is as follows: 1x RME 12MicD (mic and line inputs, ADAT outputs, Dante i/o), 1x Ferrofish A32D (Dante/analog DB25/ADAT/MADI i/o), 4x Focusrite Rednet AM2 headphone amplifiers (headphone and line outs), 4x Audio Technica ATND971a, 1x Focusrite PCIeNX, 2x Cisco CBS350 switches, and copies of Dante Virtual Soundcard on computers as needed.

The First Test (9/29/2023-9/30/2023)

The first test of the Dante system was conducted in my home. It consisted of setting up the RME as an interface with my Mac Mini (running DVS) recording a single input and an AM2 for headphone monitoring. I then recorded a few cover songs with stereo acoustic guitars, lead and background vocals. The Ferrofish A32D was also connected, and a headphone mix was routed to it to ensure operation, but it was otherwise unused.

I encountered a lot of audible clicks and pops during tracking, though unsure at the time if it was captured in the recording. Upon playback, I discovered that the majority of the pops were not recorded, however, there were still too many for comfort. This problem was later solved by addressing clocking issues and learning that there were multiple EEE settings within the switch, only some of which were previously adjusted during my initial set up.

The purpose of this first attempt was to both ensure I had a working knowledge of how to route audio using Dante in a closed network and to test all the equipment for functionality. While testing, the RME, Ferrofish, and ATND microphones were all fully functional. Of the four AM2 headphone boxes #1 and #3 were fully functional, #2 worked but would not accept POE as a power source and had to have an external supply while #4 would accept POE but would not connect to a network. #2 and #4 were sent to Focusrite for repair.

Testing the AM2 boxes also threw an unexpected curveball, especially with this being my first hands-on attempt using Dante. Initially, none of the AM2 boxes would allow routing with my network, though they (the ones that were operational) could be seen in controller. After some time, it was discovered that the previous owner had assigned static IP addresses to each of the amplifier boxes and thus they would not connect with my Link Local network. To solve this, I assigned my Mac Mini a static IP in the same subnet as the static IP addresses and was able to change the AM2's IP inside Dante Controller. Doing this took a while, however, as the AM2 boxes had a static IP of 192.x.x.x which caused problems with colliding IP address numerals with my Wi-Fi network that the Mac Mini was also connected to. Now, however, if faced with this issue I believe it could be solved simply by selecting the piece of equipment and clearing the network settings.

During this test I discovered that the switch’s IP address did not have to reside on the same subnet ID as the pieces of gear it was connecting. The switch IP address was 192.168.1.254 (standard switch IP at initialization), while all Dante equipment were using Link Local’s 169.x.x.x addressing. The link local addresses were expected, but I was surprised by the switch not adhering to the same addressing. To make configuration changes to the switch I learned to set a manual IP address on my Mac Mini to be in the 192.x.x.x network to be able to access the web portal for the Cisco switch. This is the method I continued to use throughout my time in this project.

It was learned, after some amount of time, that if the controlling computer is connected to the switch at 192.x.x.x while simultaneously attempting to connect to a separate local Wi-Fi network that is also using the 192.x.x.x architype that there is interference and accessing the switch’s web portal becomes impossible. To solve this, turn off or disconnect from the Wi-Fi, or connect the switch to the same network supplying the Wi-Fi. Doing the later, of course, takes over for Link Local using DHCP from your new network.

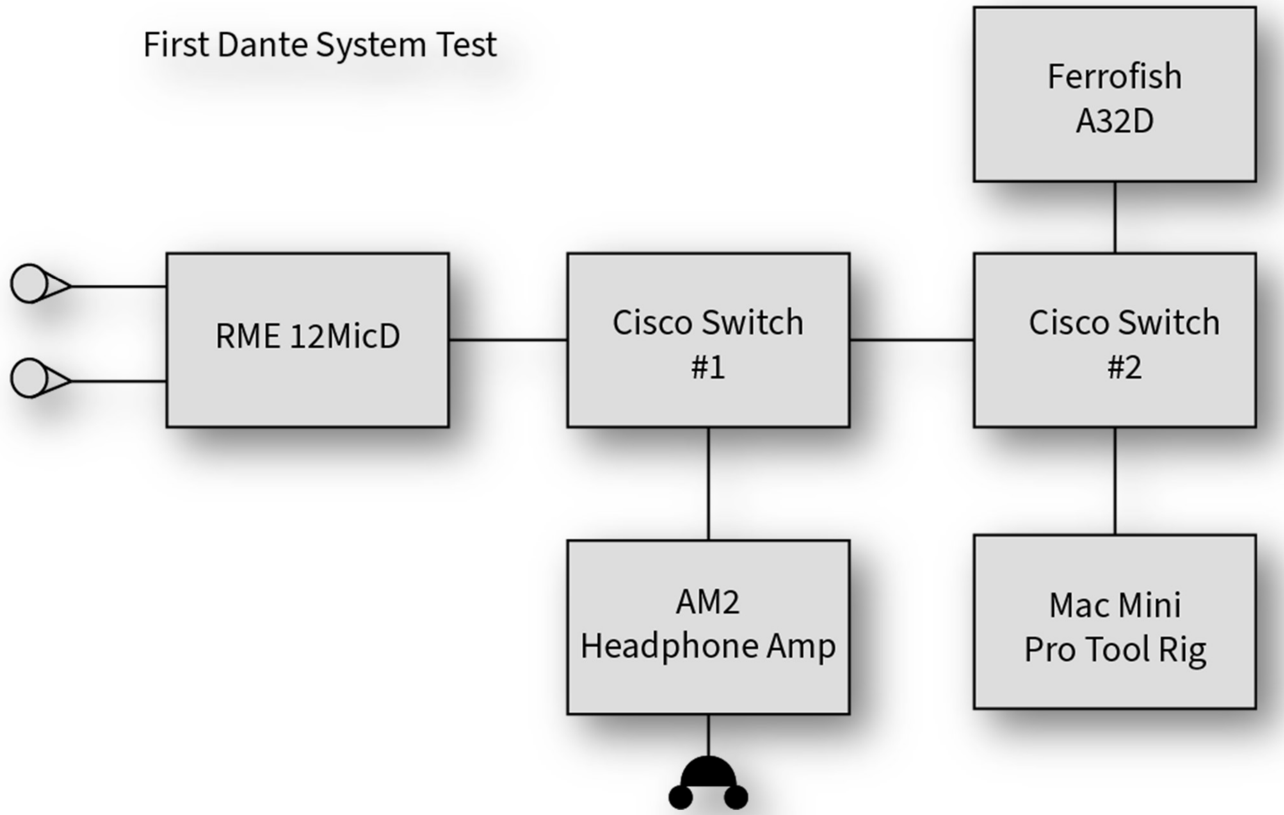

Figure 8: First Dante System Test

Lauren Gunn Recording (10/15/2023)

The first field recording conducted with this set up was of the artist Lauren Gunn singing her newly released song "Open Fire". This was produced accompanying a tiny-desk style video for the music business podcast Successfully Unsigned. Pre-production notes stated she was going to sing and perform with an electric guitar and amplifier while playing to pre-recorded tracks. I prepared by bringing a Rode NT2 for vocals, and an SM57 for the amp. There was a sizeable and interestingly shaped closet nearby which was going to be used as an isolation chamber for her amp, and I intended to make use of the ATND microphones as a room mic. However, when the artist arrived, she brought an acoustic guitar and no pre-recorded tracks. Plans shifted, and a second Rode NT2 was brought out to capture the guitar in stereo, and a SE large diaphragm condenser of unknown model was borrowed from another member of the crew to capture her vocals.

The Dante setup for the actual tracking was simple: 1x switch, 1x RME 12MicD, 1x computer running DVS and Pro Tools, and 2x AM2 headphone boxes. One AM2 was given to the artist while the other was kept for the engineer to monitor the recordings.

There were still a few clicks/pops during the recording. Clocking was set to the RME as the leader. The thought during the session was that perhaps DVS was not configured properly. Thankfully, there were no clicks or pops present in the actual recorded files. This should have alerted me to DVS *not* being the issue.

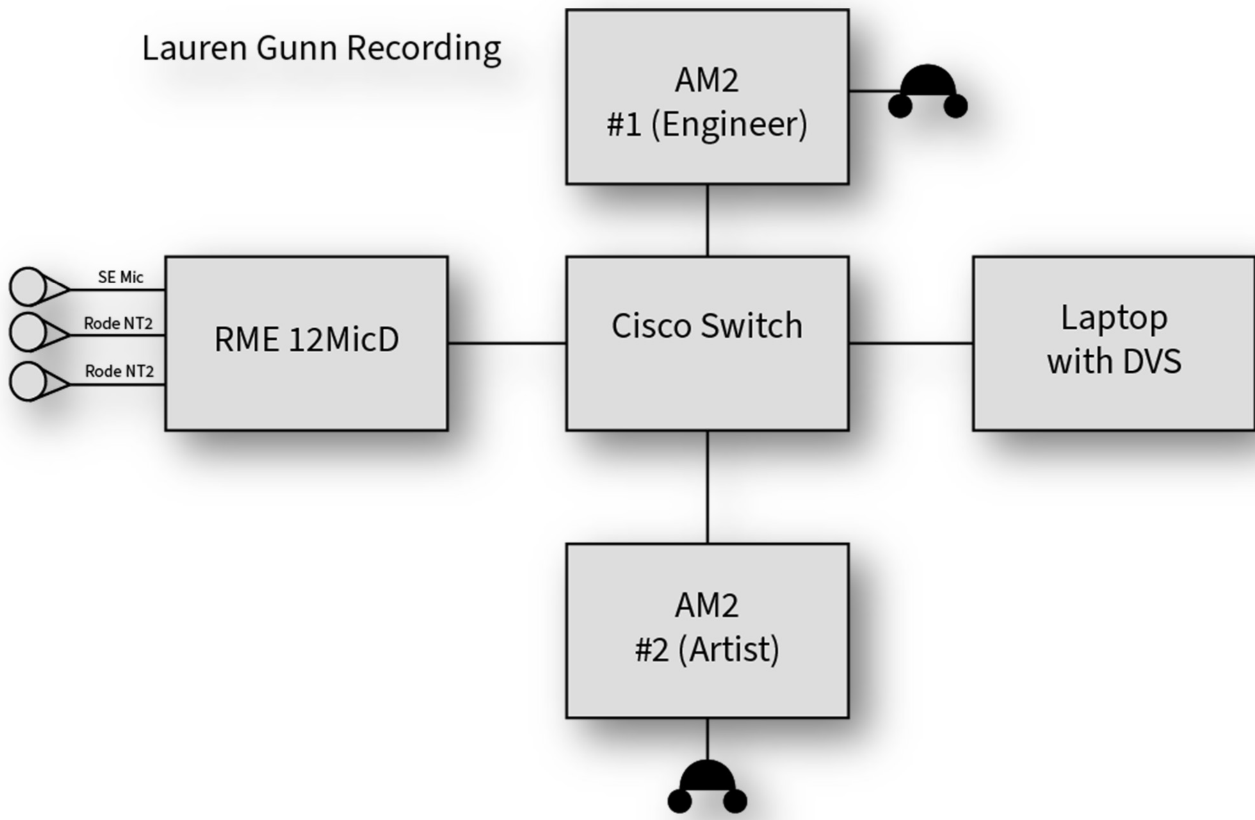

Figure 9: Lauren Gunn Recording

Chris Young Café Meeting (10/18/2023)

Met with both advisors, Michael Hanson and Frank Baird in the Chris Young Café (CYC) on campus. I demonstrated my current Dante setup and how I was using it to date. I also updated them on what I had already accomplished and what was planned next for the project.

During this meeting, to prove a little more understanding of Dante, they had me setup then connect my system to a DigiCo console that was present in the room and I successfully integrated that into my closed network, both sending and receiving audio from the console. Interestingly, though the console would transmit and receive audio, it would not show an IP address on the console. This led to a conversation that they had been having issues with some of the Dante compatible equipment inside the CYC, so we spent time looking and attempting to diagnose.

During my demonstrations of my system, I learned that the Cisco switches in my rig had reset their settings to factory defaults including the usernames and passwords. I also learned that there are two stages to applying and saving other settings (such as EEE or QoS) inside the Cisco switches and both stages must be engaged for it to become permanent. That resetting of settings is most likely why there continued to be pops and clicks during the recording session with Lauren Gunn.

While at the CYC, we also booked MTSU Studios D and E for 8 hours simultaneously for me to experiment with. To aid this I requested ADAT (light pipe) cables and D-Sub to XLR connectors to interface with the hardware in D and E.

Inter-Studio Test #1: D and E (10/23/2023)

The goal of this test was to see if I could successfully connect two studios together using Dante, and if so to have an artist performing (vocal overdubs) in one studio while I engineered from the other.

The Ferrofish A32D was set up in Studio D while the RME 12MicD was placed in Studio E. To get signal from the rooms onto the Dante network in studio D lines were patched from the console to the producer tie lines where it went into the Ferrofish via a D-Sub connector. In E, the RME device was connected to the studio’s host Pro Tools HDX via ADAT ins and outs, and clocking was set in the HDX system to follow the ADAT input from Dante. Both Dante devices were connected to each other over MTSU’s campus network directly at a wall panel. Since they were on the same subnet, Domain Manager was not needed.

Inside Studio D, I tested this set up by playing music from YouTube out of the loudspeakers which was captured by a SM57. The SM57 then went through the typical order of operations for that console, routing through Pro Tools, back to the console, then out to the producer tie line as before mentioned onto the network. Success. Inside Studio E, I was able to listen in real-time to the source material in Studio D.

After this test was completed, I brought in an artist friend, David Overstreet, to record vocal overdubs. At this point, I reversed the set up placing the RME in Studio D with the artist with the Ferrofish inside Studio E where I intended to engineer. This was done to take advantage of the quality of the RME's microphone pre-amps. At this point, the console in D was skipped, opting for a direct path from the microphone tie line being patched to the producer tie line then out XLR at mic level to the RME for amplification. From there, Dante routed the signal to the Ferrofish in E which was connected to the HDX system via ADAT optical cables like the RME before it. Once through Pro Tools in E, the signal was then patched out of the DAW outputs in E to a D-Sub and back into the Ferrofish to complete the circuit for the artist to be able to hear. This too was successfully completed; however, the artist's original tracks were all at 44.1kHz and I was unable to swap the Ferrofish to any other sample rate except 48kHz. Up-sampling his session caused issues with timing and tone, thus for the sake of the session I moved to a traditional use of Studio D with no Dante connections. Time constraints kept me from experimenting further that night. Later, I would discover that the Ferrofish's clocking needed to be changed within Dante Controller rather than on the hardware interface which I was attempting to use.

In this session, everything worked fully duplex between the two studios. The only two exceptions to success in this session were the simple sample rate issue, and a more complex issue of the Cisco switches being unwilling to connect to campus's network or pass DHCP to devices connected through them. The solution to that was to patch the Dante equipment directly into MTSU network ports rather than via the switch first. A conversation later with RIM IT faculty member Chris Dilday revealed that it is possible that MTSU ITD has security protections against managed switches on the network without approval. Latency was not accounted for in this session and may have proved an issue if we had been able to continue.

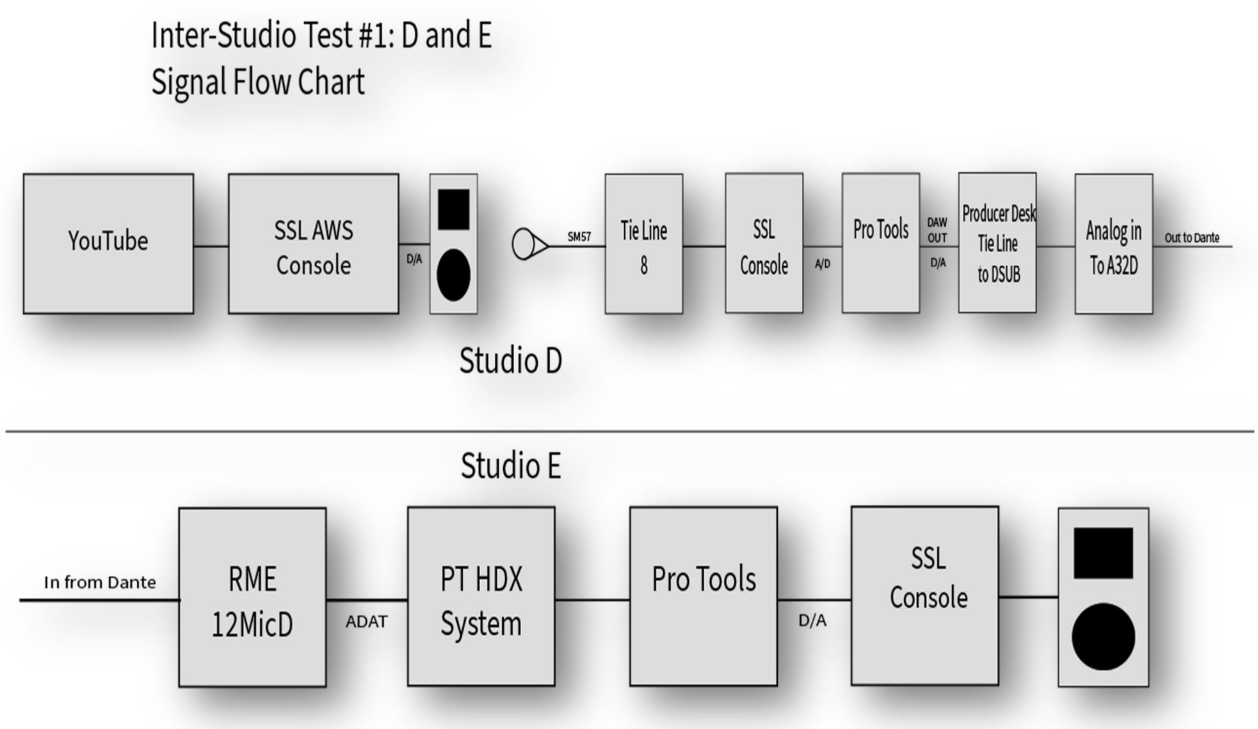

Figure 10: Inter-Studio Test #1 D and E Signal Flow Chart

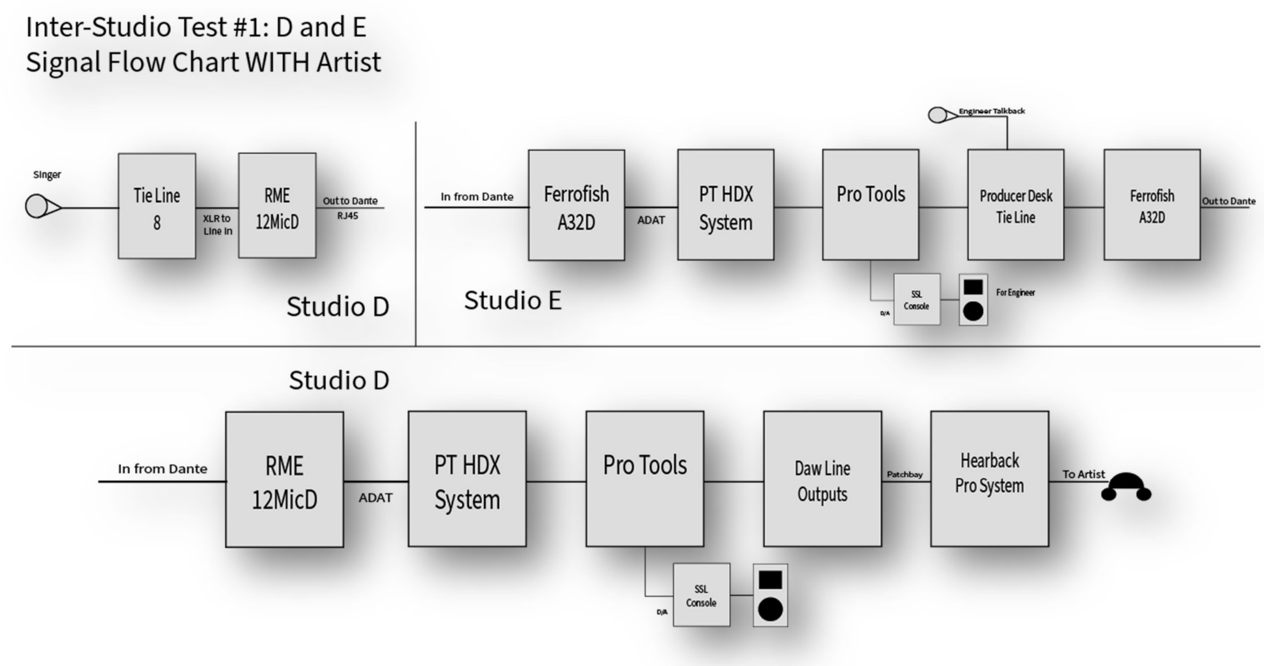

Figure 11: Inter-Studio Test #1: D and E Signal Flow Chart With Artist

Jazz Location Recording (10/20/2023)

On October 20th, 2023 I produced a jazz location recording at St. Paul's Episcopal Church in Murfreesboro, TN. Tevin Turner engineered and assisted on this session and was the primary mind behind microphone placement among many other things. We recorded a jazz quintet comprised of MTSU School of Music majors under the pseudonym The Mighty Men of Murfreesboro which played four songs: "Stella by Starlight", "Lady Bird", "What is this Thing Called Love", and "Stringing the Jug". The quintet was comprised of piano, upright bass, electric guitar, saxophone, and trombone. You can find the microphone list in Table 1.

SOURCE	MICROPHONE	NOTES
BASS	Lauten LA-120	Omni capsule
PIANO (STEREO)	Rode NT2	Cardioid
E GTR	Royer R101	
SAXOPHONE	AEA N8	
TROMBONE	AEA 28	
ROOM #1 (STEREO)	Neumann KM184	ORTF
ROOM #2 (STEREO)	Neumann KM183	Wide spaced pair

Table 7: Jazz Location Recording Mic List

The RME 12MicD was placed in the middle of the ensemble who were spread out in the parish auditorium's oration area. This room was chosen for its sonic imprint and did not disappoint. All reverb heard in the mix is natural except a small amount placed on the bass and piano due to their volume not exciting the room as much as the other instruments. A Cat6 cable was then run to the foyer of the building where Mr. Turner and I sat recording the session. The network cable entered a Cisco switch which split to a laptop with DVS and Pro Tools, an AM2 headphone amp, and the Ferrofish which was also used for a headphone feed. The second functional AM2 was placed in the live room with an active monitor connected to its line output as a method of talkback to the artists while tracking. The input for the talkback was an ATND with the engineers. The routing for the talkback was done entirely within Dante and did not touch Pro Tools since it did not need to be recorded.

The session itself ran smoothly. There were some occasional clocking clicks/pops, however, those were not printed on the .WAV files. There were, however, two moments of complete audio drop out in the first song “Stella by Starlight”. The longest dropout was almost a half of a second, but this problem did not return anywhere in the rest of the session. These holes in the music were patched and disguised in the mixing phase as best as possible. While attempting to check if EEE was fully turned off on the switches I learned that I could only access one as they both by default have the same IP address. I do not know which one was checked, but it did have the proper settings. To remedy this, I would need to change the IP address of one of the two switches to something different or connect to them directly while the other switch was not present on the network.

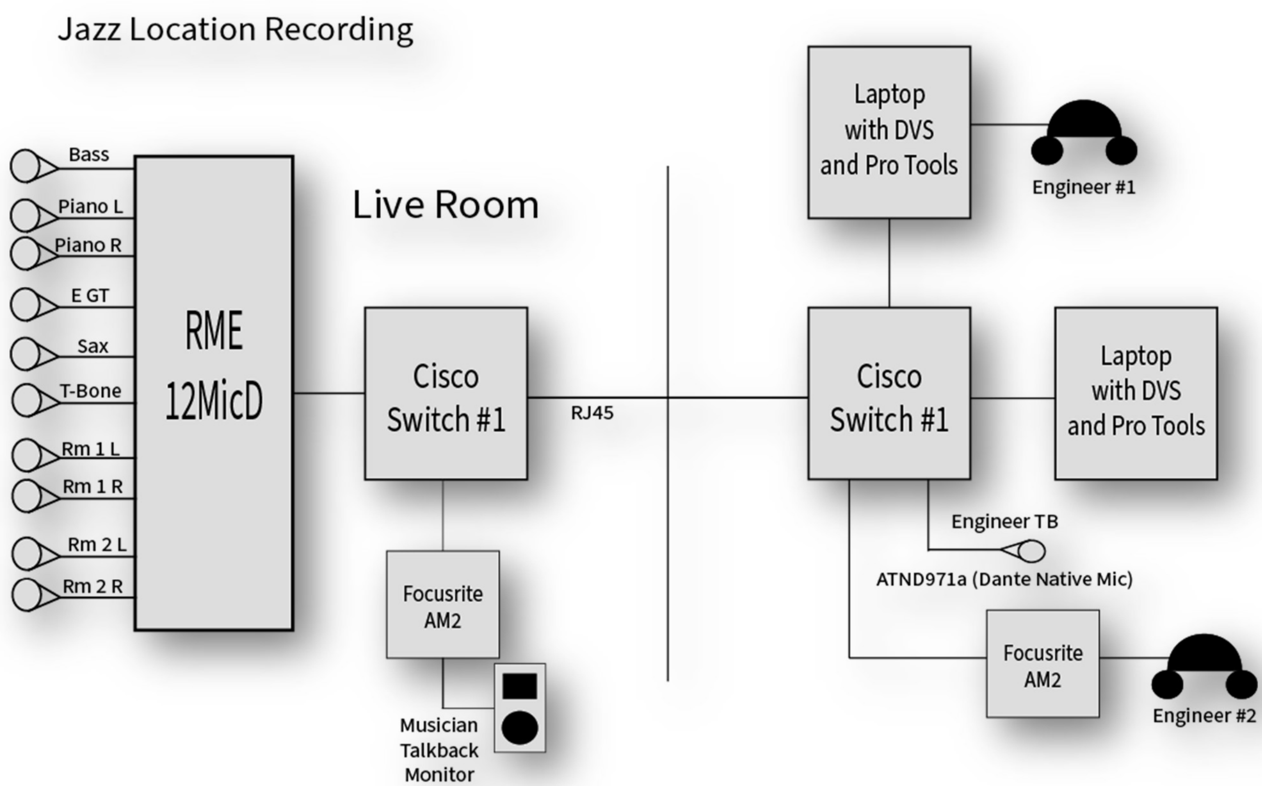

Figure 12: Jazz Location Recording Block Diagram

Testing Switches at Home (11/10/2023)

While enjoying downtime at home, I went about solving some issues with the switches' configuration. To begin, both switches were reset to factory defaults to create a clean slate. EEE was thoroughly set properly then re-checked, and proper usernames and passwords put into place. As all of the networks I intended to use these on are small and dedicated for audio I did not adjust QoS settings. The main question I wanted to find the answer to, however, was why the managed switches would not connect to the Campus network. Again, it was suggested by at least two faculty members that it could be security protocol, but I did not want to let that be an excuse for me to not dive deeper. After testing, the switches also refused to pass network connection along to any device if connected to my home's router as well. This told me that at a minimum something was not set correctly inside the switch. After some online reading and research, I discovered a setting called "DHCP Relay" which defaults to "OFF" and can only be found in the Advanced settings view mode. Once engaged: success. It then accepted my router's DHCP, and all devices now registered with 192.x.x.x IP address and communicated properly, replacing their link local addresses. With this new information, the goal was to try it on campus at the next opportunity, which would be on the next day.

Hinton Hall Switch Test (11/11/2023)

Fellow student, Bob Simmons, had a recording session at Hinton Hall of a piano trio and had requested the use of the RME 12MicD for recording. I met and delivered to him the RME around 12pm but was unable to stay more than an hour due to other commitments. In that hour however, I aided them in loading in and setting up the Dante network. Bob wanted the RME to be placed backstage where he would sit with a laptop running DVS and Pro Tools, but otherwise was not using any other Dante equipment. While they continued to set up microphones and other necessities, I opted to find other ports inside Hinton Hall and test the newfound DHCP Relay settings on another network.

One switch was connected to a wall panel backstage where his laptop and the RME branched off from. The second switch found a home in the sound booth and was connected to MTSU's network where the Ferrofish and my laptop (also running DVS and Pro Tools) was connected. Immediate success was accomplished. I could access and control both the Ferrofish right next to me and the RME down at the stage thought IP addresses given from MTSU's DHCP.

In Dante Controller, Simmons' laptop was grayed out which I would later realize due to him running at a different sample rate. The Ferrofish still refused to budge from 48kHz (still operator error), but that was of no consequence to the session since it was not needed for recording. This did mean that I could see and "record" audio in Pro Tools but was unable to listen through the Ferrofish due to the sample rate issue. This audio had significant issues due to the sample rate differences though. There was not enough time to check an AM2 to see if it could monitor the inputs, but I recorded some of the room noise as they were setting up the session.

Another interesting thing of note, Simmons' computer could not see mine listed in Dante Controller though I could see his. Initially he could not see anything on the network, but that was solved as I checked his PC's network settings and made a change of discoverability form "private" to "public". Even after that change, though, his computer still did not see mine. Time did not allow for more experimentation, but enough was completed for their session to function and for me to learn that the DHCP Relay function worked on campus as well.

Tennessee Valley Winds (12/12/2023)

At their annual Christmas Concert, the Tennessee Valley Winds performed eight songs which I recorded with the assistance of Mac Palao. This was done on location at The Washington Theatre and was a very simple setup. The RME was placed backstage left with all microphones were connected to it. A Cat6 cable run took the network into the green room where a switch lived with a recorder laptop, the Ferrofish, and an AM2 headphone box. There were no networking or hardware issues during this session, it was simple and straightforward.

This recording was made with eight microphones (see Table 2) placed around the ensemble and room. The main stereo spaced pair resided behind the conductor with two cardioid microphones flanking six feet to either side. An X-Y pair was placed in a direct line upstage from the main array to capture the spread out percussion section. The final two microphones were a decorrelated pair of omni microphones placed in small open ended hallways that hold the short stairwell from ground level to the stage which had previously been noted as having a pleasant sounding reverb.

SOURCE	MICROPHONE	NOTES
MAIN SPACED PAIR	Lauten LA-120	Omni capsules
X-Y PERCUSSION SPOT	Rode M5	
FLANKS	Lauten LA-120	Cardioid Capsule
HALL	Rode NT2	Omni

Table 8: Tennessee Valley Winds Mic List

Christmas Livestream (12/21/2023 - 12/22/2023)

During December the music business podcast, Successfully Unsigned, decided to host a Christmas themed livestream show. This included both podcast host segments, live singer-songwriters, and pre-produced advertisement videos. This event was streamed to both YouTube and Facebook on December 22nd, 2023. The day before was also used for setup and testing. There were three on-screen hosts (David Overstreet, Patrick Glover, and

Dale Shackleford) and 5 performing artists (Lauren Gunn, Missy Ecker, Luke Robins, David Overstreet, and Dale Shackleford). The show alternated back and forth between artist performances and talk-show style hosting with advertisements played at various intervals. I designed, setup, and mixed the audio and video for this show while hosting. David Overstreet took over operations during my performance segment.

Dante was used heavily in this production, and it would not have been possible without it. The rig for this event consisted of the RME 12micD, one Cisco switch, three Focusrite AM2 boxes, two laptops running DVS, and one Mac Mini connected through the Focusrite PCIeNX card. All twelve inputs to the RME were reserved and setup for use though only nine inputs were used during the actual show (see Table 3) Similarly all ten ports of the Cisco CB350 switch were used, with one connection redundant (see Table 4).

SOURCE	MICROPHONE	NOTES
VOCAL #1	Peluso PS-1	
VOCAL #2	Rode M2	
GUITAR (STEREO)	Lauten LA-120	Cardioid Capsule
ELECTRIC GUITAR AMP	AEA N8	

ELECTRIC GUITAR DI	n/a	Dual 1/4" inputs
ROOM (STEREO)	Rode NT2	Cardioid
TALK HOSTS (X3)	Shure MV7	

Table 9: Christmas Livestream Mic List

CONNECTION	**NOTES**
RME 12MICD	
LAPTOP #1 (OBS)	DVS, OBS, ATEM Control
LAPTOP #2 (MULTI)	DVS, Adobe Audition
FOCUSRITE PCIENX	Thunderbolt connection to Mac Mini
MAC MINI	Pro Tools; Unnecessary connection with the thunderbolt bridge in place
REDNET AM2 #1	For Engineer; POE
REDNET AM2 #2	For Artist; POE
REDNET AM2 #3	Line out for house PA; POE

BLACKMAGIC ATEM MINI	Non-Dante, for software control
INTERNET ROUTER	For direct line for best internet stability for streaming computer

Table 10: Christmas Livestream Network Switch Connections

From the RME pre-amps the signal progressed to two direct locations, the PCIeNX card and Laptop #2. Laptop #2 captured multitrack unprocessed audio from the pre-amps. The PCIeNX also received the unprocessed audio, mixed said audio, then created the various mixes needed for the event (2Mix for livestream, engineer mix, artist mix, house PA mix). Each of these routed in and out of the Mac Mini via Dante and the PCIeNX card and were mixed live in Pro Tools. The end audio product was recorded in two places, the streamed 2Mix was recorded in sync by OBS and the multitracks were recorded raw by Laptop #2.

As I did not have any Dante AV productions, all video was routed to a BlackMagic ATEM switcher, except for one iPhone source that connected wirelessly. Three Black Magic Pocket 4k cameras, one Sony A7s, and a screen capture of the Mac Mini mixer screen via capture card were connected via HDMI to a Black Magic ATEM Mini Extreme ISO switcher which connected to Laptop #1 via USB. Laptop #1 was operating OBS (Open Broadcaster Software) to create the stream that was sent to Restream.IO which duplicated the stream for multiple locations at once (YouTube and Facebook). OBS recognized the ATEM switcher as a single source, the iPhone as a second source, and the pre-produced videos as additional source scenes for all video needs. For audio, a copy of DVS was also operating on this computer where it was set to receive only 2 channels (Left and Right) sent from the mixer. That Dante input was manually programmed into each of the video scenes needed inside OBS.

This was a challenging event as it was the largest and most complex setup to date which I had to design using Dante. It pushed me to try, fail, and try again all while pressured by it being a live public performance. I ran into various new issues that I had to learn ways around such as Laptop #1 running OBS would not see or recognize Dante inputs in OBS. With experimentation and extensive online research, it was discovered that for Dante and OBS to play nicely together, Dante Virtual Soundcard must be running in WDM (Windows Driver Model) mode, not DVS's default ASIO (Audio Stream Input Output) mode for windows.

The first day (setup day) all devices worked properly with no problems, but on the second day (show day) other issues began including particular Dante devices refusing to be discovered inside Dante Controller. All power and network cabling were checked for layer 1 routing issues, but all was correct. No amount of powering devices off and on provided a solution. During this time there were issues that I can only describe as a "ghost IP" issues where a device would be named inside Controller, but had no IP address, information, or controllability. You could refresh Dante Controller and the devices affected by this disease would change or swap with other devices erratically. Eventually the solution was found after remembering advice given by my advisor Michael Hanson weeks prior in a different conversation to adhere to a strict power-up sequence: with *all* equipment powered OFF, turn the network switch on and wait a few minutes (let it "warm up"), followed by turning on all hardware equipment, leaving Dante Controller to be started last.

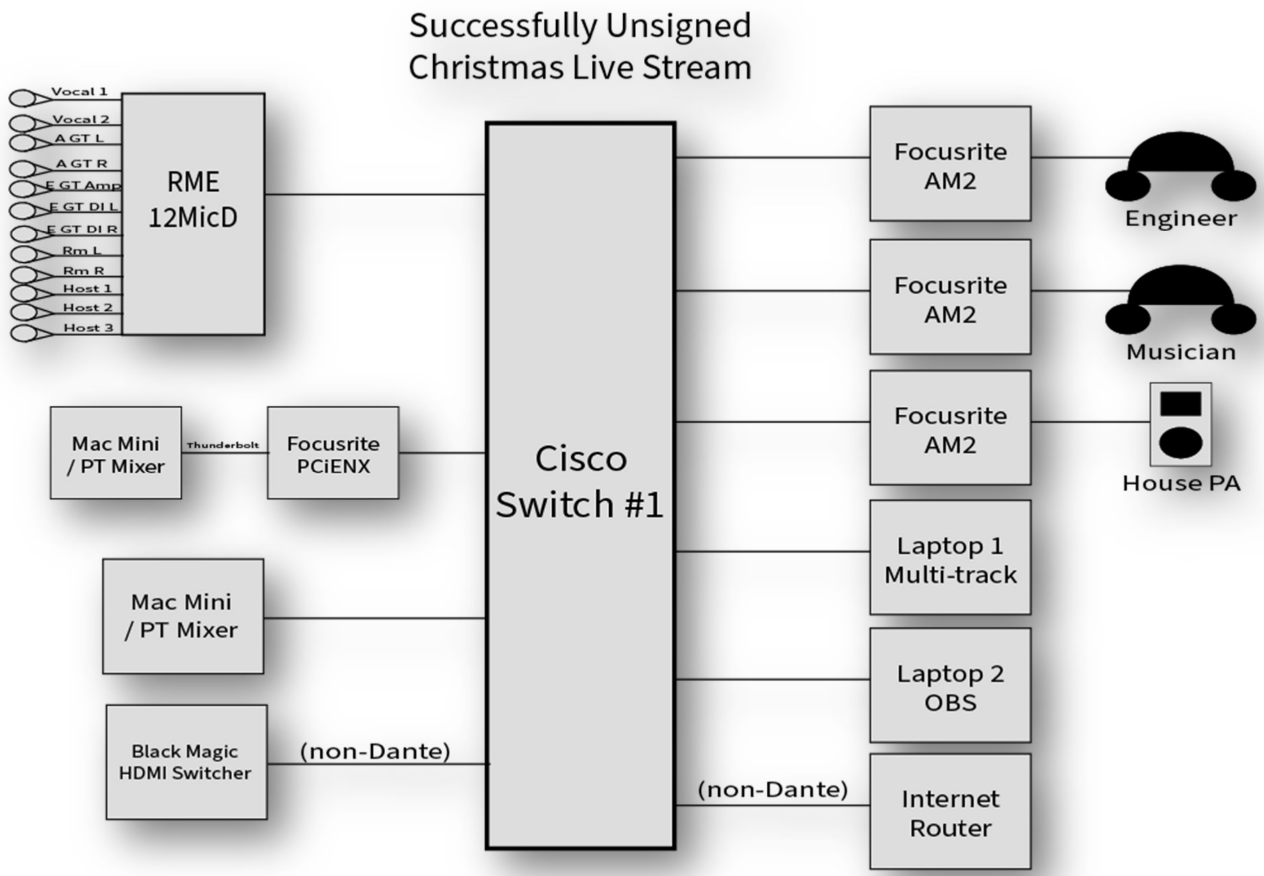

Figure 13: Christmas Livestream Block Diagram

Acapella Demos (12/30/2023)

While amongst family, it was requested I record some acapella demos of songs my sister had written. This consisted of an extremely simple set up: 1 microphone into the RME connected to a switch and laptop running DVS with two RedNet AM2 headphone boxes for each of us. The majority of the time was spent rearranging the written sheet music to better reflect the song and make the notation more singable. As such, the recordings were not finished, and it was decided to scrap them and begin again at a future date.

Inter-Studio Test #2: B, C, and E (01/13/2024)

Since the campus was quiet the weekend prior to Spring semester classes beginning, three studios were booked (Studio B in the Bragg Building, Studio C downstairs in the James Union Building, and Studio E in the West Main St. building) to attempt a multi-studio connection test with a focus on determining the latency between locations. To aid in this test, two first year graduate students, Mark Smith and Riley Jermier, were recruited.

Going into this session I was under the wrongful impression from a test conducted the previous summer that the Bragg building, The Chris Young Café, and Jones Hall were all on the same subnet and that DDM had not been used. This previous test had all three buildings connected with a small two-channel transmissions sent. At that time, I was under the impression that DDM was on an unconnected dead-end server. Believing this, during this test I hoped that if Bragg, the CYC, and Jones Hall were all on the same subnet that it was possible that at least two of the three studios, Studio B (Bragg), Studio C (JUB), or Studio E (West Main St.) might be on the same subnet. After a few hours spent traveling between the studios and attempting various available network ports this was discovered to be false and none of the equipment would connect.

STUDIO	DEVICE	IP ADDRESS
B	RME 12micD	161.45.134.113
C	Ferrofish A32D	161.45.228.59
E	Focusrite PCIeNX	161.45.246.156

Table 11: Inter-Studio Test #2: B, C, and E IP Address List

Aside from subnet mismatch and a lack of DDM, there were two small issues encountered. The first and lesser was that the Ferrofish still refused to show its IP address on the hardware LED interface. After consulting the technical manual for the device, I found the menu page the IP is intended to be displayed on but the information line for IP addressing is simply non-existent. I searched every other menu available on the hardware interface with no luck, and my assumption is that the IP address line must have been added to the programming in the second version of the model and that the manual was created with that second version in mind. To find the subnet of that room I used a laptop plugged into the campus network and recorded the assigned IP address.

The second issue was that the PCIeNX card refused the be recognized in Dante Controller. The LED indicators on the device itself showed both power and network connection and the thunderbolt bridge forwarded internet access to the connected Mac Mini, but Controller would not see the PCIe card. Many attempts at restarting the device or refreshing Dante Controller proved unsuccessful. Eventually the device decided to begin working properly without any understanding as to what changed or why.

Inter-Studio Test #3: Bragg, and Faculty Room D (02/16/2024)

After several meetings with faculty, I learned that Dante Domain Manager was indeed installed and connected to MTSU's campus network. I gained full access to DDM and permission to test transmission on campus as needed. My hope for the final project session was to connect one of the studios in Bragg (A or B) with either Studio D or E and have musicians performing in both rooms. In addition to this I hoped to have another engineer in Studio C crafting live-to-2mixes. To see if this was even a possibility, I needed to test the latency between the Bragg

Building and the building D and E reside in. With the help of faculty, I was able to set up the PCIeNX card inside the Audio Maintenance shop in Bragg and the RME 12MicD in the teacher's facility next to Studio D. As before, the network switches did not want to connect to MTSU networking so both devices had direct connections to wall ports.

For this test I set up a custom domain inside Dante Domain Manager and enrolled both devices in it. This allowed me to bridge the audio transmissions between the two subnets. 64 channels of audio from an old studio session were looped and transmitted from the PCIeNX card to the RME for approximately an hour. Dante Controller revealed no transport issues or clocking issues. However, a lengthy latency time of 280ms was reported. At over a quarter second, this is of course well outside allowable latency for synchronous performances. This time did decrease during the hour to an average of 250ms as it was late in the afternoon and students began leaving campus. While this extensive latency made using Studio B and any studio in a different building an impossibility, everything did function properly.

One major caveat to this test is that due to locations, timing, and parking issues I was unable to go to the RME and listen to the transmission. What I sent was in good faith and in reliance upon the reports from Dante Controller. Even though the connection was secured, and the latency test should speak true, in the months after this test I began to wonder if the audio was actually arriving properly. Now, I realize that while I had enrolled both devices in the same domain in DDM, I had not set up the domain to share clocking across subnets. This would cause the transmitted audio to cut in and out or not arrive at all, but should not alter the accuracy of the latency times observed.

Inter-Studio Test #4: D, and E (02/27/2024)

Since having musicians in Bragg and another building was now out of the question, it was necessary to test latency between another two studios. Studios D and E (which are in the same building) were booked for this purpose. Since both are in the same building there should be no latency challenges to send audio between the rooms, however, prior to the main session this needed to be confirmed.

First, both control rooms were confirmed to have six network ports (most already in use by MTSU equipment) with both studio’s respective live rooms also boasting four ports, all of which were available, supplying PoE, and were on the appropriate network. The Focusrite RedNet PCIeNX card was connected to a Mac Mini and the campus network in Studio D’s control room while the Ferrofish A32D was connected to the network in Studio E’s control room. The same 64 channels of audio were sent from the PCIeNX card through the Dante network to the Ferrofish and a stereo mix was sent to its' headphone amp and an AM2 amp. The headphone amplifier I took to each port in the live rooms and listened to the mix. Latency in the rooms was confirmed to be under 1ms, an acceptable amount.

Final Session Setup Day (03/05/2024)

Studios C, D, and E were the final chosen locations for the project's culmination session. Inside Studio D and Studio E musicians were dispersed to perform with engineers in both rooms taking care of their talent. Inside Studio C a stream from all inputs was sent for live-to-2mixes to be made, creating a "broadcast"-like scenario. The studios were booked for approximately 48 hours consecutively to facilitate setup and the performance of the session.

The plan for the setup day initially had three participants: Mac Palao, Bob Simmons, and myself. The day before the session I learned that Simmons had come down ill and was unable to help, and that Palao's job called him in earlier than expected by several hours allotting him only one hour he would be able to aid setup. While this made things more difficult, it did not stop progress.

One major goal of this session was not only to use the three studio rooms, but to also incorporate as much existing gear as possible. This would showcase that not only is using multiple live rooms at once possible but that the consoles and outboard equipment in the control rooms are not made obsolete by this process and that they can still contribute to the sonic quality of the recordings even if they are not necessary. Because of this desire, the signal routing is considerably more complicated than needed in order to use said equipment.

Setup began with getting all Dante equipment connected inside D and E. The RME 12micD was placed inside D with post-fader sends from the console channel path going into the RME’s line inputs which were then routed to the Dante network. Inside E, a Ferrofish A32D was placed where ADAT inputs and outputs to the Pro Tools HDX system were connected feeding and being fed by the Dante network. A separate system was installed in D with a Mac Mini running Pro Tools connected to the Focusrite RedNet PCIeNX card. This Pro Tools rig would receive the inputs from both the RME and the Ferrofish to capture multitracks and to distribute the headphone cue mixes.

The initial plan was not to use the PCIeNX card for the headphone mixes, but instead use the HDX rig in Studio E. There was a concern, however, that the ADAT conversion process would increase latency beyond what would be acceptable for the musicians. I needed to know what the longest round-trip latency of the conversion processes would be. This was tested by setting up a microphone inside Studio D plugged directly into the RME. From the RME this signal was sent two places in order to create the comparison: (1) directly to the PCIeNX rig for the shortest possible route, and (2) to the Ferrofish which converted from Dante to ADAT to the input of the Pro Tools HDX system which would then convert the audio to an analog signal for the console channel path line input. After going through the console, this signal would return to a digital state for the HDX system to be able to capture it in Pro Tools then would again go through an ADAT conversion process to finally reenter the Ferrofish and Dante network, sending to the PCIeNX rig. By making a transient sound into the microphone, recording both signals at the PCIeNX rig, and measuring the distance between both transient returns it was determined that the latency induced by the conversion processes was 12ms. This was too long. There was another suggestion from Professor Hanson that switching from ADAT inputs to analog inputs going into the Ferrofish might reduce this latency by nearly half. However, upon testing this theory there was no change. Because the analog run was routed and running, I chose to leave it instead of returning to the ADAT inputs. Due to this discovery, it was decided to use the PCIeNX card as the headphone cue mixer because it would receive the direct input from both the RME and the Ferrofish with the shortest available runs, creating headphone mixes with the lowest latency possible.

Initially, switching to the analog inputs to the Ferrofish caused signal issues such as distortion, noise, and drop-outs. There were two issues discovered when solving this, one was simple cabling needing replacement. The second involved clocking issues. All the Dante equipment clocking was set properly and was communicating correctly, however, the HDX system needed to use the same clock as Dante system for proper use. This was achieved through selecting the ADAT input as the clocking source inside Pro Tools. With these two things changed, all signals from both rooms transmitted correctly.

Studio C was the last multi-track destination that needed configuration. Studio C resides in the basement of the James Union Building across campus in a different subnet. A second Ferrofish A32D was borrowed from the university to be the inputs from the network into Studio C’s SSL console. However, upon arriving at the studio I discovered that the back of the equipment rack was locked, unlike every other rack on campus, and it was after hours to be able to call anyone to unlock it. As a backup plan in the situation that I was unable to secure the second A32D, I requested that a copy of Dante Virtual Soundcard be installed on the Studio’s host computer. I supplied the license and enrolled the copy of DVS into the Dante Domain.

The A32D was left plugged into the network even though no signal was going to be routed to it because of the necessity of a hardware piece of gear be present in the subnet to act as a boundary clock.

To test the reception of multitrack information in C, I routed all channels in Dante Controller (from a drum kit set up earlier in the day), began recording in C, drove to Studio D and played on the kit, and drove back to C. Nothing showed up in Pro Tools on the system in C. The drum kit and associated setup was confirmed to be operational from tests earlier in D and E, and I confirmed that all signal routing was correct, but no audio was present in the recordings. After attempting a second time with slight variations and no success I chose to go home for the night as it was 2am and I had been working for approximately 15 hours. A solution needed to happen, but the plan then was to reattempt the next morning after some sleep.

The last set of Dante destinations were the AM2 Headphone amplifiers. These were set up as needed throughout the day, but all four were used for this session. One resided in Studio D's live room for the drummer, two in Studio E's live room for the guitar and bass player, and the last lived in Studio D's control room with the line outputs feeding two monitor return inputs on the console for engineers in D to be able to hear the guitar and bass from Studio E. Again, all these mixes were created in the Mac Mini Pro Tools rig connected to the RedNet PCIeNX card.

Other interesting things of note during setup day included learning that RedNet Control (Focusrite's network equipment controller) will not recognize/connect to any equipment on the network if the equipment is currently enrolled in a domain in Dante Domain Manager. Campus network settings did not allow for automatic discovery of some equipment on the network either inside Dante Controller. Unlike all other equipment, the Audio Technica ATND microphones do not have a LED panel or control software which showed the current IP address needed to enroll the device in a domain. To solve this, I had to find the MAC address on each device and cross reference it to a Mac terminal prompt of "arp -a" which listed MAC addresses and their associated IP addresses.

Another challenge presented itself during setup of the AM2 headphone amplifiers (and a little later the PCIeNX) where they dropped from recognition on the network for nearly an hour. All equipment showed power and network connection via their LED indicators, but none were routable or controllable inside Controller. After no success I backtracked all steps and found the solution to the issue to make no logical sense: one unrelated piece of equipment also needed to be present on the network. Earlier in the day, I had tried solving the distortion and drop out issue by swapping the Ferrofish in Studio E with the secondary Ferrofish borrowed from campus. Though that turned out to not be the cause of the distortion, I found that for some reason, removing the first Ferrofish from the network caused all the other equipment to not be recognized in Dante Controller. As soon as it was plugged back into the network, all other equipment returned in Controller. I was able to recreate this problem and it maintained the same solution, but never saw a reason for this to happen.

The Final Session (03/06/2024)

The morning of the session there were still a few things to finish, but the biggest task was to work with the crew on what their roles would look like in this atypical workflow. For the session day itself there were five crew members, three musicians, Michael Hanson, and myself.

NAME	STUDIO	ROLE
SADIKA ANDERSON	D	PCIeNX Multitrack and Headphone Mixes
CHRISTIAN HANNAH	All	Video Documentation
DANNY MALONEY	C	Live-to-2mix
DALE SHACKLEFORD	All	Producer
MARK SMITH	D	Drum recording
AARON WALDEN	E	Bass and Guitar Recording

Table 12: The Final Session Crew List

NAME	STUDIO	ROLE
JOE BASS	E	Bass
JASCO DUENDE	E	Guitar
MATTHEW KEEGAN	D	Drummer

Table 13: The Final Session Musicians List

I spent time with each of the crew members discussing what their roles looked like and what was expected of them. Mark Smith would not record but would craft the sound of the drums through a console’s channel path before it went to Dante. Aaron

Walden would focus on recording the guitar and bass via the console (and outboard gear) and would also record the input from Dante of the drums (though I suspect the drums to be delayed by 12ms in this recording). Danny Maloney’s purpose in C was to create live mixes of all the multitracks come from both other studios. Sadika recorded the multitracks on the PCIeNX card while also interacting with the musicians to ensure they had the best possible headphone mixes. Christian Hannah wandered between studios taking behind the scenes footage of the event. Each of the members of the crew did excellent jobs, I could not have been happier. After the session, the musicians even went out of their way to compliment the headphone mixes.

Even though I set up the drums and placed the microphones myself, I wanted each of the engineers present to have their own creative input on the process. In a conversation with both Mark Smith and Aaron Walden, I told them that they could use whatever processing “to tape” they wished in order to put their mark on the project “as long as it sounds good”. For the bass and guitar, I instructed Aaron to choose and set up whatever microphones he thought would work best with their amps and instruments being brought in. I did not look over their shoulders during this process and do not know what the final microphone and processing choices were made as sonically I was happy with what was coming out of the speakers. However, below you will find the mic list for the drums I set up the day before.

INSTRUMENT	MICROPHONE	NOTES
KICK	Beta 52a	
KICK OUT	MD 421	
SNARE TOP	Sm57	
SNARE BOTTOM	SM7b	No wind sock
HIGH HAT	C451	-10 Pad; 75Hz roll-off
TOM 1-3	E604	
OVERHEADS	KSM44	Carioid; -15 pad; wide spaced pair
ROOM	QTC30	Spaced pair

Table 14: The Final Session Drum Mic List

Upon arrival in the morning, the issue with C not receiving the inputs was quickly resolved. All devices were registered in the correct domain but were not in a shared clocking group. Additionally with C, because we had to switch to Dante Virtual Soundcard, it was necessary to create an aggregate IO for the host Mac computer to be able to also output to the console for monitoring. Initially the idea was to create an all-analog board mix, however, the aggregate IO did not wish to play nicely with the console, and we opted to mix in the box with only a stereo feed going to the console for monitoring.

All Dante devices functioned properly without issue except for one talkback microphone placed in Studio C. The ATND microphone was routed properly and could be heard in Studio C when tested locally, but for whatever reason did not transmit any audio to any other location. We attempted to fix this issue but ran out of time before the session began.

With the time it took to explain how the session was going to operate, there were several things that made the minutes right before the session feel rushed such as setting the final microphones for the bass and guitar, ensuring everyone got to the proper locations, and testing talk backs which was difficult to accomplish while alone the day before. However, once the session began all ran smoothly. There was approximately 30 minutes of non-stop recording with no punch ins, and the session only halted briefly twice. Once to swap a suddenly noisy pre-amp, and the second because the guitar channel sporadically cut in and out for 4-5 seconds. By the time we were aware of what was happening the issue passed and never reoccurred.

During the last ten minutes of the session, individuals from MTSU’s IT department approached us during a song saying their systems were showing a network loop inside Studio D. They cited ports 102D and 102E as the culprits and said that if the problem was not resolved they would shut those ports down. The six ports were not labeled so which ports were 102D and 102E was unclear. We were able to ask for a few minutes to wrap up the session which they graciously gave us while they attempted to find out more about the port configuration. The song continued being performed, and the session did not stop. Right after the session ended, the IT employees returned and said they had closed the ports. However, it turns out that neither of the ports having issues were ports we were using on the session.

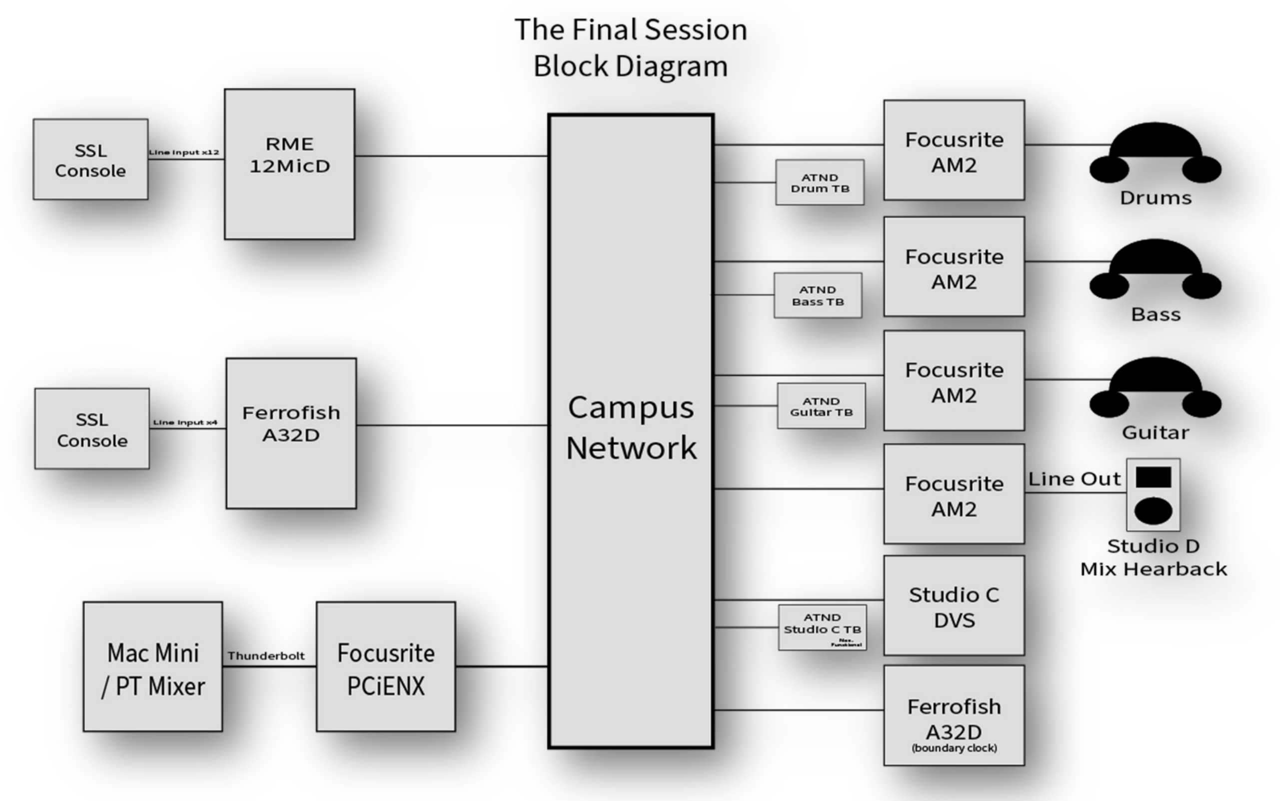

Figure 14: The Final Session Block Diagram

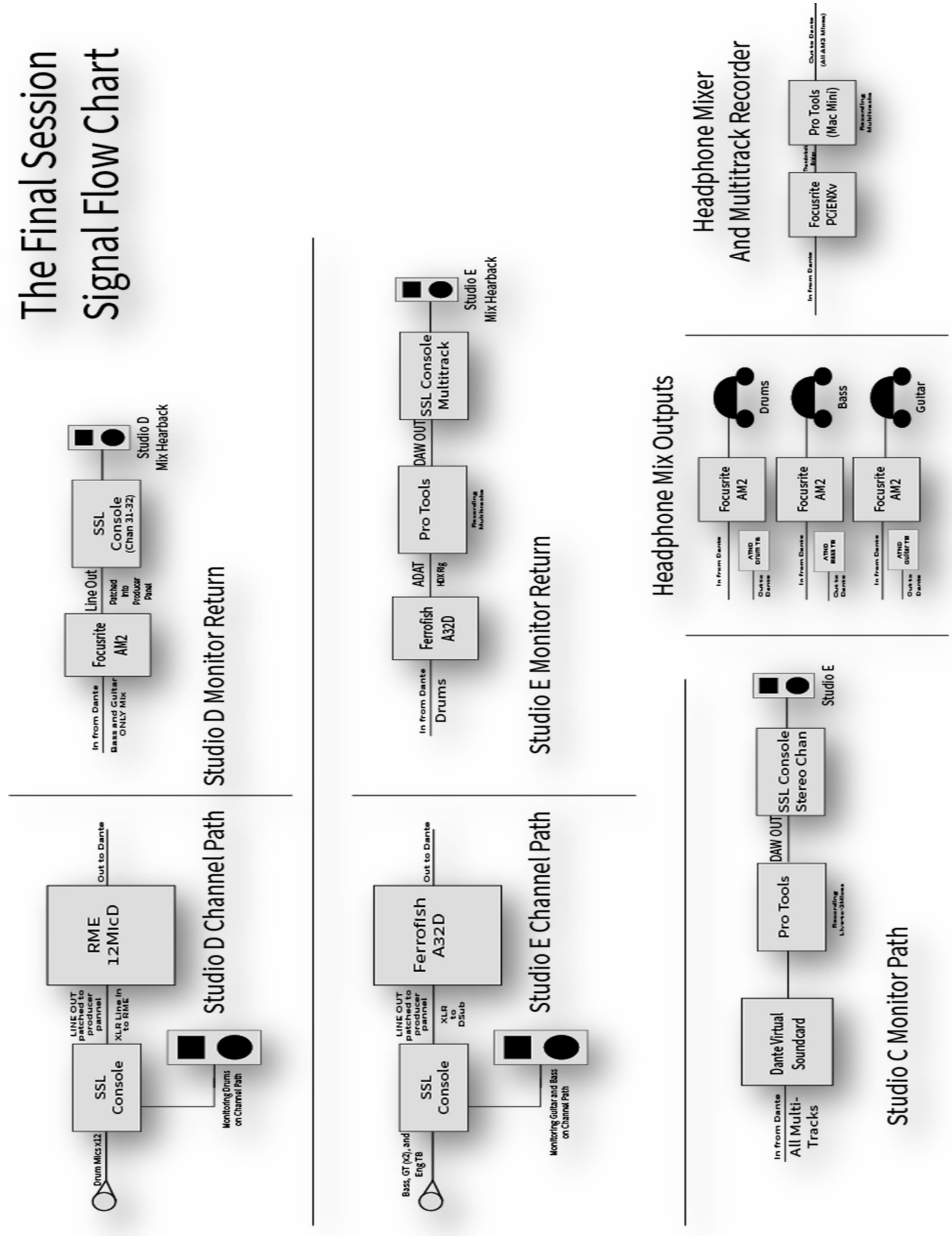

Figure 15: the Final Session Signal Flow Chart

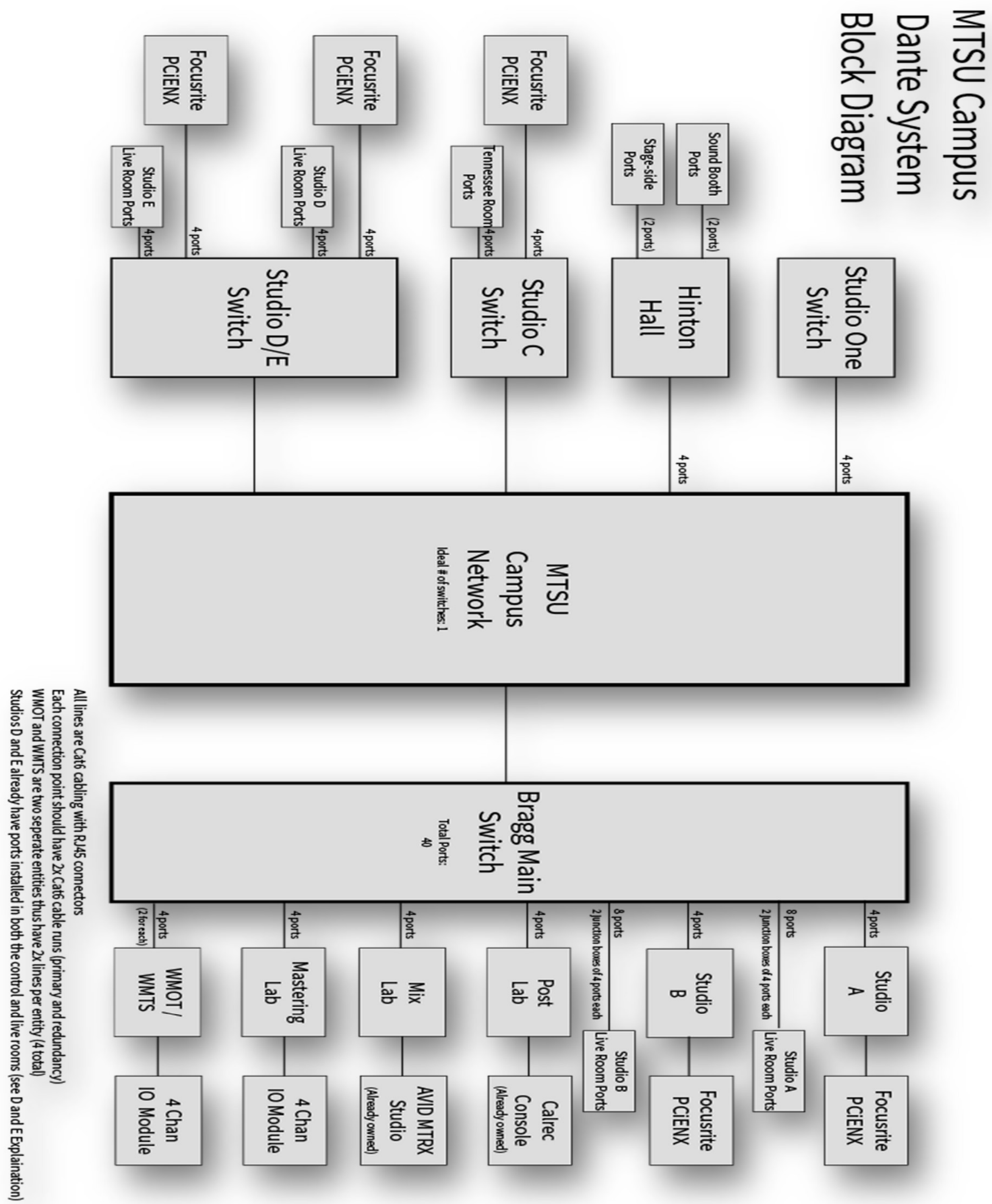

Figure 16: MTSU Designs Block Diagram

The system designs for MTSU are simple. As seen in the research analysis, the more switches or network points that a signal must pass through to reach its destination the longer the latency and more propensity to have issues. Also as seen in the research above, latency must be below the Hass Effect line for musicians to be able to accurately perform synchronously with each other. These principles lead to the simplistic design and a *strong* recommendation for MTSU to dedicate network connections between each of the desired working locations via one switch. Again, each switch encountered adds approximately 1ms of latency, and the latency must be under 10ms *at a maximum*, with less being far preferred. If outboard gear is to be used (such as in The Final Session), then the allowable time for network induced latency must be reduced even farther to allow for analog to digital and digital to analog conversion processes.

Inside this design diagram you will find two types of endpoints: recommended gear and open-ended network ports. Open ports are for expansion or removable gear (such as headphone amplifiers or special equipment). The gear recommended in the diagram is for installation purposes.

Inside the Post Lab and the Mix Lab Middle Tennessee State University already owns two pieces of installed equipment that can be added to this network. Other equipment I recommend for

MTSU’s facilities are Focusrite RedNet PCIeNX cards, some form of 4 channel breakout boxes (i.e. Four Audio D8S1 or Arista ARS-0202-A00), and a small handful of 1GB unmanaged switches without EEE that can be moved as needed.

Focusrite RedNet PCIeNX cards are capable of 128x128 channels of 24bit audio up to 192kHz sampling rate, which will be more than enough for MTSU student projects. For the creative aspect of my project, I was able to use and test one which proved to be very beneficial. For each Studio at MTSU I recommend adding one to a thunderbolt chassis which can be connected to the host computer and run as the playback engine in Pro Tools. The PCIeNX cards feature extremely low latency which is crucial for this installation. With the use of these cards, however, comes the need to create new hybrid Aggregate Devices inside the Mac ecosystem to capitalize on both the Dante channels and each room’s analog consoles. This is currently the optimal solution to adding Dante capabilities inside the studios.

The second option for adding Dante to the existing infrastructure is to take advantage of the ADAT i/o available on the Pro Tools HDX systems inside the studios. This option has several drawbacks, however. First, latency becomes an issue with the need for the extra D/D conversion process from Dante into

ADAT into the digital forms used by the host computer. As demonstrated in The Final Session, this conversion process added upwards to 12ms of latency. Other drawbacks include that ADAT optical is only capable of eight channels per cable (and less at higher sample rates). The HDX systems inside MTSU studios at maximum have two available ADAT connection points for each input and output. While 16 channels may be enough, for larger sessions that could easily fill up quickly. This option is also generally more expensive than the first as there are no dedicated Dante to ADAT converters, only multipurpose devices such as the Ferrofish A32D or the DAD AX Center. Though with this option there would be no need for an aggregate device, there would be a need to change the I/O settings inside Pro Tools from Analog inputs to ADAT inputs along with the system clocking settings (see The Final Session). While this is not a major issue in itself, it requires more steps than changing the playback engine and could be confusing or even session breaking for new students who do not understand these principles or realize the need to look in those sub-menus.

For the Mastering Lab, WMOT, and WMTS I recommend purchasing a four-channel analog input device such as Four Audio’s D8S1 or Arista’s ARS-0202-A00. Other possible recommendations should these no longer be available are AEQ’s NetBOX 4 MH (though this requires PC-only control software) or

RapcoHorizon’s Wall Plate. All of these are simple devices that allow for two channels of XLR inputs and two channels of XLR outputs; enough for stereo mixes to be sent to and from each room. All of these allow for Line Level inputs, some of them allow for microphone level inputs as well and feature control software to make adjustments as needed. This addition will make WMTS and WMOT able to accept feeds from anywhere on the network allowing them to broadcast live performances or events. This addition also turns the Mastering Lab into a viable space for remote mixing of live events and broadcasts.

Studio C has no live room. However, next to it in the designs you will see “The Tennessee Room”. This is a venue space in the halls above that can be used as a recording location, though seldom taken advantage of. Adding Dante capabilities to this room will turn it into a more easily accessible and viable space to track music in, making Studio C’s value greater. Adding a PCIeNX card and networking into C’s infrastructure will also add value turning the Studio into a place perfect for remote broadcast mixing.

As seen in the designs, each location that features installed equipment should also have two separate Cat6 lines installed to it, serving two major purposes. The first being lines for the redundancy ports available on most Dante equipment which operate on the incase of network failure. The second reason is for future expandability. In the designs, some rooms have four ports listed, which are for both the above stated reasons and for the ability to connect any additional gear needed for a particular session that are not a part of the installation.

Studios D and E already feature network ports that are connected to MTSU's primary network. These are what were utilized for The Final Session. I recommend that as many of these ports as are possible be moved to a dedicated network and that new ports be added to make a combined four ports in each Control Room and either four or eight ports be available in each Live Room. There are currently four ports already available in the live rooms of Studios D and E that can be repurposed to fill this need while Studios A, B, and C will need fresh installations. Eight ports are recommended in Studios A and B's live rooms due to their size and the likeliness that larger session take place in these rooms over that of D and E. Should the university decide to invest in Dante headphone amplifiers (like the Focusrite RedNet AM2 used in this project or Yamaha's MLA200) these ports could be filled quickly. Because of this, I also recommend the purchase of small (either four or eight port) unmanaged switches, in which EEE can be turned off, that can be placed in the live rooms for additional ports as needed.

SECTION 7: CREATIVE PROJECT CONCLUSION

If asked about Dante or networked audio two years ago I would have been able to say nothing more than it existed. Though not exhaustive, this project has built a foundation of knowledge on the topic that has aided me while opening the door of opportunity for MTSU to see the value of networked audio on campus. This project serves as proof-of-concept that MTSU is currently equipped with a network that is able to utilize Dante on campus with certain technical restrictions. With additional infrastructure such as equipment and dedicated network lines, it could expand its' abilities greatly.

Inside this project you will find the basic information needed to understand how AoIP works underneath the hood. It will give you the knowledge required to begin designing small format networks, will show you the viability of the research through a live case study, and will provide you with working designs which can be implemented by MTSU should they choose to do so. This project crosses over the lines of the theoretical and enters the realm of practicality both for myself and for the University's advantage.

It is my hope to see Dante equipment spread throughout the studios and live spaces on campus to facilitate the ability for greater sessions and projects. It is also my hope that with networked audio expansion on campus, Audio-Over-IP can be taught to new students as it possesses a prominent place in the professional world. It would better equip students for success and new opportunities as even more of the industry pushes into using networks for audio transmission. The study of this topic has already aided me numerous times in unrelated scenarios outside of this project in the professional realm, and I foresee it continuing to be of personal benefit as time goes on.

I am pleased with the time spent studying Dante technology. I am more pleased with learning the ease of which so many new workflows can be achieved that allow for innovative ways to work audio. Without Dante I would never have been able to accomplish projects like the Christmas Livestream, much less the inter-studio sessions. Once an understanding was held of how to use Dante, it made all other projects considerably easier than they would have been without it. Though I know that networked equipment is not always available, it has become an option that I will likely choose whenever possible for its ease of use and flexibility.

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APPENDIX

THE FIRST TEST

Figure 17: Setup for first at home test

Figure 18: AM2 headphone amp used in first test

LAUREN GUNN RECORDING

Figure 19: pictured Dale Shackleford (left) and Lauren Gunn (right) during video shoot

Credits:	Role
Lauren Gunn	Artist/Singer/Performer
Dale Shackleford	Producer/Engineer

Table 15: Lauren Gunn Recording Credits List

JAZZ LOCATION RECORDING

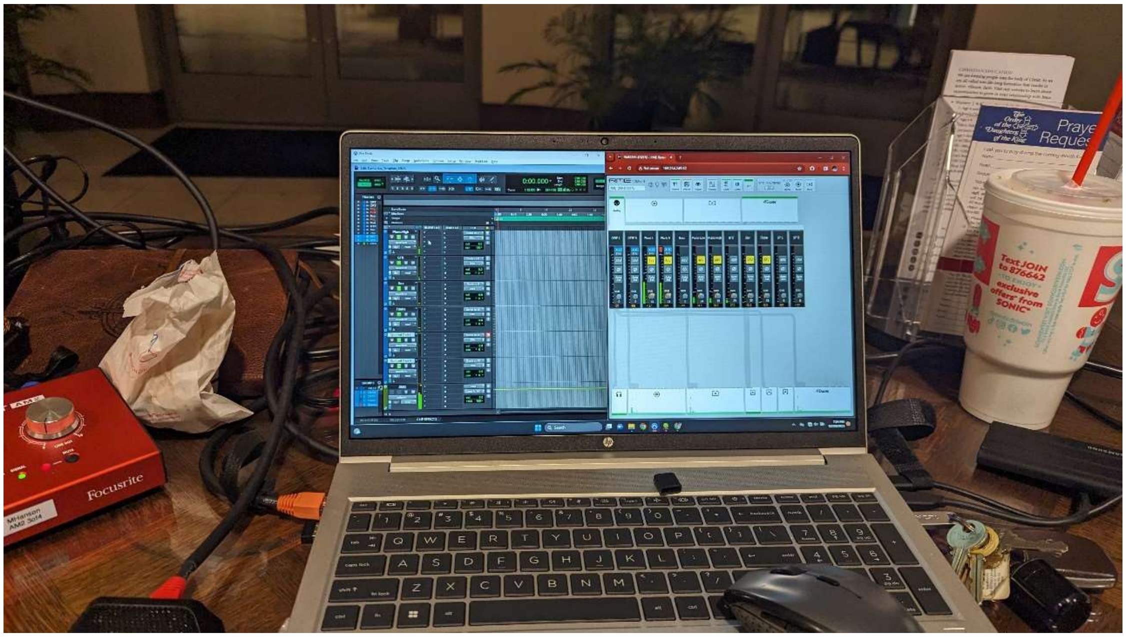

Figure 20: Mobile recording rig for jazz session. Pro Tools and RME network control web page being used.

Figure 21: Mobile rig recording setup

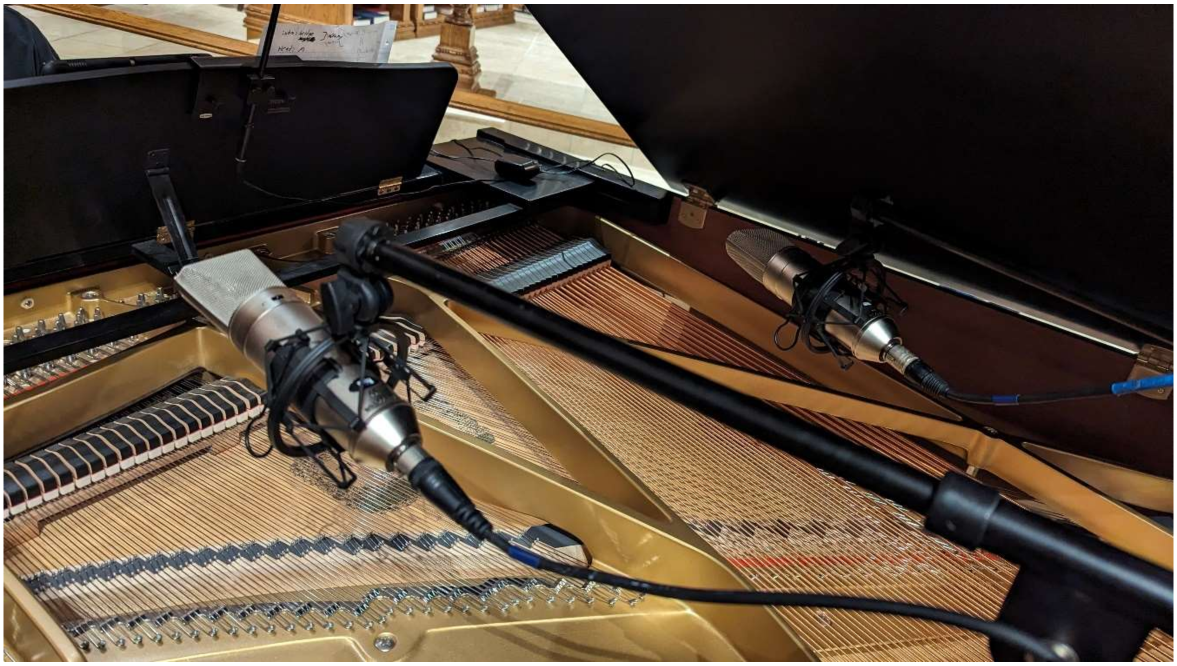

Figure 22: Piano micing for jazz session

Figure 23: Saxophone mic placement. Graham Broome (left), James Orme (right)

Figure 24: Ferrofish and Cisco CBS350 network switch

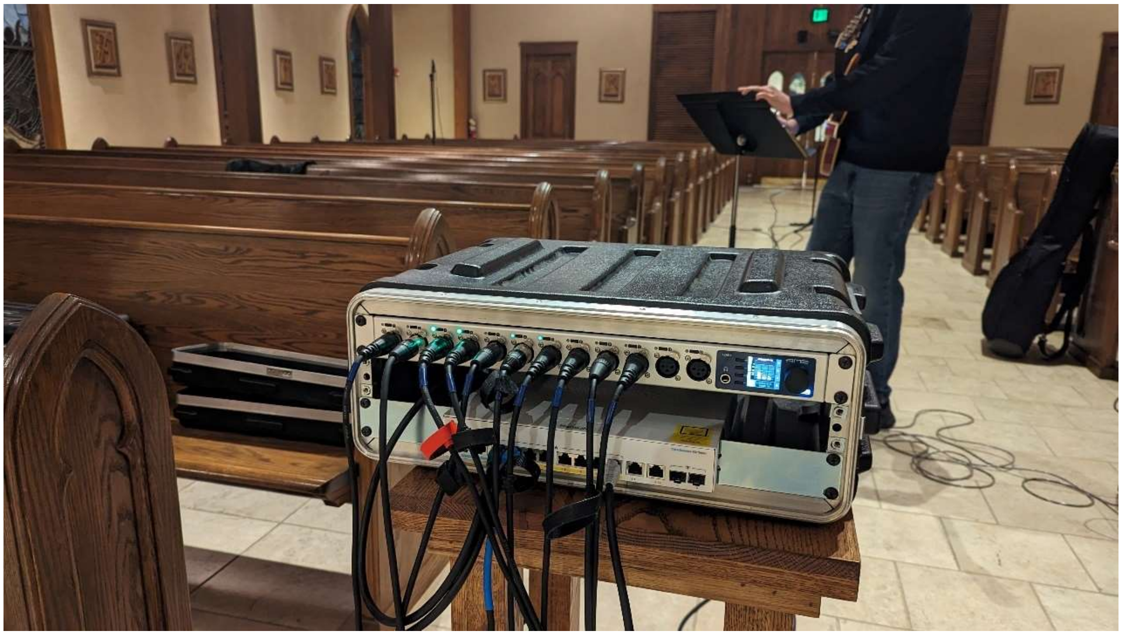

Figure 25: RME 12MicD microphone inputs, and Cisco CBS350. Pictured: Daniel Mazur

Credits:	Role
James Orme	Bass
Daniel Mazur	Guitar
Kenneth Jiang	Piano
Ryan Hungerpiller	Trombone
Graham Broome	Saxophone
Dale Shackleford	Producer/Engineer
Tevin Turner	Engineer

Table 16: Jazz Location Recording Credits List

HINTON HALL SWITCH TEST

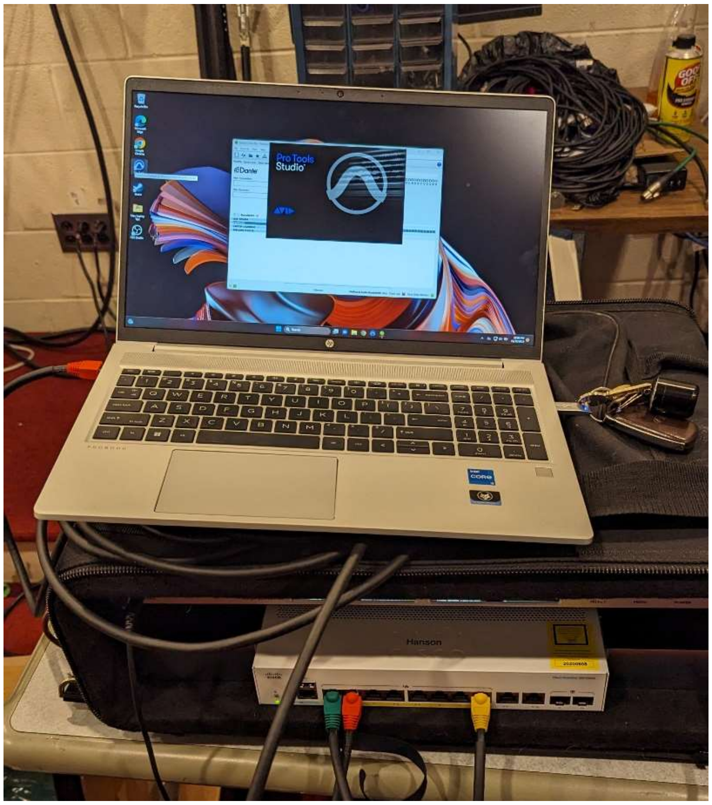

Figure 26: Mobile Pro Tools rig

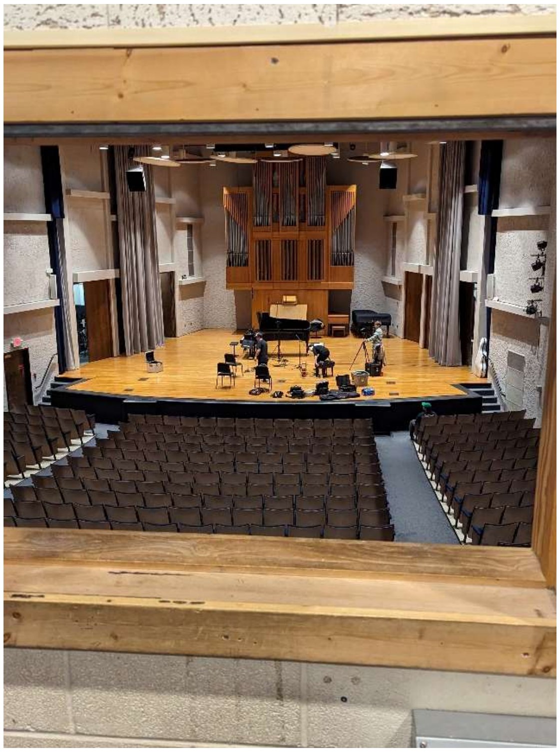

Figure 27: View from Hinton Hall's sound booth

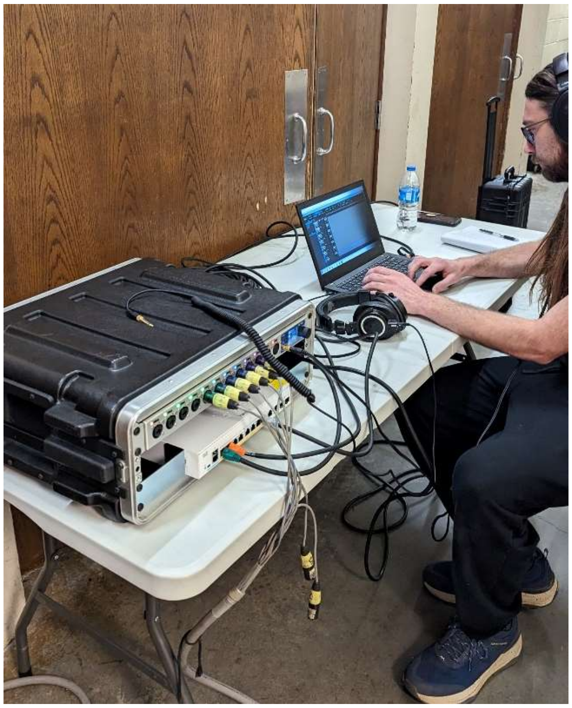

Figure 28: Robbie Dunham working with the RME 12MicD

TENNESSEE VALLEY WINDS

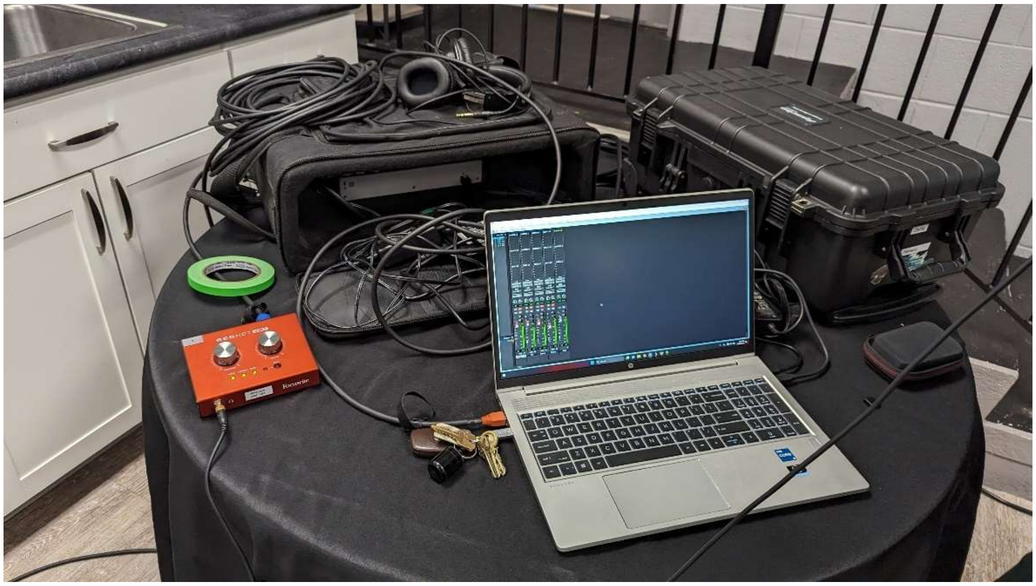

Figure 29: Mobile recording rig set up in Washington Theatre's green room

Figure 30: The Tennessee Valley Winds preparing to perform

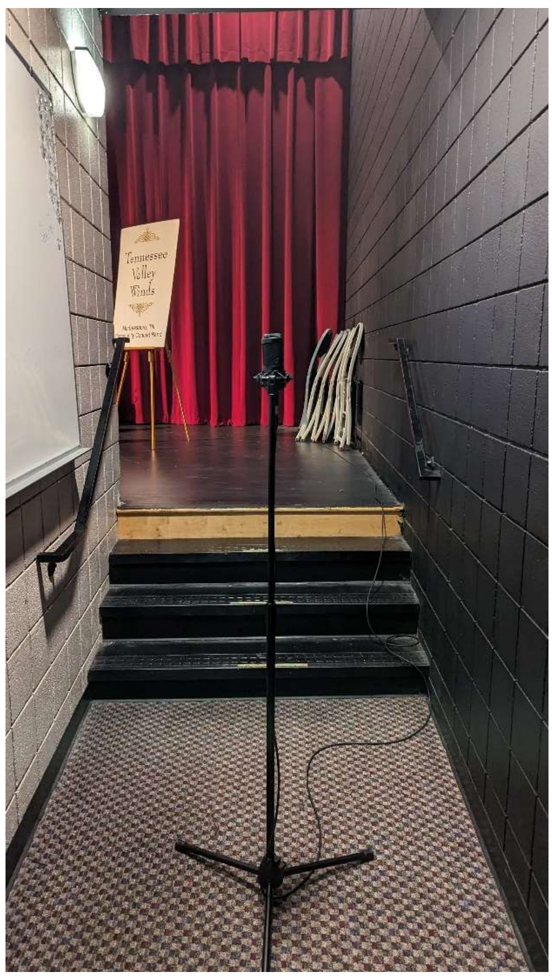

Figure 31: Hall microphone

Figure 32: XY supplemental microphones

Credits:	Role
Tennessee Valley Winds	Performing Ensemble
Dale Shackleford	Producer/Engineer
Mac Palao	Engineer

Table 17: Tennessee Valley Winds Credits List

CHRISTMAS LIVESTREAM

Figure 33: Dale Shackleford's point of view during Christmas Livestream. Pictured: Patrick Glover (left) and David Overstreet (right)

Figure 34: Studio view of livestream setup. Pictured: Patrick Glover (left) and David Overstreet (right)

Figure 35: all three computers and workstation used for Christmas Livestream

Figure 36: RME 12MicD and Cisco CBS350 switch inputs for livestream

Credits:	Role
Lauren Gunn	Performer
Luke Robins	Performer
Misty Ecker	Performer
Patrick Glover	Content Producer/On-air Host
David Overstreet	Content Producer/On-air Host/Performer
Dale Shackleford	Technical Producer/Engineer/On-air Host/Perfomer

Table 18: Christmas Livestream Credits List

THE FINAL SESSION

Figure 37: Final project primary advisor Michael Hanson (left) and Dale Shackleford (right)

Figure 38: Final project technical crew. Pictured from left to right: Danny Maloney, Mark Smith, Dale Shackleford, Aaron Walden, Sadika Anderson, and Christian Hannah

Figure 39: view of PCIeNX rig and into Studio D's live room from Studio D Control Room

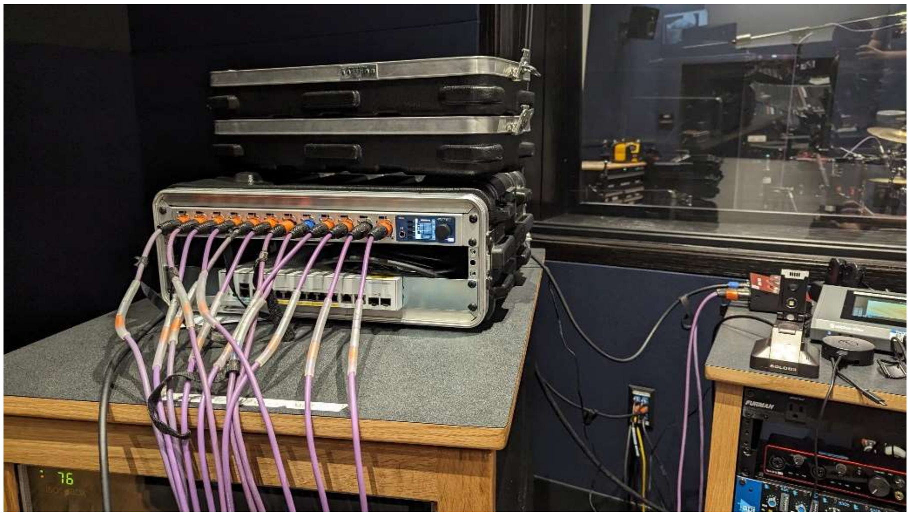

Figure 40: RME 12MicD inputs during Final Project Session

Figure 41: Mark Smith working on Studio D's SSL console for Final Project Session

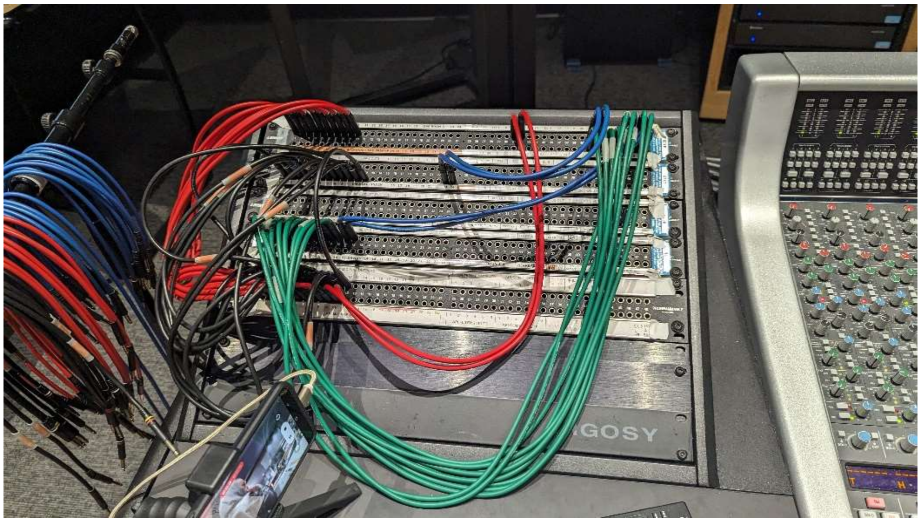

Figure 42: Studio D's patch bay during Final Project Session

Figure 43: Final Project Session players and producer. Pictured from left to right: Joe Bass, Dale Shackleford, Jasco Duende, and Matthew Keegan

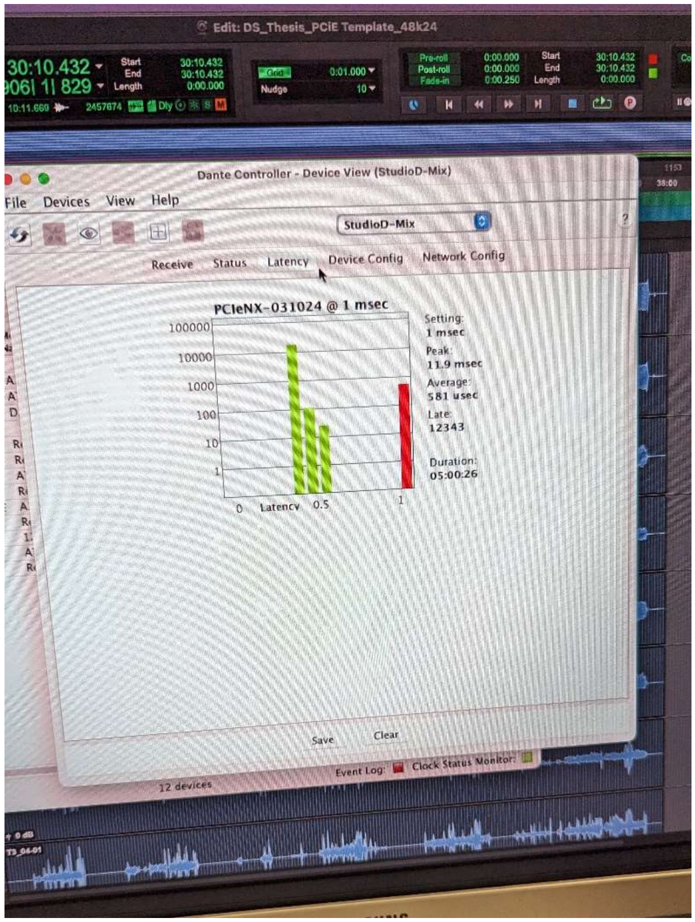

Figure 44: Dante Controller latency statistics after Final Project Session

Figure 45: View from Studio E's Control Room into Studio E's Live Room. Pictured: Jasco Duende (left), and Joe Bass (right)

Credits:	Role
Matthew Keegan	Drummer
Joe Bass	Bass

Jasco Duende	Guitar
Sadika Anderson	Engineer/Headphone Mixer
Aaron Walden	Studio E Engineer (Guitar/Bass)
Mark Smith	Studio D Engineer (Drums)
Danny Maloney	Studio C Engineer (Live-to-2Mix)
Christian Hannah	Behind the Scenes Video Operator
Dale Shackleford	Producer
Michael Hanson	Advisor

Table 19: The Final Session Credits List