AUDIORAIL® TECHNOLOGIES

ADAT® rx32tx32 user manual (February 23, 2004)

4 x 2 ADAT® Lightpipes over AudioRail CAT5 daisy chain network

AudioRail Technologies

ADAT rx32tx32 user manual

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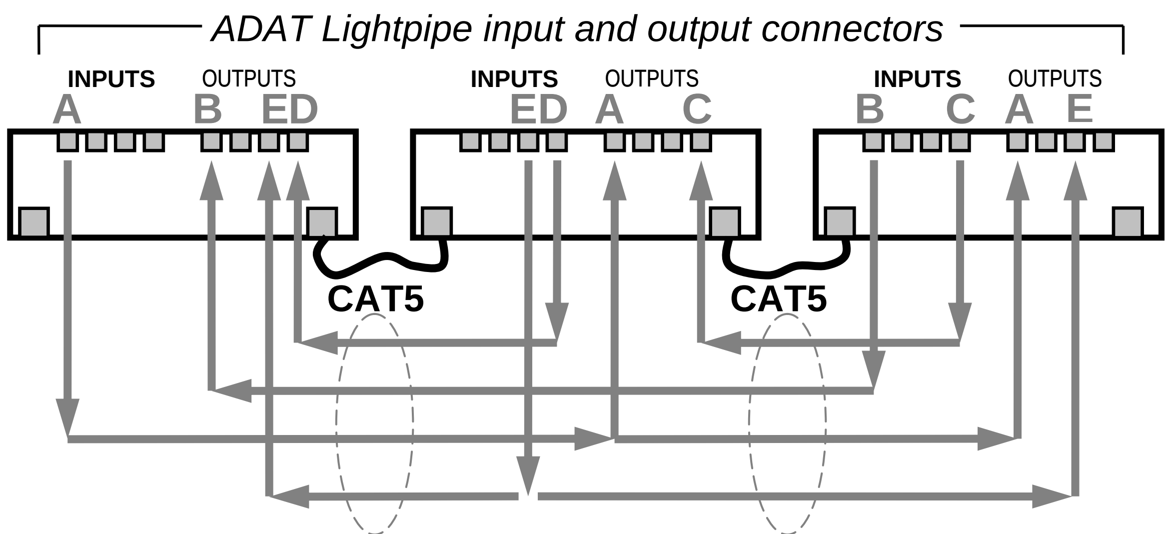
Contents

Chapter 1

Getting started with AudioRail

AudioRail is a daisy chain network with time division multiplexed (TDM) streams of up to 32 channels of audio traveling in both directions at once (64 channels total). At each node in the network, digital audio is inserted into or extracted from the TDM stream.

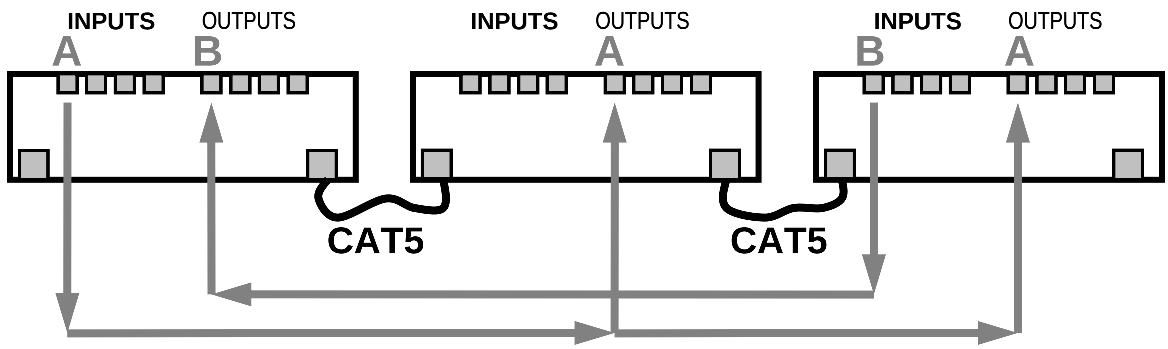
The ADAT rx32tx32 transports the Alesis ADAT Lightpipe digital audio format.



Arrows show signal flow through **CAT5 cable connections**

The picture above has several simultaneous signal paths configured in an example daisy chain of 3 nodes, showing the routing flexibility of the system, entirely configured by front panel push button switches. ADAT Lightpipe stream **A** is being sent from the input on the left AudioRail unit to corresponding outputs on both other units. On that same channel, Stream **B** is being sent from the input on the right AudioRail unit to the corresponding output on the left unit. Stream **C** is being sent from the right AudioRail unit to the middle one (but no further), and on that same channel, stream **D** independently goes from the middle unit to the left unit. Stream **E** is being sent from the middle unit to both other units, traveling in both directions at once.

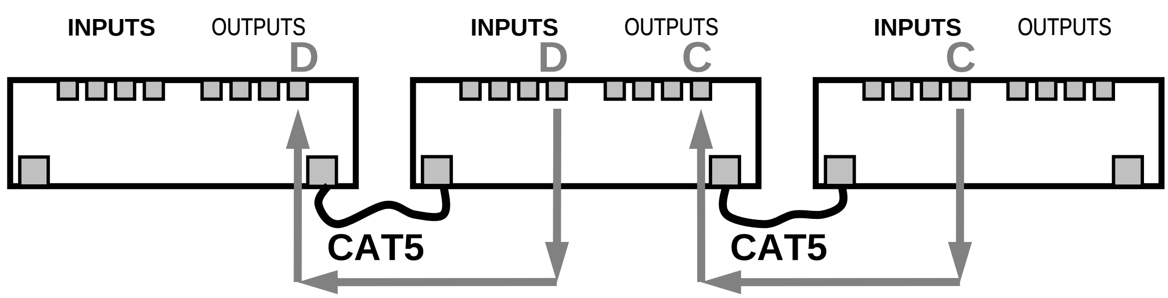
The following pages will break the scenario above into some simpler components, for the sake of better explanation of the functional possibilities.



signal flow through **CAT5** connections

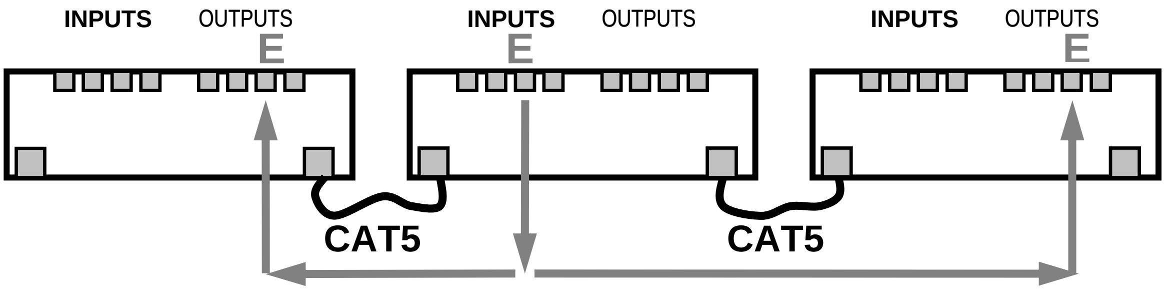
In the picture above, an ADAT Lightpipe input on **A** on the left is configured using front panel switches to go to the right. On the middle and right units, a switch will select the corresponding **A** outputs to come from the left.

AudioRail has two completely independent paths on the CAT5 cable, one going in each direction, each being capable of 32 channels of audio (4 Lightpipe's worth). In the picture above, the ADAT Lightpipe input on **B** on the right unit will be configured using front panel switches to go to the left. On the left unit, a switch will select the corresponding **B** output to come from the right.



signal flow through **CAT5** connections

The above picture shows that a signal need not travel along the entire daisy chain. It can travel over just part of it. A front panel switch on the rightmost unit is pushed to send the **C** lightpipe input to the left. A front panel switch in the middle unit is pushed to get the **C** lightpipe output from the right. Another front panel switch in the middle unit is also pushed to send the **D** input to the left. Note that this is the same lightpipe channel as on the right unit. It is still only taking 1/4th the bandwidth capacity of AudioRail in that direction, leaving all the other channels free and available. The input **D** replaces what could have been stream **C** continuing down the daisy chain, and keeps **C** from going further. Then a front panel switch on the unit on the left AudioRail unit selects **D** to come from the right.



signal flow through **CAT5 connections**

The above picture shows that a lightpipe stream can be sent from one input to go both directions at once. Two front panel push button switches on the middle unit are pushed, selecting the **E** stream to go both to the left and to the right.

A sound system designed around an AudioRail CAT5 network can be very versatile, in terms of its routing ability, the length of the CAT5 cable segments (up to 100 meters, or 328 feet, each segment), the minimal thickness of the CAT5 cable, and manageability of a CAT5 cabling scheme.

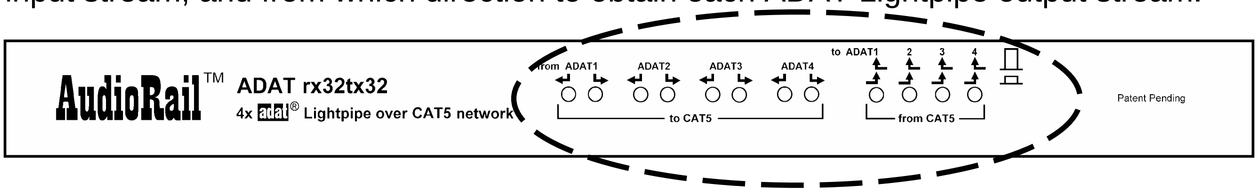
Each ADAT Lightpipe output stream follows the timing of its source (44.1K/48K, or 88.2K/96K with ADAT channel doubling format), making each connection appear as a virtual wire in a digital audio snake.

No software, microprocessor, DSP, or remote configuration is employed. Each unit powers up and is operational in less than 2 seconds. Front panel switches are the only user control.

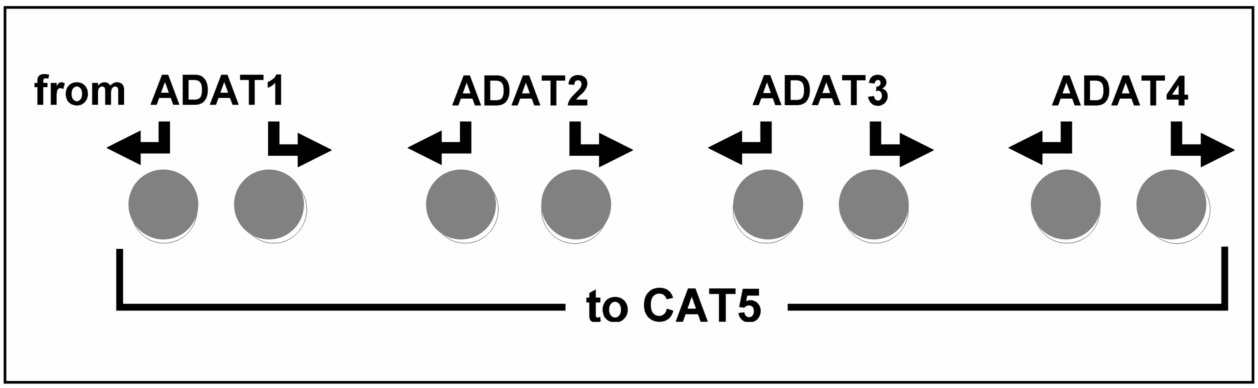
In the following section, the front panel switches are described which select how to route ADAT Lightpipe streams across the daisy chain connections. Chapter 2 will provide application examples.

Front panel switches

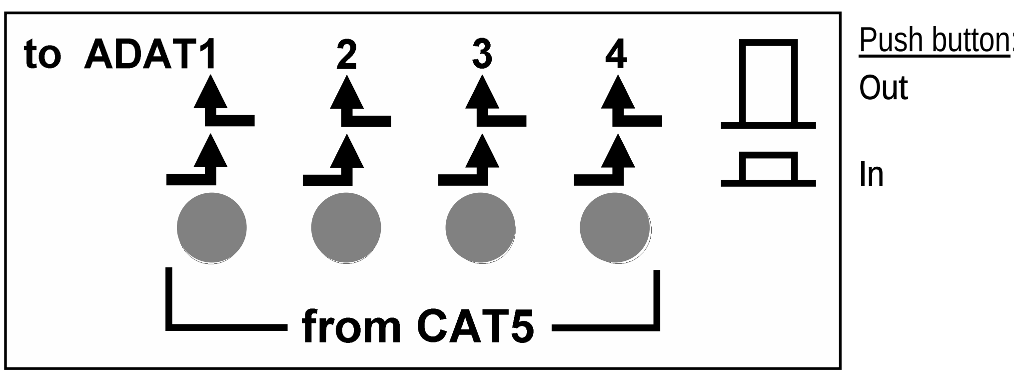
Switches on the front panel determine which direction(s) to route each ADAT Lightpipe input stream, and from which direction to obtain each ADAT Lightpipe output stream.



The picture below shows the front panel **push-button switches** associated with the ADAT Lightpipe **inputs**. The switches direct each ADAT Lightpipe input stream to the left, to the right, to both, or to neither CAT5 cable. Any switch combination is valid.



The picture below shows the front panel **push-button switches** associated with the ADAT Lightpipe **outputs**. Each ADAT Lightpipe output stream is obtained from either the left or the right CAT5 cable, depending on whether the push-button switch is in or out. Any switch combination is valid.

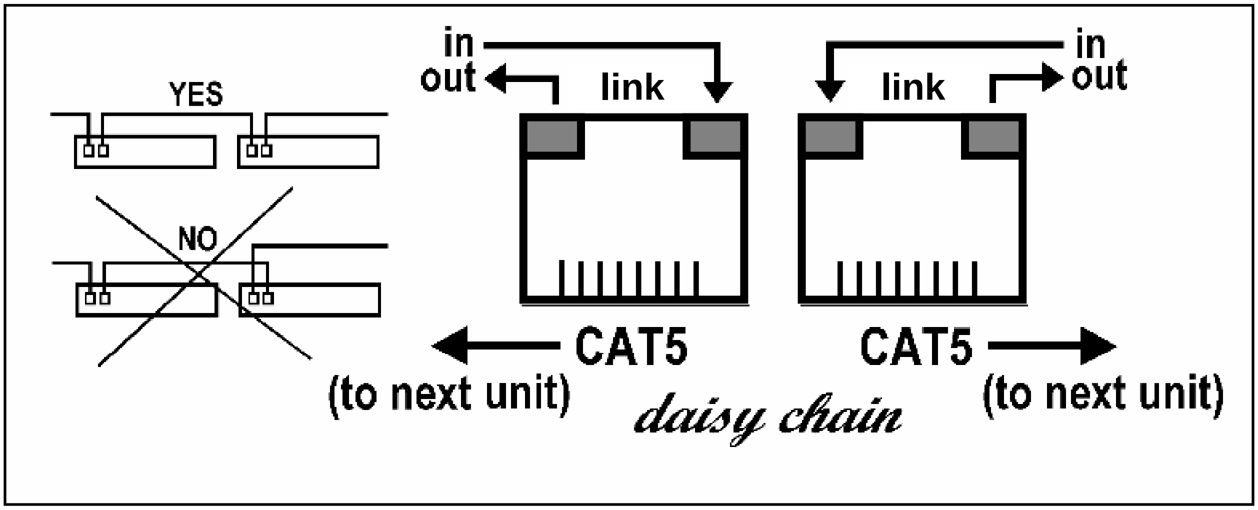


It is important to keep in mind that a particular ADAT connection can only be directed to corresponding ADAT connections on other AudioRail units (ADAT 1 goes to ADAT 1, ADAT 2 goes to ADAT 2, etc.) It is also important to keep in mind that AudioRail does not convert digital audio formats. An AudioRail device that transports Alesis ADAT Lightpipe can co-exist with other AudioRail products that transport other formats of digital audio (such as S/PDIF), but cannot convert to and from the other formats. It can only transport the Alesis ADAT Lightpipe format from one place to another.

Rear panel connections

The rear panel has the CAT5, Alesis ADAT Lightpipe, and input power connections.

The picture below shows the **CAT5 cable connector** area of the rear panel. (CAT5E or other higher grade compatible cable can also be used.) As the "YES" vs. "NO" picture depicts, the cables must be strung from unit to unit between **opposite** CAT5 connectors (not corresponding ones).

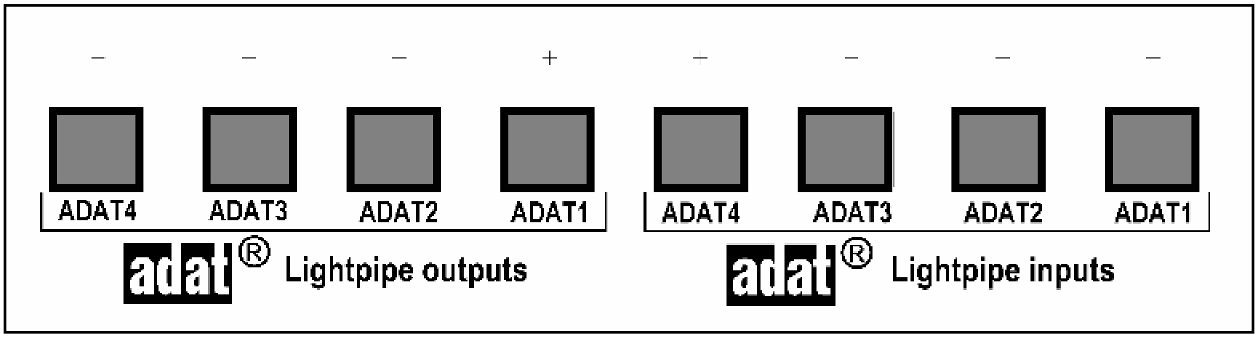


Two green LEDs on each connector indicate outgoing and incoming link status. If the unit is powered on, the outgoing LEDs should always be lit. The incoming LEDs should be lit when another unit capable of transmitting onto the CAT5 cable is connected to it with a CAT5 cable, and is also powered on. These LEDs indicate the presence of a proper link only. They do not indicate the presence or absence of ADAT Lightpipe audio, nor do they blink with data activity.

The physical cabling guidelines are the same as for 100 Mb/s Ethernet. Any Category 5 ("CAT5") Unshielded Twisted Pair ("UTP") wiring scheme can be used in a residential through light industrial environment. In a heavy industrial environment, such as near broadcast transmitters, use Shielded Twisted Pair ("STP") cable. Use normal straight cable and connections, not crossover cable or crossed connections. (Note: If crossover cable or connections are already a given, simply wire according to the "NO" diagram. In this case, two wrongs make a right. Inside the AudioRail enclosure, the only difference between the two ports is that one is crossed and one is not.)

Any number of AudioRail units can be daisy chained together. Each CAT5 cable can be any length, up to 100 meters (328 feet) maximum, per hop. Each hop adds about a quarter of a microsecond delay inside the unit, plus a half a microsecond for each 100 meters (328 feet) of cable. The encoding and decoding of the ADAT Lightpipe format adds about 4 to 5 microseconds total delay, from end to end. (This 4 to 5 microsecond delay is not compounded from hop to hop. It is incurred only once, from end to end.)

The picture below shows the **ADAT Lightpipe** connection area of the rear panel.



Depending on the setting of the front panel push button switches, each ADAT Lightpipe input can be inserted into the stream going out either or both or neither CAT5 cable, and each ADAT Lightpipe output may be tapped from either one or the other CAT5 cable. In any case, each input can only appear on correspondingly numbered outputs on other AudioRail units (ADAT1 to ADAT1, ADAT2 to ADAT2, etc.)

The AudioRail ADAT rx32tx32 product works with Alesis ADAT Lightpipe, running at either 44.1K or 48K sample rate. It also works with the 88.2K/96K ADAT Lightpipe doubling format (which accomplishes its purpose by simply doubling up on the normal 8 channels available, to yield 4 audio channels per lightpipe, instead.) AudioRail will not work with the old ADAT 8-track tape machines that have variable tape speed controls.

Synchronization is on a per ADAT Lightpipe channel basis, from end to end. Each of the eight channels (four inputs and four outputs) is completely independent of the other. If synchronization is required between channels, this must be accomplished at either end, as appropriate. Each ADAT Lightpipe stream follows the timing of its source. Consult the instructions for and follow the corresponding guidelines of the equipment connected at each end to determine what, if any, synchronization needs to be done.

ADAT Lightpipe is based on Plastic Optical Fiber (POF) Toslink connections which are very economical, compared to conventional glass fiber optics used in telecom and computer networking applications. However, it is important to keep in mind that this optical signal degrades profoundly with distance. Different brands of plastic optical fiber yield different maximum distances. By convention, 10 meters (33 feet) is called out as the limit. Because of the eight channel density of the Alesis ADAT Lightpipe protocol, ADAT Lightpipe connections are the most susceptible to degradation. Keep these connections as short as practical. Corruption of data will be apparent as harsh crackling noises and static (not hiss, hum, loss of high end frequency response, or subtle sonic variations, as experienced in traditional analog connections). If corruption occurs, then use shorter or better quality fiber connections. AudioRail uses the highest grade Toshiba Toslink transmitters and receivers, designed for up to 15 Mb/s NRZ data rates, to maximize margins on the AudioRail end. The quality of mating professional audio products and the quality of plastic optical fiber cables on the market vary from product to product, which altogether determines maximum deployable length of the plastic fiber connections. Be advised that this could be either more or less than 10 meters (33 feet).

The picture below shows the standard IEC **power** connection to the rear panel.

The switching power supply used inside the AudioRail product is itself rated for 85-264 VAC, 47-63 Hz. To provide additional margin, the AudioRail ADAT rx32tx32 product is specified to accept 95-240 VAC, 50-60 Hz. The power supply, 8-fold over rated to provide for ample margin, longevity, reliability, and cool operation, requires less than 8 Watts in actual operation at 120 VAC (12 Watts at 240 VAC).

This equipment is of CLASS I construction and shall be connected to a mains power socket outlet with a protective earthing (i.e. grounded) connection.

WARNING: To prevent fire or shock hazard, do not open the unit's enclosure. Refer servicing to qualified service personnel only. To prevent fire or shock hazard, do not expose the unit to rain or moisture. Do not operate if wet. Do not operate if visibly damaged.

Other operating considerations

- AudioRail initializes and becomes operational within about 2 seconds after power up. Power is indicated by the green front panel LED. There is no warm up period, no software, firmware, DSP or any other type of microprocessor, and nothing to configure except the switches. AudioRail recovers from unexpected power, CAT5, or Toslink cable insertions or removals in under 2 seconds.

- Removing and inserting cables with the sound system on should receive the same consideration as with conventional analog systems. Turn any amplifiers off that drive live speakers, so as to prevent high SPL transient clicks, pops, and crackle noises that can sometimes occur while plugging and unplugging cables.

- Changing clock sources in any digital audio system should never be done with live speakers on. The process of losing and regaining clock synchronization can cause high SPL transients. This is not a problem particular to AudioRail. It is a system level problem with any digital audio system.

- AudioRail is rated for 0-50 degrees C (32-122 degrees F), 10-90% non-condensing humidity. If in a rackmount or stack of equipment, make certain that it is not overheated by adjacent equipment.

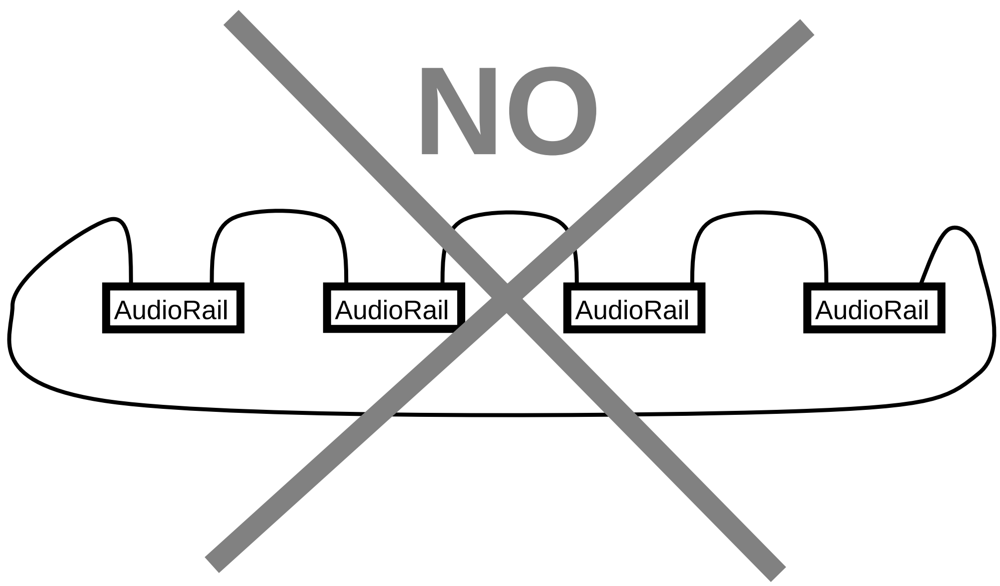
- When relocating any equipment suddenly from a very cool to a warm humid environment, condensation can occur. In this case, allow ample time to ensure that the unit is dry before applying power.

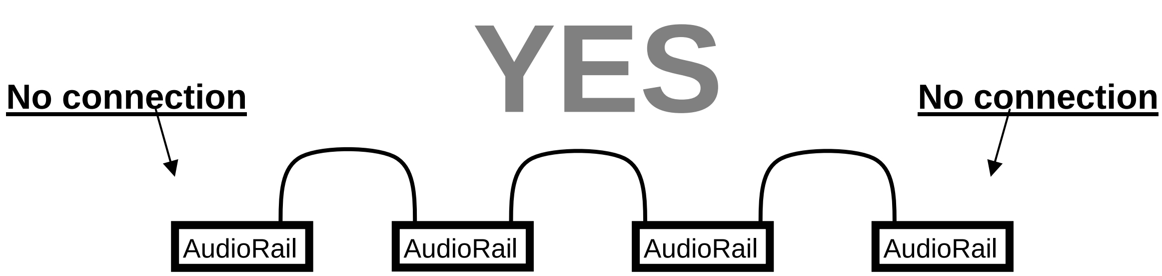
- The unit must not be exposed to dripping or splashing. No objects filled with liquids, such as vases, shall be placed on the equipment.

- No, there is no power switch. Why should we spend extra money putting in a component that causes more problems than it solves, being able to take down an entire sound system at the unintentional slip of a finger? However, as a safety requirement the user must assure that the appliance coupler (the connection between the unit and the power cord) remains readily accessible, since it is the power disconnect device for the equipment.

- AudioRail is not compatible with conventional Ethernet LAN connections. Electrically, the signals are compatible, but interconnecting a computer LAN with AudioRail will cause both to stop working until the connection is removed. After the connection is removed, both will recover.

- Do not configure AudioRail as a ring network. It will not work! See figures below:





- AudioRail has been tested for electromagnetic emissions in accordance with EN55103-1 (all categories: E1 through E5), EN55013, and FCC Class B, with substantial margin, and can be used in diverse environments ranging from residential to heavy industrial.

- AudioRail has been subjected in immunity testing to power line electrical fast transients (EFT) of up to 2000 Volts and direct electrostatic discharge (ESD) events of up to 4000 Volts, in accordance with EN55103-2, category E5 (corresponding to "heavy industrial environment and environments close to broadcast transmitters", the most stringent). Events of this magnitude will not damage the product or cause it to lock up. However, users should be advised that powerful EFT/ESD events can result in a single audible crackling noise, of a nature that is commonly experienced in analog audio systems. If events such as these, as uncommon as they may be, are unacceptable in an application (such as a one-shot live recording venue in a harsh environment), then two courses of action may be appropriate: 1.) The AudioRail enclosure should be mounted in a solidly grounded 19" rack enclosure, or otherwise connected to a better ground to the metal enclosure than what may be provided by the power cord. 2.) Power line surge protection devices and/or power line conditioners as commonly employed in professional audio work should be used to guarantee power that is free of such transients.

- AudioRail has been subjected in immunity testing to conducted RF (radio frequency) levels of up to 10V/m, in accordance with EN55103-2, category E5 (corresponding to "heavy industrial environment and environments close to broadcast transmitters", the most stringent). AudioRail is immune to conducted RF levels of up to 3V/m, corresponding to categories E1 through E4 (residential through light industrial in nature) when using UTP (unshielded twisted pair) cable, and radiated RF levels of up to 10V/m, corresponding to category E5 (heavy industrial). However, to fully comply with category E5 immunity requirements, the user operating in this environment class must employ CAT5 STP (shielded twisted pair) cable, rather than UTP cable. *(Note: The term "radiated" refers to immunity to RF energy in free space. The term "conducted" refers to RF energy coupled directly onto the cables through induction or direct electrical connection.)*

The term "emissions" refers to the product's ability to not generate electromagnetic interference that might cause other devices in the environment to malfunction. The term "immunity" refers to the product's ability to continue to function properly in the presence of environmental electromagnetic interference.

Chapter

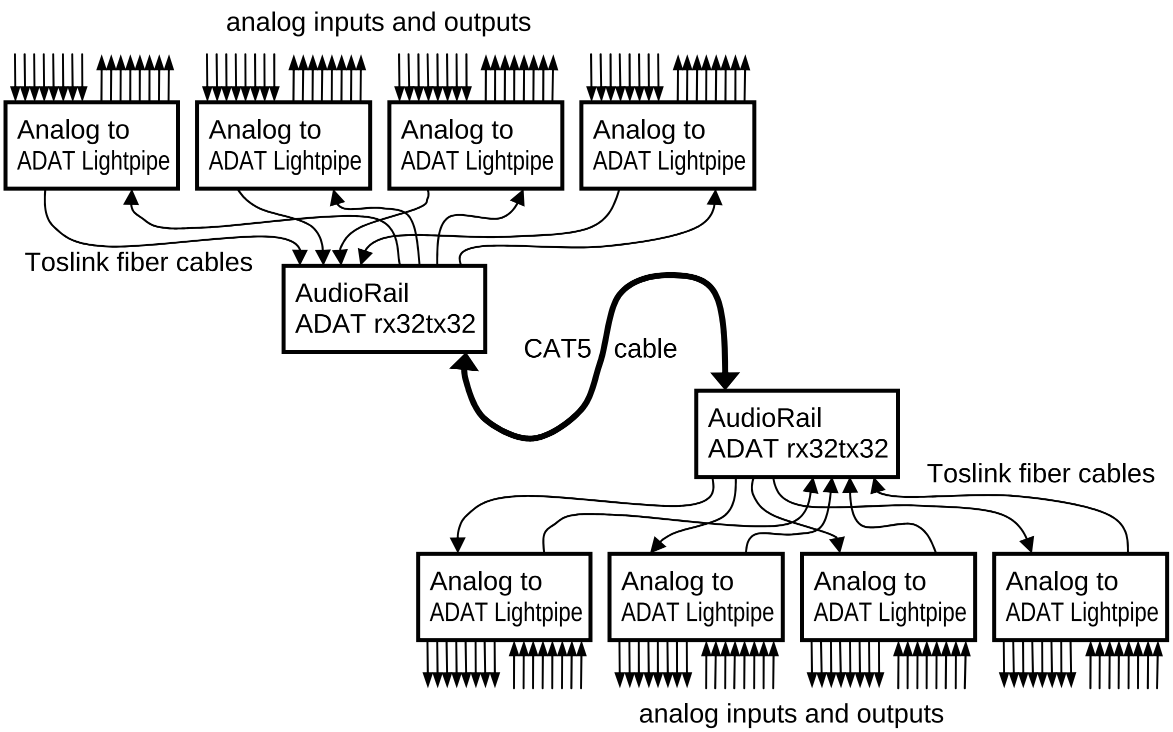
2

AudioRail application examples

AudioRail is both a transport mechanism and a switching/routing device. Its applications in professional audio are many. This chapter gives a few examples of how it can be used.

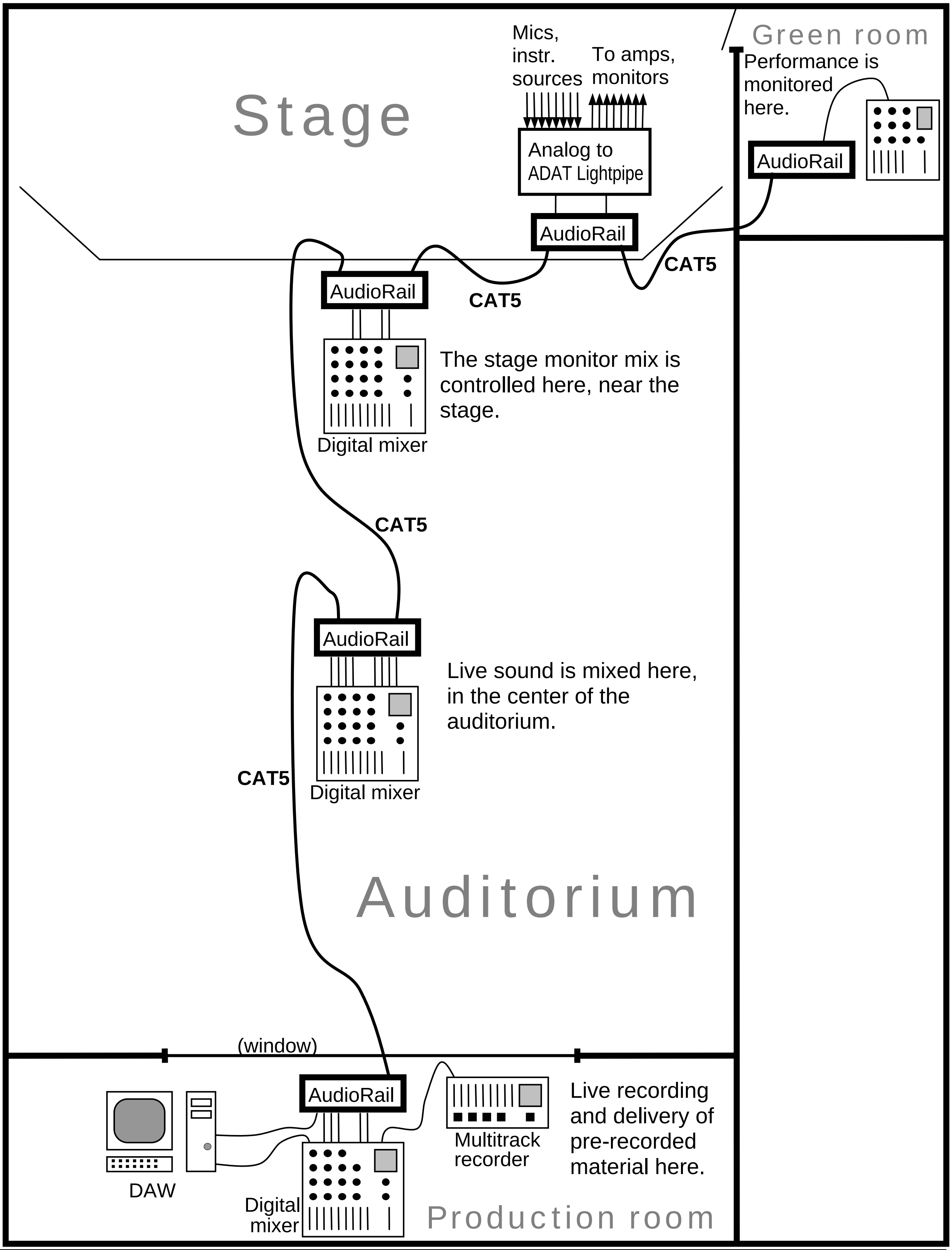
A simple 64-channel digital audio snake

Two AudioRail ADAT rx32tx32 units and digital/analog converters with ADAT Lightpipe interfaces make a simple digital audio snake up to 100 meters (328 feet) long:



An example of an analog to digital converter that supports Alesis ADAT Lightpipe protocol as illustrated above would be the Behringer ADA8000, or the Alesis AI3, both of which provide 8 analog inputs and 8 analog outputs in a single enclosure. Any other professional audio product that uses the ADAT Lightpipe interface can be used, except for the old ADAT tape decks with variable speed tape controls.

A complex live sound configuration



The picture above diagrams an example of AudioRail in a complex live sound configuration. Five AudioRail nodes are employed, with four CAT5 cables interconnecting them. The functions at each location in the venue are described below:

Stage

The AudioRail unit on the stage connects to analog/digital conversion units where the live performance microphones, live instruments, live recording microphones, and outputs to stage monitors and main speaker amplifiers connect.

Stage monitor mix

The AudioRail unit in front of the stage connects to a digital mixing console (or analog mixing console with outboard analog/digital conversion). The sound operator here taps off of all the live microphones and instruments, as well as the pre-recorded material sent live from the production room and mixes down into each of the stage monitor outputs as appropriate for the performers.

House mix

The AudioRail unit in the center of the auditorium connects to a digital mixing console (or analog mixing console with outboard analog/digital conversion). The sound operator here taps off of all the live microphones and instruments, as well as pre-recorded material sent live from the production room, and mixes down into the main speakers driving the house mix.

Production room

The AudioRail unit in the production room connects to a digital mixing console and digital audio workstation (DAW). The digital mixing console is also connected to a digital multitrack recorder. The sound operators there take all of the live performance microphones and instruments, along with additional recording microphones in the auditorium and stage, and do a live multitrack recording of the performance. Sound operators also deliver the pre-recorded material that is used in the performance. This room can also be used as a delivery point for live broadcast, and a mixdown to the Green Room.

Green room

The AudioRail unit in the Green Room connects to a digital mixing console (or analog mixing console with outboard analog/digital conversion). The Green Room is where the inactive performers reside while waiting for their turn on stage. A sound operator there takes a stereo mixdown originating from the production room, and perhaps combines it with a few other inputs from recording microphones or direct talk-back feeds from the production room, to generate a Green Room monitor mix.

"What?? No control of mic. pre's at F.O.H. mixer?"

It is true. By going digital and, particularly long distance digital, the sound operator has lost "arm's reach" control of a whole row of knobs having to do with the input trim adjustments. They are now at the A/D conversion units, up at the stage. This has its disadvantages, but there are some advantages, as well as some objections that used to be valid in analog (especially older analog) systems that are no longer valid considerations in either current analog systems or (especially) digital audio systems. These are important to point out.

Logistically, having input trim level adjustments physically near their sources can be as much of an advantage as a disadvantage. Particularly today, with most instruments having their own built in, powered preamplification stages, it makes more sense to adjust the instruments and the A/D converter input trims together in the same physical vicinity, at the stage. Often this involves communication and coordination between the sound technician and the performer whose instrument is of concern, which is easier done at the stage.

Keep in mind that the trim adjustments are meant to normalize levels traveling through the sound system, rather than adjust live mix levels, and is something that should be accomplished during sound checks, and not during a performance. Particularly where a stage monitor mix is employed, changing trim adjustments at the sound board during a performance is not good practice, as it affects both the house mix and the stage monitors at a time when the sound technician and performers can no longer discuss the outcome of the adjustments' impact to the stage monitor mix.

From a technological standpoint, many mixing boards built in the mid 1980s and prior had "sweet spots" of one sort or another that called for optimizing levels of signal paths to prevent significant signal to noise ratio problems, prevent (or perhaps bring out) compression effects, and so on. Mackie changed the state of the art of live sound reinforcement in the mid 1990s by introducing studio quality microphone preamps in their lineup of general purpose mixing boards, and their competitors followed. Advances and cost reduction of analog integrated circuit components also contributed along with this, resulting in even low cost mixing consoles having a more linear and clean response, largely eliminating the old "sweet spots".

Digital audio then takes this issue yet another quantum step. A 24-bit digital audio system, for example, has 144 dB of dynamic range (up from 96 dB for 16-bit CD digital audio). Once the signal is in the digital domain, numbers are manipulated mathematically. A digital mixer, for example, can operate entirely in the digital domain, and its "faders" and other control knobs are simply passing positional information about the knob or fader slider to computer chips inside the unit, like the mouse or scroll-wheel of a computer does.

The increased headroom (dynamic range) and lack of "sweet spots" means that for a system designed around AudioRail and digital audio, all input trims can (and should) be reduced to a point that securely precludes the possibility of overload (clipping) during a live performance, without fear of losing any perceptible sonic fidelity. This eliminates the need for the "arm's length" access to trim adjustments during a live performance.

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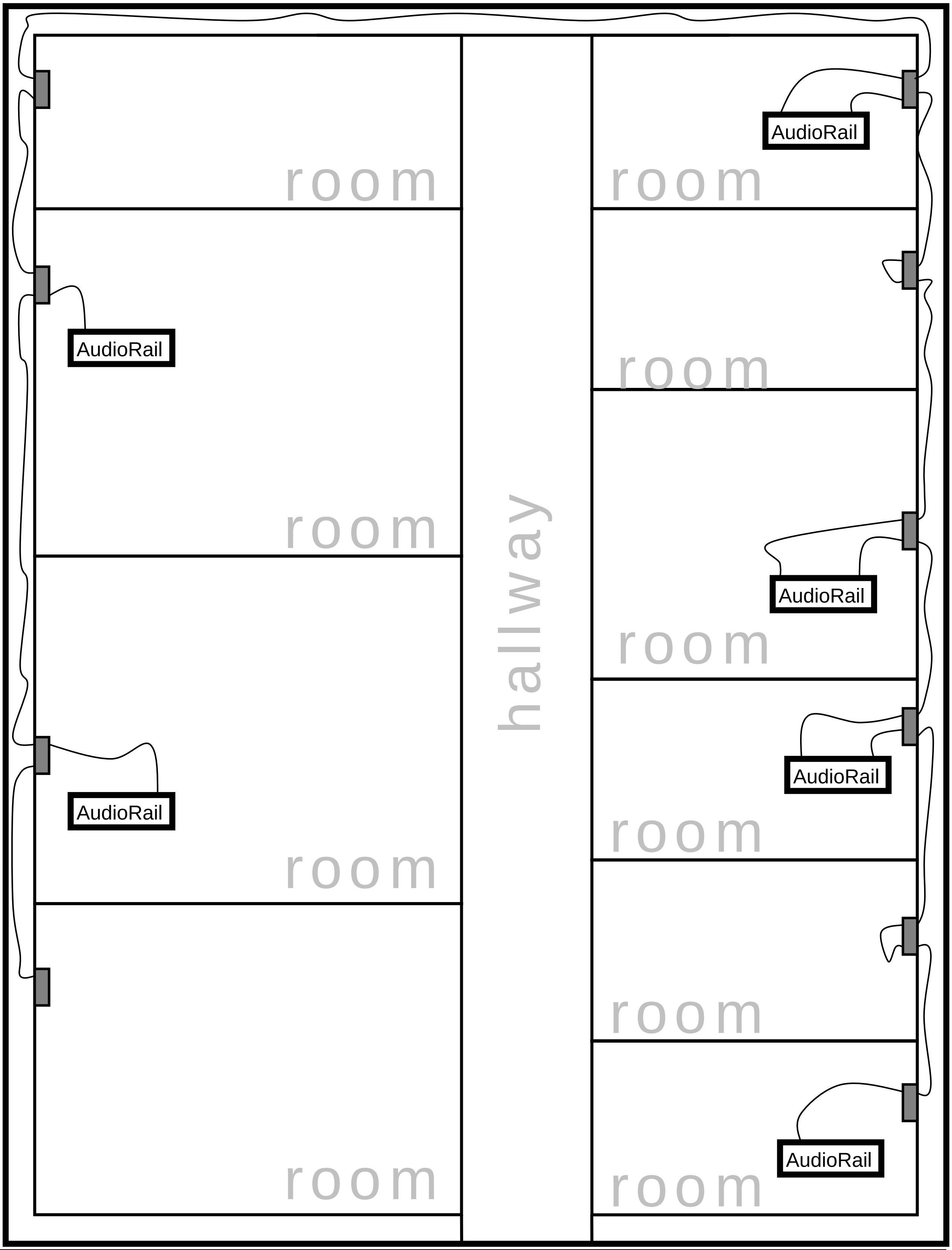
A building wired for AudioRail



In the above picture (preceding page) a building is wired for AudioRail using standard LAN wiring outlets and cables, and CAT5 cable routed through the walls.

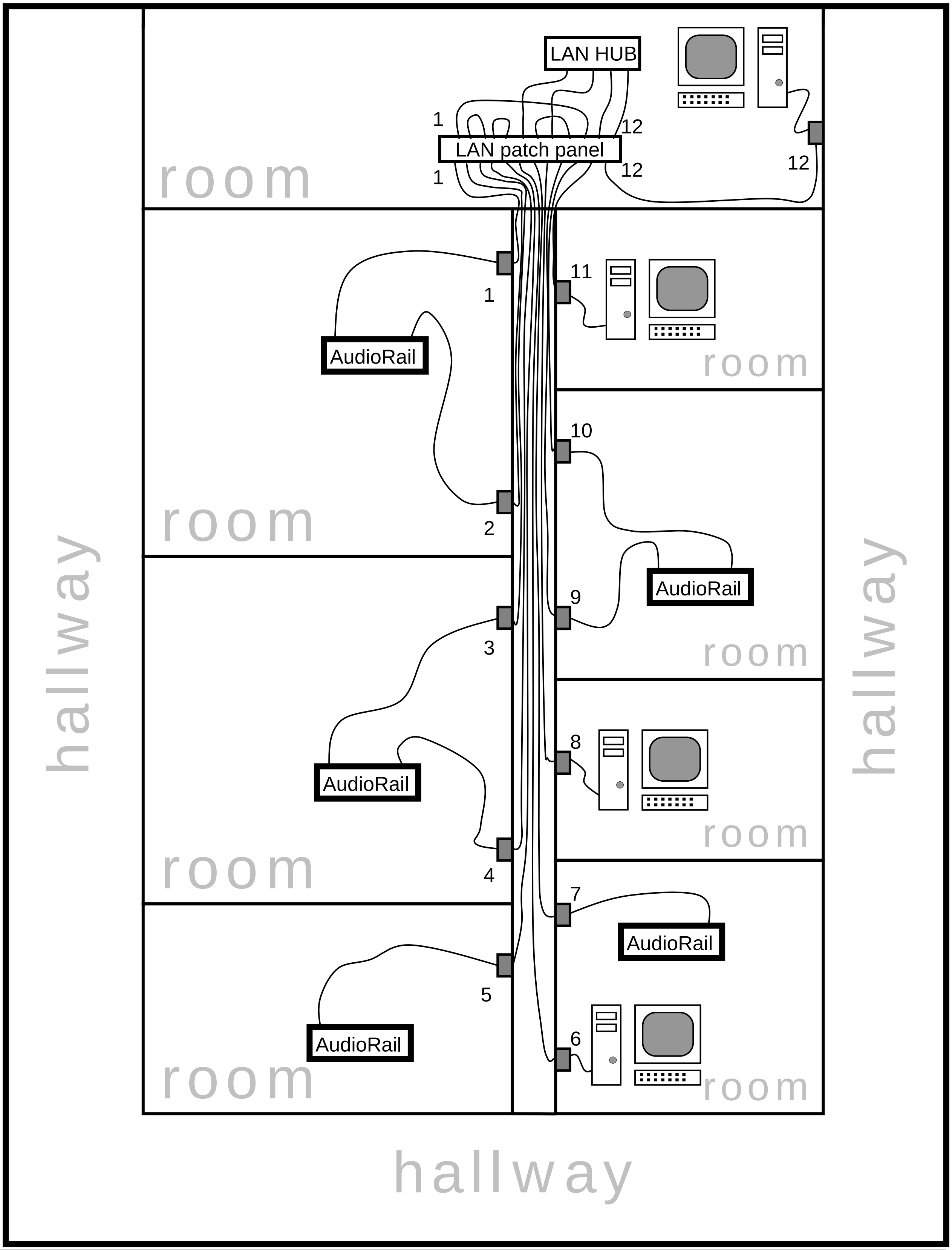
Each outlet has two RJ45 receptacles. Inside the wall, CAT5 cable is daisy chained from outlet to outlet, room to room.

Wherever AudioRail enclosures do not exist in a room, a short CAT5 cable must connect between the two receptacles in that room to bridge a desired connection across that room.

Notice in the particular example above, the four AudioRail units on the right side of the building are all connected together, but are not connected to the two AudioRail units on the left side of the building. The two AudioRail units on the left side of the building are connected to each other, but nothing else. So there are two separate AudioRail systems operating in the building. (Note: Even if they were connected, this would not preclude using the front panel switches to make them run separately.)

Notice that the above picture looks like an office building, but it could just as well be a performance facility, replacing traditional analog audio wall outlets distributed around the facility with CAT5 connector outlets.

A building wired for LAN, reconfigured for AudioRail



In the above picture (preceding page) a building was already wired assuming a standard Ethernet computer LAN would be used. Standard LAN wiring outlets, with CAT5 cable routed through the walls, all connect to a patch panel in one room.

Each outlet has one RJ45 receptacle. The larger rooms have two outlets. (The smaller rooms with only one outlet will pose a problem for AudioRail, except at the ends.)

There is a computer LAN, consisting of four computers, in this building. The four computers are connected to an Ethernet hub, the usual way.

The AudioRail units in all the rooms are patched together into a daisy chain at the patch panel. Rooms with AudioRail units that are not at the ends of the AudioRail daisy chain need two outlets in the room. All five AudioRail units in this example are connected together in a daisy chain.

The computer LAN uses the building wiring, and so does AudioRail. But the computer LAN and AudioRail are not connected together at any point. They are entirely separate, and should never be interconnected.

Chapter

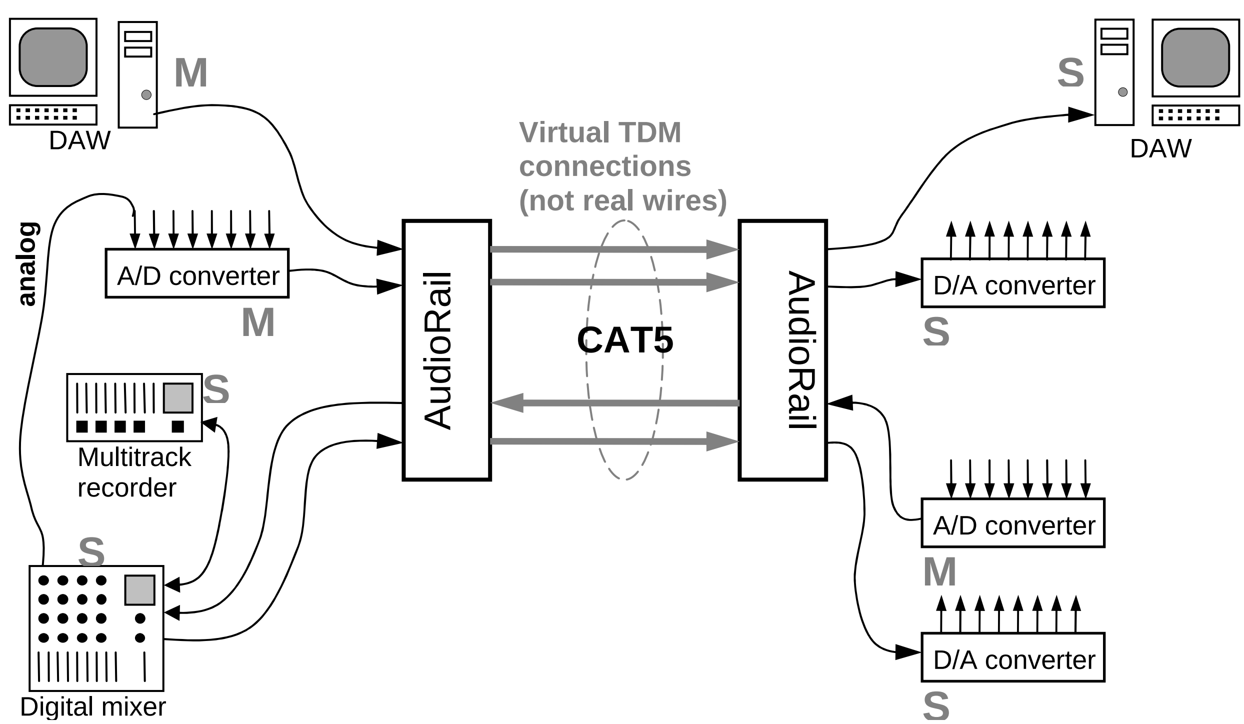
3

Word Clock synchronization

In any digital audio system, care must be taken to properly select Word Clock master and slave devices in the system. Failing to do so can cause very audible problems. This is not an issue particular to AudioRail. If you are new to digital audio, you must read this chapter carefully to understand what is involved.

AudioRail is very versatile and flexible, in that each ADAT Lightpipe output connection follows the timing of its source, as it appears at its input connector. Different end to end connections can be run at different speeds and use different clock synchronization schemes, even though they are time division multiplexed over a single CAT5 cable. Each connection appears as a virtual wire in a digital audio snake, as if it was a wire in a bundle of separate, individual digital audio cables. AudioRail has no common clock.

With this versatility, the user must consider each digital audio path and verify that exactly one clock source is selected to be master of each subsystem that requires synchronization. Consider the example pictured below:



In the example on the preceding page, there are three independent digital audio subsystems:

1. The two digital audio workstations (DAWs).

2. The mating pair of A/D and D/A converters, below that.

3. The digital mixer and multitrack recorder on the left, connecting to the A/D converter and D/A converter on the right (bottom).

In each subsystem above, exactly one device must be chosen to be the master (**M**) and the rest are designated as slaves (**S**).

Notice that there is an analog cable connecting subsystems 2 and 3 above. This does not matter. You do not have to synchronize components that have only analog connections between them. (If it had been a digital cable, the A/D converter would have had to be configured as a slave.)

The user's manual for each product describes how to configure it. Either a software or hardware switch will be available to select "master" or "slave", also often designated "internal" or "external" clock. Often a BNC connector is available to connect a 75 Ohm cable to externally slave one device's word clock to another. Configuration options vary considerably from product to product. Sometimes limitations in the available clocking schemes of one product will limit the user's choices, such as when a device can only operate as a slave, or defaults to slave mode when another device is connected to it.

Why does digital audio need a common clock?

The reason for the necessity of a common clock in any digital audio system is that no two independent clock sources can maintain perfect synchronization. If two digital audio sources running at a 48K sample rate were to each be driven by quartz crystal oscillators that had an accuracy of 100 ppm (parts per million), then one could actually be running at 48005, while the other was running at 47995. The difference between these two numbers means that the slower one would drop samples at the rate of 10 per second, or else the faster one would double-clock samples at the rate of 10 per second.

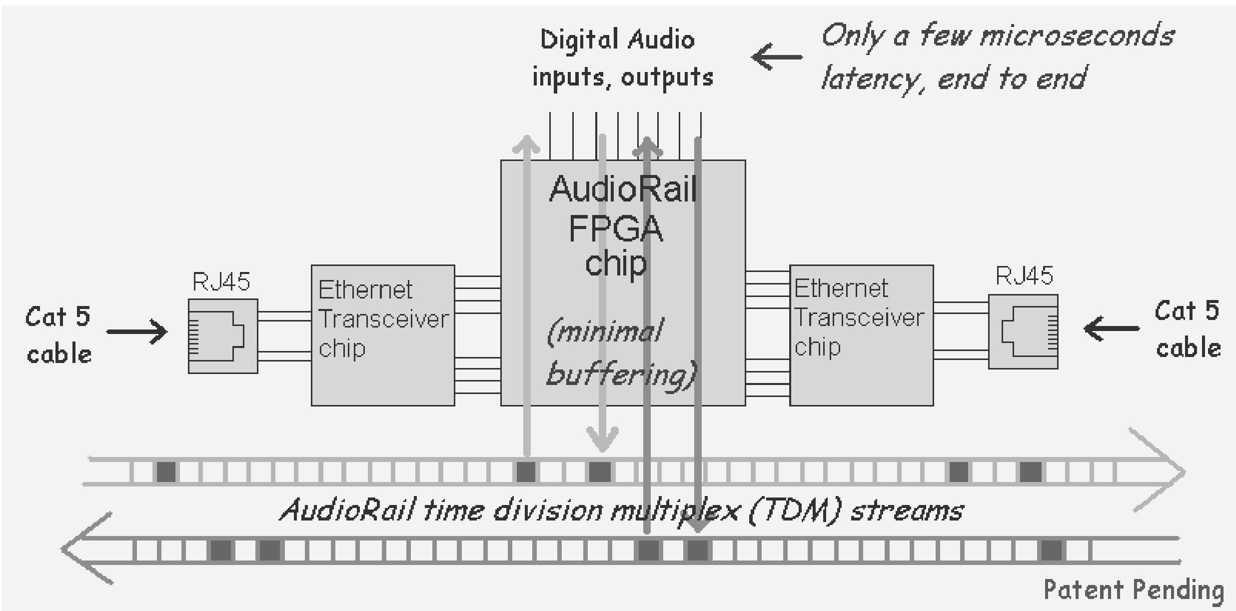
When a device is configured as a "slave" it extracts the clock from the digital audio stream and locks onto it, instead of its own quartz crystal oscillator.

What about sample rate conversion?

In some cases digital audio components have the capability to do "sample rate conversion." Wherever a specific digital audio port has this capability, there is no need for a common clock. Sample rate converters re-sample the digital audio, and can convert a few parts per million difference as easily as they can convert between entirely different sample rates. Again, each product's user's manual should be consulted.

Chapter
4

How AudioRail works: A tutorial



AudioRail™ is a Patent Pending method for using a time division multiplexing (TDM) scheme to transport many channels of digital audio over a daisy chain of standard Category 5 or other LAN interconnect media using inexpensive LAN transceiver technology and a minimum of logic.

Instead of using conventional packet/frame store and forward methods, AudioRail uses time division multiplexing to transport digital audio more simply and with far less latency than most other schemes. No software, firmware, DSP, or any kind of microprocessor is employed.

Ethernet transceivers are normally employed in computer LAN applications using Carrier Sense Multiple Access with Collision Detect (CSMA/CD) in a packet store and forward scheme to provide for the local area networking of computers. Putting together digital audio and audio related signals with a TDM stream and an Ethernet transceiver was an unlikely, but novel marriage of technology that works elegantly to provide a simple and trouble free solution for the transport of digital audio.

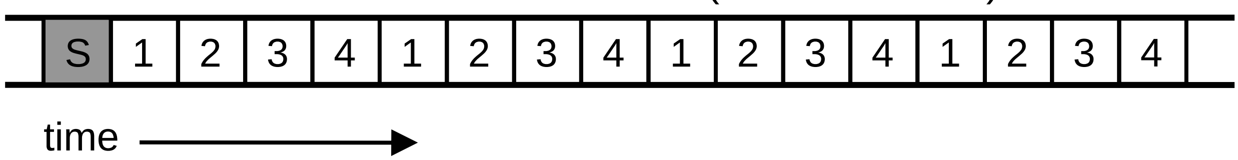
What is Time Division Multiplexing?

“Time Division Multiplexing” refers to a continuous stream of data in which there are regular, repeating slots assigned for each channel. It is not a new concept, most widely recognized as having been used for so many decades by the telecommunications industry for sending many channels of telephone voice signals over a single wire.

Coincidentally, the Alesis ADAT Lightpipe protocol itself is also based on a time division multiplex protocol.

In the case of the AudioRail ADAT version, each ADAT Lightpipe stream, itself a time division multiplex protocol of 8 channels each, is then time division multiplexed on a once per four slot basis on AudioRail, to transfer up to 32 channels repeatedly in sequence.

AudioRail ADAT time slots (4 bits each slot)



There are periodic additional synchronization points (the “ S ” above) in a time division multiplex scheme to make sure that the receiver lines up the time slots to match the transmitter. All this is organized as a repeating, back to back sequence. The sequence is continuous and deterministic, causing the data to be predictable in time.

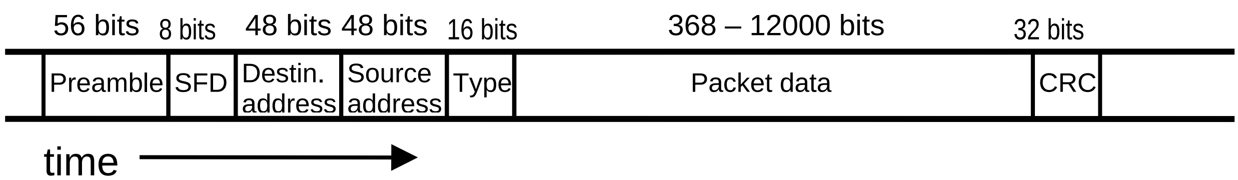
Each Alesis ADAT Lightpipe of 8 bundled channels takes up 25% of the available bandwidth of AudioRail ADAT rx32tx32 in one direction. Other AudioRail products can interoperate on the same AudioRail daisy chain, taking only as much of the share of slots as they need. For example, AudioRail S/PDIF products assign each S/PDIF stream (each S/PDIF stream contains 2 channels of audio) to only one of every 16 consecutive slots, for the case of 44.1K/48K sample rate transmission, or one of every 8 consecutive slots, for the case of 96K sample rate transmission (AudioRail can only transfer up to 16 channels of 96K audio in each direction).

The telephone company employed TDM streams, giving us decades of glitch-free long distance telephone service. Serial MADI, the Audio Engineering Society standard AES-10, released by the AES in 1991, also uses time division multiplexing, for point to point, unidirectional 75 Ohm coaxial cable connections. And again, Alesis ADAT Lightpipe is itself a time division multiplex scheme.

What AudioRail is not: Ethernet, packets, CSMA/CD

The physical and electrical technology that AudioRail uses to send and receive its digital audio over a TDM stream is taken from standard Ethernet LAN components. This, however, is where the similarity ends.

Computer LANs use packet-based transmission and reception, employing complex, sophisticated protocols. Each Ethernet packet has the following format:



Within the “Packet data” are additional layers of more protocols, for the case of computer networking. Embedded within that is the actual user data.

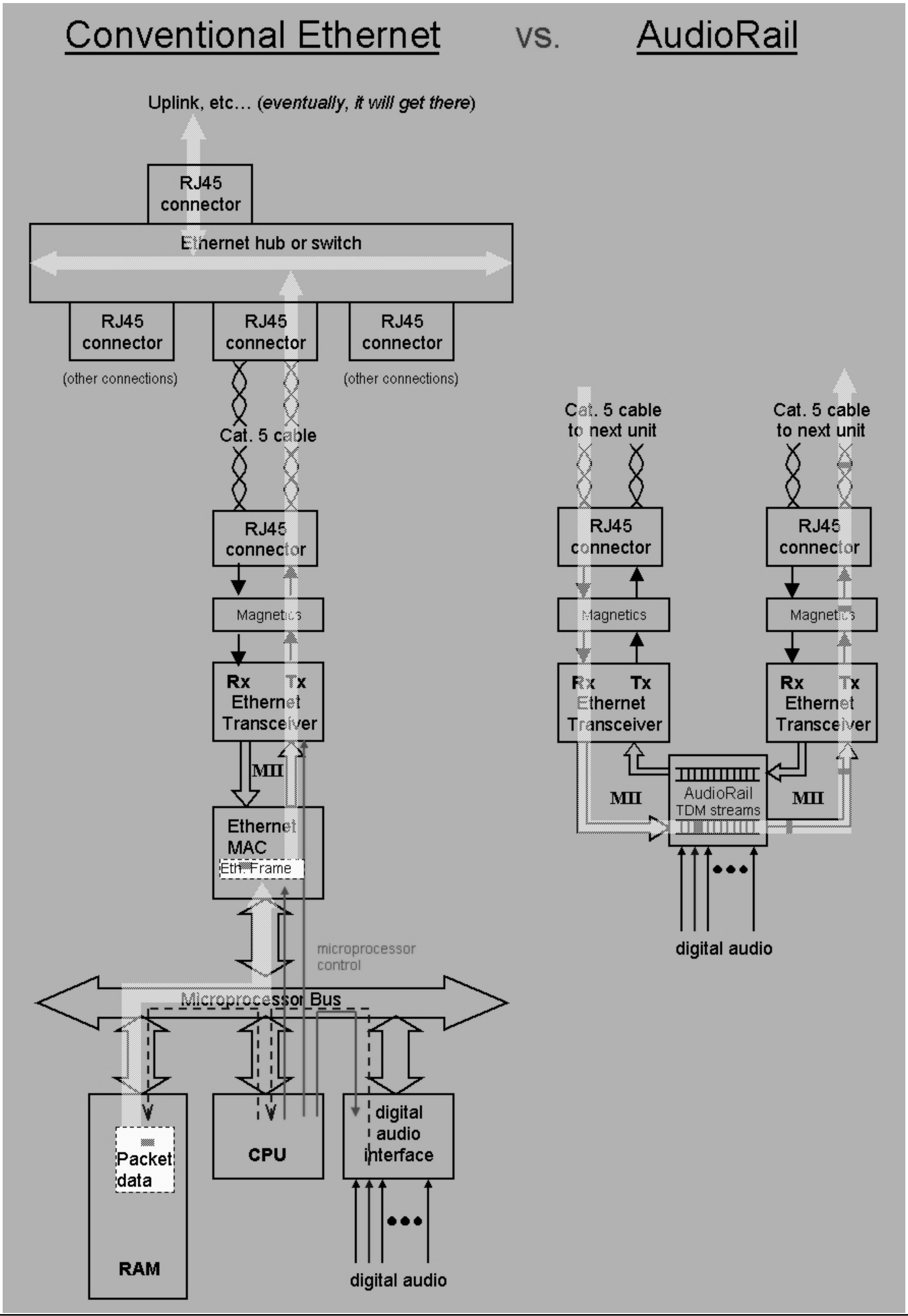
An Ethernet packet can be transmitted over a LAN connection at any time, and is not synchronized to anything in principle. CSMA/CD (“Carrier Sense Multiple Access with Collision Detect”) means that the device originating the transfer first checks to see if there is another transmission taking place on the wire. Then it just starts transmitting data. If two devices on the same wire happen to start transmitting at the same time (because they coincidentally perceived the wire to be available), then there is a collision. The devices have a collision detection mechanism that recognizes that this has happened (after the fact), after which they both back off and try again at slightly different times, according to a random algorithm that will reduce (but not eliminate) the probability of them both retrying the transmission again at exactly the same time and colliding once again.

The above works fine for the nature of computer local area networking. But for audio, transmission of data must be timely, deterministic, and guaranteed.

A second problem with doing things the “computer LAN” way is the complexity of doing so. AudioRail eliminates the entire mechanism for constructing Ethernet packets, including the Ethernet MAC (“Media Access Controller”) itself, and replaces the whole thing with a much simpler TDM mechanism.

The downside of AudioRail is that it has eliminated the mechanisms for interoperating with computer LAN components. The data format of one is unrecognizable by the other. An Ethernet “Hub” cannot be used, because it combines the transmit and receive paths into one “wire”. An Ethernet “Switch” cannot be used, because it is looking for a “Source” and “Destination” address to decide which port to direct the data. AudioRail can only be connected to a product specifically designed to communicate via the AudioRail protocol.

The tradeoff is simplicity, reliability, determinism, latency, and cost, for computer LAN interoperability. The picture on the following page illustrates the difference.


Conventional Ethernet
vs.
AudioRail
Uplink, etc… (eventually, it will get there)
RJ45 connector
Ethernet hub or switch
RJ45 connector
RJ45 connector
RJ45 connector
(other connections)
(other connections)
Cat. 5 cable
RJ45 connector
Magnetics
Rx Tx Ethernet Transceiver
MII
Ethernet MAC
Eth. Frame
microprocessor control
Microprocessor Bus
Packet data
RAM
CPU
digital audio interface
digital audio
Cat. 5 cable to next unit
Cat. 5 cable to next unit
RJ45 connector
RJ45 connector
Magnetics
Magnetics
Rx Tx Ethernet Transceiver
Rx Tx Ethernet Transceiver
MII
MII
AudioRail TDM streams
digital audio

Several other professional audio firms have other solutions that address the problems with the computer LAN paradigm described above, to come up with ways to deliver many channels of audio over Ethernet, or Ethernet-like connections. Each has its tradeoffs, as does AudioRail. AudioRail is the simplest, most straightforward, most reliable, and most cost effective product and methodology of its class.

Visit **http://www.audiorail.com** on the web for more information about AudioRail Technologies.

Chapter

5

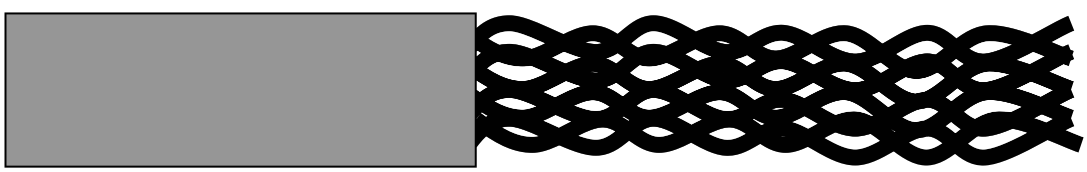
Making your own CAT5 cables

AudioRail works with any standard Category 5 cabling scheme, using standard straight-through (not crossover) cabling. Any off the shelf, ready made CAT5 network cable or existing CAT 5 compliant building wiring can be used (UTP or STP, as appropriate.)

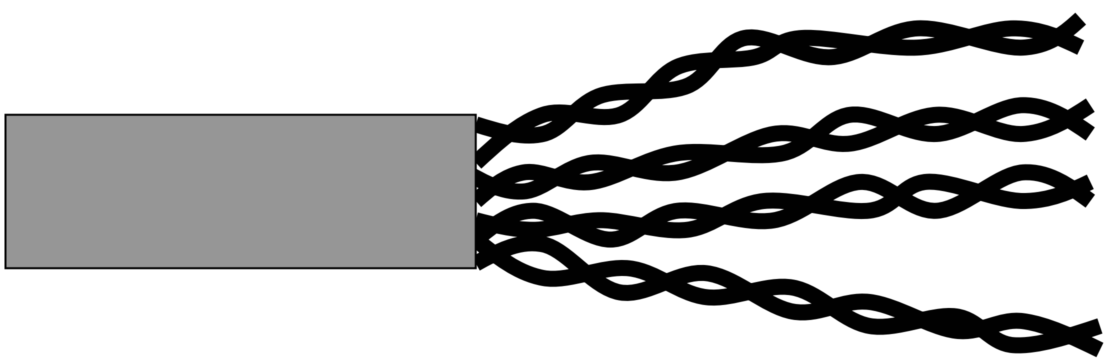
Bulk Category 5 cable can also be bought in 1000 foot spools for what amounts to as little as 5 cents a foot. Radio Shack sells it by the foot for about 15 cents a foot (#278-1583). 8-pin UTP modular plugs can be bought for a fraction of a dollar each. These are also available at Radio Shack (#279-406), as well as from any number of other sources. We actually recommend the approximately $32 Radio Shack crimp tool (#279-405) over other, more expensive brands we have used.

Making your own cables can result in lower costs and cables better tailored to your installation. However, **careful attention to detail** is **important,** or else **intermittent, noisy,** or **unreliable** connections could result. Read the following instructions carefully.

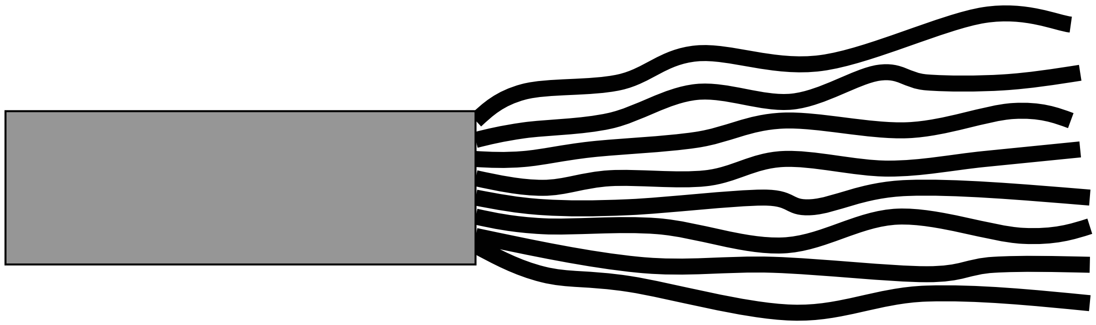
Step 1: Strip the cable outer sheath back about three quarters of an inch. **Check carefully to make sure you have not nicked any insulation from the wires inside.**



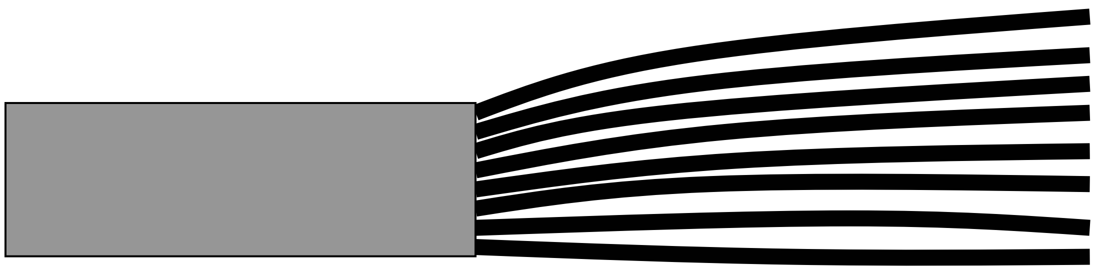
Step 2: Pull the four twisted pairs from each other. **Note the <u>color code</u> of the pairs.**



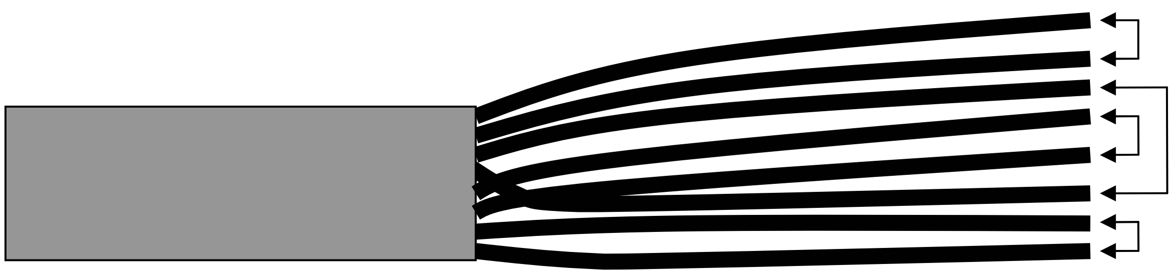
Step 3: Untwist the individual wires in each pair.



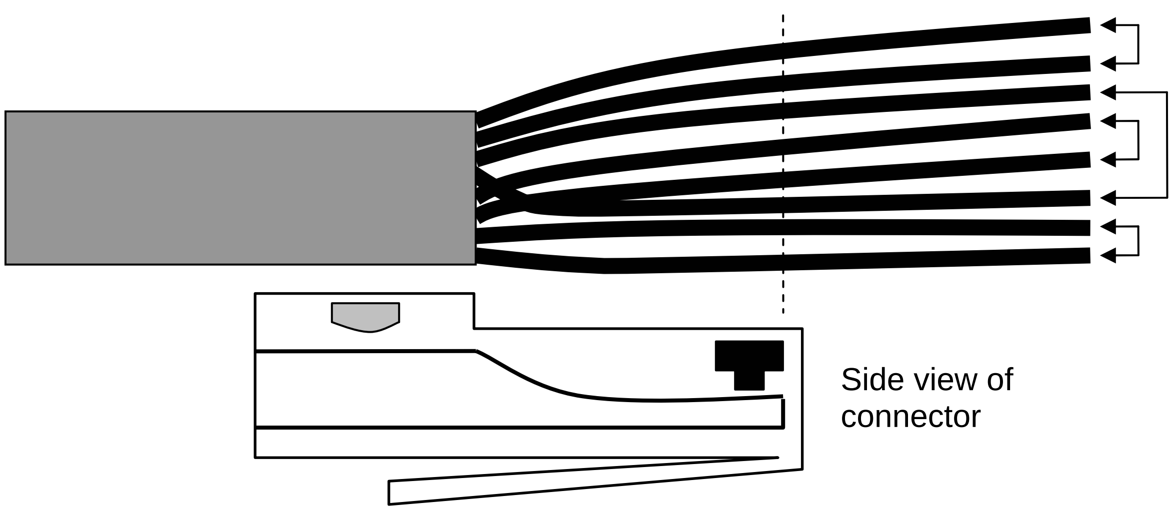
Step 4: Straighten the wires. **Make them very straight.**



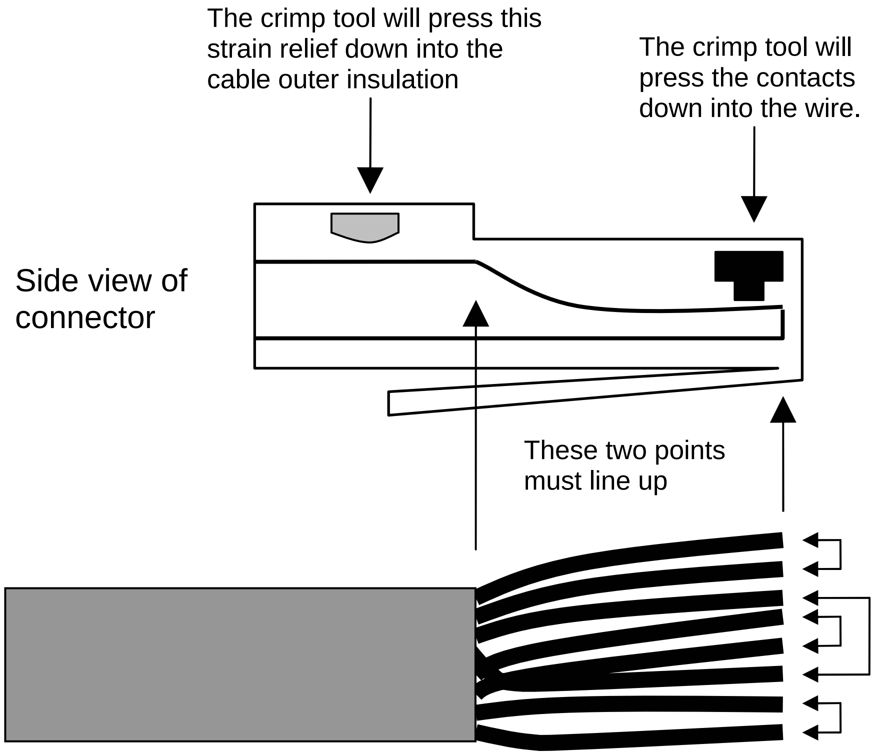
Step 5: Re-arrange the wires so that the pairing fits the following arrangement.



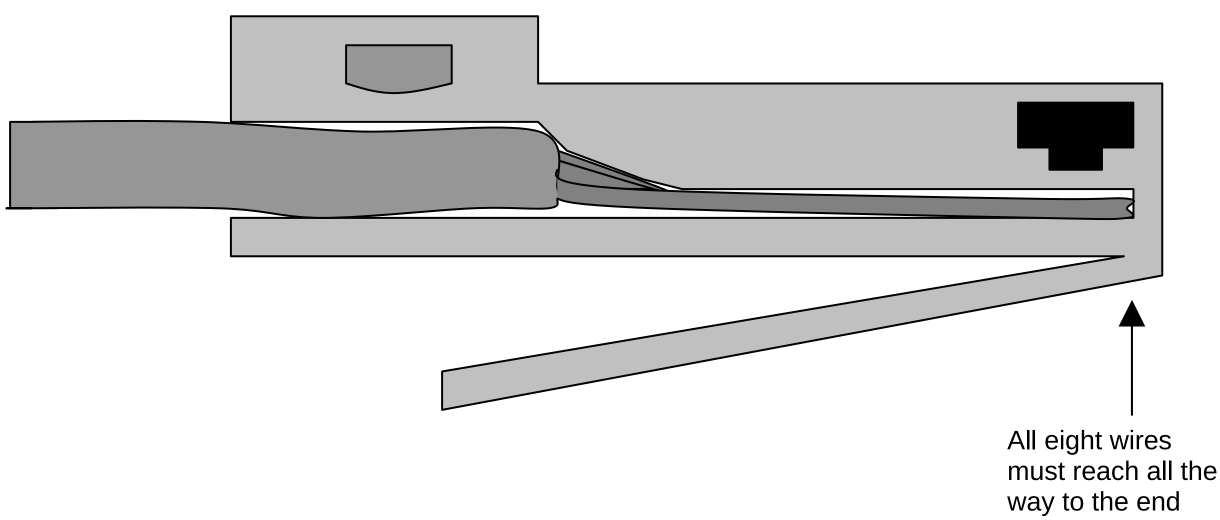
Step 6: Look at the connector, and cut the wires down to match. **Cut very straight.**



Step 7: Make sure the wires are the right length to reach the end of the inner guide, while the outer insulation reaches the point where the inner part becomes narrow. This is for electrical, as well as mechanical reasons. **You must <u>not</u> strip any insulation from the individual wires.**

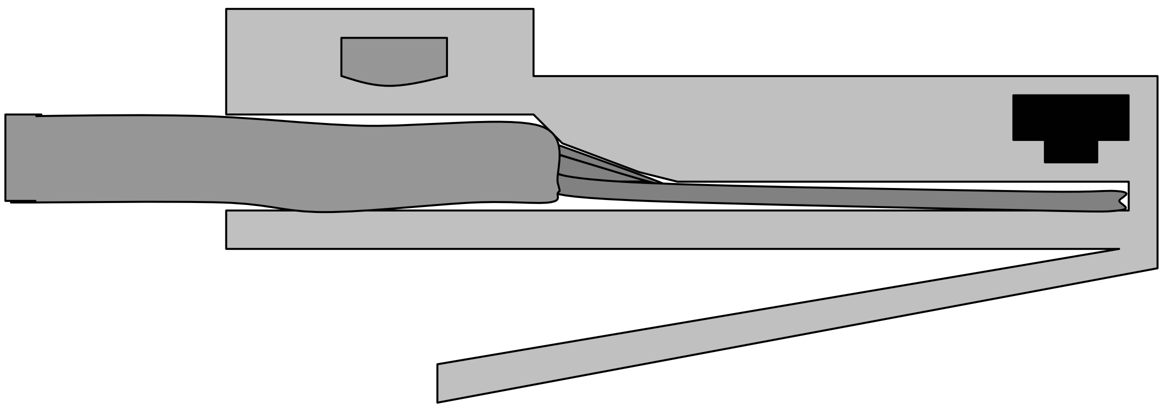


Step 8: Carefully insert the cable into the connector. Inspect it very closely, looking from the top, bottom and at the end to make sure **all eight wires** went in **all the way to the end.**

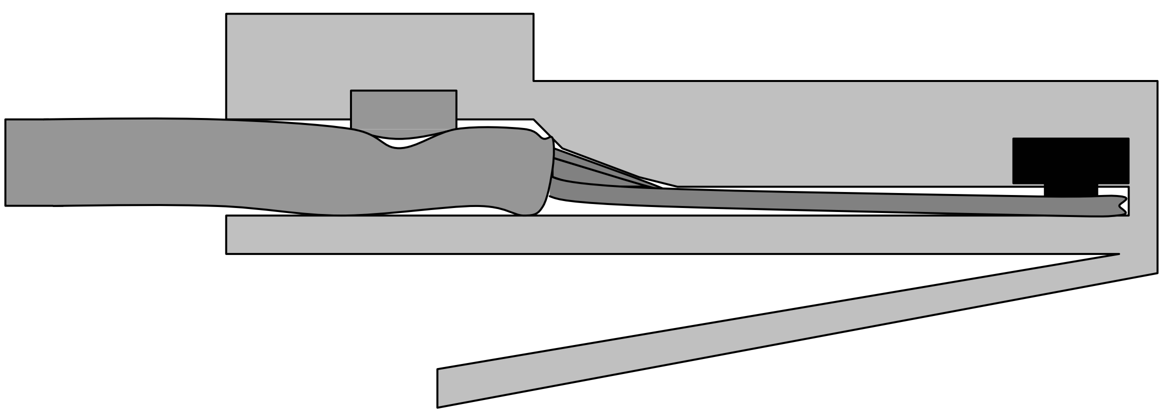


Step 9: Crimp the connector with the crimp tool. **You must use a proper crimp tool, designed to crimp 8-pin modular plugs, <u>not</u>** a screwdriver and hammer or some other improvised tool.

before crimp



after crimp



Step 10: Check again to make sure that the wires are properly paired, and did not get inadvertently switched in the process of putting them into the modular plug.

Repeat the above for the other end of the cable. **You must observe exactly the same pairing and color code scheme on the other end,** creating a properly wired normal (not crossover) CAT5 network connection. There is no "MDIX" feature in the product to switch transmit and receive pairs internally if miswired. There is also no automatic polarity inversion feature to automatically switch wires that are reversed within a pair.

Chapter

6

5-year limited warranty

AudioRail Technologies warrants this product to be free of defects relating to material and workmanship for a period of 5 years from the original date of purchase, subject to the conditions below:

- Minor cosmetic problems associated with the normal wear and tear that professional audio equipment is subject to, or which have no bearing on proper operation of the unit are not covered.

- Problems associated with accident, abuse or neglect by the user are not covered.

- Problems associated with subjecting the unit to operational and environmental conditions outside those specified by this manual are not covered.

- A unit that has been modified in any way by the user is not covered, if AudioRail Technologies, at its sole discretion, determines that the user modification has a cost impact upon the warranty claim or affects compliance to EMC or safety regulations.

- Repairs not carried out by agents specifically authorized by AudioRail Technologies may or may not void the warranty. AudioRail Technologies will review the nature and workmanship associated with unauthorized repairs. If it is determined by AudioRail Technologies that the unauthorized repair will cost AudioRail Technologies more than if it had been repaired by AudioRail Technologies in the first place, or the unauthorized repair affects compliance to EMC or safety regulations, then AudioRail Technologies reserves the right to consider the warranty void and charge fees it considers appropriate to properly repair the product.

AudioRail Technologies will either repair or replace, at its sole discretion, defective equipment legitimately under warranty, at its expense, and pay for shipping charges both ways (limited to U.S. domestic ground transportation). Contact AudioRail Technologies first, before shipping any product back to AudioRail Technologies.

This warranty is transferable. It is not affected by transfer of product ownership.

Please do not hesitate to contact us at (978) 461-0177 or info@audiorail.com on any issue that requires support. Your successful product application is of utmost concern to us.

Chapter 7

Specifications

Digital audio inputs..4 Alesis ADAT® Lightpipe optical connections

Digital audio outputs..………………….……….4 Alesis ADAT® Lightpipe optical connections

Sample rates supported…..................44.1K, 48K, (88.2K, 96K using ADAT® doubling format)

Maximum word length……………………………………………………………….…24 bits

Clock synchronization scheme…..................Each of eight ADAT® streams follows its source

Network…...AudioRail™ CAT5 daisy chain, 100 meters max. per hop

Latency......................................…..approx. 4.5 µs + 0.25 µs/hop + 0.005 µs/meter, end to end

Power……………………95-240 VAC (8 Watts @120 VAC, 12 Watts @240 VAC), 50-60 Hz

Operating environment……………….……………….0 – 50 °C, 10-90% RH, non-condensing

EMC compliance: Emissions…………..FCC part 15 (Class B), EN55013, EN55103-1* 1997

EMC compliance: Immunity……….……………………………………….EN55103-2* 1997

Safety...…...UL6500, EN60065

Size……………………………………………...……..…1U (1.75”) H x 19” W x 6.5” D

Weight………………………………………………………………………approx. 4.5 lbs.

*Complies with all EN55103 categories (E1 Residential through E5 Heavy Industrial)

Specifications subject to change without notice.

FCC Part 15 Class B compliance statement

RADIO AND TELEVISION INTERFERENCE

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- **Reorient or relocate the receiving antenna.**
- **Increase the separation between the equipment and the receiver.**
- **Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.**
- **Consult the dealer or an experienced radio/TV technician for help.**

You may also find helpful the following booklet, prepared by the FCC: "How to Identify and resolve Radio-TV Interference Problems." This booklet is available from the U.S. Government Printing Office, Washington D.D. 20402.

Changes and Modifications not expressly approved by the manufacturer or registrant of this equipment can void your authority to operate this equipment under Federal Communications Commissions rules.

CE Declaration of Conformity

Manufacturer's Name: AudioRail Technologies

Manufacturer's Address: 3 Silver Hill Road, Maynard, MA, 01754 (United States)

declares, that the product:

Product Name: AudioRail ADAT rx32tx32

Model Type: Alesis ADAT Lightpipe transport over AudioRail network

conforms to the following Standards:

EMC: EN55103-1:1997, EN55103-2: 1997

Safety: EN60065

December, 2003 Garth D. Wiebe
Principle Engineer and Proprietor