HARRIS

Radio Systems Products

Key To Digital Radio

Your Complete Guide To Current Digital Radio Broadcasting Options and Features

next level solutions

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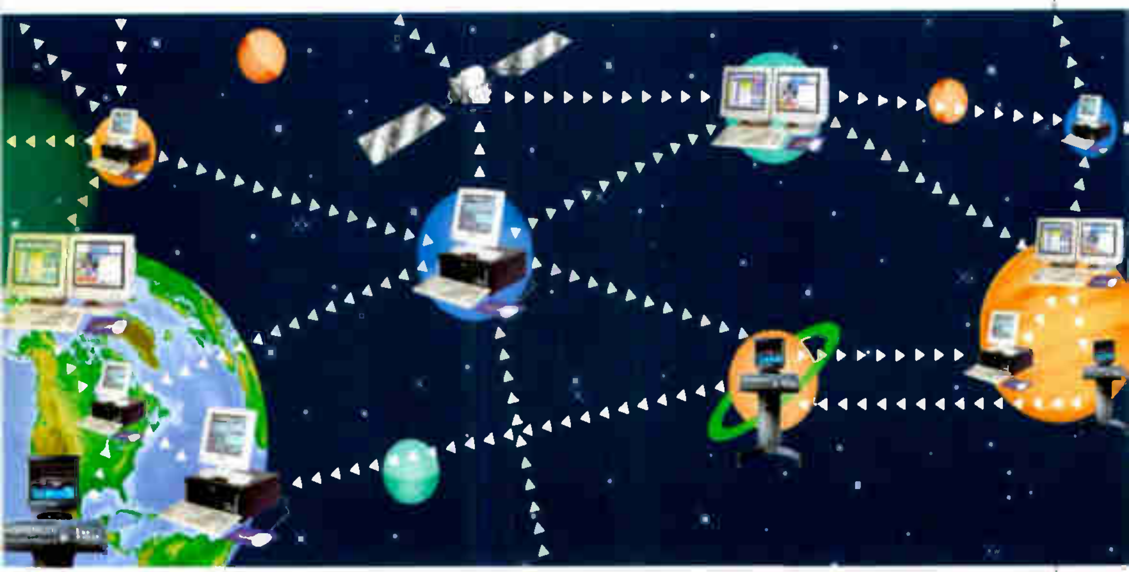

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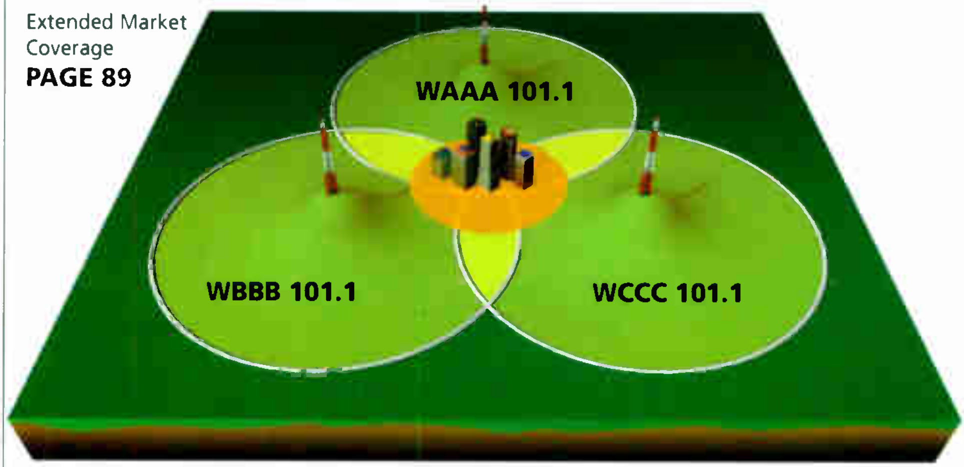


Introduction

The next few years present exciting and challenging times to radio broadcasters. Exciting, because the world of digital audio electronics and computer technology brings more options and flexibility than ever before, allowing radio stations choices of equipment and technologies that didn't exist even a couple of years ago. Challenging, because fundamental shifts in broadcast technology and the business plans they enable will demand that everyone in our industry have a plan to get from "the here" to "the there" of digital broadcasting as it becomes a reality.

Harris has prepared this *Key to Digital Radio* as a guide to the current state of digital radio broadcasting and to suggest options and strategies for the journey from the analog present to the digital future. We don't assume to know all the answers. Every broadcast facility has its own operational requirements and its own unique mix of existing equipment. And, as we all know, there are still regulatory issues to be resolved and, no doubt, technological and engineering advances waiting to be discovered.

We do know, however, that the journey is already well underway. Equipment and systems are available today that make it possible to install an all-digital broadcast chain from the production suite, through the on-air studio, to the STL or program transport system, all the way to the transmitter. Digital systems offer broadcasters tangible benefits today, even as they lay the groundwork for over-the-air digital broadcasts that will change the shape of radio in the opening years of the 21st century.

We hope you find this *Key to Digital Radio* a helpful guide as you sort through the options and decisions your station faces now and over the next few years.

James Woods

JAMES WOODS, VP, RADIO SYSTEMS
HARRIS BROADCAST COMMUNICATIONS DIV.

HARRIS

Why The Key?

Q & A

Why is Harris coming out now with *The Key to Digital Radio*?

JIM HAUPTSTUECK

Jim Hauptstueck DIGITAL PRODUCT LINE MANAGER

This guide is a tool for today's broadcasters, to assist them through the difficult decision-making processes in the new digital environment. It will help broadcasters expand efficiencies, capitalize on new capabilities and resources to help improve the bottom line now, and assure forward compatibility toward emerging technologies.

DON SPRAGG

Don Spragg AM PRODUCT LINE MANAGER

Harris wants to help broadcasters who may be confused about what digital is and how it relates to many products. We want to provide answers to the people who are trying to do the best in setting up their stations' operations.

DARYL BUECHTING

Daryl Buechting FM PRODUCT LINE MANAGER

It will be a great reference tool for broadcasters and engineers, a way for them to get the latest information they need to upgrade to digital equipment.

DAVE BURNS

Dave Burns STUDIO PRODUCT LINE MANAGER

The radio/audio broadband highway now has a map and supporting infrastructure. *The Key to Digital Radio* is this industry's technical Good Book.

Is there a void in the industry as to this kind of information?

Buechting I think there is, as far as the existence of manuals or reference materials designed specifically for broadcasters. There is a lot of information available from different places, but *The Key to Digital Radio* will be one reference for finding everything they need to know.

Will having this book make things easier for Radio stations?

Spragg Yes. There is so much information out there. Hopefully, our book is compiled in such a way as to provide a good reference that is useful.

What will be the main benefit of the book?

Spragg *The Key to Digital Radio* is one source from one company, Harris, that is able to supply total solutions.

Buechting Hopefully, it will allow Radio managers to avoid pitfalls that others have encountered by not knowing the latest information on the best way to upgrade.

Advantages of an Uncompressed Digital Studio-to-Transmitter Link

Digital electronics touches many aspects of our lives. In our home appliances, our office equipment, and in the cars we drive, digital technology has established new and higher standards of functionality, reliability, and value. Today, thanks to many recent advances in product development, the benefits of the digital world extend to broadcast stations. Broadcasters can now experience the many advantages that come with implementing an uncompressed all-digital path from the studio to the on-air signal.

Rapid growth in computing power has paved the way for a new generation of audio and broadcast equipment. Digital signal processing (DSP) uses software that runs on specialized, high-speed, digital microcomputers to simulate functions formerly performed by analog equipment. Microprocessors perform many functions that previously required separate analog components. DSP-based products can replace analog filters, mixers, modulators and other system components, and provide precision and added functions that were unattainable with analog technology.

The Harris Platinum ZD20CD is a high-power FM transmitter with solid-state reliability, performance and versatility. The Harris DIGIT™ CD digital FM exciter is supplied as standard equipment with all Platinum Z transmitters.

Defining Technical Standards

Some years ago, a working group of radio and studio broadcast equipment manufacturers defined data format standards for digital audio. The working group selected the serial data standard defined by the AES/EBU (Audio Engineering Society/European

Broadcast Union). Equipment that complies with this standard is typically described as AES/EBU or AES3 equipment. The AES/EBU standard supports the commonly used data rates of 48.0 kHz, 44.1 kHz, and 32.0 kHz. Because the current FM stereo transmission standard limits the frequency response of left and right channels to 15 kHz each, a 32 kHz data rate is often used. However, there are situations, particularly in the studio environment, where the higher 48.0 kHz data rate may be desired. Ideally, all interfacing equipment should have the ability to accept all three rates, although sample rate converters can be used where direct connections are impractical.

The Harris CD LINK™ allows transmission of uncompressed AES3 digital audio from studio to transmitter using standard 950 MHz STL frequencies.

Connectors on consumer-grade equipment designed to comply with the S/PDIF (Sony/Phillips Digital Interface) standard may appear to be compatible with the AES/EBU interface, but the data formats of the two standards are different. If consumer-grade audio sources, such as CD players and R-DAT record/play machines are used, data rate conversion and other special accommodations will be needed at the digital inputs to the mixing console.

Since the acceptance of the AES3 data interface standard by all major broadcast equipment manufacturers, a growing number of digital audio processing components have come into use. It is now possible to build an all-digital studio with an uncompressed all-digital studio-to-transmitter link (STL) using standard off-the-shelf equipment. The new generation of audio processing equipment accepts a digital audio input, processes this data fully in the digital domain, and outputs the processed data with no analog to digital (A/D) or digital to analog (D/A) conversions.

The Studio-to-Transmitter Link (STL)

Broadcasters can convey digital audio from their studio to their transmitter by leasing digital T1 telephone lines or installing fixed fiber-optic lines. Alternatively — to avoid fixed monthly costs or right of way issues of land lines, for example — stations can send audio to their transmitter by means of spread spectrum radio (see "Broadcast Applications of Spread Spectrum and T1 Technology" p. 65) or 950 MHz radio channels. Until recently, the narrow bandwidth (300 kHz) of the 950 MHz radio channels, which was adequate for the audio and technical standards of the all-analog world, was insufficient to handle the much greater bandwidth of the digital signals of newer-generation studio equipment. If stations wanted to use 950 MHz radio channels for their studio-to-transmitter link, they had to compress their high-bandwidth digital audio to fit the limited bandwidth of the STL.

Effects of Data Compression

There are two types of data compression schemes: lossless and lossy. Lossless data compression does not degrade the quality of the retrieved data. However, current lossless compression technologies used on personal computers, such as STACKER, DBLSPACE, or PKZIP, are not effective in reducing audio data. Therefore, virtually all data compression schemes used for reducing

audio data are of the lossy type. Lossy data compression schemes take advantage of perceptual masking and other psychoacoustic effects in human hearing perception to discard some of the audio information without being very noticeable to the listener. There are several lossy systems in broadcast use today that provide satisfactory results under certain conditions. The most sophisticated systems such as MUSICAM MPEG Level-II and PAK offer the least audio degradation.

While a single digital compression changes the digital

Broadcasters can now deliver high-quality uncompressed digital audio all the way from their studio source equipment, through their production and broadcast consoles, and on to their transmitter using standard off-the-shelf products.

The Harris DIGIT® CD digital FM exciter can connect directly to source audio from a digital studio-to-transmitter link, eliminating the need for analog-to-digital and digital-to-analog conversions, which add distortion, noise, and system cost.

information in a way that may not be obvious to most people, when a signal is compressed two or more times, listeners begin to notice reduced audio quality. For example, if you use digital compression for music storage and also have a studio-to-transmitter link that uses digital compression, many listeners will be able to hear compression artifacts (noise) in your signal.

Benefits of an Uncompressed STL

With the introduction of newer-generation digital equipment, it is now possible for broadcasters to implement a digital studio-to-transmitter link without the previously required lossy data compression. A fully digital STL can transport uncompressed AES3 audio data with the same "bit for bit" transparency of a T1 digital phone line — within the 300 kHz bandwidth of an existing 950 MHz STL channel. In addition to providing undegraded audio quality, newer digital STL equipment can deliver improved path reliability by adding up to 20 dB of fade margin to an existing STL path.

If any type of lossy data compression, such as APTx, DOLBY AC-2, MUSICAM, MPEG, or PAK is used in the studio-to-transmitter path, the digital audio processing device should be located at the transmitter site for direct connection to the digital exciter. Lossy data compression schemes introduce changes in the absolute peak levels produced at the output which can result in overmodulation of the transmitter. By locating the digital audio processing after the data compression and expansion, the audio peak levels can be tightly controlled before feeding this data to the digital exciter. An uncompressed digital RF STL or an uncompressed T1 digital phone line gives the flexibility of locating the digital audio processing at either the studio or transmitter end of the data link.

Digital Exciter

A digital exciter is the final link in the all-digital FM airchain. The digital exciter's input provides data rate conversion and transforms the left and right channel audio data from the AES3 serial data format into the digital equivalent of analog composite stereo baseband. This data is fed into the numerically controlled oscillator (NCO) of the FM modulator. Using Direct Digital synthesis (DDS) the output of the NCO is converted into a synthesized RF waveform.

Advantages of an Uncompressed Digital Path

There are many advantages to using an uncompressed all-digital path from your studio to your transmitter. Among these are:

- The elimination of all A/D and D/A conversions and the distortions they induce.
- Full digital quality delivered to the "on-air" signal without the noise and distortion build-up of an analog system, and without the artifacts or noise introduced by multiple digital compressions.
- Plug-and-play interfacing between equipment without worries about level adjustments or hum pickup.
- Absolute frequency response and amplitude matching between stereo channels.
- Absolute phase matching and differential phase stability between stereo channels.
- Absolute stability and repeatability, day-after-day, year-after-year, without adjustments.
- Better resistance to interference.
- Greatly improved fade margin for radio links.
- Half the cable population — one AES3 cable replaces two analog cables.
- Ability to transport some auxiliary control data along with audio data on one cable.

The Harris DRC 2000 digital audio console uses 32-bit internal precision floating point digital signal processing technology to deliver maximum audio quality.

Sample Rate Conversion

A matter of central importance to consider when designing digital systems and purchasing digital equipment is sample rate. An analog audio source, such as the output of a microphone, travels through cables as a continuous signal. In contrast to the analog signal's continuous wave, a digital signal is a stream of high and low voltage "bits" arranged in sequences that encode information. An analog signal is converted to a digital signal by sampling sequential slices of the continuous wave and then quantizing the result of each sample — that is, encoding the sample as a numeric representation of the wave's height at the point being sampled. If enough samples are taken, it is then possible to subsequently recreate the original analog wave and thereby recreate the original sound. The number of samples taken per second is called a sample rate. For example, audio on compact discs is sampled at 44.1 thousand times per second — thus, CD audio has a sample rate of 44.1 kHz.

A number of different sample rates are in common use. Hard disc audio systems operate at a sample rate of 32 kHz, digital audio tape is sampled at 48 kHz, and sound cards or other studio components may operate at yet other sample rates. The variety of sample rates isn't significant in the realm of analog broadcast systems — in which a station takes the analog output from various digital sources and keeps them in analog form through the rest of its broadcast chain. But as broadcasters make the transition to digital workstations, consoles, and studio-to-transmitter links, they need to make allowance for differing sample rates.

Digital audio consoles, such as the Harris DRC 2000, can accept digital audio at various sample rates and analog audio from many audio-source devices, and automatically convert it to a digital signal at a desired output sample rate.

It is desirable to keep digital audio input from sources such as CD or DAT entirely in the digital domain as it travels through the other digital studio and broadcast components. Digital equipment designed to operate at different sample rates cannot work together unless sample rates are first converted to a common standard. Once a station selects a single sample rate as a station-wide standard, it becomes easy to convert all audio to that standard rate. But, given the range of sample rates in use, what criteria can guide a station's decision in choosing a facility-wide standard?

A Bit of Theory

Although several different sample rates are in common use, these rates are not arbitrary. Sample rates are related by both physics and biology to a listener's hearing ability. The human ear can detect sound frequencies between approximately 20 Hz and 20 kHz (and most people hear little or nothing above 16 kHz). FM transmission standards limit the practical upper audio bandpass to a maximum of about 15 kHz. As a result, a station can deliver an audio signal that extends across most of a listener's hearing range. Attentive listeners may be able to perceive equipment and system enhancements that improve a station's ability to deliver high quality audio across this entire 15 kHz range. But while equipment specifications that exceed the range of the available 15 kHz audio channels are useful in the recording studio, these higher specifications can't further enhance the broadcast audio an FM station transmits to its listeners. In selecting a sample rate, then, a station needs a rate that will support the highest possible standard of broadcast audio. An even higher rate may be of value, but only if it helps a station meet needs, such as recording for archival storage or for non-broadcast audio production.

Digital audiotape equipment, such as the Tascam DA-P1 portable DAT recorder, supports multiple sample rates. Users can select among the standard sample rates — 48 kHz, 44.1 kHz, or 32 kHz.

An important bit of mathematics explains the relationship between sample rates and audio frequencies. According to the Nyquist theorem, a sample rate must be at least twice the highest analog frequency to enable accurate encoding (and later decoding) of the signal. In light of Nyquist, we can see that the 44.1 kHz sample rate used by the CD industry enables encoding of audio across the maximum 20 kHz human hearing range. In the broadcast environment, a sample rate of 32 kHz is sufficient to encode the signal at the highest level of quality attainable on the permitted 15 kHz audio channels. Many broadcast stations, therefore, use 32 kHz as a facility-wide standard, while audio recording studios typically use the higher sample rate of CD or DAT.

Once a station has chosen a standard, whether 32 kHz or higher, all digital audio used in the broadcast chain needs to be converted to the standard. The output of devices, such as CD or DAT, is passed through a sample rate converter to change it to the station standard. But even after all audio is converted, pops, clicks, or audio level changes may occur when switching from one audio source to another unless the signals are also synchronized to a common time reference.

The PR&E Integrity digital console has an ergonomic control surface less than 1.5 inches high. All 16 inputs can handle analog signals. Ten also accept digital inputs at any sample rate.

Synchronization

The information contained in a digital signal is in the form of a stream of digits grouped into digital "words." When a broadcaster switches from one digital audio source to another, there may be an audible glitch if the digital timing of the stream of "words" from the new source isn't exactly synchronized with timing of the signal it replaces. Adding a reference clock makes it possible to make audio transfers without audible pops.

Digital consoles provide inputs for many types of audio equipment and typically contain all of the circuitry necessary to convert sample rates automatically and to synchronize all audio to a common clock. For a small broadcast station, a single console with its own internal clock may be sufficient. Where a station uses several consoles, it may be possible to designate one console as a master (for clocking purposes) and to synchronize other consoles to the master's internal clock (for more information on consoles, see "Console Timing Issues" p. 29). However, even if all equipment in a studio is synchronized through the console, a facility with many studios can experience audio glitches when switching broadcast audio from one studio to another if the facility-wide audio routing isn't also synchronized.

Facility-wide routing can be made synchronous by installing a master clock and wiring all digital equipment in the facility's broadcast chain to use the master clock as its synchronization reference. Components not directly in the broadcast chain can remain asynchronous. Station personnel can do pre-broadcast work such as sampling and editing on equipment not referenced to the master clock and later bring the finished work into synchronization at broadcast time.

Digital Interfacing:

What are the Standards?

Connections and cables are critical parts of broadcast systems. Today, as stations replace older equipment with new digital components, interconnection standards are more important than ever. Cables and connectors that served reliably to carry analog signals can fall short of the technical specifications digital signals require.

The good news for station general managers is that installing new digital wiring can be considerably less expensive than installing an analog wiring system. Unlike analog wiring, which requires two cables to send left and right audio, digital wiring carries both channels on a single cable — meaning less cable and less labor is needed.

Over the past few years, a handful of standards has evolved to specify the architecture of digital signals and to define the characteristics of cables and connectors that can reliably deliver these signals. An understanding of the commonly-used digital standards is key to understanding specifications and product descriptions of digital audio equipment.

The AES3 Standard

The digital interconnection specification described by the Audio Engineering Society's AES3-1992 standard makes it possible to send two stereo channels on a single 110 ohm twisted-pair audio cable. Most professional digital audio equipment comes with AES3 connectors. The AES3 standard updates earlier standards described by the AES and the European Broadcasting Union (EBU) that are jointly known as AES/EBU. Although there are minor differences between the American and European standard and between the earlier and more recent standard, the standards are substantially similar.

In everyday use, people use the terms AES3 and AES/EBU interchangeably. In addition to describing connection and cable requirements, the AES3 standard defines the digital format of the information that is sent from one piece of equipment to another.

Because digital audio equipment manufacturers use commonly agreed data formats and connector standards, audio equipment, such as this Orban Audicy digital audio workstation, can send and receive audio programming that is compatible throughout a broadcaster's facility-wide audio distribution system.

Digital systems, such as Enco's $DAD_{PRO}32$ digital audio delivery system (dual monitor configuration shown), can be configured to share data via a dedicated file server and Local Area Network (LAN).

The S/PDIF Standard

Consumer-grade audio components often use the Sony Phillips Digital Interface (S/PDIF) standard. S/PDIF uses 75 ohm coaxial cable and RCA connectors. Although the S/PDIF interface may be physically compatible with connectors of some AES3 equipment, the data formats of the two standards are incompatible.

The AES3-ID Standard

AES3-ID is the professional equipment standard for delivery of digital audio over video-quality 75 ohm coaxial cables. Distances of up to 1,000 meters are possible, making AES3-ID especially useful where cable runs must exceed the 100-meter maximum recommended for AES3 twisted pair. Many broadcasters find AES3-ID isn't cost-effective for facility-wide use.

Because most audio equipment comes with 110 ohm AES3 connectors, an impedance transformer is typically needed between the equipment's connector and the AES3 coaxial cable. The additional cost can be considerable when you count the number of such connections in most facilities.

The AES3-IE Standard

AES3-IE is the standard for fiber-optic cable. Fiber-optic cable's high bandwidth makes it ideal for delivering large volumes of digital information without data compression. Because fiber-optic systems encode digital information as bursts of light rather than as electrical voltages, the signal is immune to electrical and RF interference. As with coaxial cable, the added cost of making equipment compatible with fiber-optic cable usually makes it more cost-effective for long runs than for facility-wide use.

STANDARD	IMPEDANCE	CONNECTOR	COMMENTS
AES/EBU	110 OHM	XLR	ORIGINAL STANDARD FOR 110 OHM TWISTED PAIR
AES3	110 OHM	RCA	LATEST STANDARD FOR 110 OHM TWISTED PAIR
S/P-DIF	75 OHM	RCA	SONY/PHILLIPS DIGITAL INTERFACE
AES3-ID	75 OHM	BNC	PROFESSIONAL STANDARD FOR 75 OHM COAX.
AES3-IE	N/A	OPTIC I/O	STANDARD FOR FIBER-OPTIC INTERFACE

Digital Source Equipment

Digital technology is reshaping all aspects of the broadcast world, but it is as audio source equipment that digital products first demonstrated their advantages. Beginning with the arrival of the compact disc, broadcasters have had a growing number of digital source devices from which to choose.

Digital players and recorders bring several advantages compared to their analog counterparts, including: extremely low-noise audio that resists degradation with repeated playing; compact storage media with high capacity; fast audio retrieval via high-speed random access; and a new world of computer-based editing and production possibilities. Following is a summary of the types of digital audio source equipment and the basic features of each.

Compact Disc (CD)

Compact disc was the first digital audio product commercially available. Audio CDs hit the consumer market in the early 1980s and quickly replaced vinyl as the medium of choice for commercial audio recordings. The features that made CDs popular with the "record" buying public also made them a hit in the broadcast studio. Because CDs are read by the light of a laser, rather than the physical contact of a stylus or tape head, audio quality doesn't deteriorate with repeated playing. A CD's digital audio is also free of the background hiss of cassette and reel-to-reel tape. Digital CDs, combined with digital editing and production equipment in the recording studio, can deliver clean and extremely accurate audio across a wide dynamic range. Another feature that makes CDs popular with broadcasters and consumers alike is the random-access ability of CD players that makes it possible to select a

The Harris CD2001 CD Cart machine is designed for reliable performance in the broadcast studio. A precision cast aluminum deck plate and heavy-duty printed circuit boards protect working parts and electronics from harm. The CD2001 is designed to select digital information down to the exact frame.

Panasonic's SV-4100 professional DAT recorder features on-board RAM for trim, rehearsal and instant start. The recorder is equipped with multiple industry standard digital I/O interfaces.

single cut easily or play several cuts out of sequence.

One drawback of CD, at least as compared with audio tape, is that it is a read-only medium. This has changed with the arrival of recordable CD-R. CD recorders combined with computer-friendly editing software now make it possible to use CD-R for studio-created or edited program material. However, CD-R remains less adaptable than tape because it can be recorded only once (and is, therefore, referred to as a WORM — write-once read-many — medium).

Digital Audio Tape (DAT)

Digital Audio Tape combines the high quality audio of the CD with tape's ability to be erased and re-recorded many times. DAT's adaptability makes it increasingly popular in the professional audio studio. DAT supports all the of the commonly-used sample rates — 32 kHz, 44.1 kHz, and 48 kHz. Unlike analog tape, DAT has built-in digital error-correction that greatly extends the tape's useful life by removing much of the noise introduced by wear, scratches, and dust. A DAT recorder player's high-speed tape transport mechanism is much faster than the typical forward/back controls of a standard cassette recorder, but, nevertheless, can't match the nearly instantaneous random access speed of a CD player.

Digital Source Equipment Comparisons

	PLAY/RECORD	RANDOM ACCESS	SAMPLE RATE	CAPACITY
CD	PLAY ONLY	YES	44.1 kHz	74 MINUTES
CD-R	WORM	YES	44.1 kHz	74 MINUTES
DAT	PLAY/RECORD	NO	32, 44.1, OR 48 kHz	VARIES WITH SAMPLE RATE
MD	PLAY/RECORD	YES	44.1 kHz	74 MINUTES (COMPRESSED)
HARD DISK	PLAY/RECORD	YES	32, 44.1, OR 48 kHz	DEPENDS ON SIZE AND NUMBER OF HARD DISKS

Minidisc (MD)

Minidisc combines some of the best features of compact discs and audio tape. Minidisc recording and playback technology use a magneto-optical (MO) process that employs both the laser optical technology of CD and the magnetic recording technology of analog tape. A Minidisc has the fast random access capability of CD, yet can be re-recorded again and again, just like tape. Minidiscs are housed in cartridges and, when combined with a minidisc cart player, can perform the same commercial and spot audio programming function as the familiar analog cart player. A minidisc holds the audio content of a full-size CD on a much smaller disc by using data compression. As discussed elsewhere in this *Key to Digital Radio*, a single data compression/decompression is not likely to be noticed by a listener, although multiple compression/decompressions in a station's airchain can degrade audio quality to detectible levels.

The Sony MDS-B3 minidisc recorder/player has high quality digital sound, instant non-linear access, and auto-cue making it ideal for both on-air applications and audio archiving.

Hard Disk Systems

Computer hard disk systems offer high speed random access, extremely high storage capacity, and limitless read/write capability. Hard disk storage costs more per minute of recorded audio than other media and is less easily transportable than CD, MD, or DAT. However, hard disk prices continue to drop, even as storage capacity grows. And some hard disk equipment can record and play via removable discs, such as Iomega's Zip™ disk.

360 Systems' DigiCart II Plus combines the features of three machines: a hard disk recorder, a mini audio workstation, and a digital cart machine. The DigiCart II Plus has space for two high-capacity internal hard disk drives and also provides low-cost removable storage on 100 MB Zip™ disks.

Things to Consider

When Buying a Hard Disk System

Thanks to the enormous demand for hard disk storage created by the personal computer industry, disk storage capacity has grown geometrically in the past few years while the cost of digital storage has plummeted. Equipment manufacturers now offer systems custom-designed, combining the high-volume storage and fast read-write access of computer hard-drives with user interfaces and software configured for the unique needs of the studio environment.

Hard disk systems come in a range of sizes and with an array of standard and optional features. No one system has all the answers for every facility. Fortunately, the systems in today's market assure that there is an appropriate selection and configuration to meet virtually any facility's needs. Here are five areas to consider when purchasing your system:

Enco Systems' DADPRO32 Digital Audio Delivery System provides powerful production recording, editing, on-air reproduction and management, and interfaces to scheduling and billing systems. It can be used for both live-assist and automated operations.

Your Objectives

Begin your selection process by considering what you want to accomplish at your facility. One facility may be looking for a system that can help implement broadcast automation. Another facility may be concerned primarily with storage and flexible access to music or other audio programming. Look at what you are doing now and how you may be doing things in the future. Even if your station has no plans to automate, you might decide that having automation capability would be a handy fallback to cover a staffing crisis — such as the flu taking out most of your airstaff.

Look for a system that meets your facility's functional requirements. You may find that the high channel capacity or networking ability of a new hard disk system can bring new cost-saving efficiencies to your operation, such as the ability to run multiple stations simultaneously from a single control room.

Most on-air systems accommodate basic back-to-back playing of commercials and music. But if you want to make that back-to-back airplay as seamless as possible, you may want a system with a segue editor that allows you to customize the transition from one event to another.

Look for ways the digital system can help your airstaff. There is a whole series of events that happens during a four or five hour air shift. A hard disk system can help your staff do everything — from automating timed recordings for news feeds and count-down programs, to having your staff meteorologist call in and record weathercasts via telephone DTMF tones.

360 Systems' Instant Replay uses a unique design to store and retrieve an audio library. Instant Replay stores up to 1,000 cuts on an internal hard disk and provides instant access via hot-key presets.

Work Flow

The second thing to consider is how a hard disk system will interface with your existing equipment and procedures. If you are buying or already have a digital editor, consider how it will interface with the hard disk system. Ideally, you will want direct file transfer between the editor and hard disk system. At minimum, you will need an AES/EBU port on both components so you can keep your audio digital as it travels between the two systems. Having to mix down to analog just to go back to digital undermines the enhanced audio quality you expect a digital system to bring to your station's presentation.

Many stations like to trade production back and forth. If this is important to you, make sure a digital system makes digital transfers easy for you by giving production directors the ability to exchange programming via the Internet. Your production director may also want the ability to import sound effects and imaging material easily from the Internet to the digital editing system.

Expandability

The third thing to consider is where you want to be two to three years from now. Knowing the ability for, and cost of, expandability should be major considerations. If you expect to run your digital system on a network, plan in advance for the likelihood of additional storage needs in the future as your network grows.

Hard Disk System Checklist

- ☐ ARE YOU LOOKING PRIMARILY FOR MEDIA STORAGE AND ACCESS CAPABILITY, ON-AIR BROADCAST AUTOMATION, OR BOTH?
- ☐ DOES THE SYSTEM HAVE SUFFICIENT CHANNEL CAPACITY FOR YOUR PRESENT AND EXPECTED OPERATIONAL NEEDS?
- ☐ CAN YOU IMPLEMENT DIRECT FILE TRANSFER BETWEEN YOUR DIGITAL EDITOR AND THE HARD DISK SYSTEM?
- ☐ DOES THE SYSTEM ALLOW FUTURE EXPANSION OF STORAGE CAPACITY?
- ☐ HOW WILL YOU BACK UP YOUR SYSTEM, AND HOW WILL YOU CUT OVER TO YOUR BACKUP IN THE EVENT OF SYSTEM FAILURE?

Don't overlook the Internet when assessing your future system capacity needs. Trade publications report projections of 50 to 60 percent of all radio stations having live audio on the Internet in the near future. Your Internet programming may be identical to your on-air broadcast, or it may have a different commercial load. Be sure to allow for both the additional storage capacity and channel capacity that may be necessary for an Internet presence.

Backup

The fourth thing to consider is how to insure against system failure. Regular system maintenance and faithfully observed backup procedures will assure that you are always on the air. There are many ways to back up your audio files, including redundant drives, mirrored drives, and "cold" storage on archived backup media. Discuss with your system vendor ways to implement a backup system that will work with your equipment and your facility's infrastructure.

Creative Potential

A digital on-air system can bring new efficiencies to your facility's many production and on-air activities. But just as important, it also opens the door to new creative possibilities for your production and on-air staff. Your airstaff will "buy in" to the advantages of learning to use a digital system creatively when they see it as a problem solver and not just another new thing to learn. Often, there are one or two people at a station who quickly grasp the potential of a digital system and find ways to use it as a creative tool. By giving these people an opportunity to lead the way in adapting new features and capabilities, a station can encourage other staff members to take creative advantage of the new tool.

Digital Editing

Digital audio editing tools for broadcast facilities range in functionality and price from software designed to run on standard personal computers to digital audio workstations (DAWs) with user interfaces custom designed for use by broadcasters. There are no "one size fits all" solutions in the DAW world. The goal is to look beyond superficial "features per dollar" comparisons and determine the long-term impact on your production department.

PC Software

At its least expensive, an editing system can consist of multi-track editing and mixing software installed on a personal computer system. Professional audio editing programs come in a range of prices, typically offering additional editing features and the ability to mix a greater number of audio tracks as price increases. Be sure the computer on which you plan to install any program you are considering exceeds the software's minimum system requirements. At today's low computer prices, equipping a computer with processor, memory, and storage that greatly exceed minimum requirements is money well spent. Pay particular attention to the computer's audio board. While editing software should run on most consumer-grade audio boards, you will need to install a professional quality board in your computer if you want maximum audio functionality and AES/EBU connectivity to your other studio systems.

The Orban Audicy digital audio workstation combines intuitive screen displays with an "analog feel" control board that quickly becomes a seamless extension of the operator's hands.

Digital Workstations

Digital audio workstations (DAWs) combine custom-written audio editing software with computer hardware and user interfaces specifically designed for audio editing. A workstation's user interface can include familiar controls, such as sliders, scrub wheels.

Audion's VoxPro digital sound editing system was specifically designed to replace reel-to-reel tape recorders on the air. The VoxPro system allows a user to record and edit phone bits, contests, sound effects and more — on the fly. An optional remote control panel makes one-hand operation possible.

and push buttons, which many users find more familiar and convenient than working with a computer keyboard and mouse.

An interface designed specifically for audio editing takes advantage of "muscle memory." Tasks you perform hundreds of times a day become second nature, as in driving a car or playing a musical instrument. While mouse-based systems are perfect for some people and can add editing capabilities to a computer that is also used for other functions, a pointing device, such as a mouse, has inherent limitations. A point-and-click interface requires careful visual attention to the position of a cursor on a screen and processes only single commands in sequence. In the audio studio, this can limit some people's ability to "mentally multitask." Think of a drummer playing a syncopated beat with all four limbs, watching bandmates for visual cues, and all the while listening to the tune. People who work in a fast-paced, deadline-driven production room how that drummer feels. An advanced interface design takes advantage of people's ability to devel-

Digital Audio Workstations

"A whole different ballpark" is how Frankie Blue, a production director for Clear Channel Communications stations XHRM and KJQY in San Diego, describes the versatility of the Orban Audicy digital audio workstation (DAW) he began using a year ago. "This is my first digital workstation, and it's not even conceivable that I would go back to working with analog now."

Blue, who admits to being "a dinosaur" when it comes to working with computers, was nevertheless using his new workstation with confidence within a couple of days. Learning to use the new machine was a "no brainer," says Blue because all the controls and symbols, such as play, record, and fast forward, were the same as those he was familiar with on analog reel-to-reel machines.

Clear Channel began using DAWs in San Diego in 1996. First to arrive were Orban DSE 7000s, followed by additional Orban Audicy stations. According to Clear Channel staff engineer Ron Foo, production editors first shared digital files by "sneakernet" transfers of removable disks. But as the numop "muscle memory" to make tasks performed hundreds of times become second nature.

System size and feature considerations depend on your facility's needs. A digital editor can be lightweight, portable, and able to work independently of a computer display screen, making it ideal for field work. On the other hand, an editing system can be a full-featured workstation designed to be a production department workhorse. You may find that your facility has appropriate uses for different editors in different situations. Make sure, however, that your workstations are able to communicate with your existing (or planned) hard disk system by direct file transfer or AES/EBU port.

Finally, when planning your digital editing purchase, give some thought to return on investment in addition to direct costs. All new equipment carries the indirect cost of user training. So look for features, such as compatibility with existing equipment and familiar user interface, that will quickly and smoothly turn your workstation into a productive asset. Consider both your purchase price and your costs over several years, including the cost of training and the potential gains from higher staff productivity.

ber of workstations increased, the convenience of a network became compelling.

At present, workstations are networked within each facility by means of standard Windows 95™ networking which, says Foo, is "very simple, but robust enough to handle what we want it to do." However, now that all 11 Clear Channel stations in the San Diego area are using digital workstations, plans are underway to switch to a Novell™ network that will make wide area file transfers possible among all facilities.

Says Foo, the change from analog to digital editing was like going from an electric typewriter to a click-and-drag word processor. "Our production people are creative to begin with, but they can be even more creative now. The workstations are so quick, it's easy for someone to say, 'I don't like the way this sounds, so let me do it one more time.'"

Console Timing Issues

Whether a facility consists only of a studio and a production room or is a multi-station, multi-studio broadcast center, the console is where it all comes together in the studio. A console's well-designed user interface makes it easy for on-air talent to select and switch among audio sources for their broadcast. And "under the hood," a console has to perform as reliably and as transparently to the operator as the engine does in your car as you drive to work, oblivious to spark plugs and pistons, engrossed, no doubt, in your station's drive-time program.

As the variety of source equipment proliferates in the studio and as stations migrate toward all-digital airchains, a console is more than an audio patching and mixing center. In the digital realm, the console plays traffic cop, preventing collisions of incompatible audio signals at the busy intersection where all your audio sources merge. The audio data from your various audio components — DAT, CD, hard disc — operates at different digital sample rates. A digital console receives inbound audio in any of the commonly used data rates and then converts it to a data rate that has been selected as a studio- or facility-wide standard. Many consoles can convert audio to any of the most commonly used data rates (32 kHz, 44.1 kHz, or 48 kHz) making them adaptable to many different studio and facility environments.

The Yamaha O2R digital recording console (shown at top) features an "open system" selection of interface cards that permit direct connection to a wide range of modular digital multitrack recorders, digital audio workstations, and other digital equipment.

The Harris DRC 1000 digital audio console (shown at bottom) allows each member of an airstaff to create and store a custom configuration. The console is designed to be expandable, so it can grow as a station makes additions and changes to its audio equipment.

The other important behind-the-scenes function a console performs is synchronization of individual data "words" by referencing all data to a common clock. In studios that are not tied to a facility-wide router and reference clock, the console's internal clock assures that all digital data words share a common time reference. And where there is a facility-wide time reference, the console and all other equipment can be linked to the common reference generator. Either way, on-air talent can concentrate on their broadcast, knowing they can make smooth transitions from one audio source to another without transitional audio "pops" or "clicks."

Console Features

Digital consoles come in sizes and with features to satisfy a wide range of studio requirements and station budgets. When shopping for a new console, consider which features will benefit your studio's present operations or will assist with plans for future growth.

The User Interface

Although a digital console is a computer at heart, most consoles have the familiar user interface of faders and buttons that your on-air talent already know from analog world. Familiarity minimizes the need for retraining and reduces the chance that talent will make embarrassing on-air mistakes as they wrestle with unfamiliar equipment. Digital consoles offer features such as LCD screens that can display various information, programmable user profiles that make it possible for each user to store a favorite console configuration, and software-based logic functions that aid in controlling source equipment directly from the console.

PR&E's AirWave digital broadcast console is designed to be easy to use, flexible, and cost effective. Input modules are user-reconfigurable from analog to digital. All digital inputs do sample rate conversion.

Growth Potential

As with most things, more money typically buys more capability. When choosing a new console, consider what you need at present and what you realistically expect to need in the foreseeable future. Select a model that has enough analog and digital inputs to handle your source equipment. Consider the number and types of output busses that fit your operational requirements. If you're planning a gradual transition to a digital airchain, you will need both analog and digital output busses. Other features, such as equalizers, compressor limiters, or automatic gain control, may be standard on some consoles and optional on others. Consider which functions are "need to have" as part of your console purchase and which functions may already be taken care of by other equipment at your facility. With the variety and flexibility of the consoles on today's market, it's possible to meet any broadcaster's needs and to assure maximize return on the purchase dollar.

Console Purchase Checklist

- ☐ Are there enough analog and digital inputs to work with existing source equipment?
- ☐ Does the console have extra capacity for anticipated future growth?
- ☐ Can the console convert all data rates in use in your studio?
- ☐ Are there sufficient output busses to meet present and expected needs?

Wiring Digital Audio

Digital audio is different from analog audio. And because digital is different, it stands to reason that cable that carries digital audio is also different from cable that carries analog audio. Before installing a digital system, you need to understand the basic types of digital cable, and the capabilities and limitations of the choices available to you.

It's a Question of Wavelength

Analog audio works at frequencies from 20 Hertz (Hz) to 20,000 Hertz (20 kHz). A Hertz, or cycle, describes one full wave of a frequency. The formula for wavelength tells us that at 20 kHz, one full wave is 15,000 meters long (about nine miles). If you have a cable that long carrying an audio signal at 20 kHz, then the impedance of the cable is important, and must match the impedance of the source and destination devices to which it is wired. But no broadcaster runs cables nine miles, so the impedance at analog frequencies is not important. In fact, most cable manufacturers don't even tell you what the impedance is, since it doesn't matter.

CABLE TYPE	PART NUMBER	CAPACITANCE	IMPEDANCE	WIRE GAUGE
ANALOG	BELDEN 8451	34 PF/FT.	UNSPECIFIED (ACTUALLY ±38 OHMS)	22 AWG
DIGITAL	BELDEN 1800A	12.5 PF/FT.	110 OHMS	±20% 24 AWG

The two key parameters in analog cable are resistance and capacitance. The lower the resistance, the farther you can go (lower attenuation). Resistance doesn't change the audio signal; it just makes it a bit weaker. Even smaller gage wire, such as 24 AWG, can go thousands of feet before the small gage begins to affect performance.

Capacitance, unlike resistance, is frequency-dependent. Capacitance tends to absorb high frequencies — the higher the frequency, the more the absorption. This effect, called "high frequency roll-off," is inherent in every cable design. By using a cable with

This article is adapted with author's permission from *Wire, Cable, and Fiber Optics for Video & Audio Engineers*, by Stephen H. Lampen, Technology Development Manager, Belden Wire & Cable Co. This book is available for purchase from Harris.

lower capacitance, you can go farther before roll-off becomes significant. A cable with a capacitance of 30 picofarads-per foot (pF/ft.) can carry a signal farther than a cable with a capacitance of 50 pF/ft. And a cable rated at 15 pF/ft. could run farther yet. The important thing to remember about capacitance is "the lower the better." An analog cable should always indicate its capacitance. Knowing the capacitance is one way you can tell good-performing cable from average cable, or at least which cable is suitable for long runs and which should only be used for short runs.

Digital Audio is Different

Digital audio is a digital representation of an analog signal. It follows guidelines established by the Audio Engineering Society (U.S.) and the European Broadcast Union (Europe), called the AES/EBU standard. (see "Digital Interfacing: What Are the Standards" p. 15) In a digital signal, ones and zeros represent the original analog signal. These ones and zeros take up much more room (bandwidth) to describe each wave.

A digital audio system operates at 3 MHz — a frequency 150 times higher than the 20 kHz of an analog system. Things are different at such a high frequency. For one thing, the wavelength at 3 MHz is only 100 meters (328 ft.). Many

BELDEN

TWISTED PAIR PART NO.	DESCRIPTION	NO. OF PAIRS	WIRE GAUGE AWG	NOM. CAP. PF/FT.*	NOM. IMP. OHMS
1800F	French Braid	1	24	13	110
1800A	Beldfoil aluminum-polyester shield	1	24	13	110
1801A	Beldfoil aluminum-polyester shield, non-conduit plenum	1	24	13	110
1802A	Beldfoil aluminum-polyester shield	2	24	13	110
1803A	Beldfoil aluminum-polyester shield	4	24	13	110
1805A	Beldfoil aluminum-polyester shield	8	24	13	110
1806A	Beldfoil aluminum-polyester shield	12	24	13	110

COAX PART NO.	DESCRIPTION	NOMINAL OD (INCHES)	NOM. CAP PF/FT.*	NOM. IMP OHMS
1865A	Sub-Miniature, RG-59/U type	.150	16.5	75
1505A	RG-59/U type	.235	16.3	75
1506A	RG-59/U type	.199	16.1	75
1694A	Low-loss serial digital, RG-6/U type	.275	16.2	75
1695A	Non-conduit plenum, RG-6/U type	.234	16.2	75

MEDIATWIST PART NO.	DESCRIPTION	NO. OF PAIRS	WIRE GAUGE AWG	MAX. CAP UNBAL.PF/100M	NOM. IMP OHMS
1872A	Non-Plenum	4	24	49	100±12
1874A	Plenum	4	24	49	100±12

TWISTED PAIR PART NO.	DESCRIPTION	CANARE NO. OF PAIRS	WIRE GAUGE AWG	NOM. CAP. PF/FT.*	NOM. IMP. OHMS
1806A	Beldfoil aluminum-polyester shield	12	24	13	110
DA206	Large O.D. for long runs, shielded	2	20	14.6	110
DA202	25 AWG conductors allow use with punch down block audio patchbays	2	25		110
DA202-P	Multi-channel version of DA202	4	25		110
DA202-AT	Foil shield with drain wire	2	25		110
D403-AT	Foil shield with drain wire, for MIDI routing	4	22		64
D202-4P	Multi-conductor, braided shield	8	25	16.8	92
D202-7P	Multi-conductor, braided shield	14	25	16.8	92
D202-12P	Multi-conductor, braided shield	24	25	16.8	92
D202-18P	Multi-conductor, braided shield	36	25	16.8	92
D202-25P	Multi-conductor, braided shield	50	25	16.8	92

*CAPACITANCE BETWEEN CONDUCTORS

facilities are likely to have cable runs of this length and longer, which makes the effects of impedance an important consideration when wiring digital audio systems. The AES/EBU standard specifies impedance as 110 ohms, which means that the sending and receiving devices and the cable itself must all operate at 110 ohms. However, the standard allows a tolerance of ± 20 percent. Thus, you could operate without problems with an impedance as low as 88 ohms or as high as 132 ohms. But even at 88 ohms, the impedance is far above the 30 to 40 ohm impedance of typical analog cable.

Also, the capacitance required for a digital cable (12.5 pF/ft) is lower than capacitance of a typical analog cable (34 pF/ft). Remember, capacitance tends to absorb higher frequencies, and the lower the capacitance, the longer the maximum possible cable length. Frequency absorption can be a problem even in the 20 kHz range of analog. At 3 MHz range of a digital signal, absorption is an even greater problem. In the world of digital audio, low capacitance is not just a good thing: it is essential if the system is to operate at its full potential.

Connectors are often the same for both analog and digital systems, so you might be tempted to use your old cable with new digital equipment. But now you realize that the analog cable's impedance is wrong and the capacitance is too high for digital signals.

What About Using Digital Cables for Analog?

Digital cable will work with both your digital and analog equipment. Like analog cables, digital AES/EBU cable is twisted pair and usually foil shielded. And because the capacitance of a digital cable is one-third the value of analog cable, digital cable may actually improve performance of analog equipment. If you are rewiring or if you are wiring a new facility, it makes sense to wire up for digital now. You will get fantastic analog performance and be digital-ready. You will be able to use the wiring interchangeably: you can hook up a cable to an analog device today, and a digital device tomorrow.

What About Coax for Digital Audio?

There is no technical difficulty in using coaxial cable for digital audio wiring — in fact, the American Engineering Society has established a standard (AES3-ID) for digital coax. Because the structure of coax cable is much more precise than twisted pair, and the impedance variations are lower, coax will support longer cable runs than twisted pair. However, if you don't need unusually long cable runs, coax may not bring you any added benefit and could add considerable cost to your installation. Most digital audio equipment comes with connectors designed for use with balanced 110 ohm AES/EBU wiring. You will have to buy baluns for all such connectors to adapt them for use with unbalanced 75 ohm coaxial cable. You will need two baluns (one at the sending device, the other at the receiving device) for every cable you install. Of course, if you buy a digital audio device and it has a coax output, you won't have to buy a balun to use it with coax. (Although you will need a balun to use it with twisted pair.)

Coax doesn't have the common-mode noise rejection of bal-

GEPCO INTERNATIONAL

TWISTED PAIR PART NO.	DESCRIPTION	NO. OF PAIRS	WIRE GAGE AWG	NOM. CAP. PF/FT.*	NOM. IMP. OHMS
552404EZ		4	24	13	110
552408EZ	Alum.-polyester shield	8	24	13	110
552412EZ	Alum.-polyester shield	12	24	13	110
D5524EZ	Alum.-polyester shield, dual-pair Zip	2	24	13	110
552604GFC	Alum.-polyester shield, extra-flexible	4	26	14	110
552608GFC	Alum.-polyester shield, extra-flexible	8	26	14	110
5524EZ	Alum.-polyester shield	1	24	13	110
5524SD	Alum.-polyester shield, solid conductor	1	24	13	110
5526EZ	Alum.-polyester shield, thin profile	1	26	14	110
5526	98% coverage 36 AWG OFBC, spiral serve shield	1	26	12	110
552M	95% coverage 38 AWG TC braid	1	22	15	95
5524TS	Alum.-polyester shield, plenum	1	24	13	110

*CAPACITANCE BETWEEN CONDUCTORS

anced twisted pair and, therefore, can more easily pick up unwanted electrical or RF noise. On the plus side, you can use video patch panels (which are impedance specific) and video distribution amplifiers, which don't exist in the twisted pair AES/EBU world.

One place where coax makes sense is in an installation that has both video and digital audio because you can use the same cable for both applications. Using one cable, one connector, one stripping tool, one crimping tool can make installation and maintenance really easy. However, if you're sending only audio and your cable runs are typical of lengths at most facilities, coax will bring you no functional improvement. On the other hand, there is no question that it will meet the technical requirements of digital audio.

Gepco 5522M

There are many digital audio installations today that are entirely coax cable and they work just fine.

S/PDIF

There's another kind of coax-based digital audio called S/PDIF (Sony/Phillips Digital Interface). It can use the same coax as AES3-ID, but the cable has RCA connectors. S/PDIF is a standard for consumer-grade systems and is not fully compatible with AES3-ID. While music data is convertible, most of the supporting data is not. Moreover, S/PDIF is intended for only short-run systems.

Be cautious when you see the word "coax" in product literature. Check to see if it is specified as S/PDIF rather than as compliant with the AES3-ID standard. If necessary, you can convert S/PDIF to AES3-ID and vice-versa, but conversion will require purchase of active devices to reconfigure the data stream. These devices are even more expensive than baluns.

A Word About Plenum

A plenum is usually a dropped ceiling or raised floor that is used for the return air in a heating ventilation or air-conditioning system. A plenum is a convenient out-of-sight space in which to run cables. But you can't install just any cable in a plenum. Plenum-rated cable is made with special materials that will not support a fire. Whether you need to use plenum-rated cables, regular cable, or cable installed in conduit is not simply a matter of choice. The local fire code, the fire marshal, building inspector, or similar local authority has the ultimate say when it comes to installing cable in a plenum. Before installing cable, be sure you know what those authorities require.

Belden MediaTwist?

What if there were a cable which could support all of the signals you wanted to use? What if there were one cable which could support telephone, FAX, modem, Switched 56, ISDN, T1, analog audio, analog video, digital video, machine control, broadband/CATV, 10baseT networks, 100baseT networks, gigabit Ethernet and, of course, digital audio?

Well, such a cable exists, and it's unshielded. The cable, called MediaTwist, is made by Belden Wire & Cable Co. It is the first cable to be application nonspecific. MediaTwist is covered by a dozen patents, but the key to the cable's performance is how well the twisted pairs are made. MediaTwist often outperforms old-style shielded pairs. Improved quality and consistency of the twisted pair construction takes the place of the shielding needed to prevent noise getting into (or coming out of) poorly constructed pairs.

MediaTwist is the first (and so far the only) twisted pair cable to be tested and pass the FCC certification for Class B Digital Devices. This is a test intended for digital coax. And yet, MediaTwist is four-pair unshielded twisted pairs (UTP) and takes standard connectors like RJ-45s.

You can even combine signals (say two channels of digital audio and RS-422 machine control) on one cable. You can't go as far in some cases (such as broadband/CATV or serial digital video) as you can on a coax cable made for that application, but you gain system adaptability. The key is that the jack on the wall can be a telephone today, a 100baseT network tomorrow, and digital audio the day after that. Try that with any other cable!

The downsides to MediaTwist are that it is only an install cable. It uses solid conductors, and is not flexible enough to be mic cable or rugged enough to go on the road. It works only with balanced line signals. Professional analog audio and digital audio come in balanced formats, so they can go directly into a MediaTwist pair. Unbalanced signal, such as S/P-DIF or video, require something to convert them to 100 ohm balanced format, such as an appropriate balun. These baluns are made by a number of companies. The largest number specifically for MediaTwist are made by ETS in Menlo Park CA (800-752-8208).

Most communities (but not all) follow the National Electrical Code (NEC) which is a voluntary guideline. If your community does follow that code, a copy of the NEC guidebook can be very helpful to you in making sure your plant gets the seal of approval from local authorities.

Digital Routing and Connecting Products

Just as in the analog world, a facility-wide digital audio system requires a variety of connectivity, patching, and switching components. Many such digital components, while different electronically, are similar in function to their analog counterparts. A digital distribution system also includes equipment in addition to devices found in an analog system — such as analog-to-digital (A/D) and digital-to-analog (D/A) converters, clock generators, or other components that accomplish tasks unique to the digital environment.

Here is a guide to some of the most common types of routing and connecting products. The variety and number of such products needed are different at virtually every facility, depending on routing and distribution requirements and the capabilities of other major system components (such as a digital console with built-in clocking or A/D conversion).

The Harris DA 1600.B analog distribution amplifier features four inputs and 16 outputs. A level meter for each input assists users in setting optimum operating levels. Each of the 16 outputs has a front-panel volume adjustment potentiometer.

A/D and D/A Converters

Given the reality that most stations will make a gradual transition to digital equipment over a number of years, making existing analog equipment work seamlessly with new digital components is essential. Analog-to-digital and digital-to-analog converters are the bridge between the digital world of ones and zeros and the world of the analog sine wave. Because different digital components operate at different standard sample rates, it is important that the converter be able to match the sample rate of the digital component to which it is connected. Converters can be purchased with either 110 ohm or 75 ohm connectors (or both).

Impedance Converters

A common connectivity problem at many facilities is making 110 ohm and 75 ohm equipment or cables work together. There are a variety of in-line and rack-mounted impedance converters designed to match balanced 110 ohm and unbalanced 75 ohm system components.

The Harris HDDA digital audio distribution amplifier has one input and six outputs. It is available in both 110 ohm and 75 ohm configurations. The HDDA features cable equalization, data reclocking, and signal error detection.

Digital Audio Switches

Digital Audio Switches allow selection among multiple digital audio inputs for connection to a common output. Selection can be made by front panel controls, and on some models, by means of a remotely located control panel.

Digital Audio Monitors

Digital Audio Monitors allow technical staff to monitor the level and quality of audio signal at key points in the audio distribution system. Monitors may include headphone or speaker connections, in addition to indicator lights and audio level meters. Some monitors include features, such as phase monitoring, user-selectable over-range threshold settings, and alarm systems to alert operators to audio problems.

Distribution Amplifiers

As in analog signals, long cable runs or signal splitting in a digital distribution system decreases signal strength. Digital distribution amplifiers boost a weak signal back to specified levels. Because the signal is digital, the amplified signal is an exact copy of the original, without the added noise that is typical when a weak analog signal is amplified. Distribution amplifiers also re-clock digital signals and perform error detection. When an error is detected, the distribution amplifier can trigger an external alarm.

The Harris HDAS digital switch may be used as a pre-selector or a source selector for any type of AES/EBU equipment. Users can select input via illuminating front-panel pushbuttons or from an optional remotely located control panel.

Routers

Routers make it possible to control "many-to-many" equipment and signal path decisions without the need to reroute cables or re-plug patch panels physically. Routing systems can be configured to control both digital and analog audio and can be purchased with different routing capacities. Some routing systems are designed to be expandable by means of adding cards to available slots in the router chassis or combining several routers into a single system.

Clock Generators

AES3 digital signaling consists of a serial stream of data frames or "words." When switching from one audio source to another, there will be audible glitches if the starting points of the digital words in both streams are not perfectly synchronized. A broadcast facility can avoid audible pops and clicks when switching audio by installing a master clock and configuring all audio equipment to use the master clock as a common time reference.

Frames, such as the Harris FR880 MB, are designed to fit easily into a rack for use in housing several different types of cards.

Controllability of Routing

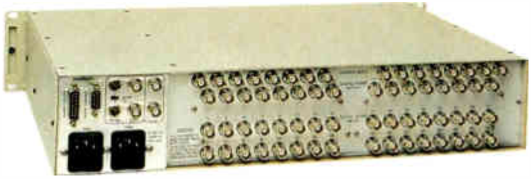
Routing a digital system is very much like routing an analog system. You have to create a system that can connect many audio components in different and flexible configurations, and you then need a way to select routing changes. When a system is very small, as in a home stereo system, it's reasonably easy to replug wires physically from one component to another when you want to change your system configuration. But as the number of components and the number of possible routing configurations increase, it quickly becomes impractical to reroute wires around a room or an entire facility. For years, the way to tame what would otherwise be an unmanageable jumble of wires has been to route equipment through patch panels.

Sierra Automated Systems' SAS 64000 audio routing system is a high-performance, microprocessor-controlled audio switching system designed for large-scale operations in professional broadcast facilities. It can route up to 256 inputs and 256 outputs, and can be configured to operate as a mono, stereo, or mixed mono/stereo system.

Patch Panels

Patch panels make order out of chaos by bringing audio from many different sources and routes to many possible destinations through a central location. An operator can quickly and easily reconfigure routing by simply re-plugging jumpers in carefully labeled, rack-mounted panels.

Digital routing systems can also use patch panels. However, unlike an analog system, in a digital system, the act of inserting a patch plug to make a new audio connection will cause a momentary audible click or pop in the audio as the digital AES3 signal re-clocks to the start of the next 32-bit data word. Depending on programming format or the number of times it's necessary to repatch equipment, one station may find an occasional audible glitch acceptable, while another might find it to be an major annoyance. See "Console Timing Issues" p. 29 for additional information of data clocking.



The Harris VIA32 digital router (top photo) controls up to 32 inputs and outputs. Users can create multi-level systems by combining with other routers.

Harris XPress series stereo audio routing switchers (bottom photo) provide monitor switching for analog stereo audio signals. Four outputs are provided so that distribution amplifiers are not required at the output of the switcher.

Routers

With digital signaling, broadcasters have another choice in routing hardware: digital routers. A digital router performs the same function as a patch panel. It allows an operator to patch many possible sources to many possible outputs. A router can be relatively small, permitting connection of a handful of devices to several possible audio routes. Or a large routing system can permit the patching of hundreds of sources with hundreds of outputs — the equivalent of a large rack of patch panels. Unlike a patch panel, however, the router is controlled by software. An operator at a central location can make routing selections without having to go to the location of a patch panel and reconfigure jumper selections. Some router software can also be programmed to schedule and automate routine switching events.

A routing system can be either asynchronous or synchronous. An asynchronous system gives a facility software-driven routing control. But like a patch panel, an asynchronous router will introduce an audible glitch as the new AES3 audio source enters the broadcast audio chain. And as with a patch panel, some stations may find this audible glitch acceptable and others may not. In a synchronous routing system, all equipment in the audio chain is synchronized to a common reference clock generator. Because digital signals from all components in the system share a common time reference, when you switch audio sources, the first data word from the new source is exactly synchronized to begin after the last digital word of the former audio source. Thus, the new digital signal flows perfectly into the audio stream without an audible glitch.

POTS vs. ISDN

When you are planning a remote broadcast, you have two telephone alternatives available to link your remote site to your studio. You can use either POTS (Plain Old Telephone System) or ISDN (Integrated Services Digital Network) in conjunction with an appropriate codec ("coder decoder"). There are advantages and disadvantages to each approach.

POTS codecs give you the adaptability to establish a remote site in a hurry — virtually anywhere a standard phone line is available. There are some performance tradeoffs, however, when you compare POTS to ISDN. While POTS codecs can often deliver audio comparable to an ISDN connection, the quality of the POTS telephone line can sometimes affect performance. Outside influences on telephone lines, such as noise and crosstalk, can cause the modems in POTS codecs to fail during a broadcast, producing several seconds of muted audio while the modems reestablish the link. Also, make sure you have a landline POTS connection available at your remote site — a POTS codec won't work with a cellular phone because cellular connections do not provide a sufficient synchronous data throughput.

For fixed studio locations, ISDN is the best choice for program delivery. And for remote broadcasts where you will need to be on the air solid for hours or where there is a lot of money riding on the success of your broadcast, make ISDN your first choice if at all possible. If ISDN is available to your remote site, can be installed in time, and isn't cost-prohibitive, it will give you the highest possible assurance of a consistent, high-quality audio link. On the other hand, for a spur-of-the moment road trip, a POTS codec can be a valuable asset that will get you on the air quickly and easily.

Comrex's HotLine POTS codec combines a high-speed modem and a digital audio codec to deliver up to 10 kHz audio over a standard dial-up telephone line. The Comrex Nexus ISDN codec delivers up to 15 kHz bidirectional audio on an ISDN telephone line.

ISDN POTS

ADVANTAGES

ISDN:
- BEST QUALITY AUDIO
- CONSISTENT PERFORMANCE

POTS:
- FLEXIBLE, EASY TO USE
- LESS EXPENSIVE PHONE SERVICE

DISADVANTAGES

ISDN:
- MORE COMPLICATED TO SET UP
- MORE EXPENSIVE PHONE SERVICE

POTS:
- MAY NOT BE AVAILABLE EVERYWHERE
- PERFORMANCE DEPENDS ON LINE QUALITY

Off-Premise Extensions

A station may want to establish an audio link between a studio and an off-premise site for a number of reasons. A broadcaster who needs to connect two (or several) facilities for daily or frequent exchange of live or recorded audio will want the security and reliability of an owned or leased dedicated connection. For most stations, this will mean leasing a T1 circuit (although if proximity and right-of-way issues permit, a fiber optic or microwave link might also be an option). When the need is to report breaking news, to cover a sports event, or to establish a temporary link for whatever reason, using the existing telephone facilities at the remote site can be a practical way to connect to the studio.

A station looking to establish a temporary link can connect through the telephone lines that are already present at the remote site by using a POTS (Plain Old Telephone System) codec. However, to avoid communication problems, it is important first to understand a bit about the off-premise site's telephone system before plugging in a codec. A broadcaster who wants to establish a permanent connection between facilities needs to understand the basics of ordering and implementing a T1 connection. So, whether looking for a temporary or permanent audio connection, it is useful to know a bit about PBX (private branch exchange) telephone systems and T1 circuits.

The Harris/Intraplex STL PLUS, when equipped with the OPX (Off-Premise Extension) option package lets you connect a telephone at the transmitter site to the PBX at the studio, just as if it were a local extension.

The Operation of the PBX

There are two sections to the modern PBX: the trunk side (outside lines) and the station side (inside lines). The PBX makes station-to-trunk connections, station-to-station connections, and in some cases, trunk-to-trunk connections, depending on the nature of the call.

The options available on any PBX, as well as the procedures for programming it, will vary depending on the manufacturer.

Tips for Using a POTS Codec

• BRING A STANDARD ANALOG TELEPHONE WITH YOU AND USE IT TO TEST A PHONE LINE BEFORE HOOKING UP YOUR CODEC. IF YOU GET A DIAL TONE AND CAN DIAL OUT, YOUR POTS CODEC SHOULD WORK.

• USE AN UNINTERRUPTIBLE POWER SUPPLY (UPS) ON YOUR AC FEED AT REMOTES TO PROVIDE SURGE PROTECTION AND GIVE YOU A BATTERY BACKUP.

• POTS (AND ISDN) CODECS ARE COMPLETELY TWO-WAY, SO USE INTERRUPTIBLE FOLD BACK (IFB) IN THE RETURN FEED TO REMOTES. A PUSH-BUTTON CONTROLLED RELAY CAN INSERT THE STUDIO OPERATOR'S VOICE INTO THE STUDIO CODEC RETURN AUDIO, ALLOWING COMMUNICATIONS WITH THE REMOTE SITE WITHOUT THE NEED OF AN EXTRA PHONE LINE.

On most systems, a person makes a station-to-station (internal call) by lifting the receiver and dialing a three- or four-digit extension. To make an outside call, you must typically first dial a "9," which signals the PBX to select an available outside trunk line. The call then routes to the local telephone company central office (C.O.), which in turn routes the call to its destination.

However, if many of your communications are between two fixed locations — your Chicago and your Springfield facilities, for example — you quickly reach a point where a leased T1 circuit becomes much more cost-effective than many toll calls routed through the telephone C.O.

With a PBX at each end, you can set up an off-premise extension (OPX), a type of circuit that enables two or more facilities connected by a T1 circuit to make telephone connections as if everyone were at the same location. A user in your Chicago facility can pick up a phone, dial a three- or four- digit extension, and connect to someone in your Springfield facility as if they were just down the hall. Because you are already leasing the T1, there is no additional telephone charge for what would otherwise be a long-distance call.

In addition to reducing telephone expenses, a T1 brings benefits of particular interest to broadcasters. A broadcaster who is already leasing a number of analog lines to send radio programming will be able to combine the traffic of many existing phone lines, reducing monthly telephone costs and gaining improved audio quality as well. Computers can also network via the T1 to exchange accounting, air-time billing, and e-mail between facilities.

Some PBXs can also handle trunk-to-trunk connections, making it possible to use your T1 and the local telephone lines at your remote facility to bypass long-distance telephone charges. Users dial a distinctive digit, such as "8," to tell the PBX to select the T1, rather than the telephone central office. The T1 carries your call to the remote site and then the PBX at the remote site patches the call to a local trunk line and sends it through the telephone system as a local call.

Using a Codec with a PBX

A POTS codec is a handy tool for establishing an audio hookup from a remote site back to your studio. A POTS codec contains both a codec (coder/decoder) and a modem. Using standard telephone jacks, you can easily hook up your codec, dial your studio, and have a broadcast-quality audio connection.

But before grabbing a codec and hitting the road for a live broadcast, plan for the possibility — the probability — that the remote site will have a PBX phone system and that the PBX will be incompatible with your codec. Digital PBX systems can present problems for a codec's high-speed modem. Also, many electronic phone systems have voltages and currents that are very different from those on a telephone company line. Some can even damage the modem in a POTS codec (or a laptop computer). You can't assume that it's safe to plug in a modem just because a phone system has a modular telephone jack.

As a rule of thumb, it is best to avoid using POTS codecs over in-house phone systems. Fortunately, these days, there is rarely a problem finding lines at most facilities that bypass the PBX. Fax lines and dial-up computer modem lines are typically routed past the PBX to a telephone company analog line. If possible, try to borrow the use of a modem or fax line, or ask if a member of the facility's technical staff can jumper an analog line for you directly from their telephone switch room.

Ordering a T1 Circuit for Broadcast Use

- Before calling your telephone company to order a T1 line, you need to gather information to describe your requirements.
- If the person you reach at the telephone company is not familiar with T1, ask to speak to someone who handles digital data circuits. Don't give up; this may take a few calls.
- When you reach the right person, tell them you want a quote for T1 service between two locations. Be prepared to give them the address and telephone exchange for each site.
- T1 circuits are available with two types of framing, ESF and SF (SF is also called D4), and two types of line coding, B8Zs and AMI. Specify that, if possible, you want a circuit using ESF framing and B8ZS line coding. Many multiplexers can run on any T1 circuit, but those using ESF and B8ZS are preferable.
- Ask the salesperson whether CSUs (Channel Service Units) are required for connection to the T1 line. They usually are, in which case you will need two CSUs, one for each end of the line. The phone company may provide these; if not, you can purchase them from your equipment vendor.
- You should expect to find a one-time installation charge and a monthly service charge in the quotation. The monthly charge is based on providing service to the two ends of the circuit, plus a mileage charge between the two sites.
- In many areas, you can get competitive bids from alternate carriers, such as MCI and Sprint, as well as your local phone company.

Digital T1 STL For My Station?

While there are many reasons to consider moving to a digital T1 STL for your radio station, most stations that make the move are motivated by either necessity or cost.

Necessity: "I'm losing line-of-sight to my tower — what are my options?"

To survive in today's radio market, radio stations must be able to adapt to constantly changing conditions. Sometimes, these changes mean that a station no longer has a direct line of sight to its transmitter. This can happen when a station moves its studios or when a new building goes up that obstructs its view of the transmitting tower.

Without line of sight, a microwave radio STL won't work. One alternative is to lease one or more high-fidelity analog telephone lines. A much better alternative is to lease digital T1 lines. Used with T1 multiplexers, T1 digital circuits provide higher fidelity audio than equalized analog lines, can carry multiple channels within a single circuit, and, in many cases, cost less than analog solutions.

Cost: "I'm leasing two (or three, or more) analog lines now. Can T1 save me money?"

Many stations have a need for multiple channels between the studio and the transmitter site. These channels might include:

- Dual STLs, when one transmitter site serves both AM and FM transmissions
- Satellite feeds from the transmitter site back to the studio
- Remote Pick-Ups (RPU) from the transmitter site back to the studio
- Off-premise extension phone lines at the transmitter site
- Control lines and other data channels

A 900 MHz radio link can't handle this variety of channels, and a single T1 line is more economical than leasing multiple analog lines in these situations.

The Harris/Intraplex STL PLUS is a fully integrated STL system designed to transmit broadcast-quality stereo audio over T1 lines with no digital compression.

Leased Line Alternatives — Analog vs. Digital

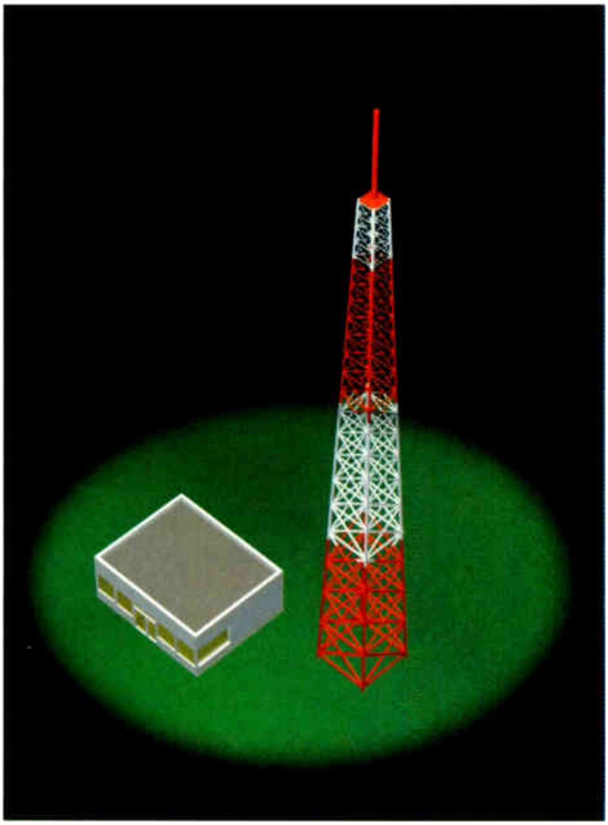
Local telephone companies have traditionally provided analog services for use by radio stations. While analog services may still be available, there are many reasons today to seek a digital alternative:

- Telephone companies are moving toward an all-digital network and no longer promote their analog offerings.
- As fewer technicians are trained in analog system maintenance, prices for these circuits are under constant pressure to increase.
- Digital circuits offer better transmission accuracy than analog circuits. Over distance, analog signals must be amplified, and noise is amplified along with the audio signal. Digital circuits are not amplified but regenerated, eliminating any noise picked up during the previous stage.

Also, when noise is present on an analog carrier signal, it directly affects the signal-to-noise ratio (SNR) of the audio signal being carried. But when noise is present on a digital line, there is virtually no effect unless the noise level becomes so great as to disrupt the signal completely. The result: the highest level of SNR under all circumstances.

Digital circuits available from the telephone company include T1, 56 kilobit digital data service (DDS), and, in some areas, fractional T1. DDS service does not have sufficient bandwidth to transmit quality audio, even with today's compression algorithms. T1 and fractional T1 can both be used for full fidelity 15 kHz stereo, and full T1 can do this without the need for digital compression. ISDN digital lines are now available in most areas. ISDN can be an excellent choice for remote broadcasts, but ISDN's per-minute costs and smaller bandwidth than T1 makes it impractical for use as a station's primary STL.

There are several ways to link a broadcast facility to its transmitter. A T1 digital circuit delivers high-quality, two-way audio, and will work even where a station lacks the line-of-sight path needed for a microwave link or 950 MHz link.



T1 for Radio Broadcasters

A growing number of radio stations use T1 today for audio transmission. T1 technology is important to radio stations for the following reasons:

- T1 is digital, providing a cleaner signal than analog and compatibility with new digital broadcast equipment.
- T1 is available everywhere and can operate over any distance.
- T1 is more cost-effective than analog, especially when carrying capacities are compared. T1 prices continue to drop while analog prices rise.
- T1 is reliable. The average annual downtime of a T1 circuit is negligible and constantly improving.
- T1 is flexible. Any information can be combined and transmitted over the T1 circuit.
- T1 is full duplex (transmits and receives in both directions).

For the radio station, this means one T1 line can be used for STL, TSL, RPU, satellite downhaul, voice channels, and equipment monitoring circuits.

Duopolies and LMAs

Many radio stations are combining to improve cost effectiveness, and T1 is an important part of the economics of duopolies and local marketing agreements (LMAs). Many operations connect two radio stations with a T1 line and generate programming services from a single site, saving thousands of dollars in personnel and operating costs.

Radio stations that are joined by common ownership or other business relationships can share live or archived programming via leased T1 circuits.

Using ISDN for Backing Up Your STL

Broadcasters frequently want the security of backup circuits on their critical audio paths, such as STLs, satellite downlinks, and network feeds. But backup can be expensive — setting up redundant audio links can double the cost of transmission systems. An ISDN codec offers a simple, straightforward way to back up studio-to-transmitter links and other critical audio paths at minimal cost.

Dial-up ISDN circuits provide an ideal alternative for critical path backup. They are relatively inexpensive to install, readily available in most areas, and once installed, carry only a low monthly charge plus billing for actual usage. Used in conjunction with ISDN codecs, a loss-of-signal sensor, and an audio switch, the ISDN line can assure that your broadcast will remain on the air even if you lose your primary studio-to-transmitter link.

The Harris/Intraplex OutBack (bottom) ISDN codec and the Silence Monitor (top) made by Broadcast Tools can be combined to create a reliable backup for a station's STL. An absence of audio will close a relay in the Silence Monitor and switch the audio feed to a backup link via the OutBack codec and ISDN line.

Implementing Automatic STL Backup With ISDN

Backing up an STL requires two codecs, one at the studio and one at the transmitter site. At the studio, connect the codec to a source of program audio and to the ISDN line, and press the network connect button (or similar function provided by the codec's manufacturer). This codec will remain in standby mode, permanently ready to accept an incoming call from its counterpart at the transmitter.

Digital transmission circuits are inherently more reliable than analog. However, occasional transmission errors ("bit errors") are unavoidable. The Harris/Intraplex STL PLUS uses digital signal processing to detect and correct bit errors as they occur.

At the transmitter site, connect the codec to the ISDN line and program a call profile with the ISDN phone number and audio parameters of the codec at the studio. Connect the audio outputs from both the codec and the primary STL to an audio switcher with two stereo inputs and one stereo output (such as the Broadcast Tools Model 2x1 or equivalent). To detect primary STL failure automatically, use a loss-of-audio monitor (such as the Broadcast Tools Silence Monitor or equivalent). Alternatively, you may use STL alarm contacts or squelch contacts.

Once your backup equipment is installed, failure of the STL will send a switch closure to the codec at the transmitter site, causing it to dial a call to the studio and bring in replacement audio. The audio switcher then routes the replacement audio from the codec to the airchain.

Direct File Transfer

The ability to manipulate, edit, and transport audio programming as digital files can greatly enhance efficiency and productivity in a broadcast organization. Once audio content is created or imported into the digital system, there is real advantage in keeping it digital throughout the rest of the production and broadcast process. Producers can edit programming on digital audio workstations (DAWs), store audio content on digital hard disk systems, and transfer program content to other digital media, such as DAT or minidisc, as needed to meet a station's operational requirements.

A key to getting the maximum efficiency from a digital audio system is to assure that direct file transfer is possible between DAWs and hard disk systems. Any digital workstation will be able to communicate with any manufacturer's hard disk system by means of an AES3 connection, but AES3 is not the most efficient way to move audio files from one place to another. An AES3 digital signal is a stream of digital words running in real time. Transporting five minutes of programming takes five minutes via an AES3 connection.

Direct file transfer is different. In a direct file transfer, you transport the entire digital file without opening it — just as you would send a file from one PC to another on a local area network. Although a file might contain five minutes of audio, you can send the file at the fastest rate the data connection permits, reducing the transfer time to perhaps a minute or less. As with other computer files, you are not limited to transfers between devices at your own location. You can move files between remote facilities via a wide area computer network, or you can use the Internet to send and receive files anywhere in the world. Data formats are often not compatible between DAWs and hard disk systems made by different manufacturers. So check with manufacturers and confirm that all equipment uses compatible direct file transfer formats before making a purchase.

Using direct file transfer, production staff can send audio files to workstations across the room, across town, or across the world.

Benefits of an All-Digital Path

Broadcasters have added a growing number of digital components to their editing rooms and on-air studios in recent years. Today, there are also several ways to create a digital studio-to-transmitter link (STL), making it possible for a station to implement a digital path from the source equipment in the studio all the way to the transmitter site. An all-digital airchain makes future over-the-air digital broadcasts possible. Equally important, it can deliver superior noise rejection and error correction capabilities compared with an analog system. Many broadcasters will realize cost savings by replacing leased telephone lines with a digital RF link or consolidating several analog telephone lines onto a single digital telephone connection.

The various types of digital studio-to-transmitter links are discussed in detail elsewhere in this *Key to Digital Radio*. The types of links include:

• **Digital 950 MHz Radio Links:**

Until recently, the narrow bandwidth of 950 MHz radio channels required broadcasters to compress digital audio, a practice which can add noise to a signal if it is compressed and decompressed several times. Now, thanks to newer generation equipment, it is possible to send digital audio uncompressed via an RF link.

Digital Signal Processor (DSP) synthesis of stereo in the Harris DIGIT® CD digital FM exciter provides perfect stereo, without the signal degradation resulting from the amplitude and phase limitations of analog baseband.

• **T1 Telephone Circuits:**

Because T1 has high bandwidth and can carry signal simultaneously in both directions (full-duplex), it can carry uncompressed digital audio and still have additional capacity for other signaling, such as control lines, satellite feeds, and remote pickups. Although broadcasters using T1 will incur monthly lease charges, it is an ideal solution where lack of a line-of-sight path makes an RF STL impossible.

The Harris CD LINK™ provides an economical way to implement an uncompressed AES3 digital studio-to-transmitter link using standard 300 kHz RF channels on the 950 MHz STL band.

• **Spread Spectrum RF:**

Like a 950 MHz link, a spread spectrum STL can connect a studio and its transmitter without need for fixed land lines. Spread spectrum delivers the bandwidth and full-duplex operation of a T1 link over a line-of-sight path at distances up to 30 miles. Spread spectrum systems for broadcast use operate in the 2.4 GHz frequency range and do not need an FCC license.

• **Fiber-optic Cable:**

A fiber-optic link is a possibility for stations that have right-of-way for the cable run between their studio and their transmitter. In some areas, it may be possible to lease a fiber-optic link from a telephone or cable company. For long runs, fiber-optic is typically much more expensive than an RF link.

Discrete vs. Composite Studio-to-Transmitter Links

Today, many broadcasters are moving to digital studio-to-transmitter links for improved sound quality — a digital STL can sound better than even the best analog system. Broadcasters are also motivated by the operational benefits and programming flexibility of a completely digital airchain from their studio to their transmitter.

When upgrading from an analog STL, a station must first select among three available digital technologies: 900 MHz radio links, spread-spectrum radio links, and T1 digital line links. For many stations, lack of a line-of-sight path to the transmitter or lack of available frequencies makes T1 the logical choice. Others may be motivated by cost considerations. A single leased T1 line can replace multiple analog lines at lower cost.

The Harris Aurora™ spread-spectrum digital radio offers T1 wireless service for line-of-sight distances up to 30 miles. Aurora radios operate in the 2.4 and 5.8 GHz bands.

But whatever a broadcaster's reasons for selecting T1, before installing a system, it will be necessary to make one further choice. There are two ways to adapt T1 for use as a broadcast STL. Digital T1 STL systems can be designed either to transmit discrete left- and right-channel stereo, or to transmit a composite stereo signal.

In a discrete transmission system, the stereo generator is located at the transmitter site, while a composite transmission system places the stereo generator at the studio.

There are sometimes good reasons for having the stereo generator in the studio. For example, many manufacturers combine an audio signal processor and stereo generator in the same chassis, and the broadcast engineer may wish to make frequent adjustments to the audio processor.

However, a composite digital STL system can introduce compromises in four important areas:

- Dynamic Range and Signal-to-Noise Ratio
- System Headroom
- Stereo Separation
- Channel Capacity

Let's take a look at why this happens.

Dynamic Range and Signal-to-Noise Ratio

Analog-to-digital conversion requires a sampling frequency at least twice as high as the analog bandwidth. Digitizing a radio program signal whose highest frequency is 15 kHz requires at least 30,000 samples per second. In actual practice, 15 kHz audio is generally sampled 32,000 times per second. And to maintain CD-quality audio, each sample must be at least 16 bits.

Using 16-bit coding, a composite stereo signal would have to run at 1.792 Mbps. The problem is T1 circuits run at 1.54 Mbps, with a payload capacity of 1.536 Mbps, so it cannot transport a 1.792 Mbps signal. To get composite stereo onto a T1 line, the sampling resolution must be reduced from 16 bits to 13 bits or less, which results in a worse signal-to-noise ratio and a decrease in dynamic range.

System Headroom

Digital audio encoders have a limited amount of headroom on their analog audio inputs. Any signals above the maximum headroom level must be sharply limited to prevent overmodulation and severe distortion.

On a composite system, when both channels have high peaks at the same moment (a common occurrence), the signal needs more limiting than either channel of the same audio on a discrete system. The result is an even further decrease (of up to 6 dB) in the effective dynamic range of the composite system compared to a discrete system.

Harris CD LINK™ sends uncompressed digital audio over conventional 950 MHz STL frequencies. Because the audio is uncompressed, stations have the flexibility of locating their audio processing at either the studio or transmitter end of the data link.

Stereo Separation

No audio transmission system has an absolutely flat frequency response. The variation over the analog audio bandwidth is typically something like ±0.5 dB, meaning that the transmission system could create as much as a ±1.0 dB error in relative levels between the lower frequencies and the higher frequencies.

On a broadcast audio signal sent to the transmitter using either an analog STL or a discrete digital STL, one decibel of difference is barely perceptible to the listener. In fact, a decibel is, by definition, the smallest difference in loudness that the human ear can detect.

The Harris/Intraplex STL PLUS connects directly to a T1 line, making an uncompressed studio-to-transmitter link possible even where lack of a line-of-sight path prevents use of an RF or spread-spectrum link.

However, when the analog audio is converted to composite stereo before digital encoding, even one decibel of variation between these components on the STL link makes a big difference in stereo separation when they are assembled at the listener's radio receiver. The reason lies in the way the receiver uses these two signals to reconstitute the original left and right channels. Discrete transmission has no such potential effect on stereo separation, because the stereo generator is located at the transmitter, after the digital STL link.

Channel Capacity

One of the advantages to a broadcaster in using a T1 digital STL is its ability to carry several signals simultaneously. Subsidiary communications channels, transmitter remote controls, data, and voice can all be combined with the broadcast audio signal for transmission on a single T1 circuit. And the duplex nature of T1 lets you use the same system in two directions for both studio-to-transmitter and transmitter-to-studio links.

The payload portion of a T1 circuit consists of 24 64 kbps channels called time slots, for a total of 1.536 Mbps (24 x 64 = 1536). The discrete STL approach using 16-bit linear coding produces a 1.024 Mbps signal. Even after adding some overhead for error mitigation, the discrete stereo occupies only three-quarters of the payload capacity of the T1 circuit. This leaves one-fourth of the T1, or six time slots, available for other STL or bidirectional channels.

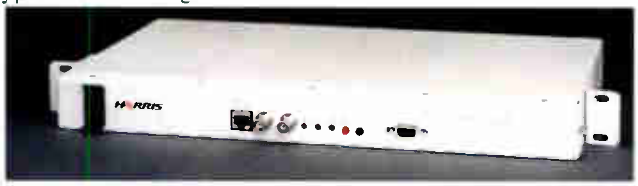
As noted above, a T1 can't carry composite stereo at a full 16-bit sample rate. Even when sample rates are reduced (with consequent audio degradation), a composite system will still have less bandwidth available for other uses than a discrete system. If sample size in a composite system is reduced to 13 bits, the T1 will have only one time slot available for other uses. Cutting the sample size to 12 bits will free a total of three time slots for other uses — still only half the number of slots available on a discrete stereo system.

The bottom line is that by choosing a composite stereo T1 system, a station can locate its stereo generator at the studio. For operational requirements or for reasons of compatibility with existing equipment, this may be the right choice for some stations. For many stations, however, the higher audio quality and the additional channel capacity of a discrete stereo T1 make it the ideal choice.

Broadcast Applications of Spread Spectrum and T1 Technology

Spread spectrum technology, originally used by the military, is now used in many commercial applications such as point-to-point wireless communications. When operating under Part 47 Section 15.247 of the FCC Rules and Regulations, licensing is not required, making implementation of spread spectrum systems simple and cost-effective.

AM and FM broadcasters have used wired T1 STL/TSL systems since 1980. Many broadcasters choose T1 over a 950 MHz STL because T1 is bidirectional and can transport larger amounts of data including uncompressed AES3 audio. Spread spectrum technology can deliver the benefits of a T1 connection via a wireless radio link in the 2.4GHz range, making it an attractive option for broadcasters who want to bypass T1 toll charges or who are in areas where the 950 MHz band is congested.



Harris Aurora™ spread-spectrum radio operates in the industrial, scientific, and medical (ISM) radio service. Aurora can be used for a point-to-point STL without the need for frequency coordination or FCC licensing.

Regulatory Issues

Before 1985, spread spectrum use was limited almost entirely to the military and the industrial, scientific and medical (ISM) services. In 1985, the FCC modified the rules to permit commercial use of spread spectrum techniques. Today, spread spectrum is used for personal communication systems (PCS), personal communication networks (PCN), wide area networks (WAN), local area networks (LAN), and unlicensed point-to-point radio links.

Regulations vary widely from one country to another. FCC rules govern the use of 2.4 GHz spread spectrum in the United States. If implementing a spread spectrum system outside the United States, it is important to investigate the local regulations related to 2.4 GHz radio emissions.

Enco's DAD_{JR} 32 Digital Audio Delivery System can be configured for either stand-alone use or can be networked with other DAD systems via a local area or wide area network.

Spread Spectrum Primer

Spread spectrum differs from familiar types of modulation such as AM or FM. As the name suggests, spread spectrum spreads data and power density over a wide bandwidth and then de-spreads it in a spread spectrum receiver.

Spread spectrum systems must satisfy at least two technical specifications. First, the total occupied bandwidth must be wider than the bandwidth of the information being transmitted. Second, some type of coding data must be added to the information data. Many spread spectrum systems use special pseudo-random noise (PN) codes to spread the transmitted signal. The length of these codes will ultimately determine the final bandwidth of the signal. Commercial spread spectrum systems use bandwidths 10 to 100 times the bandwidth of the information being delivered. Military systems often use spectrum bandwidths from 1,000 to 1,000,000 times the information bandwidth.

A spread spectrum receiver de-spreads the received signal using a locally generated replica PN code and a receiver correlator to retrieve only the desired coded information from the signal.

Advantages of Spread Spectrum

- Spread spectrum techniques have a proven track record as reliable and secure methods of transporting digital data in the U.S. military for the past 50+ years.
- Spread spectrum radios operating in the 2.4 GHz band can implement dedicated point-to-point links of up to 30 miles line of sight and do not require FCC licensing.
- Spread spectrum radios can transport T1 data and are compatible with current T1 STL/TSL systems.
- Spread spectrum radios coupled with T1-based STL/TSL systems can replace T1 line leases or can back up to an existing T1 system.
- Spread spectrum radios are capable of other dedicated connections, such as LAN/WAN, or temporary installations, such as remote broadcasts, where line of sight or near line of site links are possible.

The correlator filters out unwanted noise or interference. Most commercial spread spectrum radios have several different code sequences to select from to ensure that transmitters will only reach receivers using an identical code sequence.

A spread spectrum system's ability to reject or resist interference and jamming is one of its attractive features. Spread spectrum's low-power density aids in preventing interference with narrow-band radio systems. Also, the despreading process in spread spectrum receivers rejects interference from narrow-band signals.

By networking digital audio workstations, such as the Orban Audicy, broadcasters can distribute workloads and share audio files among many facilities.

Combining Spread Spectrum and T1

Many broadcasters use T1 phone lines for STL/TSL links. If a line-of-sight path (up to 30 miles) is available, these T1 based STLs may be coupled to 2.4 GHz spread spectrum radios. Many stations may be able to realize cost savings by installing a spread spectrum system. Depending on hardware and configuration requirements, a station may see a return on investment of three to five years by substituting a spread spectrum system for a leased T1 link.

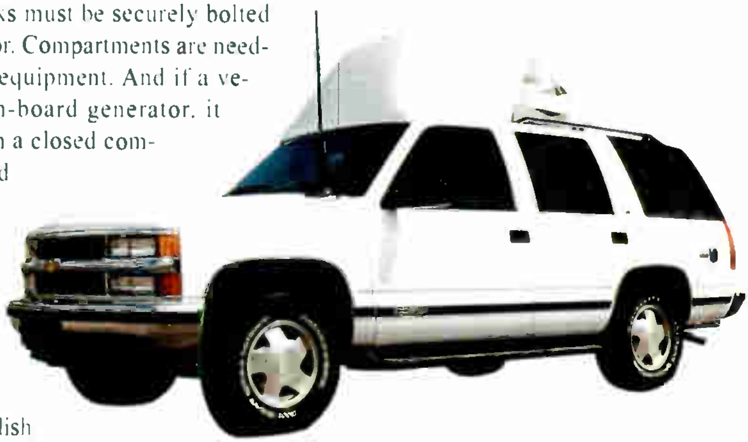
Another application of 2.4 GHz spread spectrum radios is for dedicated LAN/WAN connections between two intercity studio sites or offices. A few stations are using this configuration to allow the transfer of hard disk-based digital audio data, as well as traffic, billing and payroll information between co-owned stations. Spread spectrum is ideal for this application due to the secure nature of the signal. It is very difficult to intercept spread spectrum due to the wide bandwidth, low RF power level, and coding techniques. It is also possible to use spread spectrum for remote broadcasts, where a line-of-sight path or near-line-of-sight path is available, opening new opportunities for remote broadcasting.

Digital Remote Trucks

If your station does frequent remote digital broadcasts for promotional events, news reporting, or sports events, a remote truck will assure the best possible production and broadcast reliability and may save hours of set-up time with every broadcast. Vehicles equipped with RF and telephone STL systems, and ISDN and POTS codecs can establish a link back to your station in almost any situation.

A remote truck is more than a vehicle to haul equipment to a broadcast site. Configuring a remote truck involves careful attention to equipment requirements, floor-plan layout, power and electronic demands, and vehicle mechanical requirements, such as gross vehicle weight, suspension system capacity, and the load placed on the engine's cooling system. The Harris design and integration team has many years of experience in custom building remote broadcast trucks to meet broadcasters' unique requirements for remote links, and electronic and satellite news gathering.

From the outset, a remote truck needs to be designed with safety in mind. Equipment mounting techniques that are fine in a studio can be deadly in a traffic accident or sudden stop. All equipment has to be mounted securely in racks, and equipment racks must be securely bolted to the vehicle's floor. Compartments are needed for non-racked equipment. And if a vehicle carries an on-board generator, it must be installed in a closed compartment and vented to the outside in such a way that generator exhaust fumes cannot enter the vehicle. A truck with an extendable antenna mast or satellite dish



The Harris S-15 is based on a medium-size, four-wheel drive vehicle, such as a Chevrolet Suburban or GMC Yukon. The S-15 is equipped with a digital MPEG-2 DVB exciter and receiver with 15 Mb/s capability, a 40-watt solid-state power amplifier, and a carbon-fiber antenna system.

will need stabilizing jacks to secure the truck when the antenna is extended and an ignition lock-out system to prevent moving the vehicle until after the mast or antenna has been retracted and secured.

By working closely with an experienced design team, a station can create a digital remote truck with exactly the equipment it needs — installed in virtually any size or make of vehicle.

Processing in the Digital Airchain

For years, radio stations have used audio processors to shape the sound of their on-air signal. Audio processors allow stations to filter unwanted noise and to manipulate dynamic range, amplitude, and audio delay to create a sound that suits their broadcast format and appeals to their listeners.

Designed for stations on a tight budget, the Orban OPTIMOD-FM 2200 is based on the pioneering digital audio processing technology of the OPTIMOD-FM 8200.

The emergence of the digital airchain eliminates some quality-limiting elements that often prevented stations from transmitting the best possible audio. For example, compared to analog RF technology, the new digital studio-to-transmitter links (STLs) are much less prone to noise and distortion. Nevertheless, to obtain the best results from a digital airchain, stations need to avoid several pitfalls.

Orban's OPTIMOD-FM 8200 digital audio processor includes factory presets and controls for user-defined settings.

The first pitfall to look out for is overshoot. It is desirable in a digital audio processor to produce a very tightly peak-controlled signal at the AES/EBU outputs. When accurately passed to the digital exciter, the resulting signal will have no overshoot and will produce maximum loudness. If the link between the processor and the exciter does introduce overshoot, the exciter's over-modulation protection limiter must remove it. Since overuse of the exciter's protection limiter will degrade signal quality, STL-induced overshoots should be avoided.

Orban's OPTIMOD-FM 9200 digital audio processor for AM can automatically switch presets in sync with dayparts or special events.

A digital STL can be either compressed or uncompressed. Compressed STLs always introduce overshoot, which if not removed by the exciter's protection limiter, will result in a significant loss of on-air loudness. If any type of "lossy" data compression is used in the STL, the audio processor should be located at the transmitter site for direct connection to the digital exciter.

Eventide's DSP4000B stereo signal processor features hundreds of unique sound environments instantly ready for air or production use. Effects include reverbs, pitch change, time compression and expansion, and more.

New generation digital STLs use advanced modulation schemes to transmit uncompressed audio. Digital STLs will pass the tightly peak-controlled output of an audio processor directly to the FM exciter for maximum on-air loudness. However, if you are locating a digital or analog audio processor before an uncompressed 32 kHz STL, the processor should be band limited to 15 kHz to prevent overshoots in the STL. Otherwise, the audio processor should be located at the transmitter site.

Another potential pitfall is headroom in the analog-to-digital converter. Any clipping in your A/D conversion will cause distortion that can be greatly exaggerated by the downstream processing in the audio processor. Proper set-up of your processor can help you avoid this pitfall.

The Yamaha Rev 500 stereo digital reverb processing system features 32-bit digital signal processing coupled with 20-bit AD/DA converters. The Rev 500 includes 100 preset programs and 100 user programs with organized libraries.

The audio processor is a key component in a digital airchain. A high-quality processor with proper attention to set-up will give a station a louder, cleaner, and more competitive on-air signal.

A Word About Analog Audio Processors

Most people in the radio industry agree that digital electronics has brought higher standards of quality and operational flexibility to audio production and broadcasting. From the computer-based editing capabilities of digital audio workstations, to the fast-access and high-storage capacity of hard disk systems, to the dependable audio delivery of a digital STL, few will dispute that digital equipment improves upon the functionality of the analog components it replaces.

But when it comes to audio processors, some people in the industry continue to prefer analog technology. Backers of analog audio processing maintain that analog equipment has superior sonic performance, in part due to the years of experience that has gone into designing and perfecting analog audio processing circuits. Some users cite a "graininess" or "grittiness" that they perceive lurking just below the desired signal in digital audio processors. Also, some suggest that when lossy compression has been used in audio storage or transmission, a high-quality analog processor is superior to a digital processor in its ability to reconstruct to original signal.

While we can measure aspects of sound, such as frequency and audio response, ultimately, every listener evaluates sound quality subjectively. So it's not surprising that the choice of analog or digital audio processing systems comes down to each station's preference in equipment that will give them the sound they hope will grab their listeners.

Planning a Future-Proof AM Transmission Airchain

Of all the changes on the horizon for radio broadcasters, few are more dramatic than the audio quality enhancements that will arrive once it is possible to transmit digital AM signals. The broadcast industry is preparing for the day in the near future when in-band-on-channel (IBOC) digital broadcasts will bring digital audio to the AM band that combines the noise-free clarity of FM with the signal strength and carrying power of AM. Many of the critical components of the digital AM future are being designed and tested today. Two major U.S. groups are currently working to develop digital broadcast systems for domestic radio. Once these two proposed systems are tested and the FCC makes a decision to approve one as a standard, the beginning of digital AM broadcasting will be underway.

So, what does a broadcaster do in the meantime if faced with the need to replace or upgrade aging AM transmission equipment? How can you be sure that the choices you make today won't compromise your ability to handle new digital formats in the future? Here are some things to consider as you plan an AM system that will accommodate your requirements today and give you the means to handle digital AM broadcasts in the future.

All Harris DX transmitters have direct digital synthesis of the RF envelope using true digital modulation. The DX 50 AM transmitter is 100 percent solid-state and has a power output range of 10-60 kW.

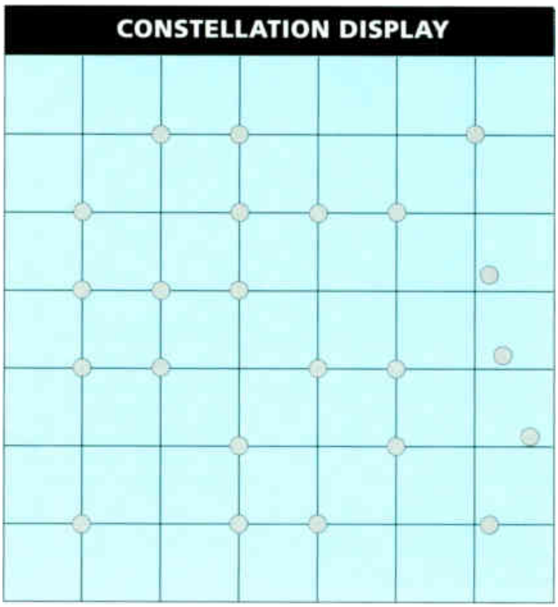
Creating a Digital AM Signal

The equipment and communications requirements for creating a digital AM broadcast in the studio and delivering it to a transmitter site are the same as for an FM broadcast. Digital source equipment, editing workstations, consoles, and studio-to-transmitter links discussed in this *Key to Digital Radio* can be used for both AM and FM (see "Advantages of an Uncompressed Digital Studio-to-Transmitter Link" p. 5). Equipment requirements differ significantly, however, once we get to the digital AM exciter, the transmitter, and the antenna system.

The two companies working on the digital AM broadcasting standard are also working on designs for AM digital exciters that will make IBOC broadcasting possible in the near future. Equipment designs and modulation techniques currently in testing will need FCC approval and, possibly, further refinement before they are made available for purchase. The major systems proposals now under development, however, share in common the use of some form of OFDM (Orthogonal Frequency Division Multiplexing) to create the digital AM signal.

We are all familiar with the carrier frequencies we tune in on our radio dial. In present-day analog AM transmission, broadcast equipment imposes a station's audio signal on this single RF carrier by manipulating the wave's amplitude. Another way to impose an audio signal on a carrier is to manipulate its phase, as is done in FM broadcasting. OFDM imposes a digital signal on the AM transmission by inserting multiple RF carriers on both sides of the main carrier, and then phase- and amplitude-modulates each carrier. By manipulating combinations of amplitude and phase on each of these carriers, it is possible to define multiple digital values, which form the digital "words" needed to reconstruct the digital information. This can be shown graphically using a constellation diagram.

The constellation display illustrates a modulated IBOC AM sideband carrier. Four errors appear on the right of the diagram.



The constellation display here shows how each carrier would appear on a vector scope. The horizontal lines represent levels of amplitude. The vertical lines represent positions of phase modulation. When a signal is modulated at the proper amplitude and phase, it appears exactly where the line of phase and amplitude cross. The weakness of the system is that it does not tolerate phase distortion or time distortion. Successful AM IBOC requires a linear system a minimum of

10 kHz wide on each side of the center frequency.

The OFDM signal consists of several hundred precisely pre-defined points on such a grid. At any given time, each point can be electrically present or absent. Thus, at any moment in time, the arrangement of points on the constellation diagram can be manipulated to represent a complex digital word. A constant stream of such words carries the digital transmission to listeners who have digital receivers. As you can imagine, the key to making OFDM work is precision.

The Harris DX 15 AM transmitter has a power output of 2-15 kW with three adjustable pre-set power levels.

Transmitter Requirements

Several characteristics can affect a transmitter's ability to deliver an OFDM signal with the precision needed to assure reliable reception. Moreover, when AM broadcasters begin to transmit digitally, they also must continue to deliver an analog signal for listeners who don't yet have digital radios. The need to operate both in analog and digital (hybrid) fashion will last for a period of 12 to 15 years and places additional demands on a transmitter. Key parameters to look at when buying a new transmitter (or upgrading an existing transmitter) are:

• Noise Rejection: Noise can come from several sources, including intermodulation distortion, incidental quadrature modulation (IQM), thermal noise, and electromagnetic noise. Today's solid state AM transmitters generally have good noise specifications in the range of -65 dB below carrier, which is sufficient for reliable hybrid AM operation.

Older tube-type transmitters' noise performance is generally in the range of -55 dB, placing them at the minimal threshold for reliable operation. While it may be possible to convert a tube-type transmitter for digital use, doing so will require more work and attention to detail than converting a typical solid-state transmitter. Also, the continual change in operation characteristics as tubes age will require periodic fine-tuning of the equipment to maintain signal integrity.

• Incidental Quadrature Modulation: IQM is the phase shift of an RF carrier that occurs with the amplitude modulation of the carrier. Because OFDM relies on its ability to manipulate phase intentionally, if the phase is varied in an undesired way with the AM modulation, while the same RF phase is being modulated in a desired way by the digital exciter, there will be a conflict that results in inaccuracies in the digital signal. IQM should be less than -36dB at 95 percent modulation for good digital service.

• Modulation Bandwidth: Normal modulation bandwidth for an analog transmitter is ± 4.5 kHz for international service or ± 10 kHz in the U.S. (Region 2). Digital transmission can require a modulation bandwidth at least ±10 kHz or 20 kHz total (10 kHz on both sides of the carrier frequency). Newer digital transmitters are designed to meet the bandwidth requirements of digital broadcasting. However, a number of technical factors (including group delay and modulator switching frequency) can add to the difficulty of converting some existing transmitters to handle the additional bandwidth needed for digital transmission.

• Peak Modulation Capability: The final consideration is the peak modulation capability of the transmitter. OFDM modulation can require this peak modulation (or crest factor) to be as high as 10 dB. However, current testing indicates the possibility of transmitting good digital quality signals with a crest factor of 4 to 6 dB, depending on the linearity of the RF amplifier. To be prepared for the future, the ability of a transmitter to handle high modulation peaks should be considered when buying new equipment.

The bottom line is that a broadcaster may be able to use an existing transmitter for digital (and hybrid) broadcasts. But before planning on such a conversion, a station's engineering staff should first assess the capabilities of existing equipment and carefully evaluate cost and reliability issues. Depending on existing equipment, one broadcaster may be able to accomplish a reliable upgrade at reasonable cost, while another may be better off buying a new transmitter. If buying a new transmitter

Harris' 300-2000 kW DX transmitters extend the benefits of digital amplitude modulation and solid-state to high and super power applications.

now for analog use, make sure that it is also capable of handling input from a digital exciter in the future and that it will be able to operate in hybrid digital analog mode.

Within the next few years, in-band-on-channel (IBOC) digital broadcasting will deliver an enhanced, noise-free signal combined with the signal strength and carrying power AM listeners expect.

Antenna Systems

The final, but also critical, area for attention is the antenna system. Even if your transmitter meets all specifications for digital broadcasting, an antenna system that is not properly adjusted can add distortion that shifts the amplitude or phase relationship of your broadcast signal. Whether you are upgrading your present transmitter or installing a new one, you will have to retune your antenna system.

The antenna system's Antenna Tuning Unit (ATU, also often called a phasing network) will have to be properly matched and adjusted to assure that that the signal that leaves your antenna is as accurate as the signal that left your transmitter.

Harris Broadcast Technology Training Center

Proper operation and maintenance of broadcast equipment is essential for its long-term reliability. Courses and training materials offered through the Harris Broadcast Technology Training Center (HBTTC) provide engineers with the skills and knowledge needed to ensure maximum equipment longevity and performance.

Since its inception in 1975, more than 2,500 broadcasters from around the world have participated in RF training programs at the HBTTC. Each year, the training center offers more than 30 regularly scheduled programs for engineers at all levels of experience. To meet special needs, the HBTTC also designs customized training programs for broadcasters. Courses can be taught at Harris or at most customer sites.

Since the Harris Broadcast Technology Training Center's inception in 1975, more than 2,500 broadcasters from around the world have participated in the center's RF training programs.

In addition to offering courses directly to the broadcast community, Harris joined forces with John Wood Community College in Quincy, Illinois, more than a decade ago to offer a fully accredited two-year program in Broadcast Technology, leading to an associate's degree and SBE certification. More than 300 students have graduated from this program, and over 90 percent of them have been employed in the electronics industry within 90 days of graduation. Others have transferred to four-year colleges and universities to

Each year, the Harris Broadcast Technology Training Center offers over 30 regularly scheduled programs for engineers at all levels of experience.

complete Bachelor of Science requirements.

Training center capabilities and course offerings are designed with the needs of the broadcast community in mind.

- The training center is equipped with classrooms and laboratories used exclusively for training.
- All instructors are professional teachers with more than 60 years of combined teaching experience.
- Instructors are involved in field service, which keeps them up-to-date and in touch with real world conditions.
- Classrooms contain fully operational radio and TV transmitters for demonstration, hands-on training of adjustments/alignments, and troubleshooting exercises.
- To meet the broadcast industry's need for RF experienced personnel, the training center offers special RF courses designed to bring non-RF technicians up to a level of competence to work on and maintain RF systems effectively and safely.
- RF courses are enhanced with hands-on lab exercises/activities utilizing actual RF hardware and test equipment essential in maintaining typical broadcast transmitting facilities.
- All classes are guaranteed. If for any reason an attendee does not believe the training has met their expectations, Harris will offer free enrollment in any Harris training class of the customer's choice.

Avoiding the Pitfalls of Upgrading FM Stations to Digital

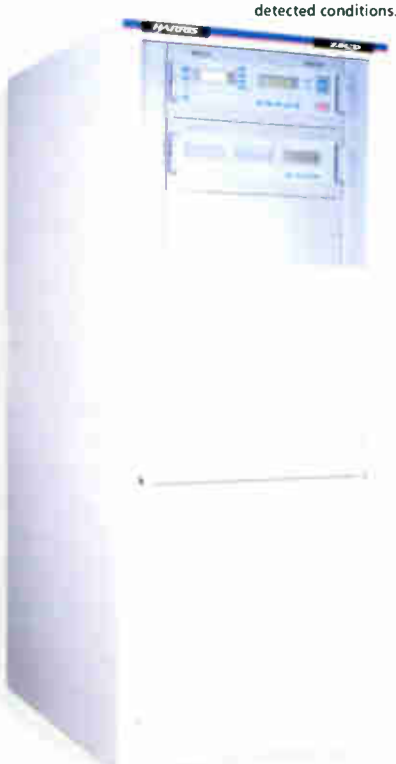
Digital replacement of analog broadcast equipment at radio stations is at an all-time high. Not only are studios converting to digital, but the RF portions of radio stations are also rapidly changing to digital products. An FM broadcast station can now keep the ones and zeros from their source equipment in digital form all the way to the generation of the FM signal. Few broadcasters, however, have the luxury of creating a new digital facility from the ground up. Most will upgrade progressively as they replace existing equipment or expand existing facilities. How, then, can a station most efficiently make the transition from analog to digital — and how can a station avoid pitfalls along the way? Here is an overview of the technical standards, equipment, and systems involved in making the change.

You can see that the capacitance of analog audio cable is almost three times higher than AES3 rated cable. Because the bandwidth of the AES3 signal is so much higher than analog audio, this excess capacitance can affect the digital signal, especially slowing the rise and fall times on the data edges, making decoding more difficult and more errors possible.

The chart on the next page summarizes characteristics of the two types of cable rated for digital audio.

Most digital equipment supplied today uses 110 ohm inputs and outputs. If you want to use

The Harris Platinum Z10CD FM transmitter is equipped with a microprocessor-based controller that monitors more than 100 operating functions and makes intelligent operating decisions based on detected conditions.



Characteristics of Digital Audio Cable

AES3-1992	AES3-ID
110 Ohm	75 Ohm
Twisted Pair	Coaxial
XLR Connector	BNC Connector
Balanced	Unbalanced
Greater Noise Immunity	Less Radiation
100 Meters Maximum	1,000 Meters Maximum
XLR Pins	BNC
1 — Shield	Center Data
2 — Data +	Shield
3 — Data -	

75 ohm coaxial cable, you will have to buy impedance transformers for most of your digital equipment connections. For short runs, you will save money by using 110 ohm cable for most of your wiring. There are times, though, when you may need 75 ohm coax for long runs that exceed the maximum length specified for twisted pair.

System Synchronization

The Audio Engineering Society has established a specification — AES11-1991 — for a synchronization reference signal for use when switching between two digital signals. This signal is a highly stable clock running at the sample frequency selected by a broadcaster for studio- or facility-wide use. The AES11 standard specifies accuracy to 10 PPM for use in a typical studio and sets a more stringent specification of 1 PPM if the signal is used to synchronize a suite of studios.

Facility-wide synchronization is hindered by the lack of a synchronization input port on much of today's equipment, but this situation is changing as newer equipment comes on the market. For new installations, consider installing two AES cables to every piece of new digital equipment in anticipation of the time when all digital equipment will allow input of synchronization signals. For now, you can purchase sample rate synchronizers and converters to synchronize signals to a standard reference, such as your digital audio console.

Digital Audio Consoles

When choosing a digital console, look for the features that you would expect an analog audio console to have. Most analog features are now available on digital audio consoles.

You may have to choose between 32 kHz, 44.1 kHz, and 48 kHz sample rates as your fixed rate when selecting a console. Several model digital consoles can be set to any of the three sample rates. Many broadcast radio stations choose 32 kHz as their standard because FM broadcasting is limited to 15 kHz audio channels. A sample rate must be at least twice the analog frequency (according to the Nyquist theorem), so 32 kHz provides the maximum 15 kHz frequency response that can be broadcast in FM.

For stations using a digital audio storage system, a sample rate higher than 32 kHz will add to the cost of hard disk storage space. For example, a 48 kHz sample rate will take up one third more space on a hard disk. Some stations may choose higher sample rates for non-broadcast audio production or storage reasons. But for broadcast purposes, 32 kHz is sufficient for most stations.

Digital Music Storage

Most digital music storage systems offer some form of digital compression that decreases the amount of space required for stored material. There are two ways to compress digital audio. One way uses statistical/mathematical coding. The other way is by psychoacoustic coding. Psychoacoustic methods take advantage of the ear's ability to focus on certain sounds while ignoring others. The sounds that would be ignored are not stored for reproduction. Many compression schemes use both statistical and psychoacoustic bit-reduction methods. The more familiar compression algorithms are APTx, MPEG ISO Layer II and III, Franhoffer, Dolby and PAC. All of the systems make choices to reduce the number of bits to be stored.

Compression isn't obvious to most people if the audio is compressed only once. But when you compress and decompress digital audio two or more times, people start to notice. For example, if you use digital compression for both music storage and for your studio-to-transmitter link, many people will be able to hear digital compression artifacts, or noise, in your signal. Compressing and decompressing two or more times is called "cascading algorithms" and is a practice best avoided. You should also pick one compression scheme for use with a storage system. If you mix compression schemes at the studio, you could have compatibility problems between components, and your station may suffer from

The Yamaha 03D digital mixing console is equipped with 18 analog and eight digital inputs, 32-bit signal processing, and over 300 factory-preset and user-definable libraries for effects, dynamics and other audio characteristics.

inconsistent sound quality.

Successful implementation of digital storage requires a backup system for stored materials. A station without backup is in jeopardy of losing programming and air time. And finally, because direct file transfer capability is important, be sure your hard disk system and your digital editor can talk to each other. Harris and other equipment manufacturers can provide technical support to help you make the decisions that are right for your facility.

Digital Audio Processors

For years, stations have used analog audio processors to help create the station's "exclusive sound." In the last few years, several companies have begun manufacturing digital audio processors.

Digital audio processors may come with either fixed or selectable sample rates. When buying an audio processor, make sure it operates at the same sample rate as your digital console's outbound audio. While it is possible to convert sample rates, it is best to keep conversions to a minimum because each conversion may slightly change a digital signal. A single rate conversion is usually not perceptible. But, each time you sample rate convert, the small digital changes accumulate and may become noticeable. By standardizing on 32 kHz, an FM station can minimize the need for sample rate conversion as well as reduce the cost of digital audio storage. However, some stations may want to use a 48 kHz standard and sample rate convert to 32 kHz. This is acceptable as well, especially if you are archiving programs, such as symphonies or music concerts for future rebroadcast or distribution. If so, you may want to use higher sample rates and greater than 16-bit resolution for the highest quality archiving currently available. The higher quality of your archived programming will not improve your broadcast audio quality, but the digital future may hold methods of storage and playback not dreamed about today.

Studio-to-Transmitter Links

There are several ways digitally to get the AES3 signal to the transmitter, including digital telephone connections, such as

T1 (or a subset of T1), fiber-optic cable, spread spectrum RF, and classic RF STLS. Each method has advantages and disadvantages.

T1's advantages are that it can be uncompressed or compressed, it can have many channels, and have two-way (full duplex) operation. T1's disadvantages may be higher initial equipment and set-up expenses, continuing monthly payments to the telephone company, and loss of control of your link. However, many radio stations successfully use T1s where there is no line of sight available or STL RF spectrum available.

Fiber-optic cable is another choice for some people who want to send the AES3 signal a considerable distance and who have or can obtain the right-of way for their cable run. Fiber-optic cable maintains signal integrity over greater distances than coaxial cable. If you don't own the fiber-optic cable run, the telephone company or cable company will have control of your link and you will pay monthly fees. Although fiber optic can be economical on a short run, for longer runs, it will be more expensive than a RF STL path.

Spread spectrum RF technology can deliver uncompressed AES3 audio to your transmitter without the need for land lines. Spread spectrum systems operates under FCC rules Part 15 which means that a station is not exclusively licensed for a specific frequency or set of frequencies. However, due to the characteristics of spread spectrum, the chances of having significant interference to your STL are very low.

For most broadcasters, the classic RF fixed-frequency STL offers the best and most cost-effective way to get their studio audio to the transmitter site. There are two types of fixed-frequency digital STLs, those that use digital compression and those that don't. A compressed STL can degrade audio if you also use compression with your digital audio storage. Cascading the two compression algorithms is unavoidable and will almost certainly add noticeable noise and distortion to the station's sound.

The Harris CD LINK™ transmitter and receiver units have similar control layouts to simplify the learning curve and shorten set-up time.

Using an uncompressed STL eliminates the problem of cascading algorithms. And an uncompressed STL brings the added advantage that the digital audio processor can remain at the studio to do all the "sweetening" you want and permit the processed audio to remain pre-emphasized. The digital studio audio is delivered to the transmitter completely processed, uncompressed and unaltered by the STL.

In contrast, compressed STLs cannot properly carry pre-emphasized digital signals. If you use a compressed STL and had the audio processor at the studio, you would have to de-emphasize the audio and then pre-emphasize it again in the FM exciter at your transmitter site. When you use a digitally compressed STL, the digital compression makes having the audio processor at the studio less desirable. It may be best to send relatively plain, unprocessed audio through digitally compressed STLs, and then do the audio processing at the transmitter site.

Once again, it is important to consider sample rates. Ideally, the sample rate of your STL should be the same as the sample rate of your digital audio console and your audio processor, otherwise you will have at least one sample rate conversion in the digital path.

And finally for 950 MHz STLs, consider your RF neighborhood. If your digital STL is in a high RF environment shared with nearby high-power paging transmitters, amateur radio repeaters, microwave links, or other RF sources, you may want to consider an on-channel RF filter for your STL frequency to limit interference between the STL and neighboring services.

Digital FM Exciters

The FM exciter is one of the first upgrades a station should consider when planning a transition from analog to digital. The exciter has the job of taking all the talent, music, creativity, and the desired station sound and turning it into an FM broadcast signal. All the digital upgrades in the studio, audio processing, and transportation of the signal can't become FM without the exciter.

Most digital FM exciters utilize a numerically controlled oscillator (NCO). Direct digital synthesis of FM with an NCO is true digital generation of an FM signal. Because AES3 encodes left and right channel information in a single digital stream, digital FM exciters must have the digital stereo generator built-in. A composite digital baseband can then be generated from the stereo AES3 digital input audio. This process is analogous to creating an analog composite stereo signal from left and right audio information.

To upgrade your FM exciter to digital, you will need to consider the exciter RF power output requirements of your transmitter. Also, physical size

Harris' DIGIT® CD digital FM exciter delivers absolute digital control of carrier center frequency and deviation. Its digital circuits cannot drift and do not need adjustment.

may be important, especially if you want to install the exciter in an existing transmitter. However, if faced with physical incompatibility, FM exciters may be mounted in a rack cabinet adjacent to the FM transmitter with no operational penalty.

If you currently have an analog transmission system with only a few digital items at the studio, the logical first step when converting to a digital airchain is to install a digital FM exciter. Consider ordering an exciter with the digital input module and an outboard A/D converter if you do not have an AES3 input available. If you select the analog input module as standard for your FM exciter, then you will still have to supply an analog stereo composite input signal to the digital exciter. However, users who provide a digital input to the new digital FM exciter will get the benefit of the built-in digital stereo generator that is part of the digital input module.

The next step is to purchase either a digital audio processor and I/O or digital STL. Both pieces of equipment can do an A/D conversion for you so that you won't need to buy any external A/Ds when you get a digital input FM exciter.

Digital Upgrade Checklist

☐ WIRING FOR DIGITAL

- ☐ Use only cables rated to handle AES3/AES3-ID.
- ☐ When doing new installations, run two pair, one for audio and one for future synchronization signals.

☐ DIGITAL AUDIO BROADCAST CONSOLES

- ☐ Look for the digital version of analog features you use now.
- ☐ Choose a console with selectable sample rates or 32 kHz only.
- ☐ 48 kHz may be the best choice for some stations. Review station needs.

☐ DIGITAL MUSIC STORAGE

- ☐ 32 kHz saves hard disk space.
- ☐ Choose one digital compression system or none.
- ☐ Have a backup system in place.
- ☐ Make sure your editor and hard disk system are compatible.

☐ DIGITAL AUDIO PROCESSORS

- ☐ Choose an I/O card that matches your digital console and digital STL.
- ☐ For FM, 32 kHz allows full 15 kHz frequency response.
- ☐ Compressed STLs limit your audio processor placement or may require additional steps of pre-emphasis and de-emphasis.

☐ STUDIO-TO-TRANSMITTER LINKS

- ☐ Choices include T1 digital, Fiber-optic, Spread Spectrum, Conventional RF STL bands.
- ☐ In STLs, compressed and uncompressed are the common choices.
- ☐ Compressed STLs alter the signal coming from the studio.
- ☐ Compressed STLs may exhibit overshoot from the compression processing.
- ☐ Compressed STLs may make cascading algorithms unavoidable.
- ☐ Compressed STLs need to keep the audio processor at the studio.
- ☐ Uncompressed STLs are like an RF digital wire, bit-for-bit transport of the audio. You have total flexibility in audio processing and freedom from cascading algorithms.

☐ DIGITAL FM EXCITERS

- ☐ Choices include T1 digital, Fiber-optic, Spread Spectrum, Conventional RF STL bands.
- ☐ A Digital FM exciter is your best upgrade choice to improve station signal quality.
- ☐ NCO digital exciters utilize true digital FM generation. Not all digital exciters have NCOs.
- ☐ Check exciter RF power requirements, transmitter interconnections, and physical size when upgrading.
- ☐ Order a digital FM exciter with a digital input.
- ☐ If you don't have a digital input now, use an A/D converter, like a Harris A2D2A, any digital audio processor, Belar DSD-LA (converts analog composite stereo) or CD LINK, to convert analog to digital for your exciter.
- ☐ Finally, for greatest broadcast quality improvement, a general digital improvement rule is to work backwards from a digital FM exciter, unless you have a really poor audio processor or other weak analog link in the chain.

Extended Market Coverage with Simulcasting

The Harris/Intraplex SynchroCast™ system allows FM broadcasters to increase their geographic coverage dramatically by simulcasting programming over multiple transmitters on the same frequency.

Harris/Intraplex SynchroCast™ synchronizes FM broadcast transmitters using precise time from Global Position Satellites (GPS) and digitally induced delays.

SynchroCast technology lets broadcasters use multiple, overlapping transmitters on the same frequency while maintaining phase-alignment, thanks to Global Positioning Satellite (GPS) technology, to attain greater geographic coverage.

With SynchroCast, a broadcaster can ring a metropolitan market with lower-power suburban stations, providing combined coverage far greater than that of a single, high-powered metro station. These facilities enjoy much lower acquisition and operating costs, giving them a competitive edge.

Alternatively, using SynchroCast systems in several communities along a highway means that listeners can keep their radios on the same program as they travel, without the need to retune.

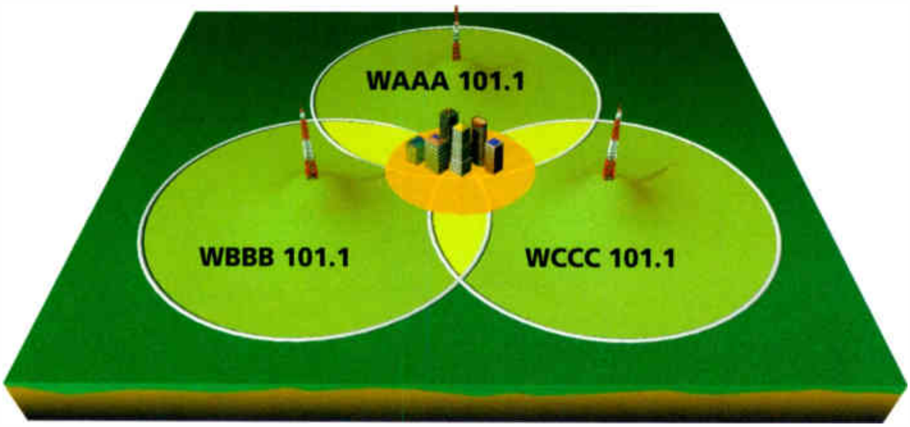
Why SynchroCast?

Past attempts to increase coverage by simultaneously broadcasting the same programming on different frequencies have been only marginally successful because the listener needs to retune the radio when moving from one coverage area to the next. The same is true when stations use translators to fill in coverage gaps. The only way to keep people listening is to stay on the same frequency.

Using traditional methods, broadcasting the same signal from two nearby transmitters on the same frequency can create serious reception problems in the areas where the transmitters' footprints overlap. This occurs for two main reasons.

• For simulcasting to work effectively, the broadcast signal must arrive at each transmitter at a precisely controlled time. Because the broadcast signal takes a different path to each transmitter site, the inherent, uncontrolled STL path delay will be different for each transmitter.

• Each transmitter's local oscillator frequency can drift, causing phase errors between carrier frequencies in the overlap area.



Using SynchroCast™, a broadcaster can expand market coverage by using synchronized transmitters on the same frequency.

How Does SynchroCast Work?

SynchroCast technology, originally developed and refined for use in land-mobile radio systems, makes this type of broadcast possible by integrating two state-of-the-art technologies: GPS satellite receivers and digitally induced delay.

First, SynchroCast uses the GPS receivers to determine absolute reference times at each location. This timing information is passed along with the broadcast audio from the studio to each transmitter site. At the transmitter, the timing references are compared to determine the actual path delay. Once the actual delay is established, digital delay modules introduce a slight but precisely controllable additional delay, allowing exact alignment of the adjacent transmitter signals.

The system operates automatically once the initial installation and alignment are complete. SynchroCast continually monitors the timing of each link, keeping the total delay to each transmitter constant even if the actual path delay changes, as may occur, for example, if a T1 circuit gets routed to an alternate path.

Finally, at the transmitter sites, the GPS receivers provide the same 10 MHz reference signal to every transmitter, locking all of their carrier frequencies to the satellite-delivered reference.

The SynchroCast system operates in conjunction with Harris/Intraplex STL PLUS digital STLs which can be used on leased T1 circuits, microwave radio links, or fiber-optic links. The STL PLUS system provides bidirectional transmission paths for program audio STL/TSL, data for remote control and LAN interconnect, and voice channels for off-premise extensions and intercoms, all on the same digital circuit.

The equipment requirements are:

- One Harris/Intraplex STL PLUS, equipped with the user's choice of channel modules, for each studio-to-transmitter link
- One Spectracom Model 8195 GPS receiver for each site in the system
- One SynchroCast Equipment Package, which includes all the GPS modules, timing transmission modules and digital delay modules necessary for the studio and two transmitter sites
- One SynchroCast Expansion Package for each additional transmitter site beyond the first two

Using SynchroCast in Europe

In Europe, broadcasters can extend coverage by using RDS radio systems to provide automatic frequency switching. But even with RDS systems, broad geographic coverage can require large numbers of frequencies. A more efficient way to deliver wide area coverage is to run multiple adjacent transmitters on the same frequency.

Once allowances are made for the differences in telecommunications and equipment standards, installing a SynchroCast system in Europe is essentially the same as in North America.

Equipment requirements for a European system are:

- One pair of Harris/Intraplex E1 multiplexers for each studio-to-transmitter link, equipped with channel modules for audio programs and other optional user traffic
- One GPS receiver for each site in the system
- One SynchroCast Equipment Package, which includes all the GPS modules, timing transmission modules, and digital delay modules necessary for the studio and two transmitter sites
- One SynchroCast Expansion Package for each additional transmitter site beyond the first two

Digital Test Equipment

Troubleshooting digital systems is essentially the same as troubleshooting analog systems. It's done with precision test and analyzing equipment — the most important piece of which is the inquiring mind of a skilled technician. The troubleshooting skills engineers and technicians have always used to debug analog systems will also lead them to the source of trouble in digital systems. It remains a matter of isolating and testing components and sub-systems and following a logical trail to the source of the problem. The more experience a technician has, the more likely he or she is to isolate the source of trouble quickly, having wrestled a similar problem to the ground somewhere in the past.

Where things differ in the digital world is the nature of the signal and the technical standards of service and test equipment needed to work with digital signals. One key difference is that in an analog system, it's possible to plug in a headset anywhere and hear both the presence and the quality of the audio. Even without plugging in a piece of test equipment, a technician can hear the strength of the audio, estimate signal-to-noise ratio, and hear electrical interference or crosstalk. There isn't such a simple shortcut with digital signaling. A technician with a set of headphones may be able to tell there is something on a line but can't readily tell whether the noise is a usable AES signal, a meaningless scramble of ones and zeros, or just plain noise.

The Harris Audio Bit Buddy is a low-cost, portable unit for monitoring digital audio (AES/EBU, S/P-DIF) or analog audio signals. The Audio Bit Buddy has a headphone output and LEDs to display basic signal parameters and audio signal quality.

You have to use digital test equipment to troubleshoot digital audio. Careful system design can make maintenance and repair much easier. Digital audio monitors installed at key points throughout an audio system can provide visual monitoring of audio

The portable, battery-powered Audio Bit Spitter generates digital and stereo analog audio test signals that can be injected into the signal path when testing device performance or signal path continuity with an Audio Bit Buddy or other audio monitoring device.

quality and can also provide headphone jacks through which you can hear the actual audio, rather than a stream of digital noise.

An engineer can use a digital audio analyzer to trace a signal through a system, to hear the digital audio signals, and to take precise and comprehensive measurements of parameters, such as audio level, frequency response, total harmonic distortion, intermodulation distortion, and frequency. Small, hand-held analyzers can be used for quick troubleshooting and can be used to listen to the digital audio and to monitor the status of validity, parity, channel status, and user bits in the digital signal.

As with analog systems, engineers use a "fox and hound" system for tracing digital lines. An audio generator places a digital signal on one end of a line, and the engineer uses an audio monitor to test the presence or quality of the signal at the other end of the line. Lightweight and easily portable test devices make this kind of troubleshooting easy and efficient. One option is to use a small, battery-powered signal generator at one end and a small, digital audio monitor at the other end. Another is to use a pair of hand-held digital audio analyzer/generators — one in generator mode at one end of the line and the other in analyzer mode at the other. But either way, it's helpful to have troubleshooting generators and monitors on hand that can operate at the standard digital sample rates and can measure basic audio parameters.

Some of the audible noises an engineer will hear while monitoring digital systems will be different from familiar analog system sounds. But with the aid of test equipment designed to analyze digital signals and with the troubleshooting skills an audio engineer already possesses, it won't take long to learn to hunt down and repair problems in the digital world.

Harris Broadcast Communications Division

The Harris Broadcast Communications Division is North America's oldest and largest broadcast manufacturer. The division was founded in 1922 as Gates Radio and Supply Company, which in 1957 became part of what is now Harris Corporation, an electronics company with annual sales that today exceed $3.4 billion. Over the years, Harris Broadcast has introduced more than 60 major innovations in broadcast transmission technology, including many breakthroughs that have become world standards.

Within the U.S. radio and television markets, Harris is the leader in advanced transmitter equipment and systems for digital radio broadcast and digital television (DTV). In the United States, close to one-half of all radio stations and over one-third of all television stations use Harris transmission equipment. Transmission equipment developed by Harris is used in over 150 countries. In addition to designing and manufacturing products for the broadcast industry, the Broadcast Communications Division distributes more than 10,000 studio products and provides studio systems integration for broadcast companies worldwide.

The Broadcast Communications Division is headquartered in a recently completed 160,000-square foot facility in Cincinnati, Ohio. The new facility consists of an administration

Harris manufactures radio and TV transmitters at its Quincy, Illinois facility. The main Quincy factory occupies 125,000 square feet.

engineering center and an industrial building. The administration/engineering center includes offices for Broadcast Communications management, sales, marketing, and engineering staffs, as well as customer demonstration and meeting areas. Three engineering laboratories located at the facility were built from the ground up to meet the stringent requirements for RF engineering. The adjoining industrial building contains an assembly area for Harris' systems integration business as well as warehouse space for products distributed by the Broadcast Communications Division.

The Broadcast Communications Division operates its main U.S. manufacturing facility in Quincy, Illinois; a European manufacturing center in Cambridge, England; and a European product development center in Rennes, France. Harris sales and service facilities are located in nearly 90 countries around the world.

The Broadcast Communications Division serves three primary areas of business: Radio Systems, Television Systems, and Systems Integration.

Radio Systems

As the world's leading supplier of radio broadcast transmission equipment, studio products, and systems, Harris offers a comprehensive range of analog, digital-ready, and digital broadcast transmission equipment. Harris is leading the digital transition in the radio broadcast studio. From the introduction of the first digital

The Harris Broadcast Communications Division is headquartered in a recently completed 160,000 square foot facility in Cincinnati, Ohio.

audio storage systems to the first digital on-air console, Harris has pioneered the introduction of many digital audio solutions.

Radio products, systems, and services include:

- Digital solid-state AM and FM radio transmitters
- Analog AM and FM radio transmitters
- Digital Audio Broadcasting (DAB) systems
- Phasors, tuning systems, and AM and FM antenna systems
- Digital studio-to-transmitter links (STL)
- The world's most comprehensive range of analog and digital audio studio products, including storage and retrieval; mixing, routing, and switching; and source products
- Satellite communications systems
- Pre-wired radio studios
- Digital audio and video storage and retrieval systems
- 24-hour service for Harris-manufactured products
- Factory authorized service for many distributed non-Harris manufactured products
- Custom-integrated radio broadcast transmission and studio systems
- Training

Television Systems

Harris has set the pace for television's transition from analog to digital technology. The RF Test Bed that Harris developed for the Advanced Television Test Center in 1990 enabled all digital

television systems proposed for the United States to be evaluated equitably. Harris also is a leader in technology for the European-standard Digital Video Broadcasting (DVB-T) market.

Harris offers the industry's largest range of field-proven digital and digital-ready television transmission equipment. Such Harris innovations as the first commercial digital television exciter and transmitter have enabled broadcasters to move swiftly toward the implementation of digital television.

Television products, systems, and services include:

- Digital television transmission systems
- Solid-state VHF and UHF transmitters
- Digital video broadcasting (DVB-T) equipment
- Digital television encoding systems
- Antenna systems
- Custom-integrated broadcast systems
- Digital television RF conversion studies
- Installation
- 24-hour service
- Training

The Broadcast Communications Division delivers trusted and dependable products and full-service solutions throughout the world at pit stop speed.

Systems Integration

Beyond manufacturing and distributing broadcast equipment, Harris integrates custom, future-ready systems for fixed and mobile applications. The systems integration team has designed and integrated systems for the broadcast industry for more than 20 years. With access to products from leading equipment manufacturers, Harris can design systems with components that precisely meet customer requirements.

Major products, systems, and services include:

- Electronic News Gathering (ENG), Satellite News Gathering (SNG), and mobile production systems
- Production, post-production, master control, and transmission systems
- Proprietary components for digital television, including MPEG-2 encoding systems and decoding systems
- Broadcast microwave systems
- Satellite communications systems
- Turnkey facilities
- Consulting, budgeting, and planning
- Technical design and documentation
- Equipment acquisition
- Systems integration and commissioning
- Custom product fabrication
- Technical furniture design and fabrication
- 24-hour service

GLOSSARY

AES/EBU
Audio Engineers Society/European Broadcasters Union — Two organizations that agreed upon a standard for the exchange of digital audio within the studio environment.

ALGORITHM
A formula or set of steps used to simplify, modify, or predict data. Complex algorithms are used to reduce high digital audio and video data rates selectively. These algorithms utilize physiologists' knowledge of hearing and eyesight.

AM
Amplitude modulation.

ANALOG AUDIO
A continuous, varying voltage which represents the electrical equivalent of sound. Analog audio has infinite precision within limits of S/N, since it is continuous. Analog audio of sufficient level may be used directly to drive an audio speaker, which changes the electrical energy back into acoustic energy (sound).

ANALOG-TO-DIGITAL CONVERTER
A device, usually an integrated circuit, which samples the incoming analog audio and outputs digital audio of defined work lengths, such as 12-bit A/D, 16-bit A/D, or 20-bit A/D.

ASYNCHRONOUS TRANSMISSION
Data transmission at irregular time intervals, not referenced to a master clock.

ATM
Asynchronous Transfer Mode — A specification of a protocol defining an ATM data cell and its handling in a network. Allows dynamic bandwidth allocation and may support multiple media.

ATU
Antenna Tuning Unit — Also called a phasing unit or a Line Tuning Unit (LTU).

BALUN
A device that converts between balanced cable (twisted pair) and unbalanced cable (coax).

BANDWIDTH COMPRESSION
Reducing the bandwidth that is required for transmission of a given digital data rate. Compression makes it possible to reduce gigabits of raw data into a DTV signal of about 20 megabits per second. This signal can be transmitted over the allotted 6 MHz bandwidth.

BER
Bit Error Rate — This term is an important indicator of the performance of the digital system. BER is the number of bits received in error divided by the number of bits sent. In various systems, there is a qualitative as well as quantitative aspect to bit errors. Some bit errors can cause more problems than others.

BINARY NUMBERS
Numbers based on the power of 2 and expressed as either a 1 or 0. For example, number 1 appears in a 5-bit word as 00001; number 2 appears in a 5-bit word as 00010; number 4 appears in a 5-bit word as 00100, etc.

BPS/HZ
Bits per second per hertz is a spectral efficiency number. Bps divided by occupied bandwidth gives bit/sec/hz.

BPSK
Bi-Phase Shift Keying —Digital bits are sent one at a time by shifting the carrier phase back and forth 180 degrees. This method also requires wide bandwidth. However, it does have the advantage of being least susceptible to noise. (Also, BSK Binary Shift Keying.)

CCD
An abbreviation for Charge Coupled Device.

CCIR
Comite Consultatif International des Radio Communications (The International Radio Consultative Committee). This standardization committee is now called ITU-R.

CCIR REC.601
The most important studio standard for digital video based on analog component 4:2:2 signals. Describes how to sample the component signal and how to structure the data.

CCIR REC.656
The standard describing how to interface to a CCIR Rec.601 formatted signal (serial and parallel 4:2:2 interfaces, commonly known as D-1 interfaces).

CCITT
Comite Consultatif Internationale Telegraphic et Telegraphonique (The International Telegraph and Telephone Consultative Committee). This standardization committee is now called ITU-T.

CLEAR CHANNEL
AM radio station allowed to dominate its frequency with up to 50 kW of power. Their signals are generally protected for distances of up to 750 miles at night.

CODEC
A device that compresses and decompresses digital audio (or video) signals.

COMPRESSION
The technique of reducing the amount of data needed to represent a video or audio signal. International standards are ETSI, MPEG, JPEG, G.722, etc.

DAB
Digital Audio Broadcasting is a generic term which is used for the delivery of digital audio by any method such as satellite, microwave, BC carrier, etc.

DAR
Digital Audio Radio — DAB as it refers to digital radio broadcasting.

DAT(RDAT)
Stereo digital audio recording format using rotary heads on 3.81 mm tape cassette.

DATA INTEGRITY
General term for any action or strategy which minimizes the proportion of data bits in a system which are corrupted.

DATA PORT
The physical and electrical protocol used by a codec and the DSU (Data Service Unit) or TA (Terminal Adapter) to transfer data between each other. A codec comes with either a V.35 or X.21 protocol built in. These numbers refer to CCITT international standards with V.35 very common for networks in North America and X.21 popular on European-manufactured ISDN terminal adapters.

DBM
Unit of audio level which dissipates one milliwatt in a load. With a load of 600 ohms, 0Bm is 0.775 V rms.

DBS
Direct Broadcast Satellite.

DBU
Unit of audio level where 0dBu is 0.775 V rms.

DIGITAL
A discontinuous electrical signal that carries information in binary fashion. Data is represented by a specific sequence of off-on electrical pulses. (see Bits, Binary Numbers)

DIGITAL AUDIO
A representation of analog audio created by taking samples of the analog audio at specific points in time. Digital audio uses bits to record the analog voltage. Because digital audio is finite, the more bits used at each sample point, the better the quality of the recreated audio when converting digital audio back to analog audio.

DIGITAL-TO-ANALOG CONVERTER
A device, usually an integrated circuit, which converts digital words into a standard varying audio voltage which is representative of the original analog audio.

DIRECTIONAL ANTENNA
An antenna that directs most of its signal strength in a specific direction rather than at equal strengths in all directions. Used chiefly in AM radio operation.

DISTORTION
Change in the waveform of an audio or video signal. Linear distortion due to phase shift does not change the signal spectrum. Nonlinear distortion introduces harmonics.

DITHER
Noise added to linearize quantizing step.

DOWNLINK
Earth station used to receive signals from satellites.

DSP
Digital Signal Processing is the process of changing analog audio signals to a digital format that may also be compressed and stored, or transmitted in the digital domain.

DSU/CSU
Data Service Unit/Channel Service Unit — Required between a codec and SW56 or DDS circuits, used to interface and condition the data coming on and off the network. It may contain diagnostic testing functions and will perform dialing tasks for switched services.

DTE/DCE
Data Terminal Equipment™ and Data Computer Equipment™ — To avoid confusion, the data protocols mentioned above designate equipment and ports as either DTE or DCE. In the case of the codec, the DSU/CSU or TA is always the DCE, and the codec is always the DTE.

EARTH STATION
Equipment used for transmitting or receiving satellite communications.

EMC
Electromagnetic compatibility — General term for set of regulations and procedures to ensure that electronic equipment neither suffers from nor generates interference.

ENG
Electronic News Gathering.

ENTROPY
The unpredictable part of a signal which has to be transmitted by a compression system if quality is not to be lost.

ETSI
European Telecommunications Standards Institute which produces standards through a number of technical committees.

ETV
Educational Television

FEC
Forward Error Correction — A system that substitutes for an error bit what it believes the correct bit should be. It may also be used in a generic sense for any system that encodes data to help minimize bit errors on the receiving end. For example, 3/4 FEC means for every 3 data bits sent, 1 FEC bit is sent.

FIBER-OPTIC CABLE
Wires made of glass fiber used to transmit video, audio, voice or data. Provides vastly wider bandwidth than standard coaxial cable.

FIELD
Half of the video information in the frame of a video picture. The NTSC system displays 59.94 fields per second.

FM
Frequency modulation.

FOOTPRINT
Area on earth within which a satellite's signal can be received.

FRACTIONAL
Relative to a part of a data rate. For instance, 64 kbit/s fractional of the 1.544 Mbit/s T1 service.

FRAGMENTATION
After writing information to a hard disk for a period of time, the disk develops areas too small to fit complete files. The system must spread the file over several areas of the disk to store. Because the read/write head must jump between these locations, the access time is slowed down, sometimes drastically.

FREQUENCY
The number of cycles a signal is transmitting per second, measured in hertz.

GEOSTATIONARY ORBIT
Orbit 22,300 miles above earth's equator where satellites circle earth at the same rate earth rotates.

GHZ
Gigahertz. 1,000 megahertz (cycles)

GROUP DELAY
This term defines the amount of time delay that may occur in a modulated signal as a function of frequency. Any unwanted delays in time affect phase relationships which may upset the proper de-coding of the QAM signal. In some cases, group delay problems may be improved by transmitter precorrection circuitry.

HEADROOM
Range of signal levels between standard operating level and level at which clipping or overload occurs.

HERTZ
A measurement of frequency. One cycle per second equals one hertz (Hz).

IBOC
In Band On Channel — A form of DAB in which the digital audio is broadcast within the same channel allocation as the standard analog program audio.

INTERLACED SCANNING
The concept of splitting a TV picture into two fields of odd and even lines.

IQM
Incidental Quadrature Modulation — The phase shift of an RF carrier that occurs with the amplitude modulation of the carrier.

ISDN
Integrated Services Digital Network — The worldwide standard for digital telephony. It actually describes a complex set of international standards.

ITFS
Instructional Television Fixed Service.

ITU-R
See CCIR.

JITTER
Unwanted frequency or phase variations of a data stream, or phase noise of a carrier signal.

kHz
Kilohertz. One thousand hertz (cycles) per second.

LAYER II
A specific subset of the audio part of the MPEG-1 and MPEG-2 standards. The Layer II specification features 32 to 384 kbit/s data rate; compression in 32 subbands with adaptive bit allocation determined by a psychoacoustic model; sampling rates from 16 to 48 kHz; and various coding modes.

LPTV
Low-power television.

MDS
Multipoint distribution service.

MHz
Megahertz. One million hertz (cycles) per second.

MICROWAVE
Frequencies above 1,000 MHz.

MJPEG
An abbreviation for Motion JPEG, a method of compression where each frame or field in a video signal is compressed using JPEG.

MPEG
Moving Picture Experts Group (or often Motion Picture Experts Group) — A working group that sets standards for the coding of moving pictures and associated audio. Standards are the MPEG-1 and the MPEG-2.

MPEG-1
A standard for the coding of moving pictures and associated audio for digital storage media up to about 1.5 Mbit/s. Includes separate specifications for audio and video. Used for broadcast audio (Layer II), CD video, etc.

MPEG-2
A specification that defines genetic syntax (that is, universally applicable rules of handling data) for transport of video and associated audio by means of data compression. Includes separate specifications on audio and video.

MTBF
Mean Time Before Failure is the estimated life for devices such as hard disks, etc.

NON-LINEAR
An editing system in which random access storage is used so that non-linear access to the material is possible.

NRSC
National Radio Systems Committee — A U.S. radio industry-sponsored group interested in setting standards for the improvement and compatibility of radio systems. NRSC is particularly interested in IBOC DAB systems.

NTSC
National Television Standards Committee — The American TV transmission standard which uses an interlaced 525-line, 30-frames-per-second picture.

OFDM
Orthogonal Frequency Division Multiplexing — A method for imposing a digital signal on an AM transmission by inserting multiple RF carriers on both sides of the main carrier and then phase- and amplitude-modulating each carrier.

OPTIMIZATION
A process where the computer reorganizes the information on the hard drive in order to eliminate the fragmented areas. This, in turn, improves the access time for the disk. Depending on the size of the disk, this process can take many hours and precludes any other operation on the computer.

OVERSAMPLING
Temporary use of a higher than necessary sampling rate in converters in order to simplify analog filters.

PAC
Perceptual Audio Coding — A source compression algorithm which also uses psychoacoustic knowledge and statistical information.

PAL
Phase Alternating Line — A European TV transmission standard found in several variants. Uses an interlaced 625-line, 25-frames-per-second picture — except PAL-M (only in Brazil), which uses an intelaced 525-line, 30-frame-per-second picture.

PBX
Private Branch Exchange — A privately owned or operated internal telephone system which is connected to the public switched telephone network.

PCM
Pulse code modulation. Conversion of voice signals into digital code.

PCS
Personal Communications Service — New digital wireless telephone technology with smaller and less expensive outdoor cells and consumer telephone sets than current cellular service. Cable television operators are among the entrepreneurs for PCS. Also called PCN (Personal Communications Network).

POTS
Plain Old Telephone System — Used to describe standard dial-up telephone lines.

PROGRESSIVE SCANNING
A method of displaying the horizontal video lines used in computer displays. The horizontal video lines used in the latest computers — and those that are planned for the future — use progressive scanning. Progressive scanning will be used in both DTV and computers. See Interlaced Scanning.

PUBLIC RADIO
Radio stations and networks that are operated on a noncommercial basis.

PUBLIC TELEVISION
Television stations and networks that operate as noncommercial ventures.

Q
Used to represent a phase shifted signal or vector when compared to the reference 1. It is used when referring to a 90-degree phase shift, also known as quadrature.

QPSK
Quadrature Phase Shift Keying — This digital modulation method sends four digital data states. Four phase positions are utilized 45, 135, 225 and 315 degrees at a fixed amplitude.

QUANTIZING
The process of converting the voltage level of a signal into digital data after the signal has been sampled.

RAM
Random Access Memory — Memory that can store data that can be changed.

RANDOM ACCESS
Storage device like a disk where contents can be output in any order. Contrasts with serial access.

RECLOCKER
A combination of a slicer and a phase-locked loop which can remove noise and jitter from a digital signal.

ROM
Read Only Memory — It can store data but cannot be changed.

SAMPLE RATE
The number of times per second (frequency) in which the analog audio is read and a digital word is created to represent that particular voltage at that point in time.

SAMPLING
A process in which some continuous variable is measured at discrete (usually uniform) intervals.

SCA
Subsidiary Communications Authorizations — Authorizations granted to FM broadcasters for using subcarriers on their channels for other communications services.

SECAM
Sequential Encoded Color Amplitude Modulation — 625/50 television encoding and transmission standard. B-Y and R-Y incompatible with PAL.

SHF
Super high frequency.

SHORTWAVE
Transmissions on frequencies of 6-25 MHz.

SIGNAL-TO-NOISE RATIO
The ratio between the strength of an electronically produced signal to interfering noises in the same bandwidth.

SDI
Serial Digital Interface — Digital video interface for production equipment.

SNR
Signal to Noise Ratio — Same as RF S/N.

S/PDIF
Sony/Phillips Digital Interface — Consumer grade digital audio interface.

STL
Studio-to-Transmitter Link.

SYMBOL
A grouping of a defined number of bits that is sent together as encoded data. For example, it could be 1, 2, 3 or more bits per symbol.

SYNCHRONOUS TRANSMISSION
Data transmission at a fixed rate in which timing is controlled by a master reference clock shared by transmitter and receiver.

TA
Terminal Adapter — This, in effect, is a CSU/DSU for an ISDN line. It adapts non-ISDN equipment to the ISDN user rate.

THD
Total Harmonic Distortion.

TRANSLATOR
Broadcast station that rebroadcasts signals of other stations without originating its own programming.

TRANSPONDER
Satellite transmitter/receiver that picks up signals transmitted from earth, translates them into new frequencies, and amplifies them before retransmitting them back to ground.

TSL
Transmitter-to-Studio Link.

UHF
Ultra high frequency band (300 MHz-3,000 MHz) which includes TV channels 14-83.

UPLINK
Earth station used for transmitting to satellite.

VHF
Very high frequencies (30 MHz-300 MHz) which include TV channels 2-13 and FM radio.

WORD
The number of bits expressed in binary form is used to define or represent a particular sample port (12-bit word, 14-bit word, 16-bit word). The more bits, the better the ability of the digital information to recreate the analog audio. Number of bits define the level of audio quality to be recovered from the digital audio.

WORM
Write Once Read Many — A data storage device such as CDs and some magneto optical disks.

YUV
A component video signal consisting of a luminance signal (Y) and two chrominance difference signals (U and V).

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HARRIS
next level solutions

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