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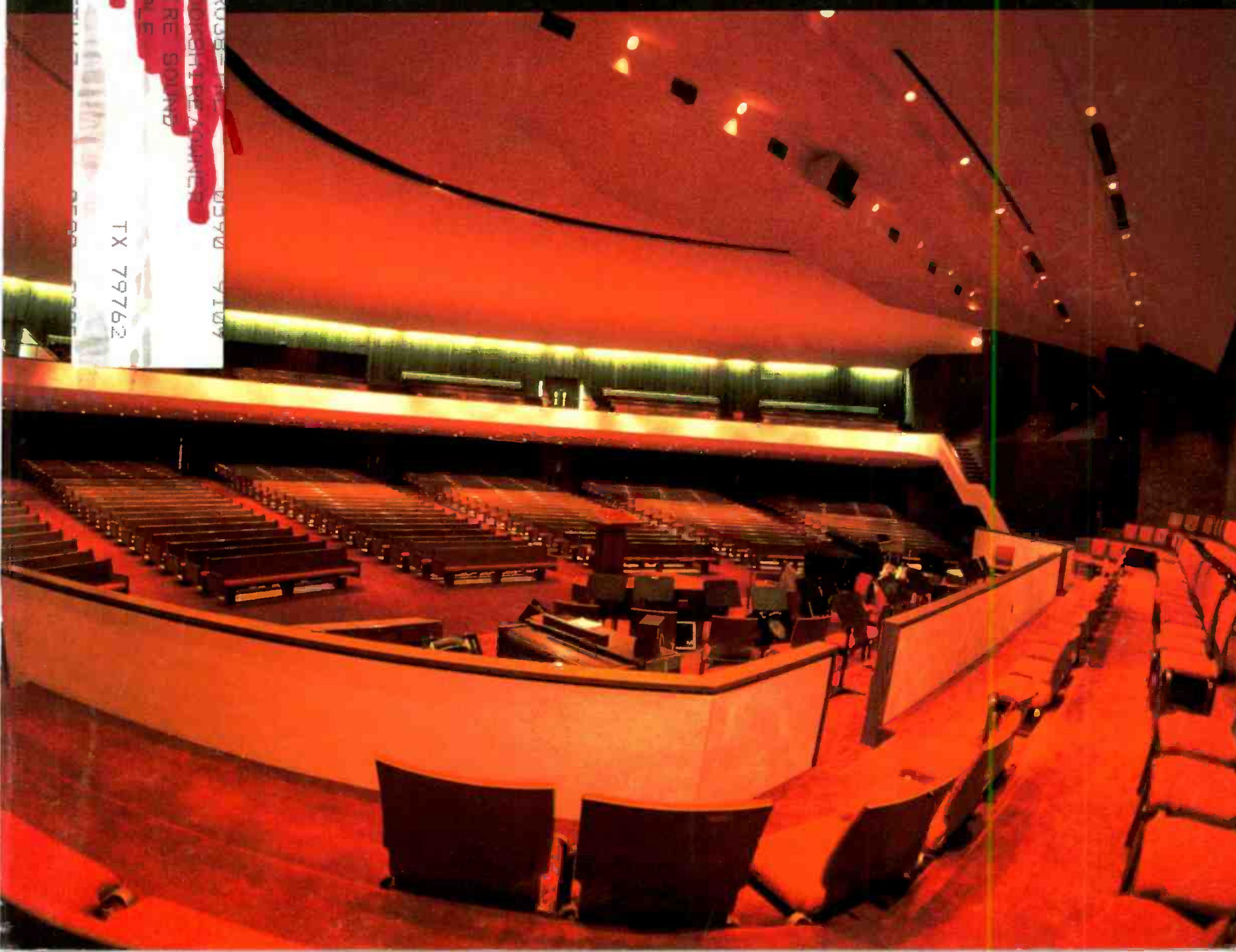

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May/June 1990 VOLUME 24 NO.3

db magazine

serving: recording, broadcast and sound contracting fields

the sound contracting engineer

the recording engineer

THE ELECTRONIC COTTAGE

the broadcast engineer

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About the Cover

• On our cover, the Sanctuary of the Salem Church of God in Dayton, Ohio. It seats 2,200 people, and has a sound system installed that is tied into the multi-track recording studio described in Brent Harshbarger's *Church Audio* column beginning on page 35. Photo by Robert "Bob" Shively.

Calendar

• SPARS (The Society of Professional Audio Recording Services) will host a weekend technical conference and interface with leading manufacturers of digital audio work stations on May 19 and 20 in Nashville, TN. This gathering follows last September's Chicago conference. SPARS President David Porter will chair the events, with Executive Director Shirley Kaye and SPARS board members participating. Manufacturers giving presentations and offering hands-on demonstrations include AMS, DAR, Dyaxis, Lexicon, New England Digital, Solid State Logic and Waveframe. On Saturday, May 19, the day-long meeting will include in-depth presentations by each manufacturer, a working lunch and cocktails/dinner. Attendees will have an opportunity for hands-on demonstrations and interfacing with manufacturers during the evening. There will be a panel discussion Sunday, May 20 with the manufacturers, followed by further opportunities to experience hands-on operation of the seven different audio workstations.

• A Call for Papers for the 132nd SMPTE Technical Conference and Equipment Exhibit has been issued by Frank J. Haney (Capital Cities/ABC, Inc.), Editorial vice president. The conference will be held at the Jacob K. Javits Convention Center in New York City, Saturday, Oct. 13, through Wednesday, Oct. 17, 1990.

Kerns H. Powers (consultant) has been appointed program chairman. John L. Baptista (Consolidated Film Industries) and Alan S. Godber (National Broadcasting Co.) are the program vice chairmen. John Erwin (Eastman Kodak Co.) is coordinating the film papers.

The 132nd conference's theme is "Film and Television—One World?" Authors are asked to submit a 500-word synopsis and a completed author's form to SMPTE headquarters by June 15 to have a paper considered for the conference.

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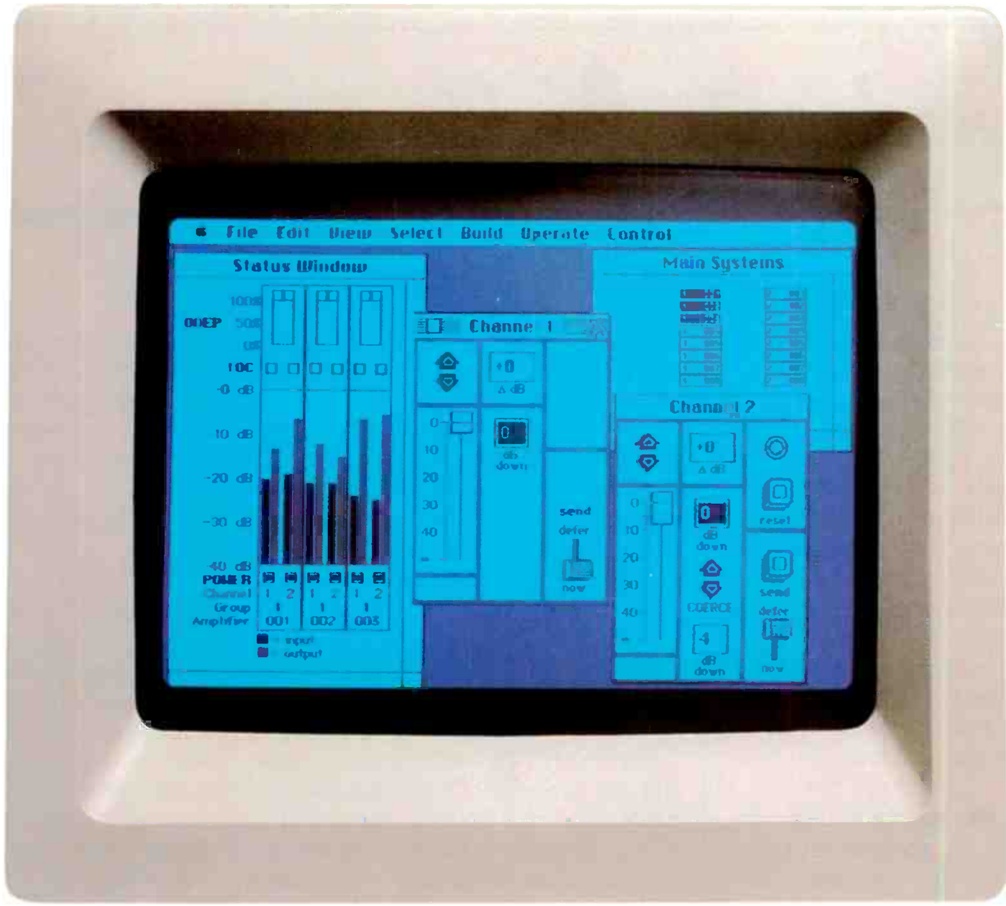

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Broadcast Audio

RANDY HOFFNER

Audio for HDTV: The Production System

• High definition television (HDTV) production systems will ultimately afford viewers improved definition, wide-screen aspect ratio and improved audio. However, the television viewer is not the only anticipated beneficiary of HDTV production systems. The promoters of HDTV production envision that a video production system with resolution comparable to 35 millimeter film will, to some degree, replace film as a production medium for television and for ultimate transfer *to* film for theater projection.

As I have previously stated, improved audio to complement the HDTV production system's improved video is an important aspect of the total HDTV production picture. The expanded uses of HDTV production generate audio considerations beyond the traditional concerns of the television production and broadcasting community, and the higher resolution and wider aspect ratio of high-definition video systems make greater demands on audio than those exacted by the current NTSC system as well.

HDTV FOR A FILM FRONT END?

Several HDTV endorsers envision the use of an HDTV system for the production and post-production of a video product which may be transferred to film for theater projection. If that scenario becomes reality, then the HDTV production system used must be capable of providing the proper audio parameters for such applications. This implies quantitative and qualitative improvements in audio for high-definition production, including that traditionally weak link in the television audio chain, the video tape recorder.

Just as 35-millimeter film's wide video dynamic range demands commensurate dynamic range performance from a high-definition video system used to produce a product it will be transferred to, film's wider audio dynamic range will likewise require audio dynamic range improvements. Such an application requires full audio frequency response of 20 Hz-20 kHz as opposed to the traditional broadcast frequency response of 50 Hz-15 kHz, and distor-

tion levels as low as possible. Fortunately, these attributes, plus absence of interchannel phase or timing error, are inherent qualities of state-of-the-art digital audio recording methods, which HDTV production devices will almost certainly employ.

A major issue regarding audio for high-definition production is the number of audio channels required, and whether these channels should be discrete or matrixed. In the production stages of the process, all audio channels will necessarily be discrete to permit editing. There may be matrixing of certain signals for the final release version of a production, as there is now. Film production for television often employs six audio channels; two each for music, effects and dialogue.

Although standard practice in television and cinematic stereo production is to generally keep dialogue centered, there are exceptions. When someone is speaking from off-screen, for instance, that dialogue is usually panned to the appropriate side of the sonic image.

AUDIO IN HDTV

The end product of stereo-television audio production is usually two audio channels, left and right. While some theatrical release prints are also in this form, many consist of more than two channels. A common multichannel film sound format is the *4.1* channels of surround sound, in which the left and right stereo channels are augmented with a hard, center speaker for centered elements of the sound track, rather than relying on a phantom center image. This is necessary in wide-screen presentation situations where many people are not seated in the "sweet spot" where they will properly hear the phantom center channel generated by left and right speakers.

There is also a surround channel containing ambience sounds which is diffused through the rear of the theater using a number of speakers and a sub-woofer, which accounts for the "0.1" channel. The sub-woofer permits the augmentation of the low-frequency audio spectrum from around 100 Hz down. It is well-known the human ear is less sensitive to sounds in the low-frequency region than to those in the midrange. The use of separate sub-woofers driven only by the low frequency components of the sound track permits these sounds to be played at a higher level than would otherwise be possible, overcoming the human ear's deficiencies in the lower registers. Fortunately, there are virtually no directional cues in the low-frequency range, making it quite practical to combine all low-frequency sounds into one audio signal.

The format for carrying the signals described above varies with the medium. For 35 millimeter film, the four channels of surround sound are matrixed into two to facilitate their inclusion into the two available sound tracks, and they are subsequently decoded into four signals on playback. This format is also used on some programs broadcast on television. For 70 millimeter release prints, the separate components of the surround system are recorded on discrete audio tracks, and indeed, there may be more than four channels. Common supplemental signals include left and right surrounds replacing the single surround channel (five channels) and for projection to very wide screens, augmentation of left, center and right signals with left-center and right-center signals, producing five front and two rear channels for a total of seven.

A high-definition production system that generates product for transfer to film for theater projection must accommodate all audio signals required for the chosen projection format. In current film-production practice, this is often done using a double-system approach, in which audio is stored on a separate medium from video and the two are synchronized for playback. In addition to audio signals, channels must be made available to accommodate whatever ancillary data it may be desirable to include with the audio signals, and multiple languages must also be accommodated. The multiple-language factor is becoming an important one not just for European film producers and broadcasters, but in the United States as well, as an ever-increasing amount of U.S. film and television production is marketed around the world.

STILL MORE TO BE WORKED OUT

Once all the above questions are answered, there are other details to be worked out. These include devising a channel-assignment scheme and the determination of monitor-speaker arrangements for production facilities.

The goal of the architects of HDTV production systems is to produce systems that deliver the video resolution of 35 millimeter film, a wide screen aspect ratio and audio of the highest possible quality. The particulars of an audio system for HDTV production are still in the formative stages.

SUMMARY

Currently, there are several groups working on the standardization of audio parameters for HDTV production. On an international scale, the International Radio Consultative Committee (CCIR) has recently convened an International Working Party, *IWP 10/12, Multichannel Sound Systems Especially Suited to Accompany High-Definition and Enhanced Television Systems*. IWP 10/12 is charged to make recommendations regarding optimum arrangements for audio production and presentation for HDTV. Here in the United States, the Society of Motion Picture and Television Engineers (SMPTE)'s Working Group on High Definition Electronic Production has an ad hoc group addressing High Definition Electronic Production Audio Systems.

There are a great many facets to the high definition audio problem, and at this point in its development, there are definitely more questions than answers. The future should, however, see the development of an excellent high definition audio production system to accompany high definition video. db

JOHN BARILLA

THE ELECTRONIC COTTAGE

The Art of Equalization: Part 2

• In Part one of this article, I surveyed some of the "tools" we can use to craft the tone color (or timbre) of a sound. I also introduced the idea of "conceptual EQ" where one deduces from known information the sonic regions where equalization will be most effective.

But on what basis do we make these deductions? Knowing something about the physics of sound can be very helpful, but before we can truly appreciate the various "tricks" used in the art of equalization, we should first take a look at the theory behind them.

THE COMPLEXITY OF SOUND

Real sounds are tremendously complex, but they can be analyzed and reduced to basic elements: layers of pure sine-waves. Sine-waves are to sound what atoms are to molecules; they are the elemental building blocks from which a foundation may be constructed. The sterile blandness of a droning test-tone generator is an everyday witness to the elemental nature of the sine-wave. Plug the test-tone into an oscilloscope, and you can see the utter simplicity and purity of a sine-wave as it traces its smooth bell-shaped excursion on the screen (see *Figure 1*).

But plug in a real musical sound (a voice, a violin, anything), and the screen will look like a spilled plate of spaghetti. Examine the waveform closely, and you will see an intricate "dance" of sine-waves; various *size* sine-waves all woven together into one cohesive fabric. "Size," of course, is merely a descriptive term. I am referring here to *amplitude* (the graphic depiction of relative loudness) and "frequency" (the number of times per second the waveform goes through a complete cycle which is correlated to pitch). In other words, what we are seeing is a multiplicity of sine-waves, each having a characteristic pitch and loudness, but somehow, perceived by the ear as a single, unified sound.

THE OVERTONE SERIES

Depending on the nature of the sound, many of these interwoven sine-waves may bear a predictable mathematical relationship to each other. This is particularly true for musical sounds or any sound with a definite pitch. This relationship, known as the "overtone series," describes the reality we can see on the oscilloscope. For sounds of definite pitch, there is one main sine-wave responsible for identifying that pitch to the ear, and numerous other sine-waves which are weaker in intensity that give "color" to the sound, but do not contradict the basic pitch. Instead, they support it much like a picture frame highlights a photograph.

Sine-waves can do this without confusing the ear because they are harmonically related and therefore, "masked." They fulfill what the human ear naturally perceives as harmonious (rather than alien) to the "fundamental" sound, and they are able to blend into a total oneness, giving it a unique character.

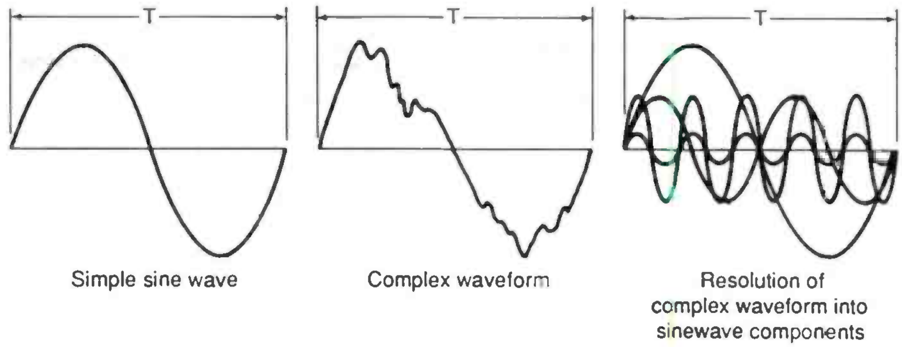

Figure 1. The progression from a simple sine wave to complex waves.

To illustrate this, you can listen to a violin and a viola play the same concert pitch, or do the same with a trumpet and a flugel horn. Same type instrument, same note, but each has an easily identifiable sound, with the differences not being in the fundamental pitch, but in the overtones. This harmonic overtone series follows a simple mathematical relationship; the harmonic overtones are always whole number multiples of the basic pitch (known as the "fundamental").

For example, a second octave A-note on the piano vibrates at a fundamental frequency of 110 Hz (see *Figure 2*).

If we mic'd the sound, ran it through a tightly-cued parametric and swept the center frequency upward from 110 Hz, we should expect to find a distinct resonance at two, three, four, five, six, etc. times the

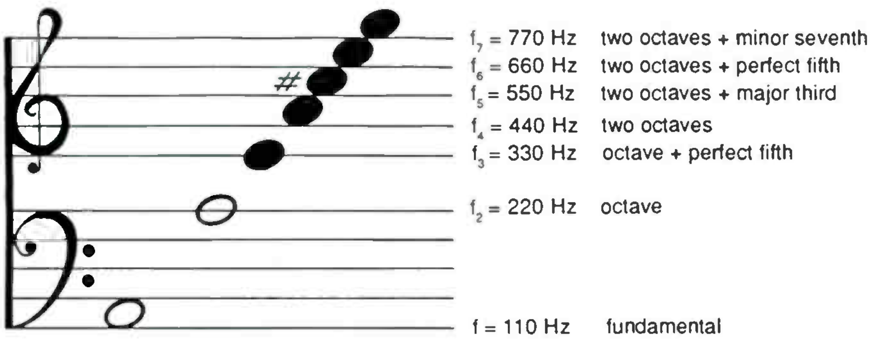

Figure 2. Harmonic overtone series for an A=110 Hz fundamental.

fundamental frequency. In other words, a harmonic overtone resides a 220 Hz, 330 Hz, 440 Hz, 550 Hz and so on.

As we swept the equalizer, we would be able to identify an area of harmonic activity and accentuate or diminish it, thereby changing the "timbre" or tonal color of the instrument to suit our needs. Since you are making music rather than conducting a controlled experiment in a physics laboratory, you are not going to do your equalization by computation. Nevertheless, being aware of the harmonic overtone series, and that you are looking for whole number multiples, can be very helpful in finding a region where equalization can be most effective. Knowing, for example, that the first harmonic above the fundamental (2f or two times the frequency) is an octave above can be very useful. If you find an active frequency, you should automatically try twice, and perhaps even one-half that frequency, to determine which harmonic region can best be manipulated to the desired effect.

Sometimes EQ can be done in a less obvious way in an alternative octave. If you have found one octave, then essentially, you have access to them all, because they are harmonically related.

Another item worth noting about the overtone series is that generally, the harmonics decrease in intensity the higher up they go (there are just a few notable cases where an upper harmonic might have a greater intensity than a lower harmonic, but in general, this is not to be expected). Theoretically, the harmonic series can stretch out pretty far—to the 16th harmonic and beyond—but practically speaking (in terms of equalization), there are limits to where equalization will be effective. Beyond those limits, we may be fishing for harmonics, but what will be dredged up will mostly be noise.

FORMANTS

One very powerful way to accentuate a sound through equalization is to locate the formant regions and boost them. A little bit of boost can be effective here, because formants are not necessarily related to harmonic content. In other words, a vocalist can be singing up high, or down low, on any pitch whatsoever, but there will be a formant region which resonates throughout. The same phenomenon is true for other instruments as well. The formant or resonance band is a constant factor, adding an important component to the characteristic sound.

Formants can be unharmonic because they are closely related to the instrument's architecture, rather than the relationship of overtones. For example, in a stringed instrument, the harmonic motion of the strings will be responsible for creating harmonics that change with the length of the fretted string, whereas the body, the resonant structure, would contribute the formants. To illustrate this further, let's draw on several acoustical factors. Rooms, for example, are said to have certain tonal colorations to them. Much of this has to do with the dimensions of the walls, ceiling and so on. Since audible sound waves have discrete wavelengths ranging from more than 40 feet to the tiniest fraction of an inch, it seems clear that when wavelengths of a sound approach

room dimensions, resonances may build up, giving definite coloration to the sound (it is for this reason that acoustically designed recording studios utilize tuned bass traps and all manner of damping devices to iron out the inherent resonances in the room).

Similarly, other shaped spaces, from a partially-filled drinking glass to the back of a guitar, to the human chest, head and throat, all have real dimensions which vibrate in a characteristic way. Because these instruments have so little mass (relative to a studio wall), they vibrate quite easily, irrespective of harmonic content.

While a resonant harmonic might really be accentuated by the formant, the formant will nevertheless respond to any kind of stimulation—striking the glass, plucking the guitar or the passage of wind through the throat, even if the note played bears no whole number relationship as harmonic overtones do.

Due to the complexity of the structures, there is usually more than one formant area—at least a low formant and a high one—of practical value in the art of equalization.

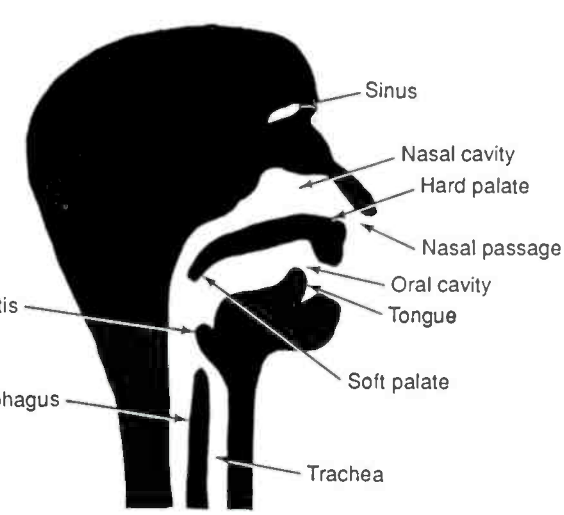

Figure 3. Resonant chambers of the human voice that are responsible for vocal formants.

Formants are still an area somewhat veiled in mystery, particularly for flexible instruments like the human voice in which the shape of the sound chamber changes to form different vowel sounds (see *Figure 3*).

However, research has shown an extremely strong vocal formant for mens' voices centered around 2800 Hz (plus or minus a few hundred Hz to compensate for individual differences). The region is more or less around 3200 Hz for females. There are other formant areas for the human voice. Five hundred Hz for males and 1000Hz for females has been pointed to as a low formant region. It is a good practice to poke around in these regions when looking for an area to EQ a vocal. "Natural" musical instruments all have formant areas that help characterize the sound.

It is for this reason that some digitally sampled grand piano sounds don't quite cut it. Unless the sample was so carefully done that each note was recorded at the proper pitch, with formant in tact, playback at other pitches will cause the formant to be shifted, thereby breaking the natural continuity of the sound.

SUBJECTIVE FACTORS IN EQUALIZATION

So far, we have been dealing with objective factors—quantifiable facts about the structure of the sound source itself. But there is another side to the coin. The inherent biases of the human ear, or the subjective factors, play an extremely important role in how we ultimately approach equalization. Since the aural-perceptual network is the final arbiter of sonic experience, it is very important we understand the typical ways it interacts with a given sound.

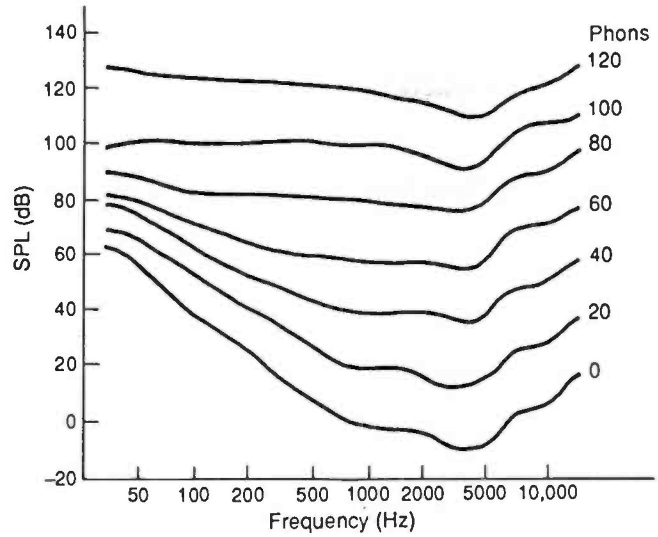

Figure 4. Equal loudness contours.

A classic study in this area done by Fletcher and Munson resulted in a series of curves which depicted "equal loudness contours" for the human ear. While these curves can seem, at first glance, like mere irrelevant academics, the fact is that some great practical lessons can be gleaned from carefully evaluating them.

Each equal loudness contour represents the typical human perception of equal loudness throughout the frequency range (relative to a known standard). In other words, a 60 phon equal loudness contour indicates if a person were to listen to a 1kHz tone at 60 dB SPL, and were allowed to compare and adjust the volume of other tones throughout the audible frequency spectrum until sounding "equally loud" to the 1k tone, the results would be far from linear.

Very low and very high frequencies would have to be pumped up to a much higher SPL in order to be perceived as equally loud. At different reference levels (from 0 phons to 120 phons), the "shape" of the curve changes, indicating that at lower listening levels, substantially more low and high frequencies would have to be added (in order to be perceived as equally loud) than at high listening levels.

At 120 phons (the ear-splitting level frequently reached at your neighborhood disco), the bass response becomes nearly linear, and the treble response is also somewhat less curvaceous than at lower listening levels (*see Figure 4*).

There are many producers, however, who care little about repeated listenings. These people want your attention for 15 or 30 seconds on prime-time television, so midrange excess is often the rule.

The result is that the sound pressure level in the studio while mixing, and that in the listening space of the end-user, is interactive in some very predictable ways.

For example, if a dance-music recording was mixed at an unrealistically low SPL, it would seem horribly low-end heavy when played at the typical high SPL of a dance-club. Likewise, if your generic "dentist's office music" was mixed at a high SPL, it would sound cold and austere, grotesquely lacking in the warmth and brilliance of both low and high frequencies if played back at the typical low SPL of the dentist's office.

These examples all point to the deceptiveness of trying to attain "true" equalization. While the relativity of it all can be a little confusing, the safest hedge seems to be to periodically check a mix at the level it is likely to be played by the typical end-user.

The equal loudness contours also demonstrate one factor not quite so relative. You can rely on this reality no matter what your mixing level: "the ear is hyper-sensitive to midrange frequencies between 2.5 kHz and 5 kHz." Herein lies a blessing and a curse. A good example is in mixing.

Chronically buried tracks, those that never seem to "sit" right in the mix, can easily be made to rise above the pack with a moderate boost at these frequencies. But do that same EQ on too many tracks, and the entire mix will sound edgy and unpleasant. The ear is apparently quite titillated when it hears these frequencies. In proper ratios, these frequencies spice up a meal just like salt does, but in excess, they become cloying. The initial titillation can give way to a subtle, non-descript annoyance for the naive listener, such that the recording will not bear repeated listening.

There are many producers, however, who care little about repeated listenings. These people want your attention for 15 or 30 seconds on prime-time television, so midrange excess is often the rule.

These commercial producers can cram more "apparent loudness" into a TV commercial by loading up on ear-sensitive frequencies, with the effect being universally observable; commercials invariably sound much louder than the program, even though they do not exceed broadcast specifications for loudness. The important academic research of Fletcher and Munson really comes to life here, for it becomes clear that every decibel of sound is not equal to another, in terms of human perception.

This discourse on the art of equalization will continue in the next issue of **db Magazine**. Having surveyed both the tools and the theory of equalization, we will move on to some very practical tricks of the trade—some "how-tos" for various stages of the recording process, from tracking to mastering.

For the uninitiated, we will also include a Berlitz-like manual for speaking "EQeez," including such jargon as "that's a little too fat for me. Trim up the low-end and give me some shimmer on top. While you're at it, try and get rid of that boxiness...." A couple of lessons, and we'll have you speaking like a native.

Stay tuned. db

BRIAN BATTLES

Ad Ventures

Equipment Concerns

• In our ongoing discussion of how to produce great radio commercials, we have made a distinction between two major categories of productions: jingles and voice spots. If your primary interest is in music, you're more likely to be successful in writing and producing jingles.

With all the MIDI equipment available today, and the number of serious recording artists who have ready access to such gear, it has become more profitable than ever to prepare customized musical advertisements. When I began writing this column about six years ago, MIDI was still fairly new, and few smaller studios could afford the hardware. The larger studios, whose clients could afford to pay live musicians, were the only ones who could take advantage of much of the existing MIDI technology. Today, however, MIDI has become almost a consumer commodity, and thousands of musicians, studios and hobbyists own impressive setups, often connected to rather advanced personal computers, and completely configured to handle all but the most exacting recording projects.

Most of the time there is little need for SMPTE code capability, computerized editing and digital sound storage in radio production. Unless dozens of commercials are produced each month with complex soundtracks (and a big budget), high-tech disk-based computer editing workstations with hypertext stacks will not be needed. Two simple multi-track (two or more) analog reel-to-reel decks, a rudimentary mixing board, one or two good mics and assorted accessories will work fine. If you can afford any additional outboard processing "boxes," that's even better. It can certainly come in handy to have a nice reverb unit, equalizer, compressor/expander and whatever else you like to play with.

If you're primarily geared towards recording voice spots, your audio production equipment needs list is the same. If you know how to bounce tracks for taping music, understand how to arrange your mics for the best results and your equipment is maintained properly, you should be able to produce just about any commercial you need. It's not an array of extravagant appliances that distinguishes you from the competition; your creativity, marketing savvy and copywriting skill will be your most meaningful advantages.

In the case of dialogue, humor and dramatic ads, however, you may need to add one more item to your list—sound effects. These are commercially available on conventional LPs, compact discs, open-reel tapes, cassettes and even DAT. You'll need the hardware to play them, but generally, the largest investment is in the "software," or the sound library itself.

Sound effects packages can range from unbearably pitiful to startlingly fabulous. It depends on the source. Prices vary considerably, so you should shop around carefully and talk to as many other people as possible about the sfx libraries they have used.

The point is that you can easily begin a successful and profitable commercial production business with a limited amount of equipment. Since you're in the recording industry, the odds are good that you'll be able to get the most out of your facility when producing commercials.

NOISY RADIO

I've spent a lot of time over the last few months doing unscientific surveys of commercial radio station engineering folks to find out what is the real state-of-the-industry in noise reduction. I receive a hefty stack of professional broadcasting magazines every month, and I read many advertisements and product reviews. Of course, most of the audio suppliers exclaim how their gear incorporates the latest in dbx, Dolby and whatever other noise reduction systems are around.

The area I see as most in need of noise reduction improvement is in broadcast cartridge (cart) equipment. Cart machines are commonly known for low fidelity and high noise. There are many reasons for this. Inferior or badly worded tape, misaligned heads and guides, defective cart housings, infrequent cleaning and so forth, all contribute to the problem. Cleanliness is of utmost importance in a recording studio or radio station. Constant scrubbing of heads, rollers, guides and other associated parts must be done regularly. This is your best and cheapest form of noise reduction. Proper head alignment and degaussing is critical, too.

Yet, advantages can be gained by using noise reduction devices. The better-engineered units are absolutely amazing, while the worst ones can butcher your work. The 24 or so radio station engineers I've contacted make it clear that most of the country's stations are not using any form of noise reduction in their audio chains. Radio broadcasters seem to agree that cart machines would benefit from a simple, standard type of noise reduction scheme. The problem is that the lack of technically competent or experienced equipment operators makes it a headache to implement. For example, dbx and Dolby systems require periodic alignment, calibration or adjustment to ensure proper performance. It's quite a chore to train radio station staff members to handle these tasks, so sometimes the gear inevitably goes "out of spec" through neglect, abuse, hardware failure or improper operating.

Regardless of what equipment manufacturers may want you to believe, the use of commercial outboard noise reduction processing is *not* a universal practice in radio broadcasting today. Maintenance and cleaning must still serve as the primary means of minimizing tape hiss and other problems at the station. As providers of recorded radio commercials, private recording studios should strive to keep tapes as noise-free as possible throughout the recording/editing/mixing/duping stages, understanding and expecting broadcasters to introduce a perceptible level of noise in their own tape playback equipment.

AUDIBLE FEEDBACK

If you are a broadcaster, or if you have experience or a question relating to producing or airing radio commercials, or if you would like to send a photo and brag about your station or studio, send a letter to me in care of **db**. For debate, discussion and arguments, you can get in touch with me via your personal computer and modem. Just log on to any local private bulletin board system (BBS) that carries the Fidonet Broadcast Echo or Fidonet Media Echo, and send me a message. If you happen to be an amateur radio operator, send a packet message to my station, **WA1YUA @ N1API**. By the way, fellow recording/broadcasting hams, just before Christmas, I became Copy Editor for **QST**, the journal of the ARRL, so I'll be exploring ways Amateur Radio experimentation may be of value in commercial production and audio. db

ED LEARNED

Higher Ground: The Winans—Live!

The 80s saw a dramatic increase in the general public's awareness of gospel music. Music once relegated to the church is now enjoyed in the concert hall by both secular and religious listeners. At the forefront of this performance boom is the Winans family of Detroit, Michigan.

This talented family has owned the gospel Grammy awards for the better part of the 80s. The Winans (Marvin, Carvin, Ronald and Michael) have won five Grammys, with BeBe and CeCe Winans picking up one each this year.

One need look no further than the Winans to understand the technical sophistication demanded by today's gospel artists. I've had the pleasure of working with the Winans several times, both as front-of-house and monitor engineer.

THE INTERVIEW

I recently spoke to Steve Fisher, the Winans' current house engineer, prior to their proposed spring tour. Besides a lengthy discussion of the Winans sound requirements, Steve also offers sage advice on the handling of one-nighters and the importance of "making do."

db: In 1985, I worked a Winans performance in Washington, D.C. They appeared as part of a large show sponsored by a local Baptist church. I ended up mixing the group, as they didn't carry their own engineer. Obviously, a lot has changed in five years. How did you get the job as Winans' house engineer?

SF: I'm a staff engineer with Aerial Enterprises Inc., the large sound company in southeastern Michigan. Aerial was contracted by the Winans' promoter to provide sound for their 1987 fall tour. I wore two hats, as Aerial house engineer and Winans mix engineer. Eventually, we dropped the production due to budget considerations, but the Winans asked me to stay on as their traveling engineer.

db: You have an extensive pop music background with artists such as Lionel Richie and Al Jarreau. How different are the technical requirements of a contemporary gospel group such as the Winans from a pop group?

SF: The Winans' show consists of a high-tech pop group that backs an exceptional vocal group singing gospel music. So their requirements are very similar to any high-tech pop group today. The quantity of gear may not be as extensive, but we have the same keyboards—the DX-7, the D-50, even an Emulator—that the big pop groups have. The levels involved are not as great as rock, but the audience is looking for a pumping, funky sound, and the band does deliver that. So you need to have a sound system that will reproduce it.

db: What do you feel is the most important element in your gospel mixing?

SF: Gospel is a vocal-oriented music: you have the word, and the word needs to be heard and intelligible. I concentrate on the vocals as the major thrust of a mix.

db: Describe your most recent dates with the group during the last few months.

SF: Typically, the Winans go out only on weekends, doing little regional jumps by plane. We fly in the day of the show, rent all our band gear, do it and leave the next day. The only gear we carry are the guitars, saxophones, a snare drum, kick drum pedal and a few cymbals. It's a carry-on luggage band all the way.

db: In addition to stage gear, you must also procure your sound equipment locally. What are your minimum channel requirements for the Winans?

SF: We use a real 22 channels from the stage, sometimes 24, so we can fit the group into a 24-channel desk if I use returns instead of inputs for effects. It's much easier with a 32, which I ask for, but don't always get. Often, we do a lot of church concerts where production is not something that has a high priority due to budget. If we do a show where multiple acts are involved, we are pretty much stuck with whatever is there. "Make do" is quite often the order of the day.

db: Can you give us some idea of your range, from best to worst?

SF: High end would be a 40 channel Yamaha PM-3000. Low end would be when a church acted as promoter, and things were deleted out of ignorance, not because of budget. The church technically didn't understand the stuff that was needed; somebody told them the stuff they did have was fine, and they were ready to go with it.

The gig I recall was in Baltimore. We showed up at a major downtown venue that was part of a gospel festival. We were the headliner for the day; the only other artists before us where choirs who didn't demand the sophisticated equipment we did. There was a 16-input Yamaha console on stage, intended for use as a monitor desk. Another console was supposed to be rented for house use, but it never showed up! We had to take the 16-channel desk out into the house and interface into a house PA system that was medium, to be polite. The monitor sends on this desk had to run our monitor system, where we usually have a separate desk and multiple mixes, and then we had to reduce our input list to 16 channels. All this took place at 4 p.m. for a 7 p.m. show, so there was some real scrambling involved. That was definitely the worst.

MONITORS

db: By the way, what are your current monitor requirements?

SF: If we carried production, we'd be using 10 mixes. However, for one-nighters, eight mixes are plenty. We try to keep it as simple as we can, because it's difficult to do one-nighters if your audio needs are complex; there's just not enough time. Our monitors break down to mono side fills, drum monitor and six floor mixes.

db: Let's talk about the individual band setup; start with the drums.

SF: Our drummer, Dana Davis, is a heavy hitter, but his fills are never overbearing and he does leave plenty of space for the other musicians. Typically, we look for a professional Yamaha kit. It's not a huge mondo setup or anything; after all, these are one-nighters and you have to tune the drums and make the setup work. But we do look for two mounted toms and one floor tom in addition to kick, snare, hi-hat and cymbals.

MICROPHONES

db: How many mics do you use on his kit?

SF: The normal number is seven, although that can change with the actual size of the set. Dana does a lot of nice cymbal work, so an overhead is a must. He mounts his cymbals higher up off the drums than a lot of drummers, so bleed isn't as great a factor as it might be. From there, it's kick, snare, hat, two rack mics and a floor tom mic. I can get away with a single mic between the two mounted toms, but if the channels are there, I'll use a separate mic on each tom.

db: What mic do you use on each drum? Start with the kick drum.

SF: I've been using an EV RE-20 for a long time, and I'm very satisfied with it. It doesn't overload, doesn't have proximity effect and it picks up the necessary depth. You can turn up the bass EQ on the kick channel with an RE-20, and it won't sound overloaded or mushy; it's still tight. In some instances, depending on the drum's sound and the sound system, I've found a Sennheiser 421 to be effective because of the way it tailors the low end.

db: How about snare drum?

SF: Snare is as much mic placement as anything. Or, you don't have much time for mic placement and it gets jiggled around to a degree. We usually use a heavy-duty dynamic,

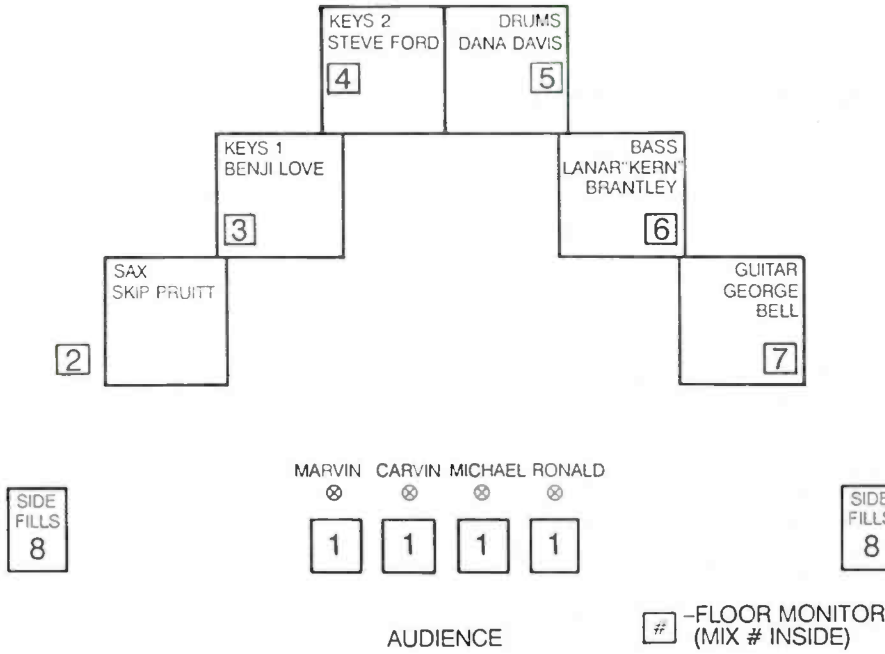

Figure 1. The stage setup used by the Winans.

like a Shure SM-57 or SM-58. Both mics have that enhanced presence peak in the high end that seems to work well on a snare drum.

db: Hi-hat comes next. Would you be likely to use the same type of mic on hi-hat and cymbal overhead?

SF: Not necessarily. For overhead, you'd want a condenser, because it picks up high frequencies extremely well and works at a distance. With a hi-hat, you can get the mic real close. If I only had one high-quality condenser, I would use that for overhead. I'd then use a dynamic with good high end on the hi-hat, close mic'd. But if there were enough condensers, I'd use them on all the cymbals.

db: Which specific mics do you favor here?

SF: The AKG 451 or 460 is something I see a lot, and they work very well; also the Shure SM-81. A mic that I don't see all that often, but one that is excellent for both cymbal work and hi-hats, is the Beyer 160 ribbon mic. Aerial used it extensively at the Montreux-Detroit jazz festival last year, and it sounded fabulous. But you don't often find anyone who will hand you some 160s.

db: Are you using the same type of mic on all your tom-toms?

SF: Yes, and it's what probably has become the industry standard, the Sennheiser 421. Most sound companies have them as their one specialty mic. It does do a great job, and it also works well on percussion: congas, timbales, that sort of thing. We actually used to have some percussion in the Winans, but we no longer do the songs that require it live. Another great mic for toms is the Beyer 201. However, it's a mic that you just don't see that often.

db: We now move to electric bass, played by Lanar "Kern" Brantley. What's he like?

SF: Kern is extremely quick; his lines are melodic and funky at the same time. He is a young player who will be making a big impact very soon. The people in gospel certainly know who he is. A lot of people come out to see the players in this group as well as the Winans. Kern also plays a Roland D-50 keyboard in addition to his electric bass.

BASS RIG

db: What kind of bass rig do you prefer, and how are you taking Kern's different instruments?

SF: We try and use Gallien-Krueger stuff whenever we can. Their stuff is reliable, it has good tone control and it sounds good with a bass guitar. That gives us a constant, which is important, because we're renting. If the channels are there, I take the D-50 on a separate D.I. The bass guitar is also a D.I., usually taken off the amplifier line output. On a Gallien, that's an xlr-out on the back of the amplifier. If the bass rig is lousy, then I have to take the D.I. off the instrument. In those situations, we have to over-EQ the amplifier to get a good sound on stage, so the raw bass guitar ends up sounding better. We do a lot of multiple-act shows, so there really isn't a lot of time to deal with this stuff.

db: You're hitting and running.

SF: That's right. You plug in where you KNOW it's going to work, where you know you'll get a clean signal and you deal from there. If we really hit and run, we will Y the D-50 into the Gallien-Krueger along with the bass, so it's all in one D.I. out. We're assured that by making one line work, we've got everything he's doing out in the house. He has to hear the D-50, so he turns it up to match the bass guitar as he hears it, and that gives me a fairly balanced mix between them.

db: What about electric guitar?

SF: We usually mic the guitar amp, using a general purpose mic like a Shure SM-57 or SM-58. If I have the chance, I'd use a Beyer 88 or EV RE-20. We use a Roland Jazz Chorus amplifier, and our guitarist, George Bell, carries his own pedals and guitar, so essentially he's got his own "sounds" with him. Again, the Roland amp is something that most band rental places anywhere in the United States are likely to have.

KEYBOARDS

db: We now come to the keyboards, played by Benji Love and Steve Ford. Let's start with Benji—what parts does he play, and what keyboards does he use?

SF: Benji plays a digital piano, midi'd to a DX-7. We use a digital piano for its ease of setup and the fact that there's no room on our stage for an acoustic piano. Then we have the sound quality issue. We don't carry a monitor engineer, so now you'd have an acoustic instrument on an electric stage, and have to count on somebody you haven't worked with trying to engineer that for you. We take these concerns right out of the picture by using an electronic piano. The midi'd DX-7 adds a couple of overtones to the piano sound. Sometimes he plays the DX-7, but its primary role is tonal enhancement. I take both the piano and DX-7 with individual D.I.s.

db: As I recall, Benji is a real church-oriented player.

SF: When the Winans depart from their planned program to sermonize or ad-lib between songs, they look to Benji for musical background.

Benji knows the gospel repertoire—he knows all the songs played in the church.

So for background behind preaching, the band looks to Benji for musical direction.

db: What about Steve Ford, the other keyboardist?

SF: Steve is an exceptional keyboard player who gets great sounds. He uses lots of keyboards midi'd together: two DX-7s, two D-50s and an Emulator. He carries his own sounds on discs, and uses the Emulator to call those up. He plays the DX-7s and D-50s for individual stuff. The Emulator is really the only piece of gear we have to rent that's proven to be difficult to find in some places.

db: Do you take all of Steve's keyboards D.I.?

SF: When we carried production, we took individual D.I.s. But since then, we had to change, due to the quick-hit situation that we were often in. We incorporated a mixer and amplifier into Steve's setup, so Steve could get his own sounds without a monitor engineer and monitor system. His stuff is heavily midi'd, so if you were dealing with him for the first time, you might have stuff coming at you from different angles that you wouldn't necessarily grasp. We use a typical stereo system: one side drives his stage system, the other side drives a key mix D.I. It's been very successful for the limitations involved. He's very good at what he does, so it's pretty well pre-mixed by the time it gets to me.

db: Our last band member is saxophonist Skip Pruitt. How many reeds does he play?

SF: He plays a soprano and alto sax. Mic-wise, we've done a number of things. Initially we used a wired mic, which was an AKG D-58E, a mic originally designed as a control room talkback mic. Its small size lent itself well to this application, although its frequency response was not what you'd call warm. Nowadays, he's got a medium-priced Samson radio rig with a small Shure mic; I'm not certain what the mic's model number is. It does take a lot of EQ, but the advantage to a clip-on mic is constant proximity; Skip moves around on stage a lot, and the mic must travel with him.

db: Does he change the mic from one sax to the other, or do you have a channel for each sax?

SF: He changes it himself. Skip is the kind of guy who's audio inclined; he's aware of the mechanical transmission of sound, and is adept at changing the mic without generating a lot of handling noise. I think he mutes his wireless before he makes the switch, but whatever he does, I know I don't have to worry about noise through the system.

db: Now we come to the Winans themselves. Do you have a specific vocal mic of choice for them, and do you carry these with you?

VOCAL MICS

SF: At some of our jobs, the ones I can advance, I do get a choice of vocal mics. But not all our jobs can be advanced. If I'm in a strange place, working on a strange system, I'll ask the local engineers what they normally use for a vocal mic. If they have a mic that works well for them, on their gear, that's what I'll throw up there. I don't want to put a lot of changes into their sound system. If I put up a bunch of esoteric mics with extended frequency response, all of a sudden, the monitor engineer has to deal with a whole new situation and change a bunch of EQs. You (the author) mixed monitors for some of our shows, so you know how that is. Unfortunately, we just don't find many engineers of your competence out there. I find I get the best results by throwing the fewest curves at the engineers. If you minimize the variables, you can get consistency where there might not be any.

db: When I first worked with the Winans, I felt they had the most beautiful blend of male voices I'd ever heard. Each also has his own personal slant on vocal styling. What can you tell us about that?

SF: The Winans have all-encompassing voices, and really put it out there! They are all strong singers, but there are dynamic differences between them. Michael has the softest voice, and the loudest is probably a tie between Ronald and Marvin. Carvin contributes a lot of beautiful falsetto work and high notes. Both Michael and Carvin can hit the high notes, and often sing the high parts in four-part harmonies, but there is a lot of switching around in who sings what part from song to song. It makes it difficult to track, but it certainly lends different sounds to the ensemble blend from song to song.

db: Do you feel you've got a good handle on who's singing what at this point?

SF: I've been mixing these guys for two years now, and there are still things that surprise me. When we were touring steadily, I had it in my back pocket: you were doing it every night and could get a good groove going. With these intermittent one-nighters, it's not that I forget words, phrases, or harmonies. It's like "where are they in this mix?" Things come and go, and it takes a while to find that comfort zone. These guys work the mics well, and they know how to sing together. So as far as vocal blend is concerned, they make my job easy.

MAINTAINING A VOCAL BLEND

db: What do you feel is the most important technical consideration for maintaining a good vocal blend?

SF: Good monitors! We take great care to set the vocal monitoring situation so they can sing within themselves and keep the harmonies balanced on stage. I can recall two shows with these guys where I didn't really have time to walk through the monitors and check vocal balances. As a result, the vocals fell apart onstage, which meant they fell apart in the house.

db: The Winans' setup keeps the whole downstage area clear for the singers. The downstage wedges and side fills are primarily vocals, as I recall.

SF: That's right. The downstage wedges are predominantly vocal, with a bit of Benji's piano for the brothers to sing against. Sometimes we do two mixes down there: a mix for Marvin and Carvin, and a mix for Ronald and Michael. Marvin's and Carvin's voices would be slightly hotter in the wedges in front of them, for instance. These wedges are never all the way downstage; we keep them at least three feet upstage of the front edge of the stage so the brothers can work the downstage area if they want to.

db: You keep the monitors tight to the mic stands, leaving a downstage apron in front for them to work.

SF: Someone might say "why use wedges at all?" Because it gives them all something to go back to. If they are struggling for level, or they want to harmonize in a very tight manner, they can go back to their mic stand, put the mic in and have tight harmonies right in their face. Side fills are very important because they do wander all over the stage, and they want to hear a good mix of their vocals as they travel around. The wedges are really used as spot monitors.

db: Besides vocals, what else do you have in the side fills?

SF: Again, we have some of Benji's piano. If we're playing a large show on a big stage where sounds don't travel well, we will put a taste of snare and hi-hat through there for timing purposes. When we carried production, we ended up with a lot of drums and other stuff in there, but for one-nighters, we want to keep it simple. We don't want a lot of sounds going in a lot of different directions. We take some of the bass out of the piano to improve its clarity, and do the same with the snare.

db: With respect to the other mixes, each musician gets his own mix.

SF: And those get pretty extensive because of our stage setup. The band is not tightly packed; they are standing away from each other and the center is open. It looks great, but it is hard for the musicians to hear each other acoustically. So they need to hear a lot of each other in their respective mixes.

db: Is there much vocal in these instrument mixes?

SF: Not really. The side fills throw enough back there to do the job, although we might throw a little vocal into the drum mix because Dana is the furthest back.

SPECIAL EFFECTS

db: What do you use in the way of effects?

SF: I use reverb on the vocals in almost any situation. It really enhances the Winans' voices; we don't overdo it, but it's there to be heard. I usually ask for two digital reverbs and a digital delay. I like the delay to be capable of one second at full bandwidth. I take the highest quality reverb available and use it on the voices. The delay is also used on the voices. We use some very decayed long echoes and, in one song, a doubling effect. The first is around 630 ms, the second around 100 ms. The second reverb is used on the drums. We do a lot of poppy stuff, and with the way Dana plays, there is room for that.

db: What sort of reverb programs do you favor?

SF: I try and set up the reverb by myself in headsets, with the vocal mic *du-jour*. Then I listen to it over the sound system. I'm looking for the biggest, most lush sound I can get with a crystalline high end. I'm not looking for a small room. I really like the Yamaha REV-7; with the onboard EQ I can tweak the reverb and be very successful in enriching the sound without losing clarity in a given room. On a REV-7, I would use the large hall program, varying the reverb time in response to room reverberation. The stock decay time for this program is 2.6 seconds: I would never go longer than that, and if the room was really live, I might have dialed down to around two seconds. I'm happy with the stock 30 ms of pre-delay. On the drum reverb, I use gated reverb, early reflection programs and some regular reverb for cross-stick snare on ballads.

USING VOCAL COMPRESSION

db: Let's talk about patchable compression a bit. I'm sure you use compression on the vocals—with these guys, you'd be crazy not to.

SF: It would be too much, because of the dynamics of the Winans' voices, not to compress. It just helps the levels stay in control. The best thing that's ever worked is to have a DBX 900 rack, and insert compressors on each vocal channel, because dynamics are totally different from voice to voice. Of course, I don't often have that many limiters available. I'll sometimes use two submasters, with limiters inserted in each, and put the lead vocal in one, the backgrounds in the other. If I only have one high-quality compressor, I'll insert that on a single vocal sub and assign all the vocals there.

db: How do you handle compression ratios and threshold settings?

SF: The obvious reason for compression is control. If somebody decides to scream, we don't want to blow the audience away. But I'm not looking to compress the stuffing out of it—there are a lot of dynamic changes in these songs, and I really don't want to change that too much. Often I come up with a 2:1 with a slightly lower threshold than I'd use for pure scream protection. If you watch it, you can see it working most of the time, but the signal can still rise and fall. If I can keep the vocal dynamics in a tighter window, so to speak, I'm better off for it and so is the audience.

db: Do you use compression on anything else?

SF: I definitely need one on the bass guitar, and also on the keyboard rigs: two for Benji and one for Steve. These guys can funk with the best of 'em, and when the band really gets cranking, those compressors come in real handy.

db: Speaking of cranking it up—how is stage volume with these guys?

SF: It's quite good, which is probably our saving grace. Kern, the bass player, has a tendency to go overboard and really turn it up, and the other guys tend to follow him because he's a natural leader. Fortunately, the Winans have an exceptional grasp of the *feel* of gospel music, and really are adept at controlling dynamics. Whenever the volume starts coming up, Marvin or Carvin will turn around and ask "what's going on? We need to look at what we're doing here—we need to relax and regain our balance." That's great, because I'm in the house and don't always have time to tell them that myself. It is important to have powerful music, but in gospel, the word is more important. Keeping stage volume at the appropriate level for singers is a big part of our success. db

WINANS INPUT LIST

The following input list shows the complexity of a full Winans production. This is what the Winans would use if they carried production, and does not reflect the trimming required for one-nighters.

CHANNEL #	INPUT	MICROPHONE/D.I.
1.	Marvin Vocal	EV ND-857
2.	Carvin Vocal	EV ND-857
3.	Michael Vocal	EV ND-857
4.	Ronald Vocal	EV ND-857
5.	Off-stage Announce	EV ND-857
6.	Sax	Radio Clip-on Mic
7.	Key #1—Digital Piano	D.I.
8.	Key #1—DX-7	D.I.
9.	Key #2—D-50 #1	D.I.
10.	Key #2—D-50 #2	D.I.
11.	Key #2—DX-7 #1	D.I.
12.	Key #2—DX-7 #2	D.I.
13.	Key #2—Emulator	D.I.
14.	Key #2—Mix L	D.I.
15.	Key #2—Mix R	D.I.
16.	Bass Amp Line Out	D.I.
17.	Bass Key (D-50)	D.I.
18.	Guitar Amp	EV RE-20
19.	Kick	EV RE-20
20.	Snare	Shure SM-58
21.	Hi-hat	Beyer M-160
22.	Rack Tom #1	Senn. MD-421
23.	Rack Tom #2	Senn. MD-421
24.	Floor Tom	Senn. MD-421
25.	OH L	Beyer M-160
26.	OH R	Beyer M-160
27.	SPX-90 L Return	
28.	SPX-90 R Return	
29.	DDL Return	
30.	REV-7 L Return	
31.	REV-7 R Return	

HOT LINE

DUPING AND BUSINESS

I have been a subscriber of yours for some time now, and have just opened my own demo recording studio in Gastonia, N.C. Several people have asked me about cassette duplication. There are several companies in nearby Charlotte, N.C. that duplicate, but I would like to have my own duplication machines to make the clients' copies myself. Can you give me the names and addresses of individuals or companies that I could write to about purchasing new or used duplicating machines?

Also, are there any books on the market that deal with the start up of an independent record company? I need sample recording contracts, guides to record distribution, how to copyright songs, how a publishing company works, etc. Any information you can supply is appreciated.

Doug Henry
Tornado Recording
Gastonia, N.C.

Replies from Senior Editor John Barilla:

• To answer your question about tape duping, I contacted Ed Adams of Amphion Recording in Rockville, CT. Ed has been involved in large-scale duping for over a decade, and was able to offer some sage advice. He mentioned that the field can be divided into two types of systems—high speed (which dupe at the rate of four or eight times normal speed) and real-time or near-real time systems (which may go up to two times normal speed).

While acknowledging there are many good systems on the market, Ed did have a few recommendations (he assumed, of course, that you want professional equipment, which is much more rugged and also more expensive than consumer gear). Of the possibilities in high-speed duping, Ed recommended both Otari and Telex as quality/cost-effective systems. Both offer master with multiple slaves (the master can either be a reel or cassette).

Ed's personal choice was the system offered by KABA Research, which runs at two times normal speed. The KABA system is a four-track system (meaning it records both sides simultaneously), and each slave has two transports. KABA also offers integrated start-up systems with cabinets and shrink-wrap machines all tied in. To find out more about KABA, Ed suggested calling Kenneth Bacon at 1-800-231-TAPE and mentioning Ed Adams.

The question about starting your own record company is more difficult to give a definitive answer. To my knowledge, there is no one resource strictly dealing with this issue. There are, however, several books that can offer some valuable insights on the music business as a whole. The classic volume is "This Business of Music" by Shemel and Krasilovsky. Other good resources would be "The Encyclopedia of the Music Business" by Harvey Rachlin, "Making It In the Music Business" by James Riordan and "How To Make and Sell Your Own Record" by Diane Rapaport.

AUDIO EDUCATION

As a college sophomore interested in pursuing a career in sound engineering, I am looking for a university where I can graduate with such a degree. Would you please send me a short list of popular colleges and the course curriculum names I should be looking for.

Carl Ware
Wimberly, TX

• Well, Carl, we can't send you a list of colleges featuring recording

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arts, music technology and recording engineering (or whatever they want to call it).

However, we do know where you can get a list. The Society of Professional Audio Recording Studios (SPARS) has an educational committee. Since 1985, SPARS has offered a National Studio Examination to prospective recording engineers, so employers may have a gauge to measure competency.

Colleges, universities and trade schools have different approaches to the curriculum, some featuring more music, more mathematics or more hands-on. So pick a school carefully! But if anyone can give you a comprehensive list, SPARS can.

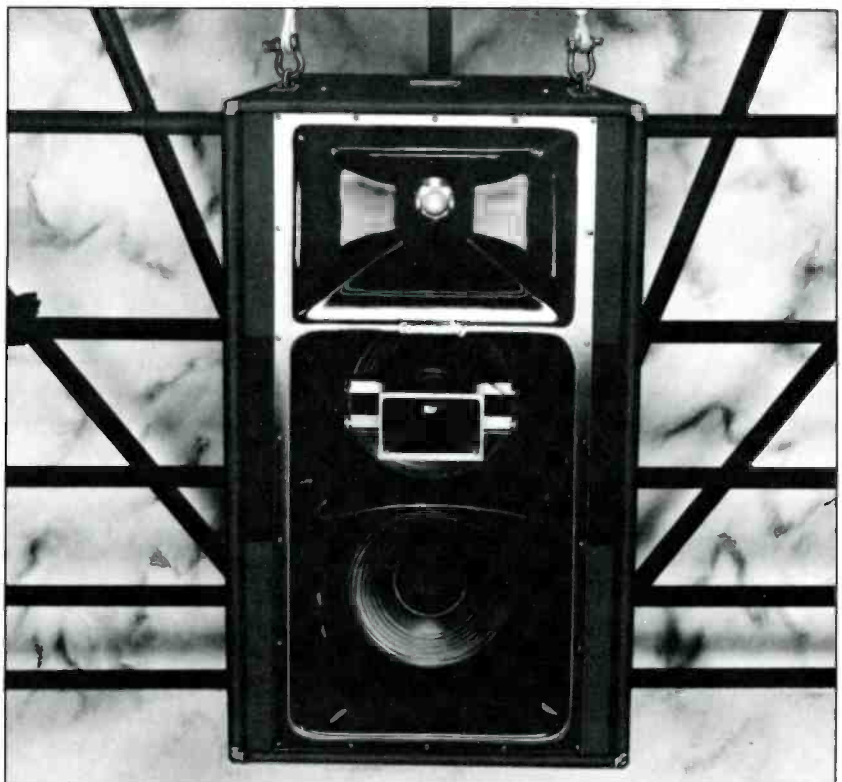
Contact them by mail or phone: SPARS, P.O. Box 11333, Beverly Hills, CA 90213. Tel:(213) 651-4944.

RECORDING SERMONS

I am in charge of recording, editing and duplicating my pastor's Sunday morning sermons. My problem is this: The pastor is a wonderful, dynamic preacher, but he sometimes goes from a barely audible whisper to an extremely loud delivery. He also moves on and off mic quite a bit. I can't really ask him to change his method of preaching as the message is the most important component, but I'm not very happy with the quality of the recordings I'm getting. In order to record the sermon without distorting the tape, I need to run it at a relatively low overall level, which, unfortunately, makes for a very noisy tape. I did try compressing the mic signal, but found that compression made it sound quite unnatural and it seemed to actually add noise. Do you have any good ideas for upgrading the system without bankrupting the church?

Eli Grandefield
Canyon City, OR

• Your problem is not unusual, and I believe it can be solved fairly inexpensively. I am assuming you are using a quality microphone and cabling. That being the case, the setting you are using on your compressor may be part of your noise problem. Most compressors can function also as limiters, so I suggest you rethink the settings in this direction. A typical mistake people make is setting the threshold too low so compression is almost constantly working. This can give an audible pumping action which sounds a little contrived for a sermon. Also, since it is reducing the gain of the microphone so radically, the noise inherent in the system is, in effect, being exaggerated when you compensate for the loss by increasing your recording level. It would be better to operate the compressor in a "hard limiting" mode: high ratio (10:1 or more) and high threshold. This way, the gain reduction would occur radically on extreme peaks only, but would be negligible at all other times. I am sure this



kind of setting would improve the overall performance of your compressor.

I would also look for an improvement of the sound at the source, rather than relying totally on downstream signal processing. That's how you will get your best signal-to-noise ratio on tape. Most pulpit microphones are stand-mounted cardioid pattern mics. The narrowness of the cardioid pattern may be the wrong choice for a speaker who moves around a lot.

The sound of this type of mic also changes radically when someone is far away or close. You really need to have a pretty stationary speaker, or one with consciously good mic technique, to get a good recording this way.

Perhaps you might best switch to a PZM microphone mounted on the pulpit (PZMs are manufactured by Crown and several other companies who license the PZM concept). PZMs exhibit a wide pattern that doesn't change much in frequency response even if the speaker turns off axis. One contingency: if your pastor is fond of pulpit pounding, you should mount it on a stand independent of the pulpit.

Another way to go is with a lavaliere (tie-clip mounted) microphone. These pick up, predominantly, vibrations from the chest cavity, so they are relatively immune to the kind of needle pinning a direct blast of air can give to a conventional microphone. One contingency here as well: tripping over the wire can sometimes be a problem, so most pastors prefer to use a wireless system.

Finally, suppressing noise from your analog tape recorder can also improve the quality of your tapes. Since you say you are editing, I assume you are recording on a reel-to-reel system. I would highly recommend an encode/decode noise reduction system like dbx Type I (or similar systems made by other manufacturers).

The difference in sonic quality can be truly inspiring. Let us know how it goes. db

SHELLEY HERMAN

A Specialized Synagogue Sound System

In 1962, the building committee of Valley Beth Shalom in Encino, Calif. decided to erect a Sanctuary appropriate for what was to become the largest and best-attended synagogue in the San Fernando Valley section of Los Angeles

Being in the sound business, already maintaining the existing equipment and singing in the choir, I was approached to design and install the sound system.

I examined the plans and suggested they be examined by a competent acoustics firm to ensure the acoustics were correct before a sound system was installed. The leading acoustics firm at that time was Bolt, Beranek and Newman, and they made the necessary changes. They did an excellent job, and many people feel the Sanctuary portion of the building is the best-sounding small auditorium in the city. I also convinced VBS to remove the sound system from the main building contract and out of the hands of the electrical people, except for conduit runs, a.c. to the correct places and necessary support devices for speaker hanging.

The building is a 130-ft x 50-ft. room divided into three sections. The largest section is the Sanctuary, at the south end of the building. The Sanctuary houses the Holy Ark, two choir-lofts, pulpits and other necessary items to conduct services. This area, a raised stage, is called the *Bima*. The Sanctuary has about 600 fixed seats, with a sloping floor in the classic three-aisle auditorium layout. The second largest room, the Auditorium, is at the north end of the building. It has a full stage, a flat floor with movable seating and a coverable dance floor. The third room, the Foyer, is between the Sanctuary and Auditorium. The three sections are divided by two moveable, sectored walls made of wood, which when fully extended, are never straight. They present a zigzag pattern to oncoming sound so it is never reflected directly back to the microphone positions (this was a BBN suggestion). The moveable walls do not have very much isolation, but that was not a design criteria, and "Air Walls" had not yet been invented.

ALTERNATE ROOM USES

The building is used in several different modes. For regular Friday evening and Sabbath services, the Sanctuary section is used alone. If attendance exceeds Sanctuary capacity, the Sanctuary/Foyer wall is opened, and the two sections are used. For the High Holy day services (Rosh Hashana and Yom Kippur), when maximum capacity is needed, both dividing walls are opened and all seats are faced to the Bima, including about 100 seats on the Auditorium stage. This gives a maximum capacity of about 1,150 for services.

If there is a stage presentation, the chairs in the Auditorium are faced toward the stage. If the crowd is large enough, the Auditorium/Foyer doors are opened for a 550-seat capacity. Because the seats in the Sanctuary are fixed and cannot be turned around, the Sanctuary/Foyer doors remain closed.

WHAT THE SOUND SYSTEM MUST DO

The sound system has to cover all aforementioned situations, including the added problem that the choir is situated behind the Cantor and cannot see his face nor hear him, yet it is necessary for the choir to respond to the Cantor, preferably in rhythm and on pitch. The three sections of the facility demand several sets of loudspeakers which are turned on and off, with the resultant echo and Haas effect. It must be an unattended system, able to be used by non-technical personnel.

The original proposal was to install a *Big Green* system. However, budget constraints dictated a Bogen system be installed. Bogen constructed the rack at their factory, to our specifications, and shipped it to the site ready for installation. The mixers were Bogen MXMs and the amplifiers were MO60" and MO100s. There was no equalization, as that technology was only academic then. All units were vacuum tube devices, because the transistor had not yet found its way into sound reinforcement systems. The only device able to supply delay at that time was a tape loop system which was too expensive and insufficiently reliable for an unmanned system. We had to live with the echo and Haas effect problems until something better came along.

The Cantor/Choir problem also had to be solved. In 1962, nobody had ever heard of anything called *foldback*, so the simple expedient of providing a separate mixer for the choir-loft microphones and the Bima microphones was used. Both mixers fed the main amplifier system, while the Bima mixer also fed an amplifier connected to the speakers in the choir-loft. By judiciously positioning the microphones and loudspeaker in the choir-loft, the choir was able to hear the Cantor, while being amplified without amplifying the sound emanating from the choir-loft loudspeakers.

In the mid-70s, the tube equipment started to tire, and all amplifiers and mixers were replaced with solid state devices. At the same time, another dual room building was added to the complex, and similar equipment was used in that building, ensuring exchangeability of equipment in case of failure.

The loudspeakers used in the original design were University Sound

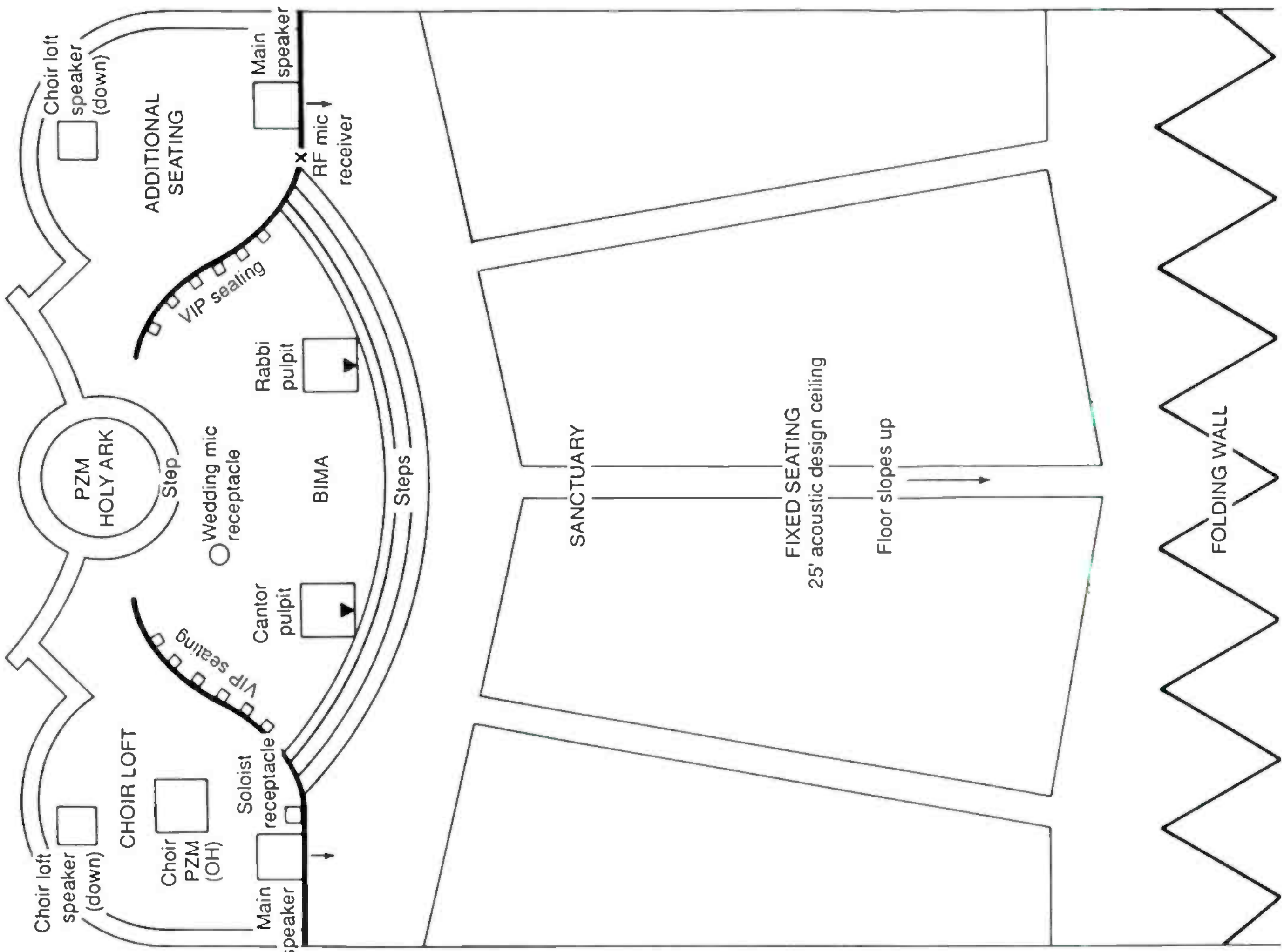


Figure 1. (Left Side) The complete Sanctuary layout. Bima seating has speakers located between the seats.

Columns. Twenty-eight years later, this does not seem to be a very good choice, but at the time, sound columns were the *hot* speakers. Today, two of them are still in operation, doing an excellent job.

As the years went by, and the congregation became more sophisticated in their listening habits, the sound columns were replaced with two-way systems, with the resultant increase in quality. Of course, when solid state delay devices became available, a Lexicon 92 was immediately installed to handle the Haas effect problems, which it still does very well.

SUDDEN RADICAL CHANGE

In the early 1980s, the entire sound system was stolen. The mic lines were unplugged, speaker lines cut and the whole rack was gone. After the initial shock, I arranged for a temporary system to handle Sabbath services and set about designing and building a new system. In designing the new system, I was handed a clean sheet of paper, but was unable to improve on the original 20-year-old design philosophy. I was, of course, able to use newer equipment, add processing to three of the main microphone channels and add the latest power amplifiers, but the basic layout remained the same. It is the system as it exists today, that I will describe.

The system, like the building (and Gaul), is divided into the Sanctuary System, the Auditorium System and the Foyer system.

THE SYSTEMS

SANCTUARY INPUTS

The Sanctuary System has 10 inputs: Rabbi's Mic (AKG 451); Cantor's Mic (AKG 451); Holy Ark mic (PZM); Radio Mic (Vega); Choir overhead mic (PZM); Choir soloist mic (SM58); Bima Wedding mic (SM58); Tape playback; Local rack mic plug; and one unused input. The Rabbi, Cantor and Radio mics are each processed by a Valley People device that provides compression, expansion and equalization.

SANCTUARY MIXERS

The mixers are a Shure M67 with a M677 expansion device. These are sufficiently quiet devices, and by using the M677 expander for all of the Bima mics, and the M67 for the choirloft mics, the necessary split is accomplished by using the microphone level output on the M677 to drive the choirloft system.

AUDITORIUM MIXERS

The mixers are a Shure M67 plus M677, through a small jack bay to input connectors around the stage area. Appropriate microphones are used for most small productions. When a major production is attempted, a console is connected to the system.

LIMITERS

The limiters used for both main systems are UREI LA-5s. They were

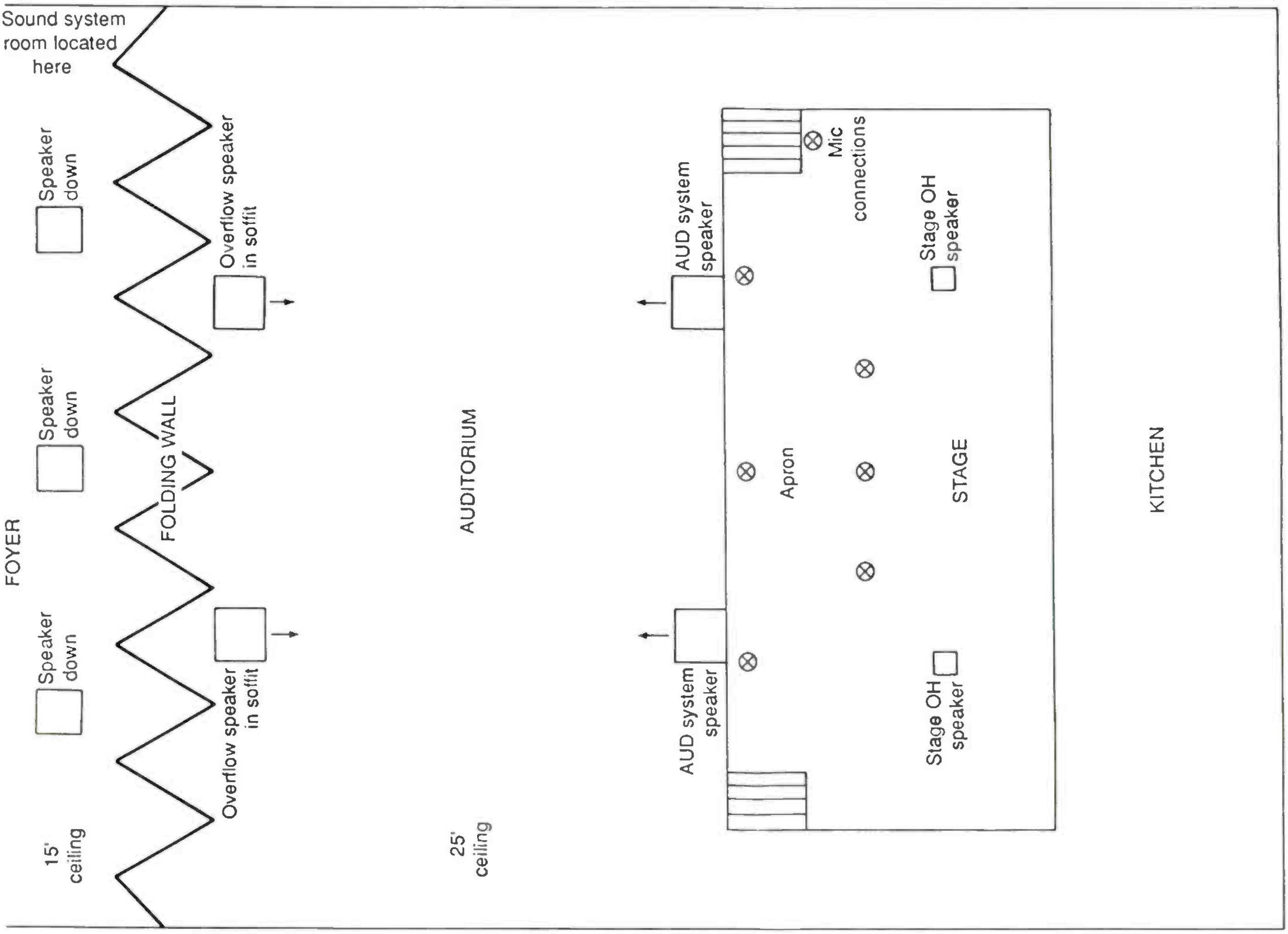


Figure 1. Right Side.

chosen because they were specifically designed for this type of service.

EQUALIZERS

The Sanctuary main system equalizer is a UREI 527 27 band. The equalizers for the Choirloft, Overflow and Foyer are Shure 107 octave band equalizers. The Auditorium equalizer is a White 4001 with a 500Hz crossover built in.

DELAY

Lexicon 92 two-output delay is connected to the main mixer output after the limiter, but before the equalizer. The outputs go to the Foyer and Overflow systems. No delay is necessary when using the Foyer with the Auditorium.

AMPLIFIERS

The amplifier for the main Sanctuary speakers is a BGW 250D. The Foyer and Overflow amplifier is a BGW 320B, the Choirloft amplifier is a BGW 2125 and the Stage Overhead and Hallway speakers are driven by a Crown D75 through 70V transformers. The Auditorium amplifiers are Altec 1594 solid state amplifiers, which are scheduled for replacement.

LOUDSPEAKERS

The Sanctuary loudspeakers are two Altec 9844 two-way 800Hz systems with self-contained crossovers. It was the original intention to use larger units, but space and budget constraints dictated this unit, and they work very well in this service.

The Auditorium speakers are Altec two-way 500Hz systems, biamplified, with the crossover located in the equalizer.

The choir-loft and Foyer speakers are 8-inch University devices, in BBN specially designed boxes. They have been used since 1963. The choir-loft amplifier also has several small 'box' speakers attached to allow people seated on the Bima behind the Rabbi and Cantor to hear.

The Overflow speakers are University CS4 sound columns, also from the original installation. The drivers have been replaced as a result of a well-known brand of amplifier with no speaker protection. The stage overhead and hallway speakers are consumer units.

SWITCHING PANEL

All switching is done at line level. There are no switches at speaker level.

A three-position switch allows the Foyer system to be connected to the Auditorium system, to be silenced, or connected to the Sanctuary system via the first delay output.

A two-position switch connects the Sanctuary program, via the second delay output and an amplifier, into two speakers facing the Sanctuary end of the Auditorium for fill when the Auditorium is being used as part of the Sanctuary. This is called Overflow.

Two three-position switches control the inputs to two halves of a small amplifier. One allows either program to be heard in an outside

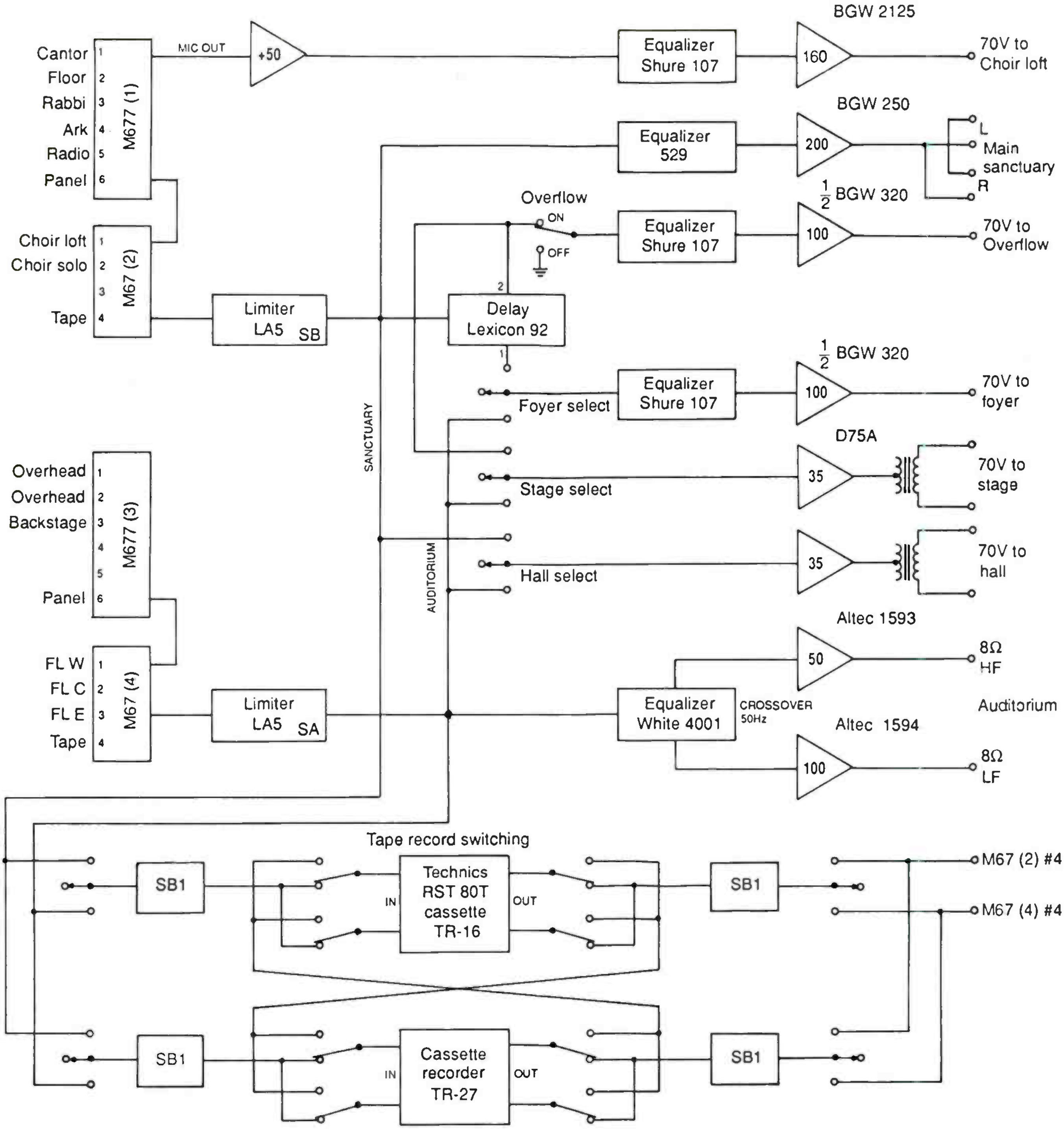


Figure 2. The tape-switching system used.

hallway, and the other allows either program to be heard in two speakers mounted overhead in the stage area; the Sanctuary program, via the second delay output for High Holy day Services; the Auditorium program for cues when being used for stage production.

Two switches route the inputs and outputs of two cassette recorders into both systems. I have selected Technics RST 80R dual cassette decks as the ideal machine for this service, as two 90-minute cassettes can be inserted in the deck, and the unit will record for 180 minutes without interruption.

One additional switch allows connection to a system in another building. XL Connectors are provided on the switch panel to access the inputs and outputs of both systems.

CONCLUSION

The design and construction of this system proves it is not necessary to use the very latest, fanciest devices in every system, especially when they are not required.

The old KISS principle works very well; the less there is to the system, the less there is to fail. If the quantity of devices is kept to a minimum, the sophistication does not exceed the level necessary and the equipment is of the highest quality, you will be assured of a reliable system. db

JOHN BARILLA

New Sound, New Songs

In the past decade, churches may have come to view audio technology in an entirely new light. Once the intention of installing a public address system was simply to allow the minister's message to be heard throughout the congregation, whereas today, a much more extensive sound system is generally the rule.

Why do we have this shift in priorities? It is certainly a reflection of a movement within the Church itself to restore authentically Biblical modes of worship. The scriptures—especially the Psalms—fairly abound in prescriptions for worship which call for powerful, jubilant celebration using diverse musical instruments. The modern Church has, in a sense, rediscovered these ancient directives and appropriated them as a word for today. The result of this theological emphasis has been the elevation of music to an extremely important position in the worship service.

One church which exemplifies this movement is Redeeming Love Christian Center (RLCC) in Nanuet, New York. Located in a beautiful suburban setting less than 30 minutes from New York City, RLCC is a ministry that has long understood the power of music as a vehicle for spiritual instruction. The church, which is co-pastored by Clinton Utterbach and his wife, Sarah, has gained a reputation for its excellence in music—most of which is written and performed by Pastor Clinton. db Magazine recently visited RLCC, and was fortunate enough to spend an afternoon with Media Manager, Ed Pfundstein, and Sound System Designer Wally Duguid.

HISTORY OF RLCC

Coming upon a hill crest in Nanuet, one can see from the main road the silhouette of a large, domed edifice located on a sizeable campus. Aside from the obvious symbolism of a steeple, the clean modern lines emit none of the architectural cues we normally associate with a church. As I visited with Pfundstein, I realized the building was designed for purposes other than worship, and yet how these original purposes were, providentially, to be transformed into a worship space. Pfundstein recounts the history of the building RLCC now occupies:

"It was a former dinner theater—actually (first) a theater-in-the-round—mainly for Las Vegas-type entertainment during the early 70s. That went bankrupt and was purchased by another company and called The Coachlight Dinner Theater," Pfundstein said. "They modified the interior replacing the bucket seats with flat levels for dinner tables and they put on off-Broadway shows. That didn't pan out for more than a couple of years, and it was a mess—vandalized and so forth," he said.

The saga unfolded as Pfundstein described how the defunct dinner theater was then purchased by the RKO corporation who intended to make this sizeable facility into a 10-plex theater. RKO, however, had purchased the facility without gaining final approval of the town planning committee in Nanuet. The town did not see it in the public interest to have a multi-theater complex at that location, and the building permit was denied. So the facility lied empty once again. Upon hearing of the church's interest in the property, RKO was eager to negotiate. The church purchased the building in 1985.

Of course, the ministry of Redeeming Love Christian Center was not formed overnight. Pastor Clinton Utterbach, a well-known choir director since the 1960s, became a recording artist and is still under contract with a major record company. During this time, he and his wife attended a Bible college and upon returning home, began teaching a Bible study group in their living room. This grew into a 1,000-member congregation in Hackensack, N.J., and subsequently, the 3,000-member Nanuet congregation—all in 10 years.

Pfundstein has been with the ministry for most of their stay. Having a background in electrical engineering and computer programming, Pfundstein was aptly suited to his position, although he admits the "hands-on" aspect of running an audio ministry is something that required much learning and growing. Today, Pfundstein is not only in charge of sound system operations, but is also responsible for the radio production which reaches an international audience. Herein lies the focus of this article. While the newly-installed sound system at RLCC is now a model of excellence and ease of operation, this was not always the case.

THE SONIC PROBLEM DEFINED

RLCC inherited an obsolete sound system when they bought the building in 1985. The components were mostly 16-year-old Altec "Voice of the Theater" speakers, and while some improvements had been made in terms of power amplification, the system was still fraught with engineering problems. The system was designed for theater, not music; certainly not the powerful gospel music created by Pastor Utterbach and the various other music ministers and artists who frequently concertized at RLCC. What appears as a barely noticeable acoustical problem when distant mic'ing dialogue (as in a theatrical presentation), becomes extremely aggravated when dealing with complex contemporary music featuring both traditional (grand piano, drums, guitar and bass), electronic instruments (the Kurzweil 250 and Roland D-50), soloists, vocal ensembles and a large choir.

Ivory Audio, of Lindale, Texas, which specializes in designing and installing church sound systems, was called in to analyze and remedy the problem. I was fortunate to catch Ivory Audio's chief engineer supervise the implementation of final

Figure 1. The former revolving stage of the dinner theater is now the main Sanctuary.

"bells and whistles" on the new sound system. Wally Duguid (pronounced "do good") has a background in broadcast and studio mixing, as well as maintenance, having been chief engineer at Cherokee Recording and several other studios in Los Angeles prior to moving to Texas. He has also done live sound for an assortment of touring gospel groups, and has done television mixing for the Billy Graham Crusades for the past 16 years. Six years ago, Duguid decided to focus on sound system installation, and found himself doing a lot of retrofits on growing churches. On typical installations, such as RLCC, Duguid designs the renovated system, and then has Acoustical Consultant George Augspurger of Perceptions, Inc. of Los Angeles, offer his feedback on the plan before laying a single cable. This redundancy in the design process enables Duguid to install systems that conform quite well to the design specifications, with a minimum of hassles.

As previously mentioned, the major problem with RLCC's system was the obsolescence of its components. But there were other considerations which impacted on designing the renovated system. The original system was operated from a tech booth one story above the congregation, which was essentially out of the house's sound field. While this may serve adequately for dialogue mixing, it is impossible to get an accurate mix at this location. High frequency sound components, which are extremely directional, never make it up to the sound booth. Hence, a mixer might naively assume that the muddy woofiness he hears in the booth actually portrays what is being heard in the house. If he does not temper his perceptions, he will probably fall to the folly of making the mix sound good to his own ears—and consequently, unduly harsh to the congregation below.

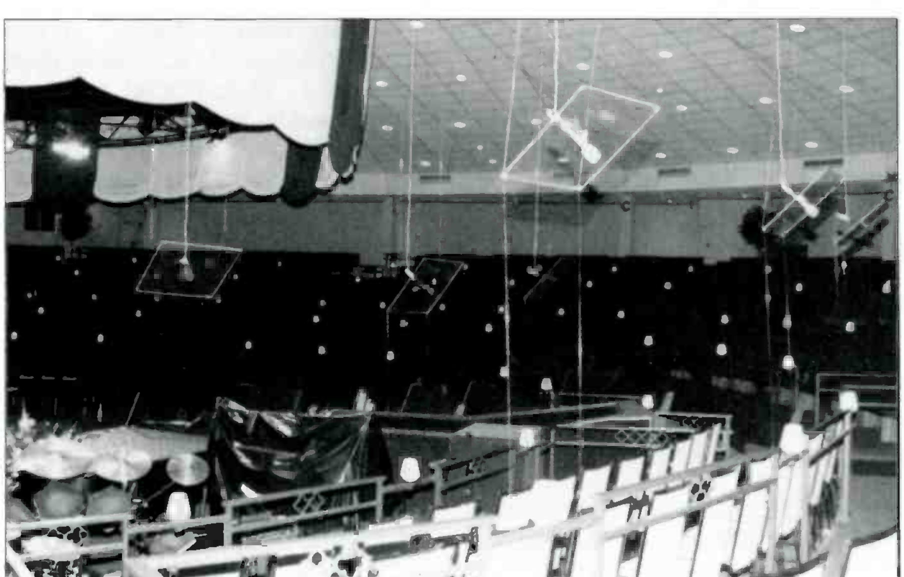
As a stopgap measure, some congregations rely on "plants"—strategically placed people who report to the sound booth via transmitter, but that procedure is never satisfactory. Duguid's first task was making a case for putting the mixer in the house where a single person could become accountable for the mix.

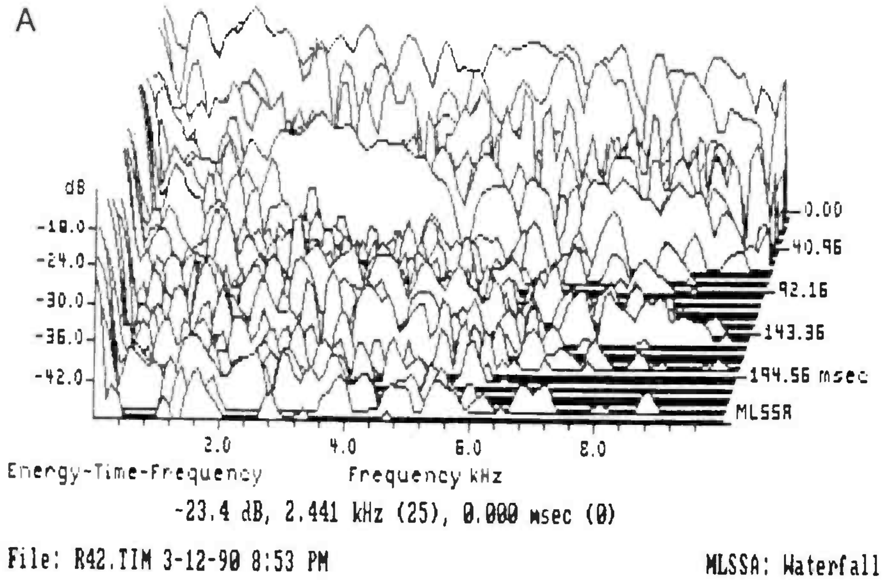
That being agreed upon, the task was to select a suitable console to handle the various sub-mixes and cue mixes necessary for live music mixing. A 40-input TAC Scorpion board was selected. The Scorpion's eight auxiliary sends were an important feature for the church, since they required five sends just for various monitor mixes. This left them a comfortable three aux sends for reverb and effects. To make room for the mixing station, a few of the seats were removed and replaced with a very unobtrusive, and aesthetically pleasing mixing platform with custom-made hardwood cabinets.

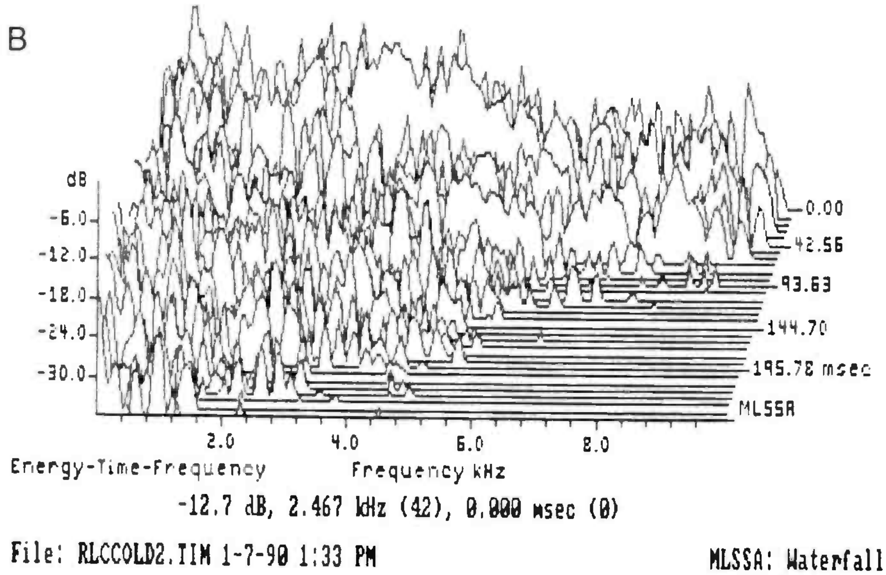
Acoustically, Duguid says, the facility posed very little problem. The hemispherical ceiling and circular (actually octagonal) periphery of the room made coverage a rather simple matter. Speakers were flown along the perimeter of the circular proscenium ring (above the stage area) and directed toward the rear walls in line with the eight "wedges" of the circular space (See floor plan, *Figure 1*). The seating being on progressively elevated platforms, and the relatively short throw (about 55 feet from the speaker ring to the last row of seats) made rear and side fill speakers unnecessary.

Duguid chose to replace the aging Altec speakers with a state-of-the-art processor based system: the Reinkus-Heinz Smart systems. Behind the translucent veil of the proscenium ring hang seven SR-1 (full range) cabinets and four LR2A subwoofers (each with double 18-inch speakers). The eighth wedge, which

Figure 2. The choir section. Note the plastic baffle to prevent feedback from the choir monitor.







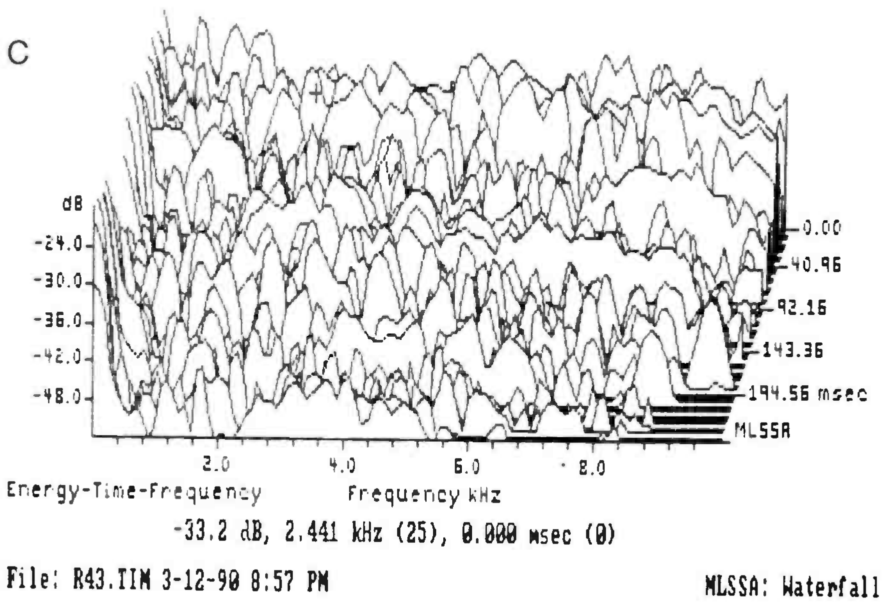


Figure 3. At (A) Block mic position, new system with final EQ. (B) shows Old system before gutting and (C) you see the final EQ with the BBE 822 processor inserted.

faces the choir, was given its own dedicated Smart system (a smaller SR 121) for reasons we will explore shortly. The amplification was punched up with additional Crown Macrotech 1200s, bringing the total system power up to about 15,000 watts.

HOW DOES THE SMART SYSTEM WORK?

According to Duguid, the processor stage of the system is "placed before the amplifiers in the typical position of an electronic crossover—tweaking for particular horns (alignment of delay between components) and system protection as well. It senses the output of the power amplifier. If the operator loses it, or if there is any extremely high signal that would damage the voice coils, it (automatically) pulls that signal back in the protection mode," Duguid said.

It was mentioned earlier that coverage was considered to be good. It was also true that intelligibility was also good due to the short reverb time of the room (0.8 seconds)—since the room was acoustically dead with carpeting and other treatments. The flown speaker cabinets, however, were shown to be inadequate to cover the front rows, so separate horns and drivers (matching those in the Reinkus SR1 cabinet) were mounted, pointing down towards the first four rows.

There were no time-delayed speakers in the house, except for one directly behind the choir section. "The old system had a house speaker pointing directly into the choir section and they always had a problem with gain-before-feedback," Duguid said. "So we put in a choir monitor speaker on a dedicated auxiliary send—for the choir to hear what they needed to hear, without hearing their own mics in that speaker, and then behind the choir (about halfway to the back), we put one delayed house speaker for the people in the audience," he said (Obviously, since that house speaker was emanating sound from a different location than those behind the proscenium ring, a delay was necessary to synchronize the sound from both sources).

One interesting problem that had to be dealt with was an extremely annoying build up of low to low-mid frequencies towards the center of the room. This resonance was maximal

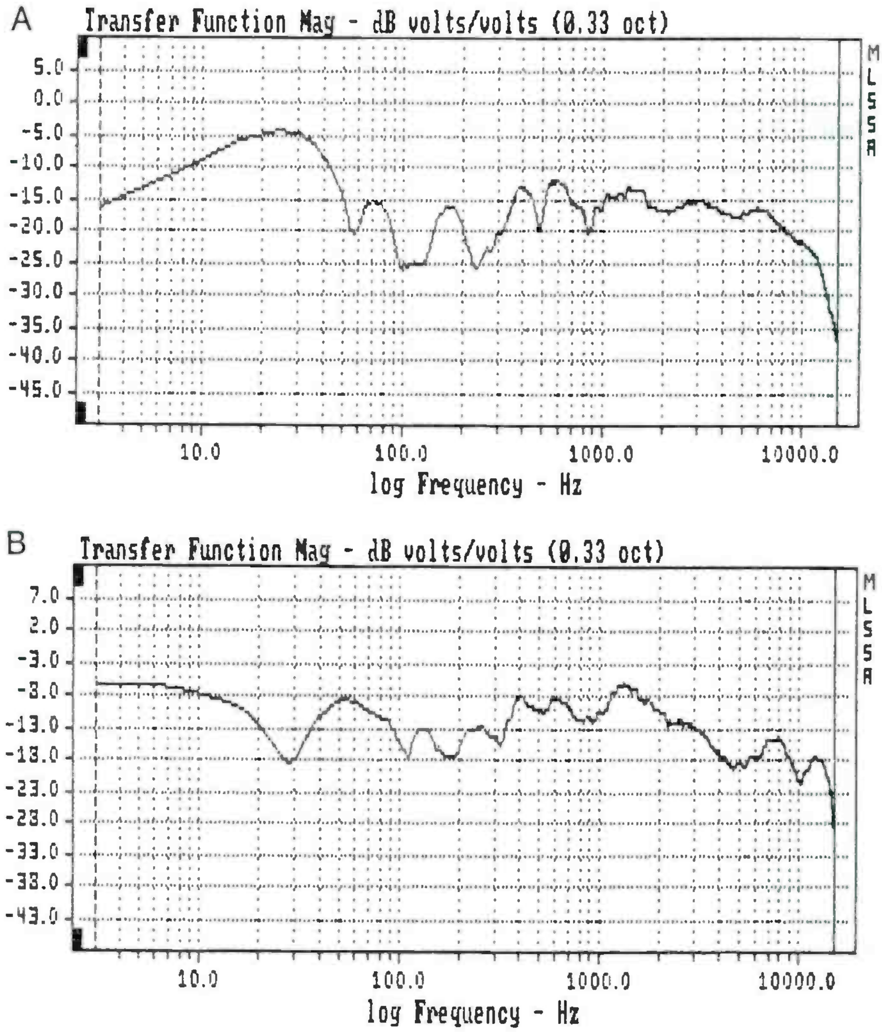


Figure 4. At (A) response at the mix position, while at (B) we see response at row 6, section G.

at dead center (where the pastor placed his pulpit). It seemed that everything from 60—200 Hz was hot, but the chief resonance was exactly at 170 Hz. Why was this so? It seems that symmetrical reflections coming off the rear walls, which are sides of an octagon, act like the collector on a parabolic microphone, focusing the sound towards the single point in the center of the room. Duguid has dealt with this by strapping in a tuned notch filter on the choir bus, and he may end up doing so on the speech bus as well.

BELLS AND WHISTLES

One interesting modification implemented by Ivory Audio is sort of a confidence light built into the wireless microphone system. RLCC uses a number of wireless mics for the sake of convenience (no tripping on wires) and aesthetics (the round stage offers no place to hide cables from view). In fact, virtually everything is on wireless except for the microphones stationed at the musical instruments and the choir microphones. There are at least six wireless mics potentially operational at all times. The dipole receivers are kept close to the stage to get the best reception and the signal forwarded to the mixing console with distribution amplifiers. With a possible six wireless mics up at the same time, the mixer and the speaker/singer need to have confidence that the transmitter/receiver is powered properly and receiving signal. Duguid wired relays into the squelch indicator which turns on LEDs at the mix panel (and also at the microphone) when the system is working properly. This facilitates quicker trouble-shooting, in case of failure.

TAPING FOR BROADCAST

The tech booth does not stay idle, now that the sound mixer resides in the house. Pastors Clinton and Sarah Utterbach have a daily radio ministry entitled, "Listen To Jesus" that airs on 33 stations in North America and is also broadcast into Africa, Russia and the Eastern Bloc nations via a four million watt shortwave transmitter located in Indiana (LESEA Broadcasting). These 15-minute segments are recorded in the tech booth which is divided into a studio and control room. Here, Pfundstein records, edits and prepares duplicates for the various radio audiences they seek to reach.

Pastor Clinton Utterbach also continues his ministry in song by writing and recording his compositions. While the actual recording of his albums may occur in an outside professional recording facility or sometimes in the church utilizing a mobile

Figure 5. Media Manager Ed Pfundstein and Audio Consultant Wally Duguid at the TAC Scorpion board.

Figure 6. Ed Pfundstein in the production studio editing a tape for Pastor Utterbach.

recording service, the most interesting part of the production process occurs during the Sunday worship service. Here, the Pastor frequently departs from the normal songs, sits down at the piano and spontaneously plays and sings an unfamiliar song. This creative process finds its inspiration in a scriptural exhortation from the 98th Psalm which states "Sing to the Lord a new song..."—(a new song being taken as spontaneous musical praise towards God). Many of Pastor Utterbach's "new" songs are very nearly complete compositions, though some are pieces which can be later put together into a finished work. These songs are recorded and catalogued for later use. His most recent album features an entire collection of songs created in this manner.

So as one can see, audio plays an integral role in the quality of worship at RLCC. The system installed by Ivory Audio is sonically all it had promised to be.

A quick glance at the "waterfall" plots in *Figure 3* will give some indication of the overall system improvement, with a remarkable increase in high-end smoothness. In fact, a quick audition indicated the system is very clean, powerful, bright—but not fatiguing, and very consistent in its coverage irrespective of location.

The usage of audio technology at RLCC may well serve as a model for other ministries who seek to propagate the gospel message with excellence and power.

Tech Notes From Wally Duguid

The final system check at RLCC was made using a Tek 2225 oscilloscope, Ivie IE-30A with an Ivie IE2P microphone, DRA Maximum Length Sequence System Analyzer (MLSSA) and a Leader 120-B audio generator. Printouts of the data were made on-site with a small ink-jet printer.

Data acquisitions were made at four locations in each of the eight seating sections, as well as specific locations relating to the choir monitor, delayed rear-fill speaker, mix position and center stage.

The geometry of the room generates a significant low frequency build up which is very pronounced within a three foot circle at dead center. This is the pulpit location. Complicating the matter was a large, undamped pulpit resonating freely at about 200 Hz with the least possible stimulus. One-inch-thick pieces of wood were secured to the underside of the offending surfaces in the pulpit, dramatically reducing the resonance. Several acoustic treatments to the room are being considered at this time.

The Reinkus-Heinz Smart systems have proven to be incredibly smooth in the installations we have used them in. Measurements of the unequalized system response indicated no need for dramatic equalization on the speakers. A Rane SP-15 notch filter originally intended for use on the house mix was removed and installed near the mixing console on the insert point of the choir subgroup. This was adjusted for maximum gain from the choir.

The only other challenge gain-wise was on the pastor's wireless lavalier. The low-end bump previously mentioned was handled sufficiently with the TAC Scorpions' input channel equalizer for this microphone. Consequently, a miniscule amount of EQ was used on the house Rane GE-30, allowing the operators to mix full-range program material without having to compensate for major notches in the house system (See *Figure 4*).

George Augsburger of Perception, Inc. did a review of the system design and calculated the predicted 2 kHz coverage to be plus or minus 2.5 dB, which was proven to an accurate prediction in the final testing.

MLSSA Calculations for distortion were made at 91dB SPL. THD was found to be 0.553 percent. TR60 using the Schroeder method was 0.8 seconds. Speech intelligibility using MLSSA's RASTI was 0.764.

The pastor has been involved in solo and choral gospel music for a number of years, and the level of expectation for the system's performance was high. After several weeks of use, the staff has indicated a great deal of satisfaction with the sonic and functional performance of the system. db

BRENT HARSHBARGER

Audio for the Church

• With the decreasing cost of multi-track machines, it is becoming very affordable for churches to take advantage of multi-track systems. Multi-track in the church makes it possible for choir directors to make custom tracks for the choir, the missionary group to make effective slide presentations and for the soloist to have tracks of currently unavailable songs. Multi-track machines come in all shapes and sizes, from four-track on cassette to the monster digital 48-track recording machines, and ranging in price from several hundred to the thousands of dollars.

Multi-track works through simulsync, making it possible to listen to one track while recording on another. Moreover, multi-track makes it possible for a few people to record the majority of the music on a project. For example, two people with drum machines and samplers can orchestrate and record an entire arrangement.

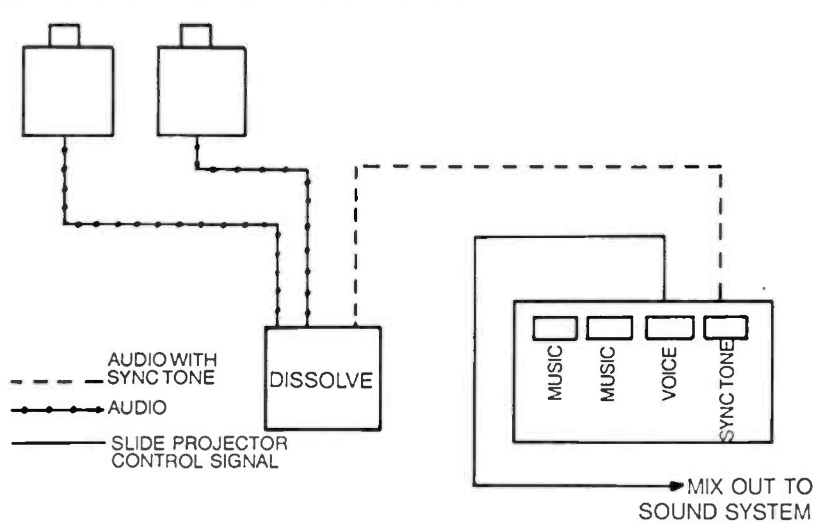
USE AN 8 TRACK MACHINE

An eight-track could help increase the gain of a choir or ensemble by mixing a choir live with a pre-recorded choir. If using an accompaniment track, re-record it to tracks seven and eight. Next, record the choir on the remaining tracks, recording each part separately. When the choir sings, play the tape and mix the parts with two or three choir mics open. Do not put any of the microphones into the choir monitors; run only the pre-recorded choir. This will allow some room to increase the gain before feedback.

One of the most effective ways for a church to use a multi-track is to make multi-media presentations. Motivating people to work in Bible school, or to show the church how missionary funds are being spent are applications that may make positive lasting impressions for everyone.

A minimum of a four-track, two slide projectors and a programmable desolve unit are needed to make simple, but impressive, multi-media presentations. Multi-track has two purposes: music with voice-over, and sync control to lock the slides with the audio. The system would look like *Figure 1.*

Figure 1. Layout of a slide-control system.



WORK UP A THEME

Making a multi-media presentation needs a theme. To make such a presentation, plan which slides will be used, and in what sequence, or write a script and select slides to fit it. Afterwards, record the narration on track one. The music, which should fit the mood of the text, and sound effects will go on two of the remaining tracks. The last track will be what is recorded with the sync tones. Sync tones are also known as cue tones, and are made up of 60hz tones which tell (or cue) the desolve unit to advance to the next slide.

Now that the groundwork is done, the multi-track can be transferred to a stereo cassette or reel to reel. This is purely for convenience. I used to mix it live from the four-track, but you will find as you get more creative, your mixes will be intensive, and by adding outboard effects, will be difficult to mix live.

If a service is broadcast, or a radio program is made around a Sunday service, multi-track can add flexibility and make professional sounding programs. Using the multi-track as an editing unit will help create smooth transitions from the program's music introduction, with a voice- over, and fading into the message of the radio program.

Several churches are installing fully operational multi-track recording studios ranging from four to 24 tracks. One such church is the Salem Church of God in Dayton, Ohio. The 2200-seat church has an in-house 16- track studio. Although the church does not own it, the studio interacts with the sanctuary.

CHURCH OF GOD STUDIO

The studio is owned by a church member who started a recording studio at another location. He opened a studio after going to Nashville and spending almost $3,000 recording a song. After considering the

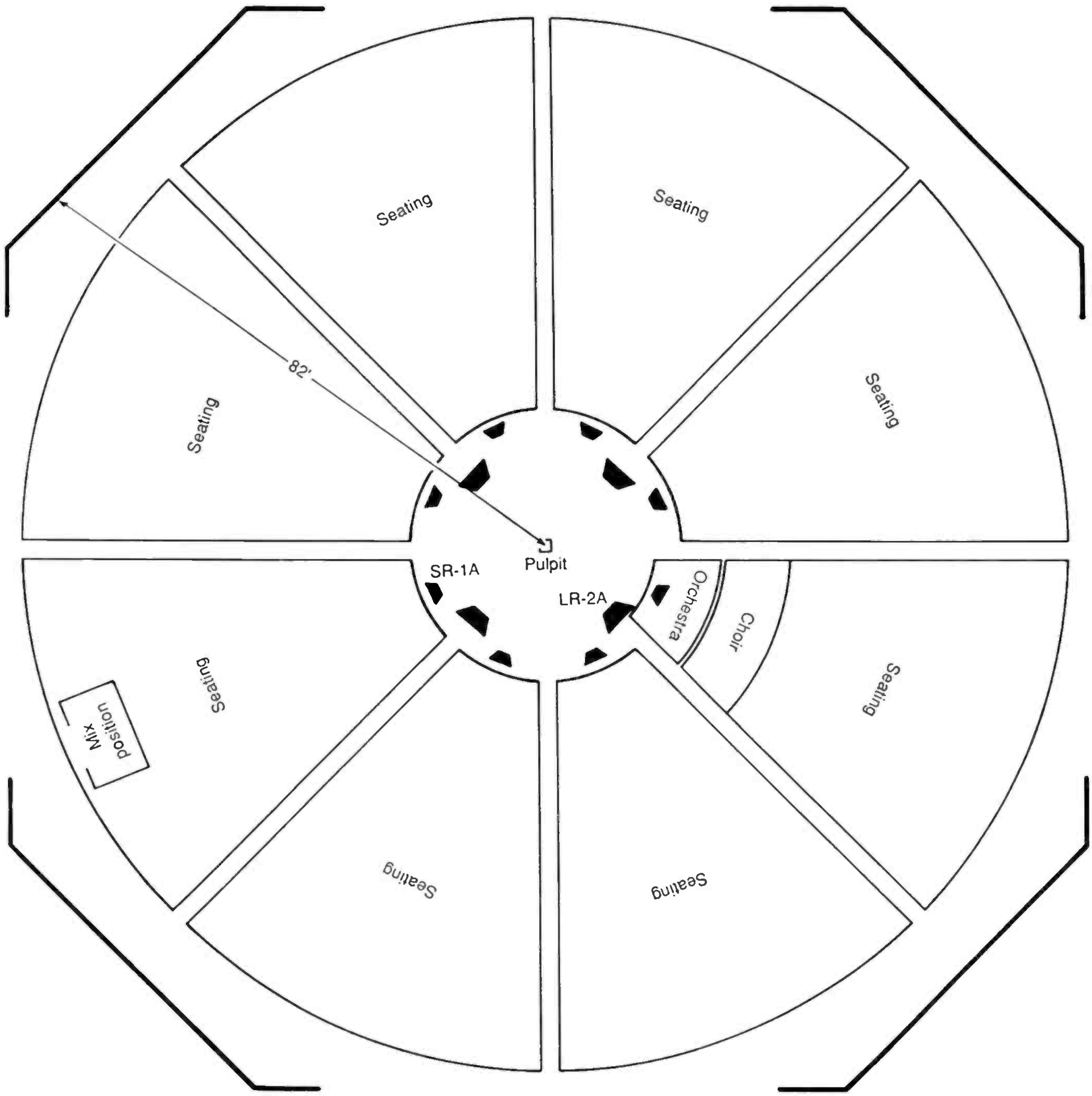


Figure 2. Floor plan of the church.

amount of money he spent to record, and how much the average album project will return in the gospel market, he started Logos Productions Recording Studio.

Meanwhile, the church's music department was growing, the church had a trio and brass ensemble that was gaining popularity in the region and the music minister at the time was interested in having an in-house studio. Therefore, Studio Owner Jim Hazelwood decided to locate his studio at the church, giving the church its in-house studio for recording special projects and its Sunday services. During the week, the studio is rented to artists for reasonable rates.

The church uses the studio for creating accompaniment tracks, special group album projects and producing Celebration Music, a daily radio program highlighting the pastor's Sunday message in a five minute mini-sermon surrounded by contemporary Christian music.

IT BEGAN MODESTLY

The studio started as a modest 16-track facility, consisting of an Allen & Heath System 8 16-channel mixing console, a Fostex 16B multitrack, two keyboards and outboard gear.

As money allowed, the church expanded to a 32-channel TS-12 Soundcraft mixing console and has incorporated the Fostex 16B to lock to a Macintosh and an Emulator III, expanding the track capacity to 32. The studio recently produced two ra-

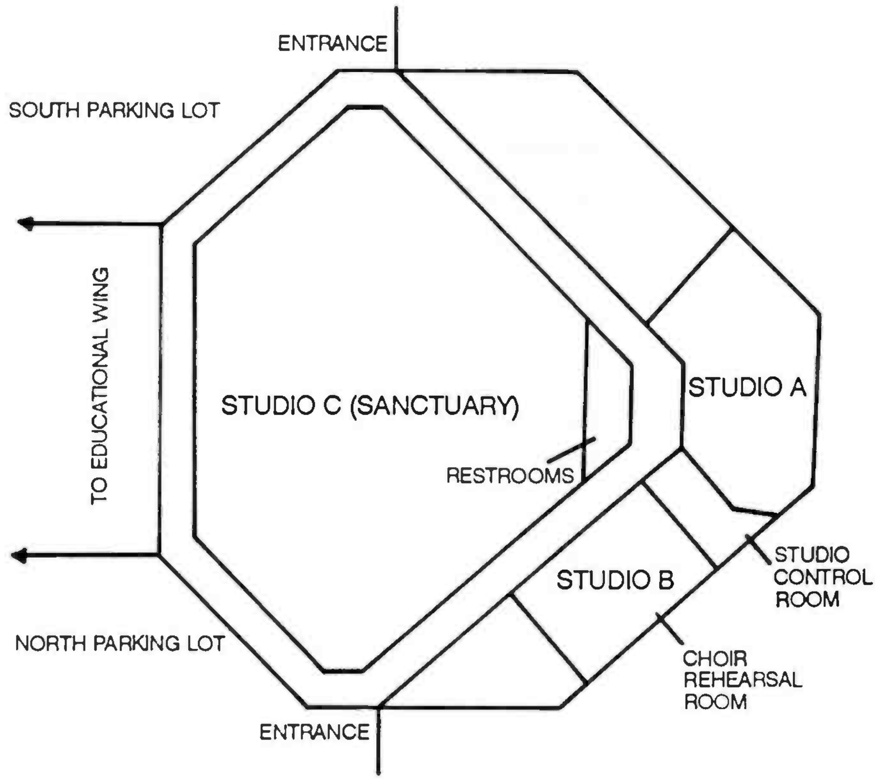


Figure 3. Layout of the recording complex.

Figure 4. It's a traditional-looking recording studio.

dio commercials entirely on the Emulator III, including voice-overs.

The studio is inter-connected to the church, meaning the studio can be used in conjunction with the house sound system or be independent of it. This is done through a patch bay, which makes the studio capable of recording, without interference, what is going on in the sanctuary Sunday morning, Wednesday evening and while the church choir records in the studio.

STUDIO RATES

Logos Productions Recording Studio has hourly rates starting at $35 for basic analog 16-track recording and mixing, including an engineer. The rates are slightly higher for the use of the EIII, producers and sequencer programming. The studio currently uses a Mark of the Unicorn Performer sequencing package for the Macintosh, which can be synced to the Fostex 16B 16-track recorder, using the Fostex 4050 SMPTE to MIDI converter. The Macintosh drives a bank of synths which includes the EIII, Yamaha TX616, Roland D-110, Ensonic Mirage, Yamaha TX802 and Yamaha RX5 drum machine.

The outboard gear and mixing facilities are excellent, including the in-line Soundcraft TS-12, T C Electronics 2290 delay unit, two Yamaha REV 7s, Yamaha SPX90, Eventide H3000, Lexicon PCM 60, Orban 422A, Symetrics quad noise gate, Aphex Aural Exciter, Dbx 160x and a Rockman rackmounted for direct guitar work.

The studio is wired for a 24-track tape machine which is available upon request, but to keep costs down, the Fostex 16B is the main tape machine. Considerable thought was taken to purchasing a 24-track machine, but it was decided that the 24-track machines could not offer a significant gain in sonic quality for the price. Therefore, purchasing the EIII and adding Dolby SR for mixing was a more flexible alternative.

DIGITAL RECORDING TO COME

Logos is looking to a digital future, with current efforts looking towards a DAT machine. The Fostex D-20 and the Panasonic 3500 are being considered for this purchase.

To round up the digital recording, serious thought is going toward getting the Studer Dyaxis hard-disk recording system, which will complement the existing EIII and Mac system.

Although not every church can afford state-of-the-art equipment, or to dedicate a room as a recording studio, multi-track in the church can be a valuable tool. Today, multi-track prices are such that any church can afford even a modest multi-track system.

In the next issue, we will look at what goes into designing a sound system, and how to go about purchasing one for your church. db

Grammy Time

This year's Grammy Awards show was the most complicated television show in which I have ever been involved. There were eleven live performances, four live to tape and some very fast stage movements (we had fifty seconds for the Stevie Wonder set-up).

To give you an idea of what is done to get this show on the road, let me describe the opening act, performed by Billy Joel. Two weeks prior to the show, Joel decided to illustrate his nominated hit *We Didn't Start the Fire*. In order to do this, the production company rented two large video walls. The producers, Walter Miller, Ken Ehrlich, Joel's Sound Man Bryan Ruggles and I had a conference call. It was decided I should record Joel performing this number. To make it easy for the video tape editors, we decided to record Joel and his band on one-track in mono. We would put a click track on track two.

The only available time to make this recording was during a sound check while the band was on tour. Luckily, they were coming to the Rosemont Horizon, a venue in Northwestern Illinois. We went to the Rosemont with two DAT recorders and used the P.A. board as our console. As the band was warming up, we set our levels. Joel came on stage. We let the click track machine run for about 30 seconds (I will explain why later on), and since we only had one chance, we did it right the first time.

Figure 1. Left to right, Stevie Wonder, Anita Baker, and Sting.

Murray Allen is Universal Studios in Chicago.

TRANSFERRING THE DAT

Afterwards, we brought the DAT back to the studio and transferred it to a one-inch video recorder, adding time code locked to house sync. We placed a graphic in the picture path explaining everything we had done, and then rushed it to Hollywood so the video editors could start their animation. Every time Joel mentioned someone's name, their face would appear on the video wall. The editors made the edits right in time with the clicks.

Since this was the opening act, we decided to start running the animated video tape with click track 15 seconds before the show went on the air. This would give the musicians plenty of time to lock into the tempo. We played the click through speakers placed right next to the drummer and synthesizer players. We then added a count-off on the track so the band would know when to start playing.

I met with the drummer and synthesizer players just before rehearsal to explain the process and the count-off. The first rehearsal went perfectly. Every cue hit at the right time.

Everybody was quite relieved with the exception of the Fire Department. We had live fire cued to the end of the song. During the next rehearsal, we lit the fire. It was hot on stage, but no one was hurt.

Our crew consists of eight men in the broadcast truck, including three mixers, a mixing assistant, a sweetener, two maintenance men and a tape playback operator. In house P.A., we have three mixers, an equalization expert, a P.A. assistant and an executive in charge of P.A. equipment.

In the monitor section, we have two mixers. In addition, we have one man dedicated to all the RF microphones, keeping them on frequency and in good working condition. We may use as many as 24 RF mics during a show.

MEETINGS

We had our first tech meeting Feb. 7, 1990 at the Shrine Auditorium in Los Angeles. At this time, we decided where everything was going to go and who was responsible for what. As always, the next day everybody was faxing their requirements to anybody whose path they might cross during the show to make sure nothing would slip between the cracks.

The actual selection of personnel is usually handled way in advance by myself, Greg Sills, the executive in charge of production, and Producer John Cossette. We pull in the best people from L.A., San Francisco, Chicago and New York. There are many mixers who are afraid of doing this show because of its complexity, But our regular crew is thick-skinned and can stand the gaff of four 18-hour days finishing with a live show that ran over three-and-one-half hours.

Our next meeting was Feb. 14 where we actually talked our way through the show, seeing if the stage moves and changes are possible. We came out of the meeting ready to do a show.

That night, Ed Greene, head music mixer, and I went to CBS Radford to record the orchestra cues and prerecords. We had 84 songs and cues to record. Jack Elliot, the musical director, and orchestra ran from 6 p.m. to midnight, and we did not finish. We decided the unrecorded numbers would be picked up at the Shrine during rehearsals or we would perform them live on the show. As it was, we already had a list of numbers we would perform live in any event.

We handled all the music mixing in the Greene/Crowe truck. On Feb. 15, right after the truck was parked at the Shrine, Greene started to set up the mix of the previous evening's prerecords. Arranger Jon Charles came in to act as producer for these mixes, and the mixing was completed late that night.

In addition to the regular mixing console, we added several more mixers and consoles to this truck to handle all music needs. This truck handled the house orchestra (41 musicians), all self-contained groups (eight on this show), all RF microphones and all production mics plus some audience. We used well over 200 microphones on this show. This truck also handled tape playbacks

Figure 2. Bonny Raitt thanks the Grammy committee.

where the act performs to a track, as did Milli Vanilli. Hank Nueberger helped Greene with effects and when there were too many pots to grab, Nueberger helped with some of the mixing. This truck also has visiting producers and mixers helping with the chores.

The signal left this truck and went to the Record Plant truck. Here, Production Mixer Don Worsham handled the overall levels of the show. This includes the music feeds from the Greene/Crowe truck, the cart and video playbacks, the sweetener plus the podium mics and audience mics (we used 24 audience mics this year) and tape playbacks of bumpers and underscores.

CARTS

Paul Sandweiss, the cart mixer, is in charge of carts. This year, he made every cart from a CD. When nominations were announced, he played a cart. When the winner was announced, he punched the winning cart. I think the sound of the carts this year were the best I have ever heard. The presenters also spoke up in full voices, allowing us to play the carts louder. Sandweiss also has his own console for controlling the levels of these carts.

SWEETENING

Carol Pratt is the sweetener for the Grammy Awards. He adds applause and laughter to make up for a lazy audience. He also covers bad transitions, making everything sound smoother. He sits right behind Worsham and directs himself to the sweetening needs.

One of the problems of sweetening is when the audience claps in time to the music, their clapping always sounds late because they are far from the band. If we bring them up in the audience mics, they actually mess up the band's groove. To avoid this, I will usually sit right behind Platt and clap in time with the band. He is very good at staying right with me. We then add just enough real audience down in level to give a live feeling without really messing up the "feel" of the performance. Since the cameras only catch the front row clappers, the slight variations in sync are not noticeable (roughly one-half frame out of sync).

The Record Plant truck is equipped with the Shure Stereo-surround Format. We pull part of the audience towards the rear, giving people with surround decoders the feeling of sitting right in the audience. The effect is really quite dazzling.

Our stage crew is headed up by Jeff Fecteau. He makes sure every snake, every splitter and every mic is in the right place at the right time. He also makes sure that levels on amps do not get changed. We did one show without him, and he was missed. Another key person who has been on the show for many years is Murray Siegel. He is in charge of making sure every artist gets the right mics. Of course, everybody in the crew covers for everybody else. On this show, we had a switch-over from Ray Charles performing with the orchestra and four background singers to Stevie Wonder on a band cart with three background speakers. All of the monitor speakers had to be changed, too. This all happened in about 50 seconds.

It is very exciting to watch the entire stage crew on a big change-over. Stagehands are moving sets; other stagehands are pushing band carts on and off stage; stage managers are steering talent on or off stage; audio is making connections and running the snakes; monitor assistants are positioning new monitor speakers; camera cranes are being moved to new positions and no one gets in anyone's way.

MONITOR MIXING

Our monitor mixing team consists of Mike Abbott and Kevin Wapner. They have been with the Grammy Awards show for two years now, and

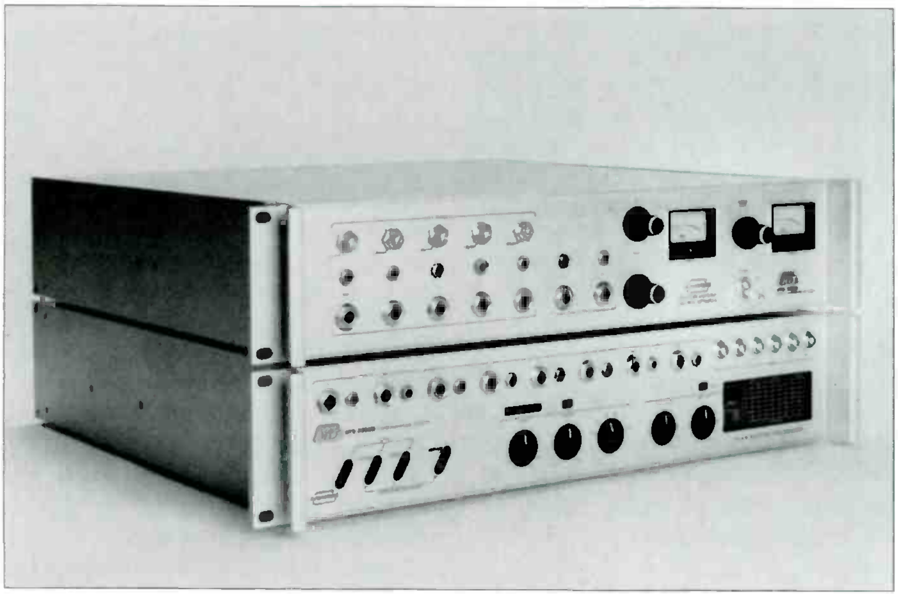


Figure 3. The Shure HTS1000 Encoder atop the HTS200SD decoder.

they have the most realistic and musical approach to monitor mixing of anyone I have ever worked with.

Another important player is Andy Strauber, who comes in from New York to keep all the RF mics going. He has his own little station backstage right across from "patch central," and across from Bruce Burns' nerve center. Burns, the executive in charge of P.A., keeps the receivers and mics in tune and all of the mics working. If you have worked a live show with rental equipment, you must know what a mean job this can be.

House P.A. is handled by Burns Audio. The mixers are Patrick Baltzell on music, Leslie Jones on vocals and solos and Steve Kibbons on production and podiums. Of course, they all overlap their functions from time to time. This year, we used four consoles to handle the P.A. sound. The P.A. gets a split on everything except the house orchestra. Relative to the house orchestra, they get a split on everything except saxes, trombones, trumpets, french horns, violins, violas and cellos. These are premixed in the Greene/Crowe truck and fed on seven separate lines to P.A. In return, P.A. has an emergency feed going back to the Record Plant truck and in case of a failure during the show, the P.A. consoles can handle the broadcast and vice-versa.

ROOM EQ

Ken DeLoria, president of Apogee Speakers, was in charge of room equalization. He placed several microphones throughout the room and constantly analyzed the sound on his Hewlett Packard. Using 53 parametric equalizers, he repeatedly made changes relative to the size of audience, temperature and humidity.

Last year, we designed a test for CBS to run so the affiliates could check their satellite reception. This 20-minute test consisted of tones at various levels, pink noise and music, and it told us the integrity of the frequency response, the headroom and the effect of any processing that might be in the signal path. The test was so successful last year that CBS runs it every week. On the Friday before the show, they ran this weekly test. Everything sounded fine.

At 1 o'clock on show day, we ran our first network audio test. It was less than perfect; in fact, it was downright bad. We let everyone know we needed it to be fixed. By 3 o'clock, we had the feeling that nobody was doing anything to correct the transmissions problems. I called the CBS vice president in charge of transmission to let him know that things had better start happening and happening fast. Within minutes, the lines between Los Angeles and New York were buzzing with activity.

FURTHER ROUTING

The show left our trucks and went to a TELCO truck who microwaved the signal to the CBS Center in Los Angeles. They set it up on the bird to the east coast. From where it was received, it was sent over several microwave links to a major building in Manhattan. From here, it went to CBS on 57th Street. The commercials were inserted in a New York CBS studio. The show was then sent back to this major building, several microwave links and up on the bird to the American public.

At 4 o'clock, we had access to the satellite. Between commercials being transferred, we ran test tones. Little by little, we were able to get compressors out of the circuit and make other changes. By 4:45 p.m., we had a clean signal. We backed our send down four dB to give us head room. Signal to noise is not a problem at our end. We went on the air at 5 o'clock.

I have a listening room where I can hear the sends, returns and anything relative to our signal. When the show went on the air, I switched around the system and everything sounded fine. I continued this process every 30 minutes. The show was broadcast in a first-class manner.

Two days later, CBS called me to see how I liked the transmission. I told them it worked, but we should try to have it set up a little earlier than 15 minutes before showtime. Outside of these problems and 10 more live bands, the show went its usual way.

SUMMARY

I would like to thank Pierre Cossette, the executive producer, and John Cossette, who paid for all of the high tech audio. I would also like to thank the National Academy of Recording Arts and Sciences for their support, especially Michael Greene (president). And lastly, I would like to thank CBS for giving us more cooperation and indeed showing a true spirit in their attempt to improve audio for video. It isn't many networks that will give you a clear path to all of their high-level executive tech people and the muscle to back it up.

Next year, we will probably have an even bigger show with a dozen new challenges. We will find a way to solve every problem. By the way, the Grammy Awards show took the entire night ratings-wise. There *is* hope for good music that is well-performed, well-mixed and well-broadcast.

Mixing Techniques for Stereosurround

ROBERT B. SCHULEIN

This is an excerpt from a technical paper by the author that was delivered at the Spring 1990 NAB Engineering Conference in Atlanta, Georgia. He is General Manager of Shure HTS, Evanston, IL.

• Developing a mix using the Stereosurround production format is very similar to two-speaker stereo and single-speaker monaural mixing, once microphones have been selected and placed, signal processing has been set and the configuration of the mixing board and source assignments have been made. In a typical setup, the input buses and outputs of the encoder are connected to the mixer patch bay, such that each input fader can be assigned to or panned between any two input buses. With this condition established, the signal controlled by each fader can then be panned in the "mixing space" to the desired position as the mix is being monitored. Based upon Stereosurround format productions to date, a number of mixing techniques and observations have emerged:

1. Due to the fact that 4-2-4 matrix systems are not discrete, and that localization problems can occur, start by mixing the most dominant sound elements first.

2. When using compression or limiting on dialogue or a vocalist, avoid processing any additional ambient signals through the same compressor/limiter, so as to avoid the dominant sound causing the ambient sounds to pump up and down. This result is particularly bothersome with a multichannel reproduction system.

3. Keep in mind that it is possible to continuously pan signals between the front and surround locations, and that it is not necessary to pan a program element completely to the surround position in order to get a noticeable rearward or surround sense to the mix.

4. When mixing music, start by mixing sound elements directly to the L, C and R positions and effects to the interior position using separate effects devices for each element. Once the mix has been roughed in, further adjustments should then be considered.

5. If problems occur regarding localization interactions between two or more sound elements, consider panning one of the signals more toward the interior position, altering its timing, or reducing its amplitude.

6. When using multiple microphones in situations where two or more microphones may pick up the same signal, confirm the polarity of each microphone between its acoustic input and the mixing console output to the encoder. This will greatly reduce the possibility of undesired out-of-polarity information being mixed into the interior or surround position.

7. In many situations where a center oriented vocal element is competing with loud ambience element (sports announcer versus crowd), it is desirable to "widen" the vocal element. This may be accomplished by either panning the element towards the surround position or by using a small amount of stereo synthesis re-

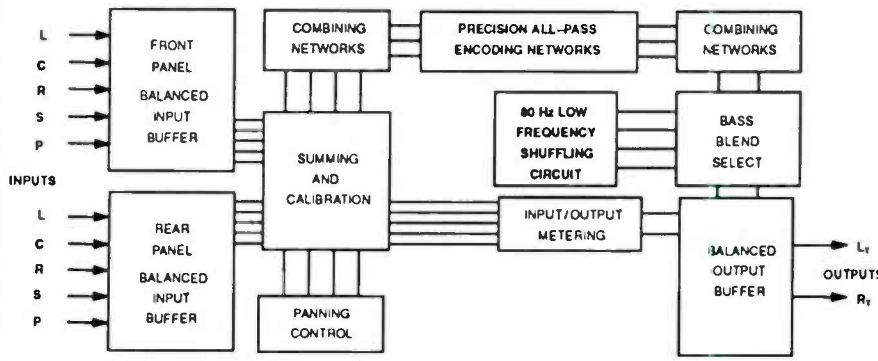


Figure 4. Stereosurround encoder system signal processing.

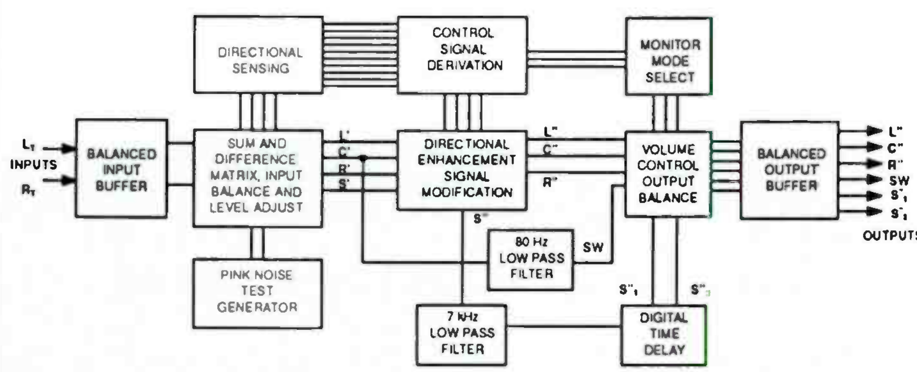
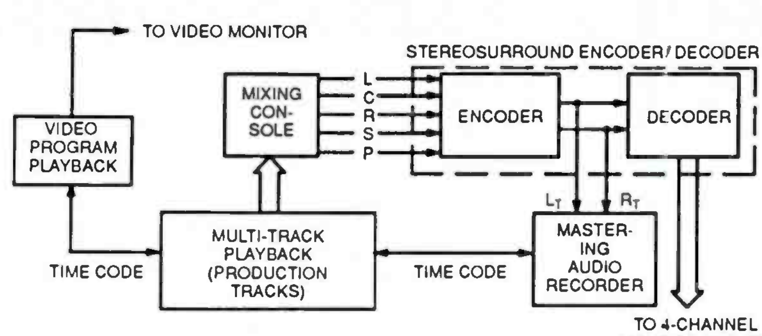


Figure 5. Stereosurround decoder system signal processing.

Figure 6. A block diagram of the Stereosurround system for post-production.



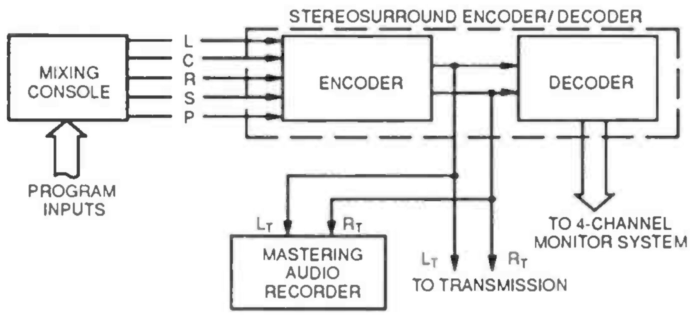


Figure 7. Block diagram for live or live-to-tape production Stereosurround.

turned to the left and right front positions. Similarly, monaural effects associated with sports productions such as baseball "bat cracks" and football player contact sounds can be given more dominance in the final mix.

8. M-S and X-Y stereo microphone techniques can be integrated into Stereosurround format productions provided proper selection of microphone patterns are made. As an example, a popular M-S microphone configuration using a cardioid mid pattern and bidirectional side pattern allows the creation of equivalent X-Y combinations that place sounds properly in the front and rear sound stage.

Realistic crowd or ambience mixes can be created by using multiple groups of microphones placed so as to sample distinctly different portions of the production environment.

Once placed, the outputs of these microphones are then positioned at different points in the "mixing" space by panning between the left or right and surround encoder input buses. db

LEN FELDMAN

Lab Report

Soundcraftsmen Model 900X2 Power Amplifier

GENERAL INFORMATION

• The Soundcraftsmen 900X2, a very powerful amplifier, boasts in excess of 675 watts per channel when operated into 4-ohm loads in its two channel mode, and nearly 1400 watts when operated in its bridged mode into an 8-ohm load. While most amplifiers of such high power rating are designed to work best into 8-ohm loads, the Soundcraftsmen 900X2 was designed to be most efficient when operating into 4-ohm loads. As a result, it can operate with complete safety even when speaker impedances fall well below 4 ohms. Another interesting fact about the Soundcraftsmen 900X2 is its ability to deliver its full-rated power even if line voltages fall below the "standard" 120 volt value.

Output stages of this amplifier employ MOSFET devices, as are other Soundcraftsmen amplifiers we have tested in the past. Among some of the more obvious features of the amplifier are its balanced (XLR) and unbalanced (¼-inch phono plug) input facilities and its recesses stepped input level controls that provide level matching from minus infinity (totally off) to +6 dB, where 0 dB is defined as a voltage gain of 20. In addition to the XLR and phone-jack connectors, there is a 5-terminal barrier strip for making input connections.

A rear-panel compressor/limiter switch protects connected speakers for those applications where the amplifier is likely to be pushed to its limits. Monophonic bridging can be done without having to go inside the amplifier, thanks to a rear panel mono-bridging switch. According to Soundcraftsmen, connected speakers are completely protected from turn-on surges by a seven-step delayed turn-on circuit. The amplifier also features a front-panel mounted 20-ampere circuit breaker and true clipping indicators for each channel. Speakers are protected from possible d.c. currents by dual output relays.

Soundcraftsmen's protection circuitry is all solid state and will cut off the amplifier's output when overheating exists. During our tests, there were periods of time when the amplifier was required to deliver steady state power levels at or near rated power. Under those conditions, we heard a built-in fan go on, cooling the amplifier by means of forced additional circulation of air. During most of our tests, however, the amplifier ran without activating the fan.

One thing I have always liked about Soundcraftsmen, and the way they present their power amplifiers, is that they include a "Certificate of Performance" with each amplifier. This document lists actual test data performed by one of their technical inspectors who signs the certificate. In all the years I have been testing Soundcraftsmen amplifiers, I have never known a sample to deviate from those measured results by more than the tolerance of my lab test equipment. More importantly, published specifications are usually exceeded by the technician's measurements at Soundcraftsmen and by my own confirming lab measurements.

Figure 1. Frequency response.

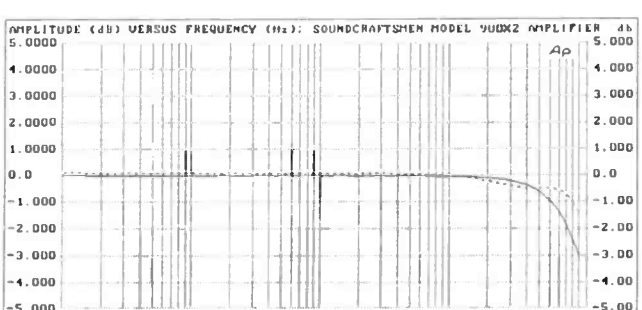
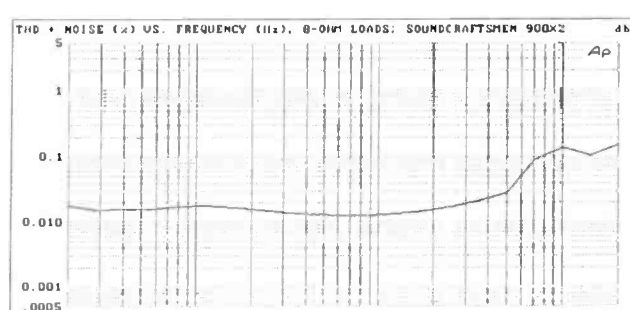


Figure 2(A). Harmonic distortion plus noise versus frequency, rated output (375 W/channel,4-Ω loads).



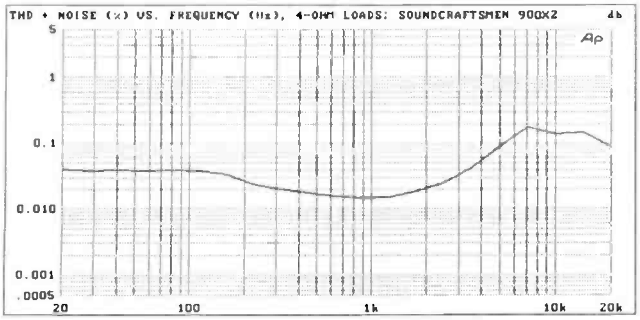


Figure 2 (B). Harmonic distortion plus noise versus frequency, rated output (675 W/channel, 8-Ω loads).

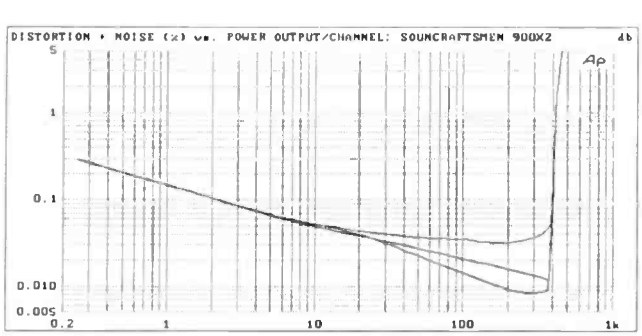
CONTROL LAYOUT

As mentioned earlier, the only control on the front panel of this amplifier is the on/off power switch, which is also a circuit breaker. Clipping indicator lights are positioned at the lower right of the panel. At the rear of the amplifier are the XLR balanced input connectors, the unbalanced phone-jack inputs and the 5-terminal barrier strip configured for balanced inputs. To either side of the barrier strip is a recessed input level control, while below the strip, there is a mono/stereo bridging switch. Still lower down on the rear panel are color coded five-way binding posts for speaker connection. When operating the amplifier in the bridged mode, connection must be made to the two "hot" binding posts. In the bridged mono mode, an 8-ohm speaker across the two "hot" terminals presents an impedance of only 4 ohms to each driving channel. While a 4-ohm speaker could be connected in this mode, it would appear as a 2-ohm impedance to each of the driving channels, and this is the absolute minimum load the amp channels can handle. Since some 4-ohm speakers actually dip in impedance at certain audio frequencies, we would not recommend using a speaker whose nominal impedance is only 4-ohms in the bridged mono mode.

LABORATORY MEASUREMENTS

Frequency response of the Soundcraftsmen 900X2 is shown in the *Figure 1* graph. The left channel was down –0.1 dB at 20 kHz while the right channel was off by –0.2 dB at that extreme frequency. The difference is not worth mentioning, and if you want to argue that the

Figure 3 (A). Harmonic distortion plus noise versus power output per channel (8-Ω loads). Best curve is 20 Hz, next best is 1 kHz, poorest curve is 20 kHz.



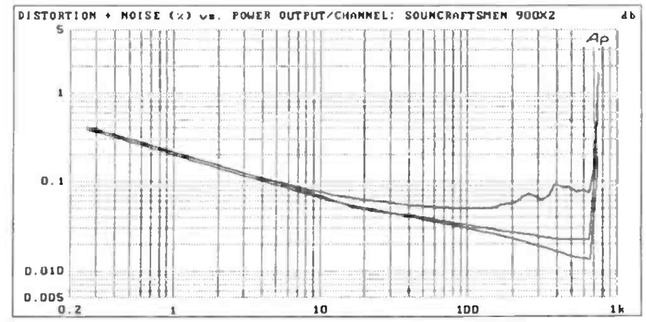


Figure 3 (B). Harmonic distortion plus noise versus power output per channel (4-Ω loads). Best curve is 20 Hz, next best is 1 kHz, poorest curve is 20 kHz.

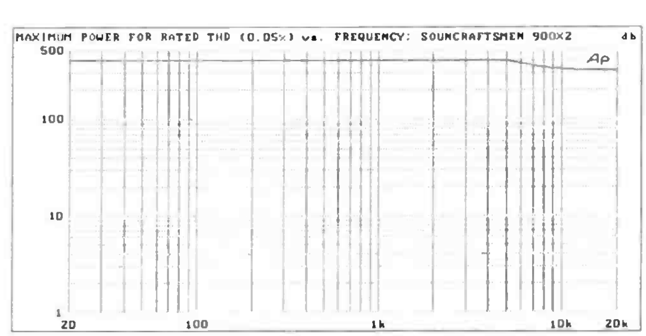
right channel is "out of spec," bear in mind the spec reads ±0.1 dB, and since there was no + deviation, Soundcraftsmen is entitled to –0.2 dB.

Figure 2(A) is a plot of distortion versus frequency at the rated output level of 375 watts per channel, using 8-ohm loads with both channels driven. At mid-frequencies, THD plus noise was well under 0.02 percent, as opposed to the 0.05 percent claimed. *Figure 2(B)* is a similar plot, but this time, the load resistors were 4 ohms. We should note that while the published specification at this impedance notes a rated power of 675 watts per channel for a test frequency of 1 kHz, the specs are a little more conservative when it comes to power output at the frequency extremes, with Soundcraftsmen claiming only 600 watts per channel over the entire frequency range from 20 Hz to 20 kHz. The amplifier easily met those specifications.

Figures 3(A) and *(B)* offer another way of looking at the power output capabilities of this amplifier. In each of these graphs, you can see plots of THD plus noise versus power output levels for 1 kHz, 20 Hz and 20 kHz. As you can see, the amplifier had no problem whatsoever delivering more than its rated power even at 20 Hz, where most amplifiers have problems because of inadequate power supplies.

Figures 4(A) and *(B)* are power bandwidth plots. That is, they show how much power can be delivered by the amplifier at all frequencies from 20 Hz to 20 kHz for the rated distortion of 0.05 percent. With 8-ohm loads, the amplifier delivered 400 watts per channel at all but the highest treble frequencies where power output for that

Figure 4 (A). Maximum power output per channel for rated THD of 0.05%, 8-Ω loads.



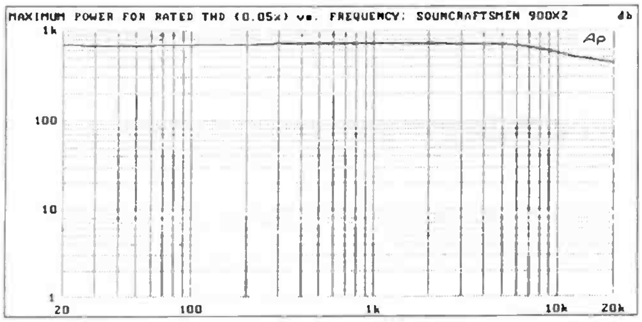


Figure 4 (B). Maximum power output per channel for rated THD of 0.05%, 4-Ω loads.

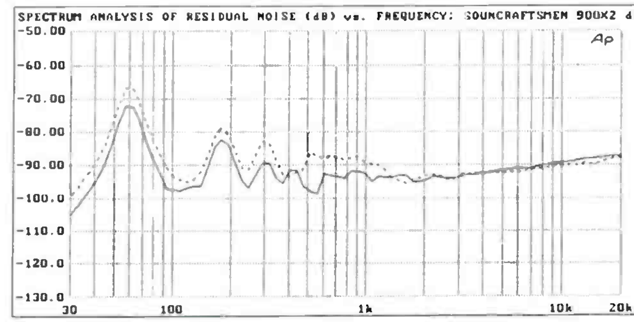


Figure 5. Spectrum analysis of residual noise at 1-watt output reference level.

level of THD dropped to about 320 watts per channel. In the case of 4-ohm loads, output reached in excess of 700 watts per channel at low and mid frequencies.

SMPTE-IM distortion measured approximately 0.07 percent at rated output. Signal-to-noise ratio was measured two ways. With 500 millivolts applied to the inputs, and the level controls adjusted to deliver one watt per channel into 8-ohm loads, the input signals were removed and the residual noise was measured. It was 78 dB below the reference one watt output level. Evidently, Soundcraftsmen's engineers measure S/N differently, preferring to set the level control full open and then measure S/N relative to rated output. Under those conditions, the A-weighted signal-to-noise ratio measured 106 dB, or just a bit better than claimed by the manufacturer. *Figure 5* is a spectral analysis plot of the residual noise as a function of frequency, again referred to one watt output levels. Notice the chief contributing components to the noise were not noise components at all, but the low-level fundamental of the power supply frequency (60 Hz) and its harmonics at 180 Hz, 300 Hz and so on. In any case, these components were at such a low level as to be virtually inaudible under actual program listening conditions.

CONCLUSIONS

The Soundcraftsmen 900X2 is about as rugged a power amplifier as we have ever tested. To be sure, it is not lightweight, and you are not likely to want to cart it around for field work or remote sound reinforcement jobs. But if you install one or more of these powerful amplifiers in a permanent setup, it is not likely you will have to worry about it standing up month after month and year after year, regardless of how long it is powered up and putting out. To most of us, that matters more than the minute differences in sound quality that one amplifier displays against another. Still, having said that, extensive listening to the Soundcraftsmen revealed that this amplifier would be as at home in the most discriminating audiophile's listening room as it would in a demanding sound reinforcement environment. db

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Power Output/Ch. at 1 kHz		
4 ohms	675 watts	700 watts
8 ohms	375 watts	400 watts
2 ohms	900 watts	N/A
8 ohms mono (bridged)	1350 watts	Confirmed
Rated THD	0.05%	0.03% (see text)
Frequency Response	20 Hz to 20 kHz, ±0.1 dB	(See text)
S/N (re: 1 Watt)	N/A	–78 dB
S/N (re: rated output)	–105 dB	–106 dB
Slew Rate	50 V/μsec	Confirmed
Input Sensitivity,Rated Output	1.22 V	1.20 V
Input Impedance		
Balanced	22 k ohms	Confirmed
Unbalanced	32 k ohms	Confirmed
Power Requirements	100-125 V, 50/60 Hz	
Weight	59 lbs.	Confirmed
Dimensions (WxHxD, inches)	19x5.25x16.5	
Price: $1,599.00		

ROBYN GATELY

The Sound Contracting Engineer

• My last article on monitors (***db***, *Sept./Oct. 1988*) focused on the premise that most monitor systems are built with only one mix available. Now let's pretend we have moved up to the position of operating an on-stage monitor system with multiple mixes.

In that article, we covered some basic concepts for proper monitor operation—the 3:1 rule, placing monitors 15 degrees off axis (not right in front of the performer), driver alignment, multiple mics and monitors on a mix and the most important concept, psycho-acoustics. If you are not familiar with all of these points, please review that article. If you are familiar, then let's move to the next level.

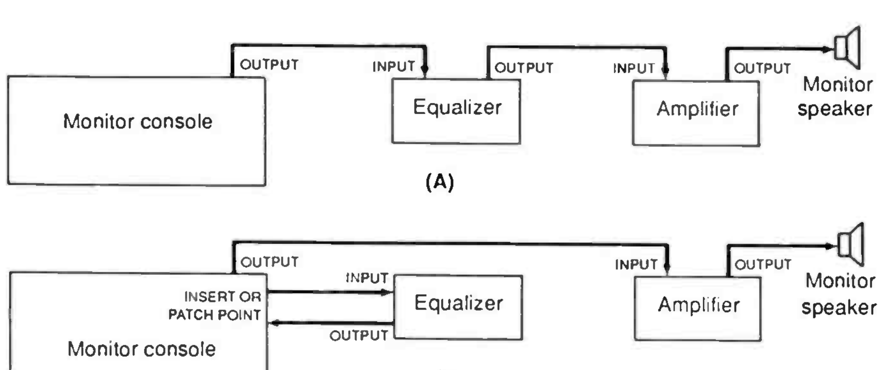
The point of multiple monitor mixes on-stage is two-fold. First, it allows you to (hopefully) give every performer their own listening environment custom-tailored to their individual needs. For example, suppose the bass player of Stage Left can hear himself fine, but cannot hear enough of the guitarist on Stage Right, or enough of the drums. Meanwhile, the drummer may be drowning in guitar volume, but cannot hear the bass or his own Kick and Snare to properly keep the rhythm section together. Stage Right, the guitarist may need to hear the snare drum to keep the beat, but feels he needs to hear his guitar as loud at the front of the stage as he does in front of his amps. Three different people, three different monitor requests. Now imagine if we added a lead vocalist, background singers, keyboardist, second guitarist and horn section.

Secondly, you are now close enough to the "action" that the performers may communicate their requests to you instantly. This may be in the form of shouts, hand signals or thrown beer bottles, but the performers will either get your attention or sulk all night.

THE CORRECT TOOLS

It is important you have the tools necessary to do a good job. Not only do you need a mixing console, but you need equalizers (1/3 octave, preferably) that are inserted into the board on the patch points. Inserting EQs here will allow you to hear the mixes as the performer hears them (see *Figure 1*).

Figure 1. It once was common practice to daisy-chain equipment together, one after the next as in (A). Today, most manufacturers provide inserts or patch points that permit the signal to leave the console, go through the EQ,and return to the console before the master fader, as in (B).



Almost all modern boards have solo or "cue" buttons allowing you to listen to individual channels or entire mixes at the touch of a finger. If you plug the output of your board into your EQ, and from your EQ into your amplifier, when you solo the lead vocalist's mix, you will hear what you have turned up in his monitor, but you will not hear the changes you have created with the equalizer. If you have drastically changed the EQ on the 1/3 octave, he may hear a muddled mess while you think things are fine.

Notice I have referred to you hearing what is going on. I do not mean running out on stage to listen. The output of your boards' solo (cue) should feed an amplifier which is connected to a monitor right next to your head. This monitor should be the same make as the monitors on-stage, so you hear exactly what is heard on-stage.

GETTING ORGANIZED

Multiple mixes on a stage require good organizational concepts that allow you to work quickly and efficiently. Just as it is important you have the right tools for the job, it is important you understand your equipment and the mind of the performer you are working with.

Assuming you have a 24 x 8 monitor board (that's 24 inputs leading to eight outputs), then conceptually what you have is eight little P.A.s, each with a listening audience of only one or two people. The first row (from left to right) of "Mix" knobs

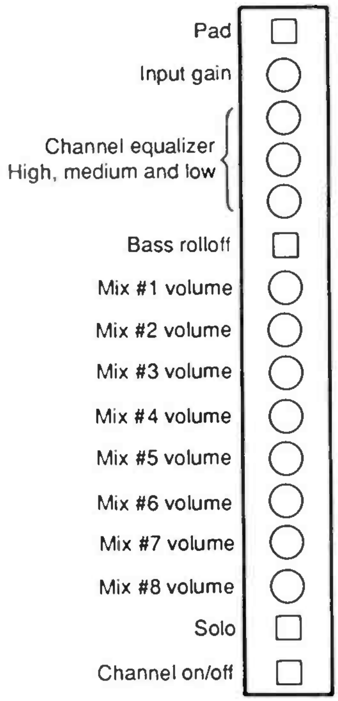


Figure 2. A typical monitor module.

adjusts the amount of each of the 24 inputs to the performer listening to the first "P.A.," the second row to the second performer, and so on (see *Figure 2*).

The more you use eight outputs, the sooner you will develop patterns that break down into some variation of the following:

Mix 1: Downstage Right floor monitors

Mix 2: Center Stage floor monitors

Mix 3: Downstage Left floor monitors

Mix 4: Upstage Right floor monitors

Mix 5: Upstage Left floor monitors

Mix 6: Auxiliary floor monitors (horns, etc.)

Mix 7: Drum monitor

Mix 8: Sidefill monitors

Mixes one, two and three cover all vocalists along the front of the stage, Mixes four and five cover the guitarists, bassist and or keyboards farther back on the stage, Mix six covers anything left over, Mix seven is dedicated to the drummer and generally requires a heavier duty monitor speaker, while Mix eight is for the "Sidefills," a P.A. system on each side of the stage turned in on the performers (see *Figure 3*).

DIFFERENT MONITORS

At Modular Sound, in Morrisville, PA., we use two different types of monitors, either a 12-inch speaker with a one-inch horn, or a 15-inch and horn. The 12-inch version is used for situations where the mix in the monitor will be mainly vocals, while the 15-inch is used when we anticipate a need to add some kick drum, bass guitar and so on.

We have developed the concept that on small to medium-sized stages, we will utilize the floor monitors to carry that person's vocals and other important vocals, while the bulk of the "Booming Kick Drum" and the rest of the band will be carried by the sidefill monitors. Generally, our smallest sidefill monitor is two 18-inch, two 12-inch and two horns per side with a total of at least 3,500 watts!

On larger stages (minimum 40 x 20 feet), we need the 15-inch monitors because individual performers are in such diverse environments that they need to be able to totally adjust their low end requirements.

Unless you are doing a "lightweight act" (Jazz, Folk, etc.), the drum monitor needs to be a minimum of a 15-inch monitor. We have been known to give drummers four 15-inch speakers with two two-inch horns. In fact, for the band *Cinderella*, we recently gave the drummer four 18-inch speakers, four 12-inch speakers and four horns with over 4,000 watts!

It's not a bad idea to set up all of the spot mixes in front of your console and go through them before laying them out on-stage. This allows you to be sure all of the mixes are working properly before setting up the stage. It can be a major problem to change a monitor or speaker cable after the stage is set up. Borrowing the CD player from the house can be more helpful and efficient than just speaking into a mic.

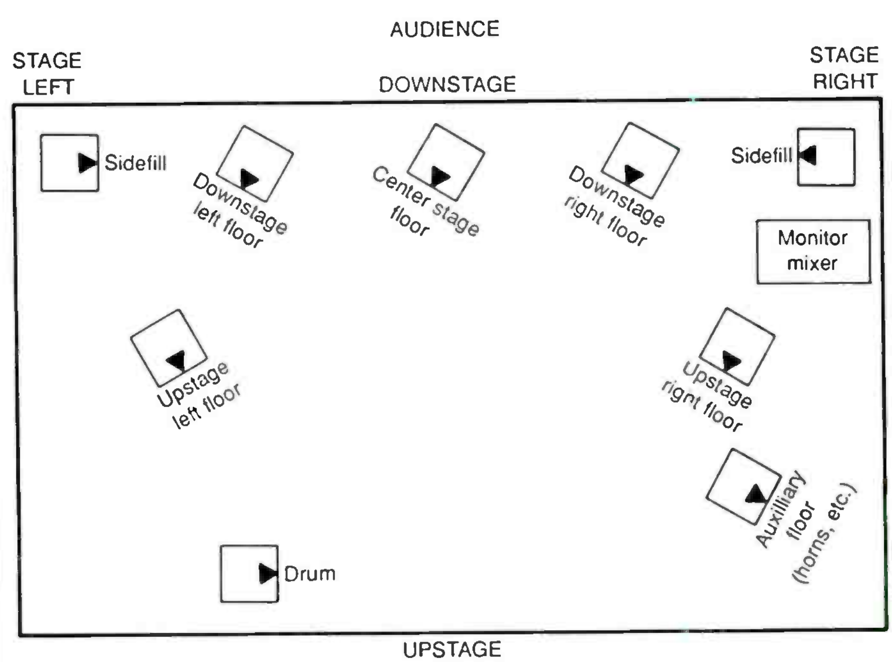
LABEL YOUR BOARD!

Make sure all outputs are labeled (on Duct, Masking, White tape, whatever) according to the person's name or position. Next to the inputs on the left side of the board, place a label next to the mixes (from Mix one down to Mix eight) that tells what those outputs are. It is often helpful to repeat this in the middle and on the right side inputs of the board. Lastly, directly underneath each input, label what that input is: Vocal #4, Flute and so on.

TALKBACK

Most large monitor systems use a *talkback* mic from the House console. This is a mic that has been plugged into a mic (or *return*) cable at the console in the audience and

Figure 3. The stage-monitor layout.



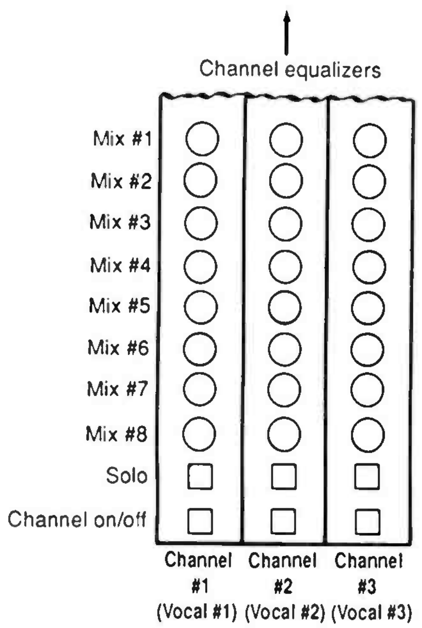


Figure 4. This is the monitor control setup used.

terminates at your console. We usually plug it into the second from the last input on the console (#23). When this mic is turned up in the sidefill and drum mixes, then the House Soundman is able to talk to the band without shouting across the arena (but don't forget to turn this off before the show).

The last input on the console (#24) is generally reserved for the monitor man's talkback mic, so he may communicate with the Band during setup and sound check.

If everything works (and the light man isn't going to drag his ladder across the stage again), you are now ready to get some basic sounds, but you're still not ready for the band.

The following sequence should not take more than five minutes. First (we assume you have labeled your board), your inputs should have their gains set in the approximate position for each input. If you can do a line check with each instrument, so much the better. We don't always have that luxury because we specialize in Festivals.

Now, with your EQs still flat, unmute (turn on, but don't turn up) all your mixes and inputs. Setting your master faders requires some experience with your board. On most boards, you are looking to set your faders at *unity gain*—whatever voltage comes in is what goes out. I have found that most boards want to be turned up about 80 percent the length of the fader. Sometimes unity gain is labeled –10dB while the label for 0 dB is all the way at the top. Other times, 0 dB is located 20 percent down. Read your instruction manual, ask questions and wonder why manufacturer's haven't standardized this.

Now, if everything works, when you turn up Vocal #1 (Input #1z0), it should come out of the speaker labeled Mix #1. If you have three vocalists across the stage and three monitor mixes (remember the mix assignments above), then Vocal #1 wants mainly to be in Mix #1, Vocal #2 mainly in Mix #2 and #3 in #3.

The problem is people want to hear lots of themselves, and a bit of everyone else. The solution we use follows: if Mix #1 with Vocal #1 ends up with its input, send at 2 o'clock, then we will add Vocals #2 and #3 to mix #1 at 12 o'clock (see *Figure 4*). Vocalist #1 will hear some of the other two vocalists, but mainly himself. Lastly, add a bit of all the vocals to the sidefill monitors.

Now, go to the other spot mixes around the stage and make sure you have the mic for that player or singer in that monitor. Then make sure you can hear the kick and snare drums in the drum monitor and add these instruments to the sidefill monitors.

A SECOND PERSON

It can be helpful to have a second person at the monitor console who keeps their hand on the correct knob while you listen to your monitors.

I've left EQing monitors out until now because it is extremely easy to over-EQ. At Modular Sound, we approach this question carefully. Too often, we see "engineers" who end up with either all of the equalizer faders near the bottom of the EQ, or we see the famous "U-EQ" (the EQ faders form a large U). Unfortunately, both EQ settings lead to "engineer burnout" trying to make a system sound good, and often make great systems resemble trash.

For best results, observe these EQ rules:

- It is best to never use more than ⅓ of the faders on an equalizer;
- Don't use all ⅓ faders before the band gets on stage. Leave room for the unexpected;
- Whenever possible, try to avoid turning up EQ faders; try to only pull out what you don't want;
- It is sometimes helpful to pull out the "prescence peak" built into many mics. To avoid dealing with feedback caused by these built-in peaks, we "Pre-EQ" our monitor system. If we know all of our vocal mics are going to be Shure SM-57s, then we will notch the EQ 2-3 dB at 630, 1 k, 1.25 k and 5 k Hz before turning the system on. We only do this because we have so much experience working with certain mics on our system, and know in advance how to speed up our job;
- Don't be afraid to use the EQ knobs on your console inputs to correct "large errors" (Bassy vocal, thin sounding mic, etc.);
- Break these rules only when absolutely necessary! But remember, certain situations require you to break rules. For instance, the "punch" of a Kick Drum is from 80-125 Hz, so it may sometimes become necessary to turn up these EQ faders in your drum or sidefill monitors.

For the simplest EQing possible, use the best equipment you can and set it up properly.

THE RIGHT QUICK FIX

Now that you know the rules, stand at Vocal #1 and speak into the mic. If it is on and sounds good, then turn it up a bit. If it doesn't sound good, fix the EQ on the input before touching the ⅓ octave EQ. When EQing your monitors, if you have a good ear for identifying frequencies, then it can be much quicker for you to stand at the mic and tell someone else what frequency to change.

Once the mic sounds good, turn it up until it starts to feedback a little bit. To do this, you must stand at the singer's position. EQing your monitors without a face in front of it does *not* give you real-world conditions. Turn up the mic until you start to hear your own voice talking, and then slowly make it louder.

When you start to hear the slightest feedback, turn down that frequency about two dB. Turning it down 6 or 8 dB doesn't fix anything, but it does remove quite a bit of the sound. Now, turn the system up slowly until you hear the next feedback. Turn down that frequency, even if it is still the same frequency as the first time.

Repeat the process until you've done it four to five times. We generally figure that if you've touched five EQ faders on a ⅓ octave EQ, then you should turn it down a little bit, listen to it and make sure it sounds good and then leave the other four EQ faders for when the band is on-stage.

For the simplest EQing possible, use the best equipment you can and set it up properly. This means use the best mics, EQ and speakers you can afford. Plan to upgrade to bi-amp monitors from passive crossovers when you can, but most importantly, have your system as aligned as possible.

Now that you are done with the set-up, you are finally ready to mix for the band on-stage. Dealing with the band is where you either make it or don't. The most important words are confidence and psycho-acoustics.

I have watched many competent "House" engineers get fired when they got behind the monitor board. The reason for this, if they are familiar with their equipment, is because their attitude caused the band to lose confidence in them.

There is nothing more important to a band than how they feel on stage. If they feel great, then you're great, but if they don't... Please realize that the most important thing to a musician is feeling confident in you and your equipment.

Unfortunately, if the house engineer refuses to fix his mix, then the only remedy is a bazooka—pointed from the monitor desk to the house desk.

If the "Nuclear Fission Gyrometer" has just stopped, don't yell across the stage in a panic, "Everybody run!" Call your assistant over calmly, and then work on the problem quietly. It's ok if the band sees you sweating bullets working on something, but if they hear a scream for help followed by a string of swear words, then they're going to worry. If a band member asks for something to be changed, they are looking for "instant gratification."

Monitor engineers talk all the time about So&So-famous-artist who asked for more vocals in his monitor, "but before I could touch the knob, said "Thanks, it's perfect." The important thing is that the monitor engineer moved towards the knob as soon as the request was made, and therefore boosted the artist's confidence.

RULE #1

Move towards the knob quickly; the problem may be fixed before you get there. If someone asks for more vocal, and you know the vocals are located on the right side of the board, then start moving to the right before you even look down to see where your hand is going. I know of one monitor engineer who always "searched the board" looking for the exact knob to turn before moving to reach it. This constant delay cost his job, because performers labeled him as "slow."

RULE #2

Look at the performer when turning up the knob. Don't just turn up the knob a bit and ask "How's that?" Turn the knob up *and keep turning it up steadily until the performer says stop*. There is no time to waste, so do what you're told; don't try to outguess the performer.

RULE #3

Appear organized. After setting up the monitor system, every band member will try to tell you all at once what they want. It helps to be able to listen to three people at once, but the confusion factor is absurd! I have found that it is quickest to first go across the stage and make sure each person has their own vocal/instrument in their own mix.

Don't get trapped into doing everyone's everything in the first 30 seconds. Introduce yourself in a loud voice to everyone on stage, and bring everything to a stop (your talkback mic can help with this.) Then explain to the band that you want to make sure everyone has their vocal and/or instrument in their own mix.

After everyone is satisfied, then explain you are going to work from one side of the stage to the other with each performer. Start with the lead vocalist and give him whatever he needs. If drums are needed in the monitor, obviously the drummer will need to play for a minute; if they need keyboards, then the keyboardist plays, and so on.

When the first mix is done, move to the next player and repeat the process. Completing the process should take no more than 15 minutes. When done, ask the band to play a song. While they are playing, take a stroll to each player and ask what needs to be changed.

RULE #4

You *must* become personable with everyone on-stage. I have seen monitor engineers lose their jobs because they were too timid to stroll confidently onto the stage to ask about the sound. This is your job and privilege.

Once the band has all of their mixes together, your job is 90 percent done for the day. This means that during the rest of the sound check, you should regularly check in with each person, making sure no changes are needed. Establish a set of hand signals so each player can communicate with you during the show without you going on-stage.

Five minutes prior to the show, go through all of your mixes. Make sure all mics work in their monitors. Play the keyboards and drums so you know they are on and work. Nothing is more embarrassing than for the band to go on stage and no monitors are on because you forgot to un-mute the mixes. It's happened to everyone.

About the only recourse you have is the knowledge that you did all you could, and that plenty of other national acts have not given you this problem.

Once the show starts, it's a good idea to keep your eyes constantly roaming across the stage so not a minute goes by that anyone can't communicate with you. In a festival situation where everything's more relaxed, it's not a bad idea to zip out on stage and ask. After the first couple of songs, everyone will hopefully be comfortable, and the roaming eye technique will suffice.

NO FEEDBACK!

The most important part of doing monitors is not to let them feed back. Nothing destroys an artist's performance more than having to constantly cringe because their vocals are suddenly a squealing mess. If you have to turn down the monitor, then do it, but don't leave a performer constantly battling the sound system. It's embarrassing to have the performer and audience go home talking about the bad job the monitor engineer did.

Obviously, there are times where the Main P.A. is the cause of feedback. Although there is very little you can do about it, make sure it is not you feeding back. Turn off your monitors momentarily. If the feedback continues, immediately contact the house soundman and let him know it has been confirmed that the house system is the problem.

If the problem continues, the band is going to get upset. Unfortunately, there is no way of showing them the problem is not you. We've learned that in festival situations, if we use the same monitor engineer and the monitors have worked fine all day, and one act is plagued by feedback, then it is probably that acts' house engineer.

Unfortunately, if the house engineer refuses to fix his mix, then the only remedy is a bazooka—pointed from the monitor desk to the house desk. You may laugh, but remember this: the band has no way of knowing what's going on. All they know is they have feedback, their monitors appear to momentarily "cut out" (as you make sure it's not you) and that they are being disturbed on-stage. They have good reason to be upset.

It's awfully hard to prove to the band you had nothing to do with the problem. About the only recourse you have is the knowledge that you did all you could, and that plenty of other national acts have not given you this problem.

You may want to explain to the band what has gone on. Don't expect them to immediately "cool out" and give you sympathy. If the house soundman admits to the band that he was the cause, that's one thing. But if he's covering himself and refuses to discuss the matter, your name will have a cloud over it until some future date where you can finally clear your name.

After a show with Clarence Clemons, I once had to wait five years to clear my name. When I finally worked with him again, I made sure he left that night thinking he'd been to visit the *monitor god*.

Once the show is half-over, your job is 99 percent done. Hopefully, you'll learn something every day. Don't be afraid to look and see how many faders you really used on the EQ. If you used more than nine out of 27, see if you can figure out why. Was the monitor positioned badly? Did the mic have a large prescence peak that used a lot of EQ to get rid of?

After you've examined your job and figured out how to improve it next time, relax, but don't take your eyes off the performers until the band leaves the stage. Make a note of any damage (on paper and make sure it gets taken care of back at the shop) and then pack it up and take it home. db

RICK SHRIVER

Movie Making in China

In 1988, I was invited to teach video production in the People's Republic of China. Marietta (Georgia) College, where I am a faculty member of the mass media department, has begun an exchange program with the Southwest University of Finance and Economics (S.U.F.E.) in Chengdu, Szechuan Province. Our department was selected to participate in the first year of this program, and I was lucky enough to go.

We were officially invited by the Chinese Government to go to Chengdu from May 1989 to July 1989. Accompanying me on the trip was a student in our broadcast program. Together, we were to conduct lectures and seminars on the basic video production skills required to produce programs with higher production values than the "talking heads," as well as chalkboard lectures which had been the norm at SUFE.

We arrived in Shanghai May 15, but that was about the last thing that happened the way it had been planned. Due to the then-happening civil unrest, we left China about two weeks before we had expected to return to the United States.

While in Chengdu, we had the opportunity to visit many fascinating sites including temples, monasteries, parks, zoos and the like. The most interesting tour we experienced was at Emei Film Studios, one of two major studios which produce most of China's feature films.

Like film studios in the United States, the Emei lot is a completely walled and fenced compound, and admittance is gained only after clearance with the guards at the gate. Once inside, things are not much different than at any other studio. The compound is populated with a dozen large buildings housing sound stages, processing labs, audio studios, editing suites, screening rooms, offices and other requisite facilities.

UNRELIABLE A.C.

A common feature to nearly every building we visited in Chengdu was the dark, damp and dirty interior. China is on a 240-volt electrical system, which is highly unstable and quite unreliable in the Szechuan Province. Therefore, lighting is minimal, and air conditioners and de-humidifiers are practically non-existent. The result is a less-than-ideal environment for electronic equipment in light of the fact that the climate is somewhat similar to the southern United States or Mexico. The sound stages are in a dismal state of disrepair. It was explained during the tour that almost all film production is now done on location, and the studios have become nothing more than storage barns.

Figure 1. Making ready a master for playback in the tape machine room.

SHARP CONTRASTS

Contrast best describes China's appearance, for as unattractive as the building interiors are, the grounds are beautifully kept. The Emei lot is landscaped with gardens, fountains, reflecting pools and lavish plantings of all sorts. The contrast extends to other aspects of Chinese culture as well. A typical sight in Chengdu is a peasant farmer with his water buffalo working a field in the shadow of a high-rise building with its roof-top satellite receiver. The streets are equally crowded with people whose only mode of transportation is a bicycle as well as those who drive Audis and Mercedes.

Similarly, the contrast is apparent in the equipping of the facilities. As dank and dreary as the buildings are, the equipment is nothing less than the best. China, as a developing nation, has received several generous grants from the World Bank to improve its media systems. As a result, Emei's audio production facility is outfitted with Studer 24-track and two-track machines, Nakamichi cassette decks, an ADM custom film-mixing console, JBL 4345 monitors and an outboard rack equipped with Yamaha, Lexicon, Urei, dbx, Ursa Major and others. All this state-of-the-art hardware appears to be a visual paradox in the environment and culture in which it resides (*Figure 1*).

CONSPICUOUS STUDIO FEATURES

A conspicuous feature of the studios and control rooms is the power conditioner. Each room is outfitted with a device attempting to level out the highly erratic voltages. It was our observation that these power conditioners operated with varying degrees of effectiveness. For example, in the video facility where we worked, the computer-assisted editor regularly "lost its mind" and the edit list contained therein.

Another curious aspect of virtually all the production facilities we toured was the method used for connecting equipment to A.C. power. The generally accepted method is to cut off the plug and insert the bared wires directly into the outlet. This was a bit unnerving by our western standards. The rationale for this practice is based on the reality of at least a half dozen different configurations of connectors at both the equipment and supply ends. So rather than deal with the myriad adaptors which would be required...

Figure 2. The ADM console. Note that it is not located in a control room, but in the studio itself.

NEWLY INSTALLED AUDIO FACILITIES

At the time we visited the Emei studios, a new audio facility which will house the existing equipment was just being completed. The old studio, a huge, cavernous room finished in dark woods, is of Soviet design. The surface treatment yields a very reverberant quality. The studio is like a dimly-lit empty cathedral with the acoustic characteristics of a gymnasium.

The main console (the ADM) is located in the studio, versus the main control room, *Figure 2*. How this works for scoring and track laying was quite beyond our comprehension. A satisfactory explanation was never given by our Chinese hosts. The studio also contained a screen and projection booth, serving as a film-mixing facility as well, and it appeared to function well in that configuration.

There is a small control room attached to the studio which houses a 12-input Studer console. This was the room of choice for doing more "pop" types of music. This control room also contained a Yamaha DX-7 and limited outboard equipment.

The new studio was designed by Chinese engineers from the Engineering University of Chongqing. The acoustic treatment of this room is very similar to that of a Western studio. The surfaces feature anechoic wedge shapes, fabric covered forms filled with an absorptive glass fiber material as seen in *Figure 3*. The studio is spacious by our standards and is decidedly non-reverberant. The walls and ceiling are finished in much lighter colors, and consequently, the room has a more open feel.

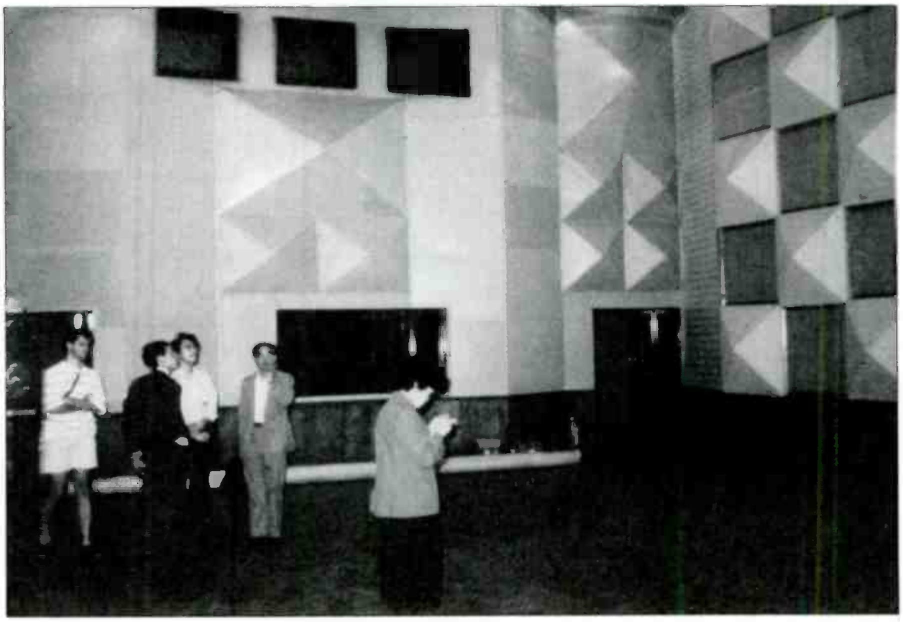
WESTERN STYLE

The new studio and control room will be patterned in a Western style with the console located in the control room. It will be used for track laying, scoring, foleying and dialog replacement. The new facility will also include a small studio for voice-over which will be linked to the main control room.

Final mixes will be done in a newly outfitted multi-function theater. This room is luxuriously appointed (even by Western standards), and features comfortable, spacious seating. The mix position is at the rear of the auditorium.

The acoustic treatment features thousands of absorptive cells along the side and rear walls. The absorbers consist of cubes roughly 10 by 10 inches, which are filled with a fibrous material and covered with colored fabric. See *Figure 4*.

Figure 3. The new main studio. The lower windows at the rear are the control room (right) and the voice-over studio (left).The windows above are for the future installation of projection equiment.



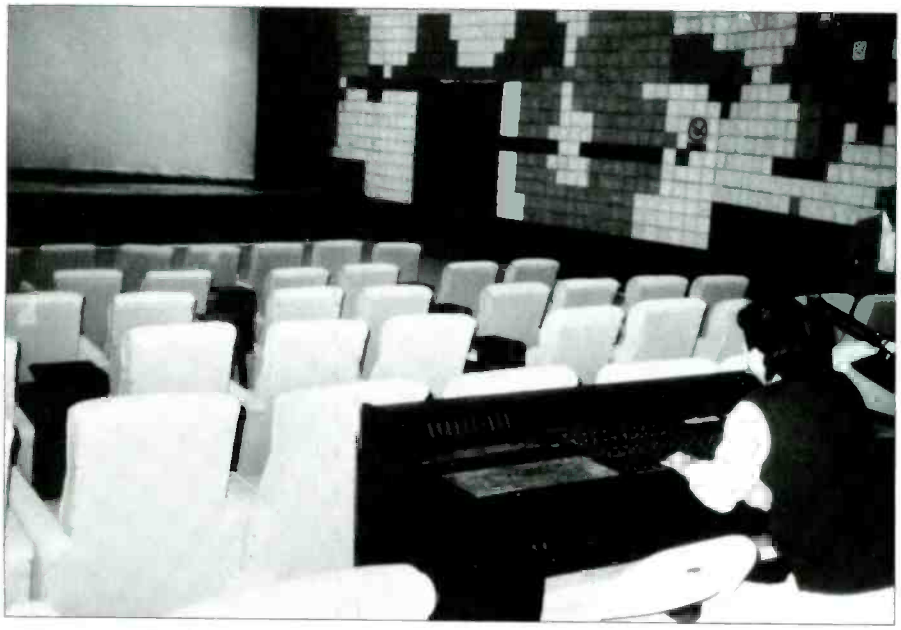
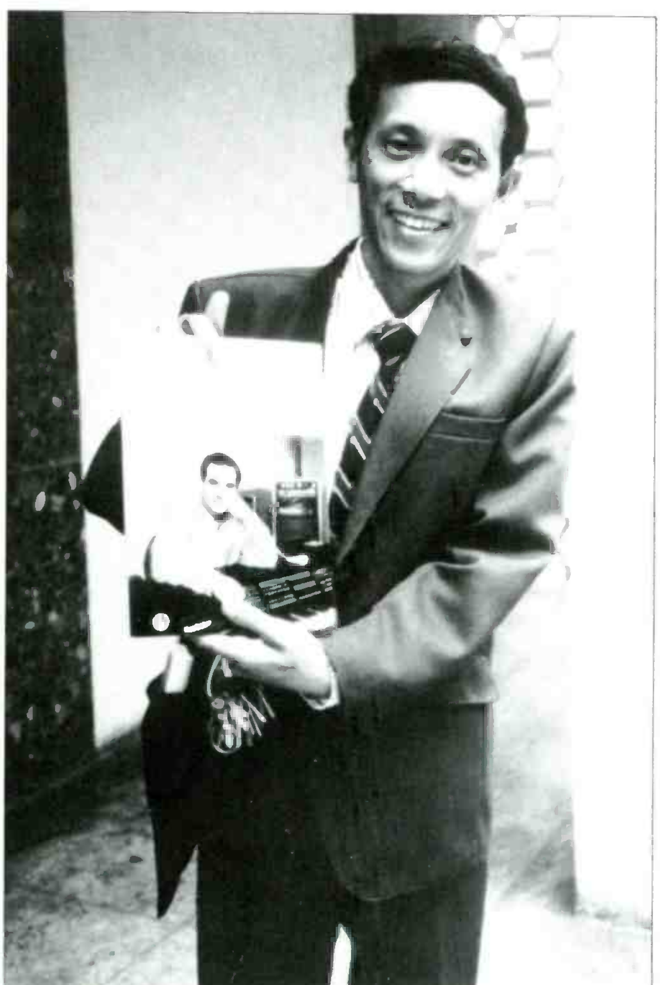


Figure 4. The theater. You are looking forward from the mix position.

Figure 5. Mr. Shi Yulin, Director of the audio facilities at Emei. He is holding a September/October issue of db Magazine which contained an article by author Rick Shriver on the construction of his Electronic Cottage studio. The photo on the cover is of Shelton Leigh Palmer, tapeless studio owner, also featured in that issue.



Another use of the theater is previewing completed features. Communist party officials regularly scrutinize films to make sure the content is "appropriate."

Because China is still decidedly Third World economically and culturally, recording has not become the flourishing industry that it has in many Western countries. As a result, popular music artists frequently find the only recording facility that exists is in film, television or radio studios. In fact, our guides explained that the studio at Emei was the only one with multi-track capability in the Chengdu area, which has a population in excess of one-and-a-half million. As a result, music recording has become a lucrative sideline.

Strangely enough, the most popular genre of music in China currently is "disco." It is the same form of dance music which dominated the United States' music scene in the 70s. Likewise, the whole disco phenomenon is enjoying this popularity. Dance clubs with spectacular light shows, lighted dance floors and polyester-clad dancers can be found in most cities. But as with many adopted western customs, it's a close-but-no-cigar imitation. A sign in the lobby of the Jingjang Hotel reads *Saturday Fever In The Roof-top Lounge*.

Another disconcerting tendency, by our Western standards, is that of the Chinese to select same-sex dance partners and to openly display same-sex friendship in a more physical way than we are accustomed to seeing. On more than one occasion, we were asked by young Chinese fellows to come dancing.

Lest this essay end with a negative impression of media production in the People's Republic of China, it should be noted that the Chinese have managed to adapt and persevere, and successfully turn out a high quality product, even in the face of these adverse conditions. While the content may not suit our tastes, the quality of the recordings we heard was excellent. The frequency and dynamic range of films scored with traditional Chinese instrumentation was spectacular.

SUMMARY

Given time to reflect on the overall experience, the cultural and technological exchange seems to have been successful. We probably learned as much as we taught. Certainly, we have a life-time of stories to tell, some about being in and trying to get out of Communist China during the "civil war" of 1989.

Chengdu was the only other city outside Beijing to experience any large-scale violence associated with the pro-democracy demonstrations. On the day we boarded the train to Shanghai, thousands of protestors began burning and looting the downtown district. Troops from Mongolia quickly moved in and "quieted" the unrest. The 18-hour trip to Shanghai lasted two-and-one-half days, as demonstrators halted rail traffic to prevent troops from moving in and out of Beijing. The train just ahead of ours into Shanghai was stopped and burned. So much for seeing history being made—we decided it was a good time to get out of China. db

Hands On

Crown's System 2000 Software for the PC

• We've been playing with a preliminary, or beta-test version of the System 2000 software, long available for the Macintosh, but now being released for the PC.

The version 0.45 we were given for this evaluation came with no instructions or other written data—just a 3.5-in. disk with a program called CROWN.EXE on it.

So we placed in an appropriate drive and invoked the disk by typing CROWN [ENTER]. We were using for evaluation an AT clone running at 8 meg.

Almost immediately, a screen came up on our monitor informing us that since we had no amplifiers connected we were to push our space bar and that got us to the main menu.

From there, the correct next step is to go to the configuration screen (a single F-key stroke) to tell the program what baud rate and other parameters the controller requires for communication. Then a single keystroke, and we are back at the main menu.

THE MAIN MENU

In the main menu you enter the information the program will use to control the amplifiers you have out in the field. Crown tells us, and we did not verify it, that you can have up to 2,000 amplifiers operational with the present program.

Each stereo amplifier is fundamentally treated as two separate amplifiers. But for each stereo amplifier you can enter an identifying number and zone.

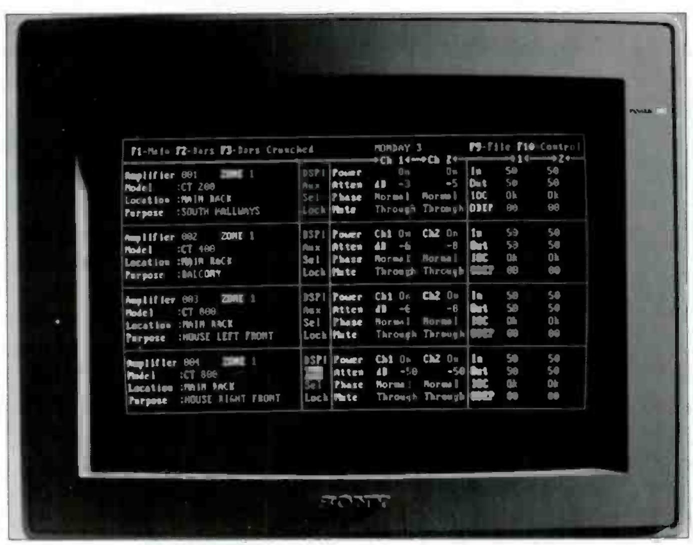


Figure 1. The main menu.

Then for each channel you can power up or down, attenuate over a wide range, reverse phase, and mute. In turn then, the Main Menu will then tell you each channel's In, Out, IOC and ODEP ranges.

ALTERNATE MENUS

There are two alternate menus. Where the Main Menu only shows four amplifiers at a time (with scroll capability down for as many as you have) the alternate menus show eight channels at a time in actual performance. In effect, you are seeing VI performance of each channel. Of course, you also would quickly be made aware of a malfunctioning channel and then switching back to the Main Menu you would identify fully the particular amplifier and could then equally quickly change its parameters or even shut it down.

DIFFERENT CONFIGURATIONS FOR DIFFERENT CONDITIONS

You may be using the same banks of amplifiers for different kinds of shows. This program permits you to save each program settings separately as a different file. Then you only need to load the correct file, and everything is automatically set for your show

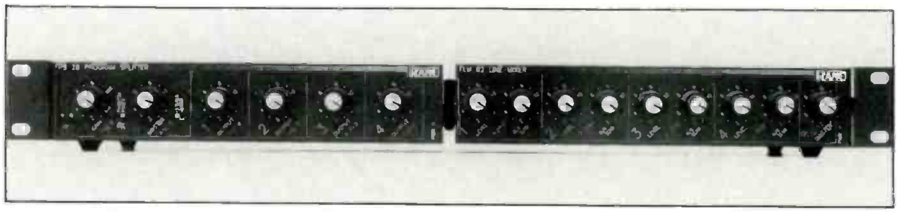
CONCLUSION

Crown has an effective and useful tool available here. The most complex setups can be controlled from a single PC. This is a logical and easy-to-use program.

Still, we would wish that Crown would include a program guide booklet. Not everyone of their customers is necessarily sufficiently computer-literate to be able to pick this one up cold. db

New Products

SIGNAL PROCESSORS



• The two newest members of the growing line of Flex Series modular signal processors are announced: the FPS 28 Program Splitter and the FLM 82 Line Mixer. The FPS 28 allows two mic or line level inputs (with switchable mic phantom) to be split to 8 mono or 4 stereo outputs via independent level controls. Additionally, these outputs may be assigned pre or post master input level control. The FLM 82 provides 8 mono or 4 stereo line inputs, each with separate level control and pre/post assignable auxiliary send control for effects processing or another independent mix. This module also features master level controls, aux loops and the Flex Bus System for single-cable connection to any number of other Bus-equipped modules.

Mfr.—Rane Corp.
Price: $399.00 each
Circle 29 on Reader Service Card

WIRELESS ACCESSORIES

• The L2 Handheld Transmitter has been added to the L Series line of wireless microphone products. The L2 is available in three different versions. Model L2/58 features Shure's well-known SM58 dynamic microphone element, while Model L2/96 incorporates the condenser element used in Shure's high-performance SM96 vocal condenser microphone. A third version, L2/Beta 58, features Shure's acclaimed Beta 58 element and is available exclusively from authorized Shure Beta dealers. Since the transmitter "heads" are easily interchangeable, any of the three elements may be used with the same L2 transmitter. Dual-trace, gold-plated wiper contacts are used, making it possible to change heads in seconds, with no wires to solder or unplug.

According to Shure, the L2's designers have devoted unprecedented attention to preserving the distinctive sounds of the SM58, SM96 and Beta 58 in the L2 wireless versions. In addition, the manufacturer is subjecting its L2 wireless microphones to the same battery of environmental and mechanical shock tests its cabled mics undergo to assure the level of ruggedness and reliability associated with its other products.

L2 users will normally get 12 hours of operating time from a standard 9-volt alkaline battery, and a lithium battery may be used if even longer operating time is desired. The battery may be replaced without removing any parts from the transmitter body. Shure L2 transmitters are designed to work with Shure L Series receivers, and may be purchased separately or as part of an L Series system. Available systems incorporating the L2 transmitter include the LS23/58 (L3 non-diversity receiver and L2/58 transmitter), LS23/96 (L3 and L2/96), LS23/Beta 58 (L3 and L2 Beta 58), LS 24/58 (L4 MARCAD diversity receiver and L2/58), LS24/96 (L4 and L2/96) and LS24/Beta 58 (L4 and L2/Beta 58).

Mfr.—Shure Brothers
User net prices for L2 systems range from $532.00 to $748.50. Systems are available in six stock frequencies between 169 and 185 MHz, and other frequencies are available on special order.

Circle 30 on Reader Service Card

db magazine

serving: recording, broadcast and sound contracting fields

BUYER'S GUIDE

CONSOLES AND MIXERS

On the pages that follow, we present this issue's Buyer's Guide on consoles and mixers. The information supplied is furnished by the respective manufacturers. Further, if a manufacturer you seek is not listed, the chances are strong that as many times as we tried we could not get information from them.

ALLEN & HEATH

SR series is a low-cost mixer designed for sound reinforcement and 2 and 4 track recording. They are available in five configurations ranging from an 8 x 2 x 1 through a 24 x 4 x 2 x 1. Features include four-band EQ and 4 aux. sends per input channel, 2 additional FX return sections.

Spectrum Series recording consoles are available in 16, 24, and 32 input channel and tape output configurations which also have features for live-sound applications. The smallest 16-channel version has 40-input capabilites during mixdown and feature MIDI muting as a standard feature.

Scepter 12-channel, 20-input rack mixer features Alps faders, direct outputs on all input channels. Balanced outs are provided on a left, right, and mono channels. Separate mono and stereo tape outputs are also provided.

Sigma consoles are 24-bus in-line consoles with input configurations up to 56 inputs. Each module can process 2 separate input signals, microprocessor-controlled MIDI muting is standard, EQ is 5-band, and there is 8-knob aux. send into a maximum of 32 buses. Full patch-bay systems are also available. The console ranges from 16 x 16 to 56 x 24.

Saber 16-bus consoles are designed specifically for 8, 16 and 24 track recording and sound reinforcement. All versions include microprocessor controlled MIDI muting of all inputs. Input channels include 4-band EQ with switchable high- and low-frequency ranges and 2 bands of mid sweep plus high-pass filtering. Aux. sends include 6 buses, all recording versions have 16-channel tape monitoring. 24-channel tape monitoring as well as patch bays are available.

SC Modular Series is an advanced version of the SR series with fully modular design. Available in 16, 24, and 32 input configurations.Modules can be intermixed and include stereo input, fixed frequency EQ, and sweepable EQ.

SRM Monitor Mixers are available in 18 or 24 input configurations. This series is designed specifically for on-stage monitor application. All units include a built-in splitter and road case.

System 8 is a recording console but can also be configured for sound reinforcement. It is 8-bus for 8 and 16 track recording or MIDI recording. All versions have Alps faders, direct outs on all input channels and 16 channels of tape monitoring that can also be used as additional FX inputs when used for sound reinforcement. Configurations are 16 or 24 input. EX7 is an optional 8-input expander.

AMEK/TAC (TOTAL AUDIO CONCEPTS LTD.)

Bullet is a 10/4/2 modular free-standing or rack-mounting multi-purpose console with hard busing system, hybrid circuitry, 15 segment LED metering, 4-band EQ with swept mids, 6 aux. send buses with 4 stereo returns.
Price:
$4,250.00-$11,500.00

Scorpion II has four chassis sizes including a patch-bay version. There are recording and reinforcement versions, all with 16 to 40 inputs, four-band EQ, and 4 to 8 aux sends.
Price:
$8,427.00-$25,504.00

Magnum is an in-line recording console, with 24-bus, and 26 or 36 input versions, and with an optional 32-T monitoring version. Optional MIDI muting and fader automation packages. Has LED or VU metering.
Price:
$29,267.00-$39,180.00

SR9000 Superconsole is for sound reinforcement and has 42 input modules (each with 2 mic and 1 line input), 16 subgroups, 8 VCA groups, 8 mute groups, 16 x 8 matrix, fully parametric 4-band EQ, swept HP filter, 16 aux. sends, in-place solo feature and 24-channel extender option.
Price:
$83,215.00

Angela 24-bus in-line recording console comes in three chassis sizes, 24 to 62 inputs. Features include: on-board or external jackfield, 4-band semi-parametric EQ plus HP and LP filters, 6 aux. sends, 6 aux. returns, flexible routing possibilities, and automation options.
Price:
$40,581.00-$99,664.00 plus automation

Mozart multi-track recording console features 32 balanced buses, 32 to 80 inputs, in-line or all-input type modules (or a combination), integral switch grouping computer for user-configurable master status switching, 4-band semi-parametric EQ, HP and LP filters, 16 aux. sends, 12 stereo returns, 4 floating patchable stereo faders and is available as standard with the AMEK Supertrue automation system. System is optionally available with GML moving-fader automation.
Price:
108,308.00-$397,231.00

APC1000 assignable production console has up to 128 all-input type modules, and is 48-bus. Channel switching and routing is all under the control of a central assignment keyboard. Three automation computers work separately and simultaneously: Recall (knob positions), Reset (switch settings), and GML moving faders. Synchronous Rest recalls switch settings against time code. Optional dynamics modules also come under automation control.
Price:
$301,975.00-$587,110.00

ARX SYSTEMS —See our ad on page 2

DI-6 is a 6-channel DI and line mixer which permits 6 independent audio sources to be interfaced with balanced low Z systems either a 6 individual channels or summed down to one master output. A 2-watt headphone output is also provided.
Price:
$549.00

AUDIO-TECHNICA U.S., INC. —See our ad on Cover II

AT4462 is a stereo field production mixer. Its features include 4-input channels, 1 and 2 mono pannable, 3 and 4 true stereo on a dual concentric central, headphone with cue system, phantom power, slate tone and mic, limiter adjustable (mono or stereo) built in IFB system, 3-tone oscillator, low cut on all channels, switchable, level warning system in operator phones when approaching clipping. Has Cordura™ carrying bag.
Price:
$1,360.00

AUDITRONICS, INC.

200 Series broadcast consoles are available in 6, 12, 18, and 24 input "drop in" mainframe configurations. Each may be configured with any combination of mic or stereo line input modules. Each module includes dual input selection, VCA fader control, dual stereo outputs, and many user-programmable logic functions. Standard features include all output amplifiers, a control-room monitor, headset amplifier, and a cue amplifier.
Price:
On request.

310 Series consoles are for radio and TV production use. Standard features include 4 aux send and returns, output submastering, 4 group master faders, VCA fader control, cue, stereo solo, stereo monitoring, phantom mic power, and a complete metering package. A wide range of optional equipment is available for custom needs.
Price:
On request.

400 Series on air and production consoles are available in 3 mainframe sizes: 18, 24, and 30 inputs with 4 or 8 submaster outputs. There are 2 aux/foldback buses, VCA faders, phantom power, and many user-defined features.
Price:
On request.

700 Series consoles are available in 24/16 or 48/24 configurations. Custom mainframes are also available.

Price:
On request.

900 Series TV on-or and production consoles are available in 6 mainframes with 4 or 8 submaster output capability. Standard features include 4 aux send and returns, 4 group master faders, VCA fader control, cue, stereo solo, stereo monitoring, phantom mic powering, and a complete metering package.
Price:
On request.

CARVIN CORPORATION

FX844 is a sound reinforcement/recording console with 8 in, 4 out capability. It has 250 watt/channel at 4 ohm amplification.
Price:
$1,349.00 sold directly only.

FX1244 is as above but with 12 by 4 configuration.
Price:
$1,699.00 sold direct only.

FX1644 is as above but has 16 by 4 configuration but no power amplification.
Price:
$1,699.00 sold direct only.

FX2444 is as above but has 24 by 4 configuration.
Price: $2,299.00 sold direct only.
MX1688 is a recording consoles with 16 inputs and 8 outputs.
Price: $3,199.00 sold direct only.
MX2488 is similar to the MX1688 but offers 24 in by 8 outputs.
Price:
$4199.00 sold direct only.

DDA—KLARK-TECHNIK ELECTRONICS INC.

DCM 232/224V are 32-bus in line format recording consoles, including central automation of module switch functions. Console status can be stored in 30 snapshot memories. Available with VCA fader automation, Alpha Boss II editor installed as option.
Price:
$120,000.00 up.

AMR 24 are 24-bus consoles with up to 44 inputs. A large number of inputs to the console is possible by utilizing all inputs, all 24 sends, and all 24 monitor returns on mixdown as inputs.
Price:
36/24/2 4 $69,740.00

DMR 12 is a 12-bus, 24-track monitor return console including a patch bay. Each input has two line and one mic input, up to 112 line inputs are available on mixdown, aux send 7 and 8 are switchable to stereo EFX returns.
Price:
On request

Arena Series VCA 8-bus sound reinforcement consoles are available with 16 to 40 inputs, they include 8 assignable mute groups, 8 assignable VCA groups, and options such as stereo returns on faders which are assignable to 8 VCA/mute groups as well.
Price:
40/8/2 $42,000.00

Q-Series 8 bus consoles are available with 16 to 40 inputs, mute version available, all versions include 8 aux returns with EQ, aux sends with EQ, direct output with each input, 8 x 4 matrix.
Price:
40/8/2 mute $26,900

S-Series Standard are 4 bus recording consoles with 16 to 32 inputs, 4 aux sends, 4 aux returns, 4-band EQ including 2 sweep mids.
Price:
24/4/2 $9.075.00

DOD ELECTRONICS CORPORATION

820 and 1220 stereo mixer series feature high and low impedance inputs, 15 dB cut and boost EQ, RCA-type outputs, phantom power, effects sends and returns and individual monitor outputs. The 8 or 12 channel mixers can be had in basic unbalanced to full balanced configurations, and with phono or XLR inputs and outputs
Price:
Dependent on configuration.

ELECTRO-VOICE, INC.

8108 is a rack mounted mono mixer with four outputs.

8208/8212/8216 are 2-bus 8, 12, or 16 channel sound reinforcement consoles. Features include equal headroom in all stages, gain-calibrated level controls for visual check of mixer input stages, transient performance not slew-rate or power bandwidth limited under any operating conditions from 20-20,000 Hz.
Price:
$1,670.80

8416/8424/8432 are 4-bus sound-reinforcement consoles with 8, 16, 24, or 32 inputs. Features include rear-panel mic inputs including 48 V phantom power, channel patching and output jacks on each input panel, flourescent bargraph metering, talkback, panning, solo, and subgrouping.
Price:
$5,481.00-$10,091.00

BK832/BK1232/BK1632/BK2432 are stereo sound reinforcement consoles with 8, 12, 16, and 24 inputs, with 4 outputs. Features include channel insert send and return, input trim, high, mid, low equalization, panning, PFL assign, LED-bar level meters, subgroup patching send and return left and right phantom power, sub, main and monitor output master faders.
Price:
$1,103.80-$2,048.80

FURMAN SOUND, INC.

Rackmount mixers MM-4A and MM-8A feature four inputs, mono (MM-4A) or stereo (MM-8A) outputs, pan pots on each MM-8A input, effects bus with send and return jacks, stereo aux. inputs with RCA jacks and level control, low-cut buttons on each input (-3 dB at 100 Hz), master fader, headphone amp with front panel jack and volume control. This is a compact utility mixer for sound reinforcement or recording use. "-B" models have balanced ins with both phone and XLR connectors and mic/line switches. "-BP" models are the same as "-B," plus 48-volt phantom powering on all inputs and phantom power switch.
Price:
$335.00 (MM-4A)
$375.00 (MM-4AB)
$405.00 (MM-4ABP)
$395.00 (MM-8A)
$435.00 (MM-8AB)
$465.00 (MM-8ABP)

GOTHAM AUDIO CORPORATION (AUDIO DEVELOPMENTS, LTD.)

AD 160 is a mono engineering mixer with the following features: 3 mic inputs, 1 line input (all are balanced), 1 line output (balanced), limiter on line output, phantom/A-B powering on mic inputs, 1 kHz oscillator, talkback mic, VU or PPM meter, monitor output, battery or a.c.(with optional a.c. adapter).
Price:
$1,765.00

AD 260 stereo engineering mixer is the same as above with the following differences:
4 mic/line inputs with pan pots, 2 line outputs, 2 limiters on line output (linkable for stereo), 1 stereo aux. input.
Price:
$2,500.00

AD 145 Pico mixer has 4-8 mic/line inputs with pan pots (all are balanced), 2 line outputs (balanced), 3-band equalization on each input, phantom/A-B powering on mic inputs, 1 kHz oscillator, talk back mic, monitor output, cue input. Battery or a.c (with optional adapter)
Price:
$4,400.00 to $6,775.00

AD 062 multi-mixer has 4-16 mic/line input modules with pan pots (balanced), 2 line outputs, 3-band equalization on each input, phantom/A-B powering on mic inputs, auxiliary send on each input module. Options include stereo line input module to replace mic/line input module, dual auxiliary return module, communications module, stereo compressor/limiter module. Standard features include monitor level, selector controls, master auxiliary sends, A + B mixdown, battery or a.c (with optional a.c. adapter).
Price:
$6,000.00 to $20,000.00

HILL AUDIO INC.

Multimix is 16 channels in a rack and can be used as a 16 x 4 x 2 x 1 or 12 x 4 x 2 x 1 or 16 x 2 x 1 console. With 16 mic/line inputs, 4 subgroups, 3-band EQ with defeat, direct outputs on all channels, 2 auxiliary sends (expandable), 100 mm faders, 48-volt phantom power, 4 RIAA equalized inputs, this console is designed for recording, broadcast or sound reinforcement.
Price:
Price on request.

Soundmix is a semi-modular console available as 24 x 4 x 2 x 1 or 16 x 4 x 2 x 1. Designed for recording, broadcast or sound reinforcement, features include: 16 or 24 mic/line inputs, 4 subgroups, 4 auxiliary sends, 4 auxiliary returns, 4-band EQ, 100 mm faders (Alps or Noble), 48-volt phantom power, direct outputs on all channels and subgroups, insert points on all channels, subgroups and masters, PFL system, and 12 segment LED display on subgroups and masters.
Price:
Price on request.

Stagemix is a 12x6 rack-mount monitor console with a 12-channel zero splitter (built-in), 12 transformer balanced input channels with 3-band EQ, peak LEDs, PFL system and individual channel mutes. Other features include 6 transformer balanced output channels with 4-band EQ and EQ defeat, individual output mutes, 12-segment LED display, PFL and AFL system, TRS effects loop and the console is color-coded for ease of operation.
Price:
Price on request.

Remix is a 24 x 8 x 16 x 2 console designed for 8 or 16-track recording or sound reinforcement as a 24x8 console with 8 effects returns. Features include 24 mic/line input channels with peak LEDs, 8 subgoups, 16-track monitoring, 6 auxiliary sends, tone oscillator, 2-track return, 4-band EQ with sweepable mids, 16 tape returns with EQ and aux. sends, direct outputs and insert points on all inputs, subgroups and masters.
Price:
Price on request.

Concept 200 and 400 Series are sound reinforcement and recording consoles. Available from 16 x 8 x 2 with an 8 x 8 matrix to 56 x 24 x 48 x 2, these consoles feature patch bays, Alps or P&G faders, 3-way speaker select, 2 types of EQ available (6-band with sweep filter or 4-band sweepable), 8 or 12 auxiliary sends using dual concentric or 4/6 sends switchable to 8/12, programmable mutes, A&B 2-track returns, d.c. controlled subgroups.
Price:
Price on request.

INDUSTRIAL RESEARCH PRODUCTS, INC.

DE-4013 automatic microphone mixer has modular construction in 2-channel increments. Its features include: up to 12 inputs in a single chassis, tandem connected chassis allows addition in 12 channel increments, 12 V phantom power, level-matic AGC. Auxiliary output and chairman override are standard.
Price:
Dependent on configuration.

DE-4018 is the same as DE-4013 but includes DE-208 TEQ 9-band transversal equalizer module. 10 channels in single chassis before tandem connecting others.
Price:
Dependent on configuration.

DE-4014 is a mixer with 4-microphone input channels plus auxiliary line level input and auxiliary output. 12 V phantom power is optional.
Price:
$1,467.00

DE-4016 is the same as the DE-4014 but with remote control via DE-207 remote control box. Remote functions are individual channel sensitivity and master gain control. DE-207 box includes 10 feet of cable terminated with an amphenol 165-9 connector.
Price:
$1,860.00

DE-4024/5 is the same as the DE-4016 but includes both auxiliary gain control on both front panel and remote box. Auxiliary output is the mix of gated microphone inputs 2, 3, and 4 plus standard (ungated) microphone input 1 plus the auxiliary input.
Price:
$1,920.00

INDUSTRIAL STRENGTH INDUSTRIES

PM160 is a 16 input (balanced and unbalanced), two internal stereo power amplifiers—400W x 2 at 4 ohms. Built in 99 program digital effects processor, graphic equalizer for stereo and mono, talkback mic, 12-point LED meters, dimensions 32 ¼ W x 21 ¾ D x 6 ½ H, 70 lbs.
Price:
On request.

PM80 is an 8 input (balanced and unbalanced), two internal stereo power amps—180 W x 2 at 4 ohms, built in digital delay, graphic EQ for stereo output, 3 EQ positions for each input channel, 12-point LED meters, dimensions 20 ½ W x 21 ¾ D x 6 ½ H, 35 lbs.
Price:
On request.

INNOVATIVE ELECTRONIC DESIGNS, INC.

4000 automatic microphone mixing system is a modular, rack mount, automatic mixing system which allows custom configurations and programmable gain control. Each frame can hold up to 8 cards creating 32 inputs and offers optional computer control for room combining, and channel on/off.
Price:
On request.

NEVE

VR series console is an automated "flying fader" multi-track recording console for the music, video post and film industries, available in 36, 48, 60 or 72 inputs, including the Formant Spectrum Equalizer, mic/dynamics unit and 8 mono/4 stereo auxiliaries. Additional benefits include a centrally positioned monitor path status indication to enable rapid console status checks and choice of metering options.
Price:
On request

DTC (Digital Transfer Console) for compact disc mastering and is used for the preparation of compact disc master tapes, the unit also provides total digital mixing and processing capabilities. All console parameters can be instantly reset under SMPTE control, also permitting the user to select or mix either of 2 stereo digital inputs and 1 stereo analog input with manual or auto crossfade from AES/EBU or 1610/1630 inputs to compatible outputs.
Price:
On request

8232 console for TV production, post-production and multi-track recording has 32 mic/line input channels with 24 mixing buses and optional stereo reverb returns. Each channel features the Formant Spectrum Equalizer, 4 mono auxiliary sends and 1 stereo cue send, and is "flying fader automated."
Price:
On request

VPR series are multi-track systems for video post and film recording with total storage and recall of all console settings. Dolby matrix monitoring on switchable 4 or 8 track buses up to 48 tracks, 4 or 8 track to stereo or mono TV feeds, separate feeds for music and effects, inset switching for Dolby DS4 matrix with solo interrupt.
Price:
On request

Prism series is a range of rackmount units derived from the V series console comprising a 4U 19-inch rack with capacity for 10 modules that may be powered from an existing console or by a 2U power supply. The 2 modules are the Formant Spectrum Equalizer and the mic amp/dynamics unit comprises compressor/limiter/gate/expander.

PANASONIC/RAMSA

WR-C900 Series are live sound mixing consoles with a 16-input frame expandable to 20 mono inputs, there is also a 24-input frame expandable to 32 inputs. Features include 4 groups, 4 aux groups,L/C/R panning.
Price:
16 input—$26,500.00
24-input—$31,500.00

WR-S852 house sound reinforcement mixing console contains the following features: 52 mic/line inputs, all inputs and outputs are balanced, 8 groups, 8 aux. groups, L/R master, 8x10 matrix, 4-band sweepable EQ and high-pass on inputs, 300,000 operation, 100mm MRP faders, discrete design using hybrid ICs.
Price:
$36,300.00

WR-S840 stage monitor mixing console contains the following features: 52 mic/line inputs, all inputs and outputs are balanced, 18 discrete monitor mixes (18 outputs), 4-band sweepable EQ and high-pass on inputs, 300,000 operation, 100mm MRP faders, discrete design using hybrid ICs.
Price:
$38,500.00

WR-8616 production/post production mixing console is also suited for sound reinforcement. It contains the following features: 16 mic/line or stereo inputs, separate 16-input tape monitoring, 4 groups, 4 aux. groups, L/R and mono master, 3-band sweepable EQ and high-pass on inputs.
Price:
less than $11,000.00 depending on configuration.

WR-T820B 8/16-track recording mixing console contains the following features: 20 mic/line inputs, separate 20-input tape monitoring, up to 48 separately mixable inputs for mixdown, 8 groups, 4 aux. groups, L/R master, 3-band sweepable EQ and high-pass on inputs.
Price:
$8,950.00

WR-S216 sound reinforcement mixing console has the following features: 16 mic/line inputs, last two inputs accept mic or stereo source, "A/B" (L/R), effect, monitor, send and mono master outputs, 3-band EQ (mid-sweepable) on inputs.
Price:
WR-S216—$2,850.00
WR-S212—$2,450.00 (12-input)
WR-S208—$1,750.00 (8-input, rack mount)

WR-133 sound reinforcement mixing console has the following features: 8 mic/line inputs, L/R, effect, monitor, mono master outputs, 2-band EQ on inputs, rack mount option.
Price:
$1,290.00

WR-M10A sound reinforcement/music rack mixer has the following features: 4 mic/line inputs, 2 stereo inputs and each stereo input can select 4 different sources for a total of 8 stereo inputs (2 are RIAA phono), automatic music mute on page (defeatable), L/R, effect, mono master outputs, 2-band EQ on mic inputs, 2-band EQ on left and right master outputs.
Price:
$900.00

PEAVEY ELECTRONICS

Mark VIII sound reinforcement console is available in 24 and 36 channels. It also has the following features: 8 submasters, 8 aux. sends, 4-band sweepable EQ, 8 outputs with LED output level indication, 3-band EQ on aux. returns, stereo L/R outputs, totally modular design.
Price:
$9,999.99

Mark IV sound reinforcement console has the following features: 16 and 24 input channels, four submasters, mono (sum) output, transformer balanced inputs and outputs, 2-monitor sends, 1 effect send, PFL, 4-band active EQ, built-in heavy duty flight case.
Price:
$2,699.99 (16-channel)
$3,499.99 (24-channel)

Mark III sound reinforcement console has the following features: 16 and 24 input channels, 2 submasters, mono (sum) output, stereo, transformer balanced inputs and outputs, 2 monitor sends, 2 effects sends, PFL, 4-band active EQ, built-in heavy duty flight case.
Price:
$2,199.99 (16-channel)
$2,699.99 (24-channel)

MS 2421 sound reinforcement console has the following features: 24 channels, stereo and mono outputs, 3-band active EQ with sweepable mid, 2 monitor sends, 2 effects sends, four 9-band graphic EQ (patchable), electronic crossover, built-in delay, PFL, two submasters.
Price:
$2,699.99

MS-1621 has the same features as the MS 2421 but only 16 input channels.
Price:
$2,099.99

SRC-421 sound reinforcement console has the following features: 16-channels, 4 submasters, stereo, 3 aux. sends (one switchable); 3-band EQ with sweepable mid, LED output level indication, channel patching, PFL, 4 outputs.
Price:
$2,299.99

MD-11 sound reinforcement console is configured in 8, 12 and 16 channel inputs. Its features include 2 submasters, stereo, 1 monitor send, 1 effects send, 3-band EQ with sweepable mid, LED output indication, PFL, channel patching, dual outputs.
Price:
$799.99 (8-channel)
$1,049.99 (12-channel)
$1,299.99 (16-channel)

502 dance music production mixer has 3-channel mixing system, stereo outputs, left and right phono inputs on program channels, low and high impedance mic inputs, 7-band stereo graphic EQ, headphone capability for program and cue.
Price:
$499.99

QUAD-EIGHT ELECTRONICS, INC.

The Virtuoso recording console has dual in-line I/O modules, 20 to 60 inputs, 24 mixing buses, 8 aux send buses, 2 stereo outputs, and VU or 60-segment bargraph meters. Each module has selectable top panel plug-in EQ, and plug-n VCA. Audio, VCA, automated VCA and moving faders are all standard as are Compumix PC automation.
Price:
$60,000 to $225,000

The Filmstar re-recording console is available in 16 to 72 inputs. Features include 8 or 24 mixing buses, 10 aux send buses, 3-channel pan bus and a 2-channel pan bus, film monitor system, 6 x 4 to 24 x 8 matrix speaker assign. Audio, VCA, automated VCA and moving faders are all standard as are Compumix PC automation.
Price:
$80,000 to $600,000

Screenmaster Post Production Consoles are available in 16 to 72 inputs. Features include 24 mixing buses, multi-track monitor mix or re-recording mode, 10 aux send buses, 3 channel and 2-channel pan, or optional 4-channel quad pan, 4-channel monitor for L, C, R and Surround or FL, FR, BL, BR speakers. Each module has selectable top panel plug-in EQ, and plug-in VCA. Audio, VCA, automated VCA and moving faders are all standard as are Compumix PC automation.
Price:
$70,000.00 to $300,000.00

RAMKO RESEARCH

Primus Series consoles feature inputs user selectable mic through high level, balanced ins and outs, indicating push-button select switches, monitor and phones feeds with muting, rack or tabletop mounting.
Price:
P-4M—$634.00
P-4S—$889.00
P-5M—$450.00

xL Series Studio Consoles are 4, 6, 8, and 12 channel stereo units. Features include rotary or linear faders, illuminated select switches, dual or single balanced outs with separate mono feed, 20 watt monitor, 2 watt cue, individual solid state metering, Programmable input levels, muting, on-air lights, remote starts.
Price:
from $1395.00 to $4,150.00

Portable Remote Mixer/Intercom is a battery-powered 4 channel mixer with balanced ins and outs, 2-way intercom, cuing, phantom powering, slate and line tones, limiter/compressor, VU melter with battery-life indicator, leather carrying case.
Price:
PM42—$560.00

SHURE BROTHERS, INC. —See our ad on Cover IV

FP31 is a compact field production audio mixer with the following features: 3 inputs, 2 outputs each switchable for mic or line level, VU meter with lamp, adjustable limiter with LED indicator, tone oscillator, built-in slate tone and mic tone and mic, headphone jacks, battery or a.c. power, phantom and A/B power.
Price:
$1,025.00

FP32 is a compact stereo field production audio mixer, a stereo version of the FP31. It includes all FP31 features plus center-detented stereo pan pot for each input channel and concentric clutched stereo master gain control.
Price:
$1,395.00

M267 is a field production audio mixer with the following features: 4 transformer-balanced inputs, switchable limiter, phantom power, VU meter, headphone jack with level control, LED peak indicator, tone oscillator, mic/line switches on each input and output, low-cut switches on each channel, battery or a.c. power.
Price:
$520.00

FP42 is a stereo version of the M267 and includes all the features of the M267 plus stereo pan pots for each channel and concentric clutched master volume control.
Price:
$990.00

FP51 is a portable gated-memory compressor combined with a 4-input, 1 output mic mixer. Up to 40 dB of compression are avialable. Other audio features similar to FP31
Price:
$965.00

FP16A is a 1-input 6-output distribution amplifier for routing multiple audio signal feeds without incurring losses. Features include up to 90 dB gain, and transformer-coupled XLR connectors.
Price:
$570.00

SOLID STATE LOGIC

SL 4000 G series master studio system is a multi-track music recording and mixing system available with 24 to 72 input/output channel, full dynamics processing and G Series EQ. Also G Series Studio Computer and Total Recall TM.
Price:
On request.

SL 5000 M series audio production system is a stereo broadcast, on-air, continuity and post-production system based on a modular cassette structure. Up to 96 channels, accepts G Series Studio Computer, Total Recall TM and Instant Reset TM.
Price:
On request.

SL 5000 M series film post-production system is a modular, cassette based film console system, available in configurations for ADR/Foley, premixing, music scoring, final mix, video post-production, and multi-operator film dubbing. Up to 96 channels, optional moving fader automation, accepts G Series Studio Computer, Total Recall TM and Instant Reset TM.
Price:
On request.

SL 6000 G series stereo video system is a stereo music, video and teleproduction system with 24 to 72 input/output channels, full dynamics processing and G Series EQ. Also G Series Studio Computer and Total Recall TM automation.
Price:
On request.

SONY COMMUNICATIONS PRODUCTS COMPANY, PROFESSIONAL AUDIO DIVISION

—See our ad on page 16 and 17

MXP-3056 VF audio recording/remixing console is intended primarily for use in recording studios. It has 56 channels which allows for interfacing with the Sony PCM-3348 digital audio multi-track recorder. Each input/output module features modular equalizers and mic/line pre-amplifiers. The Audio Group Master (AGM) function allows for audio grouping on the ACN bus as well as conventional in-line operation.
Price:
up to $190,000.00 depending on configuration

MXP-3036 VF is designed with a vacuum fluorescent (VF) light meter that displays various selectable scales including VU, BBC Peak, Din Peak, Nordic Peak and a d.c. scale. This d.c. scale indicates fader position in the automated version of the MXP-3036 VF. The automated version includes Version 2.0 software and optional wild faders that permit a user to increase the number of effects in a mix.
Price:
up to $110,000.00 depending on configuration

MXP-3000 series also has the MXP-3036 and the MXP-3040 in addition to the two consoles mentioned above. Features common to all models include user configureable with a choice of 5 different equalizers, 4 different mic pre-amplifiers and a hard disc automation.
Price:
MXP-3036—up to $99,000.00 depending on configuration
MXP-3020—up to $40,000.00 depending on configuration

MXP-2900 series audio consoles are available at 8, 16, 26, and 36 channel frames. These consoles are designed for broadcast and post-production applications. Features include 2 independent stereo outputs, 4 stereo internal monitor inputs, and video editor, interface capability.
Price:
On request.

MXP-290 is an 8-channel audio mixer. It can be controlled from the Sony BVE-900 editing control unit. Features include trim control for each balanced mic/line input, built-in 3-band EQ and VU meters with 15 segments of LEDs.
Price:
On request.

MXP210 8-channel audio mixer is designed for audio-for-video applications. It features built-in 3-band EQ and low-cut filter, as well as 2-way operation (a.c. or external d.c. 12V).
Price:
On request.

MX-P61VU is a 12-channel audio mixer. It is equipped with 12 mic/line inputs and 4 line outputs. Features include: built-in 1 kHz test tone for precise level setting, high-cut and low-cut filters for convenient bandwidth limiting, and a.c./d.c. operation.
Price:
$10,675.00

SOUNDCRAFT ELECTRONICS (JBL PROFESSIONAL PRODUCTS)

TS12 is a 24-track in-line recording console with each channel having electronically-balanced mic/line preamps, 4-band parametric EQ and 6 aux. sends with source selection. Each group module includes separate stereo effects return. Can be fitted with TS12 Automation System. VU or LED metering optional.
Price:
range from $33,100.00 to $51,100.00

Series 6000 is 16- or 24-bus recording console with optional MIDI computer retrofit. Each input has 6 independent sends and 4-band EQ with 2 sweepable mids. Features include PFL and true solo-in-place, low crosstalk routing matrix, silent electronic muting. Consoles available in 16-56 inputs.
Price:
ranges from $16,300.00 to $48,875.00

Series 8000 is a live console featuring 48-band parametric EQ, 8 aux. sends, direct routing, VCA subgrouping, LED input metering or three-way panning option for theater use. Consoles range from 16- to 40-channel inputs.
Price:
ranges from $18,785.00 to $41,500.00.

200B/VE is an 8-24 input recording consoles with balanced lin/mic inputs, assignable input channels directly to the four subgroup buses and stereo mix bus. Four aux. sends selectable as pre-EQ/pre fader, or post fader. Has 48V phantom power, individual channel inserts, balanced left/right output separate with control room output.
Price:
ranges from $5,290 to $12,230

SOUNDTRACS-PLC SAMSON TECHNOLOGIES CORP.

ERIC production console is a 24 bus split configuration digitally routed console with a high analog specification that is available in 32, 40 and 48 input sizes. It has the following features: software controlled features which include routing, muting, mic/line switching, PFL, in place solo, and record ready function (facilitating automated drop ins), all may be synced to time code in addition the 24 groups duplicate the line inputs providing up to 72 inputs for mixdown.
Price:
On request.

In line series contains a 32 bus console primarily suited to multi-channel recording with functional facilities for mixing. Dual line inputs are provided in addition to a mic on each channel which contains 8 aux. sends, 6 mono and 1 stereo. Master muting facilitates the pre-programming of 2 groups of channel mutes. Mix noise is better than –82 dB. Patchbay is standard.
Price:
On request.

PC series MIDI mixer is an "in-line" mixer available in 16 and 24 input configurations with 16 or 8 bus outs. It includes MIDI muting on all inputs, monitors, masters, and auxiliaries, which may be programmed or written in real time or as up to 100 patches. Dual line inputs enable effects to be returned to the channel allowing up to 56 channels on remix. External function allows changing of MIDI patches on remote equipment from console.
Price:
On request.

Prism is a 24-bus inline with computer automation of channels and auxiliaries. Dual line inputs as well as a mic input on each channel, 4-band parametric EQ, 8 aux inputs, monitor fader assignable as a group fader.
Price:
On request.

MRX is a traditional split console available in 24, 32, 40, 26 w/patchbay and 34 w/patchbay frame sizes. Full 16 track metering and monitoring, 4-band EQ, switchable pre/post fade, illuminated solo and mute switching.
Price:
On request.

STUDER/REVOX —See our ad on page 15

A779 is a compact mixing console with 6 mono and 6 stereo inputs, 1 aux bus, l2 main outputs.
Price:
$3,995.00

961/962 is for remote recording and broadcast production. Features include up to 16 inputs, 4 master outputs, 2 aux. inputs, an optional editor interface is available, 3-band EQ on each input, compressor/limiter on outputs. A variety of metering options are available.
Price:
ranges from $12,500 and up

963 is a recording console available with 16 to 56 inputs, up to 8 subgroups, 2-4 masters, 3-band EQ on each input, compressor/limiter on outputs. Alternate input modules, metering, monitor mixers, and machine remotes are available.
Price:
$35,000 and up

900 series consoles can be configured for on-air TV, production use, or multi-track recording. Features included 12-60 inputs with 3- or 4-band EQ, mono or stere inputs, stereo master, GML or Mastermix automation is available, outputs have compressor/limiters.
Price:
$42,000 and up.

SUNN

MX4100 Series stereo sound reinforcement mixing console is available in 8, 12, and 16-channel versions with 3-band channel EQ, 2 aux./efx sends, phantom power, gain and pan control.
Price:
MX4108—$829.99
MX4112—$1,039.99
MX4116—$1,249.99
MX4200 Series stereo sound reinforcement mixing console is available in 12 and 16-channel versions with 3-band channel EQ, 3 aux./mon. sends with pre-post assignments, gain and pan controls dual in/out channel patch, cue, phantom power.
Price:
MX4212—$1,449.99
MX4216—$1,749.99
PX2100 Series stereo powered sound reinforcement mixing console is available in 8, 12, and 16-channel versions with 250 wattsx2 at 4 ohm, 3-band channel EQ, 3 aux./mon. sends, dual in/out channel patch, dual 10-band graphic EQ.
Price:
PX2108—$1,649.99
PX2112—$1,849.99
PX2116—$2,149.99

TASCAM, TEAC PROFESSIONAL DIVISION
—See our ad on pages 6 thru 9

M-700 Recording/Production console contains the following features: 40-channel I/O console, 32 groups, quad (dual stereo) mix buses, 12 aux. send buses (2 stereo, 8 mono), 4-band EQ, high/low sweep, mid high, mid low parametric, variable high-pass filter, 3 group mutes, integral patchbay and producers desk, automation ready.
Price:
$69,999.00

M-600 Series modular recording/production console contains the following features: 16 group, stereo mic modular console that may be configured with any combination of mono or stereo input modules, and a choice of single or dial monitor modules. It has 4-band EQ, 8 aux. send buses, 2-dedicated effects returns. Minimum configuration is 16 mono inputs with 16 tape returns. Maximum configuration 32 stereo inputs with 32 tape returns. Third party automation available.
Price:
$12,999.00

MM-1 Keyboard mixers has 1 stereo line input, and 16 mono line inputs each with a trim control with 40 dB of gain. There is included 2-band EQ on each channel, small rack mount or table top operation, 4 effects sends, 4 stereo returns, direct outs on each channel, individual channel mute and solo, group mutes through 99 scene presets, MIDI control of individual and group muting.
Price:
$1,099.00

M-106 mixer has 6 stereo line input channels and 6 outputs, 2-band EQ, 1 aux. send bus, 8 mic inputs, RIAA EQ on line inputs 1 and 2, 1 effect return.
Price:
$699.00

M-200 Series PA/recording consoles are compact 1 group stereo mix consoles with 8, 16, or 24-input channels, 3-band EQ, 2 aux. send buses, 3 effect returns, PFL and full monitoring. M-208 and M-216 have 8 tape returns. The M-224 has 16 tape returns. The M-208 can be rack mounted.
Price:
$1,199.00 to $2,599.00

M-3500 Series recording/PA consoles are stereo mix consoles. There is a choice of 24 or 32-input channels, 8 tape returns in the dedicated monitor section, separate mic/line trim controls, 3-band EQ, 4 aux. send buses, 2 effect returns, full solo system with AFL or PFL, phantom power.
Price:
$7,499 to $8,499.00

M-300B series are for recording or sound reinforcement and offer 8, 12, or 20 input channels, 5 submix systems, 8 tape returns and 3-band EQ.
Price:
Dependent on configuration.

M-520 recording consoles have 8 group stereo mix consoles with 10 input channels, 16 monitor returns, 4 aux. sends.
Price:
$6,999.00

YAMAHA MUSIC CORPORATION OF AMERICA, PROFESSIONAL AUDIO DIVISION
—See our ad on Cover III

PM3000-24/32/40C Series is available in 24, 32 or 40 inputs and has the following features: 8 group buses, 8 aux. buses (each pre/off/post) and separate stereo bus, VCA assignable grouping with 8 submasters with automation interface, 8 bus muting master system with safety override. The XLR inputs are differentially balanced with 34 dB trim and 5 position pad for optimizing gain structure. There is also a 4-band parametric EQ with variable high-pass filters on each input plus 2-band EQ on the 4 stereo aux. returns, a 11 x 8 mix matrix, insert point selectable in/out on each input, extensive cue and solo system, a comprehensive talkback system with full intercom capability, VU metering, phantom power individually selectable on each mic input.
Price:
PM3000-24—$33,500.00
PM3000-32—$38,500.00
PM3000-40C—$44,500.00

PM2800M-32/40-C series is available with 32 or 40 inputs and the following features: 8 group buses, 4 aux. buses, signal assignment for the input channels via level control, 8 master mute groups with mute assign switches, 4 matrix mixes with level for all 8 channels, stereo L & R level and master. The XLR inputs are differentially balanced with 34 dB trim and 3-position pad for optimizing gain structure. There is also a 4-band parametric EQ with variable high-pass filters on each input, extensive cue and solo system, comprehensive talkback system with full intercom capability, VU metering system, phantom power individually selectable on each mic input.
Price:
PM2800M-32—$31,500.00
PM2800-40C—$36,000.00

PM1200-16/24/32 have 16, 24, or 32 inputs, 6 primary mixing buses, 3-band EQ with sweepable midband, 4 mute groups, stereo input channels, 4 aux buses, switchable pre-/post-fader, interface permits 2 consoles to be connected together.
Price:
PM1200-16—$6,000.00
PM1200-24—$7,500.00
PM1200-32—$8,500.00

DMP7 digital mixing processor has all digital mixing and signal processing with analog inputs/outputs. Features include 3 onboard DSPs (Digital Signal Processors), digital 3-band parametric EQ on each channel, preset memories: 32 internal, 67 external via supplied RAM cartridge, motorized multi-function faders, digital stereo output compressor, MIDI control of preset changes and parameter manipulations, 4 bar-graph meters and LCD parameter read-out. A digital cascade input/output ties multiple units together.
Price:
On reqest.

DMP7D digital mixing processor has all digital mixing and signal processing with digital inputs and outputs, 3 on-board DSPs, digital 3-bank parametric EQ on each channel, preset memories: 32 internal, 67 external via supplied RAM cartridge, motorized multi-function faders, digital stereo output compressor, MIDI control of preset changes and parameter manipulations, 4 bar-graph meters and LCD parameter read-out. A digital cascade input/output ties multiple units together.
Price:
$5,995.00

DMP11 digital mixing processor has all digital mixing and signal processing with analog inputs/outputs, 2 on-board DSPs, digital 3-band parametric EQ on each channel, preset memories: 32 internal, 67 external via supplied RAM cartridge, digital stereo output compressor, MIDI control of preset changes and parameter manipulations, 4 bar-graph meters and LCD parameter read-out. A digital cascade input/output ties multiple units together. This unit is a rack mount size.
Price:
$2,395.00

M406 sound mixer has among its features: 6 channels, with 3-band EQ and 6-position input level controls, high gain (84 dB) for full output, stereo program output with L & R master controls, echo/effects send bus with master send control, 2 effects inputs, each with level and pan control, dual illuminated VU meters with peak indicators, right VU meter and headphone output switchable to monitor program or echo output, rack mountable with front panel power switch.
Price:
$1,395.00

MC1204/1604/2404 series is available with 12, 16 or 24 inputs. It has the following features: 4 program mix buses, 2 effects buses, 2 foldback buses and a cue bus. Each input features a pad, gain control and peak LED for precise gain matching. There is also 4-band EQ with the two mid-bands featuring quasi-parametric control, foldback 1 and 2, and ECHO 1 and 2 strappable pre/post EQ, channel ON (mute), is post cue for proper cue monitoring, complete talkback system, illuminated VU meters, each with peak LEDs, phantom power available for mic inputs.
Price:
MC1204—$2,695.00; MC1604—$3,295.00; MC2024—$4,395.00; MC2404M—$4,395.00 (stage monitor)

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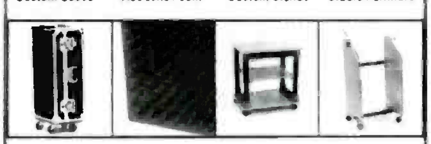

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Stolen: One pair E-V Sentry III Series 2 loudspeakers, missing since March 1988, possibly in MI or OH. Last seen boxed for shipping, one in original box with handmade inserts, one in handmade box. New condition. One has small piece of missing veneer lower left rear corner. Substantial reward for recovery. **Jim DeClercq, 17144 Dawn, Roseville, MI 48066 or call (313) 772-4687, (313) 825-5309 days**

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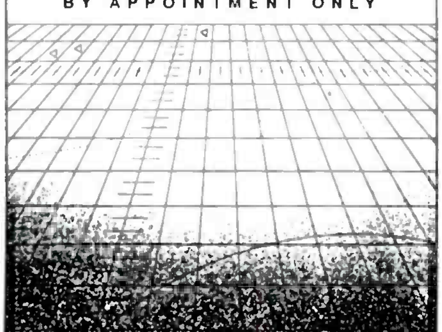

Closing date is the first of the second month preceding the date of issue.

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People, Places... & Happenings

• The **Adelaide (Australia) Festival Theatre** has recently taken delivery of several new ARX EQ60 Constant Q Dual ⅓rd Octave Equalizers. House Engineer **John Matherson** extensively tested all currently available Equalizers and chose the EQ60s for their very low noise and constant Q performance. Sydney FM Radio Station 2MMM has also recently acquired two ARX EQ60 Constant Q Dual ⅓rd Octave Equalizers. 2MMM chose the EQ60s for their low noise and found their Constant Q performance ideal for cleaning up the response of several of their more troublesome studios. 2MMM intends to update the remainder of their studios with EQ60s in the near future.

• **Courtney Spencer** has been named vice president, Professional Audio Division, as announced by **Sony Communications Products Co.** Spencer will oversee all sales and marketing operations and activities of the Professional Audio Division. He will also work closely with and coordinate the efforts of the product management and service groups in the Sony Communications Products Co.. Spencer replaces **Osamu Tamura** who is returning to Japan to assume a new position in international marketing for the Sony Professional Products Group.

"My goal is to ensure that Sony continues to improve as a company to deal with," Spencer said. "I want our dealers and our customers to know that they can expect to see a high level of quality and support from Sony that's unparalleled in the industry." Prior to his new appointment, Spencer was vice president of Sales at the WaveFrame Corp., Boulder, CO., for the past two years. He was also vice president and general manager of Martin Audio Video Corp., a major United States professional audio dealership.

• **David Yake** of **David Yake Enterprises**, and the executive staff of the **Stratford Shakespearean Festival Theatre** are pleased to announce that Tannoy's dual-concentric loudspeakers are being installed in Stratford's 2400-seat Festival Theatre. A custom-coupled cluster of six Tannoy B50s will serve as the Festival Theatre's main sound reinforcement system. Suspended above the Festival's unique Shakespearean "thrust" stage, the Tannoy B50s will address a horizontal coverage of almost 240 degrees. In addition to the B50 cluster, the Festival will be utilizing a Tannoy T300 sub-woofer for effects. The Avon Theatre has also acquired a T300, several B100s and B50s for their effects fan-out system.

• The **Indiana University School of Music** is now offering a four-year Bachelor of Science degree in Audio Recording. This is in addition to the two-year Associate of Science in Audio Technology degree, offered since 1981. The new program seeks to fill many students' needs for a four-year Bachelor's degree and addresses the musical aspects of recording, particularly the special requirements for recording classical music.

David A. Pickett, chairman of the audio department and director of Recording Arts, has bachelor's degrees in music and electronics. His doctorate was awarded for research on the orchestral conducting activities of Gustav Mahler. Pickett has been actively involved in professional recording for 20 years, including nine years as a recording engineer at EMI's Abbey Road studios in London. **Wayne Jackson**, manager of Audio Operations and assistant professor of music, has degrees in chemistry and math, and has experience in advanced telecommunications. Mr. Jackson has been active in the Audio Technology program since 1978, serving as an instructor and sound designer for many IU Opera Theater productions. His sound reinforcement background includes work for artists as diverse as Dizzy Gillespie, Queen Ida, Jay Leno and Richard Thomspson. Recording credits include Focus Records releases for the IU Early Music institute and four years as producer for the "Live from Bloomington" project, an annual benefit for the Bloomington-based Hoosier Hills food bank. Admission to the new degree program is competitive and will be based on music and scientific skills and interests, and aptitude for recording studies in serious music. Students will be assessed thoroughly during their first exploratory semester, and constantly evaluated thereafter. The exploratory semester has been in use for several years as a means of selecting students for the associate degree programs, and has proven most helpful. Full details of the admission procedure may be obtained by writing the Audio Department, School of Music, Indiana University, Bloomington, IN, 47405, USA.

• **Sandra Hale**, 10-year advertising veteran, has joined **Studer ReVox America (SRA)** as a PR/Copy Writer. For the past six years, Hale has written numerous assignments for the broadcast and print media. In making the announcement, SRA President **Tore Nordahl** said, "Her recent affiliation with broadcast and studio product as well as her strong writing and creative skills will undoubtedly be a great asset to our company."

A congregation comes to church for inspiration. To be uplifted. Moved.

All can be lost if the people in back can't understand the sermon. Or hear the music the choir has rehearsed all week.

Hence, the EMX Series Powered Mixers from Yamaha. Models come with 6, 8, or 12 inputs and include everything you'll need to make sure the congregation gets the message in the spirit it was intended.

Like digital reverberation that will add a touch of majesty to the music. A graphic equalizer so that the system response can be finely tailored and the feedback controlled. And a low distortion, high-power amplifier section that will help deliver a more powerful message.

Put all these components together in a single unit, and you've got a sound system that's almost as easy to operate as it is to listen to.

Just contact the authorized Yamaha Sound Contractor or Professional Audio dealer nearest you and ask about the EMX Series Powered Mixer. You couldn't ask for more.

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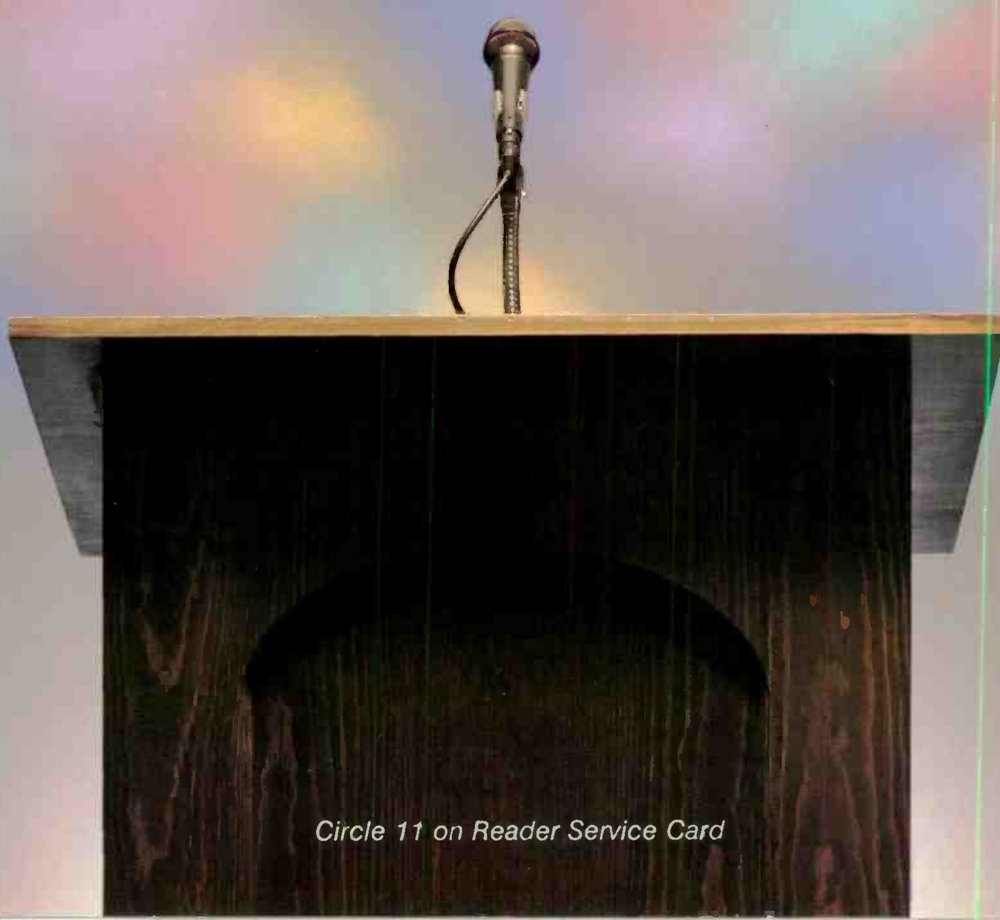

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