

Next Generation IP-based Audio

Leveraging Standard Protocols to
Simplify Setup and Promote
Sharing of Audio over IP Networks



Remember how we used to get
Finally, we're down to this
audio around our facilities



Now, Next Generation IP Audio

- Networked audio consoles have had a poor record of interoperability.
- Even if models have the ability to communicate, the complex setup and varying parameter choices by manufacturers effectively isolate one model of equipment from another.
- This is reminiscent of the early days of networking in the computer industry.

What is the Next Generation IP Audio

- Recently available protocols allow multicasting systems, like digital mixers, to automatically find and select the many settings required for network communication to work.
- Consoles can now find other audio equipment, negotiate communication methods and receive a list of available sources just by turning on the power.
- A growing demand by users will speed the adoption of standards by manufacturers and make it possible for consoles from a variety of manufacturers to send and receive audio with little or no user involvement either across the room or around the world.
- It is these new protocols that allow for the Next Generation IP Audio Systems

Who is Logitek?

Logitek has been making audio consoles for Radio for over 30 years. TV console for 3 years.

We have been making router based systems for over 12 years

Logitek is the first American manufacturer to make TDM based Console/Router systems with Over 2000 surfaces and 1700 Router Engines in service

Now that IP technology has advanced enough to become a user friendly, reliable protocol, we are introducing our first AoIP system.

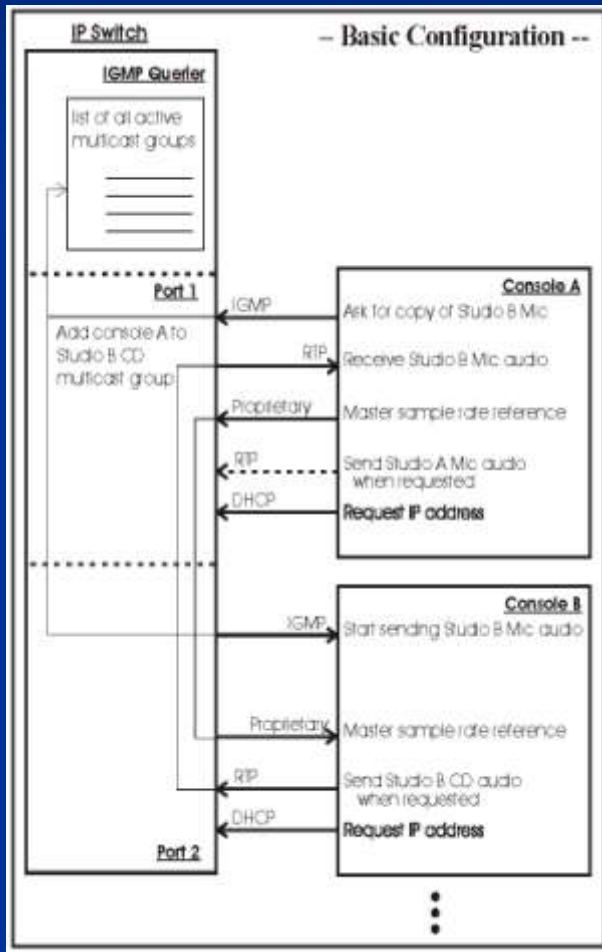
And , oh yeah, we don't make:




Logitek

Back to Basics

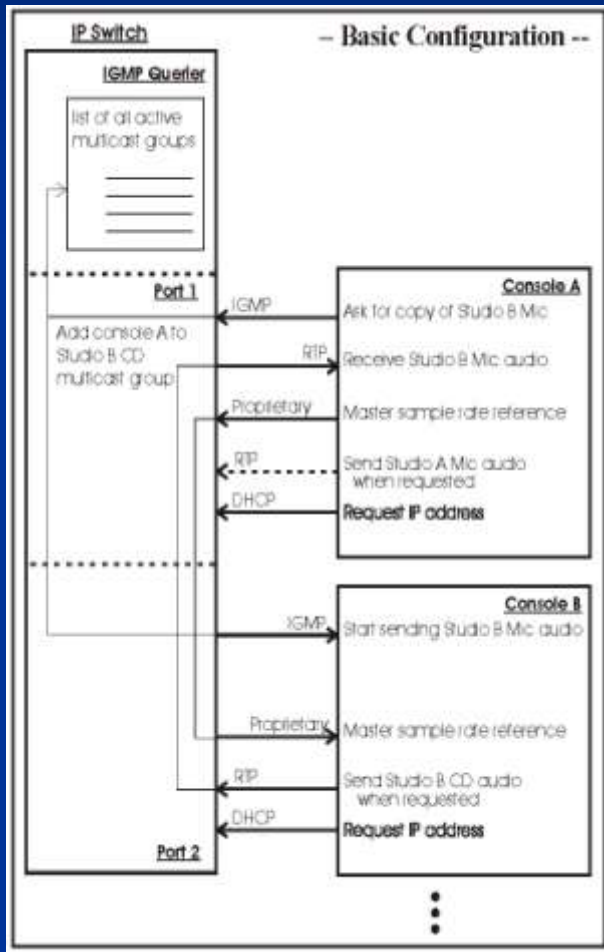
Basic Connectivity



- Current IP enabled audio systems connect together in the same way as office computers.
- Switches used for audio support the multicasting standard that allows a node or console to send one audio stream to the switch and have the switch deliver a copy to each node or console that requests one.
- The switch acts as a high capacity non-blocking audio router, only with less wiring and less cost.

Back to Basics

Basic Connectivity



Consoles (nodes) send commands in the IGMP protocol to start or stop the copying of audio from other networked nodes.

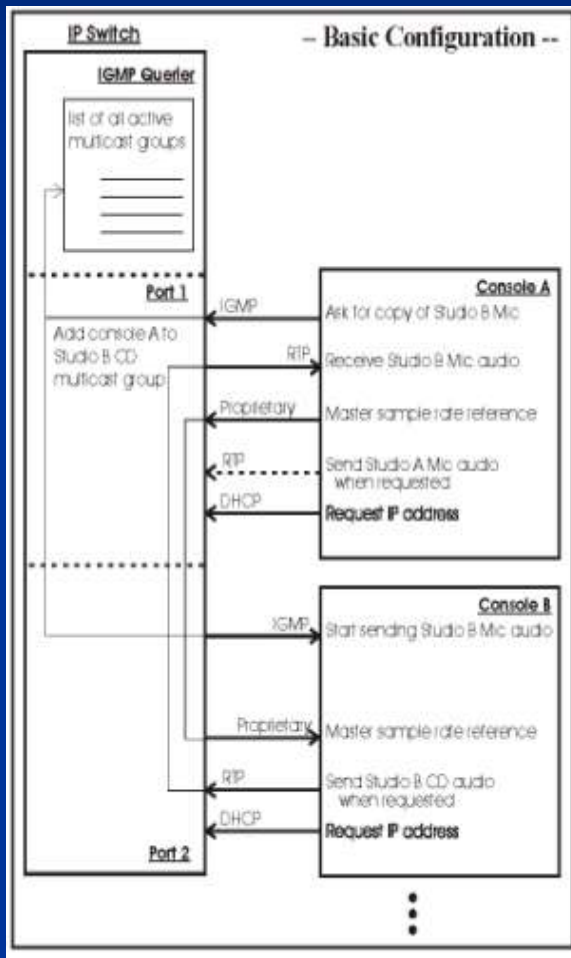
Software in the switch called a Querier keeps a list of all active audio streams and tells a console to stop sending any audio that is not being used by another unit.

The Querier is very important
It stops Network train wrecks



Back to Basics

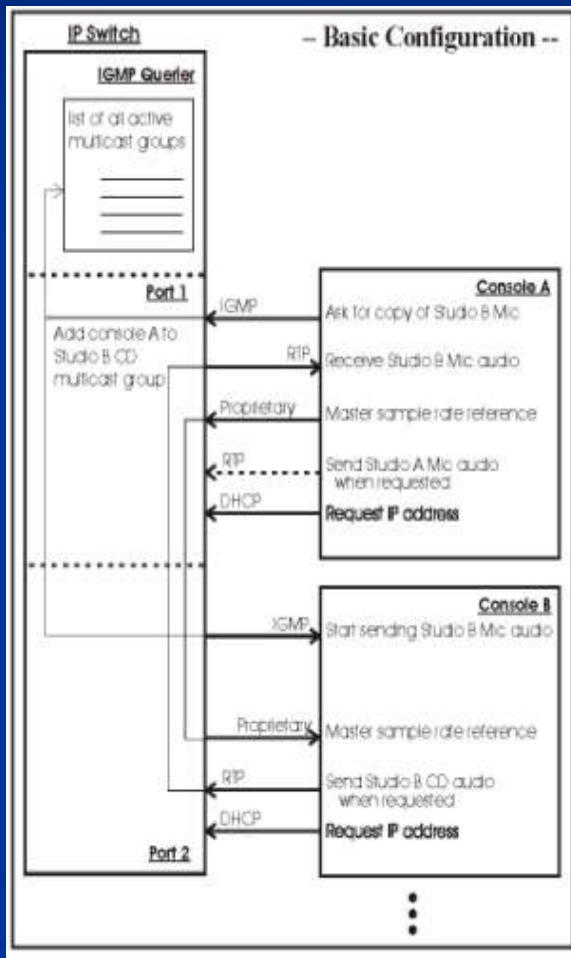
Basic Connectivity



- Audio in a console is broken into packets, labeled with a protocol called RTP and sent off to the switch on a regular basis.
- RTP provides format and sequence information plus a unique identifier so that the packets can be reassembled into audio at the receiving end.
- For speed and overhead savings, a tremendous amount of effort is saved if every piece of gear on the network produces these audio packets at the same rate.

Back to Basics

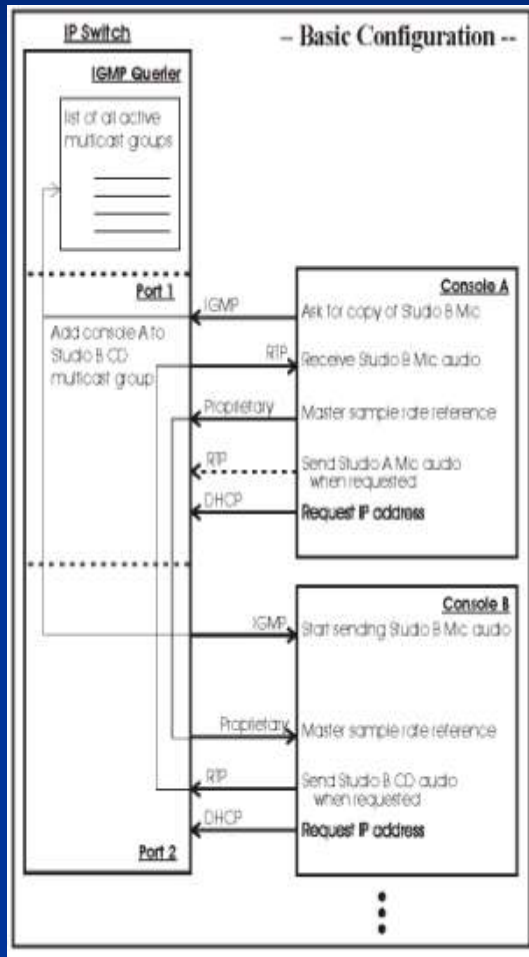
Basic Connectivity



- This is usually accomplished by having one node multicast sample rate information and having all other units adjust their internal rates to match.
- All IP based protocols use numeric addresses to accomplish their work and it is vital that these addresses be unique for each piece of hardware and each multicast audio channel.
- A protocol called DHCP is used to automatically assign unique addresses to hardware nodes but addresses for audio channels have had to be assigned manually.

Back to Basics

Basic Connectivity

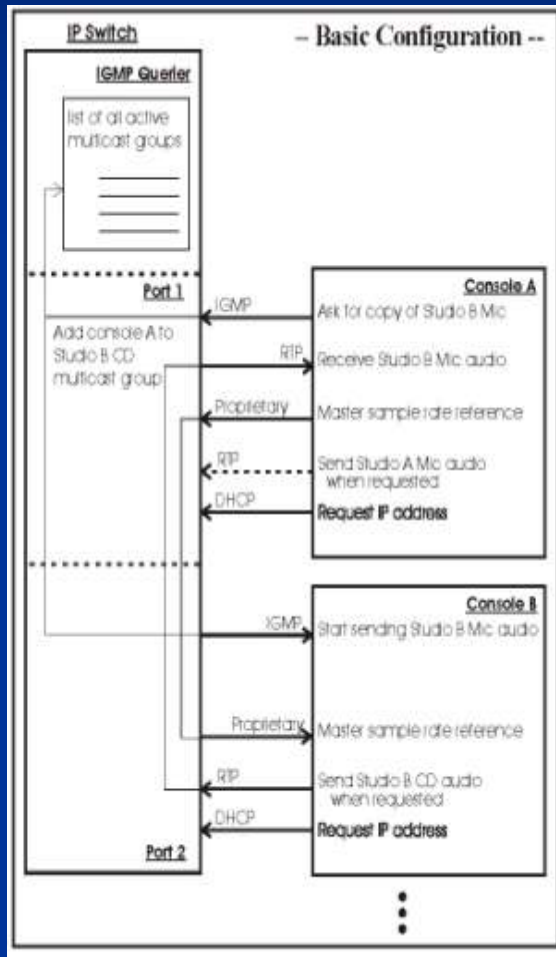


PROs – Current Systems

1. System nodes use Ethernet infrastructure with audio, control and synchronization all on a single cable.
2. The Ethernet switch acts like an audio router but at a cheaper price and with more flexibility.
3. The wiring between studios can be reduced to as few as two cables which greatly reduces the time and cost of installation.

Back to Basics

Basic Connectivity

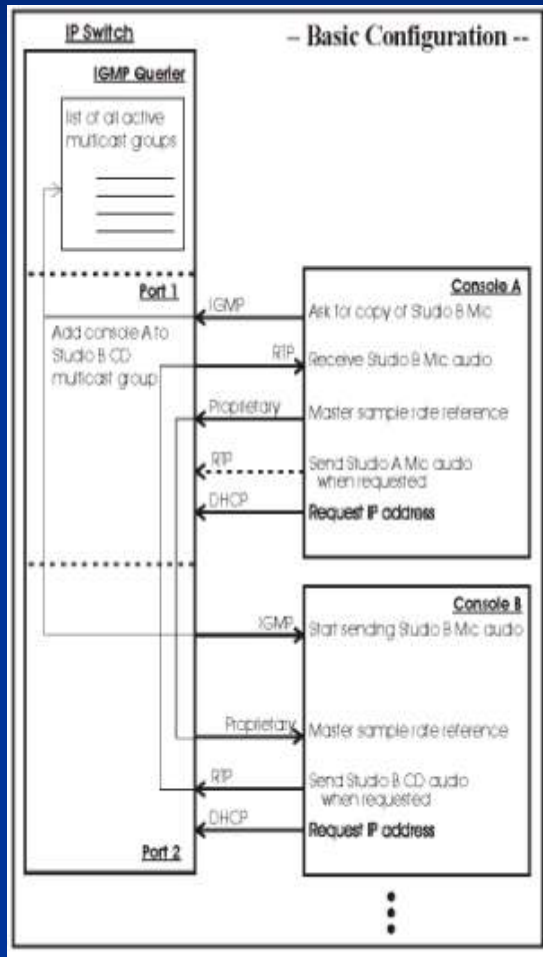


CONs – Current Systems

1. Manually assigning addresses is a time consuming, error prone task left over from the '90s. Long term management is usually a nightmare.
2. Each node added to the network needs to have the name and numeric address manually entered for each channel on the network.
3. Each new source needs to be manually entered into each node on the network.
4. There is no way to automatically advertise the name, features and address of networked channels.

Back to Basics

Basic Connectivity



CONs – Current Systems (cont.)

5. There is no way to automatically find out what channels are available from the other units
6. There is no way to easily include units outside the local network
7. There is no way to automatically handle different data formats, resolutions or sample rates.
8. Rate locking systems are inflexible and proprietary.
9. There is no common command protocol.
10. There is no inherent redundancy.

What's better

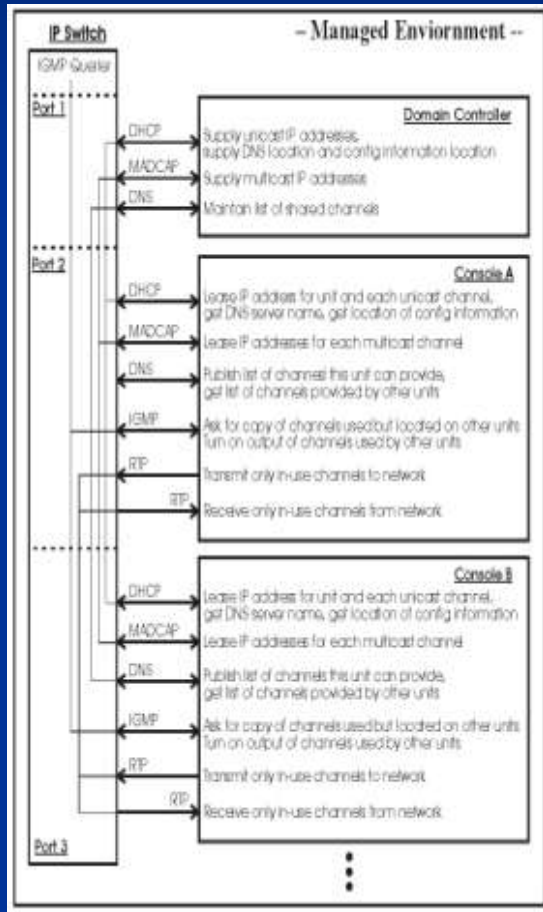
Fortunately, these issues can largely be solved by using widely available protocols from organizations like IETF (International Engineering Task force) and IEEE.

Software to implement these protocols has the advantage of being well documented, well tested, well supported, used worldwide and non-proprietary.



Advanced Systems

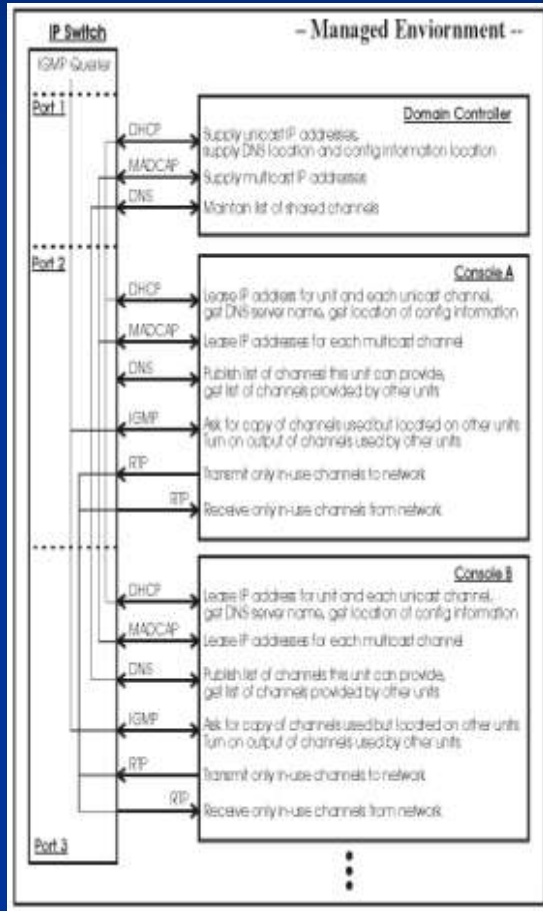
Unmanaged and Managed Environments



- Managed environments use a central configuration server to provide addressing and channel information. If no server is found, the system reverts to an unmanaged environment.
- The managed environment adds speed and flexibility to the audio network.
- In a centrally administered system, a node that has just been turned on will ask for an IP address for each of its Ethernet ports using the DHCP protocol. The DHCP server keeps a list of all addresses in use and can quickly assign an address without the need to pole every unit on the network looking for conflicts.

Advanced Systems

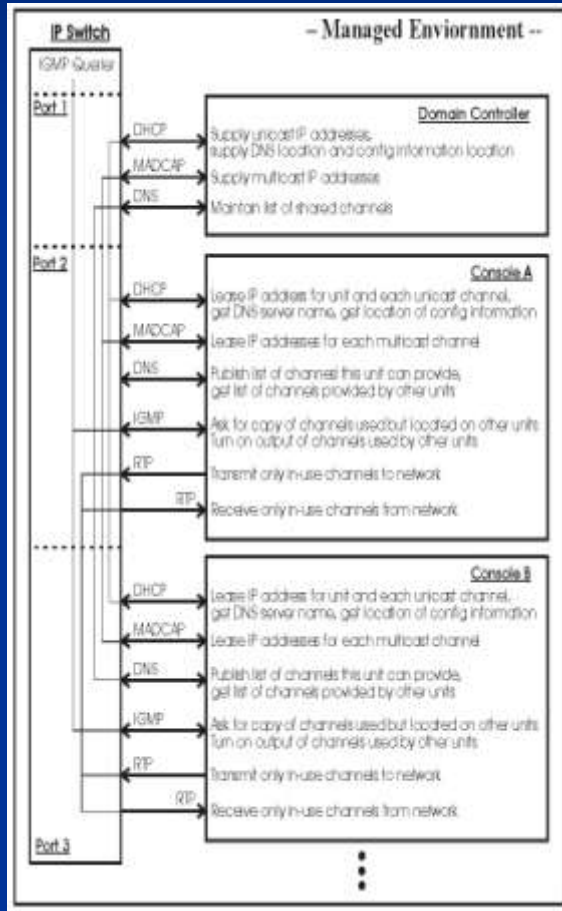
Unmanaged and Managed Environments



- The node will also be sent the address of a DNS server and, optionally, the location of a configuration folder. The node can check the folder for configuration changes and software updates.
- This makes replacing a broken unit as easy as turning it on and setting the name of the new unit to the name of the broken one. No other configuration is required.
- In the unmanaged environment these functions are performed by AutoIP and uPNP.

Advanced Systems

Unmanaged and Managed Environments

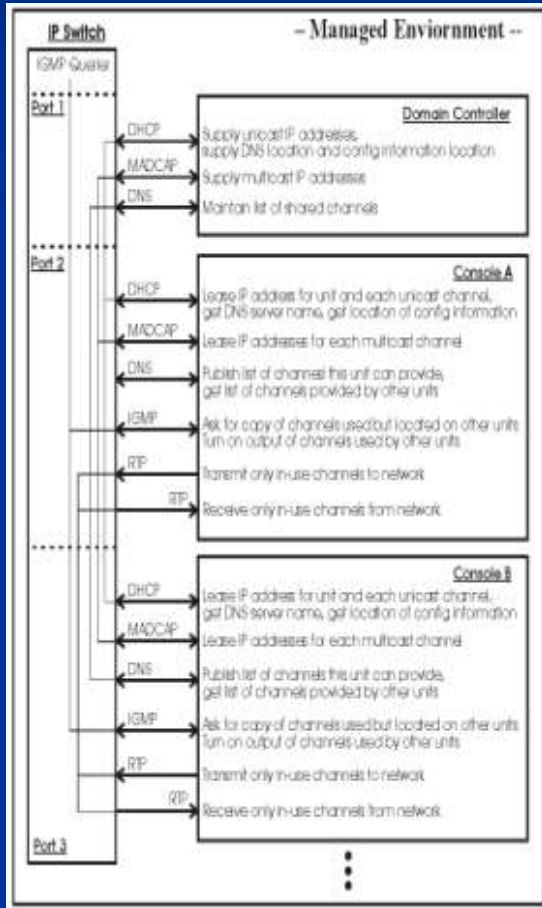


- The node will request an address for each audio channel using the MADCAP protocol (Multicast Address Dynamic Client Allocation Protocol).
- The server keeps a list of all assigned addresses and can quickly return an unused address without the need to poll other units for conflicts.
- Once all the addresses have been automatically assigned, the node will send a list of all its shared audio channels to the central server using the DNS protocol.
- It will then receive a list of channels available on other units from the server.
- The system is now fully configured and ready to use.

Advanced Systems

Unmanaged and Managed Environments

Clocking & Format Negotiation

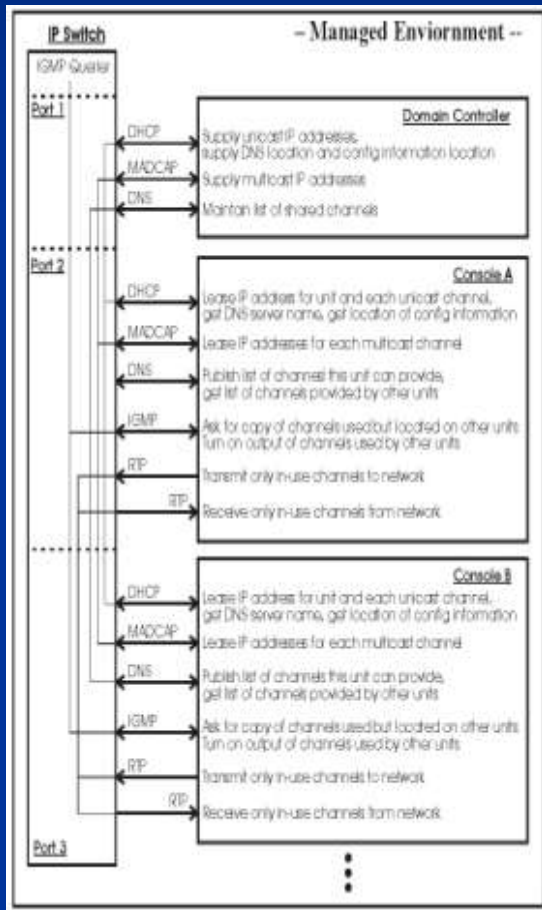


- It is possible that audio from different nodes or systems will be sampled at different rates or different resolutions from each other.
- Some audio may also be compressed for transport across a narrow bandwidth path like the internet or an ISDN link.
- The RTP protocol labels each packet with the format of the audio it contains but doesn't deal with what happens if the receiving unit can't use the format provided.

Advanced Systems

Unmanaged and Managed Environments

Clocking & Format Negotiation

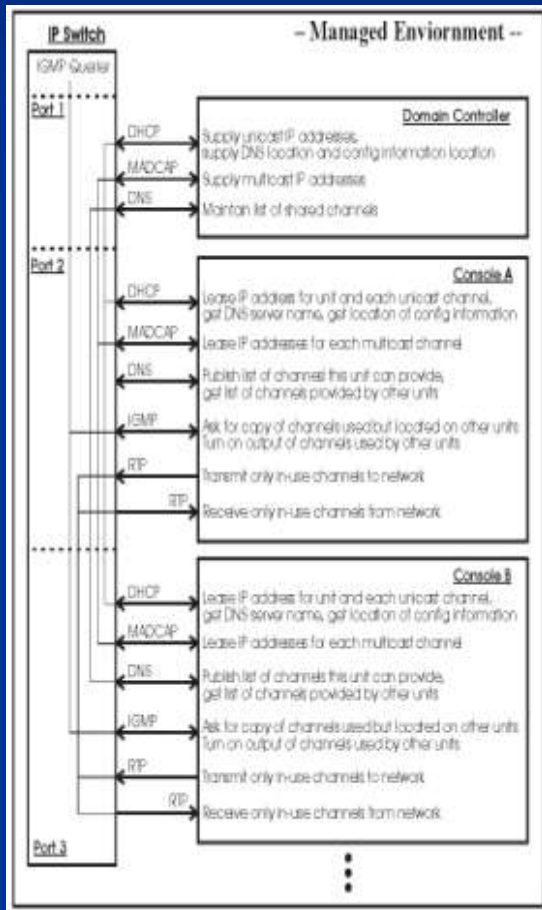


- A protocol called SIP allows a transmitting end and receiving end to negotiate an audio format or codec that both ends support.
- SIP is especially useful for outside broadcasts where the incoming program audio and the return cue channel often have different formats and quality levels.

Advanced Systems

Unmanaged and Managed Environments

Clocking & Format Negotiation

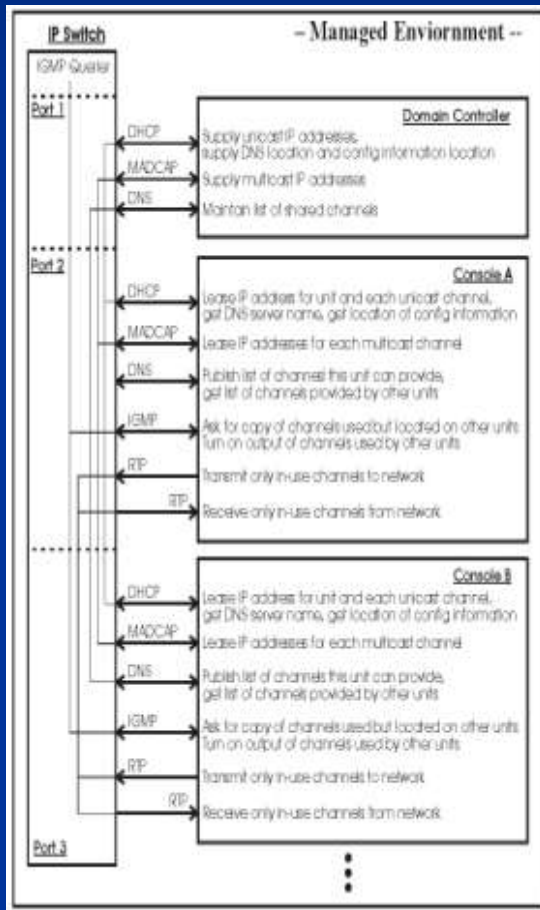


- Digital systems today all use noncompatible, vendor specific methods to maintain an identical sample rate between units. Most methods revolve around having a master unit multicast messages on a well known address at regular intervals.

Advanced Systems

Unmanaged and Managed Environments

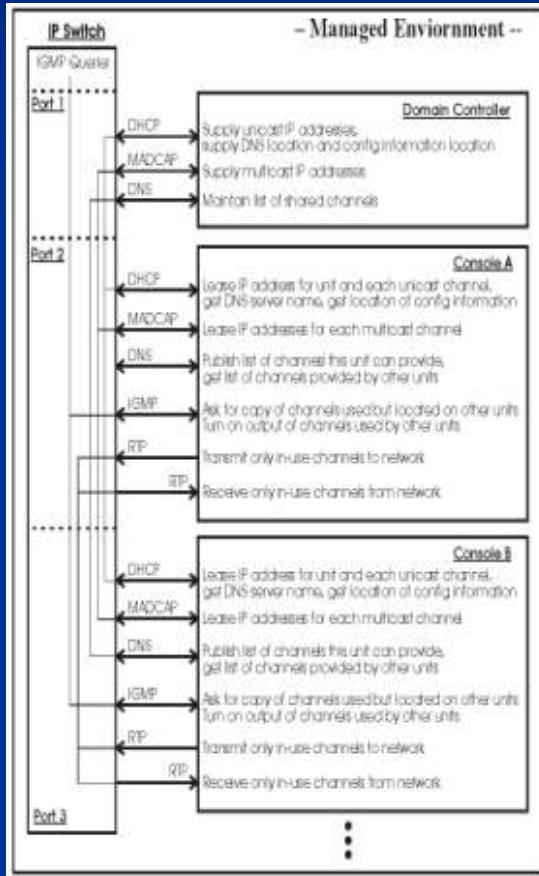
Clocking & Format Negotiation



Again, there is a new protocol designed for timing and synchronization. IEEE 1588 is a standard protocol designed to lock both the sampling frequency and the time of day to a master clock. 1588 is by far the most accurate network timing method available but requires help from the network interface controller that some older integrated circuits can't provide.

Advanced Systems

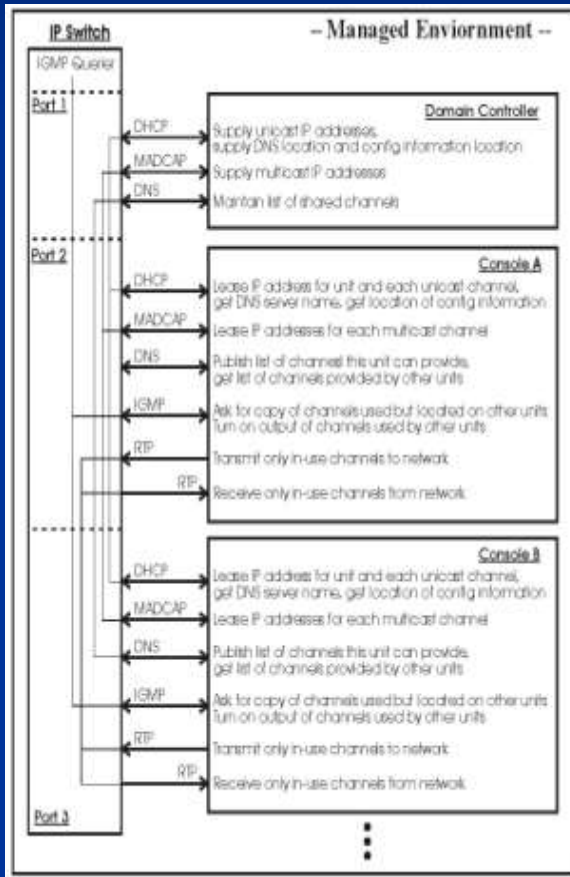
Unmanaged and Managed Environments – Standards Groups



- There are a great many standards specified for IP networks and it can often be difficult to figure out which ones are useful or have wide support among vendors.
- In an effort to promote compatibility between IP-audio products the European Broadcasting Union has issued a specification called N/ACIP that lists a minimum group of protocols and audio formats that should be supported by any piece of gear used by an EBU member.
- The N/ACIP spec is beginning to be adopted by manufacturers world wide as customers demand easier use and multi-vendor compatibility.

Advanced Systems

Unmanaged and Managed Environments

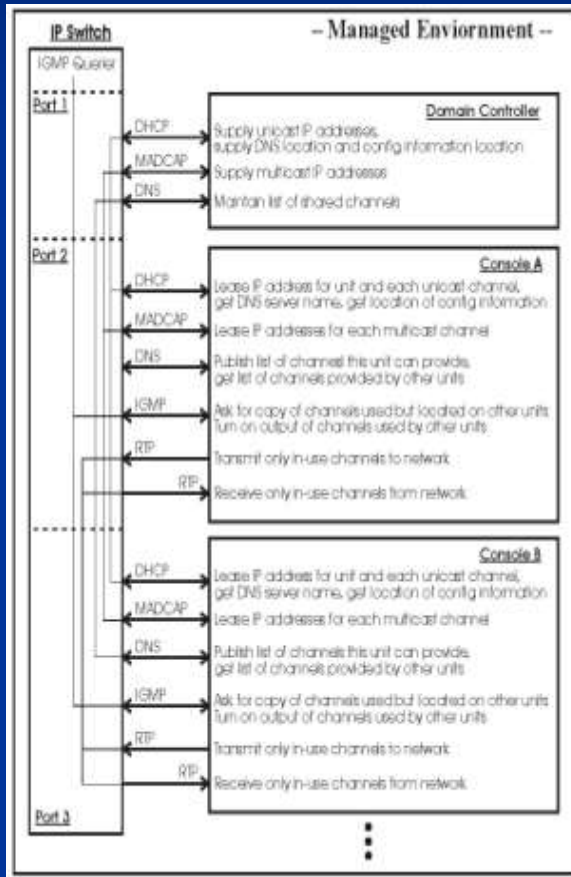


PROs – Managed Environment

1. Managed systems work well in both small and large installations.
2. A single location for setup information means less control overhead and more room for audio channels.
3. The list of shared channels can be acquired in seconds.
4. Only channel names require manual entry. Addressing is automatically generated.
5. New or replacement units are configured in seconds.

Advanced Systems

Unmanaged and Managed Environments

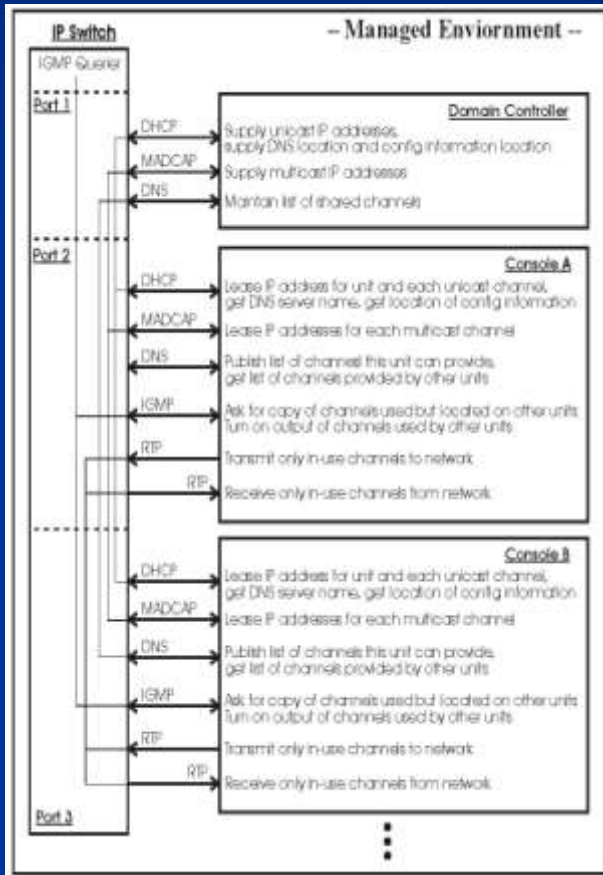


PROs – Managed Environment (cont.)

6. The channel lists can be shared between sites via Dynamic DNS.
7. Units can automatically negotiate a common audio format via SIP.
8. Mixing unicast and multicast channels conserves capacity and helps speed time sensitive switching.
9. Sample rates can be synchronized between groups or sites.
10. Support provided for N/ACIP required protocols.

Advanced Systems

Unmanaged and Managed Environments



- ### CONs – Managed Environment
1. A server computer is required.
 2. Some domain setup is needed.
 3. Managed systems can be more costly for small installations.
 4. Security setup is required for networks that are visible from the internet.

What does this mean to you

Network standards have been developed by the business world to make their systems more economical and useful by taking much of the drudgery and error out of network installation and maintenance.

The same savings can be realized by adopting these standards in the broadcast world with the added benefit of encouraging equipment from different vendors to “play” together.



What does this mean to you

- Simpler easier setup
- Lower Cost
- Easier Administration
- Opening up of the possibility of interoperability between systems.
- Automated ability to negotiate protocols with remote devices such as audio codecs.
- We are just beginning to see the first of these advanced IP Protocol systems.



JETSTREAM (MINI)



Next Generation IP-based Audio Router



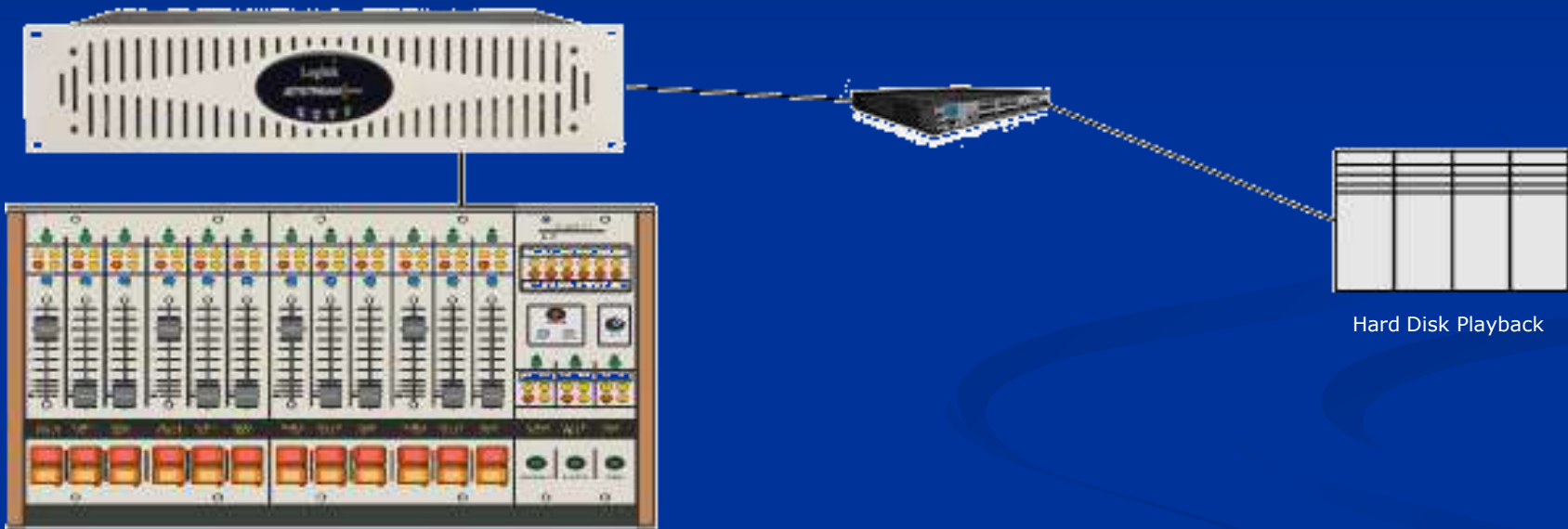
Next Generation standard IP protocols.
Console DSP with audio processing and profanity delay.
Logitek is pleased to introduce the **JETSTREAM** series of IP-based audio routers of IP-based audio routers, and embedded, microprocessor enabled, AoIP audio networking.

All this in a fanless, convection cooled, two rack unit package which uses the most recent IP protocols to make setup and maintenance fast and easy.

The Logitek logo, featuring the word "Logitek" in a stylized font with a graphic of a curved line and two red squares above it.



JETSTREAM (MINI)

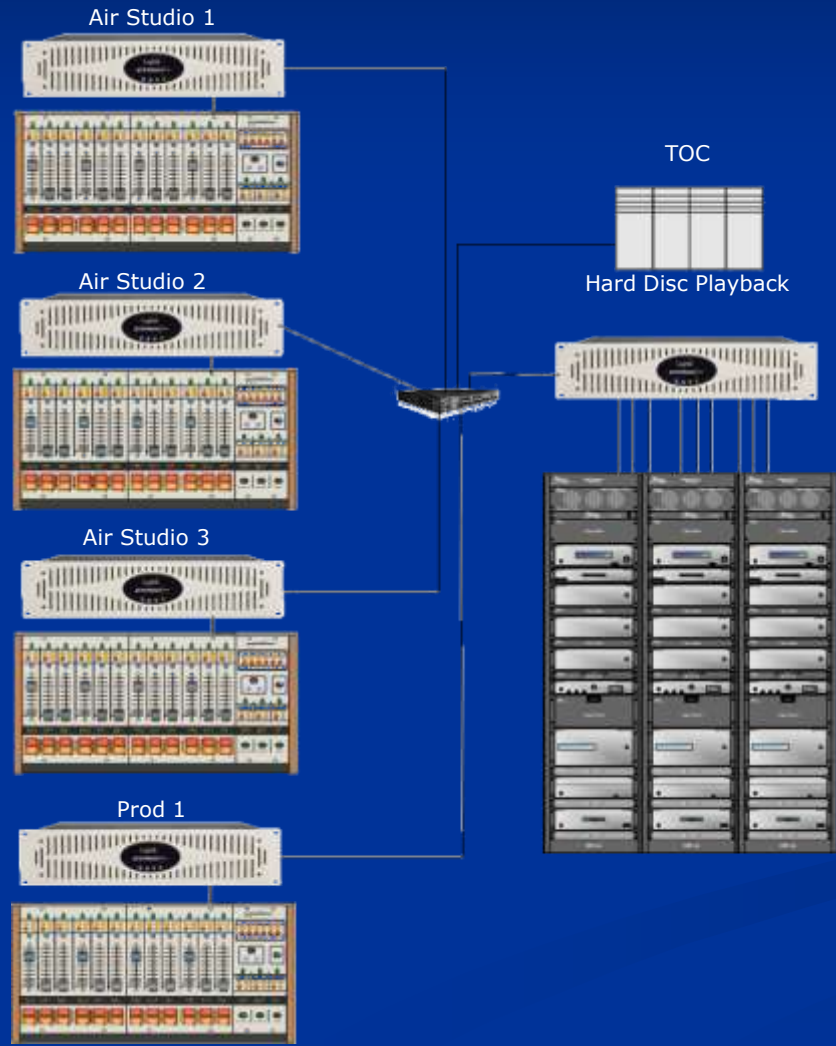


The JetStream Mini can be used to power a stand alone console in a single studio application





JETSTREAM (MINI)

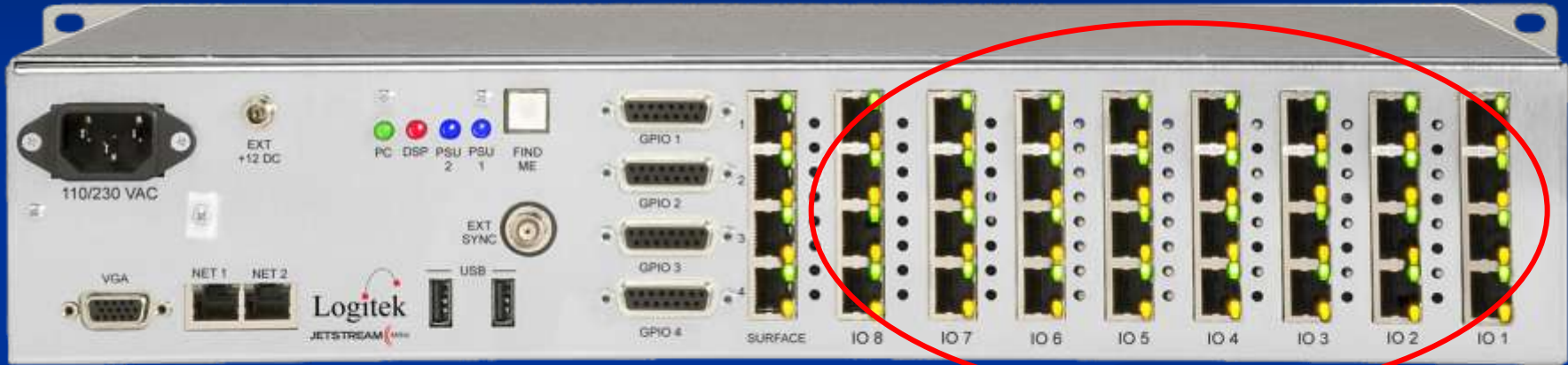


or act as the backbone in a much larger facility routing matrix.





JETSTREAM MINI



The Audio I/O for the Jetstream Mini is configurable by the user to fit their particular studio requirements. There are 5 audio cards: mic preamp input cards, digital and analog input cards and digital and analog output cards. These cards can be placed in 8 slots to uniquely configure each Jetstream Mini. A fully loaded JetStream Mini is functionally a 64 channel audio node.

The Logitek logo, featuring the word "Logitek" in a stylized font with a blue and red graphic element above it.



JETSTREAM (MINI)



The JetStream Mini is available with or without networking. For those that just need one studio today, you can save the on initial cost of the networked version,

The Logitek logo, featuring the word "Logitek" in a serif font. Above the letter "i" is a stylized graphic consisting of a dashed black arc connecting two red squares.



JETSTREAM (MINI)

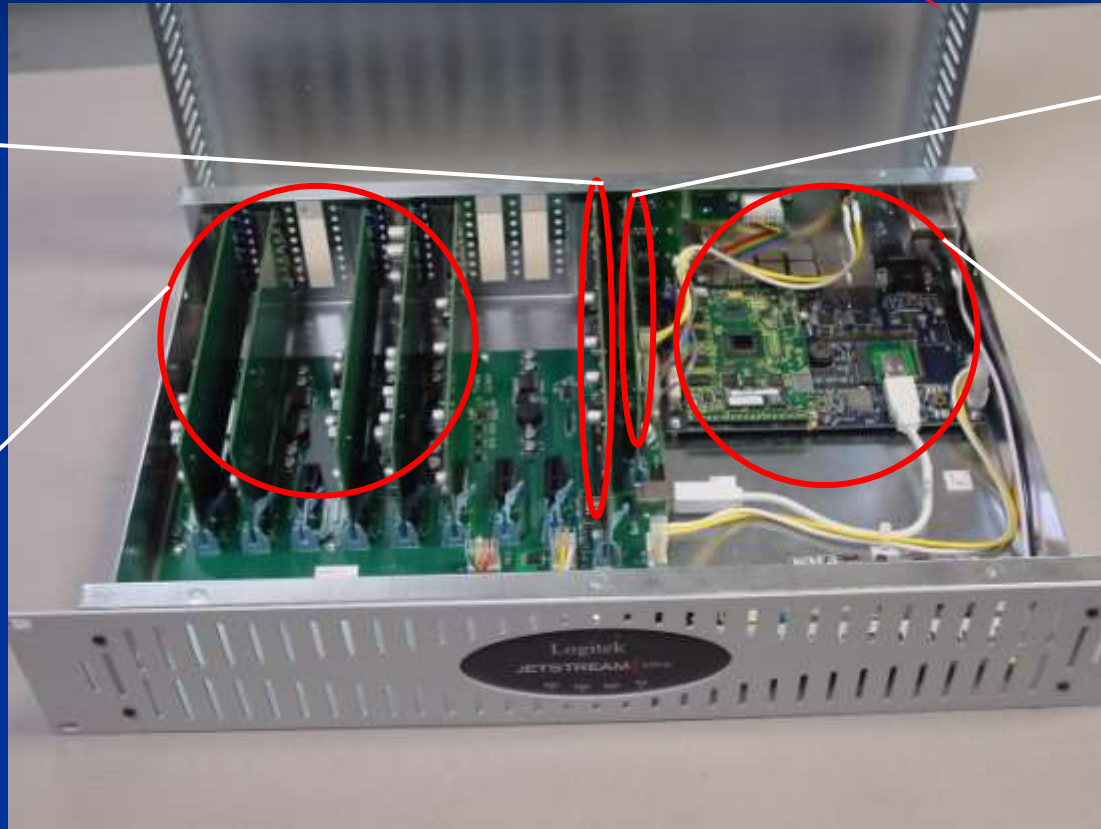


DSP

GPI/O

Audio I/O

Networking

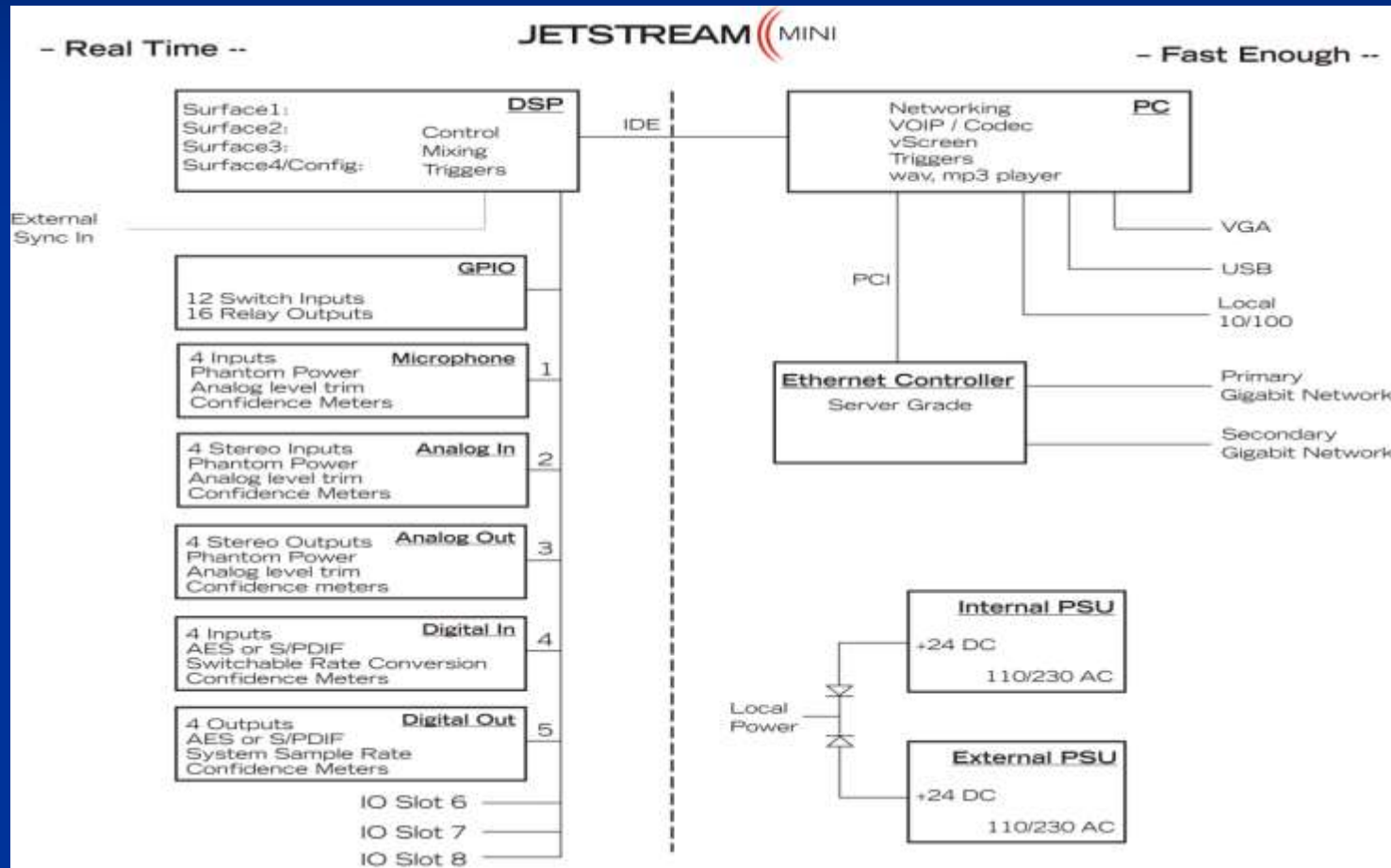


but have the ability to add the network component later. Everything about the Mini is configurable to your needs and budget.

The Logitek logo, featuring the word "Logitek" in a stylized font with a red square and a dashed line above it.



JETSTREAM MINI



Logitek



JETSTREAM MINI



FIND ME



Redundant, fail-over networking

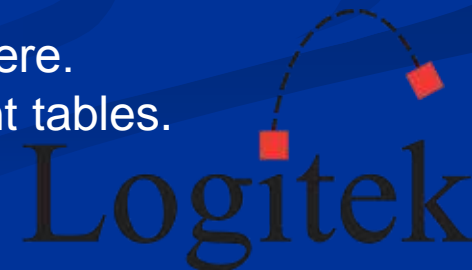
Using advanced standardized protocols such as Multicast DHCP, Auto IP and Universal plug-and-play, the JetStream Mini is easy for the operator to install. Just name the sources in a Mini, and it makes all other Minis in the system aware of its existence.

Just press the FIND ME button and the system knows you're there.

There is no longer a need for filling out cumbersome assignment tables.

And if you have a small system, three or fewer Minis can

be networked without the need for an outboard network switch.

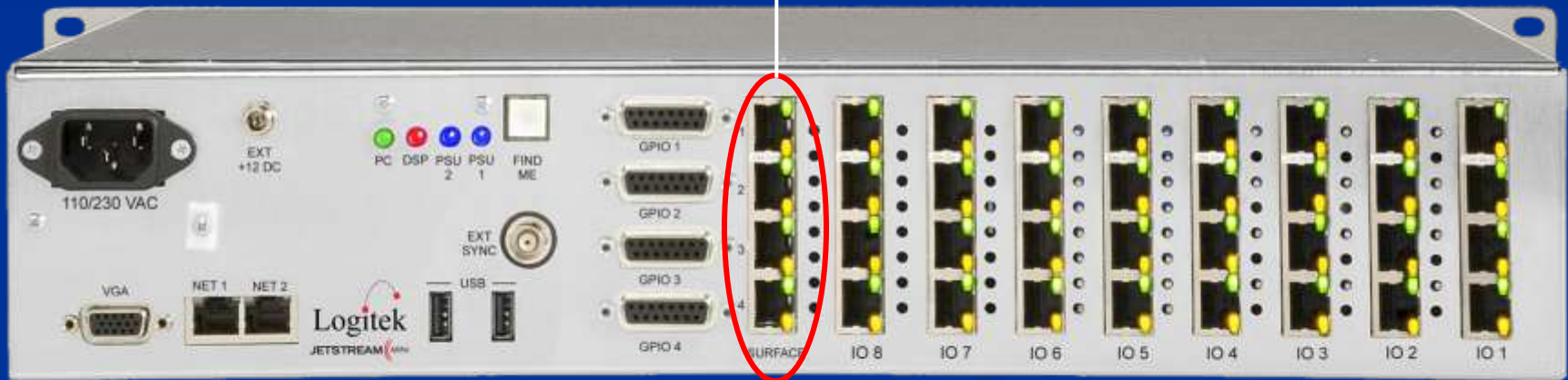




JETSTREAM MINI



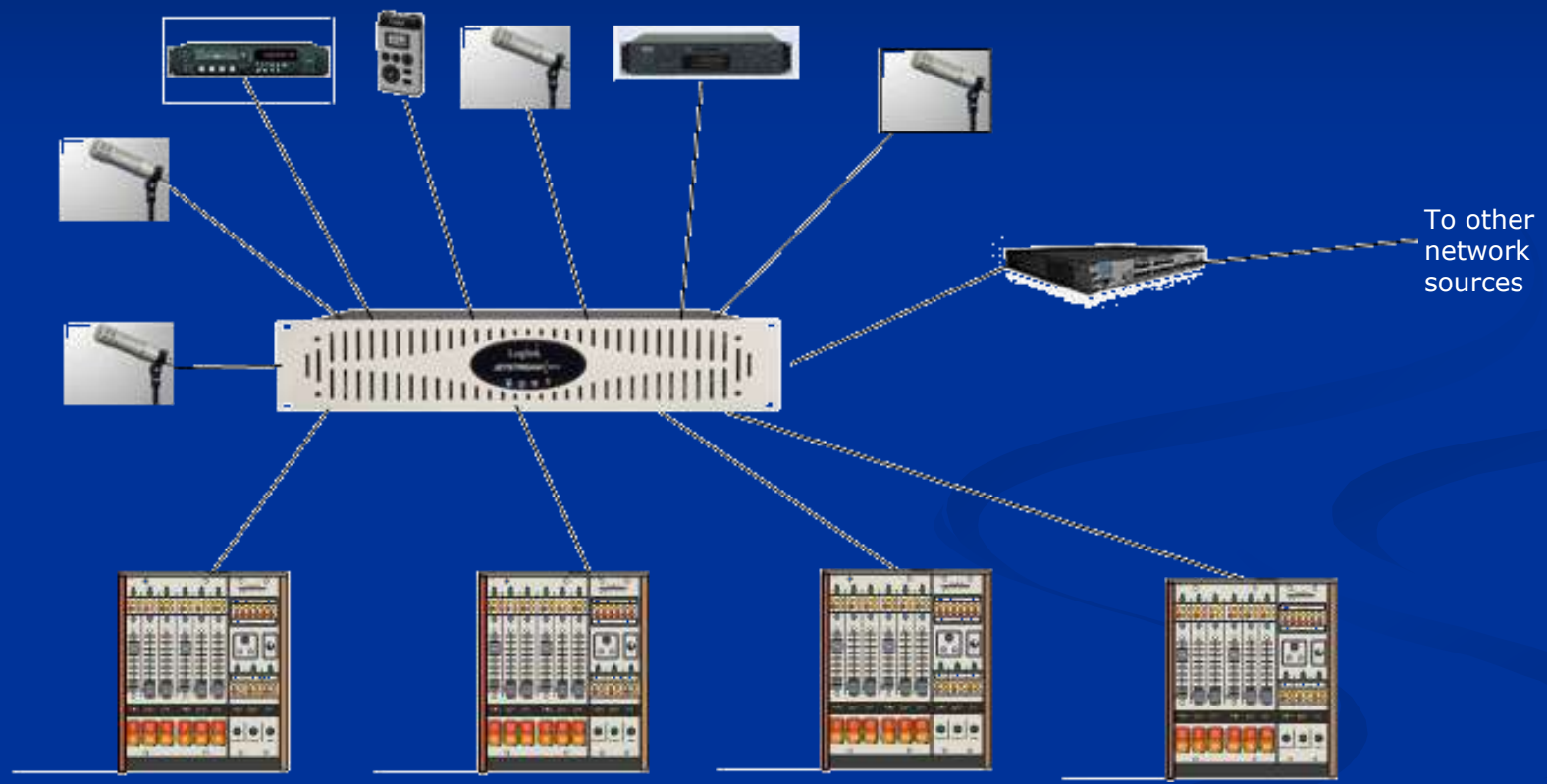
Multiple Console Support



The DSP in the JetStream Mini is also designed for the greatest flexibility and cost savings. Each mini provides multiple output busses and buckets of mix minus. A single Mini can drive as many as 4 consoles because the console DSP can be parsed out by the fader.



JETSTREAM (MINI)

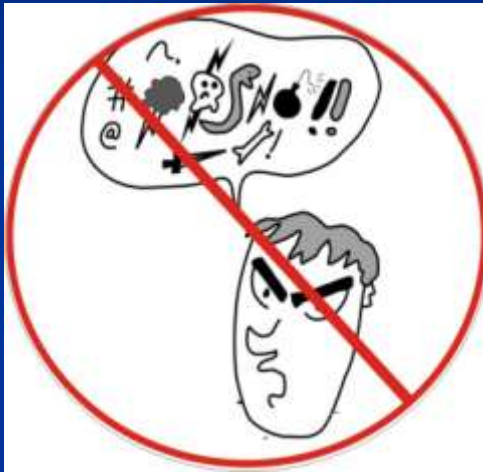


Multiple Console Powered by the same Engine with shared sources such as a News Prep area





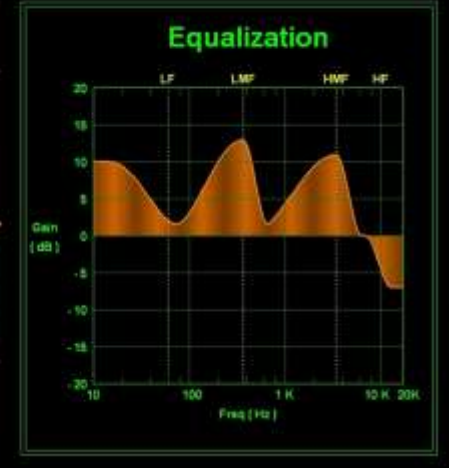
JETSTREAM MINI



Media Plyr

Hi F	10,000
Hi G	-7
HM F	3,400
HM G	+11
HM BW	3,000
LM F	370
LM G	+13
LM BW	630
Lo F	60
Lo G	+10
Mode	IN

Laptop Studio 2

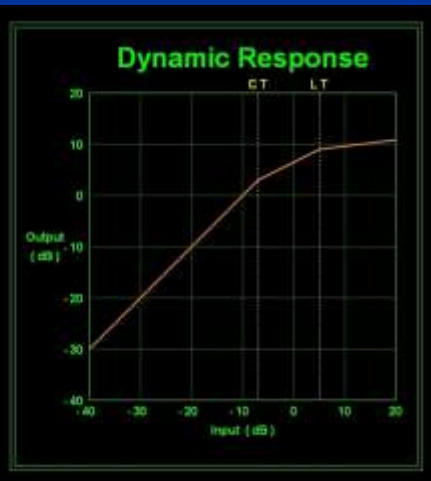


Media Plyr

L THR	+5
L RAT	8
L REL	10
C GAIN	+10
C THR	-7
C RAT	2
C ATK	40
C REL	3000
MODE	IN

Mic 1 Studio 1

Laptop Studio 2



And each JetStream Mini comes equipped with EQ and dynamics on each fader and the busses; and profanity delay as standard features.

Logitek



JETSTREAM MINI



JetStream can be used to control all the Pita and legacy consoles

Logitek

Pilot



+

JETSTREAM (MINI)



=

Low entry price and low operating cost

Logitek



JETSTREAM (MINI)



With the adoption of standard IP protocols manufacturers will be able to develop systems that will have interoperability. The huge advantage to the broadcaster is that they may once again be able to purchase equipment from different manufacturers and realize interoperability in the same facility as part of the same system.

IP Audio

More Net. Less Work.

