NTP TECHNOLOGY

AUDIO ROUTING SOLUTIONS

White Paper

NTP IP Audio routing

NTP IP Audio

NTP IP Audio is a professional high quality low latency system for distribution and routing of 24bit uncompressed audio with sample rates of 48kHz or higher on a Gigabit (1Gbps) IP based Ethernet.

NTP IP Audio is a third-generation networked audio distribution technology with great scalability and flexibility when compared to other third generation peers and previous-generation audio networks. NTP IP Audio operates over Gigabit and higher rate Ethernet variants. NTP IP Audio is a central component in an NTP routing system, which can combine both Ethernet and TDM based interconnections.

NTP IP Audio is fully integrated in the comprehensive NTP Router Control System RCCoreV3 and the VMC user interface platform. Interactive integration with the Router Control System means that NTP IP Audio can be configured and monitored using the user interface tools and software applications available on the Router Control System platform.

Network Technology

NTP IP Audio is based on the Dante™ Ethernet Layer 3 protocol developed by Audinate Ply Ltd. The protocol ensures very low latency using Quality of Service and sample accurate inter channel synchronisation. The Ethernet protocol is prepared for AVB interoperability.

Dante™ is built upon the IEEE 1588 Time Precision Protocol standard to derive a precise clocking mechanism for synchronization. This is the same protocol commonly used in robotics and other applications requiring extremely accurate clocking. Today's Ethernet switches provide high performance data transport and quality of service functions capable of supporting real-time Voice over IP (VoIP) telephone services. The Ethernet protocol meets professional Quality of Service (QoS) requirements by making use of standards originally used to provide VoIP telephone services.

As a result of this, latency less than 1ms across ten network switch hops can easily be achieved using NTP IP Audio in a gigabit network, even when using existing Ethernet switch technology.

NTP Router System

Since NTP IP Audio is an integral part of the NTP audio router system, some background on the system is required to fully appreciate NTP IP Audio. The NTP audio router system is comprised of three principal component types: The IP audio and TDM audio router hardware with various I/O capability, the QNX based router control system RCCoreV3, and the user interface software and hardware client control applications.

NTP IP audio and TDM audio router hardware serves as I/O devices and are the entry and exit points for audio in the router system. Audio signals presented to the NTP IP Audio I/O devices are packetized and routed over the network to other NTP IP audio devices where the audio data is processed, re-packetized and sent back to the same or different I/O devices which can also be access points for the TDM router system. Each NTP IP Audio I/O device has two Ethernet connections and optionally two Matrix Cores for use in fault-tolerant networking, as well as dual power supply. The NTP IP Audio I/O frame supports up to a total of 64 channels in and 64 channels out via Ethernet. The total non-blocking capacity of a 1 Gigabit Ethernet is 512 mono audio channels. The total capacity of an NTP TDM audio router system is 8.192 mono channels.

QNX™ based router control system RCCoreV3 is the main control system for all NTP IP Audio and TDM audio router hardware I/O devices and the server for the user interface software and hardware client control applications. The control system is a ultra stable database driven server, which handles all configuration and execution of cross-points and connections both from manual entries and from pre configured schedules. The control system can run on redundant controller hardware, and the real time operating system is based on QNX.

User interface software and hardware client control applications are used for operating the system. Via the VMC user interface software settings and configuration can be made for cross points, connections, and for the DSP functions in the I/O hardware. Also signal level monitoring and detection can displayed on separate PC screens. All VMC applications are running on Windows™ XP/Vista/7.

NTP IP Audio Signal monitoring in high quality is possible via the VMC user interfaces using the PC as a sound device. NTP IP Audio can be routed via the same network which connects the control system and the VMC user interface.

IP Audio Signal ingest of audio sources from a PC can be made directly to the network using a virtual sound card software operating the native Ethernet port of the PC or by using a separate Ethernet sound card installed in the PC compatible with the Dante protocols of the NTP IP Audio network.

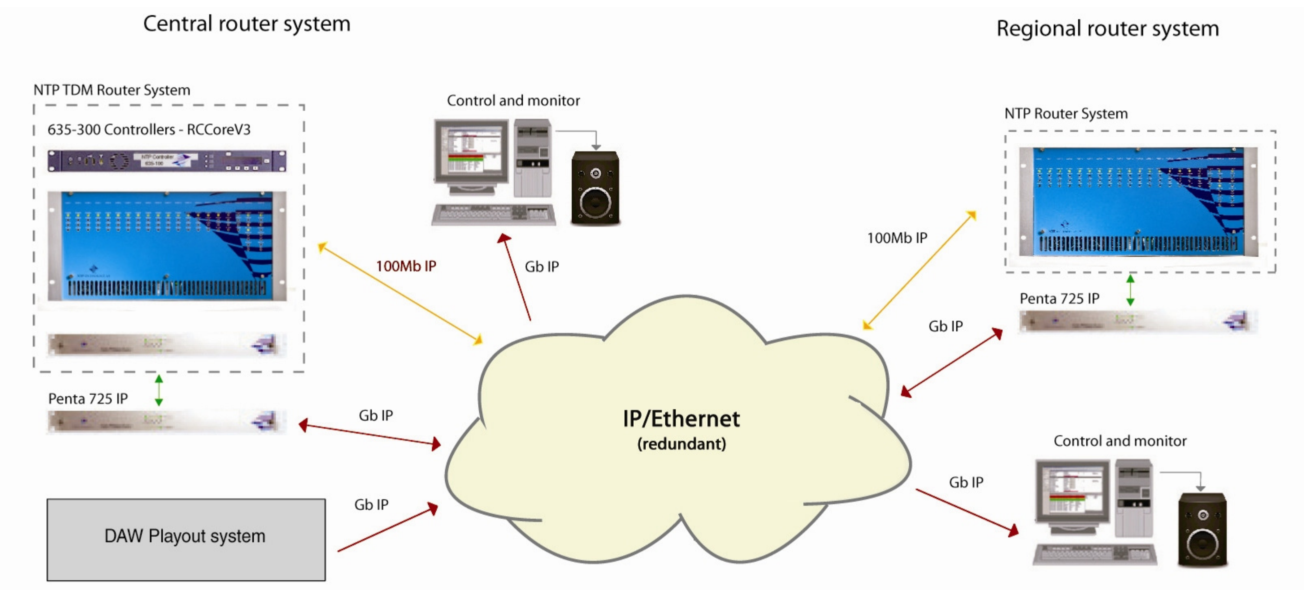

Figure 1, NTP router system with IP Audio and TDM routing

Quality of Service

NTP IP Audio relies on a network that implements Quality of Service (QoS) and differentiates between the multiple applications that operates on the network. QoS ensures that applications sensitive to network congestion and latency get preferential treatment or better service from the network.

A typical application is a network supporting Voice over IP (VoIP) telephony. Because phone calls are real time, a QoS enabled network will prioritize the transmission of packets carrying VoIP telephone calls over other packets like email and web browsing.

Once configured the RCoreV3 Router Control System can handle IP Audio switching on the complete large network configured.

Clock Synchronization

Clock synchronization is based on a simple principle: like when we set the time on a watch we synchronize the watch to another clock at that instant. Clock synchronization protocols do something similar by exchanging messages containing timestamps so that a slave clock can track the time on a master clock connected to a network.

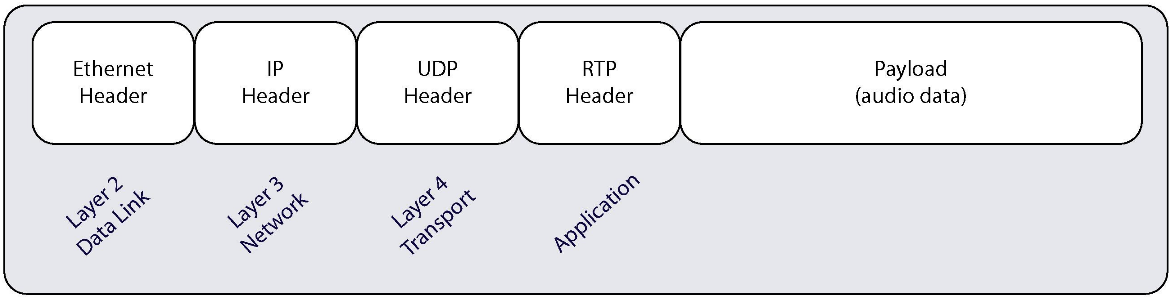

Figure 2, OSI network encapsulation of audio payload data in an IP packet

The basic method of implementing QoS is to mark packets with a priority and require switches to prioritize packet processing according to a set of rules. For example, modern switches implementing QoS usually implement “Strict Priority” which is a rule where the next packet to be transmitted is always the highest priority packet currently queued by the switch.

Should congestion occur, non-VoIP packets are dropped by the switches to allow the VoIP packets to flow unimpeded

IP Audio over network segments

In larger network structures, more Local Area Networks (LAN) or subnets are joined via gateway routers as part of an overall network. It is possible for the IP Audio Hardware to connect to IP Audio Hardware on other subnets. The Dante protocol used in NTP IP Audio is based on IP/UDP multicast, which normally does not “travel” across subnet segments of the network. NTP IP Audio uses the Dante Netspander software in order to configure the gateway router ports to be transparent for the IP Audio data and recognise the IP addresses of the IP Audio Hardware on the subnets.

A widely deployed clock synchronization protocol is NTP – the Network Time Protocol, which is used to synchronize clocks such as those in your PC with time servers connected to the Internet.

A relatively new clock synchronization protocol for local area networks is the IEEE 1588 Precision Time Protocol (PTP) (2). IEEE 1588 came out of the industrial control and test/measurement fields and can run over IP, Ethernet, and other bus structures such as backplanes. A PTP master clock sends several types of synchronization packets containing timestamps to slave clocks, resulting in the slave clock accurately tracking the seconds/nanoseconds of the master clock.

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Note that clock synchronization packets are delayed as they pass through the network. Some delays are predictable and can be easily corrected for (e.g. propagation times across links), While others are statistical in nature (e.g. queuing behind other packets sharing the same path).

In 100Mbps-only networks, queuing delays can be significant especially when there is a high traffic load, making tight synchronization more difficult to achieve. Increasing the speed of the network links to Gigabit (1Gbps) reduces queuing delays by a factor of 10 compared to 100Mbps networks, making microsecond synchronization accuracy possible with existing Ethernet switches.

NTP IP Audio Capacity and Quality

NTP IP Audio allows the network to be shared between audio distribution, system control, monitoring, and traditional network applications. A Windows PC running VMC user interface or a monitoring application software can perform control of the router system. PCs can exists on the same network with the IP Audio components. Multiple instances of VMC user interface applications can control and monitor the same router system from multiple PCs.

Other control components such as touch screens and third-party control systems may be connected to the same network and to the same system.

Standard commodity Gigabit Ethernet switches serve as the interconnect points for NTP IP Audio networking. To ensure reliable, low-latency audio delivery, these switches must meet NTP IP Audio performance and feature requirements.

NTP IP Audio can safely use up to 90% of Gigabit Ethernet link capacity. This is enough bandwidth to simultaneously carry up to 512 low-latency, non-blocking high-resolution audio channels on a 1 Gigabit network. There is no limit to the total number of channels carried by a network. Since it depends only on the speed of the core network which can also be 5 or 10 gigabit. It is possible to build systems of systems and this way, channel capacity is virtually unlimited.

Latency is the delay of a signal through a system or component. In audio, latency is very critical in live broadcast program applications. NTP IP Audio latency is typically 0,15 ms in a one switch network, and 0,5 ms in a five switch network. Typically latency will always be below1 ms. Time alignment of audio signals is assured by the high-performance PTP protocol implementation. The path through a NTP IP Audio system can comprise analog-to-digital conversion, a first pass through the network to the IP Matrix Core, processing in the core, a second pass through the network to the destination and finally, digital-to-analog conversion.

Total system latency is 2-3 ms. All audio processing and transport is carried out in floating-point format. Processing is handled at up to 32-bit resolution and network transport uses 24-bit resolution.

Fault Tolerance

NTP IP Audio supports all standard Ethernet and layer-3 fault tolerance strategies: Spanning tree protocol (including rapid spanning tree) and link aggregation. The IP Audio hardware support redundant network connection, dual Matrix Core configuration, redundant power supplies, and self-monitoring systems with secure fail-over schemes, in case of network or IP Audio hardware fault

The IP Audio hardware accommodates a fully redundant networking configuration. When this capability is utilized, two distinct and parallel networks are built. The dual network configuration can withstand any single network component or link failure by automatically switching to the secondary network. The switch over is accomplished quickly and without interrupting audio.

The IP Audio hardware also supports fault tolerance through dual connections to the same network. This alternative configuration potentially improves fault tolerance in the presence of multiple failures but does require additional backbone bandwidth and is susceptible to adverse interaction in some fault scenarios.

In addition to support for fault tolerance in the network, fault tolerance for IP Audio hardware is supported. A system can be configured with two IP Audio Matrix Cores. Cores are designated primary and backup by configuration. The primary Core initially comes up in the online state and establishes audio streams to and from the I/O devices on the network. The backup Core initially comes up in the offline state and does not transmit or request to receive any audio streams.

The two Cores are designated to perform identical signal processing. Redundancy awareness in the NTP Router Control System RCCoreV3 keep operating parameters synchronized.

I/O devices may be doubled up either throughout the system or only where deemed critically necessary.

The above described redundancy ensure that when a failure is detected in a primary device, the system switches to the backup. To avoid unnecessary interruption of audio and respond appropriately to intermittent failure scenarios, the system does not automatically switch back to the primary when the primary recovers from its failure. User interface controls via the NTP Router Control System allow manual switch over between primary and backup devices. Figure 3. shows the various configuration possibilities of redundant network and IP Audio hardware.

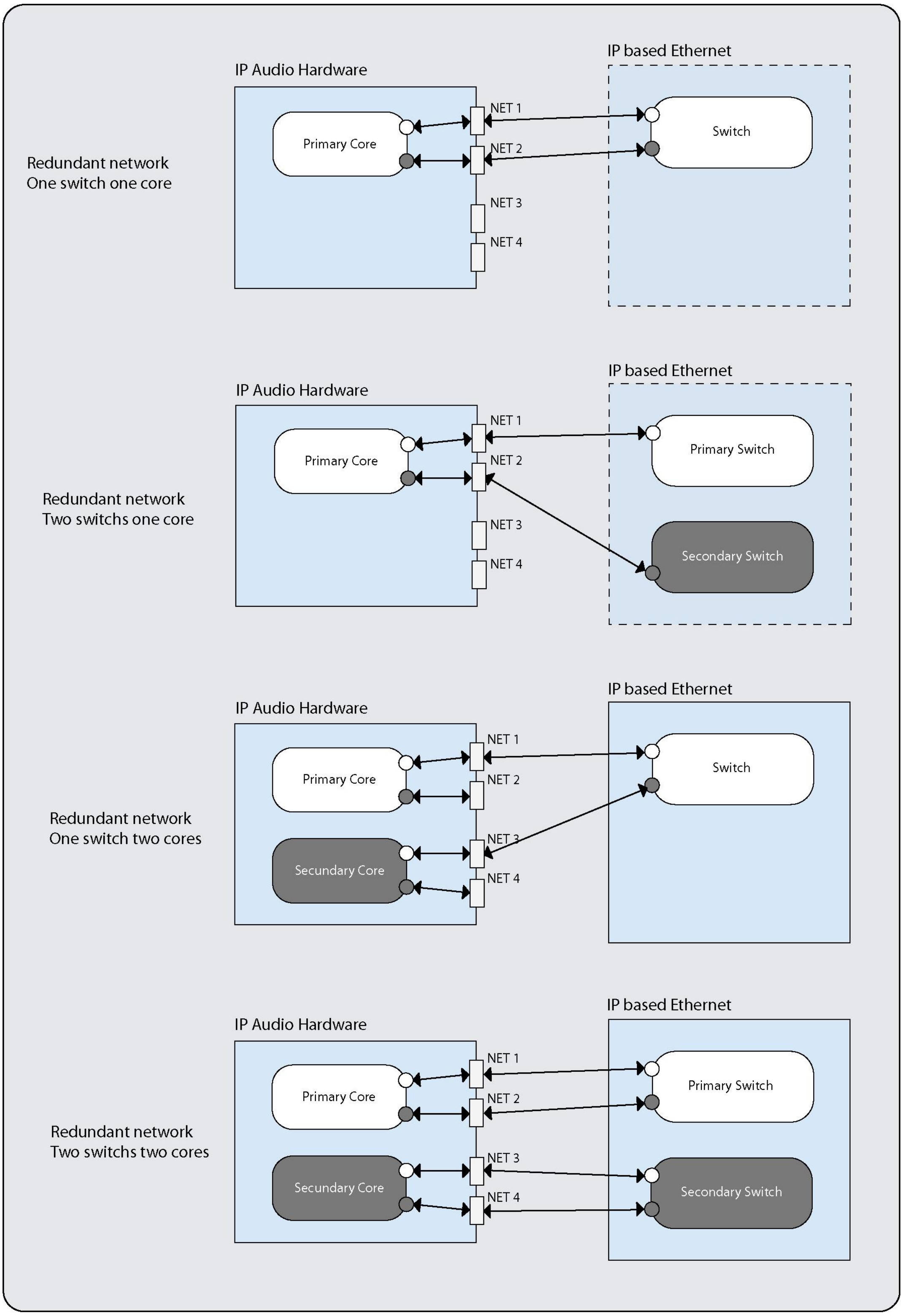

Figure 3, Fully redundant network with redundant switches and redundant cores

Audio Networks Comparison

The OSI reference model is a useful tool for understanding how networks are organized. The model is arranged from network hardware specific at layer-1 to abstract connectivity at the higher layers. Networked audio distribution technologies can be categorized based on where they insert themselves in the OSI network reference model.

Layer-1 communications technologies operate at a basic hardware level. Protocols are unsophisticated and cannot be readily translated to other network hardware. Layer-1 technology is often focused on point-to-point communications. The network only comes into existence through the addition of purpose-built switching and routing equipment. EtherSound™, AES50™, A-Net™ and Rocknet™ are examples of layer-1 audio distribution technologies.

Layer-2 systems cooperate with their respective native network technology. Ethernet is, by far, the most widely used layer-2 network. Although layer-2 systems are bound to their chosen network hardware and do not scale beyond it, insertion at this higher layer allows the use of standard network switches and often allows for the coexistence of multiple services on the same physical infrastructure: audio mixed with more traditional network applications.

AVB™ is a layer-2 or layer-3 network solution. AVB requires an uninterrupted layer-2 connection between devices and requires that connection be made through special AVB-capable switches and network equipment.

CobraNet™ was first introduced as a first-generation network requiring a separate physical infrastructure. CobraNet evolved to become a full-featured layer-2 technology. CobraNet has not evolved to compete with greater capacity and performance of the third-generation gigabit technologies.

Layer-3 networking, also commonly known as IP or TCP/UDP/IP networking, is the basis for the Internet. The layer-3 systems operate at an abstract level above dependencies on the particular network hardware on which they run.

Q-LAN™ and RAVENNA use layer-3 real-time audio distribution technologies similar to Dante™.

On a private network used in audio installations, a layer-3 protocol such as NTP IP Audio based on Dante offers increased interoperability and scalability and access to advanced capabilities in modern network equipment and tools. A layer-3 protocol operates without impediment on a layer-2 network The reverse is not true – a layer-3 network will refuse. to carry layer-2 traffic. Even if the scope of your current projects fit within layer-2 networking, there is little overhead associated with inclusion of the layer-3 capability.

Network Size

Limiting the size of the network helps ensure that performance required by NTP IP Audio is achieved. Network performance is limited by delays occurring in network equipment and, to a lesser extent, delays in wiring and fiber optic cables due to the finite speed of light.

Total latency through NTP IP Audio on a large network is typically around 1 ms. One quarter of that time is budgeted to network delays (the other half is budgeted to buffering and packetizing). 1 km of twisted pair or optical cable imparts almost 5 µs delay. The minimum delay through a standard Gigabit Ethernet switch is 12 µs; Maximum delay can be several times higher.

NTP IP Audio network design guidelines describe allowable network size in terms of hop count and network diameter. Hop count is the maximum number of switches any audio data must pass through between its source and destination. Diameter is the accumulated cable distance between the furthest two endpoints on the network. The Table shows allowed network diameter as a function of hop count.

Hops	Diameter
2	35 km
3	29 km
4	22 km
5	15 km
6	9 km
7	2 km

Conclusion

Integrated within the NTP Router Control Platform, a new networked digital audio distribution system for broadcast quality audio has arrived. Compared with previous-generation systems and competing current-generation, NTP IP Audio offers lower latency, higher fidelity, higher capacity and more comprehensive fault tolerance and routing control capabilities. NTP IP Audio operates on a cost-effective commodity Gigabit Ethernet local area network., and is also capable of handling routed networks of more subnets. These features makes NTP IP Audio extremely useful in today's comprehensive and large broadcast facilities where fast, flexible and economical audio routing is a must.

Glossary

ADC

Analog to Digital Converter. ADCs are found at the inputs of digital signal processors

AES3

Audio Engineering Society digital audio interconnect standard (3rd AES standard). Also known as AES/EBU and technically similar to SPDIF consumer digital audio interconnect standard.

AMX

Systems technology provider and control and automation platform.

ASCII

American Standard Code for Information Interchange specifies mapping of text characters to numerical values for use in communication and human-computer interface.

AVB

Audio Video Bridging is an initiative under development by the IEEE's layer-2 net-working authority, the 802.1 working group. AVB promises interoperable audio and video interconnect (similar to FireWire's offerings) on layer-2 Ethernet networks.

Core (network architecture)

A core switch is the central routing point in certain network designs. The core switch is a high-capacity hardware configurable (typically with slide-in interface cards) usually the size of a small refrigerator. Core switches are commonly set up in redundant pairs to eliminate a single point of failure for the network.

Core (IP Matrix Core)

The Core is the NTP IP Audio central processing unit. The Core is where audio signals for the system are processed and combined. Cores may be set up in redundant pairs to eliminate a single point of failure for the system.

Crestron

Systems technology provider and control and automation platform.

DAC

Digital to Analog Converter. DACs are found at the outputs of digital signal processors.

DHCP

Dynamic Host Configuration Protocol is used by IP network devices when first connecting to a network to receive an IP address assignment and other network configuration information.

DiffServ

Differentiated Services is an IETF standard for classifying network traffic by using the DSCP field in the IP header.

DNS

The Domain Name System is the service and protocol suite that converts domain names (e.g. www.qscaudio.com) used by humans to the IP addresses (e.g. 206.135.232.7) used by computers on an IP network.

DSCP

Differentiated Services Code Point is the field in the header of IP packets used in classifying network traffic under the DiffServ standard.

Ethernet

One of several variants of wired and wireless physical network interconnects. All Ethernet variants share a common packet format.

IP Audio Hardware

The I/O router devices are the entry and exit points for audio in the NTP IP Audio system system.

IEEE

Institute of Electrical and Electronics Engineers is a professional organization that, among other things, operates a standards body which is responsible for networking technologies such as Ethernet.

IEEE 1588

A time-transfer protocol that allows precise synchronization of clocks across an Ethernet network.

IEEE 802.1

IEEE standards working group responsible for networking standards related to layer-2 Ethernet networking.

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IEEE 802.3
IEEE standards working group responsible for maintenance and extension of wired Ethernet standards.

IETF
The Internet Engineering Task Force is responsible for development of IP networking protocols and standards.

IGMP
Internet Group Management Protocol is a protocol used in the management of multi-cast transmissions.

IP Addressing
An IP address is a 32-bit number that uniquely identifies an endpoint on an IP network.

IP Networking
A communication system that utilizes TCP/IP or UDP/IP encapsulation of data.

IP Routing
The process of forwarding data towards its destination based on information contained in the header of an IP packet.

LAN
A Local Area Network is a network scaled for the home or small office. LANs are typically layer-2 networks. LANs may be connected through a gateway to a WAN or to the Internet

Link Aggregation
The use of multiple links in parallel to create a single higher-bandwidth and resilient connection. Various proprietary and standardized means of designating and operating the links are available.

mDNS
Multicast Domain Name System is a scaled down variant of DNS which uses multicast messaging rather than a network of servers to distribute resource information.

Multicast
A one-to-many addressing mode. A packet with multicast addressing will be simultaneously routed to all interested listeners. Multicast is available in IP networking and on Ethernet.

OSI Reference Model
The Open Systems Interconnection Reference Model is a useful tool for understanding how networks are organized. The OSI reference model is comprised of seven layers from network hardware specific at layer-1 to abstract network connectivity at the higher layers.

PTP
Precision Time Protocol is an alternate name for IEEE 1588.

QoS
Quality of Service is a data communications discipline that includes classification and prioritization of data flowing through a network.

RCCoreV3
RCCoreV3 is a complete integrated controller server that handles the Router Control of a NTP Audio Router System.

SMPTE
Society of Motion Picture and Television Engineers is a professional organization that, among other things, operates a standards body which is responsible for broadcast media and interconnect standards.

SNMP
Simple Network Management Protocol is used to control and monitor network equipment and end stations.

STP
Spanning Tree Protocol ensures a loop-free topology for Ethernet networks. STP is also used to create fault tolerant networks. Advanced variants of STP are available: Rapid STP (RSTP) recovers from failure more quickly than the original. Per-VLAN STP (PVST) takes VLAN configuration into account in its operation.

Stream
NTP IP Audio audio is transmitted in streams. A stream is an ongoing series of packets containing one or more channels of real-time audio data.

TCP/IP
Transmission Control Protocol over Internet Protocol is the workhorse protocol suite of the Internet. The protocol suite creates reliable connections between application over a network. The TCP protocol handles error correction and connection management.

UDP, UDP/IP

User Datagram Protocol is a stripped-down protocol suite typically used for non-critical applications or real-time data. UDP is known as a "connectionless" and "unreliable" protocol meaning that it does not include the connection management and error recovery functionality found in TCP – applications are expected to provide these pieces if required.

Unicast

A one-to-one addressing mode. A packet with unicast addressing will be routed to a single destination as indicated by the IP address in the header of the packet.

VLAN

A Virtual Local Area Network is a logically segmented interconnected set of network ports or end stations. VLANs allow different services or user groups to be isolated from one another without requiring dedicated network hardware for each.

VoIP

Voice over Internet Protocol is a telecommunications protocol suite which allows conventional telephone calls to be carried over data networks such as Ethernet or the Internet.

WAN

A Wide Area Network is a network with scope larger than a LAN and smaller than the Internet. A WAN is typically a layer-3 network.

WiFi

Trade name associated with wireless Ethernet networking based on the IEEE 802.11 family of networking standards.

Zeroconf

Zero Configuration networking is a set of techniques and protocols that automatically creates a usable IP network without manual

References and literature

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(2) IEEE 1588 Precision Time Protocol, http://ieee1588.nist.gov/

(3) Whitepaper: QSC, Q-Lan architecture, Kevin Gross, October 7, 2009

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