



Audio Engineering Society

Convention Express

Paper 370

Presented at the AES 159th Convention
2025 October 23-25, Long Beach, CA, USA

This Express Paper was selected on the basis of a submitted synopsis that has been peer-reviewed by at least two qualified anonymous reviewers. The complete manuscript was not peer reviewed. This Express Paper has been reproduced from the author's advance manuscript without editing, corrections, or consideration by the Review Board. The AES takes no responsibility for the contents. This paper is available in the AES E-Library (<http://www.aes.org/e-lib>) all rights reserved. Reproduction of this paper, or any portion thereof, is not permitted without direct permission from the Journal of the Audio Engineering Society.

Sound Reinforcement System Design for Multi-Purpose Assembly and Performance Spaces

James Krumhansl¹ and Joanne Chang²

¹ DLR Group, 1422 Euclid Avenue Suite 1300, Cleveland, Ohio 44115

² DLR Group, 51 University St #600, Seattle, Washington 98101

Correspondence should be addressed to James Krumhansl (jkrumhansl@dlrgroup.com) and Joanne Chang (jchang@dlrgroup.com)

ABSTRACT

In the past 20 years, there has been a substantial change in the technology available for sound reinforcement solutions. These changes have impacted the accessibility and types of solutions for deployment in common community spaces, whether they are academic, civic, or privately owned. Lessons learned and new features in the more recent innovations have changed deployment and ownership of these systems as well as how they are used and maintained by the average end users, advanced users, and technicians for multi-use facilities. In addition to the performance of the primary reinforcement system, there have been increased mandates and requirements to meet the needs of the hearing impaired as a required supplement to the primary reinforcement system, which drives peripheral requirements to augment those systems to make them inclusive to all patrons of a space. As new practices of design, implementation, and use are presented, some fundamental concepts are still relevant and acknowledged with the new paradigm sound system reinforcement. Subject matter will cover a range of topics including consideration for system components, considerations for physical integration, addressing program needs and room acoustics needs in system design, and interaction/coordination with other facilities systems for a holistic approach to the deployment of a sound reinforcement system with an integrated design approach.

1 Rise in Need for Multi-Purpose Venues

Various types of activities may occur in an assembly or performance space. The common use types include full student, staff, or organization assembly, graduation or award ceremony, school performances (band, orchestra, choir, theatre/drama, musical, and dance), touring performances, keynote and presentation, lectures, panel discussion, cultural and community events, and religious services. With changes in the economic environment and increasingly raise of sustainable awareness, facility

owners and managers are seeking to increase revenues and maximizing venue usages and utilization rates. The cost escalation and inflation of construction and equipment also have pushed facilities, including new construction and renovation, to optimize the use of spaces. Over the years, assembly spaces and event venues have grown in favor of having the capability to serve and support multipurpose functions, and to attract different types of event presenters. It also allows supporting diverse activities of the communality around the venue geographic location.

Spaces designed to be configurable is often the response to the multi-use needs. The configurability may be presented in flexible seating layouts with loose gangable seats or retractable seats, or by introducing operable room partitions to allow a space to be divided into multiple smaller spaces or be used as one large space, or a combination of both flexible seating and operable partition features. Variable room acoustics, including passive and active means, is another method often employed for multipurpose spaces to support use types requiring different room acoustics parameters. Passive variable acoustics utilizes retractable curtains, banners, or panels to alter the room's reverberation time. In general, deploying curtains and banners to "deaden" the room or use panels to reduce the room volume for lower reverberation time that are essential to support clear speech and communication, such as lecture or sermon uses. Active variable acoustics or electro-acoustic enhancement systems use electronics comprise microphones, loudspeakers, and digital signal processing to create artificial reverberation for increased room reverberation times or enhanced acoustic characteristics. The defined set of target use types of the space will inform room acoustics design criteria.

Sound reinforcement system to support different room arrangements and uses will need to be versatile, allow varying configurations and setups, which may include audio zoning, loudspeaker reconfiguration, system presets, automation, flexible connectivity and signal routing system, flexible control positions, and integrated control system and interface, etc.

2 Challenges with Operation and Ownership

There are often challenges in managing and operating multi-functional spaces. Knowledge of different types of uses, staff skills, personnel and physical asset management, planning and scheduling, operating cost, and maintenance are the core requirements and factors involved in managing assembly and performance spaces.

Audio systems and equipment, especially for professional level products, require specialized knowledge and training to operate, troubleshoot and maintain. However, audiovisual is often thought to be a division of the information technology under the big technology umbrella by many organizations. And

in many cases, the audiovisual systems are managed, maintained, and operated by the IT/AV department with staff mostly trained in information technology and have little audio related background. The skill gaps create difficulty and issues in operating and maintaining professional audio systems.

Different types of uses call for various degrees of system complexity, which require different levels of technical expertise for the staff and operators, as well as different crew sizes. A simple presentation use may require only a few microphones and program audio sources to the main PA system, which may be operated with a simple self-serving system, or by one or two personnels with minimum technical proficiency; while a live music performance will require managing, mixing, and balancing multiple wired and wireless microphones, monitoring systems, and loudspeaker systems, for which staff having related skills and experiences, and knowledge of various aspects of audio would be essential. Events with complex system needs often entail multiple skilled staff and operators for effective and smooth operation. Determining and staffing required experiences and skill levels of personnels, and appropriate staff size are critical to successfully run venues serving a wide range of use types.

Challenges also have been observed in planning and scheduling. The complex aspects of operating multi-functional spaces involve planning and arranging the necessary equipment setups to meet each event's needs, suitable crew skills and crew size for the required tasks to set up and run the events, as well as the changeover time between events. Knowledge of the nature of various event types, such as how different types of events proceed, what coordination and support are needed, what timeframe and sequences for different tasks are required, what technologies are involved, and what skill sets are vital for the crew members to accomplish these, etc. are important to support effective managing of this aspect in addition to typical required management skills.

Highly skilled personnels and large staff/crew size often accompany with higher operation costs. Thus, define and determine the goals, and balance between the initial cost of ownership and operating costs are among the key elements to evaluate in facility planning and system design for multi-use spaces.

3 Changes in Hardware Implementation

Development of loudspeaker technology and products over the years has supported different

implementation of the system design for multiple functional space. The loudspeaker types may be categorized by their sound radiating behavior, point source and line source; by their function and physical appearance, single cabinet and modular multi-cabinet array, column array and curved array; and by their integrated components, processing loudspeaker and passive/non-processing loudspeaker. In general, point source loudspeakers, typically single cabinets, are best for smaller spaces and focused areas; and line source, multi-cabinet arrays, serve better for large areas. Column arrays have slim and clean appearance and are often desirable aesthetically integrating with interior design. And processing loudspeakers provide added features of digitally steering the sound beam to optimized coverage without physically tilting the cabinet, as well as allowing various beam size and steering configurations to be saved and recalled for different uses.

Deployment of the loudspeaker system design may be portable or permanently installed. Portable loudspeakers that are brought-in and set up per events may be utilized for spaces where permanently installed loudspeakers are undesirable for the architecture and interior design, unfeasible structurally, or due to the business or operating model determined by the owner. It is typically carried out with loudspeakers on stands or multiple rigging points throughout the space that must be planned and provided. Permanently installed loudspeakers are restricted to their fixed positions and configurations. In multi-functional spaces, the design practice generally involves defining a set of use types and room configurations that the space is targeted for and utilizing loudspeaker zoning, signal processing, and system presets to serve these functions.

With the maturity of networked audio technology and growing product adaptation, the network addressable loudspeakers have been developed and deployed in the market by many manufacturers. The ability to modify audio routed to each loudspeaker individually via network increases the arrangement flexibility of audio zoning, compared to the traditional method of predefined loudspeakers grouping fed by assigned amplifier channels. For processing loudspeaker and self-powered line-array loudspeaker products enabled with networked audio features, the cabling requirements are minimized, often reduced to a single Category cable that delivers both audio and control signals to the entire array, and between array cabinets.

Many manufacturers offer loudspeaker solutions with integrated signal processing to optimize performance. It may be presented in the form of self-powered loudspeakers with on-board DSP, or outboard DSP amplifiers, or processing settings for third-party DSP's. Loudspeaker types are available in point source, modular line-arrays, and column array loudspeakers. Some manufacturers advance their solution offering intelligent software that permits adjusting vertical coverage electronically and some with added feature for definable exclusion areas to enable more precise sound delivery control and automating the setup process. These features provide flexibility and ease of adjusting needed coverage for configurable spaces.

The introduction of modular beam-steerable column loudspeakers in the past few years fills a market gap for the processing loudspeakers. It combines digitally steerable features with modular cabinets to form different lengths of vertical column array for the throw distances and vertical beam size needed. They may be suitable for event spaces desiring capability of flexible configurations both physically and digitally, and with more aesthetically pleasing look.

Loudspeakers with variable waveguide and supporting asymmetrical pattern are beneficial to allow adjusting for different loudspeaker positions and array configurations. So that the same set of components can serve different configuration uses, which avoids needing to own and store multiple components dedicated for each room arrangement. Typically, placing waveguide in narrower horizontal coverage position for far throw and wider horizontal coverage for near throw. And in conjunction with asymmetrical waveguide position to avoid sound energy hitting the adjacent side wall that is often undesired.

The robotic line-array loudspeakers launched in recent years add robotic elements for adjusting the horizontal coverage in each cabinet and the spay angles between array cabinets remotely, eliminating the need to take down the whole array and re-assemble for different room configurations that are typical for traditional line-array loudspeakers. It may be fitted for venues desiring minimum staff for changeovers where various room configurations can be supported with the fixed loudspeaker locations and recalling the presets of the loudspeaker configurations. The design considerations will need to account for higher initial costs and heavier weights with this type of loudspeaker.

In regard to the loudspeaker mounting methods for flexibility, system design implementations include utilizing motorized rigging system, or employing motorized traveler system, in addition to typical use of hoists and strong points. The motorized rigging system allows for changing the loudspeaker elevations, while the traveler system enables reposition of the loudspeakers in plan. The combination of varied rigging methods in conjunction with different loudspeaker products and technologies may produce a desirable solution for the venues.

Modern electronic design aims for higher efficiency in response to pursuit of economic operational costs and environmental sustainability. In amplifiers, this mostly includes higher ratios of power to loudspeaker vs waste heat, and lower power draw in idle and stand-by, which have been seen increasingly in the recent product offerings. Furthermore, features of flexible power outputs admitting diverse use and distribution of the total power to different channels of the amplifier, as well as mixed impedance loads that allows serving both low-impedance (4-ohm/8-ohm) and high-impedance (70V/100V) on different channels of the same unit and freedom of assignment, provide system design flexibility with cost and energy efficiency benefits.

Digital signal processing, as previously mentioned, has been used for optimizing loudspeaker systems. It enhances signal quality and system efficiency by performing multiple types of signal processing in digital domain within one unit. Unlike traditional analog systems, in which multiple units are used for different types of signal processing adding noise to the signal chain. Open architectural digital signal processors are especially advantageous for complex system tuning and operational efficiency, as the designers and users can create custom configurations as needed with the software processing blocks for signal routing and processing. Additionally, multiple configurations and settings can be programed, stored, and recalled for different applications.

The major impact from the raise and wide adaption of the networked audio technology is the inclusion of networking hardware as the major audio equipment, and the requirements and restrictions associated with network technology standards. Networked audio requires high performance network switches, typically Layer 2 or 3, managed switches that support 10/100/1000 Ethernet standards, and with features including VLANs (Virtual Local Area Network),

QoS (Quality of Service), multicast filtering and STP (Spanning-Tree Protocol), etc. [1] However, the Energy Efficient Ethernet [2] (aka “Green Ethernet”) feature in network switches has been found to cause audio interruptions or dropouts as well as adding latency and is recommended to be disabled to ensure consistent system performance. Networked audio transports digital audio over standard Ethernet network and thus adhere to the IEEE and ANSI/TIA standards for Category 5/5e/6/6a cabling distance limitation of 100-meter [3]. Therefore, attention and careful planning must be given in designing infrastructure and connectivity system.

Devices connected over the Ethernet network can take advantage of PoE (Power over Ethernet) technology to simplify installation eliminating the needs of local power. In addition to cabling distance limitation, calculating PoE budget to select appropriate network switches is critical when designing a system that contains multiple PoE devices. Factors in determining PoE budget include quantity of connected devices, PoE class [4] and maximum power consumption of each device, and buffer for power loss in cables. With larger size or quantity of network switches in the system and that the network switches supply power to the connected devices, requirements for electrical and mechanical to support the needed power and heat generated in the equipment room also increase. The same applies to the power capacity of UPS (Uninterruptible Power Supply) necessary for the systems. UPS offers protection from voltage fluctuations and electrical noise filtering that is recommended for today’s audio electronics, typically for the digital mixing system, DSP’s, and network switchers. It also allows the technical staff to save work and properly shut down the system equipment during power outage.

Another design consideration is adapting telecom design practice in cabling organization and interconnecting method. Traditionally, audio cabling is terminated directly to equipment in rack and not necessarily through patchbays. Standard telecom practice, on the other hand, terminating all permanent cables to patchbays and then patch to network switches. This allows greater adaptability for reconfiguration and future expansion, also it reduces the expense of replacing network ports due to wear and tear from repeated plugging and unplugging, as the ports on patchbays are more durable and less expensive to replace than those on network switches. The amount of patchbays and patch cords and the

necessitated rack spaces will need to be accounted for the system, as well as the equipment layout in rack.

4 Migration to Audio over IP Solutions

Audio over IP (AoIP) solutions have been around for some time now, and with certain protocols taking precedence in the industry we are seeing mass adoption and integration into products both in the realm of live sound and commercial audio equipment. Convenience, cost, ease of use, reliability, redundancy, and quality of transmission are all factors to recognize when making the decision to utilize and plan for digital audio over network equipment for the interconnection of audio equipment.

One favourable feature of deploying Audio over IP is the inherent reduction in cabling infrastructure. A single category cable of the appropriate specifications can carry hundreds of channels of bi-directional audio simultaneously where historically analog audio distribution would require discrete conductors per channel of audio for transmission over a cable. This not only impacts the need for the size of cable, but the weight, and required pathways and bend radii needed to route interconnecting cabling between two devices, and over distance this can have a huge impact on facility planning when multiple permanent and temporary equipment locations are planned for. It should also be noted that the standard connector of category cabling is less complex than a multi-pin analog audio connector, which can be difficult to connect and easy to damage if not properly seated in a multi-pin receptacle.

The use of category cabling also offers an added benefit of leveraging existing infrastructure where historically modern construction will often have a distribution of category cabling from telecommunications, and in some cases even existing networking hardware can be leveraged when configured properly to allow for endpoints to connect to an existing network deployment or new segmented network on existing hardware to allow AoIP solutions to be used with minimal impact to network performance depending on the existing hardware and the needed bandwidth. Note that in some cases there will need to be proprietary hardware depending on the specific network protocol, but the cabling infrastructure is ubiquitous in the deployment of structured cabling for networks in commercial facilities. In many cases small channels counts will have a nominal impact, with bandwidth being a factor

of the quantity of channels and how they are packaged on the network depending on the protocol as well as the desired sample rate and bit depth of an audio channel.

Network data transmission will inherently have some level of anticipated latency depending on the specific protocol and network architecture. The amount of network interfaces and media conversions that occur between endpoints on the network will have greater impact on latency than traffic limited to a single piece of network hardware, so careful consideration for network architecture can have an impact on applications that require real time monitoring such as live performance vs the playback of prerecorded or live audio in systems where milliseconds of time can be perceived due to audio synchronization in the same space as a performance or a real time video feed. In mixed deployments where network audio may be utilized for local reinforcement and other distributed applications priority should be given to maintain audio traffic to a single switch to limit the amount of “network hops” than may influence latency in larger systems.

A value and functional add of audio over network is consideration for how audio is easily replicated on the network with no quality loss, providing higher quality outcomes for auxiliary audio feeds for applications such as streaming, recording, live broadcast, facility overflow, and any other application where multiple channels of audio can be routed to different systems simultaneously without added noise or signal segregation where distribution amplifiers and other analog breakout solutions would have been utilized historically. This also allows multiple systems to process individual channels as appropriate, meaning a live event can have sources mixed as appropriate for the delivery method, such as a live mix for a venue being different than a streaming mix, different for an overflow venue, and unprocessed audio from sources can be recorded before effected by any of those independent mixing solutions for greater control of post-production or review of archival recordings. Lastly, where the physical plant of the deployment allows full redundancy can be accomplished in supported topologies where maintaining signal for critical recording and live audio applications warrant replication of signal transport through reliable backup connections and hardware.

There are several other factors that go into the consideration of deployment and operation of

networked audio that may be explored directly with the manufacturer/developer of specific proprietary solutions as well as standardized protocols such as AVB [5]. For the purpose of this paper, the intent was to explore high level decision making of how audio over IP provides value in a multipurpose venue / performance space and the considerations for deployment of this modality of facility operation. While use of AoIP comes with a level of knowledge required for ongoing use and maintenance, it provides many benefits that lower facilities impact in regard to ongoing labor to support it and the required investment and density of cable infrastructure to meet the operational needs of a multi-purpose facility.

5 Implementation of Control Systems and Automation

With venues of varying sophistication rising and the need for spaces to quickly and effectively change modes of operation, the use of integrated control systems has become highly utilized in order to allow for ease of use and variety of use cases in multi-purpose venues. This often provides desired outcomes where technical event staff needs are limited to more complex event types and simple modes of presentation can be managed by non-technical users guided by simplified control routines and digital preset recall. The value in this automation also adds to the life and quality of life of end users and faculty providing controlled start up sequences and managed shut-down sequences.

Many systems manufacturers now offer touch screen interfaces that not only allow for customization of the user interface, but levels of control per user based on the Graphic User Interface (GUI) configuration and commissioning providing a simple interface for novice users while allowing the highly technical staff or “power users” to have access to more sophisticated system controls for troubleshooting and advanced use cases.

An example of the flexibility often occurs when a space can be used to host simple lecture/presentation with a single presenter only requiring 1-2 microphones and some level of control of audio of media sources that can be easily managed on a small GUI. If this same space also requires preparation for a larger event where multiple microphones and audio sources need to be mixed by a live audio engineer, the ability for a control system to change routing from a few simple sources to sending the output of an audio console can be easily accommodated within seconds

where the configuration of one system does not impact the use of the other but both can be easily used with the primary sound reinforcement system and feed the assisted listening system for either use case with little prep or physical configuration. Because the GUI is configurable with custom text and graphics these modes of configuration can be labelled and explained right on the interface minimizing the need for training or re-learning systems in a given space.

Control systems also help ease the integration of sound reinforcement with other facility systems such as life safety and paging systems. Logic interfaced with these systems can prioritize playback of external systems by muting audio routes instead of other methods such as power shutdown which leads to longer restart times or problematic system resets.

The intent of automation is to lower the complexity and labor requirements that go into preparing and operating a venue for event support. In the case of sound reinforcement automation allows for minimal management and interface with the sometime complex processes that go into routing and mixing sound sources for events to allow for quicker start up, shut down, and changeover between events to maximize the value of a facility where multiple event types can occur. While venues are still capable of operating and managing these events without these systems, the value rises as multiple modes of operation come into play through a venues seasonal schedule.

6 Meeting ADA Assistive Listening Compliance and System Augmentation for Universal Design

Per the 2020 American Disabilities Act of 2010 [6], there are three recognized types of systems for compliance which include Infrared (IR), Radio Frequency (RF), and Inductive Loops systems. Systems may be deployed as a portable system where local interfaces can be used to feed the amplified sound system, or as a permanent integrated part of the sound system. Challenges often arise due to physical and environmental limitations, and the appropriate system is selected based on these criteria by the system designer.

In instances of multi-purpose venues, the demographic of event patrons may vary significantly over time as types of events change. It is important to consider not only the solution that best performs in

the environment, but also how facilities staff will manage the required receivers as part of the compliance process.

With IR technology the signal propagation of Infrared will inherently require the receivers to be in line of sight of the emitter signal output since this wavelength would not penetrate walls of standard construction. This allows for more secure transmission of signals with added complexity when the geometry of the space is prohibitive for coverage from a single point. This also allows for many spaces in adjacency to operate without interference, and with deployment of the same system in multiple locations to allow users to go between spaces without a need to change channels on a receiver device.

RF based systems are another alternative and will offer better coverage when larger venues are in need of assisted listening provisions. Currently in the United States there are two reserved bands for these technologies allowing for interference free operation when coordinated with any of these technologies in a given area. Use of the 72MHz and 216 MHz ranges will inherently provide some benefits and tradeoffs to determine based on a given facility's needs. Lower bandwidth allows for more concurrent channels with reduced range while Higher Bandwidth provides more distance with fewer concurrent channels available in this range. Another consideration is the antennae design required for transmission where the lower frequency will typically require a much longer antennae than a higher frequency solution to achieve maximum distance.

Loop Based Systems leverage induction technology with a physical loop buried in the environment that will push current through a copper conductor which can be received natively with hearing aids with integrated tele-coil (T-coil) receivers which negates the needs for venue patrons with this feature to need a separate belt pack system to receive the assisted listening feed. While they provide great value for communities with a high population that can utilize this feature, special attention needs to be addressed during construction as metal losses due to floor construction types may impact performance of the system. A single loop will only provide a single channel of operation which also makes alternative use limited. Loops can be deployed with a single

conductor loop which minimizes cost but can introduce issues with bleed if multiple systems operate nearby. Where multiple systems occur deployment of a phased array is possible allowing for multiple adjacent systems but double the driver and conductor cabling needed to perform.

While ADA 2010 has not received any official update since its inception, technologies are currently in development which build on the original intent of the standardized solutions allowing for more feature sets and provide more agency for patrons to utilize these systems with their own devices in place of facility provided receivers. Currently there are two system types introduced in recent years intended to leverage the paradigm of the mobile device enabled end user. These solutions include Wi-fi transmission and Auracast.

Wi-fi based assisted listening solutions utilize standard 802.11 IEEE [7] network-based wireless communication to provide direct service to end user devices which have the appropriate application to monitor the assisted listening feed. While these offer some features such as multi-channel support and expansion as well as the ability to relay visual information through applications, a key caveat is that real time audio over wi-fi can be challenging even with dedicated access points so some level of latency is to be expected and should be validated based on the particular use case for this system deployment. It should also be noted that for equity among user groups personal access to mobile devices may be limited or restricted due to local customs that even though this is designed for end users with mobile devices that there should be some provision of dedicated receivers for patrons.

Auracast is the newest solution where an update to Bluetooth communications will now allow for one-to-many transmissions where historically most Bluetooth solutions are peer-to-peer based pairings. Use of these systems requires dedicated access points to broadcast the Auracast signal to Bluetooth enabled receivers managed by smart devices. With the typical broadcast range of Bluetooth solutions, we should expect that range will be limited in comparison to lower bandwidth and longer ranged RF solutions and something similar to other technologies in the 2.4Ghz space and per Bluetooth Special Interest Group is

currently designed to provide ~100m transmission distance [8]. For larger venues we should anticipate that multiple units may need to be chained together and operate together to provide both the required distance for coverage and capacity of endpoints to be deployed in a particular facility.

While this technology has been used primarily for compliance with hearing assistance for the hearing impaired, ongoing development and consideration for Universal Design has led to a variety of alternative use cases for these solutions. Use of communication tools can be used to augment and supplement user experience to allow for various populations of user groups to get the most out of their experience in a venue based on their specific needs. Some alternative use cases may include but are not limited to multi-language broadcast and descriptive narratives for the Visually Impaired. There are also other visual aid and communications related features available with these solutions, but the focus of this paper is to address how audio reinforcement and communication can be addressed in multi-purpose venues.

7 Conclusions

Ownership and Maintenance of Sound Reinforcement Systems has evolved beyond a single solution designed to meet the needs for a vast variety of users and use cases. System design requires consideration and accommodation of various stakeholders, operators, and patrons to lead to successful outcomes in large assembly and performance spaces. The physical environment is no longer the only driver in the design and decision-making process. Modern advancement in sound reinforcement technology has provided opportunities to tailor to the unique needs of a facility in any iteration of its intended use and adjusted to the variety of use cases that occur. Design of these systems has also evolved to become more inclusive of patrons and operators in the spirit of Universal Design. Thoughtful design practices ensure outcomes considerate of ownership, presenters, performers, and patrons that lead to greater satisfaction in use and flexibility in ongoing operations through the life of the facility and the equipment deployed.

References

- [1] “Implementing Dante AV-over-IP from an IT Manager’s Perspective”, Audinate, 2022.

“AoIP Network Recommendations”, Clear-Com, 2024.

https://usa.yamaha.com/products/contents/proaudio/docs/dante_network_design_guide/101_network_switches.html

<https://www.presonus.com/blogs/technical/selecting-an-avb-switch?srltid=AfmBOorQJsetfnD7qckqHU04mSxJrrbpXszMNhujfhckb-ToCDri2syX>

<https://www.ravenna-network.com/your-practical-guide-to-aes67-part-1-2/>

<https://www.ravepubs.com/choosing-the-right-network-switch-for-av-projects/>

<https://xchange.avixa.org/posts/av-networking-and-osi-model>

- [2] <https://www.getdante.com/support/faq/can-i-use-eee-energy-efficient-ethernet-or-green-ethernet-in-my-dante-network/>
- [3] IEEE 802.3 and ANSI/TIA-568.
- [4] IEEE 802.3 standards define different PoE classes and their maximum power levels. IEEE 802.3af /PoE Type 1 (Classes 0-3) at 15.4W max; IEEE 802.3at /PoE+ Type 2 (Class 4) at 30.8W max; and IEEE 802.3bt /PoE++ Type 3 (Classes 5-6) at 60W max, and Type 4 (Classes 7-8) at 100W max.
- [5] AVB consists of multiple standards including IEEE 802.1AS, IEEE 802.1Qav, IEEE 802.1Qat, IEEE 802.1BA, IEEE 1722, and IEEE 1722.1
- [6] <https://www.ada.gov/law-and-regs/design-standards/>
- [7] 802.11b/a/g/n/ac/ax/be/bn standards
- [8] <https://www.bluetooth.com/auracast/>