Computer-control systems for audio

Although computer control can simplify the use of the system, it can significantly complicate the task of system design.

By Steven J. Thorburn, P.E.

What is computer control of audio? Try looking up these three words in the dictionary. Webster defines a computer as an electronic machine for calculating. Control is defined as the power to direct or regulate. Audio is defined as relating to sound. With these definitions, music boxes and player pianos can be considered examples of the first computer-controlled audio sources.

Computer control of audio systems has come a long way since those rudimentary beginnings. Today's computer control is sophisticated enough to allow a technician in one country to configure, test and reprogram a control system in another country halfway around the globe.

What can you do when you need to create a sophisticated audio system with state-of-the-art capabilities but you know it's going to be operated by people who can't make their VCRs stop flashing 12:00? The answer is easy. Just build the intelligence into the system. With today's computer control, you can make the most difficult operational, configuration and testing tasks as simple as pushing a button.

Although computer control can simplify the use of the system, it can significantly complicate the task of system design. Designers must choose components to satisfy the audio needs of the system, but they must also ensure that they incorporate the right amount of machine intelligence and communications capabilities for proper system integration to meet the client's needs. It can also make it more challenging for the installer to set up a system. Three or more computers and a wealth of computer knowledge might be needed to do the initial installation set-up of the system.

When used properly, computer control can allow a complex network of diverse equipment to function together as one monolithic device. It can provide the system administrator with the ability to change configuration parameters or test and troubleshoot the system from a remote location. The designer can create multilevel, tamper-proof security to insure that only qualified people can access sensitive system areas. Reconfiguring the system by the push of a button or at a specific pre-programmed time is also a design possibility. With the low price and incredible power of today's computer processors, just about anything a client or designer can dream up can be realized.

A basic understanding of the application of computer control in the audio world is a prerequisite for any audio-video contractor working today.

Regardless of what type of computer-control system is being used, creating computer control of audio systems involves two separate components: hardware interfaces and software communications protocols.

Hardware interfaces are concerned with details such as connector types and pin definitions, allowable data rates, network topology (architecture configuration), current or voltage loops and a signal's electrical characteristics.

Software protocols define details such as data character sets, data packet (words) sequence and timing requirements and procedures for determining when an error has occurred and what to do when it happens.

Public domain standards include one or both of these components: AES 24/SC-10, MIDI, RS-232, RS-423, RS-422, RS-485, parallel, home RF and X-10 and 20mA current loops. Many manufacturers have taken the hardware interface standards and added their own software protocols to create proprietary control-system architectures.

The capabilities generally referred to as computer control of audio can be broken down into four broad automation categories: functional control; automated testing, set-up and central operations; digital signal processing systems; and computer-generated audio — multimedia or sound cards.

Functional control

This category includes control of specific equipment functions as well as user-interface design and control of required nonaudio remote devices. It usually includes control of detail, such as transport functions, volume control, routing of signals, source selection and device interface. This control can take place through infrared, MIDI, serial data links or contact closures and, depending on system complexity and integration, does not necessarily require any additional equipment.

User-interface design is at the heart of computer-based functional control. The design is normally dictated by the difference between the sophistication of the audio-control system and the technical sophistication of its users. Simple systems that are basically self-explanatory need nothing more than switches and volume controls.

More complex systems can cause frustration and anger in those not trained in their operation. Very sophisticated systems operated infrequently or by untrained personnel demand an entire control system dedicated to user-interface tasks. This type of control system can shield the users from the inherent complexities of the system and allow them to perform the required functions with little or no training. Examples of this type of user interface controllers include AMX and Crestron systems. With the right software programming, these sophisticated control systems can create simple user interfaces for even the most complex systems. An added advantage to this type of user interface is the ability to modify it easily through software to meet changing operational needs or requirements.

On a large project I was working on last year, four computers were in ac-

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tion during installation. The first computer was running a TEF audio analyzer to set the room EQ and loudspeaker alignment; the second was adjusting the programmable EQ; a third was being used to program the control system; a fourth was in the rack being configured as the system controller. It took three people and four computers just to get the system up and running. With this level of complexity in audio system installation and operation, it's no wonder that users are crying out for simple interfaces that provide them the luxury of operating the system without having to know anything about it.

Automated testing, set-up and central operations

Computer-control systems can facilitate automated testing of complex or performance-critical systems. Both continuous testing and programmed interval testing can be set up, allowing the system technicians broad discretion in implementing performance testing regimens.

The computer now provides us with the ability to tell, from a remote location, whether a specific loudspeaker is receiving the amplifier's signal. As a follow-up, frequency-specific automated tests can be performed at night when the facility is empty. This testing tells the technician the status of the system from the point where the test signal is injected to the end of the loudspeaker run and can accurately determine whether that signal path has suffered a fault. By setting up several different tests, one can easily judge whether the problem is in the high-, mid- or low-frequency range of the loudspeaker system. It can also alert the technician if the signal is not making it through the amplifier or if the problem is with the processing equipment. These testing systems are especially important to the clients, who can run these automated tests to find out whether anything needs to be repaired or replaced before it affects their next presentation or show.

Computer programmability of equipment can even make load-in and system setup a lot quicker. Once a crew has worked in a venue, it can save the basic set-up away on disk and later recall the last setting as a starting point. This particular use of computer control speeds up repetitive tasks. It can also make system setup tamperproof by limiting the number of accessible controls. The setup engineer can keep a copy of all setup parameters in case of future problems.

Automated testing and computer programmability enables effective use of central control rooms and remote amplification racks. With a fiber-optic backbone, the audio and control can be sent out to remote amplifier racks.

They said you were a twit in 3rd grade.

Communications standards

By Kevin Jennings

Most computer control of audio systems is accomplished with standard hardware and software protocols. Protocols are nothing more than an agreed upon set of definitions that allow any manufacturer to create equipment that will communicate with other equipment using the same definitions. Hardware protocols define the characteristics of the physical connections between devices, such as the number of wires, voltages, timing sequences, signal definitions and signal names. Software protocols define how the hardware protocols are translated into machine language commands and data. These protocols are administered by various organizations, including the Audio Engineering Society (AES), Society of Motion Picture and Television Engineers (SMPTE) and American National Standards Institute (ANSI). Tables A and B list some of the most widely used protocols with their vital statistics.

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Table A. Hardware communications protocol

MIDI

Vital statistics:
- A musical instrument digital interface.
- Released in 1983.
- Five-pin DIN connector — in, through and out.
- 5mA current loop, optically coupled, output to input.
- Data rate: 31,250 baud or 31.25 kbps.
- 16 channels of communication.
- Distance limitation is 98 feet (30m) between equipment.

Benefits:
- Popular in the music industry.
- Low cost to implement.
- Expansive — new features have been added since its inception.
- Links everything from musical instruments to computers.
- Has been universally supported by the industry.

Weaknesses:
- Only supports 16 channels of different conversations at one time.
- Unidirectional, with no message-received response.
- 31.25 kbps is slow for a data network in today's terms.
- Seven-bit word makes interfacing with other eight-bit systems difficult.
- Weaknesses are not a problem in small systems, but large systems have severe limitations.
- MIDI has not effectively made the transition to the professional audio industry.

RS-423

Vital statistics:
- A single-ended synchronous or asynchronous serial data hardware.
- Links between equipment are normally limited to 4,000 feet (1,275m).
- Requires three wires for asynchronous connections, more for synchronous operation.
- Unbalanced.

Benefits:
- Allows very long wiring runs.
- Higher data rates than RS-232.
- Maximum data rate is 100kbps.

Weaknesses:
- Single-ended, connects only one transmitter to one receiver.
- Relatively low data rates compared to RS-422 and RS-485.
- Sensitive to ground-plane problems.
- Availability of equipment with compatible I/O port is less than with RS-232.

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The control allows for adjustment of the delay if the amplifier is equipped with a DSP card. It can even allow adjustment of zone EQ if the stage acoustics change relative to the loudspeaker system the amplifier is serving. Automated testing allows the automation of quick, basic tests or complex, time-consuming tests. It also provides the technician with the critical information needed before sending a crew out to the zone for maintenance.

Digital signal-processing systems

Digitizing a signal provides the ability to manipulate it without loss or distortion. Digital processing affords us the luxury of ignoring transmission losses and electrical noise problems within processing systems. We can even reconstruct damaged signals through parity and error checking.

Systems based on digital signal processing (DSP) should be considered a true black box. The signal going in is processed and comes out totally different. The only limitation is how much processing power is provided by the box. DSP technology allows us to develop black-box processors that have analog inputs and outputs, with all processing done in the digital domain. The CPU will process the signal based on the mathematical models it is given; EQ, delay, mix and split can all be applied before the signal is sent to the output. In fact, any signal transformation that can be documented can be applied. This ability allows processors to be programmed to simulate natural phenomena or transformations that would be completely impossible in the analog domain.

Early units from TOA and White provided processing for cluster loudspeaker systems or projects that required a great deal of processing. They were, and still are, excellent tools if you have that specific need. IED's UDAPS was the first large-scale DSP-based signal router and mixer. The technology of Peavey's Media Matrix is now available for even relatively small projects.

Ten years ago I asked a friend why he was still collecting LPs and didn't buy CDs. He explained that at that time his pre-amp was not digital, and his power amp did not have a digital input. Because the CD player was in reality an analog source at its outputs, he didn't see a big advantage in changing out one analog source for another. (Of course, the higher cost of CDs may also have been a factor.) Ten years later, digital I/O is a relatively common feature on pre-amps and even some amplifiers. DSP technology allows us to process the audio signal in a totally digital environment.

Spice tools of the '70s and '80s developed the mathematical models for circuit and equipment design. The spice model of an operational amplifier can be modified to function as an audio power amplifier. Bandpass filters can be stacked together to form a 1/3-octave EQ circuit. A delay is a delay. From these building blocks, defining the mathematical transform of the device became relatively simple. Model, frequency response, gain and phase information could be determined for any input or output. The dawning of the digital age is past. We are now standing in the glow of its early morning sky.

Computer-generated audio — multimedia or sound cards

Digital audio recording and synthesis have created new ways of supplying messages, announcements, music and sound effects to today's audio systems. Because of the vast amount of knowledge incorporated in this field, computer-generated audio will not be covered in this article. If you want to know more about this subject, plenty of reference books and periodicals focus entirely on this subject.

How is computer control used?

Computer control is used for corporate theaters, stadiums, multi-use boardrooms, hotels, convention centers and specialty attractions. Each venue has characteristic problems that can be addressed with computer control.

• *Corporate theaters, stadiums and multi-use boardrooms:* The room setups for these types of installation can change with each presenter or event. The marketing department might have just asked to use a four-person table during a panel discussion. Shouldn't be a problem, right? The catch is that the room is turned 90°, and they want it done at a snap of a finger. Twenty minutes after the panel discussion finishes, you have to have the room set up for the corporate board meeting with video teleconferencing. Different room uses, special effects, queued sources at specific times, reconfiguration and system presets all benefit from properly implemented computer control.

• *Hotels and convention centers:* These venues can benefit from computer-controlled systems because the hotel's events can be scheduled into a database. A computer-controlled system can automatically reconfigure the room combining, reset delay from head tables at different locations, reconfigure paging zones and set volume levels and EQ, all with no operator intervention.

• *Specialty attractions:* Theme parks and museums can benefit from all of the functions provided by computer-control systems. The ability to control the system from a single point and verify the operation before the facility opens is a plus. The system will automatically adjust the volume as the crowd noise changes, and computer-generated audio can be used for announcements and special effects. User-interface design configuration depends on the attraction or venue type. Upgrades to DSP-based processing units can usually be done at low cost.

Control-system manufacturers

Computer control of audio has been around for a number of years. Most boardrooms, training rooms or classrooms with some type of control system for changing the volume or source use a computer-controlled communication network between the control panel and the central processing unit.

AMX and Crestron create both user interface hardware and software, but their systems also act as gateways that concentrate and interpret all types of control signals and redirect the information to the intended devices.

The Crown IQ system is a computerized system that uses Crown's IQ Software Command Protocol to control and monitor the various functions of an audio system over a Crown-Bus proprietary hardware interface. The IQ concept was developed in the early

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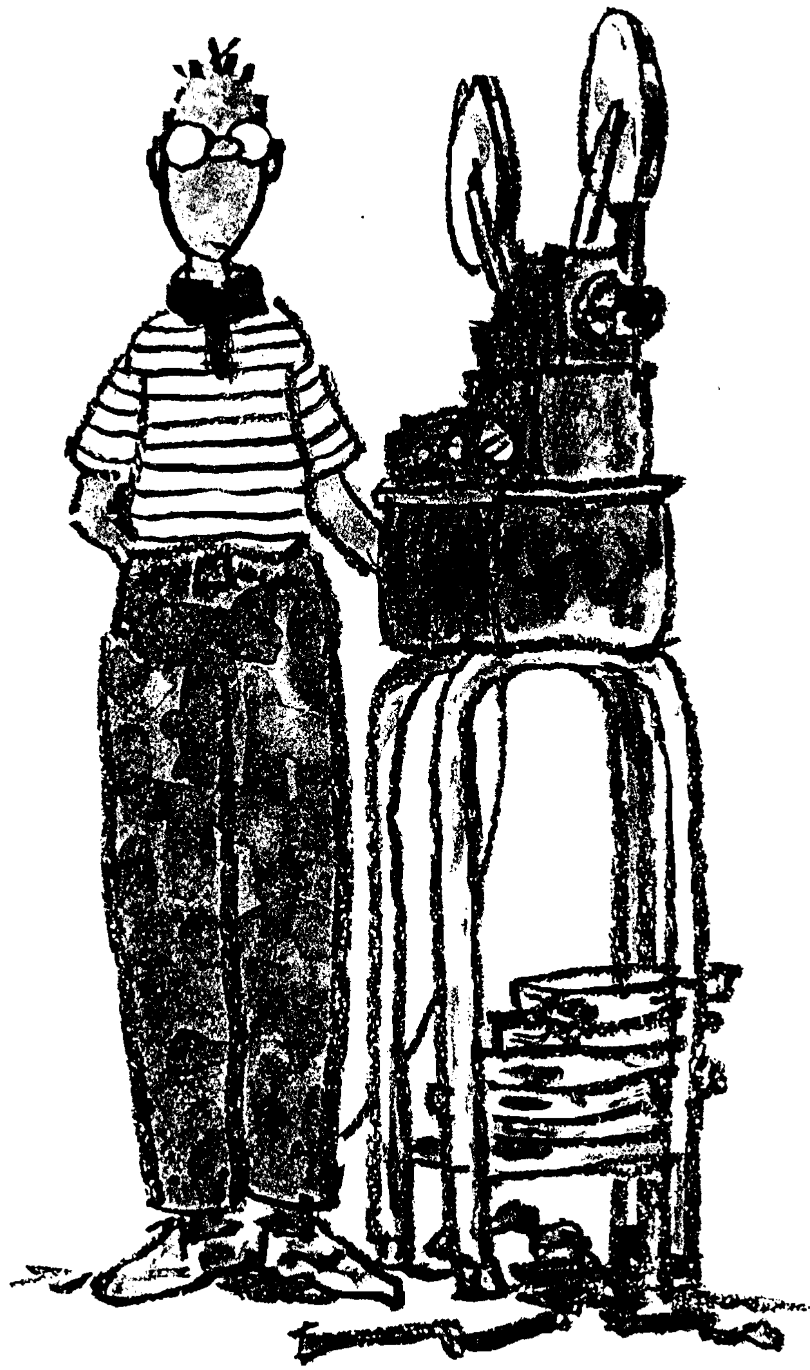

They said you were a nerd in high school.

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RS-232
Vital statistics:
• A single-ended synchronous or asynchronous serial data hardware standard.
• Links between equipment are normally limited to 50 feet (16m).
• Requires three wires for asynchronous connections, more for synchronous operation.
• Unbalanced, single-ended.
• Maximum data rate is 20kbps.
Benefits:
• Most widely supported serial communications hardware standard.
• Currently available as a standard I/O port on a large variety of equipment.
Weaknesses:
• Single-ended, connects only one transmitter to one receiver.
• Relatively low data rates.
• Unusual voltage (-25V to +25V) requires unique power supply in equipment.

RS-422
Vital statistics:
• A more robust serial digital data interface standard.
• Uses individual differential signal pairs for data transmission in each direction.
• Links between equipment are normally limited to 4,000 feet (1,275m), depending on data rate.
• Maximum data rate is 10Mbps.
• Requires four wires.
• Balanced.
• Can broadcast to up to 10 receivers.
Benefits:
• Well-supported broadcast serial hardware standard.
• Allows very long wiring runs with no perceived degradation of signal quality.
• Higher data rates than RS-232.
• Maximum data rate is 10Mbps.
• Not susceptible to ground-plane problems.
• Voltage levels can be achieved with standard ±5V power supplies.
Weaknesses:
• Availability of equipment with compatible I/O port is less than RS-232.
• More conductors required for each connection.
• Requires one pair of wires for each communication direction (transmit and receive).

RS-485
Vital statistics:
• A robust multidrop serial digital data interface standard.
• Multidrop (party line) signal path.
• Multiple devices with unique addresses share a single signal path.
• Maximum of 32 transmitter and 32 receiver devices.
• Links between equipment are nearly immune to interference, more reliable in demanding environments.
• Effective at distances of 4,000 feet (1,275m) and beyond.
• Requires four wires for full duplex.
• Capable of half duplex with only two wires because of its multidrop design.
• Balanced.
Benefits:
• Well-supported point-to-point serial interface.
• Allows very long wiring runs with no perceived degradation of signal quality.
• Higher data rates than RS-232.
• Maximum data rate is 10Mbps.
• Not susceptible to ground-plane problems.
• Multiple devices with unique addresses share a single signal path (32 max).
• Allows half duplex on one pair.
Weaknesses:
• Availability of equipment with compatible I/O port is less than with RS-232.
• More conductors are required for each connection.

Current loop
Vital statistics:
• No formal standards for current loops.
• Informal standard of 20mA of flowing current equaling a logic 1.
• No current equaling a logic 0 is commonly used.
• Voltage is present on current loops, but the precise voltage does not matter as long as it does not exceed equipment maximums.
Benefits:
• Interfaces easily with opto-isolated equipment.
• Highly immune to EMF interface.
• Extremely high signal-to-noise (S/N) ratios are obtained.
Weaknesses:
• Difficult to implement.
• Requires a closed loop for operation.

Parallel
Vital statistics:
• Single-ended synchronous data hardware interface standard.
• Links between equipment are normally limited by driver power.
• Requires 17 wires for eight-bit operations.
• Unbalanced.
Benefits:
• Availability of equipment with compatible I/O port is very good.
• Higher data rates than RS-232.
Weaknesses:
• Single-ended, connects only one transmitter to one receiver.
• Requires a large number of conductors for implementation.
• Can be sensitive to EMF problems.

SMPTE time code (ANSI/SMPTE 12M)
Vital statistics:
• Synchronization signal based on an analog 2,400Hz square wave.
• Different time code formats specific to the recording media.
• Types: Linear or longitudinal (LTC), vertical interval.

Home RF or X-10
Vital statistics:
• Broadcast asynchronous data hardware interface standards.
• Links between equipment are normally limited by driver power.
• X-10 modulates the 60Hz power waveform for transmission.
Benefits:
• Availability of compatible equipment is very good.
• Requires minimal additional wiring for operations.
Weaknesses:
• Single-ended.
• Requires a large number of conductors for implementation.
• Can be sensitive to EMF problems.

Table B.
Software communications protocol

There are three types of protocols: public systems, proprietary systems and DSP-based systems. Public systems — systems with published definitions — include AES 24/SC-10 and DMX 512

AES 24/SC-10:
Mission Statement — Recommend standards for communication of digital data within, into and out of audio systems, taking into .
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1980s and has grown considerably over the last 15 years. The IQ system has a two-wire serial data loop RS-232 for the PC and RS-422 for the Macintosh.

The Crest NexSys system is a computerized system that uses Crest's NexSys software Command Protocol to control and monitor the various functions of an audio system. The NexSys system was introduced as an amplifier controller in 1990. It operates using an IBM-PC interface card and Windows-compatible software and communicates over an RS-485 line at 57kbps. All devices are daisy-chained, with the last device requiring that termination resistor standard connectors are specified. Crest amplifiers interface through a DB 9 connector. One of the major benefits of NexSys is that it is fully compatible with MIDI equipment and can control or be controlled through a bidirectional MIDI interface.

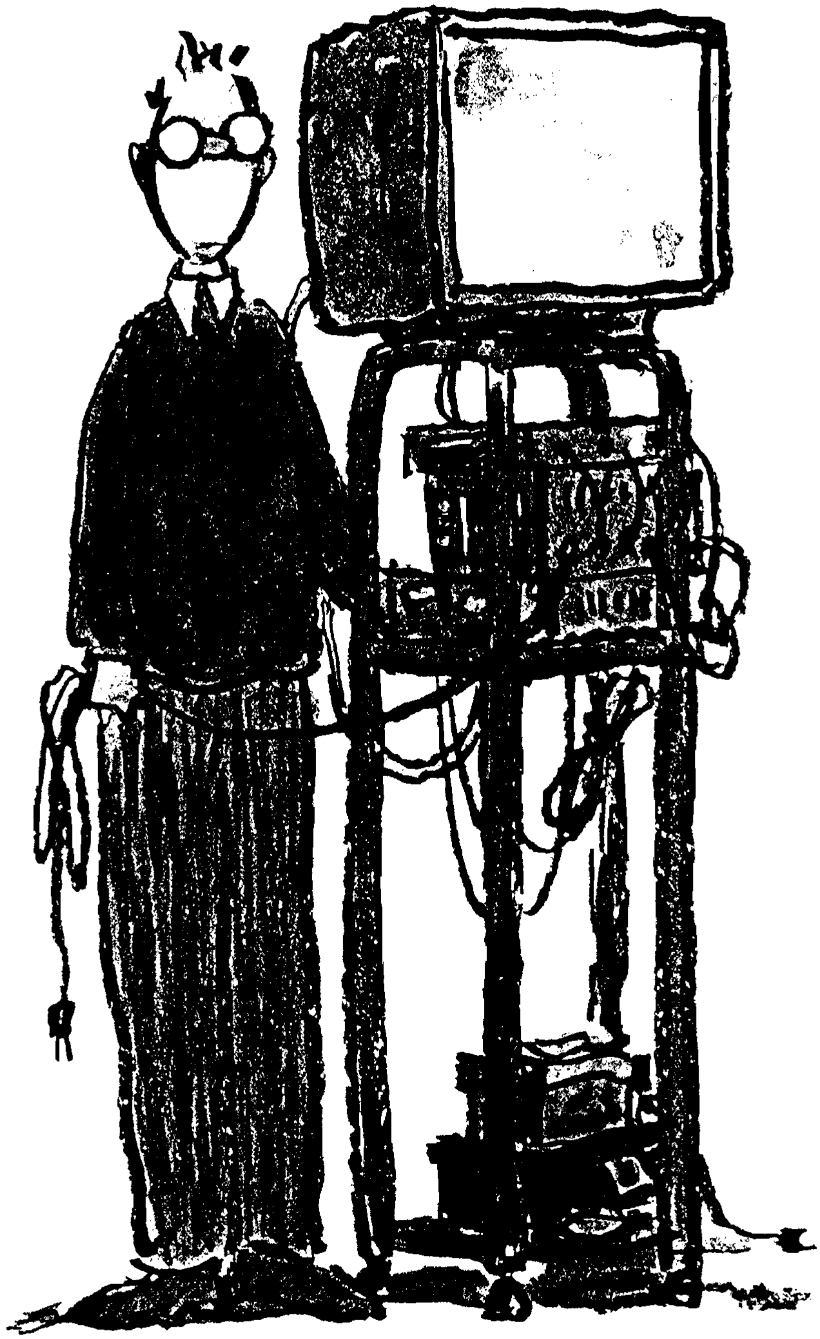
Lone Wolf's MediaLink, a format-independent network communications protocol, allows for the seamless and transparent connection of electronic devices into a fault-tolerant network. It has the unique quality of being compatible with a variety of existing hardware and software, yet it is fully adaptable to the rapidly changing performance requirements of today's network users.

Richmond Sound Design Show Control Software tracks the times at which queues for lights, sound, flys and such are called by the stage managers and then executes pre-programmed queues precisely as they are timed in the computer. The system also gives the stage manager the ability to easily override queues, start the queues early or late or skip them altogether. Examples of controllable equipment include microphones, audio mixers, source equipment, ambient volume control, switchers, routers, processing equipment, amplifiers, EQs, delays and DSP black boxes.

Other manufacturers have realized how important it will be for their equipment to be controllable from a central system. They realize that programmability control for their different mixers, routers or paging controllers would serve a larger group if they could have their equipment interconnected.

In my opinion, one of the most interesting advances has been the development of the DSP black boxes. TOA and White began their work in this area with the Saori and D5000. These boxes allow the designer to select different signal paths and add different items. IED allows for creation of large black boxes for major projects with lots of inputs and outputs. The opportunity to have a reconfigurable, true black box with only source signal in and amp outs is extremely appealing.

On a smaller scale, the TOA DACsys II



They said you were a geek in college.

"Communications Standards"

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account possible future applications in more general media systems.

DMX 512

System basics:

- Created in the mid 1980s as a lighting control protocol.
- Also used to control other systems by way of standard interface devices and alternate start codes.
- Can address up to 512 devices on each DMX 512 link.
- Communicates over an RS-485 line at 250kbps.
- A single link can accommodate up to 32 separate receivers.
- XLR-5 connectors are specified as standard.
- Transmitting devices use female connections.
- Receiving devices use male connections.

Benefits:

- Interface with lighting equipment.
- Interface with other equipment types via standard interface devices and alternate start codes.
- Highly immune to EMF interference.

Weaknesses:

- Output resolution is limited to eight bits (256 levels).
- Updates of a fully configured 512 address link occur only 44.11 times per second using typical data.

SMPTE Time Code (ANSI/SMPTE 12M):

A synchronization signal based on an analog 2,400Hz square wave, which is modulated to contain absolute time- and frame-reference information.

A number of different time-code formats are each specific to the recording medium it is used on. 24fps time-code is used for film production; 30fps is used for black-and-white NTSC video and most show-control applications; 30fps Drop Frame is used for NTSC color video to compensate for its 29.97fps sync rate; and European Broadcast Union (EBU) 25fps time code is used for PAL video.

There are two types of time-code. With linear or longitudinal time-code (LTC), the time code is recorded on an analog audio track. With vertical interval time-code, the time-code is recorded as digital data in a video signal's vertical blanking interval. (This setup allows freeze-frame video to still have a time-code reference.)

MIDI show control (MSC)

MIDI show control is a standardized language protocol for peer-to-peer controller interconnection in a show environment. Because it is intended as a controller-to-controller protocol, it is not a replacement for controller-to-device protocols such as DMX-512 or IQ. MSC is a set of real-time, system-exclusive MIDI messages that contain show commands. The messages are standard MIDI SysEx messages, which use the Universal Real-Time System exclusive ID.

MSC has two main limitations. The command-response time limitations of MIDI can cause real-time messages in large systems to be delayed. This problem can be avoided by limiting the number of devices on a single MIDI link. The second limitation is the open-loop nature of MIDI, which makes it unsuitable for controlling potentially life-threatening systems. This limitation is being addressed in the new MSC 2.0 specification by the inclusion of two-phase commit protocols. All actions are acknowledged before execution in a two-way data link.

is an interesting box that allows for adjustment and processing of relatively complex settings for a loudspeaker system. For the price of a few equalizers, limiters and crossovers, you can save your client some money and yourself some time in the setup and checkout.

Programmable EQs were really brought into vogue by

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Computer control and the AES

In 1992, the Audio Engineering Society (AES) formed a subcommittee called SC-10 to create a local-area-network standard for sound systems. AES realized that the lack of standards was not in the industry's best interest The 13th International AES Conference, "Computer-Controlled Sound Systems," was held in December 1994 in Dallas. The proceedings should be read by anybody interested in understanding the complexity of creating standards for computer-controlled audio. (AES can be reached at 212-661-8528.)

Working groups were developed under SC-10 to carry out the tasks of the subcommittee. Each subcommittee developed a mission statement:.

- **SC-10-1 on data communications:** Recommend standards for communication of digital data within, into and out of audio systems, taking into account possible future applications in more general media systems.
- **SC-10-2 on application protocols:** Design an application protocol for the remote control and metering of audio devices through digital data network. Application protocol refers to a set of data elements and exchange rules that implement the actual control and monitoring functions the end users see.
- **SC-10-3 on information:** Serve as a clearing house for SC-10 information.
- **SC-10-4 on PA-422:** AES15 is the current standard for PA-422. SC-10-4 is looking at adding information to the standard.

The latest and greatest from AES 1995

As with all projects, just as you think you have them finished, another issue arises. The main article on computer control of audio systems was submitted before the Audio Engineering Society (AES) Conference in October, and at the show, two major systems were announced.

The first system is QSCcontols 2 from QSC; the second is HAC from Harmon. The concept behind these two products is to adopt an existing network standard and file transfer protocol and develop packets of information for transfer between machines. The systems require at least one smart host because each machine has not the master control protocol, but rather the specific information on what is controllable and the function and range of the control.

The hardware protocol selected is Ethernet, and the software that is currently used is TCP. The fact that both Harmon and QSC have arrived at the same "block" is not a coincidence. They have worked closely together to reachthis point. Their work has also been available to other manufacturers, as well as to AES 24.

But what does this mean for AES 24? The comments made at the convention implied that these two companies have an "AES 24-like" system. Both Harmon and QSC are the first to admit that they are both using about 90% of the same data structure and realize they both need to move closer to each other for the last 10%. The work they have completed has raised a few logical issues with the current draft implementation of AES 24.

Having had the benefit of not being involved with AES 24, I think the industry can begin to see the light at the end of the tunnel.

The Ethernet standard

Ethernet was originally developed by Xerox, Digital and Intel in the early 1970s. The term Ethernet commonly refers to original Ethernet (now known as Ethernet II) and to the IEEE 802.3 standards. The two are generally compatible in terms of cables, connections and electronic devices, but Ethernet II and IEEE 802.3 standards are different enough to make them incompatible. Ethernet is generally used on networks with light to medium traffic (it supports a maximum throughput of 10Mbps) and performs best when a network's data traffic is sent in short bursts.

Software protocols

Software protocols are called frame types, and they define the basic structure of the data being transmitted. The frame type is the lowest level of software protocol and allows transmission of any type of digital data within its frame. There are presently four frame types in use: Ethernet 802.3, Ethernet 802.2, Ethernet SNAP and Ethernet II. Ethernet 802.3 is the original software protocol. It was released before the IEEE 802.3 standards were complete and did not fully comply with all its specified provisions.

To solve this problem, a second, more advanced protocol was released. The second protocol was Ethernet 802.2. Nearly all of today's networking software can use either frame type, but 802.2 is preferred for its advanced features. The other two frame types allow compatibility with older Ethernet II networks (Ethernet II frame type) or Apple Computers' Appletalk networks (Ethernet SNAP). To allow communication, the network server and clients must have the same frame types loaded. It is possible to have multiple frame types loaded on the same physical network simultaneously without causing any conflicts.

The next level of network software defines the actual packet type and structure transported within the Ethernet frame. At this level, the different network operating systems available for the Ethernet "backbone" become incompatible. Novell Netware network's packet type and structure are not compatible with a Microsoft NT or Windows for Workgroups packet type and structure. Some work has been done to allow noncompatible packets to be transmitted over and through a network's server. This is done by encapsulating the incompatible packet inside a standard compatible packet for transmission and routing. At the other end of its journey, the standard packet is stripped away, which allows the destination machine to receive the original data.

Hardware standards

Physically, an Ethernet network consists of a network interface card (NIC) in each device connected to the network. The cards can be connected with various types of cabling to carry the electrical signals. The NICs are transceivers that translate information from the attached machine into the proper electrical signal for transmission over the network cabling. The hundreds of different NICs all include features such as unique network address selection, support for up to three different interface connectors, cable systems and sockets for remote boot proms (this allows remote starting and configuration of devices).

Connected to the NICs are the cables that make up the "backbone" of the network. The Ethernet cabling comes in three common types. 10Base2 (Thin-Net) generally uses the NICs onboard transceiver to translate signals to and from the rest of the network. The cabling for the 10Base2 consists of RG58A/U or RG58C/U coaxial cable, 50Ω terminators and T-connectors that attach the BNC cable to the NIC. 10Base5 (Thick-Net) differs from 10Base2 in that it uses external transceivers that connect to the NIC with the Ethernet standard Digital Intel Xerox (DIX) interface connector. 10Base5 uses the same type of coaxial cable but allows significantly longer cable lengths for each segment.

10BaseT networks require concentrators (centralized wiring hubs) and are wired in a star topology with a concentrator at the center of each star. 10BaseT networks use shielded or unshielded twisted-pair cable and RJ-45 eight-pin telephone-type interface connectors within each "star." The concentrators are connected to the server with a single Thick-Net or Thin-Net coaxial cable. The allowable twisted-pair cable lengths are longer than Thin-Net but shorter than Thick-Net. This topology allows 10BaseT networks to be very large while being simpler to troubleshoot because of the ease of removing whole sections (single concentrators) from the network at once.

Technical terms

When discussing computer control of noncomputer systems, such as audio systems, the common thread underlying the different methods is the communication protocol. Machine communication takes many forms, and the computer industry has created numerous terms to define the concepts involved. The following is a list of some of the common terms.

- Current loop: In a current loop, data is represented by the flow of current through a circuit. Because any circuit requires a complete path for electron flow (a loop) to operate properly, this type of interface has been named a current loop. At this time,they have no formal standards. An informal standard is that 20mA of flowing current equals a logic 1 ("on") and that no current equals a logic 0 ("off"). Voltage is present on current loops, but the precise voltage does not matter as long as it does not exceed equipment maximums.
- Voltage loop: In a voltage loop, data are represented by a difference in potential (voltage) between two points in a circuit. Because any circuit requires a complete path for electron flow (a loop) to operate properly, this type of interface has been named a voltage loop. Small amounts of current are present in voltage loops, but the precise current does not matter as long as it does not exceed equipment maximums. All serial and parallel communications hardware standards defined by the Electronic Industries Association (EIA) are voltage loops, but each has a different voltage level associated with a logic 1 and a logic 0.
- Contact closures: The simplest form of machine communication, the contact closure is actually an on-off switch and can be used in either current- or voltage-loop systems. No timing signal or data structure are defined. The transmitter (switch operator) can be used on other machines, mechanical devices such as doors, or humans. No matter how the switch is controlled, it sends the same message -- a completed circuit -- when on and an open circuit when off.
- **Parallel data link:** In a parallel data link, each data bit is transmitted over its own wire. Numerous other signals, such as system clock and handshaking signals, also occupy their own wires. Parallel data links are fast and efficient but require a large number of wires. This factor becomes very expensive over long distances.
- **Serial data link:** In a serial data link, data bits are transmitted one after another over the same physical wires. In its simplest form, serial communication can

(Sidebar continued on page 28)

"Technical Terms" *(continued from page 26)*

be performed using only two wires. Some other signals, such as "request to send" and "clear to send," occupy their own wires in more sophisticated implementations. Serial data links are slower than parallel data links but use far fewer wires and are therefore cheaper over long distances.

- Single-ended serial link: A single-ended serial link connects only one transmitter to one receiver. In other words, only one device is at each end of the communications link (RS-232, RS-423).
- Broadcast: A broadcast serial link allows one transmitter to send messages to multiple receivers at the same time. In other words, one device is at the transmission end of the communications link, but multiple devices can be connected at the receiving end (RS-422).
- Multidrop: A multidrop serial link allows more than one transmitter and one receiver to share the same physical wires simultaneously without interfering with each other's messages (RS-485). Broadcast and single-ended capabilities are contained within these systems.
- Synchronous data link: Both the transmitter's and the receiver's system timing clocks are locked precisely in synchronization, and one bit (an on or off signal) is transmitted during each clock cycle. The easiest way to achieve this synchronization is by running a separate wire between the two systems to carry the clock pulses. This setup is fine when used on systems such as parallel data links where the extra wires needed do not effect the system, but it defeats one of the primary advantages of serial communications. It is possible for the receiver to derive its clock pulses from the incoming data stream of a serial data link if both the receiver and the transmitter agree on specific conditions of the transmission. This is how current serial hardware standards accomplish synchronous transmission because it alleviates the need for an extra wire. The drawback to this method is that both ends of the link must have a certain amount of sophistication, which can be costly and complicated. Few serial communications devices used today incorporate this technology.
- Asynchronous data link: In asynchronous communication, the receiver's clock starts and stops (re-synchs) itself at the beginning of every data byte, or word. To enable the receiver to re-sync itself, the transmitter adds a start bit (a 0 or off) at the beginning and a stop bit (a 1 or on) at the end of each byte. This ensures that there is a transition from a logic 1 to a logic 0 at the beginning of each byte (even if a byte is composed of all 1s or 0s). This scheme only works if both ends of the link are set to exactly the same parameters: bps, length of each data byte, start-bit enabled and correct number of stop bits. Anyone who has improperly set up an asynchronous data connection on a modem has seen the garbage characters that result. The start-stop bit method also allows the link to re-sync itself if the data is corrupted, because the receiver will not pass on data unless a valid stop bit is received.
- Full duplex: Communication occurs in both directions simultaneously over separate physical wires.
- Half duplex: Communication occurs in only one direction at a time. When transmission direction is changed, the transmitter and receiver must inform each other that they are reconfiguring by way of additional data bytes. This process causes half-duplex transmission to be even slower than half the speed of full-duplex transmission at the same bit rate.
- Data rate: Machine communication has no minimum data rate. The maximum data rate will depend on cable-length capacitance and any other factors that might affect transmission quality.

I remember the first time I saw the prototypes for programmability. I was amazed that I could set it, then walk away, and no one would be able to tweak my settings.

MicroAudio and ART. I remember the first time I saw the prototypes for programmability. I was amazed that I could set it, then walk away, and no one would be able to tweak my settings. The unit has no front-panel controls. The EQs don't just provide a way to set the EQ curve in one unit; the curve can be copied to the next unit. Different settings can also be called up to optimize settings depending on what is playing or the size of the crowd.

Yamaha's new digital mixing board is another product that will stir up the marketplace. The preset capability, combined with the ability to change an EQ setting or balance a mix with a MIDI, is bringing audio to the level of computer-controlled lighting boards of 10 years ago. With this board, a church could allow a study group to use the system with limited features for night meetings, yet the Sunday service audio person could still take full advantage of all of the features.

The many ways to skin a cat

Last spring, I had the good fortune to be a presenter at NSCA's yearly workshop on computer-controlled audio. The five major complete systems were available for the students to play with and use to complete the design of two projects. IED, Crown, Crest, Lone Wolf and Media Matrix all showed that they had the technology necessary to complete the project. The client would get a complete and working system in the end. The strengths and weaknesses of each system were brought out in the different design scenarios. They all got to the same end point by following different paths. One system might have a better user interface; another might provide all of the tools needed as long as there is enough memory. The total system price difference among systems designed by different groups changed depending on their approaches to solving the problem. All of the systems have some form of hardware connection and can get the project completed, but you must decide which completes it in the best manner.

So what does all of this mean? In many cases computer-controlled audio systems can make life a lot easier. Projects and clients suddenly have a choice for more flexibility than they did a few years ago. This also means that unless your paper trail, self discipline and project management is in excellent form, the project will take on a life of its own. Remember that software programmers are never finished with a project — there is always one more feature to add. If you are using computer-controlled systems for your projects, you are now a programmer.

Computer control of audio systems is not a magic elixir for system design. If you do not know the basics of the system you are designing or working on, the computer will not make the decisions for you. You will find that you will be dependent on the manufacturer's support teams. If you are going to undertake these projects, make sure that you have the data from the manufacturer. Just because something can be controlled through RS-232 does not mean that Company X's RS-232 software command structure is the same a Company Y's or Z's. Remember the goal — we are trying to make the end-users' life easier and simplify the control of the systems. S&VC