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WHITE PAPER

Q-LAN

The Architecture and Network Redundancy

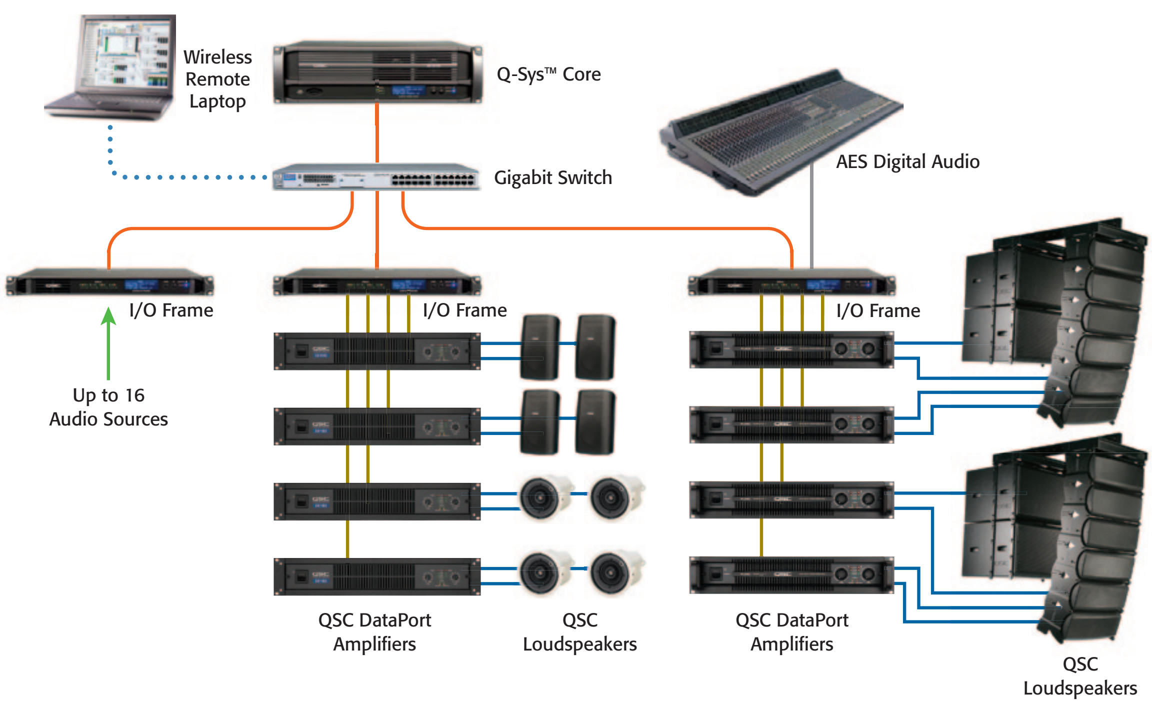
Q-LAN is a third-generation networked media distribution technology providing higher quality, lower latency and greater scalability when compared to its third generation peers and previous-generation audio networks. Q-LAN operates over gigabit and higher rate Ethernet variants. Q-LAN is a central component of QSC's comprehensive Q-Sys integrated system platform. Interactive integration with Q-Sys means that Q-LAN can be configured and monitored using the graphical and scripting tools available on the Q-Sys platform.

Q-Sys™

Since Q-LAN is an integral part of the Q-Sys platform, some background on Q-Sys is required to fully appreciate Q-LAN. Q-Sys is comprised of three principal component types: I/O Frames (and other I/O devices), Cores and User control interfaces. The components are interconnected via an Ethernet and/or IP network.

I/O devices are the entry and exit points for audio in the Q-Sys system. Audio signals presented to the I/O devices are packetized and sent over the network to the Core where the audio data is processed, re-packetized and sent back to the same or different I/O devices for output to power amplifiers or other audio destinations. Each I/O device has two Ethernet connections for use in fault-tolerant networking. The Core connects to an I/O device via Q-LAN. The number of I/O devices in a system is limited only by the size of your Core. The I/O frame supports up to a total of 16 channels in and 16 channels out through up to four modular audio I/O cards installed in the Frame. Future I/O devices may have different I/O capacity.

The Core is Q-Sys' central processing unit and is where audio signals for the system are processed and combined. Different Core models (e.g. Core 1000, Core 3000, Core 4000) are available with different network I/O and processing capacity. At least one Core is required in a system. Addition of a second Core for fault tolerance is a design option.



Q-Sys™ architecture

The Core's connection to the rest of the system is primarily through the Q-LAN network. Each Core has four gigabit Ethernet connections: two for fault-tolerant Q-LAN and another two for running control communication though connections physically separate from Q-LAN if desired. Q-LAN audio and control data can peacefully coexist (see Quality of Service) so this second set of connections is not used in typical installations.

Q-LAN allows the network to be shared between audio distribution, system control and monitoring and traditional network applications. A Windows™ PC running Q-Sys Designer software is an optional component of a Q-Sys system. The PC exists on the same network with the other components. Multiple instances of Q-Sys designer can monitor the same Q-Sys system from multiple PCs. Multiple instances of Q-Sys designer running on a single PC can control and monitor multiple systems.

Other control components such as touch screens and third-party control systems (e.g. AMX™, Crestron™) may be connected to the same network and to the same system(s).

Standard commodity gigabit Ethernet switches serve as the interconnect points for Q-LAN networking. To ensure reliable, low-latency audio delivery, these switches must meet Q-LAN performance and feature requirements (see Q-LAN Network Requirements). User interface and control components of the system may also be connected to these switches or may be connected to lower-performance segments of the network (e.g. fast Ethernet, WiFi, WAN).

Q-LAN Capacity and Quality

Q-LAN can safely use up to 90% of gigabit Ethernet link capacity. This is enough bandwidth to carry up to 512 low-latency, high-resolution audio channels. The largest Q-Sys™ Core utilizes this maximum capacity at 512 channels in and 512 channels out through one network connection. There is no limit to the total number of channels carried by a network. It is possible to build systems of systems and this way, "system" capacity is virtually unlimited.

Latency is the delay of a signal through a system or component. In audio, latency is most critical in live sound applications. Where latency is critical, generally speaking, lower latency equates to better performance.

Q-LAN latency is 2/3 ms. Time alignment of audio signals is assured by a high-performance hardware-assisted precision time protocol implementation (see Clock Distribution). The path through Q-Sys comprises analog-to-digital conversion, a first pass through the network to the Core, 1/3 ms of processing time in the Core, a second pass through the network to the destination and finally, digital-to-analog conversion. Total system latency is 2-1/2 ms.

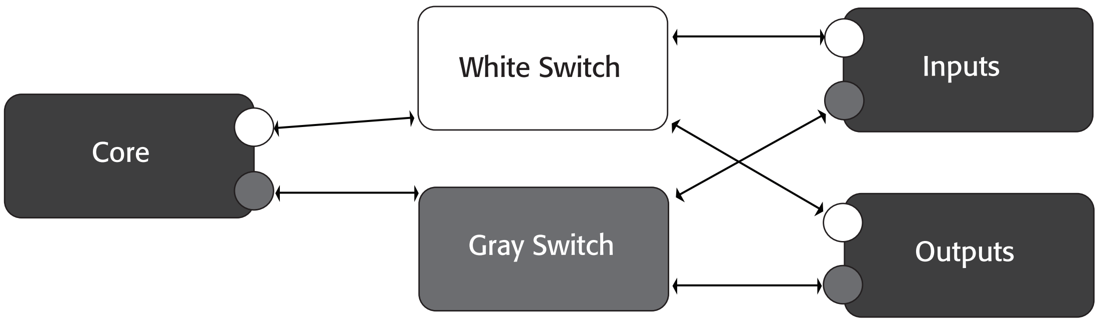
All audio processing and transport is carried out in floating-point format. Processing is handled at up to 64-bit resolution. Network transport uses 32-bit resolution. Resolution in excess of the performance of the 24-bit input and output interfaces, DACs and ADCs assures that Q-Sys processing and networking is sonically transparent. The floating-point format preserves dynamic range throughout the signal path making gain structure and intermediate overload considerations inconsequential[1]. The inputs and outputs of the system are the only places where attention to gain structure is required.

[1] Gain structure refers to the normalization of gain elements though the signal path from input to output so as to assure a constant amount of headroom. Configuring the gain structure in this way has the benefit of optimizing signal-to-noise ratio for the system. Because of the self-scaling attributes of the exponential representation used for floating-point audio data in Q-Sys, signal-to-noise ratio is inherently optimized at any gain setting or signal level.

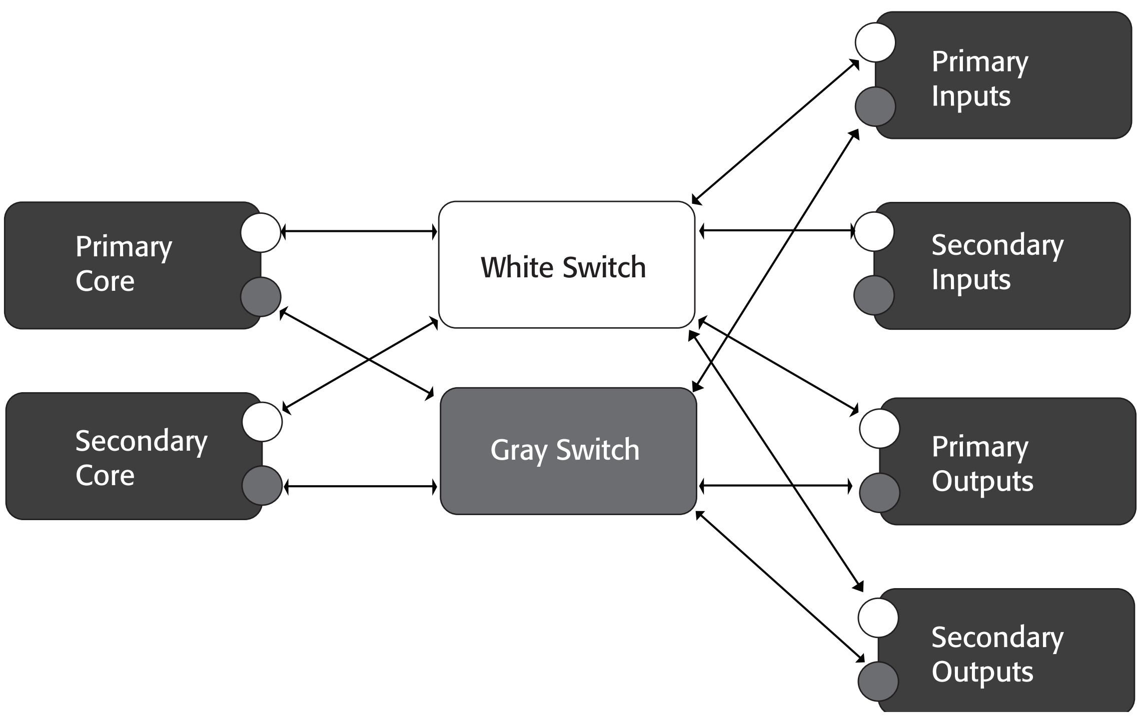
Fault Tolerance

Q-LAN supports all standard Ethernet and layer-3 fault tolerance strategies: Spanning tree protocol (including rapid spanning tree), link aggregation, IP routing, vendor-specific meshing and fail-over schemes, self-monitoring systems and redundant power supplies. Q-Sys™ 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.

Q-Sys also supports fault tolerance through dual connections to the same network. This configuration is, in some regards, more "IT friendly"[2]. 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[3].



Fully redundant network configuration



Fully redundant network with redundant Cores and redundant I/O Frames

2 Mainstream IT professionals and departments tend to think in terms of "The" network. The idea of multiple independent networks is not always readily accepted.

3 In some failure modes, it is possible for one piece of malfunctioning equipment to adversely affect everything connected to the same network.

In addition to support for fault tolerance in the network, fault tolerance for Q-Sys components is supported. A system can be populated with two Cores. Cores are designated primary and backup by the designer. 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 Q-Sys control communications keep operating parameters synchronized.

I/O devices may be doubled up either throughout the system or only where deemed critically necessary. Analog and digital audio sources to and from I/O Frames are wired in parallel to primary and backup input devices. Analog and digital outputs are bridged together and wired to downstream equipment. Internal relays open on the backup device to prevent contention. Microphone inputs configured to supply phantom power may be safely wired in parallel. The Core uses the following logic to select between primary and backup I/O devices operating as a redundant pair:

1. On startup select the primary.
2. Core continuously polls both primary and backup during operation.
3. If selected I/O device reports a fault or fails to respond and cannot be discovered and other device has recently reported good health, switch to the other device.

The above rules 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 allow manual switch over between primary and backup devices.

AV 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 network solution. AVB not only requires an uninterrupted layer-2 connection between devices but it 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/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.

On a private network such as used in audio installations, a layer-3 protocol such as Q-LAN 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.

Q-LAN and **DANTE™** are both layer-3 real-time audio distribution technologies. Telephony through VoIP and Audio over IP used in broadcast applications are examples of lower performance layer-3 technologies.

Audio Delivery

Q-LAN audio is transmitted in streams. A stream is an ongoing series of packets transmitted at a rate of 3000 per second. Each packet contains 16 audio samples for each of up to 16 audio channels. An arbitrary mixture of streams with different channel counts is permitted on a Q-LAN network. Samples are conveyed in 32-bit floating point format. The maximum payload size for an audio stream packet is 1024 bytes. The minimum payload is 64 bytes. Total packet size including all headers is in the range 1078 to 118 bytes. Bandwidth consumed per stream is 3.3 to 26.5 Mb per stream for streams with 1 to 16 audio channels respectively.

The organization of multiple audio channels into streams improves bandwidth efficiency (a single 16-channel stream uses half the bandwidth of 16 single-channel streams). Streams also remove the need to route related channels independently. Streams do not affect routing flexibility. All audio passes through the core and is processed there on a channel-by-channel basis. The Core supports a generous number of streams (up to 128 streams received and 128 transmitted).

Audio streams are transported via UDP/IP. As part of stream setup, a UDP port number is negotiated between transmitter and receiver. Transmitters and receivers find each other through a separate handshake sequence which occurs when a stream is first established. A receiver will open a UDP port in the range 6511 to 6638. The receiver has learned the IP address of its respective transmitter through the separate discovery mechanism (see Discovery). The receiver sends a UDP subscribe request packet to the transmitter from that port. The transmitter responds with a subscription acknowledgement packet and then begins sending audio data packets.

Streams are unidirectional from transmitter to receiver with regular, though less frequent (1 acknowledgment per 100 stream packets) acknowledgments from receiver to transmitter. The periodic acknowledgments ensure that the transmitter promptly discontinues transmission in the event of receiver or receiver connection failure.

All stream transmissions are done with unicast IP addressing. Separate streams are used to route the same audio to multiple destinations. The I/O Frames feature the ability to replicate the same audio channel multiple times on multiple outputs of an I/O Frame. In this scenario, the number of channels on the network is reduced with respect to the total number of output channels available from the system.

Discovery

Discovery is the process of enumerating and identifying devices and resources on a network. Without discovery, the only way to make connections is by using cumbersome network addresses.

Instead of exposing these addresses to users, Q-Sys devices are identified by name. Q-Sys uses a scaled-down variant of the Domain Name System (DNS) used on the Internet. This variant is called mDNS and it works on a local area network with or without assistance of a name server or other infrastructure. mDNS is a component of the Zeroconf protocol suite available in various forms from Apple™ and others.

Zeroconf streamlines the process of setting up a Q-Sys system. Simply attach the Cores and I/O devices to the network and the components promptly appear in a list in the web browser. Click on each item to configure and name it. Discovery combined with system integration supported by Q-Sys designer means that audio connections across the network are specified graphically. Unlike other distribution systems, with Q-LAN, there is no requirement to assign and remember names or numbers for signals. Simply drag wires from source to destination and the connection is made for you.

Clock Distribution

Any digital audio and/or video distribution system must deliver both the media data and its corresponding clock. Q-LAN accomplishes clock distribution using the IEEE 1588 Precision time protocol (PTP). A variation of PTP is used in AVB. PTP is under consideration for use in next-generation video distribution systems standards under development by SMPTE.

Under PTP, one device on the network is elected Grandmaster. The Grandmaster transmits periodic time updates to all other PTP participants on the network. Participants also periodically do timed interrogation of the Grandmaster. These measurements are used to compensate the time base at each participant for any fixed latency introduced by the network. Q-LANs 48-kHz sample clock runs synchronously with respect to the PTP-distributed clock. All audio streams operate synchronously with respect to this master clock.

This common sample clock approach maximizes routing and processing flexibility – with all audio represented according to the same clock, it can be readily routed and combined without synchronization or alignment issues. The master clock scheme does however require that any digital audio sources to the network be pre-synchronized to the network's clock or that sample-rate conversion be employed to accommodate digital signals to the network's clock. The AES3 digital interfaces in the I/O Frames feature integrated high-quality sample-rate conversion.

Network Management

Q-Sys™ is a comprehensive and integrated system platform. The capabilities of the platform include audio distribution, audio processing and system control and monitoring. The control and monitoring capabilities of Q-Sys include control and monitoring of Q-Sys hardware as well as external components. With these comprehensive capabilities built into Q-Sys, the system is self monitoring. A separate network monitoring system is unnecessary in most Q-LAN installations.

The Q-Sys system and thus Q-LAN also features the ability to be controlled and monitored over the network by external entities or systems. If desirable, Q-Sys and Q-LAN can be integrated into larger network management platforms.

Q-LAN Network Requirements

Q-LAN is a high-performance real-time audio distribution system and as such requires real-time performance from the network which hosts it. The performance is achieved through the following means:

Use of gigabit Ethernet throughout – Gigabit Ethernet provides 10 times the bandwidth compared to previous generation 100 Mb fast Ethernet. Significantly, it does so at one tenth the latency. Even if the channel count of your application does not justify using gigabit Ethernet, the reduced latency will. Q-LAN requires use of gigabit Ethernet end-to-end for all paths carrying audio through the network

Use of Quality of service (QoS) capabilities in network equipment – QoS allows network equipment to expedite delivery of time-sensitive traffic such as Q-LAN audio and timing signals over non-real-time traffic such as file transfers. Without QoS, network performance is indeterminate.

Limiting the physical extent of the network – Network design guidelines limiting the physical extent of the network assure that data is not unduly delayed in long cable or fiber runs or by accumulated latency of a multitude of network switches.

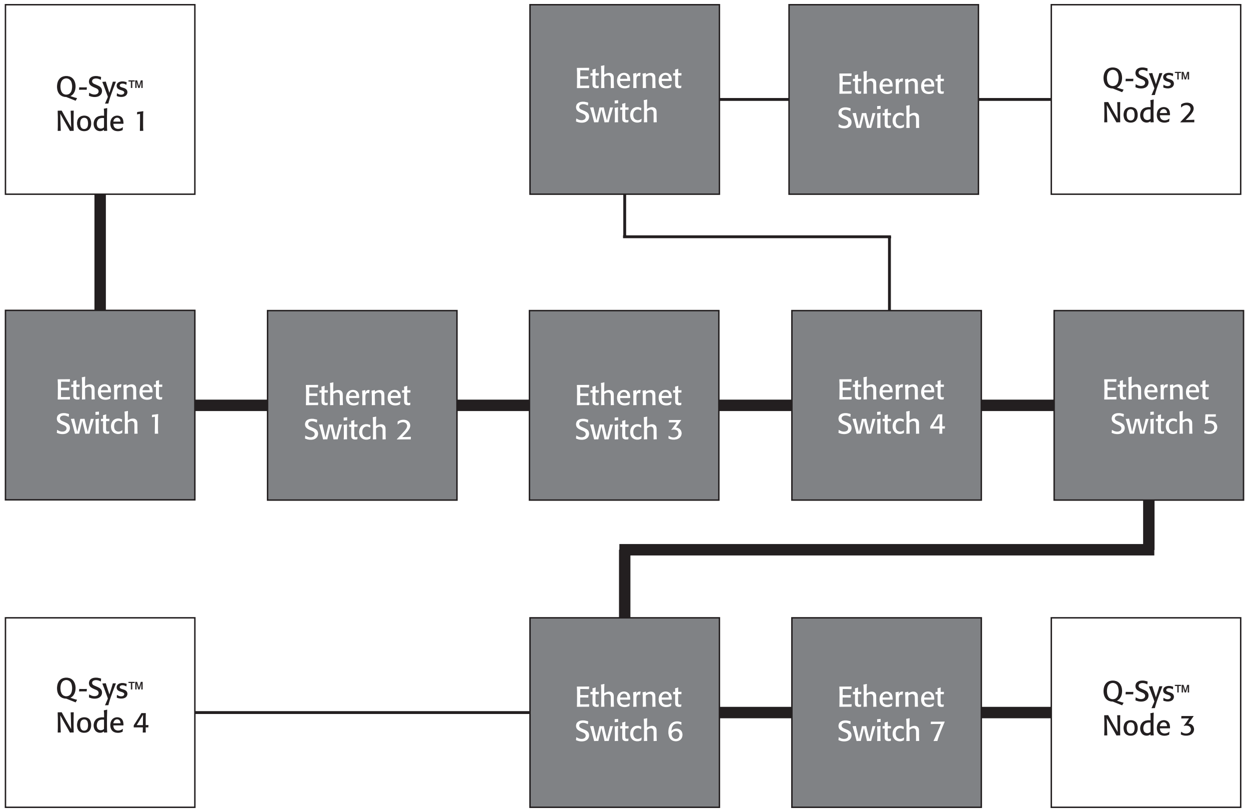
Quality of Service

Quality of service enables the network to distinguish different traffic types and for these different types of traffic to receive different treatment depending on their priorities. QoS capability is required by Q-LAN. QoS is the mechanism which allows Q-LAN to comingle with other network applications, (including other network audio distribution technologies such as CobraNet), on the same infrastructure. Q-LAN does not require a separate infrastructure. It does not require audio traffic be segregated through Virtual Local Area Network (VLAN) configurations. The QoS mechanisms do this globally for the network and with minimal configuration requirements.

Q-LAN uses layer-3 DiffServ QoS support. Q-LAN operates three distinct traffic classes.

- 101110 (46) Expedited forwarding (EF) for clock transport.
- 100010 (34) Assured forwarding (AF41) for audio transport.
- 000000 (0) Default classification for control communications.

In support of this scheme, the QoS mechanism on a switch used in a Q-LAN network must feature a minimum 4 egress queues per port. Once traffic has been separated into different classes and placed into respective queues, the switch must determine an appropriate transmission ordering. Switches typically offer several choices as to transmission strategy such as round robin, weighted round robin, weighted fair queuing or strict priority. Q-LAN requires strict priority selection. Under strict priority, the switch transmits all high priority traffic before any lower-priority traffic is transmitted.



7-hop network

Network Size

Limiting the size of the network helps ensure that performance required by Q-LAN 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.

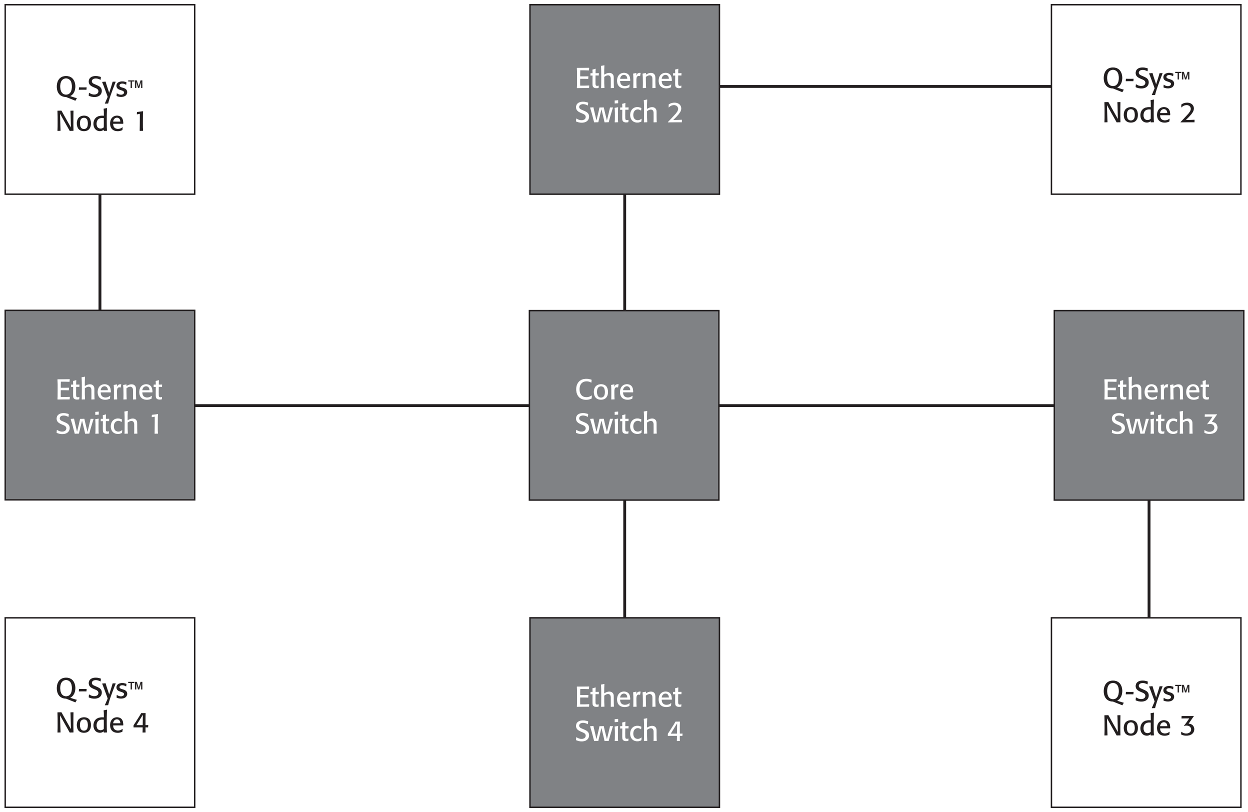
Total latency through Q-LAN is 2/3 ms. About half of that time is budgeted to network delays (the other half is budgeted to buffering and packetizing). 1 km of twisted pair or optical cable imparts almost 5 μs delay. The minimum delay through a standard gigabit Ethernet switch is 12 μs; Maximum delay can be several times higher[4].

Q-LAN 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. **Table 1** shows allowed network diameter as a function of hop count.

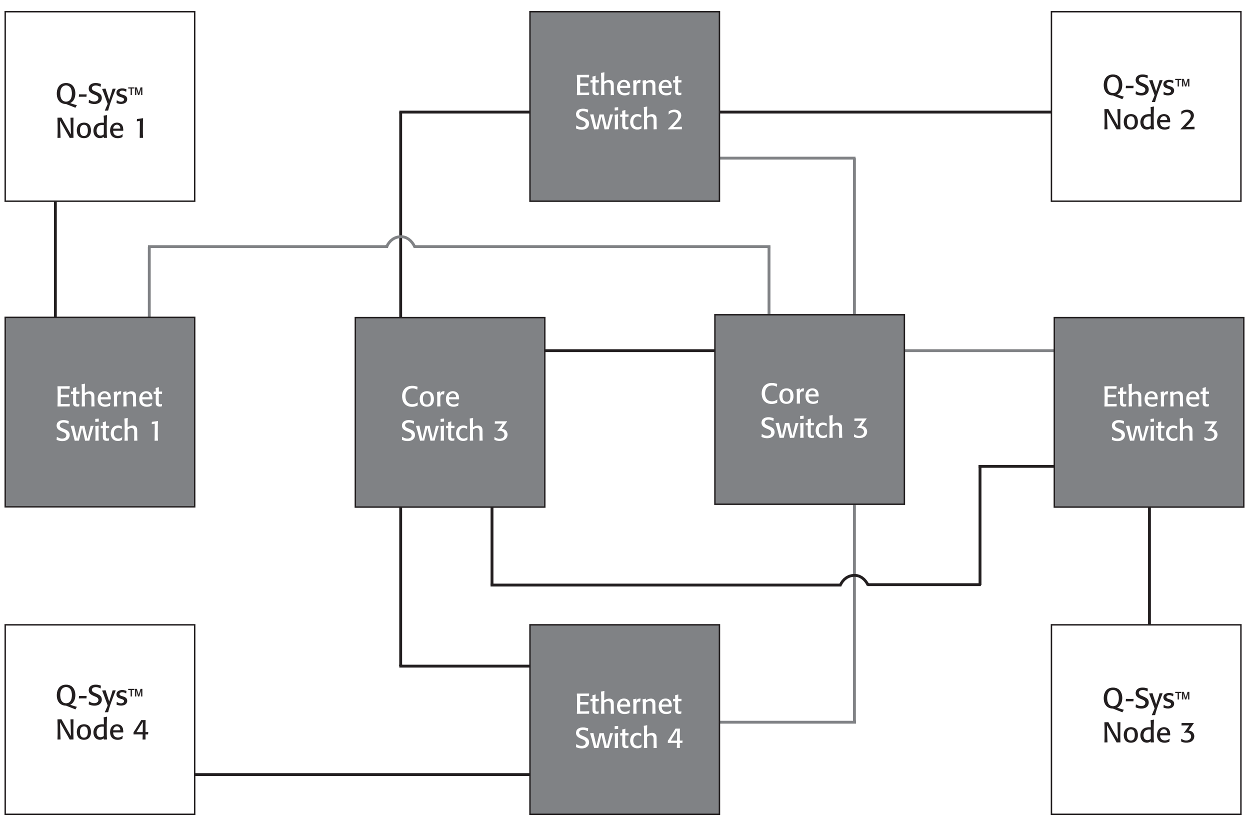
Hops	Diameter
2	35 km
3	29 km
4	22 km
5	15 km
6	9 km
7	2 km

Allowed network diameter as a function of hop count
Table 1

[4] Standard gigabit switches use store-forward switching. Exotic switches use cut-through switching and can have lower latencies. This discussion assumes QoS is deployed. Without QoS, delay can be unbounded.



3-hop core-switched network



4-hop redundant core switched network

Switch Performance

The high performance required by Q-LAN implies performance requirements for the Ethernet switches in the network that hosts it.

Internal bandwidth – Q-LAN expects non-blocking wire-speed gigabit Ethernet bridging: No dropped packets due to internal bandwidth constraints; no flow control invoked or required.

Buffer capacity – At least 80 Kbyte total egress buffering available for audio class of service with minimum of 40 Kbyte available per port for this class.

Forwarding decision time – The time from receipt of the last bit of the packet at the ingress port to the transmission of the first bit of the forwarded packet at the egress port must be 10 μs or less.

Layer-3 Networking

Q-LAN operates well with conventional layer-2 Ethernet networking. Many high-performance networking components now include layer-3 capabilities such as IGMP, WAN multicast management protocols, wire speed IP routing and support for internet routing protocols. Unlike most other networked audio distribution systems, Q-LAN can take advantage of these capabilities should you choose to activate them. These provisions make for a more scalable, robust and manageable network.

Jumbo Packets

Data on Ethernet is transmitted in packets. Ethernet packet size is limited by the IEEE 802.3 standards to 1522 bytes. A larger packet size has the potential to reduce overhead in some communications protocols. Although the IEEE has refused to condone an increase in the Ethernet packet size, many vendors, bowing to customer demand, have implemented support for jumbo packets. Jumbo packets are generally defined as having a total length between 1523 and 9216 bytes.

Network performance criteria for Q-LAN require no jumbo packet through all audio paths on the network. The presence of jumbo packets on Q-LAN-shared network links, even if on a separate VLAN, will introduce additional network latencies making a single switch hop produce the same delay as 6 switch hops with standard framing.

Jumbo packets will only be present on a network if network equipment is configured to pass them and end stations are configured to generate them. Fortunately, managed switches typically ship with jumbo-packet support disabled by default. End stations such as routers and servers also generally ship with jumbo packets disabled. A concerted and systematic configuration effort is required to enable jumbo packets on a network.

Flow Control

Low-level flow-control protocols (principally 802.3x) are used to prevent ingress buffer overflows. Modern switches have adequate internal bandwidth such that input buffer overflow is not a concern and 802.3x flow control is considered by many vendors to be a relic. Switches with internal bandwidth meeting or exceeding wire speed are not expected to initiate flow control.

Being a latency-critical application, Q-LAN cannot tolerate the delivery delays created when flow control is invoked. Flow control must therefore be disabled or not given a chance to be invoked through the network paths used by Q-LAN.

Managed vs. Unmanaged Switches

A managed switch contains an intelligent entity which can be communicated with for the purposes of configuration and monitoring. Many of the advanced features found in current-generation network equipment require configuration. Examples of these features include: STP, Link aggregation, DiffServ, Broadcast storm suppression, IGMP, SNMP, HTTP, Telnet, VLANs, IP routing. Since Q-LAN uses DiffServ, managed switches are required. Most network managers will find multiple features on the above list which they are unwilling to forgo.

With 100 Mb fast Ethernet networks, unmanaged switches were often an attractive and cost-effective option. With the new generation of gigabit Ethernet switches, unmanged switches are less common and there is little price difference between the managed and unmanaged ones.

Switch Testing

QSC has tested a number of switches whose specifications meet these Q-LAN requirements. Without exception equipment meeting specification have operated without issue.

One could use network equipment for Q-LAN with confidence based solely on published specifications were it not for the fact that not all relevant specifications are published by all manufacturers. Specifically, details on buffer sizes and QoS selection strategy are often missing from published specifications.

QSC therefore publishes a list of network equipment which has been tested to Q-LAN specifications. An updated list can be found in the Q-Sys™ Designer help system.

Conclusion

Integrated within QSC's Q-Sys integrated system platform, a new networked digital audio distribution system has arrived. Compared with previous-generation systems and competing current-generation offerings of both the shipping and gestating variety, Q-LAN offers lower latency, higher fidelity, higher capacity and more comprehensive fault tolerance capabilities. Q-LAN operates on a cost-effective commodity gigabit Ethernet local area network. Since it uses some of the same protocols used on the Internet, Q-LAN has the ability to extend beyond the confines of the local area network. Q-LAN's integration within Q-Sys makes for a point-and-click networked audio distribution experience.

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 (Q-Sys™)

The Core is Q-Sys' 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.

Fast Ethernet

A 100 Mbit/second Ethernet variant. Sub variants are available for twisted pair cabling (100BASE-TX) or fiber-optic cable (100BASE-FL).

Gigabit Ethernet

A 1000 Mbit/second Ethernet variant. Sub variants are available for twisted pair cabling (1000BASE-T) or fiber-optic cable (1000BASE-SX/LX).

Grandmaster

Source of master clock in an IEEE 1588 clock distribution system.

HTTP

Hypertext Transport Protocol is used by web browsers to retrieve content from web servers. An appliance supporting HTTP can be assumed to feature an integrated web server.

I/O Device

The I/O devices are the entry and exit points for audio in the Q-Sys™ system.

I/O Frame

The I/O frame is an I/O device that supports up to a total of 16 channels in and 16 channels out through up to four modular audio I/O cards installed in the Frame.

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.

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.

Q-Sys

Q-Sys is a complete integrated system platform that encompasses everything from the audio input to the loudspeakers. Q-LAN is the networked distribution component of Q-Sys.

Q-Sys Designer

Q-Sys Designer is the user interface used to configure, control and monitor Q-Sys audio signal processing and network routing. Q-Sys Designer is a Windows application that runs on a computer connected to the Q-LAN network.

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

Q-LAN 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 intervention or special configuration servers.