



Reinventing Intercom with SMPTE ST2110-30

Martin Dyster – The Telos Alliance
InfoComm June 2019

A Short History Lesson



Dateline: 2004

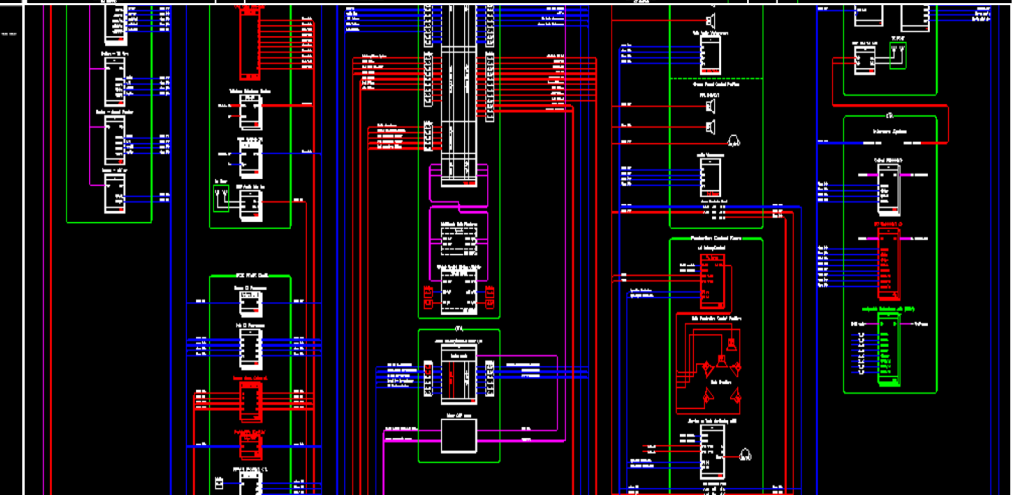
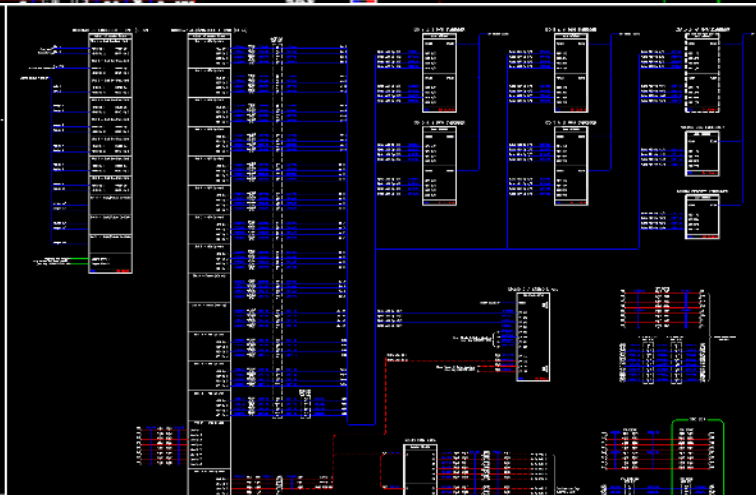
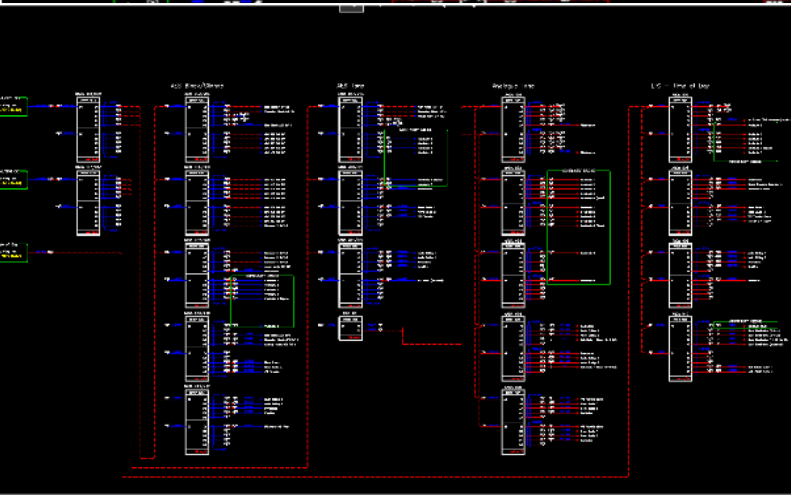
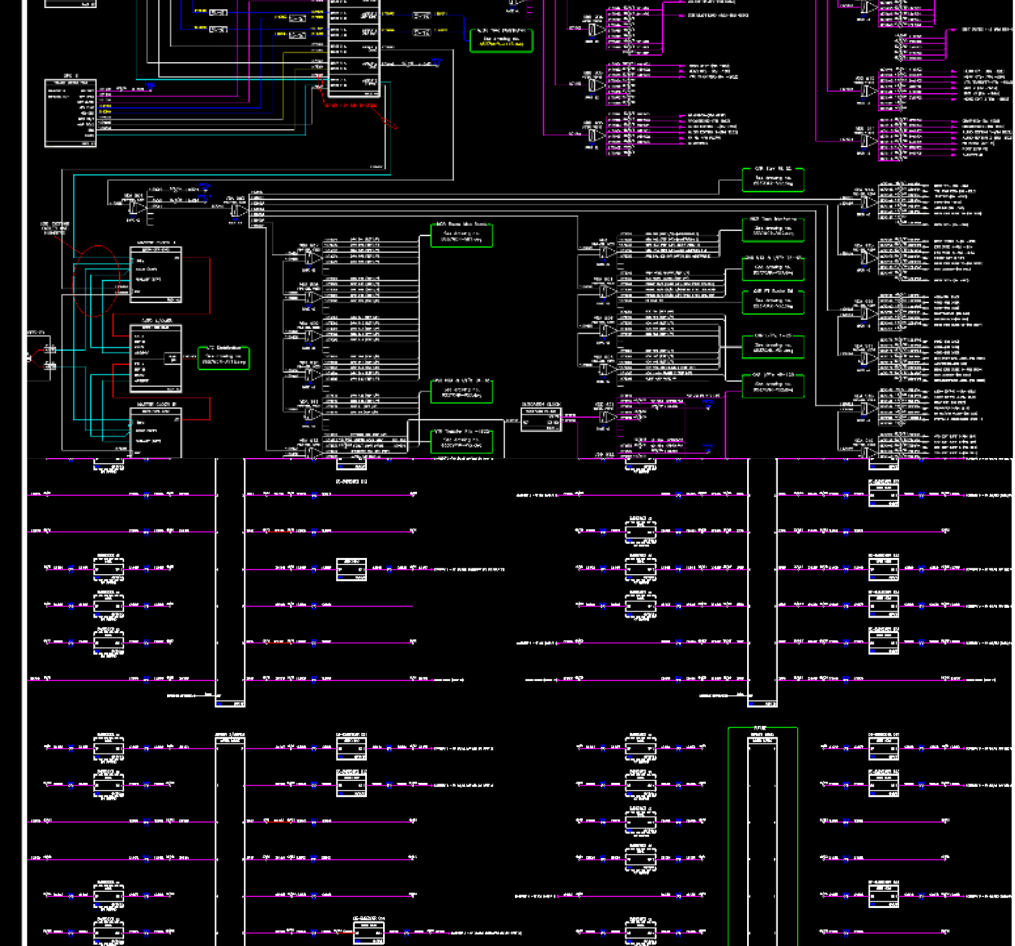
A photograph showing a person's legs and feet sitting amidst a massive, chaotic pile of network cables in a server room. The cables are of various colors (black, white, yellow, blue) and are tangled together, creating a dense, almost impenetrable mass. The person is wearing light-colored trousers and dark shoes. The background shows more cables and server racks, emphasizing the complexity of the network infrastructure.

**System Integration Before
Audio over IP!**

Typical broadcast audio system install: 2003/4

- RTP in Lisbon, new HQ went live April 2004 in time for Euro Soccer Championships
- AES-3, Analog and SDI audio throughout
- In excess of 10,000 individual audio pairs.
- Everything through patch bays
- Hundreds of miles of cable
- Tens of thousands of hand soldered, crimped or punched connections





Audio over IP: A Revolution





2003: The Telos Alliance (Axia) Introduces Livewire, The First AoIP Protocol

2004: First AoIP Broadcast Studio Created: Auburn University's WEGL-FM 91.1 in Alabama

2006: Audinate Introduces Dante IP Audio Protocol into Pro Audio Market

2008: Wheatstone WheatNet AoIP Protocol Comes to Market

2010: ALC Networks Introduces Ravenna

2010: The Audio Engineering Society Forms the x192 Working Group

2013: The AES67 Standard Is Ratified

2014: Media Networking Alliance Launches

2015: Alliance for IP Media Solutions Formed

2017: SMPTE publishes SMPTE ST 2110, adapting AES67 as the audio format 2110-30

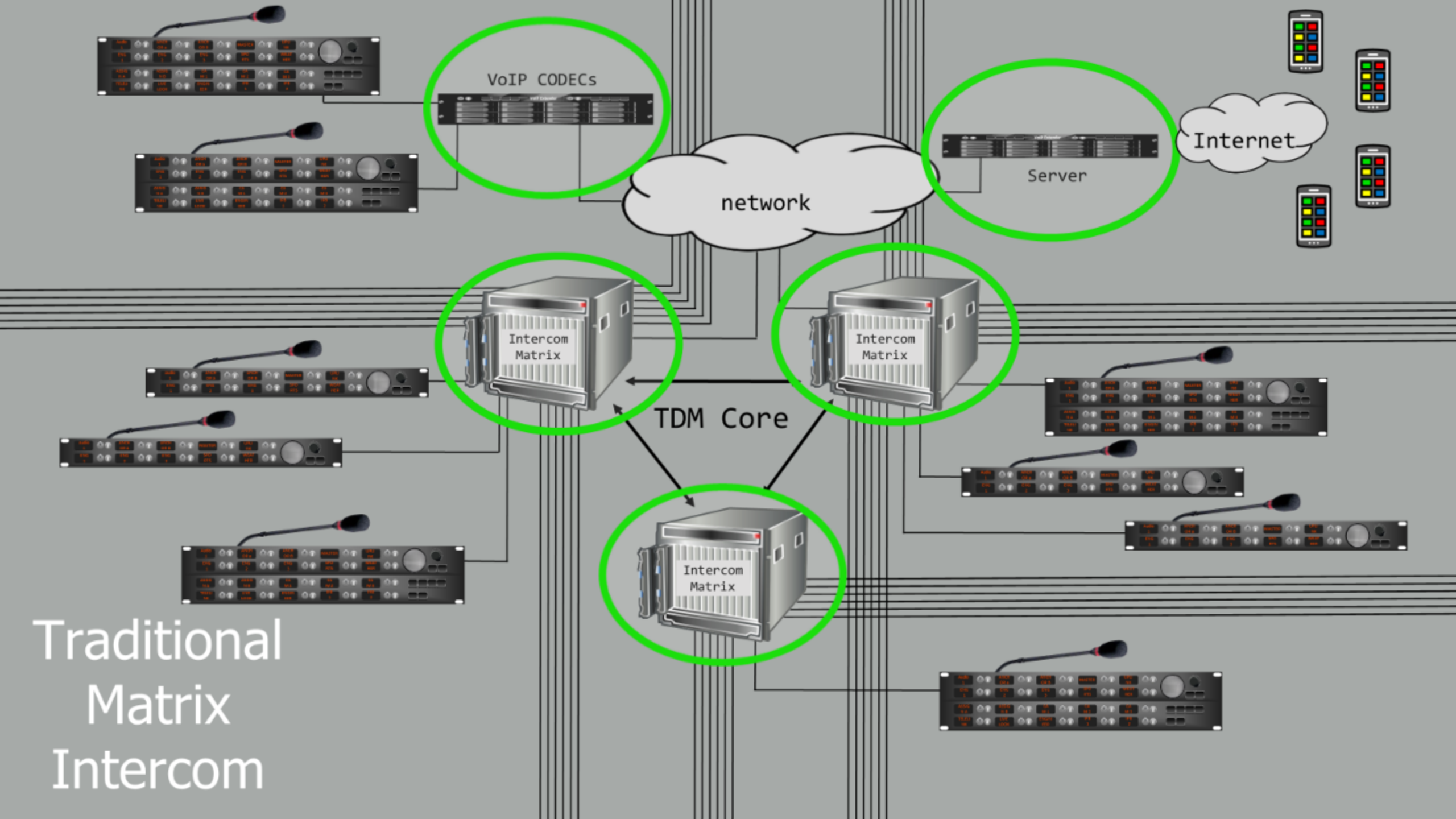
Audio over IP: The Payoff



- Standard Based – AoIP uses Layer 3 switch fabric, PTP Clock, RTP and SDP
- Switched Multicast removes the need for DAs, Patch Bays, Centralised Routers
- Reduces cabling, saves rack space, less complex design and associated costs
- Uncompromised audio quality - 24bit/48Khz low latency
- Digitized signal path at source
- No complex House-Sync distribution system
- Fewer single points of failure

INTERCOM

LIFE AS WE KNOW IT



Matrix Intercom, a practical solution?

- Partylines, IFBs and Groups all require mixing
- Matrices can store configurations
- Gain controllable I/O and crosspoints
- Signal detection and monitoring is easy
- Matrices are resilient – Dual Power/Dual CPU
- Wide range of connectivity options
- A known commodity

Matrix Intercom is not such a practical solution

- Lack of flexibility – size limited by port count
- Channel handling limited by buss structure
- Difficult (and expensive) to trunk multiple frames
- Takes up rack space
- Everything cables back to central point
- AoIP connected matrix systems also need Ethernet fabric
- Single points of failure

THE CHALLENGE: BUILD A MATRIX-LESS PURE IP INTERCOM

An IP Intercom from scratch: The Constitution



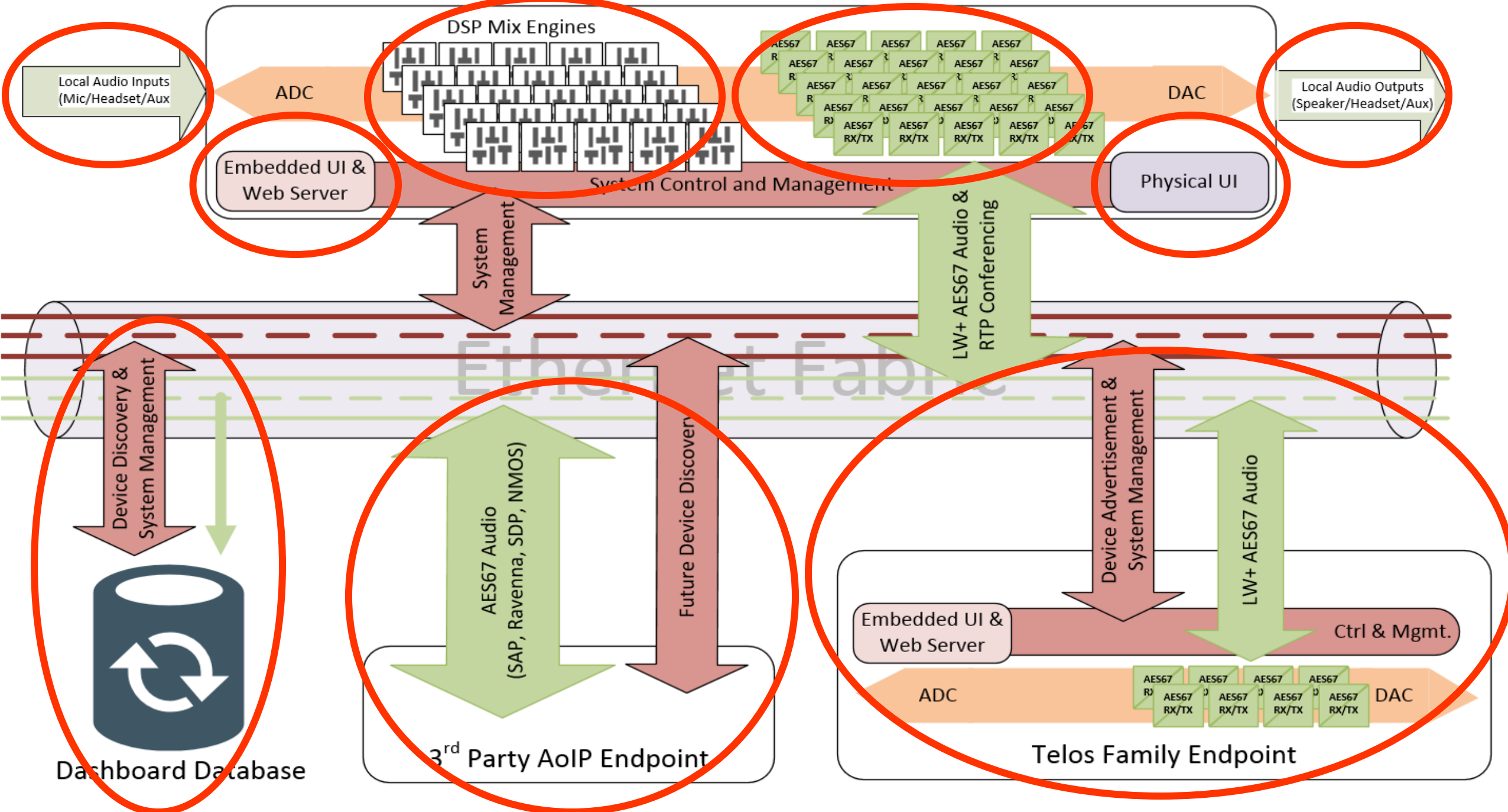
- No Matrix or equivalent Centralised Processing Engine
- Produce 'Familiar' Software and Hardware components
- Meet or exceed expected Matrix Intercom capability
- Use standards based protocols natively – RTP / AES67 / SIP / VoIP
- Develop a scalable and resilient solution
- Make it simple for users of all skill levels
- Hide complexity 'under-the-hood'
- Look beyond the limits of what has gone before

Introducing:

Telco

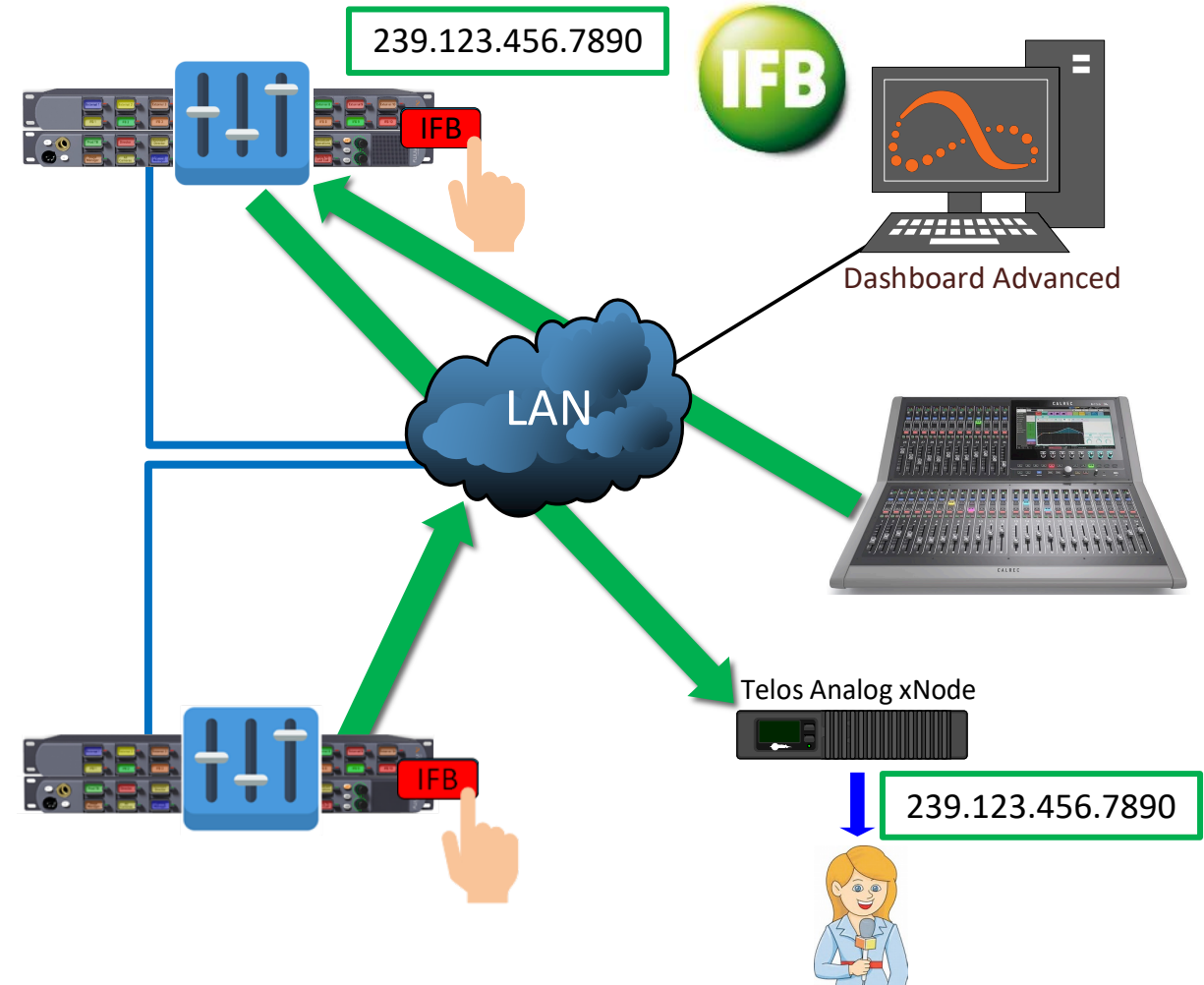
 INFINITY®

Infinity Device / Panel or Beltpack



Example: What happens when I created an IFB?

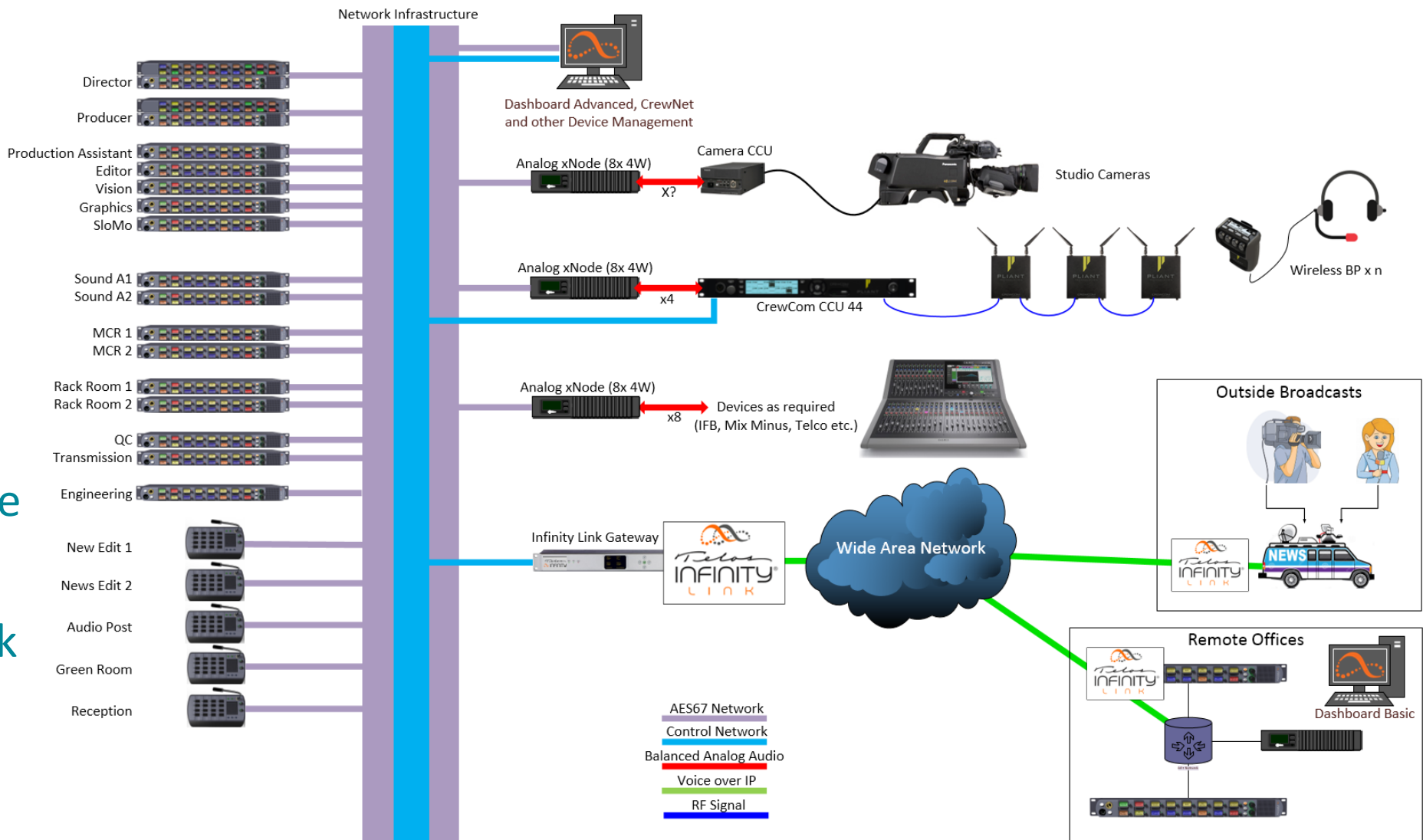
1. Create an IFB using Dashboard
2. Dashboard will assign a mix resource at two locations (primary and secondary)
3. Add a source (mix minus from console)
4. Add a destination (IFB Transmitter) IFB
5. Destination will subscribe to primary MC address
6. Mix Minus is routed to primary IFB mix
7. Destination hears mix minus
8. Add IFB key to Intercom panel(s)
9. Panels can talk across IFB feed – mixer will 'duck' mix minus per configuration



Telos Infinity - Typical Studio System

Telos Infinity has no Matrix, so communication between Intercom elements is via direct connection to the LAN

Analog and other baseband audio is converted to IP by node devices and any ST2110-30/AES67 present on the network can connect natively without an external interface



The diagram illustrates a 48-channel digital audio system architecture for a television studio, organized into four identical functional blocks arranged in a 2x2 grid. Each block represents a self-contained audio environment with the following components:

- Production Control Room:** Contains a computer monitor and four channels of audio processing equipment.
- Studio Floor:** Includes four channels of audio processing equipment and a camera operator icon, connected to an **Analogue effects** unit.
- Sound Control Room:** Features a single channel of audio processing equipment.
- Post Production:** Includes two channels of audio processing equipment and two microphone icons.

The system is interconnected via a central digital bus system, represented by a large horizontal and vertical network of lines. This bus system facilitates the distribution of 48 channels of digital audio across all four blocks, ensuring synchronized and high-quality sound throughout the studio environment.

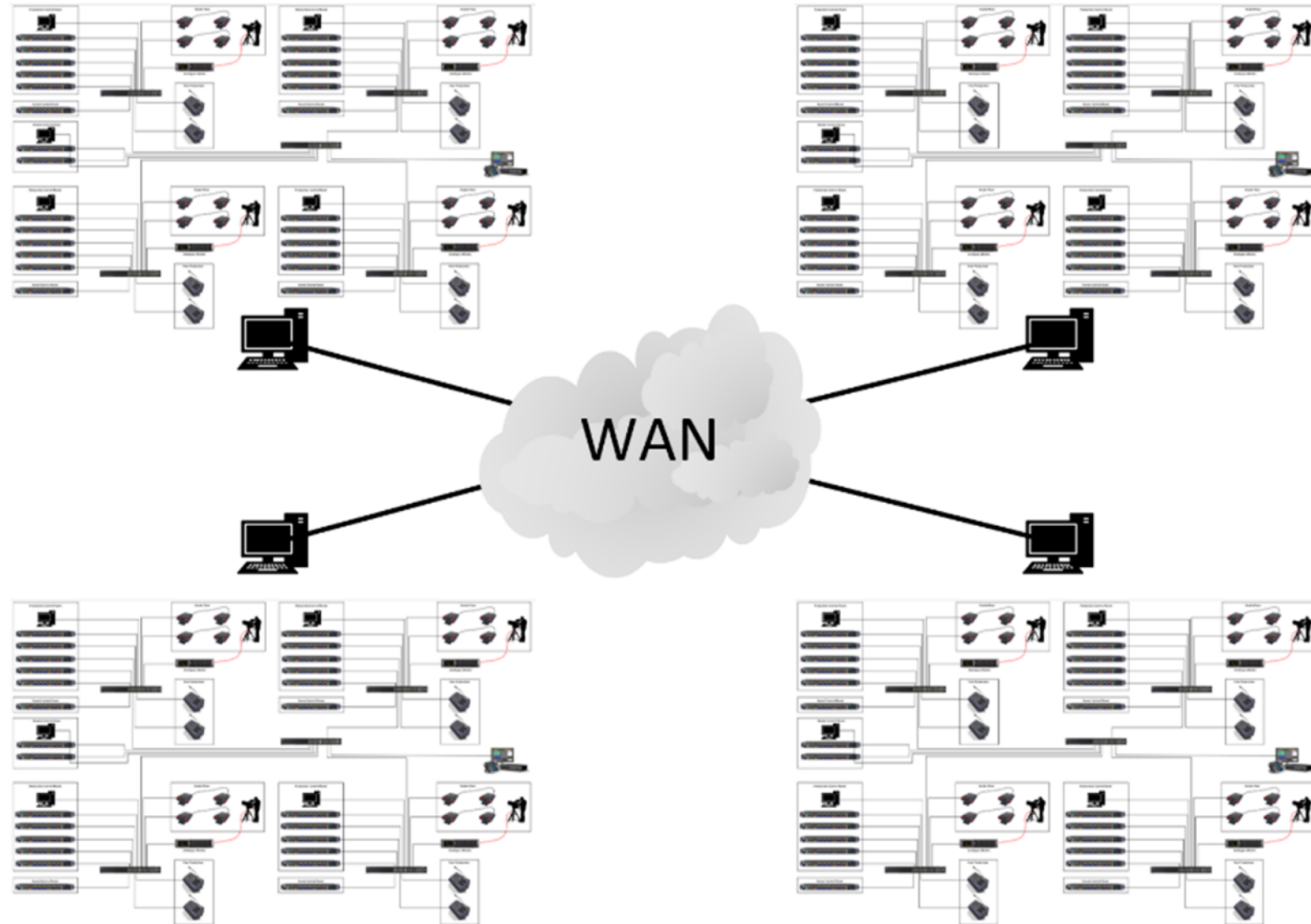
Dashboard Advanced enables the users to manage their own studio configurations while the Master Control client has full administrative rights over the entire system

Telos Infinity - Multi Site Systems

Sites are connected using Telos Infinity software servers equipped with the adaptive OPUS VoIP codec

Dashboard manages each local client with overall administrative control assigned to one or more Master Control operations

VLAN will work directly on hi-speed connections without needing VoIP compression





Thank you