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db magazine

serving: recording, broadcast and sound contracting fields

Report on DTS

Introducing—A new Church Audio Book!

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World's most frequent flyer!

Sure you'll see other brands hanging around from time to time, but when it comes to self-supporting miniature condensers, the Audio-Technica AT853 series, now in its fourth generation, has been the overwhelming choice of both contractors and end users for years. The reason is simple: It works so well. And for some very important reasons.

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The standard AT853a output level is higher than most others. But the AT853R phantom-only model eliminates the battery compartment and offers up to 3dB more sensitivity. Two more compromises avoided!

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The AT853 series is part of a complete UniPoint system of matched miniature microphones and accessories. The result is consistent response, polar pattern, and output from every microphone in the area. You get the same great sound above the choir, mounted on a lectern, or used to reinforce musical instruments. Sound system operation and EQ are simplified for superior, trouble-free day-to-day results.

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Over the years we've been able to fine-tune every aspect of the AT853 series. You can readily hear the difference engineered into this tiny microphone. It's a difference that spells customer satisfaction and builds your good reputation.

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About the Cover

● Our cover photo this issue is of The Plant, in Sausilito, California. It is but one of many studios that Carl Yanchar's Lakeside Associates has designed and constructed. Carl's *Tek Text* article on designing digital-capable audio studios begins on page 14.

Calend

Late Breaking News

The 1993 Professional Loudspeaker Workshop

● Mark your calendars for October 28th and 29th in Pasadena California Consultants and potential clients will be able to hear many different pro loudspeaker cluster systems and hear which are the best for their respective projects. Loudspeaker clusters of like classifications will be professionally ridged for the demonstrations and TEF analyzers will be used to assure

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AES 95th Convention

AUDIO IN THE AGE OF MULTIMEDIA

October 7–10, 1993

Jacob K. Javits Convention Center • New York

Science emphasizes detached observation, objectivity, and logical deduction, but most who come away from the AES 95th and New York City this fall will find themselves feeling anything but detached—the combined dynamics of this city and this event simply won't allow it.

loudspeakers are properly set up as well as EQd and level balanced. IED and White will assist in providing computer control of equalization,, gain, and routing of source material for the demonstrations. A special feature will the computer controlled random selection of loudspeakers to insure fair and impartial testing.

The manufacturers participating as of press time include: **Excel, Frazier, MacPherson, OAP Audio, PAS, Radian, Ramsa, Renkus-Heinz** and **TOA Electronics.**

Fee for this two day event is: pre-resistration $50.00 (closes Sept 30th) and $350 at the door. Registation includes manuals, continental breakfast, lunch and intermission snacks both days.

Contact for all details: Fax your name and address to (714) **898-3768 or call David Kennedy & Associates at (714) 895-7221 ext. 701.**

● **Sound in Entertainment: Spcecifying and Designing Audio Systems to Create Sound Environments** is the long title of a series of seminars, sessions and exhibits that will take place November 13-15 at the **Orange Country Convention Center** in Orlando, Florida.

The conference is supplemented by the **LDI93** show. Exhibits will be open 10:00 am to 6:00 pm November 13th, 14th, and 15th 1993. LDI93 is sponsored by *Lighting Dimensions* magazine. For information call **Jacqueline Tien** at 212 677-5997 or fax 212 677-3857. You can write them at 135 Fifth Avenue, New York, NY 10010

● **The Record Plant, Hollywood**, California will celebrate its 25th anniversary on September 22. For details on the event, contact **Rick Stevens, Record Plant, 213 993-9300.**

● It's not too early nor too late to start thinking about the **95th AES Convention,** which will be held this year on October 7-10th in New York City. This time however, it will be held at the **Jacob Javits Convention Center.**

"The AES Convention Committee members agreed that the AES exhibit requirements have outgrown the **New York City Hilton Hotel** where AES Conventions have been held for several years,"according tp Convention Chairman **Leorrnard Feldman.** "Our experience at the 93rd AES Convention in San Francisco confirmed the advantage of having all exhibits on a single floor—a benefit that could not be achieved at the Hilton."

The dates include a Saturday stay, offering advantages to attendees for lowest discounted air fares while exhibitors will benefit from weekdays being used for setup and breakdown, thus avoiding overtime charges. According to Len Feldman "the dates of October 7-10 were not available at the Hilton."

For information, contact the AES at 60 East 42 Street, New York, NY 10166, or call at 212 661-8528. db

1993 Editorial Calendar

JAN/FEB *The Electronic Cottage and Project Studios.*

Winter NAMM Show issue.

• GUIDE: Speakers: Performance & Monitor.

MAR/APR Radio and TV Audio/Sound Reinforcement in Permanent Installations.

NSCA & NAB show issue.

• GUIDE: Consoles and Mixers, Work Stations.

July/August *db Visits Houses of Worship.*

• GUIDE: Power Amplifiers.

JULY/AUG Live Sound—Intimate Clubs to Major Stadiums.

• GUIDE: Tape, Tape Recorders and Accessories, Microphones.

SEPT/OCT *State of the Recording Art.*

AES in New York Show issue.

• GUIDE: Signal Processing Equipment, Part I, (delays, reverbs, crossovers, equalizers).

NOV/DEC *Post Production—Coast to Coast.*

SMPTE in LA Show issue.

• GUIDE: Signal Processing Equipment, Part II, (noise gates, noise reduction, limiters, compressors).

NAMM—Jan 15-18 (Anaheim)NAB—April 18-22 (Las Vegas)
NSCA—April 2-4 (Orlando)AES—October 7-10 (New York)
SMPTE—October 30-Nov 2 (LA)NAMM—1994 Jan 21-24 (Anaheim)

JOHN BARILLA

THE ELECTRONIC COTTAGE

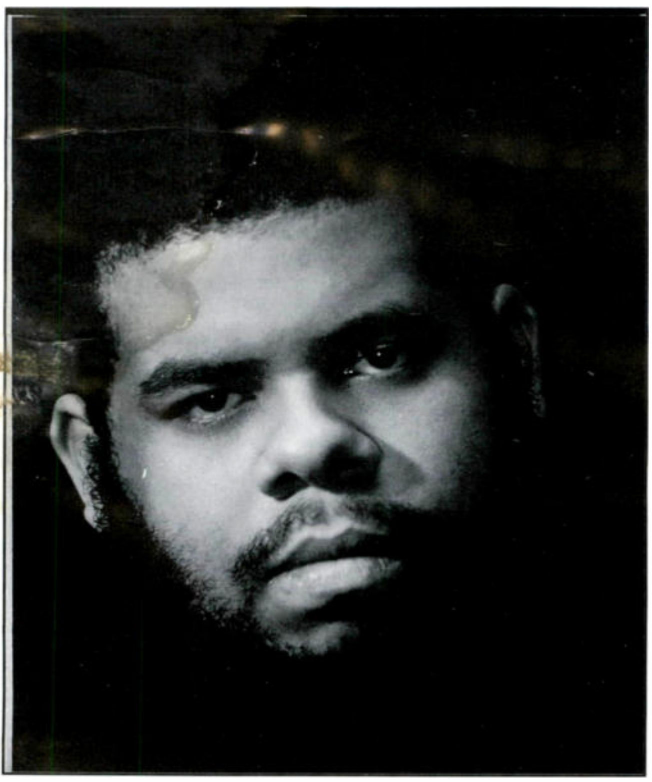
Rap Music And The Electronic Cottage

• Over the past several years we have witnessed the phenomenal rise of the compact computer-based recording studio called the "electronic cottage" or "project studio". Both terms are descriptive of this phenomenon. "Electronic cottage" implies the melding of lifestyle and workspace, whereas "project studio" refers to the artistic liberty that often prevails because recording costs are relatively low. Call it what you will, the advent of affordable high-quality recording gear continues to be a boon to both studio owners and their clients.

Certain styles of music seem as if they were designed to be produced in such studios; in fact, if the technology did not exist, perhaps such musical forms would never have evolved. There is no is better example of technology-driven music than contemporary rap music. While the combination of poetry and music has been in our cultural consciousness since the beatnik period (during the 1950's), today's rap music is really quite different. Rather than the stream-of-consciousness improvisations of the beat poets and musicians, we now have a very focused, message-oriented art form used to express the hopes, fears and frustrations of the urban underclass. Whether you like rap or hate it, no one can really deny the power of this musical form as a means of communication, and the electronic cottage has—from the very beginning—been at the forefront of the rap revolution.

Frankly, it doesn't require a whole lot of equipment to record a rap song; that's probably why it's become such a popular musical form. A drum machine, a sampler and a keyboard are really the basic sound-generating equipment; and the musical arrangement is often just a series of looped passages recorded on a no-frills MIDI sequencer. It's not that rap songs aren't recorded in state-of-the-art facilities (some of them are), but rap music is anti- establishment to the core, and its gut-level impact is self- consciously cultivated. It's probably clear that four years at a musical conservatory will not help you make an authentic rap record. So then, what does it take? Well, the first thing it takes is an authentic rapper. So, let's peak behind the scenes as I show you how I work with a man called *John the Raptist*.

John the Raptist



A VOICE IN THE WILDERNESS

The wilderness we're speaking of here is not the middle-eastern desert in biblical days where John the Baptist cried out against hypocrisy and corruption; we are speaking of the contemporary urban wilderness (in this case, the Bronx) where a New York City Board of Education employee named John Davis was dubbed John the Raptist by the disadvantaged youth he works with. John, who is also a dedicated Christian youth counselor in his church, became known on the Bronx streets as one who spontaneously composed raps, warning teenagers about the dangers of drug abuse, promiscuous sex, street violence and racism, and urging them instead, to embrace the eternal truths found in the Bible.

It was two years ago when John approached me about recording his first album, entitled, *City On Fire*. Currently, he is finishing his second album (*Wolves in Sheep's Clothing*) and touring widely, while continuing to challenge youth with his message.

CAPTURING THE MESSAGE

So how does one capture this message and enhance it with music so that it reaches the intended audience? Frankly, it is largely an interactive process involving artistic dialogue between John the Raptist (as author of the original lyrics) and me (as musician/producer/engineer). I'll share with you a typical production process as an illustration of what it takes to put together a rap song in the studio, but first let's take a look at two radically different approaches to rap production.

One, (by far the most common approach today) is to build upon samples of previously-recorded material. Here, a short phrase will be recorded into a sampler, then edited down to a tight 4 or 8 beat loop. The loop will be triggered by a note on a keyboard. That note is then recorded (at 4 or 8 beat intervals) into a MIDI sequencer whose internal clock has been set to match the tempo of the sampled phrase. Usually, this initially-recorded sequence provides the backbone of the rap song. For example, one might use a passage featuring a rhythm section from an old Motown record or a jazz record which includes bass, drums and guitar. After programming this into the sequencer, other samples may also be added for color. Since these passages were not specifically recorded for the production, but literally "lifted" from other recordings, some additional processing may be needed. For example, on a passage lifted primarily for the bass content one might roll the treble completely off to diminish the impact of higher pitched instruments which are also playing. Some of the residual artifacts (like scratchiness and surface noise on records) may sometimes be seen as desirable attributes if it gives the sample a unique twist.

On top of these samples, an additional drum beat and other instruments may be recorded. During the course of the arrangement, everything might be made to drop out except for the drums (or some other single element in order to highlight a spoken phrase) and then the full arsenal of mutated sound loops is made to resume playing.

It's an interesting process requiring a special "punkish" creative sense and a fondness for techno-manipulation. The downside is that lots of this work starts to sound alike, because people tend to access the same "hot" samples.

Sometimes they engage in a bit of technological warfare by using a snippet of their competition's song and commenting on it in their own production. But such practices are not all good-natured fun. Sampling other people's works has some serious ethical implications that sometimes have to be resolved by a court of law. When the passage lifted is extensive enough to be recognized by an average jurist, the possibility of lawsuit is ever-present. But ethical considerations aside, another reason why rappers do this is that it is an incredibly inexpensive way to produce a record by assimilating first generation creativity into a second generation audio montage.

THE ROAD LESS TRAVELED

There is another way to produce rap music, and while there are fewer adherents in this school, the result is blissfully free from the ethical problems associated with sampling the work of others. That way is to do all original tracks from scratch. Some major rap artists (such as M.C. Hammer) have recently renounced using lifted samples in favor of original tracks), and that is the way John the Raptist and I have decided to also go. There are two main reasons, we prefer this method: one is, of course, ethics. The other is strictly creative.

For us, the creative process goes something like this: John will show up at my studio with a few verses which form a rough lyrical idea for his rap. As the creative interaction goes on, the original idea will be honed into a tight verbal statement several verses long, with a repeating chorus to reinforce the central theme. But to get to that point we usually do quite a bit of live jamming, John modifying his lyrics while I experiment wildly with the music. Typically, John will say, "Gimmee a beat", and I'll manually play something from my drum samples that jibes with the poetry he is chanting. Once we agree on an appropriate basic beat, I simply loop it and let it play over and over again while I try jamming with various other synthesized or sampled sounds. Having decided on an appropriate sound—say synthesized bass—I improvise along with John's rap for a long time, recording every bit of the performance into my computer. If I hit on something that really works with the lyrics and beat I save those passages and loop them, deleting all the others. This process goes on for a third and fourth instrument until we've reached the definitive groove. Sometimes live instruments are in order—say a funky guitar lick. Sometimes, I will defer performing it until later on (after the vocals have been recorded), if it's a simple repeatable pattern, I will simply sample the passage and let my sampler play it back over and over again.

Rather than slavishly follow convention and sample other people's materials, John and I have decided to rely strictly on our own God-given talents.

When the various loops are in place and the kernel of the idea has jelled, we then proceed to do an arrangement of the parts, designating some measures of the song as choruses and some as verses. Perhaps, we will also include what is called a "breakdown" section—where most of the instruments drop out to feature the rap *au natural*. At this point, the form of the song is completed. The process, though, is not as simple as it may sound; at every stage, a creative decision has to be made, and there are lots of live performances to filter through and edit. Typically, it might take two or three sessions to refine the tracks to this point, but now the pieces begin to fall into place much more quickly.

With the previously-programmed musical tracks synchronized to multi-track tape machine, we begin doing live continuous takes of John's rap; maybe 5, 6, or 7 complete performances. Then we sit back and listen to them all. Sometimes one track is brilliant for the majority of the song, but other times the best performances are on several different tracks. So invariably, we end up making a composite rap from the various tracks, compiling the sections onto one master track. At that point, we might add some female background vocals during the choruses—which gives the raps a more universal appeal.

THE VOCALIST

The vocalist of choice happens to be John's wife, Gail, who provides a rich, soulful counterpoint. While the parts she sings are simple and repetitive by design, they provide just the right amount of diversity—enough to please your ears, but not enough to draw attention away from the message of the rap. We usually triple-track these tasty vocal parts and process them (with pitch-shifting and delay) so that they appear like six voices in full stereo. Finally, some songs may call for what I call a "linear track" —a live performance track that changes in response to the various dynamics of the song. An example of this would be a lead guitar part which is played in a "call and response" fashion between the lines of the rap. This is not a looped passage, but something that reflects the real-time development of the music. It adds a realistic dynamic you don't often hear in a typical rap song.

RAPPING IT UP

From what you've just read, you might conclude that what we're doing is a bit unorthodox, and it is. Rather than slavishly follow convention and sample other people's materials, John and I have decided to rely strictly on our own God-given talents. For us, it's more challenging to do all original tracks, and the result sounds much more organic to who we are as an artist/producer team.

Likewise, we feel this approach is much more consistent with our ethical outlook. Come what may, our work will be judged solely on its own merits.

In today's climate, it is indeed the road less traveled, but it is also the one that brings inner peace and artistic satisfaction. db

DAN MOCKENSTURM

Broadcast Audio

Transferring To The Working Medium

● After recording the source material as described in my previous articles, the next step is listening and selecting the "good takes" that are to be used as the final audio—then compiling and transferring those takes to the *work tape*. This process is done in the audio suite where tracks can be listened to (hopefully on quality monitoring) with critical ears and then adjusted as necessary with EQ, volume and sometimes outboard processing devices such as a compressor or de-esser. This should be done *before* it is transferred to the work tape. The work tape format will be selected at this stage of production to suit the final requirements of the project. The two formats that are most asked for are multi-track analog/digital tape or disk recorders. Without going into detail about all the different features and configurations of machines, I will say some things that apply within the formats of analog and digital domain.

On any format it is very important to transfer the audio source to the work tape while maintaining the integrity of the original sound. Care must be taken to be sure recorders are properly aligned and cleaned so they can perform optimally. Good levels must be maintained to keep the signal-to-noise ratio at its best. Any processing should be kept to a minimum, because you can't be sure what you may or may not need in the mix. Remember, you can do some adjustments to a track in the final mix. Some ambiance tracks and sound effects require a combination of sounds pre-mixed before being recorded to the work tape. Care needs to be taken with the balance of the blend. Write down settings and take good notes of how those premixes are made.

ANALOG RECORDERS

Most big projects will require 24 tracks or more to record the dialog, effects, and natural sounds such as ambiance and hard effects. But even smaller projects need clean and informative track sheets, reference tones at the head of the tape and the tape is usually always striped with SMPTE time code on the furthest outside track. The time code type will usually be 1 of 4 types of code, (30 frames non-drop, 30 frames drop frame, 24 or 25 frames per second). This time code is used to synchronize the work tape to the other source machines so the transfer audio has a specific place to be on the work tape. But as you might have guessed, getting that to happen is no easy job. I've heard of people cueing-up the source tape and trying to press *play* and *record* at the same time, but we can be more precise than that today. Using a synchronizer, we can set an offset that is the difference of the two time-code locate points and by controlling the two machines from the synchronizer, events can be placed exactly in sync on the work tape. Analog tape is also commonly used with noise-reduction systems. Because of the inherent noise of tape, there have been some great inventions of noise reductions units such as Dolby SR, but again, you have to be careful about the set-up of these units.

Now you can see why field recording is so important; we are now going to 2nd generation audio on the work tape (unless you are using digital).

DIGITAL RECORDERS

This format can allow you to transfer your field DAT audio to the work tape through the digital domain and not lose a generation. The procedure is the same as transferring a file from one computer to another. It's great for transferring

Circle 21 on Reader Service Card

sound effects or music from CD libraries. But if you need to pre-mix sounds or process them with outboard, you will need to go through an analog console and then to the digital work tape.

The *digital workstation* allows other advantages. Since the audio is recorded to the unit's hard drives, the sound file is random access and can be played at anytime when "triggered," the same sound can be triggered multiple times. This makes working with sound effects a lot of fun because it's just a matter of placing the effect in sync rather than placing and recording it each time.

Just as with analog tape, you still need to keep things in sync and that again makes it important to use your reference to time code. As the director/producer decides which takes they are going to use, it is important to record that take to the work tape in sync. Sometimes these decisions are made is a video edit session. When it is done this way, the video editor will keep a list of what takes it is using and a list of where those takes are to be placed on the work tape. This list is called an EDL or Edit Decision List. This list can be printed out and used as a reference for assembling the audio. But again, in these days of computers, that list can be copied onto a disk and load into an audio computer that is controlling a video machine—this process can be automated. What happens is the computer reads the list, figures out the location point of the first piece of source audio by time code and reel number, then controls the machine to locate to that position, puts the record machine into *record* and out, then repeats the procedure for the next piece of audio. After all the source audio is recorded, the program will then align all the events and assemble a completed track. All you need to do now is add effects—that will be the topic of the next article.

TIMECODE

This is the most important part of the transfer process, because it is the reference point for keeping everything in time or in sync with the picture. As I mentioned before, there are different types of time code and it is very important to use the same type throughout the entire project. Different time codes do not work together. I would strongly suggest reading and learning more about SMPTE time code and synchronizing.

To stripe a tape with SMPTE, first select the type of code to be used, determine if you need an offset, (usually time code starts at 1:00:00:00 on the first reel—depending on the project), set a level of –20 to –10 dB and record it to the outside track of your multi-track. (track 24 on a 24 track machine). The source of the time code is usually a device which can generate SMPTE time code such as a synchronizer, most video machines, and some computer systems' software.

To synchronize two or more machines together select one machine as the "master" and the other machine(s) will be "slaves". This means that the slaves will be constantly referenced to the master tape's time code and speed-up or slow-down to stay in sync with the master. To do this, connect the output of the master time-code channel either directly into the time code input of the slave, or to the input of a synchronizer which can then distribute the time code to multiple machines. A synchronizer can also manipulate the code and do things such as reshaping the signal if it is at a poor level, set offsets, and help in other problem areas as well.

CONCLUSION

I hope you're getting a feel for what it takes to do a complete audio production for TV or video, and I can tell you that a *film* is even harder. There are always little problems that come up, it is our job as engineers to be creative and knowledgeable enough to figure out solutions to these problems. The more you know, the better off you'll be. I think these articles also show some of you that there are different areas of specialization needed for production, and to specialize in a particular area can be very beneficial to a good stable job. Not everyone is cut-out to be a mix engineer. Some are better at working in field recording, sound design, effects or dialog editors. Explore the possibilities and try to find what works best for you. Till next time—Happy Audio!

NEW PRODUCTS

DUAL BASS ALIGNMENT

• Two bass alignments from one speaker can be achieved by the use of a port plug that can be pulled out or left on depending on the preference for that particular mix. The "b" series can then be acoustic suspension or bass reflex. There is also woofer and tweeter adjustments, a three level position switch on the tweeter and woofer to emulate both studio and audiophile playback response curves. Speakers are bi-wired with gold plated 5-way binding posts.
Mfr: Digital Designs
Price: $602.00—DD161-b;
$830.00—DD261-b
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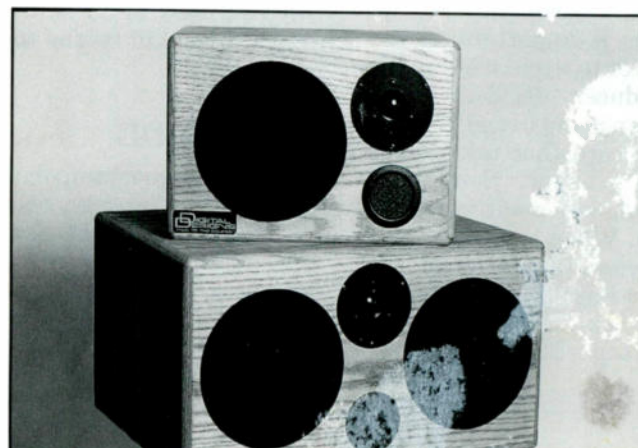

COLOR MONITOR

• This new 12-in super VGA color monitor uses an analog input for a color display palette that is virtually unlimited. The monitor will display VGA as well and super VGA 1024 x 768 resolution. they can be built in, rack mounted or free standing. Full metal industrial, open frame, or desktop tilt and swivel enclosures are available.
Mfr: Modgraph, Inc.
Price $750.00 quantity discounts available
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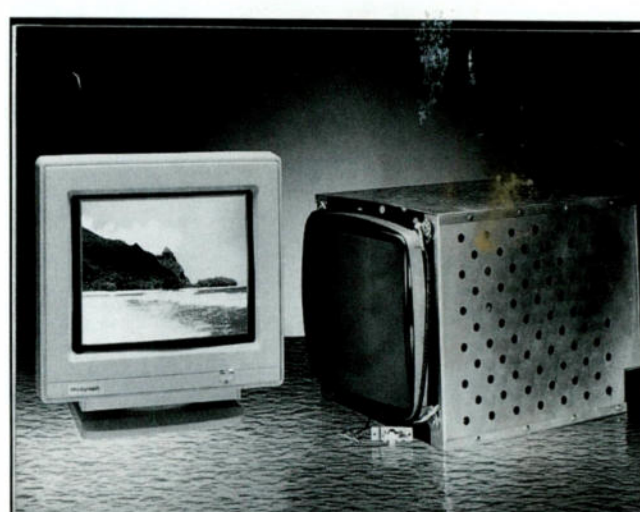

ANTENNA SPLITTER

• AD-200 makes it possible to operate up to four wireless diversity microphone receivers (as many eight antennas) using only two antennas. It also provides a high degree of output isolation, a critical feature with prevents intermodulation in multi-frequency systems. In addition, it features a newly designed High Q front-end filter design, along with low noise amplifiers, which minimize insertion loss while providing overall unity gain. A green LED on the front panel indicates power on, and a rocker type switch controls power to the splitter and receivers. The unit has an external AC power supply for 110V operation. Eight 2-in coax antenna cables are included, and power out cables are provided for up ... rs to eliminate excess power supply clutter. It covers the 150-235 MHz frequency range, and is compatible with any receiver which uses standard 50-ohm antennas and connectors.

Mfr: Telex Communications
Price: $ 499.00
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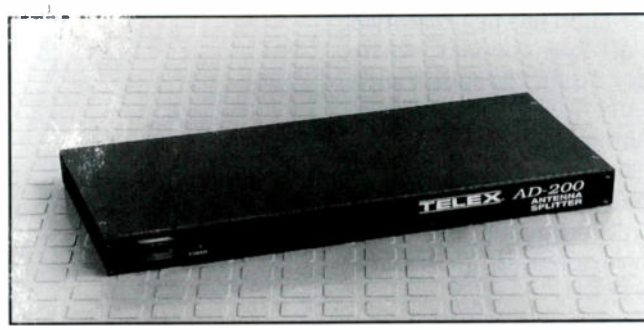
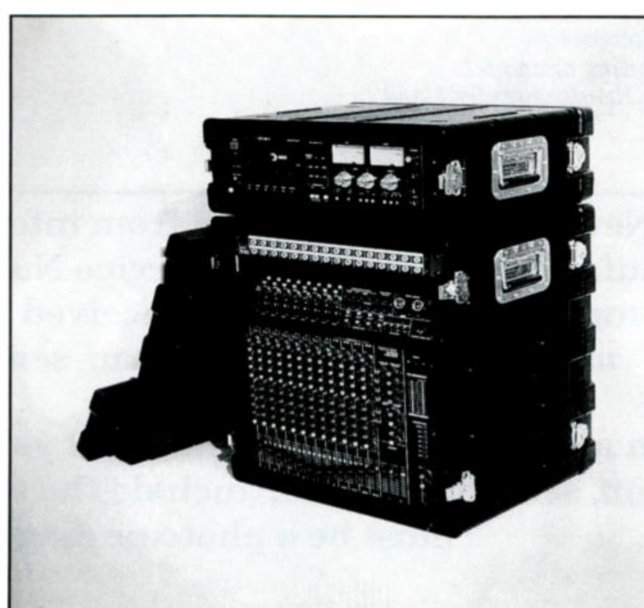


EXPANDABLE CASES

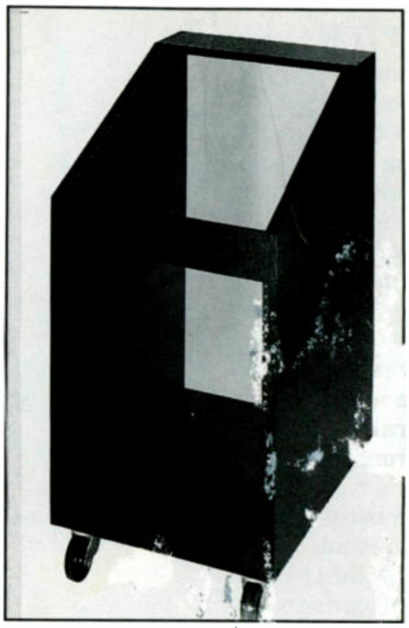
• EMS (Expandable Multi Rack Systems) are expandable, transportable and modular rackmount cases. Modules are purchased with a 3-rack space case height. Incrementally, the user removes a prescribed number of screws that in turn detaches the top or bottom of the case. The case then expands in 3-rack space increments by adding two each-side expanding modules and two each-front lid expansion modules. Additional recessed, spring-loaded steel catches and handles are also available to facilitate the need for larger rackmount configurations. Other features and options include stacking feet, reinforced corners, horizontal component installation bars and front, well as rear, rackmount rails. Ample ventilation is provided by mounting-in side vents that double as spaces to mount recessed spring-loaded heavy-duty handles.

Mfr: Star Case Manufacturing
Price: depending on modules
Circle 63 on Reader Service Card



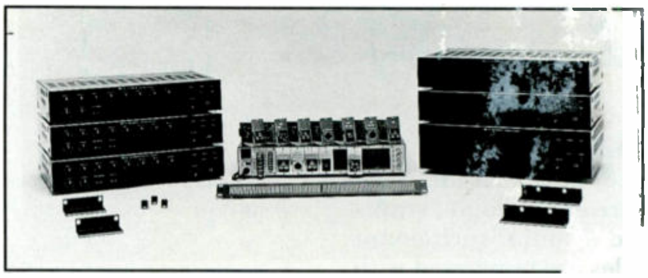
HOME STUDIO RACK

• Model LRK is ideally suited for reel-to-reel tape machines, mixers, and a variety of other components. Ruggedly constructed using ⅝-in. thick black-laminated high-density particle board and has pre-installed casters at all four corners, while high-quality 10-32 threaded rack rails (also pre-installed) are supplied for mounting purposes. Measuring 27½-in deep by 31 ¾-in high, th unit shops flat, and comes complete with all hardware, including an Allen wrench and decorative caps for the bolts.
Mfr: Middle Atlantic Products
Price: $286.67
Circle 64 on Reader Service Card



AMPLIFIER SERIES

• Seven new models comprise the 900 Mark 2 Series of commercial amplifiers. External improvements from the original Mark 900 include a new black finish an reduced rack height. While the units have become smaller, the features list has grown. Now there are eight available inputs, and a terminal strip for remote mater volume control. Two levels of mount priority have also been including the unit's design to produce added versatility, while all 37 existing 900 Series input modules will work with the 900 Mark 2 Series, and vice versa.
Mfr: TOA Electronics
Price: depending on model
Circle 65 on Reader Service Card

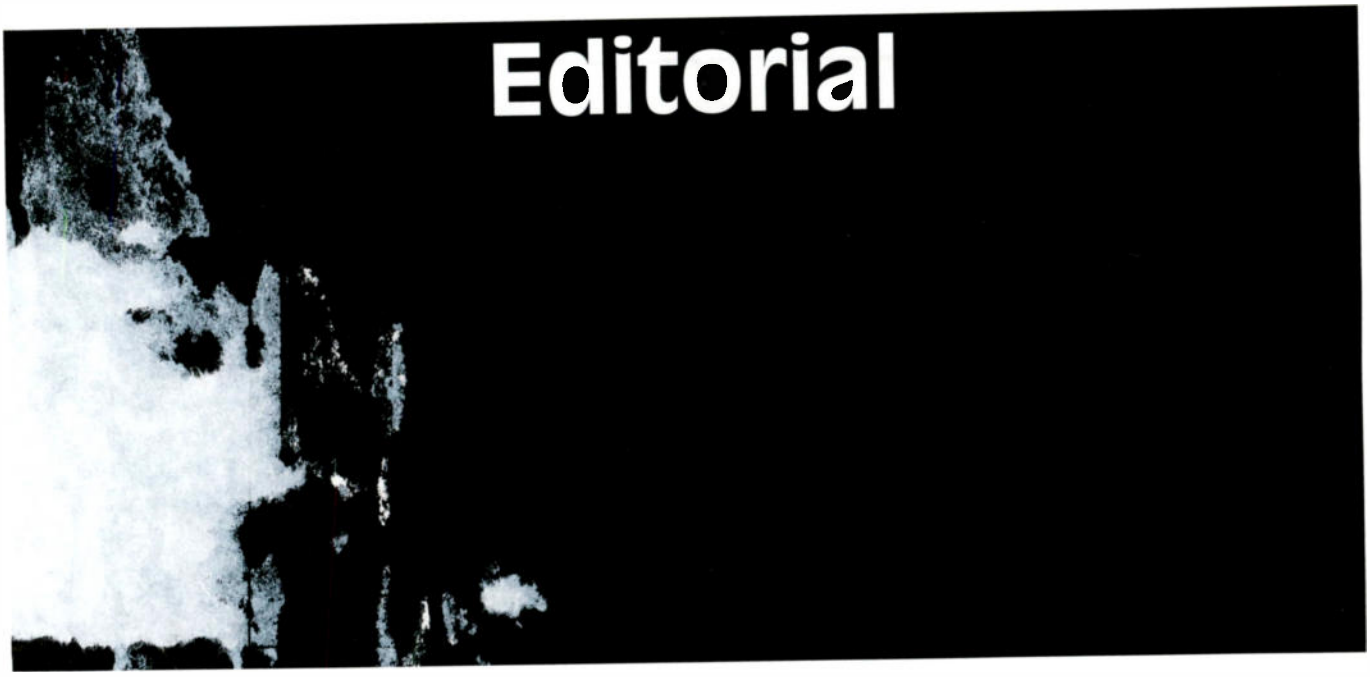


New Products are edited from information supplied by the manufacturer. The Reader Service Number under each product is computer processed when received and sent to the respective manufacturers who, in turn, send out the information.

If you are a manufacturer and want your new product listed in this section, send the release, include the suggested list price and there must be a photo or diagram included.

Send to New Products Department, db Magazine, 203 Commack Road, Suite 1010, Commack NY 11725.

Editorial



Digital audio continues to move in more directions than anyone can guess.

There have, for example, been several attempts to bring digital audio into the film theater market, all have gone. One, however, is trying again, and is likely to succeed. DTS (Digital Theater Sound) has come up with a practical system using CD-ROM as the digital audio medium; it's a placing of a proprietary time code on the film to drive the digital audio signal on the CD-ROM with the filmed picture, and in perfect sync. Tests, including cutting the time-coded film in several places did not undue the perfect sync.

This is because the film also carries two-channels of optical analog audio, and the system is smart enough to know when to switch and switch back. A theater equipped with the DTS system will play digital sound in the theater, a theater without this equipment, plays the optical tracks.

All this is by way of an article in this issue on page 57 by Contributing Editor Shelley Herman. It details this new system, but also, in Shelley's inimitable fashion, gives you an overview of where theater sound has come from.

Church Audio is an arm of sound reinforcement in general, but a specialized one. Since we now have a sizable readership employed in this discipline, we have been on the lookout for editorial material of value. We were recently contacted by a Canadian writer,Joseph De Buglio, who have created a book of Church Audio. On page 20, we present one chapter that will give you a feel for what this author is trying to say. Watch these pages for further information on this new book.

If you are planning to build a recording studio, large or small, be sure to read Carl Yanchar's article on this subject. You will learn a lot about this subject from Carl. All the illustrations only touch on the studios that he has already designed (and built). See page 14.

John Barilla details the special skills and talent necessary to produce a good rap recording, and Dan Mockensturm returns to our pages with insights into broadcast audio. LZ

CARL J. YANCHAR

Designing Digital-Capable Recording, Broadcast And Production Facilities

During the past decade, the development and advance of recording ... nology has cha... ...esign specifications of recording, broadcast and production facilities.

THE GROWING IMPORTANCE OF Stereo Television, consumer CD release, video rentals, home surround-sound systems, and exotic automobile sound systems have raised the standard by which commercial audio product is compared.

Nowhere is this advance more obvious than within the rapidly advancing area of television production. The small, three-inch internal speaker of yesteryear _ which filtered out hum and buzz, minimized the impact of distortion, excessive compression and equalization, level differences and the effects of poor mic placement _ has been replaced in newer receivers. Improved receiver design also allow easier connection to a home entertainment system. Many consumers now consider that improved audio quality enhances the perception of an improved picture quality. In audio, the standard is no longer "broadcast quality" but rather "CD quality."

Newer high-definition (HDTV) or advanced television will also provide improved picture quality and the capability of "CD quality" sound transmission. Yet the benefits of bigger and better pictures will be only fully realized with bigger and better sound. Many of these improvements can be implemented today, via relatively simple electronic upgrades, such as replacing analog cart machines with a DAT player, digital cart machine or hard disk-based workstation.

Other improvements can be accomplished with modified production techniques, and quality control measures. Greater attention is now being placed on maintaining level, loudness and phase uniformity; eliminating excessive equalization or compression; ensuring correct alignment of analog tape machines and Dolby noise-reduction processors; and ensuring a system interface that is free from hum and noise.

The extension of the usable lower-frequency limits, and the expansion of available dynamic range to 90 dB and beyond, has placed even more stringent demands on a control room's monitoring system, as well as requiring an acoustic environment of equivalent capability.

As video production and post-production facilities are constructed or upgraded to accommodate HDTV with stereo/surround sound, attention is be placed on providing an acoustical environment that does not hinder production, and one in which quality judgments can be made accurately.

TECHNIQUES FOR IMPROVED AUDIO MONITORING

At all locations where quality judgments are to be made, the audio monitoring system must be equal to the task. The system should be linear in response, and capable of accurately reproducing lower octaves. The best way to accomplish this task is to start with a linear monitor, with dispersion selected to complement the room geometry and use. In most cases, some degree of high-frequency rolloff is preferred to a ruler flat response to beyond 20 kHz.

For an audio control room, the physical conflicts of a dedicated center audio monitor for surround mixing, a video monitor mounted at a reasonable height, and a panoramic window into the studio, all present physical challenges. Dolby Laboratories recommend that for surround-sound mixing, he center monitor match both the left and right, and be placed at the same height for optimum imaging. The optimum viewing distance for video displays is given in Reference #1.

The main left and right speakers should be, and usually are, placed symmetrically about the main video monitor. In most cases, the main left and right speakers will be mounted at or near one or more of the room corners. At low frequencies most loudspeakers radiate over a 360-degree angle. Some

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Carl J. Yanchar is the president of Lakeside Associates of Irvine, California.They are studio facility designers, acoustic consultants and handle fabrication and construction.

of tne low-frequency energy will be reflected back into the room, but sligh ly delayed. Because of the long wavelengths involved, most of this delayed energy will be in-pha Uniess the speaker is designed with this type of mounting, low-frequency response will be boosted. ertain frequencies, how- dependent upon the exact ge- ill be canceled. rs when the udistant from s well as the

ng studios ent this prob- ating the audio ers, a design force the frequency en- y to radiate over a 180-degree angle rather than 360 degrees, thereby boosting the low frequencies more uniformly. Some loudspeakers have internal equalization designed to compensate for this alternate mounting method. If not, external equalization should be incorporated; a 6 dB boost in the low-frequencies does not allow for quality monitoring!

Because this rating system was devised primarily for residential and office applications, data for frequency bands below 125 Hz is not always available, nor is it reliable.

In either case, when using audio monitors with extended bass response, it's important that they be de coupled from the structure of the room. This precaution is necessary not only to prevent rattles, but also to maintain good stereo imaging. Sound travels faster through most structural materials than it does in air, meaning that the structure borne sound will arrive before the direct sound.

ENHANCED SOUND ISOLATION

All recording, production and post-production facilities require an appropriate degree of sound isolation from the outside world. For the single-room facility, its location is usually the determining factor. Proximity to external fixed as well as transportation noise sources will establish the degree of isolation required.

When choosing a site, evaluate every possible condition, because external noise levels vary greatly from day to evening. Truck traffic, railroad and manufacturing plant schedules may be quite different on weekends than on weekdays.

NC Curve	63 Hz	125 Hz	.250 Hz	500 Hz.	1 kHz.	.2 kHz	4 kHz	8 kHz.
NC-50	71	64	59	54	51	49	48	47
NC-45	67	60	54	49	46	44	43	42
NC-40	64	57	51	45	41	39	38	37
NC-35	60	53	46	40	36	34	33	32
NC-30	57	48	41	35	31	29	28	27
NC-25	54	45	38	31	27	24	22	21
NC-20	51	41	33	36	22	19	17	16
NC-15	47	36	29	22	17	14	12	11

Table 1: Sound Pressure Level (dB) for various Noise Criteria curves.

Air traffic patterns can be altered due to weather.

In a multi-room facility, the unavoidable proximity to high-level, low-frequency sources demands a level of isolation which, in most instances, exceeds that of any potential external sources. Rooms for different applications require different degrees of isolation. A Foley room, or voice-narration studio where live sound is being recorded, require a lower noise floor than a post-production room in which external noise is more a distraction. The distinction is extremely important, because the cost of each increment of isolation can increase construction costs geometrically.

A conventional method of specifying noise-level requirements in a room is the NC (Noise Criteria) curve. Data for NC curves was gathered from surveys of the effects of environmental noise on the ability of office workers to perform work and to communicate via speech. Such curves were developed for the purpose of providing criteria for reducing complaints to tolerable levels. They have been adapted by the building industry as a simplified method of communicating the allowable noise levels in a room [2],[3].

NC 15-20 used to be the standard for studios and concert halls. Digital recording and the use of Dolby SR noise reduction has lowered the standard goal to NC 5-10, and even below. Achieving this low level of noise is extremely costly, however; each 5-point reduction in noise criteria translates to a minimum 10-15% increase in construction costs [4].

At NC-20, the allowable noise level at 63 Hz is 51 dB SPL. While this value is less than ideal, for some music and most voice recording situations such an SPL may not be an insurmountable problem if, for example, the use of a high pass filter is tolerable. For Foley work, especially when the sounds being recorded fall in this frequency range and are at or below the background level, a filter will not help. And, when several channels are summed, the situation becomes further aggravated.

The human ear is less sensitive to low-frequency sound. Most microphones, however, are designed to exhibit a flat frequency response. While passing through an electronic chain, any rumble picked up can increase distortion and, in extreme cases, even cause amplifier clipping.

The degree of sound isolation actually achieved is dependent upon three factors:

1) The mass, stiffness and damping of the enclosing walls, ceiling and floor;

2) The airtight sealing of all

Figure 1. The Audio control room #2 at KRCA-TV, Burbank, located back-to-back of its companion Video Control room, is equipped with a 24-input Neve Series 55 mixing console.

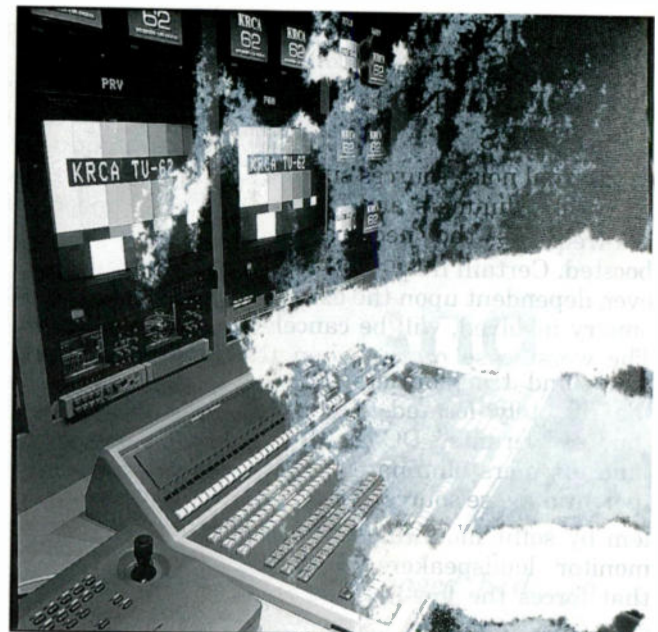

Figure 2. Production C.ntrol Room #.. TV, quipped with a Grass Valley M... 250 switcher and Pinnacle PRISM DVE system

penetrations for doors, windows, cables, and air ducts; and

3) The physical separation from internal and external noise sources [5].

The first two are summarized in the STC (sound transmission class) rating system adopted by the building industry as a simplified means of comparing various barrier constructions. Because this rating system was devised primarily for residential and office applications, data for frequency bands below 125 Hz is not always available, nor is it reliable.

At low frequencies, wall, floor and ceiling transmission loss become much more complex to predict. The transmission loss is no longer governed only by mass, but such additional factors as damping, stiffness and panel dimensions. Materials such as concrete and concrete block perform well in the low-frequency range, by virtue of their combined mass and stiffness. Theoretically, STC increases by 6 dB by each doubling of mass, a factor that works to good advantage at first, although the benefits quickly diminish.

At 2,243 kg/cubic meter (140 pounds/cubic foot), a 152 mm (6-in) thick 3.66 m (12-ft) high wall will weigh approximately 1,250 kg/m (840 pounds/ft). This is not a severe problem on grade, but on the upper floors of an existing highrise office building could present some interesting structural challenges.

APPLICATIONS OF FLOATING FLOORS

In situations where increasing STC by adding mass quickly reaches its practical limit, the only other practical method of increasing sound isolation is physical isolation. Floating or "room-within-a-room" construction is usually necessary to prevent vibration and structure-borne sound from entering the room. Ease of access into the room and the law of gravity limits the physical separation of a floated floor from the structural floor.

Internal noise sources such as lights, dimmers and fans as well as the necessary penetrations of the enclosure...

There are three basic types of floated floors. The lowest cost—and least effective—is the use of a continuous underlayment. A continuous sheet of neoprene, fiberboard, or proprietary materials is laid down, then covered with building paper or polyethylene, and finally concrete poured on top. Because of the limited static deflection, this type of floated floor is effective only in the mid- and high-frequency ranges.

The second type of floating floor employs a neoprene or coated fiberglass fixed mount, placed at 305 mm (12-in) to 610 mm (24-in) centers and covered with plywood and polyethylene. Concrete is then poured over this form. A STC of 73 can be achieved using this method.

For extreme low-frequency vibration isolation, spring mounts are required. Although fixed springs have been used, the most common system is the raised slab system. Housed metal springs are placed on 0.91 to 1.22 m (3-4 foot) centers with an integral steel reinforcing grid. Concrete is poured over the mounts and grid. After the concrete has cured sufficiently, about 30 days, the slab is raised by a process of slowly turning jack screws built into the mounts. Isolation down to sub-sonic frequencies can be obtained and STC ratings of 82 and greater are possible.

REDUCING THE EFFECTS OF INSIDE SOUND SOURCES

Internal noise sources such as lights, dimmers and fans as well as the necessary [illegible] of the enclosure [illegible] [illegible] las[illegible] can [illegible] When [illegible] lights [illegible] this [illegible] as well [illegible] reducing the [illegible] elec- [illegible] and conduits provide very efficient trans- [illegible] for sound, and [illegible] be sealed airtight, yet [illegible] able for future changes. A good solution, especially wh[illegible] cable trays must be used, are products such as Crouse Hinds' "Thru-Wall Barrier." These devices consist of mounting frames of various sizes and elastomeric sealing blocks that form a tight seal around cables and conduits.

QUIET AIR CONDITIONING

Excluding doors and windows, air conditioning ductwork represents the most significant penetration of an enclosure. Careful sealing and structural de coupling of these penetrations are crucial. Even small cracks in partitions should be eliminated [6],[7].

Air conditioning noise, especially low-frequency noise, presents one of the most difficult challenges to providing a low noise floor. Conventional techniques such as fibrous duct lining and passive silencers are effective above 250 Hz. A recent study has suggested that placing the duct lining in a manner that the fibers are normal to the duct axis can nearly double the attenuation at low-frequencies, compared to that of a conventional orientation [[8].

Passive silencers of sufficient length to control low-frequency noise place severe static pressure restrictions on the air handling units. Fortunately, active noise control systems are becoming a viable solution to solving low-frequency attenuation problems. First patented in 1934, active noise control systems consist of a microphone that detects the noise as it propagates down the duct. A DSP controller processes this signal, determines a canceling waveform and introduces this signal through a loudspeaker. A second microphone located just beyond the speaker provides error correction feedback. Attenuation is 12-20 dB between 40 Hz and 160 Hz [9].

Independent duct systems for each critical room are necessary. Whether sheet metal, rigid fiberglass, or flexible, the ductwork should be routed and isolated so that it does not pick up any noise along its path or couple sound from one room to another. The ductwork should not generate through vibration any noise of its own.

If not properly braced and damped, sheet metal duct, although offering higher transmission loss than rigid fiberglass or flexible duct, can generate popping noises when it is pressurized or de pressurized. Square or rectangular duct is more susceptible to producing aerodynamically generated noise. Round or flat oval sheet metal duct minimizes both of these problems. Flexible duct has very low transmission loss, but can generate crackling noises as it expands and contracts when the fan is turned on and off. Very often flexible duct becomes pinched, restricting airflow and generating noise. On the other hand, it is very economical and does not transmit vibration as well as the other types.

To introduce the air into the room without adding noise requires a low outlet velocity. To attain NC 15, an outlet velocity of no greater than 250 ft/min is required. Consequently, a large outlet area is required. Dampers within 6m (20 ft) of an outlet should be avoided as they reduce the net area of the duct and therefore increase noise. Grilles and diffusers must be sized so as not to restrict air flow.

Figure 3. The Audio Post Room at KRCA-TV for mix-to-picture, post-production and is equipped with a 36-input Sony MXP-3000 console.

Figure 4. Studio A control room at The Plant, Sausalito, SSL-equipped.

FINE TUNING THE ENVIRONMENT

Attention to detail is important in eliminating other potential internal noise sources. Any device with a fan should be relocated to a non-critical room if possible. Devices which must remain in the room should be treated on an individual basis to minimize their noise contribution. Light fixtures, air grilles and registers, and console and rack panels can be set into vibration at particular frequencies. These offenders can be easily identified with a sweep oscillator and damped with neoprene or foam.

In the studio, script stands or tables should also be investigated for resonance and covered with an absorbent material such as heavy felt or carpet. Chairs should be selected for quietness as well as comfort. Storage cabinets should be built so as not to rattle or resonate and prevent any items stored in them from doing so as well. Again, the best solution is to remove potential problem items from the room if they are not required.

REVERBERATION AND SOUND ABSORPTION

A reverberant room will be noisier than a non-reverberant room of the same volume. Why? Simply because sound is not absorbed when it strikes a boundary but is merely reflected back.

Unlike music-recording studios, where room character and moderate reverberation is desirable, voice-over or Foley recording areas should be as transparent as possible. Voice-over, ADR and Foley studios for video are usually very small. Because of their low volume, traditional reverberation time calculations are not meaningful. However, any reflected energy should be made diffuse and of essentially uniform frequency content.

Reflection room resonance can be controlled by the addition of absorption with porous materials, diaphragms and resonators, geometric reflection control and diffusers. All of these techniques are to varying degrees non-linear with frequency. Porous absorbers become ineffective below 250-500 Hz, depending upon their thickness. Diaphragmatic and resonant absorbers are by design frequency sensitive. Most reflective surfaces become absorptive, usually with increasing frequency, and at some point can become diaphragmatic as well. Diffusive surfaces not only vary with frequency but also can be dependent upon orientation. They also usually become absorptive and diaphragmatic at certain frequencies.

At frequencies below 250 Hz, conventional techniques require a depth of porous material or a cavity equal to one quarter the wavelength of the lowest frequency to be absorbed. At 20 Hz this is over 4.27 m (14 ft); even at 40 Hz it is approximately 2.13 m (7 ft). A cavity of this size consumes a lot of real estate but, for a critical listening environment, the improvement in low-frequency transient response is worthwhile.

Some general contractors offer pre-construction services, such as value engineering, that can save time and money, and minimize surprises and headaches later on in the project...

Recent work at the U. S. Army Construction Engineering Research Lab has developed low frequency absorbers with normal incidence absorption coefficients approaching 1 with a thickness less than 10% of a wavelength [10].

These are a few of the areas that should be given attention when constructing or remodeling a production or post-production room for critical audio recording or monitoring. Although fine tuning can sometimes be done after construction, appropriate isolation must be built in. Correcting problems after the fact will always be more costly than designing it right from the beginning.

Figure 5.. The remodeled control room suite for Minnesota Sound Gallery's Studio A.

SIDEBAR 1
Evaluation of an Existing Room or Proposed Environment

Before buying [illegible] facility, a prospective owner should minimize their investment risk by conducting an initial walk-through, preferably together with the architect, designer, and contractor. The following are some of the items that should be assessed:

- Control Room Shell of between 55 and 85 square meters (600-900 square feet).
- Studio Shell as required (55 cubic meter/2,000 cubic feet minimum).
- Support Space as required.
- Room for expansion.
- Clear Height of 4.5 m (15 ft) minimum.
- Column spacing 7.5 m (25 ft) or greater.
- Floor loading capability of 730-975 kg/square meter (150-200 pounds/square foot).
- Existing mechanical system capacity (heating, ventilation, air-conditioning).
- Electrical service; ease of upgrade; power conditioning required.
- Structural system and exterior wall composition.
- Ability of roof to accept additional loads for isolation construction and HVAC.
- Roof Condition.
- Air traffic paths.
- Railroads.
- Automobile, truck, motorcycle.
- Road condition (potholes).
- Soil type.
- Proximity to TV & radio transmitters.

On the basis of this on-site research and other information, the prospective owner should be able to determine the suitability of the existing structure. Some idea of the extent of changes required to transition from the existing structure to the new use should also be developed. Assess the scope of these changes as they affect both the interior and exterior of the building. Examples of some specific considerations would include zoning requirements, building codes, required upgrades to an older building, disabled access, additional parking, earthquake or wind reinforcement and fire codes.

[illegible] tractors offer pr[illegible] rvices, such as value engineering, that can save time and money, and minimize surprises and headaches later on in the project.

SIDEBAR 2
Grounding Schemes for Multi-Room Facilities

From the standpoint of technical power and signal interfacing , a single-room facility is relatively straightforward. However, all technical systems take on an added dimension of complexity when designing the multi-room facility. Inattention to detail can result in facilities where individual rooms function internally, but interface between more than one room may be erratic, or even impossible. Such problems will severely limit the usefulness of the facility.

Paramount among considerations is a well planned and scrupulously executed grounding scheme. For AC power, two grounding concepts predominate:

1) A true "Star Ground" with every outlet having a separate ground wire pulled back to the central technical ground point, and;

2) A devolved ground, in which each major room has a central ground point, and these points connect to the central technical ground for the entire facility.

The true Star Ground is by far the more effective method. The third-pin ground wires from all outlets should be brought to a bus bar connected to the central technical ground point. All receptacles must be the isolated-ground type, usually identifiable by their orange color or green triangle. All technical power should be completely shielded in steel conduit and raceways. Romex should *never* be used. However, conduits must be connected to ground and must not be mechanically connected to conduits for any other power system, or to anything else metallic such as water or sprinkler systems.

Technical power should never be used for any other function such as photocopiers, kitchen equipment, air-conditioning, or any non-production equipment. Only when all these conditions exist can an interface system be developed which is consistent and trouble free.

All equipment should have only one connection to the ground system. The most convenient ground connection is the third wire (usually green) in the AC power cord. According to the National Electrical Code (NEC) and other codes, this wire *must* be used for safety. Some studio operators prefer to use a separate ground wire to each piece of equipment. This approach is not usually necessary, and is very inconvenient in cases where equipment has to move from room to room.

Obviously, if you leave the AC cord ground wire intact to avoid ground loops, any other signal connection between equipment must *not* complete a ground connection. In other words, all signal shields must connect *at one end only*. Various methods exist for achieving this goal, including complex schemes involving shields cut just about everywhere and bussed grounds on patchbays.

Most of these ground schemes can be made to work as long as the cardinal rule of grounding is applied: *Do it Consistently*. However, any scheme involving ground bussing can create different ground "nodes," with varying impedance to true technical ground. In a multi-room installation, this can create certain patches or equipment configurations that never seem to work totally hum and buzz free.

The simplest shielding method is to carry all shields through any interconnects and patchbay normals, and lift the shield at one end. The most prevalent choice is to connect any shield at its source and lift it at its destination, directly at the equipment and *not* at an intermediate connector or patchbay.

The majority of equipment features some type of captive cabling, allowing easy disconnect and establishing a connector "standard" such as XLR to "Elco" for multitrack machines. Thus, all input ground lifting is done in this equipment specific cable and not in any truck or tie lines. This greatly simplifies installation as all bulk wiring is done in one simple way, with all shields intact.

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JOSEPH DE BUGLIO

Nuts and Bolts and More Tips

This article is Chapter 4 in a new book "Why is Church Sound So Confusing?"by Joseph De Buglio. Watch these pages for information on the complete book. It will shortly prove to be the complete word on church audio installations.

WHEN TO START TRAINING?

Before we can suggest a sound system design, we must look at the larger picture. It is one thing to give a church a sound system that finally does what it is supposed to do, it is another thing to know how to use the system to enhance the worship, rather than degrade it.

The operation of a church sound system is a team effort. That team includes the Minister, the Organist, Pianist, Choir Director, Song leader, Soloist, Musician, Lay readers, Guest Speakers, Guest Soloists, Guest Musicians, Children, Teen-agers, Lay members, Ushers, Deacons, Elders and of course, the sound person. (Some women have proven to be the best sound system operators in North America.) Every person in a church needs to be taught how to use a sound system, what to expect of it and how to get the same high quality results every time.

We are now approaching the third generation of churches with audio and it seems that 98 percent of the people haven't a clue about what they are doing. The purpose of the sound system is to have the people in the pews intimately involved with every part of the worship.

Example: In some churches there are lay people who pray out loud during a planned part of the service. Often, they will simply pray out loud from where ever they happen to be standing. In small churches this works just fine but in larger churches, a person's voice needs to be amplified. It is interesting to see what happens when a microphone is put in front of most people. Give a person a microphone and their automatic volume control turns down. Furthermore, they often hold the mic so far away they give the impression that the mic has teeth. Sometimes they hold the mic as if it has the pickup capsule in its side. Now, if you combined these two responses, you may conclude, as I often do, "what that person has to say must be so personal they don't want to share it with others." The only reason for these responses is lack of education. People have to be given permission to speak out with the equipment they have. Would you build a church to protect you from the elements and then have all of your services out of doors because you're afraid to wear out the building!

The worst offenders of not using a microphone properly are ministers. Most ministers act as if the microphone is a hindrance or an obstacle that well meaning people put in front of them. Considering what Bible Colleges teach and what those institutions own, it is no wonder ministers are afraid of using audio equipment. A poor system in a church literally drives people away. All the mic techniques in the world will not help.

However, in a proper sound system, the microphone becomes an extension of your voice so everyone can participate. If you treated a microphone like an ear of a very close friend, you would be well on your way to taking advantage of your ministry.

HOW TO USE A MICROPHONE

The way this works is very simple. If you were speaking softly, as you would with a close friend with whom you have an important message to share, you move closer. This means getting within 4 to 12 inches from a mic. With a good system this is whispering. When you are talking as to a friend across a living room with general comments, you should be 9 to 20 inches from the microphone. When you have important statements or a special point on which you want to raise your voice, you should be 15 to 36 inches from the microphone.

The reason for using this method is straight forward. When your voice is low, there is a lot of detail that people often miss. This detail is needed because it's not what was said but how you said it that will carry the largest impact. Your words of comfort are meaningless if people only hear 30 percent of them. Therefore, getting closer to the mic allows people to fully understand the importance of those supporting words. Likewise, when you are excited, you need to move away from the microphone.

When you speak louder, you will blow more air. As you blow more air, you will pop the mic with strong bass sound. This is annoying, distracting and easily avoidable. If you are able to move away from the mic (if you are hand holding it, it works the best), you will also keep the volume down while getting your message across more effectively.

Getting louder often does not mean getting clearer. In fact, as you increase the volume of a sound system, the room starts to fight back. If the acoustics are good, the sound can be louder but, as in most churches, there is a narrow window at which the volume can be set and it is up to the person speaking or performing to know the limits. It is insulting to everyone when a sound system is so loud you can not understand what the minister is saying.

There are many churches today that are abusing our hearing. Did you know the average sound pressure level of congregational singing in a conservative Protestant church is about 95 decibels (dB) and in a Pentecostal type church the singing is about 5 to 10 dB louder? According to most of the Health and Safety Acts around the world, long-term exposure to sounds at these levels will cause some hearing loss. In one church, the congregational singing could last up to an hour and a half. In this 2,000 seat church, the sound levels were often over 108 dB. According to OSHA, your exposure time is 30 minutes without hearing protection. Perhaps if people started to ask for hearing protection in some churches, the sound levels would come down.

(Many of these churches have serious acoustical problems and most sound men and audio companies have the attitude that if you turn up the sound loud enough, eventually, the room will not have any affect on the sound and it will eventually get clearer. Folks, the opposite is true. Turning the system down will not only make it clearer but people would get more out of it than just abuse.)

Distance	SPL dB	Increase
32 in	60	0 dB-Performance of a HIS System
16 in	66	6 dB-Normal Speaking Distance
8 in	72	12 dB
4 in	78	18 dB-Typical singer holding mic
2 in	84	24 dB
1 in	90	30 dB
½ in	96	36 dB
¼ in	102	42 dB
⅛ in	108	48 dB-Lips touching the windscreen

Sound Pressure Change When Moving Into A Microphone.

This chart shows the amount of change that occurs as a person moves around a microphone. Just as doubling your distance changes sound levels 6 dB at the speaker end, the same happens with input. When a person moves into a microphone room 16 inches to 1 inch, the sound pressure change is 24 dB. In this close distance, there is little or no room effect on the input signal.

A CONSTRUCTION DETAIL TO CONSIDER—AND MUCH NEEDED

Cable chase ways: Since there has been no standard in church sound systems, the idea of a cable chase way seem pointless. However, if a cable chase way for lighting and sound were planned, then the church could have flexible options in the future.

Cable chase ways are not new. In fact, offices have had them for years. Cable chase ways cost more during construction but later when changes are done, there would be no need for expensive conduit work after the fact.

IS THE "Q" IMPORTANT?

It is now generally accepted that a church is best served with a central cluster speaker system. The speaker system is often placed somewhere over the front of the pulpit or platform area using directional speakers that have a predictable and constant dispersion of sound. Omni directional speakers such as spheres, column speakers, low "Q" speakers and Hi-Fi speakers fail to perform in so many critical ways they are not worth the paper to criticize.

Not all speakers are the same and these other speakers work better in application where non-critical listening is required. Finding so many Soundspheres in church closets, for a "State of the Art Technology" that is considered current, is a clear case of buying the wrong tools for the wrong job. Between spheres, column speaker, flat speaker other speaker designs and audio products, if all of the churches who owned them donated this equipment to other churches who are determined to waste their money on this inappropriate equipment, the equipment could be recycled many times and churches would spend less of my money foolishly. However, as one church board chairman said, "I wouldn't want that equipment wished upon anyone else...." Those strong opinions are often made after a church has had the opportunity to compare a proper church sound system with one designed with good intentions.

In the early days of church sound, speakers generally were all horn types with horn loaded woofers. Amplifiers were expensive and rarely over 100 watts in power. (Today, many amplifiers can produce over 400 watts X 2 channels at a 4-ohm rating.) A term often used and often misunderstood is "Q". "Q" is the directivity rating of a speaker.

High "Q" speakers are designed to project sound over large distances. A low "Q" speaker is a speaker that allows the listener to get comfortably close without serious degrading of the sound. A hi-fi speaker is a low "Q" speaker. A police Bull horn is a high "Q" speaker.

Just as the installation of the Organ was resisted in the church because it was once played while Lions were feeding on Christians in Roman arenas, churches have also been slow to embrace the sound system for similar reasons. At first, churches used their sound systems strictly for speech only. Today, some of the best sounding systems in public places are in churches.

In the beginning of church sound, people soon learned that a sound system was very limited and often the room had to be fixed. Many churches did fix their sanctuaries with very good results. In the mid 50s, bigger and less expensive amplifiers arrived in the market place. At the same time, speaker engineers began to understand what "Q" was and how to measure it. A high "Q" rating could be from 10 to 25. A low "Q" is from 2 to 9.

There is a direct relationship between reverberation, "Q" and how much direct sound arrives to the people in the pews. Poor reverberation limits the performance of a speaker system. By increasing the "Q" of the speaker, you can compensate for the increase in room reverberation. However, there is a limit as to how high you can increase the reverberation of the room and have a speaker with enough "Q" that will still give the best sound coverage possible.

As more churches began to accept and use sound systems, audio contractors suddenly had to deal with churches that had reverberation times of over 1.5 seconds. (Most audio contractors rarely have to deal with as hostile an acoustical environment as that which occurs in churches.)

There are some speakers that have a "Q" of 50 . A police bull horn has a "Q" of over 50. Police bull horns don't sound very good. Many church sound systems sound like bull horns but regular two way speakers are being used which should sound very natural. Once a speaker "Q" rating passes 25, it no longer sounds natural due to too much compression of the sound within the bell of the horn. (Horns compress sound in terms of ratios. A highly projecting horn can have a compression ratio of 12-15 to 1. A natural or musical sounding horn has a compression ratio of 4-8 to 1. Low compression, high "Q" speakers give the best overall performance in a church.) Speakers with a "Q" of over 22 generally are not very musical. They have to be supplemented with woofers. The end result is the woofer drags the "Q" rating, over the full range of the speaker system, down.

Later, someone discovered that if you stacked several horns on top of each other, you could increase the "Q" of the speaker system and maintain a reasonable quality of sound.

This works fine but where do you draw the line? When is it time to stop looking to sound equipment as a magic wand to solve poor building designs or poor use of construction materials?

In one photograph (page 348 in the book *Sound System Engineering* by Don Davis), it shows a speaker cluster of 4 horns and 4 woofers stacked one on top of the other. It was demonstrated that the test worked very well but the speaker system looked awful. This was an example of creating a high "Q" array using a column speaker approach. In this case, it would have been better to fix the room rather than having a speaker system that looked like a monster.

There have been several studies and charts suggesting the limits of reverberation and the "Q" of the speaker system for churches. (These charts can be found in the books mentioned earlier. Klark Teknik has a manual that has excellent charts on this subject.)

From experience, a church with a reverb time of 2.4 seconds or longer at 200 hertz will not be able to count on their speaker system to work properly for every part of the worship service. "Q" is important but there is a limit. The higher the "Q", the more speaker compression.

For churches that have ambitious music programs, a medium to low "Q" speaker that sounds very musical and low reverberation are a must because in a loudness war, the room always wins. (A list of the 5 top speakers for the HIS System discussing their good points and bad points and when to use them will be in the next update.)

Note: Dome tweeters and bullet horns have a very limited use in a church. They have two major problems. If the speaker system was only controlling prerecorded music all day, they would work very well. However, speech requires a lot of speaker output in the mid range. Many domes and bullets fail and fall short of the power demands of speech. Secondly, most domes and bullet horns have such wide dispersions that gain before feedback is seriously limited. Although domes and bullets allow you to make a lower cost speaker, the performance limitations and high rate of failure make them impractical for use in a church.

ARE ALL SPEAKER CLUSTERS THE SAME?

It would be fair to say there are huge differences in speaker components and the jobs they are designed to do. It is also important to know that the position of a speaker in a room is super critical. Every room has a sweet spot. A speaker out by 1 foot can make $2,000 speakers sound like $50 A.M. radio speakers.

Some speaker systems are less critical than others and in some rooms there is no room for error or flying by the seat of your pants in design. You should have a detailed knowledge of your sound system supplier. The following is a list of things you should know about your supplier:

1. Learn everything about the speakers you are being recommended. An audio contractor doesn't have time to know about every speaker that is available or how they sound.
2. Avoid custom built speakers. A custom built speaker is often designed on a given set of assumptions.
3. If you have a custom built speaker for an original design, ask for the printed test results. Usually, there is no laboratory testing on custom speakers so you have no way of knowing 100 percent how the speaker will behave under church conditions.
4. A good sound contractor will have a limited number of speakers they use and will know them inside and out.
5. Learn how to read a spec. sheet and understand what it all means.

After selling over 100 complete sound systems, it never ceases to amaze me how a church can spend huge amounts of money based on a proposal without going to hear the finished product in other churches. Many speaker clusters look the same. Many audio companies use the same equipment. The end results are like day and night. When you buy a church speaker system, it must be assembled and strategically located in your church. What good is it to have a speaker shoot-out at your church when the audio company doesn't know how to install it for the best performance. Speaker testing is often a waste of time for the client. It is very important for the contractor.

In many speaker demonstrations, people are often caught up in how a speaker sounds rather than knowing how a speaker performs. In reality, the differences between speakers from the major speaker manufacturers is very small. It is the ability of choosing the right speakers for a given job, that takes skill.

In a recent speaker demonstration in a church, three different horn speakers from three manufactures were represented. Before equalizing and before setting the volumes at equal levels, all three speakers had very distinct sound differences. However, after equalizing the speakers and setting the volume levels to equal loudness, the differences were very small. In each case, the speaker system was limited by the performance of the room. Once the room was factored in, the sound quality of the speakers became very similar. What was more noticeable was speaker coverage and control. One speaker was more musical, the other could play louder and the final speaker had better control.

WHAT DOES THE CHURCH NEED? DOES THE CLIENT KNOW?

Speaker clusters have a very small window in which to work. When you buy a sound system for your church, it must include the skill of your contractor and his sensitivity to the needs of the church and the acoustics of the room. It is nice to buy the speaker system that sounds the best, but if the best speaker system can't sound better than the others because the room gets in the way, isn't it better to find the best combination supplier, speaker and room repair?

When a church asks for a demo in their sanctuary, the smartest sales person is going to win. When a church takes the time to visit other churches to hear the system in action during a worship service, the best performing system will win. Not all speaker clusters are the same and not all audio contractors know how to find the room's sweet spot.

ONE RULE—ONE FACT

One of the most important rules for all sound operators is simple: Once the minister begins to preach, you are allowed only one volume level adjustment throughout the whole sermon (unless your minister instructs you otherwise). With a good sound system, the minister should be able to control their own volume by using the microphone properly (more effectively). However, there is one adjustment most ministers won't mind.

Most sermons begin 3 to 7 minutes after singing. Since most singing is very loud, your hearing of low sounds is stressed for a while. Speech is intermittent. Speech gives our ears a chance to rest between words. When the volume is set right after a Hymn, chances are it is too loud. Some people get the impression the sound system has been turned up. Rather, it is our hearing that became more sensitive to lower level sound.

Often the sound person can lower the volume of the system by 3 to 5 dB at about 5 to 10 minutes into the sermon. The interesting thing is, if you don't do this and you need more volume from the sound system, it won't be there.

Another rule is to turn off all other microphones not in use during the sermon. The more mics that are on, the less gain you have before feedback. This is important for the times when your minister has a throat problem but is determined to continue in a lower voice. Furthermore, the sound tends to sound clearer because there are no open mics to reinforce the first sound. There are other reasons for turning things off or down, but for now, these reasons will do.

A POWERFUL PERSON

One of the most important facts is, the soundman, in churches with a good sound system, is the most powerful person during a worship service. The soundman can either enhance everything and assist people sitting in the pews to be more involved with the service or, the soundman can undermine everything the minister does without the minister knowing it. It is time we wake up to the fact the 95 percent of all churches have a sound system of some kind. Unfortunately, only 5 to 10 percent of these churches have a sound system really usable as a tool. All other churches have sound systems that get in the way. When something gets in your way, it undermines everything. Even a good sound system can have the same problems of getting in the way.

For example: In a good system where people expect to hear properly, the minister or a lay person may move to a mic that has been turned off to give the other mic better control. The sound person should be alert and see the minister moving to the other mic.

If the mic is set right, the mood of the service continues. If the minister speaks for 3 seconds or longer with the mic off, and the minister notices it, what happens? It breaks the mood and often, his concentration. This single event upsets more ministers than anything else. Moving from one mic to another at the closing of a sermon may be just what the minister needed to do to make his final plea or point. A soundman asleep at the controls is unacceptable.

Of course some will suggest using a wireless lapel mic. In my opinion, a wireless lapel microphone has some noticeable limitations. Most churches do not have the acoustics needed to use a lapel mic effectively. (Usually this means a dead room and most churches don't want dead rooms.) With a lapel mic you cannot raise or lower your voice too much without losing something or distorting the wireless. (Automatic volume controls will cause other problems.) With a wireless, all you get is plain vanilla in the presentation. Just as some ministers wear the well-earned title of word smith, a person can also be an *amplified voice smith*—a person who knows how to present a message using all of the inflections of voice. Casey Kasim has the most recognized voice on TV today. He has made millions of dollars selling his voice to television.

On PBS, the public broadcasting service, there is a weekly show called *Nature*. The success of *Nature* and its continuance is based on the voice of George Page. This is a voice that brings life to the screen. Jason Robards, Lorne Greene, Robin Williams and Dick Van Dyke are all examples of the use of voice. As they say in the business world, it's not what you say but how you say it. A good sound system accurately reproduces everything you say and how you say it — the key to a successful presentation.

A lapel mic can not give you anything more than a vanilla presentation. Then again, there are many gifted ministers who know how to make plain vanilla taste like chocolate. Of course, this is only an opinion based on many experiences. What have your experiences been?

This is not to say wireless are not effective. For drama presentations, plays and other specialty events, a lapel wireless setup is just the ticket. Today, some ministers make the usage of a wireless microphone a condition of their employment. Getting a wireless mic just to have "freedom" must be combined with planning to be more dynamic with body language. What a church should avoid is getting a wireless lapel mic for a minister who does not use body language in their presentation when they could have done better buying 6 regular microphones for most of the other events they wish they could do better. A good wireless lapel is a lot of money. If all you have is $1,000 to spend, 6 mics with stands and cables could be the wiser choice.

ASSESSING THE SANCTUARY

Room Shapes

There are six basic room floor plans with hundreds of variations of each. There are rectangles, squares, diamonds, ovals, triangles, circles and pentagons. In roof designs there are several basic shapes with many variations of each. There are Domes, "A" frames, Flat, Sloped, Vaults and waves. Almost all the known shapes will work but you need to know at which end of the church you will preach and at which end you will listen.

The type of room shape must match the nature of the service or denomination. There is no such thing as an ideal or perfect space but, it is important to recognize the shape, how it works and where the speaker system must go. Don Davis wrote in his book *Sound System Engineering* that a speaker system often goes where a wall or ceiling should be. This is very true. But as you get into larger spaces that seat 200 people or more, you need more than a reflector. The wall must amplify as well.

SOUND SYSTEM DESIGN TYPES

In churches, you will see a variety of system designs used. Some designs are chosen because of appearances, others are chosen because of the perceived cost savings but most are installed to compete with room acoustics. Unless you have a ceiling lower than 12 feet, the best system in any church is a Central Cluster design.

The Central Cluster design forces you to look at the person speaking or singing. It offers the highest levels of intelligibility. It has the lowest levels of Listener's Fatigue.

It is usually the best layout for operating and hearing the speaker system. It usually has a higher level of gain before feedback. It is a system that you notice the least.

Therefore, you can say a central cluster system is the least obtrusive available. The other sound system designs are either creators of dead spots or very expensive if done correctly.

The following is a description of the various system types and why they are not appropriate in a church.

PEW BACK SYSTEMS

The pew back system is most often attempted in churches with long reverb times. It is based on the idea that if you get the sound sources closer to the people, the listener will get more direct sound and less interference from the reverberation of the room. If it is to work, it can only be a speech system. It will not work well for music because music has to be played loud enough for the musicians to hear themselves. This means, in the pew back system, a listener could hear both the direct sound and the amplified sound coming from many directions, depending on how the listener moved their head during the live music. Music, at a medium level in a pew back system, will increase the reverberation time of the sanctuary. Remember the reason for the pew back system?

From the experiences known, there is no church that has a pew back system that works as good as a properly designed cluster system. Generally, a pew back system is abandoned after several years of trying to make it work. Robert Schuller's Crystal Cathedral in California is a prime example. They went from a cluster system to a pew back system and now they are back to a cluster system. The next step is to find a method of reducing the reverberation without interfering with the appearance of the "Glass Slipper".

DESIGN PRINCIPAL

The pew back system will have speakers mounted either on the back of the pew, under the pew or on the floor under the pew. A speaker is placed every few feet to give even coverage in each row. Each row, or in some cases, every other row gets speakers. Some people have tried one speaker for every other person or a 2 to 1 ratio, while others have tried up to 10 to 1 ratios.

For every 2 or 3 rows of speakers, you have to install a delay system. The delay is used to delay the electronic sounds from one row of speakers to the next and to delay the electronic sound from the source of the sound. Without the delay, you will hear echoes. Costs can vary. For a speech only system that works reasonably well, the cost per seating position can be well over $80. This makes it the most expensive way of doing sound. Would it not be better to spend $35 per seat on sound and $45 per seat on acoustics?

Contractors who have tried pew back systems have used all types of speakers at all angles. It doesn't seem to matter whether it is a $10 speaker or a $200 speaker, overall performance is low. Some systems are on a 70 volt or 100 volt distribution layout, while others have tried a mini amplifier for every 2 speakers. Others have also tried to series speakers together to try and keep the amplifier cost down. Running an audio signal from speaker to speaker does create other sound quality problems as well.

The only place for a pew back system is in town halls, city hall counsel chambers and in churches that have no music in their services.

THE LEFT/RIGHT MONO SYSTEM

The left/right mono system has always been called the poor man's system. It is installed out of convenience, lack of planning and a copy of what entertainers do when they only have a few hours to put on a show. Of all the books available on sound reinforcement, not one book shows how to install such a system. Rather, they go to great lengths to explain what is wrong with it. Instead, they all support the cluster system as the best way to install a permanent system. The left/right mono system is strictly a portable or temporary system setup. It was (and still is) a fast and convenient system by which entertainers could put on a show for their public.

There are four main reasons for not using such a system in a church:

1. Dead Spot

When you have two mono sounds separated by 10 feet or more, the left and right speakers will start canceling each other out in the overlap areas and whenever you are at a different distance from each speaker. When you are at an equal distance to each speaker, the sounds are summed together, often increasing the sound level 6 to 9 dB.

When you include wall reflections, the problem is compounded and it creates additional dead spots. When the speakers are 25 feet apart or wider, the areas of overlap increase dramatically. A dead spot can be easily measured with an inexpensive sound level meter such as the one Radio Shack stores sell.

In many testing experiences with computerized measurement systems, some very interesting pictures of sound began to appear. In many churches with a left/right system, it could be seen how the sound from the left speaker was louder than the right speaker but, the test microphone had been placed in front of the right speaker. This position would be about 30 to 55 feet out and about mid point of the right side of the church. By standing at this position and running the test signal, you could indeed hear which speaker was louder. Please, remember that dead spots are often frequency dependent. That is, since every frequency has a different wave length, not all sounds or notes will cancel or boost at a given position. A sound level meter can tell you whether the sound is lower or louder. A computer system is needed to determine if certain sounds are missing in a given position. In translation it means that in this pew the vowel "a" and a "Mc" are not audible and all "b" and "ch" sound are too loud. Some people can fill in the blanks better than others. People with hearing aids have problems in this setting.

2. Gain Before Feed Back

In a left/right speaker installation, the speakers are often placed behind the pulpit at equal height or slightly higher than the pulpit. Most speakers have great vertical dispersion control but poor horizontal control. This is like placing a microphone in front of the speaker, and we all know that will cause the system to feedback. The causes of feedback are a combination of acoustics, angles of incident, proximity effects and oscillation. As you turn up the level of a microphone, it is able to pick up and amplify everything. This includes any low level noise generated by your electronics (mixer, amplifier, equalizer, preamps, effects devices and other pieces of electronics connected to the sound system that is unstable. Poor wiring and electrical induction are other causes of noise.) As a result, any speaker close to a microphone will cause feedback. Any speaker that has a signal path with an angle of incident that reflects sound back to the microphone will cause feedback. When the gain of the mic is turned up so loud that the noise from the system is being recycled (that is when the noise in the system is being produced by the speakers and picked up by the microphone, the level increases in a cycle many times until you hear it. This can appear to happen instantaneously), it causes feedback.

The acoustics part of the picture is more complex. All large rooms are constantly flexing and moving. This is a constant effect altered by room temperature and humidity. Speakers move air. Microphones will amplify everything whether you can hear anything or not. At a level you can not hear, the room's flexing is being picked up by the sound system. As long as the level of sound system is kept low, the sound system is stable. When the levels of the sound system are turned up, decay time of the rooms flexing becomes longer. Now, at the same time, the speaker system is moving air and exciting the room, multiplying the energy output hundreds of times. This creates a chain reaction in which the sound system amplifies the resonant frequency of the room (room flexing) and causing feedback. Acoustics play an equal part in the causes of feedback. When there are multiple speaker locations in a room, the number of wave forms that will excite the room increases. Also, the number of hot spot reflection points increase which causes feedback. This is only a partial explanation.

3. Intelligibility

Intelligibility is the understanding of individual words. As discussed earlier, in a 45 minute sermon a minister can speak about 10,000 words. A sound system with a score of 80 percent intelligibility will mean that 2,000 words in a 10,000 word sermon were missed or misunderstood. Depending on your seating position, one speaker will either boost or cancel certain frequencies. That means every "ch" sound is canceled and every "a" sound is amplified. As a result, many words and word fragments are missed. Fortunately, our brains are usually fast enough to fill in the blanks because of familiarity with the subject matter and the person speaking. A visitor to the church does not have this advantage. For this reason, no church should accept a sound system that scores below 87 percent intelligibility. Below 87 percent people can misunderstand complete phrases. The TEF or MLSSA acoustical measurement systems can test a room's intelligibility in minutes.

4. Localization

Churches always say they don't want to see or hear the sound system, but it must be loud enough and crystal clear. When you have a sound source at ear level, you are automatically programmed to first look at where the sound comes from, then to look for the source of the sound. This is a basic protective instinct all creatures possess — to be able to tell where danger is coming from is natural. Since humans have no natural enemies that attack from overhead, man has his eyes and ears where they are, on the front and sides of our heads. A mouse is most often attacked from overhead. They need their eyes and hears closer together, near the tops of their head. A mouse can also rotate the ears to find the noises.

When the sound is coming from one direction (Speakers) and the visible source is in another direction (the person speaking), the human brain goes into overtime matching up the two events. When you compound low levels due to feedback problems, low intelligibility and multiply sound sources, you have classic listener's fatigue syndrome.

It has been shown that a poorly designed sound system can help people lose their attention span or cause them to nap sooner than when a proper sound system is used.

The difference can be as much as 20 minutes.

When the eyes and ears can focus on the same event, you spend most of your time understanding what was said. When a left/right, or distributed or pew back system is used, you really spend most of your time just trying to hear.

It would be fair to say most church sanctuaries, by themselves, do not have dead spots. Rather, most churches are so large they need amplification. If all you need is extra level, then why would anyone install a sound system that creates dead spots?

WHAT ABOUT A STEREO SYSTEM IN A CHURCH?

Some churches have successfully installed Stereo Reinforcement Systems. With the arrival of electronic instruments, stereo keyboards and tape accompaniment, some churches have felt it necessary for this kind of investment because they had the facility and talents.

Remember, a Stereo system at home is the opposite from a stereo system in a church. At home you position yourself between two speakers and the recordings play tricks with phase to give the impression that sounds are coming from different places within the sound field. In a live situation, you can see where the sound is coming from. Therefore, the sound system must give the impression the sound comes from the same place otherwise the performer will hear an echo and it will disrupt their playing ability. A live stereo speaker system is really many speaker clusters over the performer's head. The nearest cluster amplifies and the other speakers are silent. This gives the effect that sounds are coming from the different parts of the stage where the performer is, giving a better picture of localization. As a result, a true stereo church sound system can cost 4 to 5 times more than a good quality mono single cluster. (Church stereo systems will be discussed in detail in future editions of the book.)

SPEAKER LOCATION

The location of the speaker system in a room is the most important step in clinching the desired results. If the speakers are in the wrong location, the rest of the system will sound mediocre despite the quality of the equipment. In almost all churches, there is only one proper location for a speaker system. Any other location is a noticeable compromise in comparison.

The speaker system in your church is the most important part of the audio link. If this part of the system is not correct, you will not be able to successfully make any improvement through electronics. It is vital therefore, to make your speaker system the first step in correcting your sound problems. Not treating the speakers first will result in needless experimentation and expense. (It is amazing to see all of the gadgets churches try to invent. If only they knew the laws of audio and acoustical physics, they would spend millions of dollars less on audio products that don't work.) In rectangle shaped rooms, a single point is ideal for all church application.

(According to Dr. Dave Eagan and Dr. Don Davis, there is no other shape better than the shoe box or rectangle church. Boston Symphony Hall is a prime example.) Often the speaker location is always several feet in front of the pulpit, overhead. Other times it is directly overhead while in some rooms it is 2 or 3 feet behind the pulpit. The exact positioning must consider ceiling height, pulpit height, width of room and pulpit to back wall distances.

In wide or fan shaped rooms, localization of sound does present a minor problem. Those sitting on the ends will perceive two sound sources. This will not reduce intelligibility. However, it will introduce minor amounts of listening fatigue.

For this reason, it has been our practice to divide the sanctuary into several rooms. This gives all people a point source for listening comfort. Sub clusters are very effective. Phasing problems are controlled by separation, crossovers between horns and sometimes digital delay circuits.

Typical comparison of a Single Cluster Speaker System verses a Multiple Speaker System.

Situation	**Cluster Systems**	**Other Systems**
Dead Spots of 3 dB or more	Almost none	Many
Phase Cancellation	None	10 to 30%
SPL from front to back	6 dB or less	Often 12 dB or more
Realism	98%	15% or less
Articulation	+85%	75% or less
Intelligibility	Great	Poor
Max. working distance before feedback	20" to 40"	4" to 16"
System design life	Permanent-possibly unlimited	Replace when no longer tolerated
Listening fatigue	0.5&	20 to 50 %
Music Quality	Hi-Fi	Limited
Flexibility	Very good	Limited
System headroom	20-30 dB	10-20 dB
System focus	Pulpit area	To each speaker
Echo Amplification	5% or less	10& and up

CLUSTER HEIGHT

The maximum height for a cluster should be no higher than 40' and no lower than 13'. However, height is also determined by the speakers throw distance and other room restrictions. Remember, the closer to the ceiling, the more bass the speaker system will produce. If the room is bass heavy, hang the speaker lower if there is room to do it.

If you require a throw distance greater than 145', a sub cluster system may be required. Ceilings below 13' may require other system designs. This book will not discuss these requirements in detail in this edition. Send for a Supplement for your church.

Height	13'	Max. Length	48'
"	20'	"	72'
"	25'	"	85'
"	30'	"	110'
"	35'	"	130'
"	40'	"	145'

Height Ratios. The target point is usually the third row of pews from the back wall.

SPEAKER SAFETY

Hanging a speaker from a ceiling presents some concerns that need to be addressed:

1. Cabinet construction: Ceilings in many churches have a wide temperature range. During the summer in some churches, the ceiling can exceed 120 degrees for many days on end. Many speakers are only fastened with nails and glue. There are a number of stories circulating the Audio Engineering Society (AES) and the National Sound and Communications Association (NSCA) that describe how speakers are falling apart and falling down. Look for a speaker cabinet that is reinforced for roof suspension.

2.Do not use chain: Speakers vibrate and it can cause metal fatigue to the chain links. Also, a lot of bass sounds are lost with chain suspension.

3.You should not use aircraft cable because of sound quality in the bass. It has the same problem as chain.

4.Speakers should be supported from the side walls, not from the top of the cabinet only. (Some speakers have metal rods or bars that run through the speaker box to support the bottom of the speaker from the inside. Also, look for a speaker with a space frame type of construction.)

SPEAKER BRACKETS.

Custom steel speaker brackets are the best way to support the speaker system and they are very inexpensive to make. A properly welded steel bracket wins in many ways. Depending on the number of speakers in the cluster, there is a wide variety of fast and simple adjust you can make that are often too awkward with chain or cable.

In most single and two speaker clusters, the steel bracket is very cheap to make. A bracket that is safe up to 400 pound can often be made for under $50. That includes the steel, welding and paint .

THE BEST ILLUSION OF THEM ALL!!!

There are three very strong effects only a central cluster can do. The first, which has already been mentioned, is localization. From 80 to 90 percent of all the seating in the church, when someone on the platform of the church speaks, sings or plays, everyone's attention is on that person. Since the cluster is positioned in the vertical axis of our ears, the sound arrives to both ears at the same time. The reflected sounds will give the direction.

Since the reflected sounds are much lower, your first reaction is to look at the pulpit area or center to the altar area. At the same time, your eyes will focus on the first moving object or to the tallest person standing at or around the pulpit. The only people who will notice the cluster are people seated at the extreme right and left of the front three rows of pews.

The second effect clusters give is the illusion that the minister is only four or five feet away. Since you are listening to only one speaker in your seating position, there is no presence cancellation. Generally, this effect is for 70 percent of the seating and when the RT60 is below 2 seconds. When the RT60 is longer, not only does the sound system lose intelligibility but the music program is degraded.

Multi sound source speaker systems can never give this effect. This system gives the illusion the minister is further away than what he is in reality.

The third and most exciting illusion a cluster has to offer is movement of sound. Since 80 to 90 percent of the people seated have sound arriving to both ears at the same time, it is very difficult to say where the sound is being amplified from except from the source, the minister, lay person or singer. When the person moves, the sound appears to move with them. There is a limit to this. If the minister moves from a central pulpit position to the extreme left, talking the whole time with a wireless lapel mic, it leaves the audience thinking the sound followed the minister until he had moved half the distance, then the speaker is high. Example: If the cluster is 20 feet high, the minister can move 10 feet to either side of the pulpit and have the illusion that the sound is following him.

Since it is natural to look at what we hear, the central cluster approach to church sound is the only truly natural method of sound reinforcement.

SOUND OPERATOR TIP

One of the rewards of installing a high quality and affordable sound system is in the listening to the system. When a system is properly adjusted, for the many different parts of a worship service, you will have the impression that the sound system is not on at all. Although many listeners like the effect, a new problem keeps showing up.

For every new system, you try your best teaching the volunteer operators of the system. In recent years, video taping the training session has become a valuable tool in reminding the sound operators of all the tricks to using the new system. However, as good as some sound operators are, training can take many sessions.

As stated earlier, many people with audio experience have generally picked up many bad habits. The one habit that is hard to break is the most obvious. In many retraining sessions, I have found the operator has changed the channel equalizers or the main equalizer in such a way that the sound system sounds like a bull horn.

It seems that hearing the minister as though he were only 2 feet away is not enough. Many sound operators want to hear the sound system sound like a PA system so they will be convinced the system is working. The bottom line is, they don't trust their hearing. Although this is not a serious problem, it has caused some embarrassment to the suppliers of such systems.

If you operate a good sound system, trust your hearing. As long as you can understand the minister, everyone else will too.

MIXER LOCATIONS

The best location for a sound operator in a church is on the main floor, 1 to 3 rows from the back of the church and in the pews. Preferably, just inside of the outside isles.

If there is a balcony with seating under it, the mixer desk is best located 2 or 3 rows out from under the balcony's front edge. Although it is a new concept to most churches of today, historically, churches started the idea of having a sound operator controlling the sound system from within the congregational seating area in the 1940's.

As it turns out, it is impossible for a person to adjust sound levels from one area for people in another area. It doesn't work. The idea of having a mic mixer in a place like a pulpit or a room behind the altar is very awkward. How many times have you been to a church with such a setup and have heard the sound system ringing or sounding like someone is speaking through a tin can through the whole service. People come to church to pray and hear what the Minister/Priest has to say. It is annoying, insulting, and rude to have to put up with something that could have been adjusted in seconds.

And of course, when someone complains, the Minister/Priest says he didn't hear it! Well of course not. You have to be in front of the speakers to know what is happening. Well, enough with people frustrations.

Having a sound operator and a lockable secured wooden or steel mixer desk is the best choice. For some denominations, that have a very structured hour of worship, in most services the levels can be preset with 3 or 4 mics and no one needs to operate anything. The only problem with this or any automated system is that everyone speaks differently from week to week, day to day, minute to minute. Even with the most professional presenters, no one is able to speak at a constant level for everyone to hear all the sounds at the same volume. True, you can get expensive gates, limiters and compressors but their use is very limited. For example, if you set your limiters for a person with a powerful voice who is speaking very close to a microphone and seconds later pass the same microphone over to pick up the choir at 10 feet away, you will hear nothing. Move the mic 5 ft. away and you still hear nothing. But if you bypass the limiter, compressor, and gate you will find that you have more than enough audio level.

If there are several things going on at the same time in which many mics need to be turned on or off, having a sound operator is the most natural and best way to run the sound system. With a good operator, most people will not be aware that any adjustments are being made. Besides, when the operator is in the pews, he cannot day dream or fall asleep. He is forced to stay alert.

It is no secret that the fewer mics that are turned on, the higher the system can be turned up. Most churches that have tried to use an automated mixer system wind up having the bypass switch on all of the time. This translates into a $10,000 expense that is not being used. The only place where an automated mixer will work well is in rooms with RT60s that are below 1.3 seconds and the NC is 15. There is nothing better than a person operating the mixer for live sound reinforcement.

Balconies are an option under certain conditions and if the sound operator is young and in good health. Stairs are not fun. Setting up a service that has special music or concerts takes three times longer when a balcony is the location of the mixer. Operating from the balcony is a two man job unless you are willing to hold up the service from time to time to let the sound man finish the set up.

From a listening stand point, the balcony must not have an arch or beam above the railing. If there is a beam or arch, it automatically reduces the sound level considerably unless the sound operator is in from the beam by 10 feet.

A good quality custom mixer desk that matches to pew design and color can cost from $900 to $1,500 to make. The mixer desk would only contain the mixer, tape machines, wireless receivers and remote controls for lighting and AV (Screens).

SPEAKER, MICROPHONE WIRES AND WIRING

Microphone Wire
Recommended Pin connections should be:
Pin 1 ground, drain wire
[do not solder pin 1 to the shield]
Pin 2 (+) Hot Red wire
Pin 3 (−) Cold Black wire

This is a standard followed by many contractors and audio companies. However, some manufacturers use pin 3 as hot. Check the manufacturer's specification sheets before you interconnect your electronics as it can often cause some hums and noises when pin 2 and 3 are incorrect.

All microphone { LINES } shall be of a LOW IMPEDANCE TYPE.

All microphone { CORDS } shall be of a LOW IMPEDANCE TYPE.

The line will consist of 2 stranded lines with 1 drain wire or ground and foil shield.

Shield *must* be aluminum foil wrap for permanent wiring. This is currently the best available shielding that will give 100 percent protection from RF. This wire is not suitable for mic cords as the foil shield is prone to breakage or unraveling.

For movable mic cords use a stranded shielded wire. Depending on the manufacturer, the best braided shields are between 85 to 93 percent. However, short mic cords will maintain a shield under stress from bending. Braided wire is not suitable for permanent installation work as it will not give a 100 percent shielding from RF interference.

INSTALLATION DO NOTS

...llel to AC (Alternating Current)

...l to ballast routes or fluorescent

...l to unshielded speaker lines ...watts.

...ave any 90 degree turns.

...les. All mic line should run ...ut breaks. (There is one ex-... the signal for TV or Radio ... go through a splitter box

...ps

...Beldon 8451 or equivalent and 14/2 stranded ...aker cable:

Conduit Size	Mic Line	Speaker Line
½ in	4	1
¾ in	8	2
1 in	12	3
1¼ in	20	4
1½ in	30	6

Never have more than two 90 degree turns in each conduit run. In any system with more than 8 channels as a starter system, you should consider conduits for the following projects that require conduits:

1. Mic cables
2. Speaker cables
3. Video cables
4. Remote lighting
5. Remote platform lighting
6. Electrical
7. Audience mic inputs
8. Distributed system

HOW MANY MIC LINES?

The number of mic lines you need is also the size of mixer you should have. The only thing to determine a larger mic input requirement is the size of your music program.

Generally, you should have 1 mic input for every 70 square feet of altar or platform space. This does not mean you can't group all of your mics to one location. What this does ensure is that for 99 percent of your churches functions through the year, you will have enough mixer channels and mic input so that you should not have to rent equipment for special services.

To calculate this you have to measure the size of your altar area and divide by 70. The following chart will assist you in your decisions.

1 to 9—8 channel mixer
10 to 13—12 channel mixer
14 to 18—16 channel mixer
19 to 26—24 channel mixer
27 to 34—32 channel mixer
34 to 42—40 channel mixer

There is an exception to the rule. For smaller churches, you should always install at least an 8 channel mixer with 8 mic lines.

As it turns out, audio amplifiers are not very efficient but the amount of current generated does raise some safety and fidelity issues.

If you can, avoid patch bays. It is often cheaper to buy a larger mixer than to go through the expense of building a patch bay. Besides, most churches I know of who have a patch bay never use them more than once a year. Rather, these churches wished they had larger mixers.

MIXER TIP

Always number your stage and the mixer the same with mic stage number starting from left to right from the soundman's position. Therefore, if you pulpit is in the middle of the church and you have a12 channel system, the pulpit control on the mixer will be either 6 or 7. This is very helpful for people who only operate the mixer a few times a year.

SPEAKER WIRE

For those people who are looking into the audio industry and wondering what all of the hocus-pocus is about in speaker wire, have we got some bad news for you. Are the claims of the seller of expensive speaker wire telling you the truth about speaker wire?

DOES WIRE MATTER IN A CHURCH INSTALLATION?

For hi-fi people, this book will not settle any disputes. There are dozens of claims that speaker wire manufacturers are making every day. Sometimes one would think that somebody, with nothing better to do, is figuring out what the next scam for wire will be. I wonder how far they will push before they are unable to get away with it any longer?

For the church, there are several solid reasons for doing some of the things needed for a church installation.

Let's start with a high current amplifier. A 200 watt into 8 ohms 2 channel amplifier can generate a considerable amount of current. There is enough current in two of these amplifiers to trip a 15 amp. fuse.

In some of the large current 200 or 300 watt amplifiers that boast they can work at 2 ohms, the outputs either have a 10 or 15 amp. fuse for each channel. That is enough current to run two drills drilling into steel (intermittently). Fortunately, all of that power is momentary. Different frequencies and rapidly changing volume levels often avoids thermal shutdown in a church setting.

However, we must recognize that even for a millisecond, a 200 watt amplifier at 2 ohms can generate a potential of 10 amps per channel of the amplifier. If this amplifier were 90 percent efficient, it would require a 20 amp service to plug into. As it turns out, audio amplifiers are not very efficient but the amount of current generated does raise some safety and fidelity issues.

Will expensive wire help the church sound system to sound better? In most cases no. The reason is simple. Sonic differences in wire is usually subtle.

Many churches have an NC above 25 dB and/or they have a reverb time longer than 1.8 seconds. This is a very hostile environment for playing games with fidelity. If your church has either of these two problems, there will be too much interference to hear the difference. Furthermore, if the speaker is further than 30 feet away, as it is in most cases, room effect will also interfere with the sound quality.

If your church has neither of these problems, then the wire issue becomes a cost consideration. Some high tech wire can cost over $15 per foot. In a 150 foot run, the cost of 1 wire run would be $2,250. The wire costs more than most professional speakers. Therefore, unless you have the speakers to justify expensive wire, common sense should prevail. And one more point, expensive wire on cheap or poorly designed speakers is a waste of money. If you have to chose between speakers or wire, spend the money on the speakers.

It would be fair to say in most cases, the average person in the pew and the musicians will not hear the wire difference in low cost, budget speakers. The magic wire has to offer can easily be defeated by choosing a better sounding speaker.

DOES WIRE SIZE MATTER?

Yes it does. Wire size determines the amount of current you can send over a given distance. In the appendix there is a wire chart that shows the wire gauge to use over distance with 100 watts - 8 ohm, 4 ohm and 2 ohm loads. The length of the cable run, size of amplifier and the speaker's handling capacity have to be taken into account.

The following list and rules should help in choosing the wire you need for the job:

1.Always run 1 speaker wire to each speaker cabinet or speaker component. If your cluster is a 2 way system with 2 woofers and 4 horns, run 6 speaker lines. If you have 2 full range speakers in the cluster, run two speaker lines. This is a real asset in trouble shooting your system.

2.Whenever possible, keep your speaker cable runs under 100 feet. Otherwise, use 14 gauge wire on runs 100 feet or less. Use 12 gauge wire up to 200 feet. For longer runs, double up on the 12 gauge wire.

3.Don't use inch tip sleeve jacks for your speakers. Some amplifiers will not tolerate a momentary short on the output of the amplifier. Either the speaker will be damaged or the amp will fail.

4.Do not use 16 gauge wire or smaller for amplifiers with an output of 75 watts or higher.

HEARING IMPAIRED

There are four basic types tems. Each one has an advar most cases, the FM systems do of price but an FM system is not

The four system types are:

1.Hard Wired Systems

2.Loop Systems

3.FM Systems

4.Infrared Systems

Hard Wired Systems are by default the of the 4 systems. However, it is the most res

The design of the system is simple. From t you run a distributed cable under the pews yo covered. At each seating position you mount with a volume control, tone control and headset in This can be a line level system or 70 volt system. W a good quality full ear cup headset, you have the bes signal to noise ratio.

Drawbacks on the system are obvious, you can't move. Therefore, you have to provide many seats with input boxes which drives the cost up. Churches with concrete floors or closed basement ceilings can not use this option. Installation is labor intensive. Cost of the system varies. For 10 people on two rows of pews, you can spend as little as $400 plus installation. A good quality system will cost about $1,000. At 12 seats, a wireless system becomes more attractive.

FM Systems have recently become the most popular system while many public facilities have standardized with infrared systems.

Loop Systems are making a strong comeback. A popular system in the 60s and early 70s, the loop system almost disappeared in churches. In recent years, loop systems have been very effect in simultaneous translation systems, school classrooms and business meetings.

The design of the system begins with an amplifier, a coil of wire around the area people are seated and various types of receivers. A person with a "T" switch on their hearing aid will not need any additional equipment to hear.

Problems with the loop system are frequency range and uneven coverage. Where the loop system wins out is in situations where you need more than one program taking place at the same time. For example, you can have as many simultaneous translation languages you want for as much space you have available. There is no limit. This may mean people have to sit in designated areas but no wires are required. The other advantage of the loop system is privacy. Once you step out of the field there is no further signal pickup. For some churches, this is an important issue.

FM Systems have recently become the most popular system while many public facilities have standardized with infrared systems. FM systems are an alternative to the infrared systems, which are costly in comparison. The FM systems are every bit as good as the infrared system but they have one draw back. When a person leaves the sanctuary, the signal continues. Some FM systems can transmit over 1,000 feet under good conditions. That means someone could leave to go to the bathroom during the service and not miss a word. It also means that you are subject to public airwaves being listened into with radio scanners. This also applies to FM microphones. If privacy is important, FM will not give you that kind of security. For simultaneous translations, you can have up to 32 channels at the same time.

The *infrared system* has been around long enough to become a standard in public places. Some churches choose this kind of system because the theatre or concert hall down the street uses infrared. Most infrared systems seem to be compatible. The infrared system is secure for privacy and it is very good quality sound. Some of the drawbacks are light and line of sight. In some churches with large windows, the sunlight can add noise to the audio signal. Relocating the emitter can help but a second emitter is often the solution. Another problem concerns the elderly.

Some elderly do not stand when everyone else does. Depending on the receiver, a person standing in front of them can block the infrared signal completely.

As a personal preference, the FM system is the best buy where privacy is not a problem. However, newer infrared systems have been coming down in price. The hardwired system is the lowest price as long as volunteers install it. The loop system is very useful if most of your church members already have "T" switches and then they don't need anything attached to them.

Pop = - a father - a soft drink—a lot of air vibrating the surface of the microphone diaphragm generating an undesirable, low frequency rumble or bang. Words with the letter "P" or "B" are often the cause of blowing too much air = pop

OSHA = Occupational Safety and Health

Poor reverberation is usually when reflected energy is focused back onto the microphones in the platform area. This inhibits the choir, organ and sound system.

Note: Some of the best looking and most impressive spec. sheets in the industry have sometimes been found to be the worst sounding speakers.

Shoot out—OK coral—Frisbee contest

—various speakers set on a stage for side by side comparisons. All speakers must be equalized and set at the same volume using pink noise and a SPL Meter.

Sound travels through wire faster than through the air. Sound travels at 1125 feet per second or 660 miles per hour. A frequency is speed of sound divided by the length of the sound wave.

1125 x 10 ft. = 112.5 hertz
1125 x 3 ft. = 375 hertz
1125 x 6 ins = 3,125 hertz

Series wiring is when you take the negative terminal of a speaker and contact it to the positive terminal of the second speaker. This can be with full-range two way speaker boxes or from driver to driver. This is a low cost method of matching an impedance load. 8 ohms series to 8 ohms = 16 ohms. 8 ohms parallel to 8 ohms = 4 ohms. This most often degrades the overall signal because the signal path includes the voice coil and the crossover in two or three way speakers.

Always leave yourself a way out. By not soldering pin 1 to shield, you can easily isolate your audio components to trouble shoot your system for noises, hums and levels.

It was once said that if you bring too much attention to a problem, people will either try to prove you are wrong or they think that you are hiding something. Isn't this like watching someone else burn their hands in the fire and then putting your hands in the fire to see if you will burn too! Is experiencing it for yourself more important than learning from others?

SPL = Sound Pressure Level

Remember, you must always think of the cosmetics. Everything you do must appear as if it was meant to be there and not just something added on.

To be finished.

During the late 50s and early 60s, many churches did live radio broadcasts of the service. This resulted in many churches building an enclosed sound booth combined with the broadcast and live sound. This was not just a compromise, it was a handcuff to both the live and radio sound.

Always leave yourself a way out. By not soldering pin 1 to shield, you can easily isolate your audio components to trouble shoot your system for noises, hums and levels. If you need additional grounding to reduce a specific problem, pin 1 to ground in the right location can make a world of difference. However, if pin 1 is already grounded throughout the system, trouble shooting can be a nightmare.

Tip—If you don't have any 12 gauge wire available for runs up to 200 feet, you can double up on the 14 gauge wire. db

Buyer's Guide

Microphones (including Wireless Microphones), Tape Recorders, Recording Tape and Tape Accessories

On the pages that follow you will find a Guide to Microphones in chart form. This is followed by a Guide to Wireless Mics in paragraph form. This is, in turn, followed by paragraph-form Guides to Recording Tape and Tape Accessories. Manufacturers' addresses conclude the Guides.

As always, be aware that we attempt to contact every manufacturer, but not all are cooperative or prompt enough for our necessary deadlines.

Model	Type	Pat- tern	Freq- uency in dB	Imp- ed- ance	Sens itivity	SPL %dist	Dim LDW	Wgt oz	Finish	Con- nect- ion	Price	Features

AKG

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
C14B/T LII	Cond	4 patt	20-20k	180	12.5	146 0.5	5.6 1.8	11	matte black	XLR	$1,499.00	Electronic version of classic C-12.
C1000	cond	card/ hyper card	50-20k	200	6	137 1.0	8.7 1.3	9.7	grey	XLR	$399.00	Rugged dual-pattern.
D3900	dyn	hype card	40-20k	600	6	147 1.0	7.4 2.1	9.7	matte black	XLR	$319.00	For vocals.
C5900	cond	card	40-20k	200	6	140 1.0	7.4 2.2	9.8	matte black	XLR	$449.00	Vocal performance-condenser quality..
C522M/S	cond	mid/ side	50-20k		6	140 1.0	5.4 2/1.1	12.4 10	matte black	XLR	$1,699.00 to $2.700	MS-Stereo mic.
Blue Line	cond	various patt	20-20k	200	10- 25	120- 134 1	2-10 0.3 0.7	0.2- 2.8	blue matte	XLR	$179.00 to $458.00	Modular mic line with snap-together design.
Micro Mics	cond	various patt	20-20k	200	4- 13	120- 133	0.6- 9.3	1.4/ 4.6	matte black	XLR 3.5- mini	$89.00 to $299.00	Miniature condenser mics.

AUDIO-TECHNICA US See our ad on Cover 2

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
ATM63 HE	dyn	hyper card	50-18k	600	48			9.3 150 0.87	grey	XLR	$190.00	High output and extended response.
ATM61 HE	dyn	hyper card	50-18k	600	55			9.7 1.98 0.87	grey	XLR	$250.00	High gain before feedback, extended response and reduced coloration off axis.
ATM41 HE	dyn	hyper card	50-17k	600	55		2.12	10.3 10.3 0.87	grey	XLR	$210.00	High output with low handling noise. Clear even when used ultra close.
AT4051	cond	card	20-20k	250	35	143	6.10	4.2 0.83	black	XLR	$610.00	Internally direct coupled, uniform polar pattern at all frequencies.
AT4031	elect	card	30-20k	200	46	145 1	6.28	4.9 .083	black	XLR	$330 00	High output and very high SPL handling, uniform polar patterns at all frequencies.
AT4033/ SM	elect	card	30-20k	100	28	140 1	6.73	14.5 2.10	black	XLR	$699.00	The element is suspended in the center of the acoustically-transparent grille, near perfect 90 deg off-axis response, includes shockmount.
AT822	elect	X/Y stereo	30.20k	200	44	125 1	7.76 2.44	5.8	grey	minis phone	$299.00	Uniform polar response vs. frequency for realism and spacial impact.
AT825	elect	X/Y stereo	30-20k	200	46	126 1	8.43 2.44	8.5	grey	2-XLR	$399.00	Full mono compatibilty,Switchable low cut, two-way power.

BEYER DYNAMIC,INC

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
M424	dyn	hyper card	40-16k 2	200	52	144 1	3 0.64	6	black alum	XLR	$149.95	Miniature, high SPL without distortion.
M420	dyn	hyper card	100-15k 2	200	58	140 1	4 0.64	6.2	black alum	XLR	$249.95	As above.
MC740	cond	5 pos	40-20k 2	150	40	144 1	8.5 2	8.2	black alum	XLR	$1,524.95	3 position rolloff, 10dB pad, large diaphragm.
MC740	cond	multi	40-20k 3	150	46	144 1	8 3.5	10	black alum	XLR	$1,524.95	3 pos. rolloff,switchable 10 dB atten.
TG-X 180	dyn	hyper card	40-16k 3	290	54	140 1	6 2.5	4.5	black anad.	XLR	$179.00	Small, high output.
TG-X 280	dyn	hyper card	30-16k 3	290	54	140 1	6 2.5	4.5	black anad.	XLR	$229.00	Vocal instrument applications.
TG-X 480	dyn	hyper card	40-18k 3	280	50	140 1	8 3	6	black anad.	XLR	$289.00	Vocal mic with pronounced proximity effect.
TG-X 580	dyn	hyper card	30-18k 3	280	50	140 1	8 3	6	black anad.	XLR	$349.95	Fast transients and extended frequency response.

BRUEL AND KJAER, TGI NORTH AMERICA

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
4011	cond	card	40-20k 2	180	40	110 0.5	6.75 .75	5.8	black chrm	XLR	$1,800.00	Transformerless,phantom powered.Switchable 0,20dB attenuator. Flat response both on and off-axis.
4003	cond	omni	10-20k	30	24	135	6.5	5.3	black	4pin	$1,660.00	Dynamic range from 15 to 154dB SPL(A)typical.

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
4006	cond	omni	2 20-20k	30	36	1.0 135	.63 6.5	5.3	chrm black	XLR XLR	$1,660.00	Uses 2812 power supply. Dynamic range from 15 to 143dB SPL(A)typical.
4007	cond	omni	20-40k 2	30	50	1.0 148 1.0	.63 6.5 .63	5.3	chrm black chrm	XLR	$1,660.00	Uses 2812 power supply. Dynamic range from 24 to 155dB SPL(A)typical. Phantom powered.

CROWN INTERNATIONAL

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
PZM-30 D	elec cond	hemi	25-20k 6	240	65	150 3	5 6	6.5	black	XLR	$349.00	Switchable flat or rising high freq resp.
PZM-6-D	elec cond	hemi	20-20k 3	240	67	150 3	6 5	6.5	slvr	XLR	$349.00	As above.
SASS-P Mk II	elec	omni/ uni	20-20k 3	240 x2		150 1	11.5 5.7	17	black/ grey	XLR	$899.00	Stereo PZM mic
PCC-170	cond	super card	50-20k 3	150	30.5	130 1	4.8 3.4	6	black	XLR	$295.00	Directional boundary mic, available with programmable memrane switch and adjustable gate.
CM-31	cond	super card	40-20k 3	150	35 1	130 0.5	1.4	4	black	XLR	$240.00	Miniature mic, for choir, orchestra, stage suspension.
LM-300	cond	super card	80-15k 3	150	42	130 1	1.4 0.5	5	black	XLR	$247.00	17-in.goosneck, lecturn mic. available as LM-300L—. 22-in and LM-301—19-in. with 5/8-in threaded mounting.
PZM-11	elect	hemi	80-20k 3	225	45	120 3	4.5 2.75	2.5	off white	screw term.	$67.00	Security and survellance, looks like an electrical wall switch.

COUNTRYMAN ASSOCIATES

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
Isomax IV	elec cond	hyper card	70-18k	270	52	145	12 18 24		black	XLR	$395.95	Gooseneck mic, internal vibration isolation.
Isomax II	elec cond	card/ omni/ hyper/ bi-dir	20-20k 1	600	57	150	5.18 0.64		black	XLR	$241.45	4-pattern selectable
Isomax Head set	elec cond	hyper card	20-20k	600	57	150			black		$272.95	Small, lightweight adustable headset mic. Also available as wireless, same price.
Isomax EMW	elec cond	omni	20-20k	600	57	150	1.04 0.26		black +	XLR	$230.00	Small lavalier, available several colors, also as EMW Wireless at $157.45.

ELECTRO-VOICE, INC. (A MARK IV COMPANY)

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
N/D 408B	dyn	card	60-22	150	51		2.75 2.85	6.7	blk	XLR	$247.00	Pivoting instrument mic with special element for wide freuency response and high output.
N/D 457B	dyn	card	55-21	150	51		7.12 2.05	7.05	blk	XLR	$245.00	Hand-held vocal mic with hypercardioid pattern for very high gain before feedback.
N/D 757B	dyn	card	50-22	150	51		7.12 2.05	7.7	blk	XLR	$314.00	Hand-held vocal mic with extended frequency response, switchable low-frequency roll-off filter and special element.
N/D 857B	dyn	super	50-22	150	50		7.40 2.05	7.9	blk	XLR	$425.00	Acoustical path corrector provides increased sens. and a uniform polar pattern.
N/D 357B	dyn	card	55-20	150	53		7.12 2.05	7.05	blk	XLR	$206.00	Works well with both live and studio applications.
N/D 257B	dyn	card	65-19	150	53		7.12 2.05	7.05	blk	XLR	$147.00	Offers 3 dB more output than others in its class.
N/D 308B	dyn	card	65-19	150	53		2.75 2.85	6.7	blk	XLR steel	$213.00	Ideal for kick drum and other percussion applications as well as guitar amps.

HM ELECTRONICS,INC

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
HM58	dyn	card	80-14k	600	75		6.6 2.0	10	silver abs	XLR	$29.00	Designed for high quality professional applications.
RM77	elec cond	uni	150-15k	800	72		7.5 2.0	11	grey zinc	XLR	$29.00	Built-in pseudo-reverb with variable control.

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features

MTG (GOTHAM TECHNOLOGY GROUP)

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
UM925	cond	3 pat	40-18k	200		120 0.5	7.5	15	black	XLR	$1,995.00	3 patterns utilizing large dual membrane capsule,vacuum tube electronics, black or nickle finish
UM705	cond	3 pat	40-18k	200		123	8.7 0.5 1.7	10 1	black	XLR	$1,150.00	3 patterns, utilizing large dual membrane capsule m7, solid-state electronics, black or nickle finish.
UM70	cond	3 pat	40-18k	200		125 0.5	8.7 1 1.7	10	black	XLR	$995.00	As UM705 above.
M715	cond	card	40-18k	200		123 0.5	8.7 1 1.7	10	black	XLR	$895.00	As UM705 above but single pattern.
SMS200	cond	card	40-20k	150		134 0.5	6.5 0.8	4.5	black	XLR	$595.00	Small single membrane capsule with ceramic electrode, interchangeable capsule, black or nickle finish.
SMS210	cond	super card	40-20k	150		134 0.5	6.5 0.8	4.5	black	XLR	$625.00	As SMS200 above.
SMS270	cond	omni	20-20k	150		138 0.5	6.5 0.8	4.5	black	XLR	$575.00	As SMS200 above except flat response in the difuse field.

MILAB

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
VIP-50	cond	var pat		200		112 1	6.5 2	14	black alum	XLR	$1,495.00	Switchable pad, HP filters, large dual membrane capsule.
DC-96B	cond	card	20-20k	200		118 1	5.7 1	7	black alum	XLR	$795.00	Lage dual membrane capsule.
VM-44	cond	card	20-20k	200		128 1	5 1	4.6	black alum	XLR	$615.00	Switchable pad.
LSR 2000	cond		20-20k	200		133 1	7.25 1			XLR	$675.00	Switchable pad.
MP-30	cond	omni	40-20k	200		110 1	1 3	2.3		XLR	$385.00	PZM style, Crown licence.

NADY SYSTEMS, INC.

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
SP1	dyn	card	70-15k	500			7.2 7.2	9.3	grey	XLR	$84.95	Also available as SP1c (unidyne) at $89.95.
SP2	dyn	card	70-15k	500			7.2	11.35	gray	XLR	$84.95	
SP4	dyn	card	50-18k	250			7.2	11/	black	XLR	$149.95	Neodymium mic.

NEUMANN

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
TLM-170	cond	multi	20-20k	50	41.9	150 0.5	6.0 2.4	22	black	XLR	$2,450.00	Multi-pattern mic employs FET100 technology
RSM-1915	cond	shot gun	20-20k	50	32.8	134 0.5	8.4 1.2	6.0	matt	XLR	$4,050.00	Transformerless M-S/X-Y stereo shotgun mic with active matrix.
KM100 SERIES	cond	multi	20-20k	50	38.4	148 0.5	3.6 0.8	2.8	black	XLR	$950 00 to $2,590.00	Capsule and output stage can be separated to allow interconnection of accessories.
U 87 Ai	cond	3 pat	20-20k	200	31.9	117 0.5	7.9 2.2	17.6	black	XLR	$2,375.00	Multi pattern studio mic with 1o dB atten, low-freq rolloff, pattern selection.
U 89i	cond	5 pat	20-20k	150	41.9	134 0.5	7.3 1.8	14.1	black	XLR	$2,450.00	Multi-pattern mic handles high SPLs, bass rolloff.
SM 69 FET	cond	5 pat	20-20k	200	34.4	110 0.5	10.2 1.9	16.4	black	DIN 12	$4,080.00	Dual capsule stereo M/S and X/Y can be rotated to vary steeo effect.
KMR 81i	cond	shot gun	20-20k	150	34.9	128 0.5	8.9 0.8	5.1	black	XLR	$1,375.00	M/S stereo shotgun includes MTX 191 matrix, high-pass filter.
KMR 82i	cond	shot gun	20-20k	150	34.9	128 0.5	15.5 0.8	8.8	black	XLR	$1,525.00	Shogun has high directional efficiency, hig and low freq. rolloff switches, includes a windscreen.

PEAVEY ELECTRONICS

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
PVM38i	dyn	card	40-16k 3	300	56	150 0.1	5.75 1.43	7	black matte	XLR	$149.99	Unidirectional hand held vocal mic.
PVM45i	dyn	hyper card	50-16k 3	300	56	150 0.1	5.75 1.43	7	black matte	XLR	$179.99	Unidirectional instrument mic.
PVM80	neo	card	50-16k	300	52	150	5.75	14	black	XLR	$179.99	Hand held vocal mic, uniform polar response.

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
	dyn		3			0.1	1.87		matte			
PVM480	cond	card	40-20k 3	500	48	128 0.1	5.39 0.87	2.1	black satin	XLR	$239.99	Electret condenser vocal/instrument mic.
PVM520i	dyn	card	45-19k 3	400	52	150 0.1	4.8 1.95	10	black satin	XLR	$329.00	Rugged flite case included, high-performance instrument mic.

SENNHEISER ELECTRONIC CORPORATION

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
MD431	dyn	super	40-16k	200	57	120 1.0	7.88 1.25	8.8	black alum	XLR	$479.00	High gain-before-feedback, handles high SPL. Triple-layered steel mesh grill. Magnetic reed on/off switch.
MKE -4032	elect cond	super	70-20k	200	46 1.0	140 1.9	8.1	7.5	black alum	XLR	$649.00	12-48 V phantom or AA battery operation. Built-in blast filter, shock mount. Rugged, handles high SPL.
MKE-2	elect cond	omni	40-20k	200 1000	46	126 1.0	0.43 0.23	0.1	black	XLR	$285.00	Two impedance options, fleshtone color option. Small, ultra light for broadcast, church and theatre in wireless system.
MD 518	dyn	card	50-16k	200	58	120 1	7 1.25	6.5	black	XLR	$229.00	Handheld mic, high SPLs, smooth pickup.
MKE 300	elect	super	150-17k	200	35.9		9.1	2.1	black	3.5 mm	$225.00	Ideal for camcorders, integrated windscreen,.

SONY PRO-AUDIO

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
C-48	cond	multi	30-16k	150	39	128 1.0	2.2 9.1	20	satin nickel	XLR	$1150.00	Selectable patterns,10 dB pad,lo-cut switch, 9 V battery or phantom power. Vibration-proof structure.
C-535P	cond	card	30-16k	200	40	138 1.0	0.8 6.1	4.9	black alum	XLR	$555.00	Slim-line design with 10 dB pad. Rejects SCR, TV and other electronic noise. Excellent transient response.
ECM-MS5	cond	elec card	stereo	70-20k		150 1.0	40 8.4	1.9	alum	XLR	$1340.00	Three capsule design for M-S recording. Built-in M-S matrix field-rugged construction. 12-48 V phantom powered.
F-730	dyn	card	50-11k	300	60		1.7 6.5	8.8	black alum	XLR	$153.00	For vocal recording, offers extra punch in low range.

SHURE BROTHERS, INC. See our ad on Cover IV

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
BETA58	dyn	super card	50-16k	150 290	71.5		6.38 2	9.3	slvr blue	XLR	$266.00	Three-stage directional tuning network,advanced shock isolation system,humbucking coil,rugged steel grille.
BETA57	dyn	super card	50-16k	150 290	71		6.18 1.5	9.2	slvr blue	XLR XLR	$258.00	same as the BETA 58 but designed for musical instrument mic'ing,features smooth wide response.
VP88	stereo cond	multi	40-20k	150 100	66	129 1	11.43 1.56	14.7	black alum.	5-pin XLR	$995.00	MS stereo, mono compatible,built-in left-right stereo matrix adjustable side level switch, internal or phantom power.
SM81LC	cond	card	20-20k	150 85	40	146 1.0	8.3 0.9	8	black	XLR	$441.50	Ruler flat response, phantom power only.
Beta 87	cond	super card	50-18k	150	54	134				XLR	$420.00	Uniform supercardioid pattern throughout frequency range,.
SM91A	cond	hemi	20-20k	150 90	45	144 0.1	0.6 3.7 5.0	9.3	black cast steel	XLR	$310.00	Low profile boundary effect mic. External pre-amp with 12 dB per octave roll-off switch. Accepts battery ot phantom power.
SM98A	cond	card	40-20k	150	54	153 0.1	1.2 0.5	0.4	black brass	XLR	$291.50	Full-range response in miniature unit. Many optional mounting accessories.
SM57LC	dyn	card	40-15k	150	75		6.22 1.25	10	grey	XLR	$147.00	The standard for percussion and instrument mic'ing.

YAMAHA MUSIC CORPORATION

Model	Type	Pattern	Frequency in dB	Impedance	Sensitivity	SPL %dist	Dim LDW	Wgt oz	Finish	Connection	Price	Features
MH100	elec	card	10-10k	1.6k	70					phone jack	$49.00	Headset and microphone in one unit. Uses lightweight pads that are easy on ears even with extended use.
MZ101	dyn	card	40-17k	250	76		6.2 0.9		mett brown	XLR	$135.00	Noted for its clean mid-range and high-end. Poly-laminate diaphragm. Unique 3-point suspension. Gold-plate connectors.
MZ102BE		dyn	card	40-18k		250	76 0.9	6.2	mett brown	XLR	$190.00	Deep,lower mid-range quality. Beryllium diaphragm for tight response. Die-cast zinc body. Gold-plated connectors.
MZ103BE		dyn	card	40-18k		250	76 0.9	6.1	mett grey	XLR	$235.00	Wide range resistance to off-axis sound. Beryllium diaphragm, 3-point suspension and gold plated connectors.
MZ104	dyn	card	30-17k	250	77		7.0 1.4		mett brown	XLR	$145.00	Good instrument mic. Good bass response. Lowered sensitivity to avoid hig SPL overload.Gold-plated connectors.
MZ105 BE	dyn	card	40-18k	250	77		6.0 1.4		mett brown	XLR	$200.00	Designed to avoid unwanted bass buildup with close mic'ing. Beryllium diaphragm. Gold-plated connectors.
MZ106S	dyn	card	40-18k	250	77		7.2 2.0		mett grey	XLR	$140.00	Ideal for vocal use. On/off switch with switch lock for lock-On. Two-layer laminated polyester film diaphragm.
MZ205BE		dyn	card	40-18k		250	77 1.3	.3	mett grey	XLR	$295.00	Vocal microphone with right-angle XLR connector, Beryllium diaphragm. 3-point suspension. Gold-plated connectors.

Wireless Microphones

AUDIO-TECHNICA US See our ad on Cover II

ATW-1031 Wireless Microphone System Includes a model ATW-R10 receiver and a model ATW-T31 UniPak body worn tranmitter with individual controls for adjusting the sensitivity of the microphone and musical instrument inputs. Available wih a wide range of Audio-Technica microphones, including ATM73cW and ATM71cW head-worn, AT831cW cardioid lavalier, ATM35cW miniature instrument mics and MT830cW subminiature Condenser for theater and lavalier use

PRO 88W Wireless Microphone System is equipped with a flexible detachable antenna and a choice of 2 different transmitter and receiver frequencies on each system. The small and inconspicuous PRO 88 wireless system will mount on the light shoe of camcorders or on the back of a camera itself with a velcro attachment. Eight VHF channels are available for interference-free operation in virtually any location.

ATW-1032 Wireless Microphone includes a model ATW-r10 receiver and ATW -T32 handhel transmitter, with unidirectional dynamic element

BEYER DYNAMIC, INC.

SEM 700 is a condenser UHF wireless system with variable gain 450-980MGZ and supercardioid pattern.

Custom quote varies by number of channels.

S170P is a condenser VHF diversity body-pack wireless system with omni or cardioid patterns.

Price: $1499.95.

TS 190 is a condenser VHF diversity body-pack wireless system with omni or cardioid patterns.

Price: $1799.95.

TS 900 is a condenser UHF body-pack wireless system with omni or cardioid patterns.

Custom quote varies by # of channels.

S170H is a dynamic VHF wireless mic with noiseless mute, rack mount receivers with TGX 480 head. It has a hypercardioid pattern, frequency response 40-18 k, impedance 290ohms, sensitivity 50, sound pressure level 140 at 1% distortion, dimensions inlength 8-in. width 2.5-in., weight 9.5 ounces, black aluminum finish, wireless connector.

Price: $1499.95.

SDM 186 is a dynamic TGX 480 head VHF diversity wireless mic, interchangeable heads, variable gains, LED metering, rack mount receviers, hypercardioid pattern, frequency response 40-18k, impedance 290ohms, sensitivity 50, sound pressure level 140 at 1% distortion, length 8.5-in. width 2.5-in., 9.7 ounces, black aluminum finish, wireless connector.

Price: $1499.95.

SEM 186 is a condenser MC0 81 head VHF diversity wireless mic, interchangeable heads, variable gain, LED metering, rack mount receivers, supercardioid pattern, frequency response 50-18k, impedance 190ohms, sensitivity 50, sound pressure level 138 at 1% distortion, length 8.5-in. width 2.5-in., weight 9.7 ounces, black aluminum finish, wireless connector.

Price: $1499.95.

SDM 700 is a dynamic TGX 480 head UHF diversity wireless mic, interchangeable heads, variable gain 450-980 MgHz, hypercardioid pattern, frequency response 40-18k, impedance 290ohms, sensitivity 50, sound pressure level 140 at 1% distortion, length 8.5-in. width 2.5-in., weight 9.7 ounces, black aluminum finish, wireless connector.

Price: custom quote by # of channels.

ELECTRO-VOICE

The **MS-2000A dual-receiver** diversity systems are available with handheld, bodypack, and professional guitar transmitters. Precisely engineered audio and rf circuits and exclusive DNX companding result in a signal-to-noise ratio of 105 dB for silent operation—eliminating "noise tails" or compander "breathing." **The MR-2000A rack-mountable receiver** has both quarter-inch and balanced XLR outputs, switchable mic/line and with 30dB of level adjustment. A switchable internal power supply permits use anywhere in the world.

The **MT-2000A handheld transmitter** features the world-class N/D7578 N/DYM head for full, rich vocals with virtually no handling noise. Transmitter controls include on/off and audio mute switches and an audio level adjustment of 30 dB. Two LED's indicate on/off and battery status.

The **MB-2000A bodypack** has a TA4M 4-pin mini-XLR connector to accept a variety of lavalier microphones, including EV's premium C0100 omni and CS200 uni condenser mics. Separate on/off and mute switches, and 30 dB of audio level adjustment, give the user precise control. The multi-function LED indicates power on/off, battery condition, and overload distortion.

The GT-1000 professional guitar transmitter was designed specifically for stringed instruments, and two years of use by major touring acts have lead to it being considered the best-sounding guitar/bass wireless available—giving the sound and "feel" of a cable. The rugged bodypack features separate power on/off and mute switches, and 12 dB of level adjustment to match the pickup's output. The quarter-inch to mini-XLR cable is detachable. The GT-1000 transmitter is used with the GR-2000A receiver, a variation on the MR-2000A.

HM ELECTRONICS

System 50 Body-pac Wireless Mic System

Includes: RX520 Receiver and TX550 Transmitter

The System 50 offers a switching diversity receiver and comes standard with a mic-mute switch and low-battery indicator. Price: $733.25.

System 55 Handheld Wireless Mic System

Includes: RX520 Receiver and TX555 Transmitter

The System 55 offer excellent audio quality, a switching diversity receiver, locking mic-mute switch and a low-battery indicator.

Price: $743.25

System 515 Body-pac Wireless Mic System

Includes: RX522 Receiver and TX550 Transmitter

Intended for cost-effective professional applications; for portable or fixed installations; rack-mountable; lightweight; operates on AC or DC power and includes noise reduction circuitry.

Price: $525.50

System 525 Handheld Wireless Mic System

Includes: RX522 Receiver and TX555 Transmitter

Intended for cost-effective professional applications; for portable or fixed installations; rack mountable; lightweight; operates on AC or DC power and includes noise reduction circuitry.

Price: $599.50.

NADY SYSTEMS, INC.

Nady RW-1

Nady's most affordable professional rack-mounting VHF system. Features True Diversity reception, balanced and unbalanced output, front or rear mount antennas and Nady's new Surface Mount Technology (SMT) transmitters. Choose handheld or lavalier mic.

Prices start at $609.95.

Nady 401

The Nady 401 features four independent VHF receivers in one rack mount component, with any combination of Nady's new, all-metal HT-10 handheld transmitters or Nady LT-10 lavalier transmitters. More than 200-feet operating range, 120 dB dynamic range.

Prices start at $1399.95.

Nady 650

The mid-priced VHF rack-mount system with sophisticated filtering—up to ten 650 systems can operate in the same location simultaneously. Features True Diversity, balanced/unbalanced output, 120 dB dynamic range and Nady's new SMT transmitters, Handheld or lavalier mic.

Prices start at $669.95.

Nady 750

The Nady 750 features two independent VHF True Diversity receivers in one rack mount component, with balanced and unbalanced output for each, plus any combination of two Nady SMT lavalier or handheld microphone transmitters.

Prices start at $1299.95

Nady 2000

Nady's new top-of-the-line VHFsystem features hiss mute circuitry to maintain audio quality at the limit of the 2000's operating range, state-of-the-art handheld and lavalier transmitters, ultra-sophisticated filtering for up to 20 system simultaneous operation, bass boost and much more.

Prices start at $1599.95.

Nady 301 UHF

The ultra-affordable UHF wireless system, with four user-switchable UHF operating channels on the receiver and transmitters—Nady's SMT lavalier bodypack, or new HT-50 handheld mic. Features True Diversity, frequency synthesis and 120 dB dynamic range.

Prices start at $829.95.

Nady RW-3 UHF

A professional rack mount True Diversity system with four frequency synthesized, user-switchable UHF operating channels on the transmitters as well as the receiver. Also features balanced and unbalanced output and front or rear mount antennas.

Prices start at $1,099.95.

Nady 950 UHF

Nady's top of the line UHF system, with ten user-switchable channels on the receiver and transmitters. Features True Diversity, 120 dB dynamic range, bass boost, balanced/unbalanced output. Mics feature all-metal construction. Also available with 40 channels.

Prices start at $2,499.95.

SENNHEISER

VHF 1H

SKM4031-90 handheld microphone transmitter, EK2012-90 miniature receiver on VHF carrier frequency for ENG/EFP applications.

Price: $3245.00.

VHF 1B

SK2012-90 body pack transmitter with MKE2-2-R RD lavalier microphone and EK2012-90 miniature receiver on VHF carrier frequency for ENG/EFP applications.

Price: $3920.00.

VHF 2H

SKM4031-90 handheld microphone transmitter, EM2003-90 diversity receiver with ground plane antennae on VHF carrier frequency.

Price: $3651.00.

VHF 2B

SK2012-90 body pack transmitter with MKE2-2-R RD lavalier microphone, EM2003-90 diversity receiver with ground plane antennae on VHF carrier frequency.

Price: $4326.00.

UHF 2H

SKM4031-TV body pack transmitter with MKE2-2-R RD lavalier microphone and EM2003-TV diversity receiver on UHF carrier frequency.

Price: $5835.00.

UHF 2B

SK2012-TV body pack transmitter with MKE2-2-R RD lavalier microphone and EM2003-TV diversity receiver on UHF carrier frequency.

Price: $6615.00.

UHF 2EH

SKM4031-TV handheld microphone transmitter and battery operated EM2003-TV diversity receiver on UHF carrier frequency, supplied in canvas carrying bag.

Price: $6525.00.

UHF 2EB

SK2012-TV body pack transmitter with MKE2-2-R RD lavalier microphone and EM2003-90 diversity receiver operating on UHF carrier frequency, supplied in canvas carrying bag.

Price: $7305.00.

SHURE BROTHERS INCORPORATED See our ad on Cover IV

WM98 The world's finest miniature wireless musical instrument microphone. Extremely uniform cardioid pick-up pattern and a wide-range frequency response. The WM98 is a miniature condenser microphone, frequency response is 40-20,000, impedance 1200, sensitivity is -74dB, maximum SPL 144dB, length is 1 1/4 and width is 15/32 with a weight of 0.4 ounces. The finish is a Matte Black enamel, connector Tini 4-pin.

Price: $170.00.

WL84 Miniature, lavalier electret condenser microphone with a supercardioid pickup pattern. Designed for high-quality wireless applications involving speech. The WL84 has a supercardioid pattern, with a frequency response of 50-16,000. Impedance is 1200, with a -68dB sensitivity. Maximum SPL is 138dB, length is 1 1/32 and width is 7/16 with a weight of 0.21 ounces. The finish is Silver Blue enamel, connector Tini 4-pin.

Price: $125.00.

WL83 Miniature, lavalier electret condenser microphone with a supercardioid pickup pattern. Designed for high-quality wireless applications involving speech. It is omnidirectional with frequency response of 50-16,000, impedance 1200, sensitivity -65.5dB. Maximum SPL is 136dB, length is ¾ and width is 7⁄16 with weight of 0.21 ounces. The finish is Matte Black enamel, connector Tini 4-pin.

Price: $100.00.

WCM16 Head-worn electret condenser microphone intended for wireless use by performers, musicians, and lecturers requiring high quality voice pickup. It has hyper cardioid polar pattern, frequency response 50-18,000, impedance 1200, sensitivity -75dB, maximum SPL 150 dB, dimensions are length ¾ and width 7⁄16 with weight of 1.27 ounces. The finish is Matte Black enamel, connector Tini 4-pin.

Price: $250.00.

839W Miniature, lavalier electret condenser microphone with an omnidirectional pickup pattern. Designed for use with Shure Wireless microphone systems. Frequency response 50-16,000, impedance 1200, sensitivity -64.5dB, maximum SPL 135 dB, length ¾, width 7⁄16, weight 0.21 ounces, Platinum beige finish, connector Tini 4-pin.

Price: $85.00

L2/58 All the performance and features of the Classic SM58 in a wireless.

L2/Beta 58 All the performance and features of the Beta58 in a wireless. Excellent for vocal applications, particularly those involving high sound pressure levels requiring high volume monitoring.

L2/Beta 87 Excellent quality condenser microphone offered in a wireless version. Flat response with a slight presence rise and low-end roll-off make it perfect for demanding vocal applications.

SONY PRO AUDIO

C-800

Large Diaphragm tube condenser microphone intended for musical instrument applications. Ideal for very critical recording and can handle high input SPL. Cardioid and omni patterns, Frequency Response 20 Hz-18 kHz, Impedance 250 Ohms, Sensitivity -46 dB/PA, Sound Pressure Level 150 dB SPL, Dimensions: 0 2 ¼ x 7 ¾-in.,

Wt. 11 lbs 4oz., Black aluminum finish, XLR connector.

Price: $4,665.00.

C-800G

Large dual diaphragm tube condenser microphone intended for vocal applications. Incorporates the worlds first tube cooling system in a microphone. The ultimate in transducer technology. Cardioid/Omni patterns, Frequency Response 20Hz-18kHz, Impedance 100 Ohms, Sensitivity -33dB/PA, Sound Pressure Level 134 dBSPL, Dimensions: 0 2 ¼ x 7 ⅝-in. x 9 ½, Wt. 1 lb. 14 oz., Black Aluminum finish, XLR connector.

Price: $5,945.00.

C-48

Large dual diaphragm FET condenser microphone intended for various applications. Features multi-pattern selectivity as ell as 48 V phantom & 9 V battery operation. Cardioid/Bi/Omni patterns, Frequency Response 30 Hz-16 kHz, Impedance 150 Ohms, Sensitivity -38.8dBm, Sound pressure level 138 dBDPL, Dimensions: 2 ¼ x 9 ⅛-in. x 1 ⅝-in., Wt. 1 lb. 4 oz., Silver aluminum finish, XLR connector.

Price: $1,155.00.

C-535P

Small diaphragm FET condenser microphone intended for musica linstrument applications. Features a 10 dB pad and high quality gold spattered diaphragm. Cardioid pattern, Frequency Response 30Hz-16kHz, Impedance 200 Ohms, Sensitivity -40.0 dBm, Sound Pressure level 148 dB SPL, Dimensions: 27⁄32 x 6 ⅛-in., Wt. 4.9 oz., Black Aluminum finish, XLR.

Price: $555.00.

C-536P

Small diaphragm FET condenser microphone intended for musical instrument applications. Features a 10 dB pad and high quality gold spattered diaphragm. Right angle pick-up pattern. Cardioid pattern, Frequency Response 30Hz-16kHz, Impedance 200 Ohms, Sensitivity -40.0 dBm, Sound pressure level 148 dBSPL, Dimensions: 27⁄32 x 6 ⅛-in., Wt. 5.3 oz., Black Aluminum finish, XLR connector.

Price: $555.00.

ECM-23F3

Budget minded small diaphragm electret condenser microphone intended for various applications. Features "AA" battery operation and supplied windscreen and cable. Cardioid pattern, Frequency Response 20 Hz-20 kHz, Impedance 200 Ohms, Sensitivity -47.0 dBm, Sound pressure level 134 dB SPL, Dimensions: 1 ⅛ x 7 ½, Wt. 7.6 oz., Grey aluminum finish, XLR connector.

Price: $270.00.

ECM-MS5

Three capsule electret condenser MS stereo microphone intended for a variety of stereo applications including sampling, music recording and video production. Features built-in matrix. Cardioid stereo M-S pattern, Frequency Response 70 Hz-20 kHz, Impedance 150 Ohms, Sensitivity -37.0 dBm, Sound Pressure Level 130 dB SPL, Dimensions: 1 ⅞ x 8 3/8-in., Wt. 7.6 oz, Grey aluminum finish, XLR connector.

Price: $1,340.00.

F-730

General purpose dynamic microphone intended for hand-held vocal applications. Features include built-in on/off switch, integrated pop filter and replaceable capsule. Cardioid pattern, Frequency Response 50 Hz-11 kHz, Impedance 300 Ohms, Sensitivity -59.8 dBm, Dimensions: 0 1 ¾ x 6 ⅝-in., Wt. 8.8 oz., Black finish, XLR connector.

Price: $153.00.

VEGA, A MARK IV COMPANY

AX-20 VHF band wireless microphone system. Dual receiver, true diversity wireless microphone system features special noise sensing squelch circuitry for false signal control. Sensitive and selective receiver enables 25 systems to operate in one area. Transmitter/receiver system with accessories $1349.00.

"600 Series" UHF wireless system. 494-704 MHz, 150 milliwatts RF power output. Dual receiver, true diversity. 2000 feet of range and 108 db signal-to-noise ratio. Full line of accessories including directional antennas and antenna splitters.

Price: $4500.00.

Q PLUS wireless intercom system for backstage sound and lighting crew operation. Six full duplex, hands-free belt pack units can operate in one area simultaneously. 1000 feet of range and broadcast quality audio between base station which can interface to a wired intercom and program audio feed.

Price: $10,000.00.

Open Reel and Cassette/R-DAT Recorders and Accessories

FOSTEX CORPORATION OF AMERICA

Open Reel Machines

The **R8** is an 8-channel/8-track reel-to-reel recorder. It is a 2-head machine running ¼ in. tape at 15 in./sec. with a frequency response of 40 Hz to 18 kHz, a flutter rate of 0.06 percent and a S/N ratio of 78 dB. THD is 1 percent at 1 kHz.

Price: $2,800.00

The **E2** is a 2-channel/2-track recorder. It is a 3-head machine running ¼ in. tape at 15 in./sec. with a frequency respone of 30 Hz to 20 kHz, a flutter rate of 0.05 percent, and a S/N ratio of 74 dB. THD is 1 percent at 1 kHz.

Price: $3,795.00

G-16S is a 16 channel/16 track reel-to-reel recorder. It is a 2-head machine running ½ in. tap at 15 in./sec. with a frequency respone of 40 Hz to 18 kHz, a flutter rate of 0.05 percent WTD and a S/N ratio of 80 dB with built-in Dolby S NR. THD is 1 percent at 1 kHz.

Price: $8,500.00

G-24S is a 24-channel/24-track reel-to-reel recorder. It is a 2-head machine running 1 in. tape at 15 in./sec. with a frequency respone of 40-18 kHz, a flutter ratio of 0.05 percent, WTD, and a S/N ratio of 80 dB with built-in Dolby S NR. THD is 1 percent at 1 kHz.

Price: $14,500.00

Cassette Recorders

Model X-18 is a 4-input, 4-track portable (battery) recorder. It is a 2-head machine running at 1-⅞ in with a frequency response of 40-12.5k and a s/n ratio of 58 dB wtd Dolby B.

Price $399.00

X-28H is an 8-input, 4-track dual speed recorder. It is a 2-head machine. At 3¾ the frequency response is 40-14k, the flutter rate is ±0.07% and a s/n ratio of 58 dB wtd Dolby B.

Price $599.00

Model 280 is an 8-input, 4-track automatedrecorder. It is a 2-head machine. At 3¾ the frequency response is 40-14k, the flutter rate is æ0.07% and a s/n ratio of 65 dB wtd Dolby C.

Price $849.00

DAT Recorders

Model 10 is a stero master recorder with instart and RAM scrub. It is a 2-head machine with a frequency response of 20-20k ±0.5 dB and a s/n ratio/dynamic range of 92 dB.

Price: $2,850.00

Model D-20B is a stereo master recorder with on-board timecode and chase/lock sync. It is a 4-head machine with a frequency response of 20-20k ±0.5 dB and a sn ratio/dynamic range of 92 dB.

Price: $8,500.00

Model PD-2 is professional portable timecode recorder. It is a 4-head machine with a frequency response of 20-20k ±0.5 dB and a sn ratio/dynamic range of 92 dB. Weight is12 lbs with battery.

Price: $11,495.00

JRF MAGNETIC SCIENCES See our ad on page 4

T-Bar head adjustable mount has conventional bottom mount head assemblies converted to a top mount T-Bar which offers azimuth and wrap adjustments. The T-Bar mount will reduce mechanical head alignment time.

Price: (includes installation and head reconditioning) $250.00 to $450.00

PLX magnetic heads are available for tape machines such as Teac ½ in. 80-8, 1 in. 85-16; Otari ½ in. (4 and 8 track) MX5050, ½ in. (4 track) MTR10/12; Ampex 350, 440 ¼ in. (mono and 2 track) and MM1100/1200 (24 track); Studer A80 and A800 (24 track); Mincom M56 and M79 (16 and 24 track); MCI/Sony ½ in. (2 track) 2 in. (24 track).

Price: available upon request

Time-code retro-fit kits are available for many ¼ in. tape machines. Features include center track time code record and playback capability; adjustable to zero sub frame accuracy; FM playback; optional FM and mono pilot playback (MK 1 Kit) available; fully functional at 7 ½, 15, 30 in./sec.

Price: ranges from $1,795.00 to $3,200.00

PLX magnetic heads are available for tape machines such as Teac ½-in 80-8, 1-in 85-16, Otari ½-in (4 and 8 track) MX5050, ½-inMTR10, Ampex 350 440 ¼-in (mono and 2-track), Ampex MM110/120024-track, MCI/Sony ½-in 2-track and 2-in 24-track, Studer A80 and A80024-track.

Prices: Available on request

Flux magnetic record and playblck heads are available for Ampex ATR100, 102 and 104 ¼-in and ½-in model machines The new 2tk ME seriesoffers extended headroom and 30 in/sec performance capabilities. Full ¼ to ½-in conversions available.

Time-code retro-fit kits are available for many ¼-in machine Features include center-track time-code record and playback cappability, adustable to zero sub-frame accuracy, FM and mono pilot playback available, fully functional at all tape speeds.

Prices: range from $1,795.00 to $3,2,00.00

NxT Generation, Inc, a division of JRF/Magnetic Sciences offers complete R-Dat repairs, overhauls and spare parts for most models manufacured by Panasonic, Sony, Otari, Tascam, and Technics. All work is fully guaranteed to meet factory specs

Prices: $75.00 to $500.00

MITSUBISHI DIGITAL PRO AUDIO

The **X-86HS** has four heads, is DC servo controlled with a pinch roller drive with constant tension servo system; line input impedance of 10k ohms; line output impedance suitable for 200-600 ohm load; frequency response of 20 Hz to 40 kHz; dynamic range over 90 dB; and wow and flutter. Additional options available.

Price: available upon request

The **X-86** and **X-86C** are two-channel ready; have four heads; are DC servo controlled; has pinch roller drive with constant tension servo system; line input impedance 10k ohms; line output impedance suitable for 200 ohm load; frequency response of 20 Hz to 20 kHz; dynamic range over 90 dB; and wow and flutter. Additional options available.

Price: available upon request

The X-880 has 32 channels; is DC servo controlled; has a closed loop servo, pinch rollerless capstan drive system; line input impedance of 10k ohms; line output impedance suitable for 200 ohm load; frequency response of 20 Hz to 20 kHz; dynamic range of over 90 dB; and wow and flutter. Additional options available.

Price: available upon request

OTARI

MX-50 II is a ¼ 2-channel analog Recorder/Reproducer: Quartz PLL direct drive capstan and 2 Induction reel motors; Choice of 15/7.5 ips or 7.5/3.75 speed combinations; 10.5-in. Reels; % W/F @ 15 ips; 30 Hz-20 kHz 2dB @ 15 ips; Input: Transformerless active Balanced 10 kOhm, Output: Transformerless active single ended 5 Ohm; s/n 70 dB (IEC) unwtd ref 1040 nWb/m; Monitor speaker; VEM (voice editing module is among many options available. Professional User

Price $2725.00

MX-5050 B-III is a ¼ 2-channel half-track or quarter-track three-head designed analog Recorder/Reproducer; DC quartz PLL capstan and 2 induction reel motors; 10.5-in. Reels; % W/F @ 15 ips; 40 Hz-20 kHz 2 dB @ 15 ips; s/n unwtd (IEC) 72 dB ref 1040 nWb/m @ 15 ips; 3 speeds 15/7.5 ips or 7.5/3.75; 20% vari-speed; Built-in mini-locator; Gapless, seamless, punch-in/out; Transformerless +4 dBu balanced inputs/outputs; Otari parallel interface for external machine control; Rugged and reliable machine with many available options. Professional User

Price from $3459.00

MX-55 is a ¼ 2-channel Upright /Console analog Recorder/Reproducer, available in 6 models: DC quartz PLL capstan and 2 induction reel motors; 10.5-in. Reels; W/F % @ 30 ips; 30 Hz-20 kHz 2 dB @ 15 ips; s/n unwtd (IEC) 72 dB ref 1040 nWb/m @ 15 ips; 3 speeds pairs available 3.75/7.5, 7.5/15, or 15/30 ips; Built-in mini-locator; Gapless, seamless, punch-in/out; balanced Transformerless inputs/outputs; Mic/Line mixing; 10.5-in. Reels; 2-ch with time-code track and VEM (voice ediging mode) are among the many options available for this machine.

Professional User Price $4509.00 - $7135.00 depending on model.

MX-5050 MK-IV-2/4/8 ¼-in. 2-channel; ½-in. 4-channel; ½-in. 8-channel console analog Recorders/Reproducers: DC quartz PLL Capstan and 2 induction reel motors; 10.5-in. Reels; % W/F @ 15 ips; 40 Hz-20 kHz, 2 dB @ 15 ips; s/n unwtd (IEC) 69 dB ref 1040 nWb/m; 15/7.5 ips or 7.5/3.75 speed combinations; Gapless, seamless, punch-in/out; Balanced transformerless inputs/outputs; 20% vari-speed; Mic/Line mixing; Audiolocators, Remote Controllers, and Synchronizers are among the many options available for these machines.

Professional User Price from $4509.00 - $6609.00.

MTR-15 is a high performance ¼-in. 2-channel; ¼-in. 2-channel with TC and ½-in. 2-channel Desk Top/Rack Mount or Console version analog Recorder/Reproducer with 12.5" reel capacity; % W/F @ 30 ips; 40 Hz-28 kHz 2 dB @ 30 ips; s/n unwtd IEC 73 dB ref 1040 nWb/m; Multiple Auto-alignment capability and storage of record and reproduce electronics; 30, 15, 7.5, 3.75 ips speeds; Cue wheel for Shuttle/Jog operation; Selectable bal/unbal inputs/outputs; Remote Controllers, Autolocators, Chase Synchronizers, Serial and Parallel Interface are among the many options available for this machine.

Professional User Price from $9,715.00 - $13,304.00.

MX-80 is a 2-in. 24-track analog Recorder/Reproducer with Remote Controller: DC quartz PLL capstan and 2 servo ⅓ HP reel motors; Microprocessor controlled constant tension transport, noiseless and gapless record punch-in, punch-out capability at any speed; 30/15 ips or 15/7.5 speed pairs; 50% vari-speed; Transformerless balanced inputs/outputs; <0.04% W/F @ 30 ips; 30 Hz-20 kHz 2dB @ 15 ips; s/n unwtd 68 dB @ 30 ips ref 1040 nWb/m; Tach or TC based Autolocators are optionally available.

Professional User Price $31,950.00.

MTR-90III 2-in. 24-channel Master Analog Recorder/Reproducer with Session Controller: Pinchrollerless transport for quick and accurate tape response as well as servo-locked forward and reverse play; DC quartz PLL capstan with two ½ HP DC reel motors; 7 to 14-in. Reels; <0.04% W/F @ 30 ips ref 1040 nWb/m; Transformerless balanced input/outputs; 20% vari-speed; Serial I/Q interface (RS-232); Gapless and seamless punch-ins/outs; Many options.

Professional User Price $42,950.00.

MTR-100A is a 2-in. 24-channel High Performance Analog Recorder/Reproducer with Session controller: Microprocessor controlled, pichrollerless transport utilizing quartz PLL DC capstan and PWM DC servo reel motors; Automatic Alignment of Rec/Rep electronics; <0.04% W/F @ 30 ips; 30 Hz-22 kHz 2dB @ 30 ips; s/n unwtd (AES) 70 dB @ 30 ips ref 1040 nWb/m; Wind speed 2 to 474 ips; Autolocators with or without TC and Chase synchronizers are among the options available for this machine.

Professional User Price $67,950.00.

DTR-900II 1-in. 32-channel PD format Digital Master Tape Recorder/Reproducer with Remote Control/Locator featuring a 32-channel meter display: Featuring 2 separate analog and Aux digital tracks plus a SMPTE/EBU TC data track; Transformerless balanced Inputs/Outputs; 20 Hz-18 kHz 0.5 dB; s/n (Din) 90 dB; 3 Sampling Freq; Adjustable Crossfade; Digital Ping-Pong; 12.5% Pitch control; Otari's proprietary pichrollerless Transport provides positive tape control with gentle tape handling. Chase synchronizers and Remotes are among many options available for this machine.

Professional User Price $150,000.00.

R-DAT Recorders

DTR-7 2-channel DAT Recorder/Reproducer: Closed Loop Tension Servo tape transport; Transformerless balanced +4 dBu/-16dBu analog inputs and outputs in addition to both AES/EBU and S/PDIF digital audio interfaces; 1-bit Delta Sigma A/D converters and 1-bit Pulse D/A converters assure accuracy; Frequency Response 20Hz-20kHz 0.5 db; s/n X DB (DIN) Sampling monitor enables input monitoring through the converters; Precise dB indication of margin-to-peak levels is displayed for optimum recording level settings; Other features: EIAJ/DAT format assures compatibility; Selectable 48/44.1/32 kHz sampling frequencies; 32 kHz long play mode; Start ID, Auto ID Edit and Auto Renumber functions; 20 segment bar graph meters with peak hold; Wireless remote control and 3U-size rack mount adapters included.

Professional User Price: $1695.00.

DTR-90T 2-channel 4 head Professional Digital Audio Tape Recorder/Reproducer with separate timecode track, timecode reader/generator and chase syncronizer standard. Conforms to EIAJ/DAT format. Specs, 20-20k ±0.5 dB, dynamic range <90 dB. Features include standard AES/EBU digital in/out RS422 serial port, jog/shuttle wheel,±12.5% vari-speed.

Professional User Price from $8,995.00.

SAKI MAGNETICS, INC.

Professional Studios Series

SAKI Magnetics for 25 years has been the premier tape head manufacturer in the United States with over 90,000 heads installed in major record lables,studios, video post houses, TV and Radio stations world wide.

Studer (24 trk $3445.00) (2 trk 1/4-in. $445.00) (2 trk ½-in. $595.00)

Studer (A8O QC $695.00) (710/720/721 QC $535.00)

OTARI (24 trk $2500.00) AMPEX ATR (2 trk ¼ $445.00)

AMPEX (2 trk ½ $625.00)

SONY PRO AUDIO

APR-24 is a 24-channel analog recorder utilizing 2 in. tape. It features amorphous steel heads and "DC constant tension design." Frequency response at 30 in./sec. is 48 Hz—25 kHz (+0.75/-3.0 dB). S/N is 70 dB at 30 in./sec. and 66 dB at 15 in./sec. includes remote control with stand.

Price: $38,080.00

APR-5000 is a 2-channel analog recorder which can be purchased in various configurations (including an IEC center-track time-code version). The 5002W version features a 50 Hz-28 kHz (+0.75/-3.0 dB) frequency response and a S/N ratio of 65 dB. Other versions feature 9-pin serial interface.

Prices: from $8,875.00 to $12,680.00

TASCAM, TEAC CORPORATION OF AMERICA See ou ad on Cover III

Cassette

The **102** stereo cassette recorder has two heads, two motors with Dolby HX Pro; wow & flutter: 0.45 percent; frequency response of 25 Hz to 18 kHz; Unwtd. S/N ratio of less than 80 dB with Dolby C.

Price: $299.00

The **103** stereo cassette recorder has three heads, two motors with Dolby HX Pro; wow & flutter: 0.045 percent; frequency response of 20 Hz to 20 kHz; Unwtd. S/N ratio of less than 80 dB with Dolby C.

Price: $499.00

The **202WR** dual stereo cassette recorder each has two heads, two motors with Dolby HX Pro; wow & flutter: 0.06 percent; frequency response of 30 Hz to 18 kHz; Unwtd. S/N ratio of less than 79 dB with Dolby C.

Price: $499.00

The **112/112B** stereo cassette recorder has three heads, two motors with Dolby HX Pro. B versio offers XLR in/out wow & flutter: 0.04 percent; frequency response of 25 Hz to 18 kHz; THD @ 400Hz of 1 percent; Unwtd. S/N ratio of less than 78 dB with Dolby C.

Price: $679.00

The **112R** auto reverse stereo cassette recorder has four heads, two motors; wow & flutter: 0.03 percent; frequency response of 25 Hz to 19 kHz; THD @ 400 Hz of 1 percent; Unwtd. S/N ratio of less than 80m dB with Dolby C.

Price: $329.00

The **122 MKII** stereo cassette recorder with ¼ in. mic inputs has three heads; three motors; wow & flutter: 0.04 percent; frequency response of 25 Hz to 19 kHz; THD @ 400 Hz of 1 percent; Unwtd. S/N ratio of less than 78 dB with Dolby C.

Price: $1,099.00

The **238** 8-track cassette recorder has a speed of 3 3/4 in./sec.. It has wow & flutter, 0.04 percent; frequency response of 30 Hz to 16 kHz; THD @ 400 Hz of 0.8 percent IHF. S/N ratio of less than 93 dB with dbx.

Price: $1,799.00

DAT

The **DA-30** DAT Recorder with AES/EBU and consumer digital I/O performs to the following specifications: wow & flutter: 0.001 percent; frequency response: 1 Hz to 22 kHz; Unwtd. S/N ratio of less than 94 dB.

Price: $1,499.00

YAMAHA CORPORATION OF AMERICA

The **DTR2** includes four sets of Input/Output connections; digital I/O (coaxial and optical), RCA-type phono unbalanced analog connections and balanced XLR (+4 dB) connectors. A front-panel switch selects analog or digital inputs. Another front panel switch selects either 44.1 kHz or 48 kHz sampling frequency.

Price: $1,495.00

The **YPDR601** CD Recorder allows a full TOC to be written to disc either before or after recording; allows recording of audio data to disc to be interrrupted using standard pauses functions; direct input/output connections can be either analog or digital. Analog connections are made via balanced XLRs, while digital connections can be made via AES/EBU, or SDIF-2 data formats.

Price: not available

The **ATR-80/24** and **32** 2 in. track recorders have three heads; three motors; wow & flutter: 0.05 percent @ 30 in./sec.; frequency response of 45 Hz to 25 kHz; THD @ 1 kHz of 0.5 percent; Unwtd. S/N ratio of less than 67 dB.

Price: $34,999.00

The **ATR-60/16** 1 in. 16 track recorder has three heads; three motors; wow & flutter: 0.08 percent @ 30 in./sec.; frequency response of 40 Hz to 22 kHz; THD @ 1 kHz of 0.8 percent. A wtd. S/N ratio of less than 71 dB.

Price: $15,999.00

The **MSR-24** 1 in. 24 track recorder has two heads; three motors; wow & flutter: 0.06 percent @ 15 in./sec.; frequency response of 40 Hz to 20 kHz; THD @ 1 kHz of 0.8 percent; A wtd. S/N ratio of less than 65 dB.

Price: $12,499.00

The **MSR-16** ½ in. 16 track recorder has two heads; three motors; wow & flutter: 0.06 percent @ 15 in./sec.; frequency response of 40 Hz to 20 kHz; THD @ 1 kHz of 0.8 percent; A wtd. S/N ratio of less than 65 dB.

Price: $7,499.00

The **TSR-8** ½ in. 8 track recorder has two heads; three motors; wow & flutter: 0.08 percent @ 15 in./sec.; frequency response of 40 Hz to 20 kHz; THD @ 1 kHz of 0.8 percent; A wtd. S/N ratio of less than 68dB.

Price: $3,499.00

The **BR-20** and **BR-20T** ¼ in. two track recorder has three heads; three motors; wow & flutter: 0.06 percent @ 15 in./sec.; frequency response of 35 Hz to 22 kHz; THD @ 1 kHz of 0.8 percent. A wtd. S/N ratio of less than 72 dB.

Prices: $2,999.00 and 3,999.00 respectively.

The **DA-800** ½ in. 24 track DASH recorder has three heads; three motors; wow & flutter: unmeasurable; frequency response of 20 Hz to 20 kHz; THD @ 1 kHz of 0.05 percent.

Price: $99,000.00

The **ATR-60/2T** ¼ in. 2 track recorder with center track TC has three heads; three motors; wow & flutter: 0.05 percent @ 15 in./sec.; frequency response of 40 Hz to 22 kHz; THD @ 1 kHz of 0.6 percent; A wtd. S/N ratio of less than 72 dB.

Price: $6,999.00

POLYLINE CORPORATION

PolyQuick, a division of Polyline Corp., is a distributor of both audio and video production and packaging supplies. Following is a partial list of professional supplies in stock for immediate shipment.

Open Reel Audio:

blank audio tape, empty reels/boxes, editing supplies.

Audio Cassettes:

blank-loaded audio cassettes, labels, cassette boxes, albums.

R-DAT:

Many brand names and lengths to choose from.

CD Packaging:

Jewel cases, albums, mailers.

STUDER REVOX AMERICA, INC.

R-DAT

D780 R-DAT is a 2-channel, 2-head recorder with 4 motors. Wind speed is 400 times playback speed. Hall-Commutated capstan motor, Frequency response is 20Hz-20kHz, rack mountable, dimensions: 19in. X 5.25-in. X 15/4-in., weight: 23 lbs.

Price: $7400.00.

CD Recorder

D740 CD Recorder-Converter technology: Bitstream in differential mode, Sampling Frequency: 44.1 kHz; Digital inputs/outputs: Optical, Cinch and XLR; Analog inputs: Cinch and XLR; Frequency Response: (record & playback: % (20 Hz-20kHz ±0.2 dB, Total Harmonic Distortion + Noise: (record & playback): <0.008% (20-20kHz) <0.005% (1 kHz), (playback only): <0.006% (20 Hz-20 kHz), Phase Linearity: (record & playback): <F128M3 degrees(20 Hz-20kHz), (playback only): 1 degree (20 Hz-20 kHz), channel Separation: (record & playback):80 dB(20 Hz-20kHz), >90 dB (1 kHz), Channel Balance: (record & Playback): <0.2 dB (output line), Input dBm, (UNCAL): max. increase of the input sensitivity -10 dB, Output Line: +15 dBm 0.1 dB (at 0 dB, RI = 10 k/Ohm), internal adjustment range 0, +24 dBm, Parallel Remote: 25-pin remote control socket with fader start, dimensions: 19-in. rack mount verison or table top version (16 ½-in. X 5 ¼-in. X 13 ¾-in.).

Price: $11,500.00.

Cassette Recorder

A721 Cassette Recorder- 2-channel, 4 motors, 3 heads, speed: 1 ⅞-in. per second, DC direct drive spooling motors, distortion less than 1.0%, frequency response: 20Hz-20kHz 3 dB, line input: balnaced and floating, minimum 10 kOhm, LCD, THD% less than 1.0% at 0 VU, >72 dB 19-in. rack mount, 23 lbs.

Price: $3300.00.

Open Reel Recorders

PR99 MKIII- ¼-in. 2-channel, 3 motors, 2 channels, AC servo capstan, 3 heads, 10.5-in. max. reel size, W & F less than 0.1% DIN 45507, distortion 1%, S/N (A weighted) 66 dB, response 30 Hz-22 kHz, 3 dB, dimensions 19-in. x 15.75-in. x 8-in.

Price: $3495.00.

A807- ¼-in. 2-channel or ½-in. 4-channel, 3 motor, servo controlled DC capstan, 3 heads, 10.5-in. max reel size, W & F less than 0.05% (DIN 45507), distortion less than 1% (ref. +510 nWb/m), S/N 66 dB (NAB unweighted ref. =510 nWb/m), response 30 Hz-20 kHz 2 dB, optional: balanced mic., SMPTE center track time code, dimensions 24-in. x 22.5-in. x 44.5-in..

Price: ¼-in. 2 track in console $7700.00.

A812- ¼-in. 2-channel, 3 motor, servo DC capstan, 3 heads, 12.5-in. max. reel size, W & F less than 0.04% (DIN 45507), distortion less than 1% (ref. = 510 nWb/m, S/N 70 dB (NAB unweighted ref. = 510 nWb/m), response 3-Hz-20kHz 2 dB, optional SMTPE center track time code, dimensions 25-in. x 26-in. 44.5-in.

Price: 1/4-in. 2-track $13,200.00.

A820- 2-track recorder ¼-in. & ½-in., 3 motor, servo DC capstan, 3 heads, 14-in. max. reel size, W & F less than 0.03% (DIN 45507), distortion less than 1% (ref. 510 nWb/m), S/N 70 dB (ref. = 510 nWb/m), response 30 Hz-20kHz 2 dB, optional SMPTE center track time code. 1/4-in. 2-track.

Price: $17,900.00.

A827-MCH- 2-in. 24-channel, 3 motor, servo DC capstan, 3 heads, 14-in. max. reel size, W & F less than 0.03% (DIN 45507), distortion less than 1% (ref. 510 nWb/m), S/N 70 dB (ref. = 510 nWb/m), response 30 Hz-20kHz 2 dB, dimensions—30.5-in. x 29.5-in. x 57-in.

Price: $44900.00,

A820-MCH- 2-in. 24-channel, 3 motor, servo DC capstan, 3heads, 14-in. max reel size, W & F less than 0.03% (DIN 45507), distortion less than 1% (ref. 510 nWb/m), S/N 70 dB (ref. = 510 nWb/m), response 30 Hz-20kHz 2 dB, automatic alignment, internal noise reduction, dimensions- 30.5-in. x 29.5-in. x 57-in.

Price: $67,000.00.

UHER OF AMERICA

Open-Reel Recorders

4000 Report Monitor AV- Portable Open-Reel, 2 Track Monaural, 4 Speeds: $^{15}/_{16}$. 1-$^{7}/_{8}$. 3-$^{3}/_{4}$. 7-$^{1}/_{2}$ IPS, 3 heads, 5-in. Reel, 1 Channel, Belt Drive, 1 VU Meter, Frequency Response 20-25, 000 Hz., Wow & Flutter less than 0.2%, Signal/Noise 64 dB, Mic Inputs- 200 Ohms, LED funciton indicators, Switchable ALC, dimensions: 11 x 3-$^{1}/_{2}$ x 9 inches, Weight 8 lbs.

Sugg. List Price: $2,025.00

4200 Report Monitor- Portable Open-Reel, 2 Track Stereo, 4 Speeds: $^{15}/_{16}$. 1-$^{7}/_{8}$, 3-$^{3}/_{4}$, 7-$^{1}/_{2}$ IPS, 3 Heads, 5-in. Reel, 1 Channle, Belt Drive, 2 VU Meters, Frequency Response 20-25,000 Hz., Wow & Flutter less than 0.2%, Signal/Noise 64 dB, Mic Inputs- 200 Ohms, LED functions indicators, Switchable ALC, Dimensions: 11 x 3-$^{1}/_{2}$ x 9 inches, Weight 9 lbs.

Sugg. List Price: $2,169.00.

4400 Report Monitor- Portable Open-Reel, 4 Track Stereo, 4 Speeds: $^{15}/_{16}$, 1-$^{7}/_{8}$, 3-$^{3}/_{4}$, 7-$^{1}/_{2}$ IPS, 3 Heads, 5-in. Reel, 2 Channels, Belt Drive, 2 VU Meters, Frequency Response 20-25,000 Hz. Wow & Flutter less than 0.2%, Signal/Noise 62 dB, Mic Inputs- Dimensions: 11 x 3-$^{1}/_{2}$ x 9 inches, Weight 8 lbs.Sugg.

List Price: $2,169.00.

6000 Report Universal- Portable Open-Reel, 2 Track Monaural, 4 Speeds: 3-$^{3}/_{4}$, 1-/8, $^{15}/_{16}$, $^{15}/_{32}$ IPS, 3 Heads, 5-in Reel, 1 Channel, Solenoid Controlled, Belt Drive, 1 VU Meter, Built-In Voice Activation System, Memory Pulse Facility, Wow & Flutter less than 0.2%, Signal/Noise 62 dB, Dimensions: 11 x 3-$^{1}/_{2}$ x 9-in., Weight 8 lbs.

Sugg. List- $2,640.00

Cassette Recorders

CR 1600- Portable Stereo Cassette, 4 Track Stereo, Electronic Drive Control for Auto-reverse Operation in Record or Playback Mode, 2 Speeds: $^{15}/_{16}$,1-$^{7}/_{8}$ IPS, 3 Heads, 2 Channel, 2 VU Meter, Dolby B, Switchable ALC, Solenoid Controlled, Fully Remote Controlled, Built-in Voice Activation System, Memory Pulse Facility, Record Time- 6 hrs., Frequency Response- 20-19,000 Hz., Wow & Flutter less than 0.2%, Signal/Noise 64 dB, Dimensions: 9 x 2 x 7-in., Weight 7 lbs.

Sugg. List- $2,179.00

CR 1601- Portable Cassette, 4 Track Monaural, 3 speeds: $^{15}/_{32}$, $^{15}/_{16}$, 1-$^{7}/_{8}$ IPS, 3 Heads, 2-channel, 1 VU Meter, Switchable ALC, Solenoid Controlled, Fully Remote Controlled, Built-in Voice Activation System, Memory Pulse Facility, Record Time- 8 hrs., Frequency Response- 20-19,000 Hz., Wow & Flutter less than 0.2%, Signal/Noise 50 dB, Dimensions: 9 x 2 x 7 inches, Weight 7 lbs.

Sugg. List- $2,179.00

XEDIT CORPORATION

Xedit manufactures 32 standard variation of Editall precision tape splicing blcoks encompassing all video from 8mm to one inch, DAT, and of course all Analog and Digital formats. This is a wolrd class line of professional products that is unsurpassed for both quality and scope. All of these blocks are individually precision machined from premium non-magnetic alloy and are then meticulously hand finished to assure that the delicate magnetic tape is held secureley without damage. “Blade Splicing remains one of the most technically and cost effective technologies available to the recording industry.”

EDITABS are pre-formed die cut splicing tabs that are available for all tape formats through to one inch. They are in effect the other half of the “Editall Splicing System”. They are easy top use and offer the advantage of avoiding finger contamination of the adhesive resulting in permanent splices that are as strong and reliable as the tape itself. Editabs are packaged in sip lock bags of 250 tabs on sheets, or bulk boxes of 1000 tabs; they can also be ordered in rolls of 5000 tabs.

Some items of special interest:

The **S-3**D (¼) and the **S3.5D** (½in.) are deluxe blocks that include a choice of three cutting angles and that are taken through an extra polishing step that provides the trough with a mirror like finish. The **S-3/OT** one of our exact Otari replacement blocks is also finished in this manner.

The **MD-25** is a “hybrid” block encompassing the trademark curved trough holding design with a precisely located, flat splice point, configured exactly like the original Mitsubishi digital block with the added benefit of holding the tape securely for splicing. The MD-25 is an exact replacement for Mitsubishi but may be used with other brands as well.

Also specially designed for Digital splicing are our patented “EC” series of blocks. They are capable of holding the thinnest of tape securely in a flat trough utilizing mechanically operated very small holding edge clamps while allowing full access to the tape for splicing.

The **P-2** is a high quality molded plastic ¼ bloack inteded for instructional use. The P-2 is frequently provided by schools along with the comprehensive text **TB-2**, on all aspects of tape editing by Editing pioneer Mr. Joel Tall. Special arrangements are offered to schools.

XEDIT also manufactures a high quality, compact drift and flutter meter for all types of service and quality control aplications. The **Model 20-P** is an aluminum instrument measuring 8.87-in. x 6.75-in. x 1.75-in. Wt. 3 lbs.

Price: $600.00

Audio Tape

BASF CORPORATION

PEM 469 studio mastering tape features high output and low noise for a wide dynamic range and standard bias for compatibility with many tapes and machines. Low print-through minimizes the effects of pre- and post echo and excellent slitting provides consistent phase stability and superior winding characteristics.

PEM 468 features extremely low print-through, high output and low noise for a wide dynamic range. Excellent slitting for consistent pahse stability, and batch number and web position printed on the back coating for permanent tape-type identification.

PEM 369 features 1 mil. thickness, 3,600 feet for extended record time. Extremely low print and excellent slitting for superior winding and phase stability.

PEM 526 and Chrome Loop Master 921 are designed for high-speed cassette duplication, and provide excellent durability under the stress of bin-loop applications. Also features high-frequency signal retention after numerous passes and extremely low print-through.

PE 649/949/1249 have premium iron oxide, high output, low noise, standard IEC Bias I cassette tape. Features extended headroom in low and high frequencies for the most critical and demanding music duplication.

PE 619I/919I has low noise, high output, extended high end response iron oxide cassette tape for bias compatibility with Standard I designation.

PE 668/968 is a ferro-cobalt, high coercivity, IEC II bias tape. The tape has high output and very low noise. Ideal for demanding music duplication.

TDK ELECTRONICS: PROFESSIONAL PRODUCTS DIVISION

The Acoustic Master (AM) is a normal-bias ferric oxide formulation with reduced noise levels and wider dynamic range. Has jam-proof cassette mechanism with five screw construction and transparent shell. Available in 30-, 46-, 60-, 90- and 120-minute lengths.

The Sound Master (SM) is a high-bias cassette for studio and demo recordings. Uses SA tape pancake for low noise and wide dynamic range. A dual-layer anti-resonance cassette mechanism ensures phase accuracy and low modulation noise. Available in 10-, 20-, 30- and 60- minute lengths.

Instant Starting Acoustic Master (AL) is a normal-bias tape in an instant starting cassette. New pure-grained ferric oxide formulation for reduced noise levels and wider dynamic range. Housed in a jam-proof cassette mechanism. Available in 30-, 60- and 90-minute lengths.

The Duplication Master (ZM) has a normal-bias tape formulation, pure grained ferric oxide particles achieve reduced noise levels with wide dynamic range. Housed in jam-proof cassette mechanism. Available in 30-, 46-, 60-, 90- and 120-minute lengths.

The Medical Master has a normal-bias formulation for slow speed medical recording applications. Pure grained ferric oxide particle plus high-dispersion binder and advanced coating technology yield extremely low drop-out rate. Available in 60-, 90- and 120-minute lengths.

The Digital Audio Tape (DA-R) is a super Finavinx pure metal particle digital audio tape housed in a high-precision sealed cartridge. Available in 60-, 90- and 120-minute lengths.

3M

The 275 Digital Audio Mastering Tape for high density recording, offers low drop out rates and improved widablility. Compatible in a II DASH, DMS and PD formats.

The 806 Audio Amstering Tape has a 1.5 mil thick backcoated plyester backing. Developed to give good compromise in print-through and maximum output level characteristics. available in ¼ in., ½ in., 1 in., and 2 in. widths.

The 808 Audio Mastering Tape has a back coated polyester base 1.5 mil thick. Has extermely low print-through characteristics. Ideal for speech, sound effects, and other applications where low print through is required. Available in ¼ in. width only.

AUD Digital Audio Cassettes are engineered to deliver stat-of-the-art performance in the production of CDs, record albums and cassettes. Available in 30, 60, and 75 minute lengths, these units incorporate anit-static system.

AVX Audio Cassettes are normal bias (IEC Type I), optimized for heavy duty usage and high speed duplication. Available individually boxed or bulk packaged in 20, 30, 46, 60, 90 and 120 minute lengths.

IRC Audio Casettes are "instant record", normal bias (IEC Type I). A magnetically-coated leader is employed so that recording can be start at hte very beginning of tape. Available in both boxed and bulk packages, in lengths of 30, 60, 90 minutes.

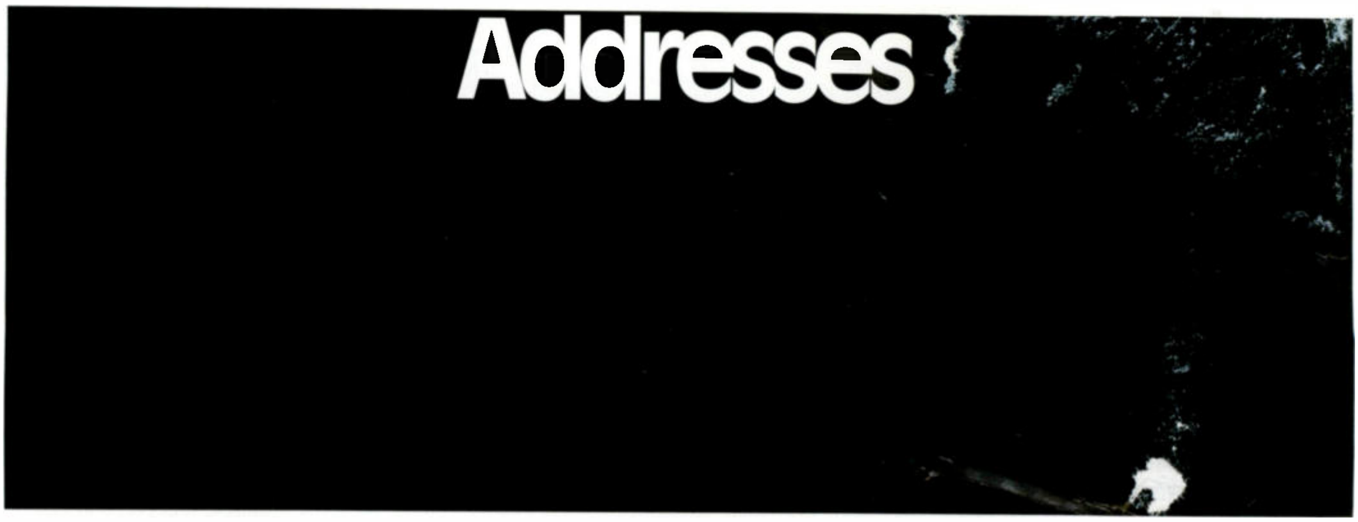
SX Audio Cassettes are chrome bias (IEC Type II) for high quality recording applications. Available in album boxes , 30, 46, and 90 minute lengths.

3M 996 audio mastering tape is the first analog mastering tape capable of recording at operating level +9 with virtually no distortion. Provides widest dynamic range of any analog mastering tape and offers improved print with almost no increase in tape noise. Available in ¼ in., ½ in., 1 in., and 2 in.

3M 226 audio mastering tape is designed for music mastering on a budget. Delivers high output/low noise performance on 1.5 mil polyester backing. Available in ¼ in., ½ in., 1 in., and 2 in.

3M DAT is designed for the demanding professional user who needs low error rates and a dependable product. Labels allow for plenty of space for documenting the project. Available in 46, 60, 90 and 120 minute lengths.

3M 186 audio meets the needs for program distribution,s torage and everyday use. Designed as a low noise tape for cost-effective use in standard bias settings. Available in ¼ in. width.



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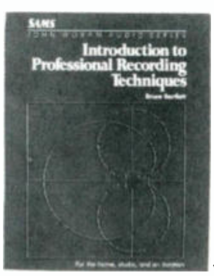

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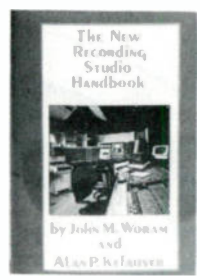

This new edition has been accepted as the long-awaited replacement to the original book published in 1976. The new edition is not "old wine in a new bottle." The revision has been done by Professor Kefauver. He is the Coordinator of the Recording Arts and Sciences Department and Director of Recording at the prestigious Peabody Conservatory of Music. The book is used by most of the recording schools and universities here and abroad. This book contains all the basics for the recording studio engineer, as well as more advanced information covering MIDI, Automated Consoles, SMPTE Time Code and Digital Audio. This book has been and remains the "bible" of the audio industry.

525pp. Hardcover

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LIVE SOUND! *by David Scheirman*

This excellent video is targeted at first-time users and musicians new to the field of sound reinforcement. However, the video contains insider tips and sophisticated approaches to using the equipment.

The video covers:

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- Loudspeaker placement and setup
- Mic selection and placement
- Monitor systems
- Mixer Position
- Processing/effects
- Crossovers/Equalization
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This is a *must have* video!

75 minutes

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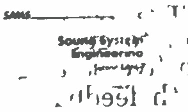
SOUND SYSTEM ENGINEERING, SECOND EDITION *by Don and Carolyn Davis*



Like the first edition, this comprehensive text provides readers with useful information for the day-to-day work of designing sound systems. This updated version contains in-depth coverage that carefully examines acoustic gain, clarity of sound, and required electrical power.

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CONCERT SOUND AND LIGHTING SYSTEMS *by John Vasey*

This book shows how to set up, maintain, and operate sound and lighting equipment for the performance of amplified music or any kind of touring production. An excellent reference and/or guide to procedure, the book provides descriptions of all the components that make up a system, explanations of how they all work together, and photographs and illustrations that show specific equipment and proper stage setup.

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This brand-new second edition has been updated to include the latest in MIDI, cinema sound, tranformers and compact discs. Readers learn the new developments in audio electronics, circuits, and equipment. There is also an in-depth examination of disc, magnetic, and digital recording and playback.

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BROADCAST SOUND TECHNOLOGY *by Michael Talbot-Smith*

This is an introduction to the technical aspects of sound in radio and television. It examines in detail the main items in the broadcast chain: studio acoustics, microphones, loudspeakers, mixing consoles, recording and replay (analog and digital), and the principles of stereo. It offers a easy technical treatment of audio principles and broadcast hardare.

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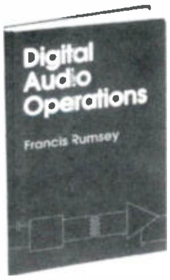
DIGITAL AUDIO OPERATIONS *by Francis Rumsey*



Leaving the higher levels of theory to other digital audio texts, this handbook emphasizes principles for the studio and those aspects of digital audio appropriate for day-to-day sound engineering operations. It describes the sampling process, error correction, editing systems and different recording options. This book is written to help producers and engineers in the studio get the best possible results from the high quality standard equipment in use today.

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STEREO MICROPHONE TECHNIQUES *by Bruce Bartlett*

This book is extremely timely for sound engineers and video or audio producers. Also, as Digital Audio Tape (DAT) production becomes less costly to use in the field, all electronic media will be trying to achieve the highest level sound production possible. This book tells how to position the correct microphones in the proper locations in order to record optimal quality stereo sound. The many illustrations and clear organization easily explain the theory behind stereo mic'ing methods, and describe specific techniques, including comparative evaluations. In addition, it offers suggestions on session procedures and stereo troubleshooting as well as recent developments in binaural and transaural stereo and stereo boundary arrays.

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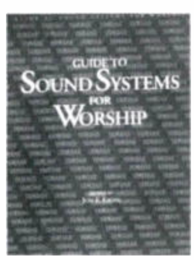
GUIDE TO SOUND SYSTEMS FOR WORSHIP *by Jon F. Eiche*



This book is written to assist in the design, purchase, and operation of a sound system. It provides the basic information on sound systems that is most needed by ministers, members of Boards of Trustees and worship and music committees, interested members of congregations, and even employees of musical instrument dealers that sell sound systems. To be of greatest value to all, it is written to be both nondenominational and "non-brand-name."

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LIVE SOUND MIXING

by Duncan R. Fry

Live Sound Mixing is the first book that shows you how to pull a great sound from a PA system. Author Duncan Fry is a survivor of many years on the live sound trail. People used to say "Ah, you can't learn how to mix from a book!" With this book, you can! This is an easy-to-read and understand "hands on" book that every live-sound engineer should have. From the basic principles of how a system works, through troubleshooting when it doesn't, the book is packed with useful information, many clear diagrams, helpful hints and detailed explanations. There are examples of every type of equipment you're likely to come across.

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Using a "hands on" approach, this video gives you an overview of the basics in setting up a personal studio. Among items covered are: hooking up instruments, connecting equipment, setting levels, working with the console, doing a recording, overdubs and mixdown. This is a very important video for all persons planning on setting up a studio.

Approximately one hour long

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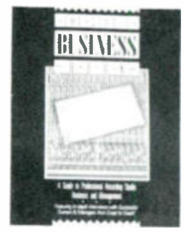
THE STUDIO BUSINESS HANDBOOK: A GUIDE TO PROFESSIONAL RECORDING STUDIO BUSINESS AND MANAGEMENT

by Jim Mandell



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New Ears: The Audio Career and Education Handbook

by Mark Drews

The newly revised second edition of New Ears: The Audio Career and Education Handbook is designed to assist those interested in studying sound engineering, music recording and music technology. It is, in fact, a comprehensive reference to audio career and education options.

Author/editor Mark Drews is senior audio engineer at the Syracuse University School of Music and College of Visual and Performing Arts. An active recording engineer, musician, and video artist, Mark Drews was the recent recipient of a Fulbright Senior Research Fellowship to assist with the development of a graduate music recording program in Norway.

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SHELLEY HERMAN

What Goes Around...A Report on DTS

● Sixty seven years ago, in 1926, a revolutionary new technology burst on the American public. Talkies! The public was already in love with the movies. In less than thirty years they made stars and millionaires out of a handful of people like Chaplin, Pickford, Griffith, and the rest of the film pioneers. Now the movies talked. In the first sound movie, *The Jazz Singer*, Al Jolson said: "You ain't heard nothin' yet". He was famous for that phrase, but he didn't know how prophetic he was.

With almost every passing year since then, the sound has improved. From transcription to optical to magnetic to digital, and hundreds of smaller increments along the way, each step has increased the quality of some facet of the film sound.

The Jazz Singer was released using a method called Vitaphone, developed by Western Electric. The Vitaphone system consisted of a film and a recorded disc. The good folks at Vitaphone calculated the length of time a roll of film lasted and figured out that a 16-inch disc turning at 33⅓ rpm would run for about the same time. Another standard was established, who needed ASA or ASTME?

The playback equipment consisted of a projector with a turntable and tonearm, both driven from the same motor. If the projectionist got the needle correctly in the groove, the sound would be synchronized for that reel, and then he had about twenty minutes to get the next reel and disc ready. Because there were a separate recording media for the sound and picture, this was called a "double system"

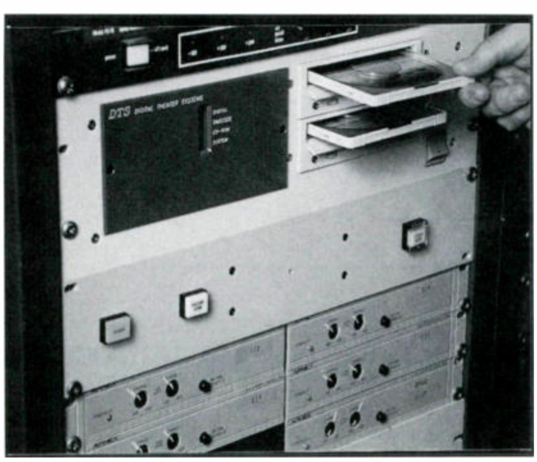


Figure 1. The DTS 6-track playback unit.

The system was cumbersome to use, so the same people, and others, figured out how the sound information could be all printed on the film making the projectionist's life a whole lot easier. The engineering staff, being the highly intelligent people necessary to develop this method, reached deep down and called it "single system"

The new system recorded sound on the film using one of two different methods to modulate light and expose the portion of the film reserved for the sound track. One method used a galvanometer, which is most easily described as a very sensitive meter, with a small mirror in place of the pointer. When electrical sound was applied to the coil of the galvanometer, the mirror would move slightly, displacing the beam of light that was pointed at the mirror and reflected onto the film. The film would then be exposed in relation to the sound. The other method used a device known as a *light valve*. It consisted of two flat ribbons of metal, touching edge to edge with a light focused through the gap between the ribbons onto the film. With the electrical signals applied to a coil that forced the ribbons apart or together, the amount of light passing through the opening would be related to the sound. In recent years, light emitting diodes and laser systems have replaced the electro-mechanical devices.

These systems worked wonderfully for many years, so long as there was just a single sound track. Then the creative minds asked: "Why not have more than one sound track. We could put two or three speakers behind the screen, one on each side of the audi- d one in back". *Fantasia*, only teen years after Jolson spoke rst cinema words, had a seven-tr stem. Features like Fantasia w ack to that double system again, only they didn't use discs, they just used more reels of film and recorded nothing but nd tracks on them. Because of nders of sprocket holes and otors that made every- run in synchronization, the sou stayed in sync.

When magnetic recording came along in the late 1940s the predictions for advancements in sound were unbelievable. The predictors forgot about one thing. If it doesn't sell popcorn, the theatre owners don't care about it. Also the magnetic system was back to double system again.

Why not just print time code on the film, put all the sound information on an other media, and have a reader on the projector read the time code, which would be exactly synchronized with the picture, and let the digits do the rest

They wanted something printed on the film. Besides the projectionists wouldn't clean the magnetic heads, or replace them when the wore out. Unsuccessful attempts were made to affix magnetic media to the film. Magnetic tape in the format of 35 mm film was used for multi-track audio recording for both cinema and record release with outstanding results.

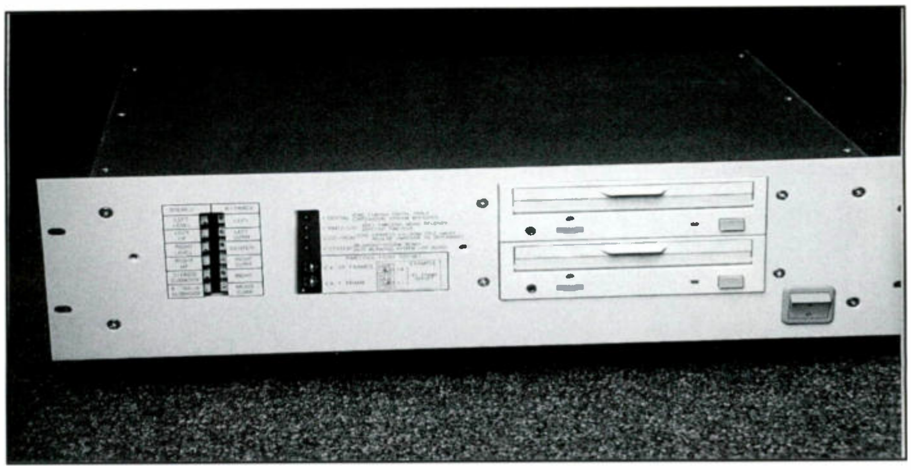


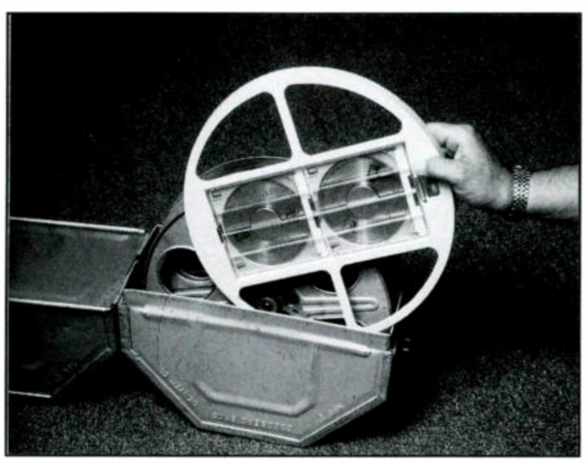
Figure 2. The same DTS 6-track playback unit with the cover plate off, showing the adjustments and controls.

As the demand for more optical sound channels increased, they were running out of places on the film to put them. Outside the sprocket holes, between the sprocket holes, between the frames, everything was tried. Then came the next revolution...digital.

I recall an incident in the 1970s when, emerging from the umpteenth demonstration of quadraphonic sound recorded on two channels, (which usually sounded like someone had a symphony orchestra on the end of a string, and was swinging it around our heads) someone said: "The Emperor still has no clothes"! Over the past few years I have heard many demonstrations of digital sound recorded, single system, on film. The results have been aurally spectacular, but the biggest problems have been with massive data compression and the inability to edit the film without messing up the synchronization. The device that reads sound from film can not be located at the same place as the lens, as the film does not travel continuously through the projector, but stops briefly each time the shutter opens, if it didn't, the image would be blurred. At the same time the sound portion of the film must travel smoothly so there is no flutter. The farther apart the image and sound are, the quieter the sound, and the more difficult editing becomes.

All these variables plus a plethora of film standards (CinemaScope, PanaVision, Todd AO, etc., etc.) add up to a real nightmare when an attempt is made to couple a sophisticated sound system to film. Not only that, but every time someone has an idea for a new sound system, new prints of each film have to be made, which is not an inexpensive process.

Figure 3. Two CD-ROM discs fit right into a standard film can for theater shipment.



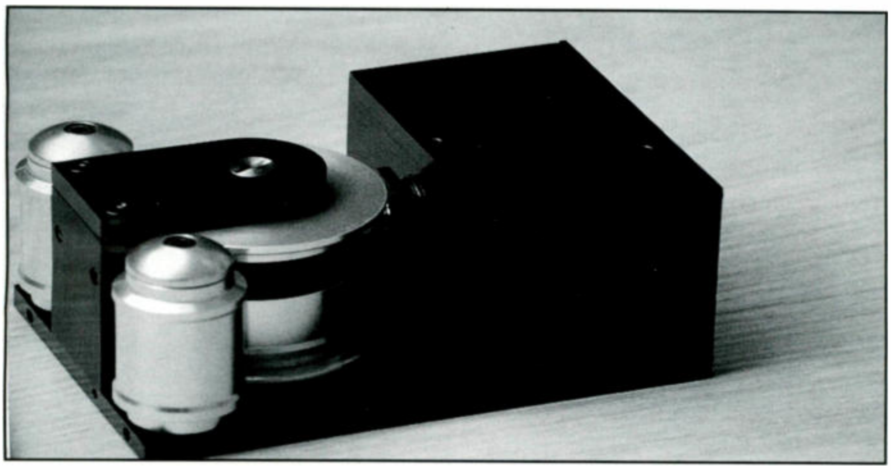


Figure 4. The time-code reader that fits onto a projector.

Meanwhile, videotape has been truckin' right along, and because of its electronic basis, (SMPTE)time code has been applied to the tapes, and those computernick wizards with their ones and zeros have made systems that lock everything together.

Along came the Digital Theater Systems people. Their idea was: Why not just print time code on the film, put all the sound information on an other media, and have a reader on the projector read the time code, which would be exactly synchronized with the picture, and let the digits do the rest. it's double system but it makes sense.

The time-code reader is placed anywhere in the film path on the projector, the distance between the reader and the lens is determined with a calibrated film, the offset number is entered into the electronics and the synchronization problems are ended. Because the sound is being reproduced from a CD ROM, the quality is exceptional, and with two discs, 200 minutes of six channel audio is available for all the wild effects one can imagine.

The Los Angeles Section of the Audio Engineering Society held its July meeting at the AMPAS (Film Academy) Goldwyn theater (see **db** July/August 1992) where Jurassic Park was screened. *Jurassic Park*, with all its roaring dinosaurs and screaming people, was released with the DTS system, and a short lecture about the system preceded the screening. Some samples of edited films were demonstrated, and with the digital control of the synchronization, the edits were undetectable. The Jurassic Park sound is as impressive as are the dinosaurs. I must admit I secretly cheered when the lawyer was eaten, but was surprised when the T/Rex didn't spit him out.

Figure 5. This logo apears on promotional materials for films to identify DTS films.

One of the greatest advantages of the DTS system has nothing to do with the electronics, but the fact that new, improved sound systems, that can sync to time code, can be used with the same prints. The sound can be updated, corrections can be made, multiple language versions can be released, all without ever touching the prints. Just send a different CD. In [illegible] where several lang[illegible] ken, several DTS [illegible] erate simultaneo[illegible] DTS people, being aware that electronics are fallible, have included a standard stereo sound track on each print, and should the digital signal fail, the system will continue to "free wheel" for four seconds, then it will revert to the optical track. This will be of comfort to those people who feel that if electricity was unavailable, we'd all be sitting around watching TV in the dark. There is a digital serial number on each reel of film and CD that will disable the system if the wrong CD is placed into the machine. The feature that is probably dearest to the hearts of the theatre owners is that the system cost is about one quarter of some of the older systems. As of this writing, about 1,000 theaters in the United States are already equipped with this system.

The playback system is based on a 386 computer system with one or two off the shelf Toshiba CD-ROM drives, and a couple of proprietary devices. The LTRT (stereo) has up to 3.5 hours of playback time from a single CD ROM. The Discrete unit has six output channels, plays back 100 minutes from each of two drives and the CD-ROMs are played sequentially. Each reel of film has a file on the CD ROM.

The recording system is relatively simple. Apogee analog-to-digital converters change the music to bits and bytes with 20 dB of headroom, an APT data compressor at 4:1 reduces the data, and the information along with the synchronizing information is recorded on a hard disc. Each recording system will record either the LTRT or six channel format. The data is then recorded on a recordable CD

The DTS System

There are two versions of the DTS Digital Sound Processors. The DTS-6 System brings discrete channel digital sound to premier 6-track theaters, while the DTS Stereo systems provides the same digital sound for stereo-only theaters. Both systems are fully automatic and installation takes less than an hour.

In operation, the projectionist simply loads a CD-ROM disc into the unit's disc drive. The system reads a time code on the film on its way to the projector and automatically plays the correct audio for every frame projected. Film edits and non-digital trailers are automatically accommodated. The system even checks that it has the correct sound track for the movie being shown.

Because DTS prints have a conventional stereo optical track, with the time code placed between the optical tracks and the picture, a singe inventory of prints can be made for use in all theaters ith an automatic fail-safe backup always available.

ROM, checked against the film, and if everything is OK, it's off to the pressing plant. At the present time there are two recording systems, one at Todd AO and one portable. Just as soon as the system is simplified enough for studio transfer technicians to use, anyone who will fork over the bucks can have one.

What started talkies, and this article, was picture on film with the sound on a synchronized disc.

What goes around, comes around. db

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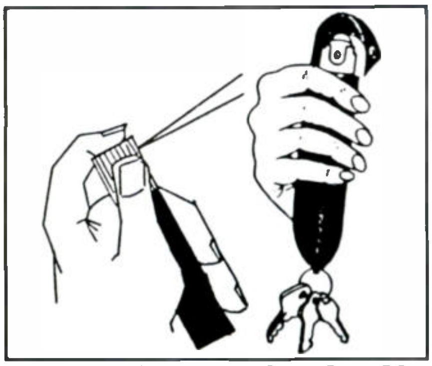

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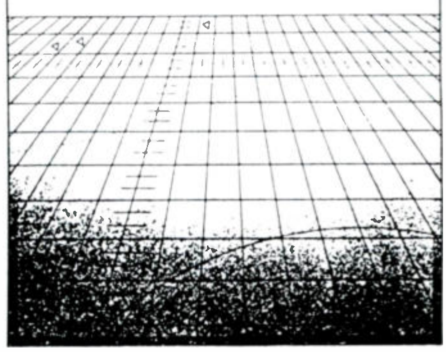

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PEOPLE, PLACES & HAPPENINGS

• **Community Professional Sound Systems** announced the appointment of **Todd Rockwell** as the new engineering and marketing liaison for the company. He serves as a link between the minds and design resources available at the company's Chester, PA operation and contractors and consultants across the country. Rockwell formerly was with **Electro-Voice** and **Mark IV Cinema**, where he held several positions. He has a BS. in Electrical engineering from Michigan Technical University in 1987.

• **Ampex Recording Systems Corporation** of Redwood City, California has formed a new division to be known as **Ampex Digital Media**. The division will concentrate on developing and supplying specialized high-performance magnetic media for use with the company's current digital video and mass data storage recorders. **Dave Davies**, formerly Vice President of Engineering at Ampex Recording Media Corp has been named Vice President of the Digital Media Division. Michael Wilke, formerly Marketing Manager at the Corporation has been appointed General Manager of the division.

• **Rose Mann** is back at the **Record Plant.** A mainstay at the legendary Hollywood Record Plant during the Seventies and Eighties, Rose is now Vice President. Studio Manager, according to the announcement by **Rick Stevens**. Record Plant will also be celebrating its silver anniversary (25 years) in September. The studio has just completed an ambitious $4 million upgrade that more than doubled the client areas of the building, adding two new state-of-the-art Studio Suites, and a new digital editing/MIDI overdub suite to the existing two studios.

• At **University Sound, Ken Koceski**, engineering manager, has been promoted to market development managers of the company's new **USI Audio** branded products; announced **Doug MacCallum**, president of University. Koceski, who has been at the company for five years, will be fully involved with USI Audio products.

• **Audio Pus Video International, Inc.** and **Tahoe Productions** have formed an alliance to provide post-production services for the New York video market. **Scott Irwin**, Director of Post Production for Tahoe has established a base of operation an APVI New York, situated in mid-town Manhattan. APVI specializes in standards conversion, duplication, film-to-tape, audio and international post-production with two other facilities in Northvale, New Jersey.

• **Allan Nichols** has been named director of sales and marketing for the Mark IV Pro Audio Group. The announcement was made by Ivan Schwartz, general manager. The Group, headquartered in Buchanan Michicagan distributes in the U.S. Klark-Teknik, DDA and Midas products, as well as Electro-Voice Concert Sound products.

• As part of its commitment to meeting the audio/video musical-instrument rental needs of facilities around the country, the **Toy Specialists** of New York City has opened a new office specifically to serve Florida and Southeast Markets. The announcement was made by **Bill Tesar**, Vice President of the company. The new office will be headed by **Mark Prater**, and it is located at 1211 N. 56th Street, Tampa, Florida. Their new phone number is **800 445-3330**.

• There will be four sub-committees and twenty-three working groups of the **AES Standard Committee** at the **AES Convention** in New York. The groups will be preparing documents on subjects ranging form computer control sound systems to low-noise grounding and wiring practices. High on the committee's agenda will be the SC-10 Sub-Committee's widely anticipated protocol standards controlling sound systems. Other documents scheduled to be completed at this meeting include listening tests for loudspeakers, methods for archival transfer of audio recordings, interfacing MIDI to sound system, conservation of polarity, and guidelines for authenticating audio recordings.

Details about these projects and the activities of the AESSC, including the schedule of meeting and list of projects, may be found in the **Journal of the Audio Engineering Society**. The meeting will begin on October 4th at the New York Hilton Hotel. This is three days before the opening of the Convention. **Contact the AES at 60 East 42nd Street, New York, NY 10165. 212 661-8528 or fax them at 212 682-0477.**

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