Design Concepts for Mixing Consoles for Multi-Channel Reinforcement Systems for Music and Theater

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DESIGN CONCEPTS FOR MIXING CONSOLES FOR MULTI-CHANNEL REINFORCEMENT SYSTEMS FOR MUSIC AND THEATER

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ABSTRACT

Over the years, a design concept has evolved for performing arts centers which includes true three-channel sound reinforcement from clusters located left, center, and right at a proscenium arch, with each cluster covering most or all of the audience seating area. Generating a viable three-channel mix for such a system using a currently available mixing console which is a standard product is problematic at best.

This paper presents the functional design criteria for a new console which meets the mixing needs of such systems. The work is based on research with clients and other users, as well as the author's experience. A complete functional description of a console is presented, including functional diagrams, along with a thorough discussion of the rationale underlying all operational concepts.

OVERALL SOUND SYSTEM CONCEPT

The possibilities for reinforcement systems in Performing Arts venues have just begun to be exploited. At the time of this writing, sound reinforcement in the vast majority of venues, even the most prestigious ones, is done through monophonic systems. Ahnert [1] and Sobel [2] have demonstrated complex time and precedence-based localization of sound implemented with the help of computer-aided mixing and panning. It is an impressive system, but is well outside the budget of most venues and productions. Some touring systems, including Alembic's Grateful Dead system [3] in the 1970's made use of separate stacks to amplify each instrument and vocal.

Full stereo reinforcement is a reasonable and realizable design concept in many venues. The author has designed and operated a number of such systems on both permanently installed and rental bases. While accurate localization is not attainable in all seats, a very

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1 Ahnert, Wolfgang; "The Complex Simulation of Acoustical Sound Fields by the Delta Stereophony System DSS"; Paper Presented at 81st AES Convention; November, 1986

2 Sobel, Norbert; "The Delta Stereophony System: A Multichannel Sound System to Achieve True Directionality and Depth"; Conference Paper presented at AES 6th International Conference; May, 1988

3 Davis, Don and Davis, Carolyn; "Sound System Engineering"; pp 193-4, 198; Howard W. Sams & Co.; Indianapolis; 1975

pleasing stereophonic spaciousness can be achieved for all listeners who can hear both left and right loudspeaker systems.

Experience in the motion picture industry has clearly demonstrated the value of a centrally located loudspeaker system in addition to left and right channels. A hard center image is generated and localization is improved. The Dolby surround system, now a standard in modern motion picture theaters, makes use of such an arrangement, as has nearly every other major motion picture multi-channel system that preceded it. Surround effects are provided by a single rear channel fed to loudspeakers around the perimeter of the seating area.

Theatrical venues require a more flexible effects playback system, capable of providing localization to nearly any part of the hall. In most halls, this is implemented in the form of permanent or portable effects playback loudspeakers and wiring at various elevations around the perimeter of the hall and at catwalk level. The loudspeakers are connected to power amplifiers via a loudspeaker patch bay. The power amplifier inputs are routed to a patch bay. A suitable matrix mixing system feeding these power amplifiers can accomplish a wide range of effects.

All of these systems benefit from the support of a sub-woofer capable of reproduction in the 20 - 80 Hz range. The sub-woofer may be utilized as a separate effects channel, or may be fed from a sum of left, right and center mix busses.

PROGRAM DEPENDENT MIXING REQUIREMENTS

Musical programs benefit greatly from three channel systems. Stars and soloists are usually mixed to the center channel. Backup vocals and an orchestra or group may be either wrapped around the soloist in stereo, or fed by a full three-channel panned mix. Once a mix is established for musical group, it is often relatively static. Adjustments may be limited to balancing solos or a performer who doubles on another instrument. The surround effects system is rarely used.

Theatrical events are often much more complex. In non-musical theater, the system often does little more than amplify fixed and wireless mics and play back simple sound effects. The wireless mics may be locked into the central cluster (motion-picture style) or combined into an active three-channel panned mix that follows actors around the stage. Effects may be as simple as a distant off-stage door slam or as complex as the sound of birds and other creatures gradually interrupted by an approaching and passing thunderstorm at a forest picnic.

Musical theater combines the musical complexity of a live band with the theatrical complexity of live actors and surround sound effects. Some productions may require two or more operators to keep up with live actors, a pit band, and sound effects cues. Each element of the mix may be quite dynamic, totally changing with each change of scene. Multiple stage configurations and multiple orchestra setups are not uncommon.

The console may be located in an ideal seat where the operator hears all three clusters, under a balcony where he or she hears only a monophonic and delayed underbalcony system, or in an isolated control room where noisy tape recorders may be operated without disturbing the audience seated nearby. Monitoring facilities must be provided to allow the console to be used efficiently in all three situations.

Non-musical, non-theatrical events (speeches, panel discussions, and the like) pose a much different problem. Here, only a few microphones may be required and the services of a sound operator may be outside the budget of the client for such uses. Simplicity in the form of automatic or pre-set manual monophonic mixing of a few mics to the central cluster is the order of the day. The console which is the subject of this paper is the one designed for theatrical and musical uses.

FUNCTIONAL OVERVIEW

The mixing requirements of these various uses are quite different, but there are a number of common elements. The main house mixing console in use for a musical or theatrical performance needs to be capable of generating a number of separate mixes simultaneously. The most common requirements are:

1. Left, center, and right mixes for the audience
2. A separate send to the sub-woofer
3. Up to four mixes for foldback to performers (also known as stage monitors)
4. Up to four sends to effects devices, often in stereo pairs
5. A stereo mix for recording, audition, or broadcast
6. An eight by eight line-level matrix for sound effects playback

Many of the mixes that will be generated can be quite complex, and can change quite rapidly. It is common in such venues for a single console to be used by as many as a dozen different operators in a one week period, for applications as different as a lecture, a Broadway musical, a local rock band, an opera, and a jazz quartet. Only one or two of those operators will be familiar with the console and the venue, so signal flow and operation will need to be very simple to understand. Very few travelling operators will have experience with three-channel panning, so its operation must be so similar to conventional two-channel stereo panning as to be functionally indistinguishable.

Conversely, the house sound operator who will the be major user of the console will know it very well but will be mixing a new show (or sometimes even two) every night. The console must give that operator the ability to quickly generate, fine-tune, and troubleshoot a mix of a band or performance he or she has never heard. It can do that by providing the signal flow and monitoring which is specifically designed to allow the operator to dissect and study a mix as thoroughly and as quickly as possible.

If the console is to be useful to the operator, it must be very logical in its layout, be very clearly labelled, possess all of the functions the operator is likely to need, and be as flexible as possible by providing a number of optional ways to accomplish the same function, so that if one way is in use for another function, other options are available. It must be rugged so that it can withstand the constant moving from one mix position to another on an almost daily basis. It must have excellent performance specifications so that it does not degrade overall system performance by adding audible noise, distortion, or crosstalk.

SIGNAL ROUTING AND PANNING

Development of the left, center, right, and sub-woofer mixes (the house mix) is the single most important function of the console. Every time the console is in use, it will be fulfilling this function. Second in priority are the sound effects playback matrix and mixing for stage monitors. When more complex stage monitoring requirements exist, a separate console and operator will be used. Other functions will be utilized only part of the time, or will be used to maximum capability less often. Complex effects and recording mixes are occasional uses.

THREE-WAY PANNING

The left, center, and right components of the house mix should be generated by a three-way pan pot. Three channel panning should function as follows: When the pan control is set to the center, the center channel only should be fed. When the pan control is set to the left, the left output only should be fed. When panned to the right, the right output only should be fed. As the pan control is operated through its range from left to center, the output should change smoothly from left to center. As the pan control is operated through its range from center to right, the output should change smoothly from center to right. For all settings of the pan control, the signal should appear in only the adjacent two outputs and the sum (i.e., the root of the sum of the squares) of those two outputs should equal the input signal within ±1.5 dB. The center should be indicated by a detent. The approximate midpoints between left and center and between right and center should be indicated by a panel legend. An ideal panning rate would be a constant number of dB per degree of pot rotation for the center 80% of rotation, with the pan going hard left and hard right respectively and center disappearing completely at the left and right extent of its travel. When a pan should be turning a buss hard off, the feed to that buss should be at least 30 dB below the buss which is on.

THREE-CHANNEL SUB-MIXING

Monophonic sub-mixing has no place in mixing for two-channel or three-channel reinforcement systems. Each sub-mix in these applications should consist of a mix of mics set up to generate a two or three-channel stereo sound field. The resulting mix has the potential to be an exciting, spacious, and open sound. String, and horn sections, percussionists, and vocal groups keep their life by maintaining their stereo sound field. By contrast, monophonic sub-mixes limit the operator to gimmicky ping-ponging one-dimensional mixes of drums to one cluster, brass to another, vocals to another, etc.

The house mix should be supported by at least three three-channel sub-mixes, a direct to left, center, right delegation, and a post-fade effects send to the sub-woofer. These sub-mixes might be used on one day to sub-mix live actors in one left, center, right sub-mix, a pit band's melody instruments in another, and a rhythm section in the third. On another day, the first left, center, right sub-mix would handle vocals, the second melody instruments, and the third a rhythm section. One yet a third day, the first left, center, right sub-mix might contain a string section, another horns, and a third the rhythm section. The possibilities are limitless. The common element is that each left, center, right sub-mix is a three-way panned mix which needs to be controlled as a group and which may need signal processing as a group. The operator must also be able to listen to each of these sub-mixes as a three-way panned group, individually and in combination.

An important implication of this requirement for stereo mixing is the limitation it places on expansion mixing. One common use of expansion mixing is to mix many mics on a string section for a large pit orchestra. If an input module is to be developed to bring many mics into the mix with simple equalization that is common to all, each mic in that expansion mix must be panned independent of every other mic in that mix. Any expansion mixer which does not allow this will be unacceptable in its ability to generate an acceptable stereo sound field. Taking eight mics, mixing them monophonically, and panning the resulting mix to some stereo perspective generates a dull, dry sounding, static mix.

Patch points are needed on all sub-mixes. It is often helpful to be able to provide additional equalization for gain before feedback on a sub-mix of vocal mics, or special level control on any of the sub-mixes.

CONVENTIONAL TWO-WAY PANNING

To allow maximum flexibility, each input should be switchable to either two-channel and three-channel pan capability. The two-channel pan function should be a conventional stereo pan pot, with outputs to the left and right channels of each three-channel sub-mix.

ADDITIONAL MIXES - SUB-WOOFER, EFFECTS, AND FOLDBACK

SUB-WOOFER

In the operation of large sound systems, one of the easiest ways to get in trouble is to lose control of low frequency energy. A useful design philosophy says that the main reinforcement system should be high-passed at a frequency on the order of 80 Hz, and a conscious decision made by the operator to send specific channels to the sub-woofer via a dedicated effects send. If the effects playback system is also high-passed at the same frequency, the sub-woofer system is compatible with both systems, and the low frequency sections of both systems are capable of higher levels. For this reason, a single post-fader effects send should be available for use as a send to the sub-woofer. Of course, the reinforcement system should be designed with appropriate crossovers complementary with the loudspeakers, their locations, and the total system. One element of this system is the high-pass filters.

EFFECTS AND UTILITY MIXES

In support of the house mix, the operator may need to generate utility mixes to feed to signal processors such as reverberation generators and special effects devices. At least two effects sends are required, or one stereo pair and one or two monophonic sends. At least one stereo and two monophonic returns should be provided that can access the house, stage monitor, and effects busses.

FOLDBACK MIXING

At least four pre-fade mixes are needed for use with stage monitors (foldback). In most venues, the house mix console will be used to generate a foldback mix about half the time the console is in use. These mixes should be post-equalization and post-insert. Nearly all of the equalization that is required on an input channel is required on all mixes generated from that channel, including foldback mixes.

There are rare occasions when a stage monitor send might need to be operated pre-insert but still post equalization. An example of such a situation is when a star mic needs to be compressed, but the soloist should not hear the effects of the compression in his stage monitor mix, which is being generated at the console. A switch or jumper should be provided on the input module to allow this mode of operation.

The author cannot conceive of any real-world applications where it is ever more useful for any mix from a reinforcement console to be pre-equalizer, and yet consoles continue to be built that way. Imagine, for example, a console with the only pre-fader sends being pre-equalizer and the operator trying to satisfy the simple requirement of generating a stage monitor mix with simple bass or high frequency rolloff on the star mic.

EQUALIZATION

In support of all of the mixes, the operator should be able to selectively adjust the frequency response of any input to all outputs to compensate for problems unique to that input. Among such problems are mic characteristics, mic placement, the nature of the sound produced by the instrument or performer utilizing that input, and wild sound (i.e., mic leakage) picked up by that microphone. Minimum equalization should be:

1. Boost/cut high and low frequency shelving with adjustable hinge points
2. One or two bands of full boost/cut parametric equalization
3. Sweepable (20 - 350 Hz) or selectable (50 Hz, 100 Hz, 250 Hz) two-pole high pass filter.

PATCH (INSERT) POINTS

In support of all of the mixes, the operator should be able to insert external signal processing such as de-essers, compressor/limiters, and gates on any input channels and sub-mixes. The insert point should follow the channel equalizer section, so that the operator may use the equalization to minimize false-triggering of effects devices (limiters, gates, etc) by undesirable microphone pickup and responses (mic popping, low frequency leakage, etc.). The insert point should be pre-fader, so that the operator is riding gain on an externally processed signal.

MIXING AIDS

The operator needs the ability to quickly switch from one pre-set mix to another. Attenuators or gain controls on pre-amplifier inputs should be detented to allow accurate pre-setting. It is common in many venues for a single console to be used with two sets of inputs for two musical acts playing alternate sets, or for multiple scenes in a theatrical production. Detented controls facilitate returning to settings which have been determined during sound checks. Level control should be sufficient to move the input overload point from -24 dBu to +18 dBu in seven detented steps of 6 dB each. (0 dBu=0.775 v)

Separate A and B mic inputs on each channel expedite switching between multiple scene or band setups. Separate input attenuators for A and B inputs are desirable for the same reason, but may be too costly to implement.

Each input needs a balanced line input on a connector separate from that used for the mic input. Line inputs will often be run through a patch bay to maximize flexibility, but it is undesirable to route mic inputs through that patch bay. Long term reliability is a lot to ask of patch bays operating at mic level. Input attenuators for line inputs which are separate from mic inputs are desirable, in that they simplify pre-setting of setups that involve line inputs.

Group muting of inputs allows quick changes between scene presets. VCA grouping is helpful if it can be accomplished without degrading signal to noise performance. VCA grouping is not a substitute for separate sub-mix busses, however, for two reasons. First, the sub-mix cannot be soloed. Second, the sub-mix cannot get its own signal processing via a common patch point (or set of three patch points).

MONITORING AND SOLO FUNCTIONS

Effective and flexible monitoring and soloing are two of the most powerful tools built into a good console. The ability to listen to one input channel at a time (the solo function) allows the operator to troubleshoot a mix. It is also a very effective learning tool, allowing an operator to develop a practical understanding of mic placement, mic leakage, off axis sound, and the components of a good mix. The following functions are the important elements of effective soloing:

* Post-fade solo should be provided on input channels. This configuration is far more powerful in troubleshooting an active mix than pre-fade solo. Examples of troubleshooting may include looking for buzzes or excessive low-frequency leakage that may be getting into several mics. Post-fade solo will quickly tell the operator which inputs are most significantly affecting the mix. In-place soloing (i.e., where the output of the pan-pot is directed to the monitoring output) would be useful, but not absolutely necessary.
* Pre-fade solo of input channels should also be possible. Many operators do not understand post-fade solo since so few contemporary reinforcement consoles provide it.
* Channel muting must allow the solo function to remain active, to allow cueing of tapes and previewing of mic inputs.

The capability to listen to one sub-mix at a time allows the operator to fine-tune a complex mix by breaking it down into more manageable elements. Because a sub-mix in this console consists of left, center, and right channels, the monitor function must allow the operator to listen to the sub-mixes in left, center, right groups. In addition to the solo functions described above, the monitor section should incorporate the following capabilities:

* The monitor section should allow selected outputs of the console to be fed to stereo headphones and power amplifiers for left, center, and right control room loudspeakers. Separate level controls should be provided for control room and headphone outputs.

* A switch should be provided to force the headphone and control room monitor outputs to a mono sum of what is selected to feed them. This allows the operator, who may hear all three channels, or may be located in an isolated control room, to monitor the monophonic signal that is being sent to auxiliary systems such as underbalcony loudspeakers. The principal importance of the mono switch is to check for mono compatibility problems caused by poor mic placement or out-of-polarity mics.

* The headphone amplifier should have at least two outputs, each capable of providing 110 dB to modern studio quality headphones having impedances between 30 and 600 ohms and sensitivities on the order of 95 dB/mw. The output impedance of the headphone output should be less than 30 ohms. Separate level controls on the two outputs allow two operators to work with different headphone levels and/or with headphones of differing sensitivities.

* It is very important for the operator to be able to listen independently to each three-channel and/or stereo sub-mix. This should be accomplished with push-buttons such that any combination of sub-mix can be monitored by the operator <u>in two or three-channel stereo</u>. The sub-mix monitor function should be available in stereo headphones as well as in three monitor speakers in the control room. For the headphone mix, the center channel should be combined to the right and left ears. The level controls for the headphone and control room loudspeaker outputs should be independent of each other.

* It is often important for the operator to listen to the production intercom system during a performance and rehearsal. Wearing an intercom headset makes mixing almost impossible, and using a loudspeaker to monitor the intercom is not practical when the console is located in the audience. A transformer-coupled monophonic input to the monitor section of the console should be provided to allow the operator to listen to production intercom via his headphones or control room loudspeakers. This input should be provided with its own level control and positive action on/off switch. The input sensitivity should be 0 dBu for full output to headphones and control room monitor line outputs.

* A stereo pair of line inputs to the monitor system should be provided. This function allows an operator to monitor a stereo tape recorder fed from an effects buss, or an off-air feed of a stereo broadcast of the event.

* Using the headphone and control room outputs, the operator should be able to individually monitor each and every output buss of the console. The operator should also be able to monitor the effects busses in stereo pairs.

A stereo pair of mic preamplifiers should be provided for use with the monitor system. These inputs should be provided with phantom power, gain trim, simple screwdriver adjustable high and low frequency equalization, and a patch point. A front panel switch should allow selecting these preamplifiers to feed the control room or headphone outputs. These preamplifiers can then be used with dummy head microphones or other suitable microphones located in the audience to allow an operator mixing in a control room to hear sound in the hall. The output of these preamplifiers needs a level control with a detented position to indicate correct calibration.

A level indicator should be provided for each and every main output, group output, effects output, foldback output, the sub-woofer output, matrix output, and the solo buss. A single set of indicators could be switched between the matrix busses and the effects send busses. The indicator should allow monitoring of both average and peak levels, and should conform to international standards for VU and/or PPM response. A useful implementation is a continuous VU display, with a PPM-calibrated LED set to flash 6 dB below maximum output. Experience has shown that by virtue of its time constants, a PPM-calibrated meter will indicate signal levels which are 6 dB below actual peak levels in the circuit.[4] All level indicators should be clearly readable in light levels ranging from darkness to brightly lit rooms. Where the performance venue is out of doors, indicators should be visible in bright sunlight.

HOW MANY INPUTS? - PROVIDING FOR EXPANSION

Contemporary musical and theatrical productions are becoming increasingly complex. Few facilities can afford a really good house mixing console with more than 32 inputs, but shows that require 48 or more inputs are not uncommon. To accommodate this fact, the console should be provided with external input to all busses to allow an external console to provide additional inputs when required. In addition to all the mixing busses, the solo buss and the solo control buss should also be extended. A second solo control input should be provided to allow an external console having either positive or negative logic to be used.

This approach is quite economical, since when many mic inputs are required for a given production, it is common for many of those inputs to have relatively simple mixing requirements, so that a much less sophisticated console (fewer mix busses, less equalization, etc) will usually suffice for the expansion unit.

THE SOUND EFFECTS PLAYBACK MATRIX

A major departure from conventional thinking about console signal flow occurs with the sound effects playback matrix. A conventional console output matrix is usually implemented by being fed by the main console output busses. For every matrix input that is used, one input module and one group must be removed from use with the main mix and

[4] Hoffner, Randy; "Audio Program Metering in the 1980's: The Work of th IEEE Audio Measurements Subcommittee; SMPTE Journal, August, 1989; pp. 590-593

devoted to the matrix. This is quite wasteful of some very expensive busses and input modules for a function that is little more than a routing matrix for line level signals.

The sound effects playback matrix should really be totally separate from the busses used to generate the house and effects mixes. Although it should be possible to access the matrix inputs by patching from the groups, effects busses, or an input module's direct output, most consoles will rarely need to be used in that manner.

It is not out of the question that the matrix function might best be implemented in a totally independent console or mixing system. Effects playback mixing for complex shows might be done by a second operator, or supported by a computer controlled system. Richmond [5] and Isobe [6] have presented systems that could be useful in such an application.

The sound effects playback matrix does not usually require equalization or pre-amplification, but it does need level control. Equalization of sound effects can usually be done at a pre-production stage when they are recorded on tape.

MISCELLANEOUS SUPPORT FUNCTIONS

TALKBACK

The operator needs to be able to use a microphone at the console to talk to performers during rehearsal, and may also need to make announcements to the audience during performances. A separate preamplifier for this microphone allows the functions to be fulfilled without tying up a full-capability microphone input. This input should be provided with minimal screw-driver adjustable equalization (high and low frequency boost and cut) to allow correction for proximity effect and excessive microphone presence. It should also provide phantom power. The talkback function should access the house center channel to talk to the audience, the stage monitor busses to talk to the stage, and the effects send busses to allow the operator to use his mic to perform a rough setup of effects devices without the assistance of performers.

PHANTOM POWER

48 volt balanced phantom power, per DIN 45 596, is required. Phantom power should be switchable on each input, so that it may be eliminated from troublesome lines or lines with defective mic cables.

MODULE DIRECT OUTPUTS

Pre-fade and post-fade direct outputs of input modules have been provided on consoles for a number of years. Their major usefulness is for applications where a microphone must be preamplified and can be routed without tying up a buss.

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5 Richmond, Charlie; "Computer-Controlled Systems in the Performing Arts"; Conference Paper presented at AES 6th International Conference; May, 1988

6 Isobe, Masahiro and Mochimaru, Akira; CSEAS - "A Computer-Aided Sound Effects System"; Conference Paper presented at AES 6th International Conference; May, 1988

THE PATCH BAY

An optional patch bay can facilitate the rapid reconfiguration of the console and the sound system. The following ports should be routed via the patch bay:

Channel line inputs
Channel direct outputs
Channel insert points
Sub-mix insert points
Mix insert points
Left, center, and right outputs
Effects and utility buss outputs
Control room outputs
Solo outputs
Buss inputs for use with an extension console

The following system ports should be routed to the patch bay:

External signal processing equipment
Tape recorders
External line level sources
Tie lines
Sound system inputs

Patch bays should be implemented in 1/4" ring-tip-sleeve jacks rather than Bantam plugs. Patch bays in these facilities get relatively little use as compared to studios, making long term reliability a problem. Patch bays should always be mounted as close to vertical as possible to minimize the buildup of dust, dirt, and assorted liquids.

The patch bay should be located with the mix console, but should be capable of being mounted alongside or under the console. This allows the mix position to displace fewer audience seats or take up less space in a cramped control room. Because of the custom nature of the requirements and the interface to external equipment, it may be more efficient for patch bays for reinforcement consoles to be fabricated by the installing sound contractor than the console manufacturer.

ERGONOMIC CONSIDERATIONS

Good layout of the console can go a long way to making it faster to operate and reducing operator errors. Different sizes, shapes, and colors of knobs attached to controls can reduce confusion about which control is which when there are many rows of controls. Slight irregularities in spacing of controls (like locating some off center in opposing directions, for example) are also helpful. Illuminated LED's of different colors and shape can unambiguously show the condition of pushbutton switches that might otherwise be difficult to see in low light.

Most often used controls should be located nearer to the operator (i.e., the bottom of the console). Controls should also be grouped in a logical fashion. A preferred layout based on the author's experience is to locate the pan control, solo button, and channel mutes near the channel fader, with auxiliary sends (effects and foldback) just above them. Of these, the sub-woofer and foldback sends are likely to be the most used, so the sub-woofer should be closest to the operator, with the foldback sends just above it, followed by effects sends. Just above the sends should be equalization, with high-pass and low frequency closest to the operator, parametrics next, and high frequency farthest away. Equalization logically goes near the top of the module, since, in the author's view, good mixing utilizes less equalization, and because equalization is relatively static in a mix once established. Finally, at the top of the module, input gain trims, polarity inversion, and phantom power switching.

A write-on labelling strip (or space for the operator to place masking tape) should be provided somewhere between the channel fader and the effects sends. The strip should be on the order of 15-20 mm high, and should run the entire width of the console. Sub-mix and send masters also need labelling in this fashion.

It is common for operators to need to mark many control settings on a daily basis, using masking tape or other expedients. Permanent labelling of console functions and control calibration should be done with paints that are resistant to commonly used adhesives and solvents used to remove those adhesives.

Consoles in performing venues are often moved, and are often subject to abuse. In addition to the dirt, dust, and chemical pollutants found everywhere, the most common forms of abuse are mechanical banging around and the spilling of assorted liquids onto the console. Mechanical design of the console needs to accept this use as a reality, and use construction techniques which minimize the damage from such hazards. Sealed pots and switches, and pots mounted to the panel but connected to circuit boards by wire jumpers are more expensive to implement but make the console less prone to severe damage or failure. Channels, bushings, and other means to route spills away from faders and connectors can be effective yet add little expense. A slight slope to the horizontal mixing surface is quite helpful in allowing the operator to see more of the console more easily, and making it more likely that spilled liquids will flow off the top surface as opposed to dripping into the console.

ELECTRICAL CHARACTERISTICS AND PERFORMANCE SPECIFICATIONS

GENERAL

All controls should be labelled to indicate their optimum, calibrated, or flat setting. Equalizer controls should be detented at their flat settings. These indicated settings act as a guide to the operator in properly utilizing the headroom of the console, and are the "normal" settings referred to in the performance specifications.

The console should comply with international industry standards with respect to input and output connectorization. All inputs and outputs should be available on 3-pin XLR connectors, with pin 2 high, pin 3 low, and pin 1 shield. Patch points should appear at 3-circuit 1/4-inch ring-tip-sleeve connectors, with both inputs and outputs balanced, wired tip high, ring low.

INPUT STAGES

All inputs should be balanced, and should be compatible with both balanced and unbalanced loads. Input impedance of all line inputs should be at least 50 K ohms. Microphone inputs should be bridging to 150 ohm microphones and their impedance optimized for best signal-to-noise.

Maximum input level should be at least +26 dBu at all line inputs, +16 dBu at all microphone inputs, and +24 dBu at all patch points.

OUTPUT STAGES

All console outputs should have an output impedance of 60 ohms. The left, center, right, effects send, and monitor send outputs should be equipped with transformers. It is not uncommon for the console to be separated from the main amplifier racks or from tape recorders and effects devices they will feed by several hundred feet of cable. Contemporary researchers have determined that this impedance is a close match to the characteristic impedance of modern audio cables.[7, 8]

All outputs should be capable of driving a parallel equivalent load of up to 0.1 microfarad in parallel with no less than 500 ohms resistance with no degradation of performance specifications other than the voltage divider loss of the termination on the output impedance of the buss, and should produce no overshoots or rising frequency response greater than 1 dB when viewed with a measurement bandwidth of 1 MHz.

Levels at patch points and console outputs should be compatible with optimizing signal to noise and headroom with professional components. Most external signal processing equipment is optimized for an operating level around +4 dBu. Patch points and other outputs which operate below that level compromise system noise performance. All buss outputs should be +4 dBu with an indicated level of 0 VU. All patch points should be +4 dBu when the signal path they interrupt is correctly adjusted for its optimum noise and headroom.

Maximum output level at all outputs except patch points and module direct outputs should be at least +26 dBu for load impedances of 5,000 ohms or greater, and at least +24 dBu for load impedances of 500 ohms or greater. Maximum output at patch points and module direct outputs should be at least +24 dBu for load impedances of 5,000 ohms or greater and at least +22 dBu for load impedances of 500 ohms or greater.

FREQUENCY RESPONSE

The response of the system outside the audio range is important, including input and output transformers. The output of modern microphones contain much energy below 20

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7 Bytheway, David; "Wired for Stereo"; Broadcast Engineering; September, 1986, pp 22 - 32

8 Jensen, Deane; "Long Line Application"; Application Note, Jensen Transformers, Inc.; H. Hollywood, CA.; 1987

Hz and above 20 KHz. Ringing, overloads, and instability outside the audio range can and do produce very audible distortion products inside the audio range.

Frequency response, referred to 1 KHz, should be ±0.5 dB between 20 Hz and 20 KHz, and no greater than + 1 dB at any frequency, measured between any input and any output.

The console should be unconditionally stable for any condition where outputs are not connected to inputs.

All equalization should be minimum phase, and should not degrade the signal to noise performance of the console except to the extent that it modifies the frequency response of the signal path it is equalizing.

No inversion of signal polarity should occur between any input and any output, including patch points. A polarity inversion switch should allow for correct of polarity errors external to the system. From a signal flow point of view, this switch should be located before the channel patch point.

DISTORTION

Non-linear distortion should not exceed its rated value for any input signal level which is at least 2 dB below the specified maximum input level and for any output level which is at least 2 dB below the specified maximum output level. Rated Total Harmonic distortion should be less than 0.01% at all frequencies between 50 and 10,000 Hz, and should be less than 0.05% at all frequencies between 20 and 20,000 Hz. Rated SMPTE Intermodulation distortion should be less than 0.05%. The conditions for measurement of distortion should be the injection of the test signal at any input and the measurement at any output with gains set at their indicated "normal" operating levels and input level controls set consistent with good engineering practice for the level of signal injected.

NOISE, CROSSTALK, AND INTERFERENCE REJECTION

Crosstalk and signal to noise specifications should be commensurate with a facility where peak sound levels can reach 113 dB and the background level is NC 15. The crosstalk of a sound effect being cued through a muted input that has leakage only 80 dB below full output can be disastrous.

The equivalent input noise of microphone inputs should be no greater than −128 dBu, unweighted, measured with a bandwidth of 20 – 20,000 Hz and a 150 ohm source.

When output level controls are set for normal operation and no inputs are delegated to any output, the noise level at any buss output should not exceed −85 dBu, unweighted, measured with a bandwidth of 20 – 20,000 Hz.

When output level controls are set for normal operation and one input is delegated to any output, the noise level at any buss output should not exceed −85 dBu, unweighted, measured with a bandwidth of 20 – 20,000 Hz.

Crosstalk from any undelegated or muted input to any output should be less than -85 dBu for any input signal level below maximum input, unweighted, measured with a bandwidth of 20 - 20,000 Hz.

Common mode and radio frequency interference rejection should be designed for the real world rather than the laboratory. Common mode voltages can often reach 20 volts or more where multiple mix consoles and feeds between them are in use. RF fields of 5 volts/meter exist in performance venues in some large cities, and higher levels can occur when a walkie-talkie is operated next to the console (a common occurrence). Careful attention should be paid to maintaining balanced input impedances of differential inputs. Input transformers should be used when required to provide adequate common mode rejection. Good engineering practice with respect to shielding, filtering, and grounding for rf rejection should always be followed.

The console should be immune to interference from unmodulated, amplitude-modulated, or frequency-modulated electromagnetic fields of less than 5 volts per meter at any frequency between 60 Hz and 600 MHz. The console should be immune to interference from unmodulated, amplitude-modulated, or frequency-modulated voltages of less than 20 volts peak-to-peak containing any frequency components between 50 KHz and 600 MHz applied to any input or output. "Immune" means that the console should meet all performance specifications in the presence of such interference, and there should be no audible products of the interfering signal.

Common mode rejection at all balanced inputs should be sufficient to prevent any common mode voltage of less than 40 v peak-to-peak from degrading performance specifications.

The console should be immune to electrical noise on the AC mains lines of up to 500 volts peak-to-peak. "Immune" means that the console should meet all performance specifications in the presence of such interference, and there should be no audible products of the interfering signal.

LEVEL INDICATORS

Level indicators which read average levels should conform to the standard which defines VU meters (IEEE 152-1953 (R-1971)/ANSI C16.5-1954 (R-1961).[9] Level indicators that read peak or quasi-peak response should conform the standard which defines PPM meters (EBU 3205).[10] Reference level for all indicators should be calibrated to within ±0.5 dB of the specified output level of the console.

POWER SUPPLIES

The 48 volt phantom power supply should be sufficiently well regulated that the open circuit phantom voltage at any microphone input does not fall below 44 volts when 11 milliamperes of phantom current is simultaneously drawn from every other microphone input, and when input voltage from the AC mains ranges between 105 and 125 volts RMS.

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[9] "IEEE 152-1953 R-1971)/ANSI C16.5-1954 (R-1961) Recommended Practice for Volume Measurements of Electrical Speech and Program Waves"; IEEE; New York

[10] "EBU Tech 3205-E (English Language Version) Second Edition, The EBU Standard Peak-Programme Meter for the Control of International Transmissions"; EBU; Brussels, Belgium; 1979

All console performance specifications should be met for all AC mains voltages between 105 and 125 volts AC.

The console's power supply should be approved or listed by a safety agency recognized by the regulatory agency having jurisdiction where the performance facility is located.

FUNCTIONAL DIAGRAMS

Figures 1 through 6 make up the functional diagram for a console which meets most of the design criteria outlined. In these drawings, no attempt is made to get "inside the circuitry" at the circuit design level. Rather these drawings are intended to communicate to the console designer the needs of the user as viewed from outside the console. Where gain stages, switching, routing, and controls are shown, they are shown to clarify functional concepts.

Similarly, technical specifications are viewed functionally from outside the console. The author acknowledges that less stringent noise, output level, and distortion specifications can reduce the cost of the console significantly. Unfortunately, the demands of the performing environment dictate these specifications, not the author.

INPUT MODULE

Figure 1 shows an implementation of the input module. A single mic input receives switchable 48 V phantom power and is routed through a switchable attenuator and mic/line selector switch to the preamplifier. By suitable wiring of a multi-level rotary switch, the attenuator and pre-amplifier gain trim control should allow input gain to be varied in 6 dB steps over a 48 dB range. The transformer is optional, depending on whether it is required to meet performance specifications.

From the preamplifier, the signal is routed through a polarity inverter, the high pass filter, and the equalizer section to a normal-through patch point (or insert). The insert return feeds the input of the linear channel fader, which in turn drives two- and three-channel pan pots, a module direct output, and the post-fade effects busses. Although the panning system is shown with switching at its input, the switching may occur at either input or output of the pan pots.

The output of the pan pots drives an interlocked switch which allows the channel to be delegated directly to the main output mix busses or to either of the three sub-mixes. Note that three-channel pan drives left, center, and right busses in the selected sub-mixes or the main output, while the two-channel pan drives only left and right busses in each sub-mix or the main output. The most practical implementation of the delegation switch is a four push-button interlocked multi-pole switch, with each button selecting a left, center, and right combination of busses.

The foldback, effects, and sub-woofer sends are shown as being individually switchable between pre- and post-fader operation because most available consoles come that way and because the function gives the maximum flexibility for alternative uses of the busses. As a cost compromise, the foldback sends will always be pre-fade, the effects sends nearly always post-fade, and the sub-woofer send always post-fade.

The solo buss is shown with a circuit-board jumper for post-fade, which is the preferred operating mode. Some operators prefer pre-fade solo, so a circuit board jumper or panel-mounted switch should be provided. In any event, the mute relay should mute all sends and group outputs while allowing the solo function to monitor the module. Muting is shown with control in eight groups. This is an optional function that has great value in practice.

A peak-indicating LED is shown monitoring the critical points for overloads. It should have PPM response and be calibrated to flash 6 dB below circuit clipping for a steady stage input.

OUTPUT MODULES

Figure 2 shows the output circuitry. In all outputs, the summed buss is routed through a normal through patch (insert point) to the master fader. Main and group outputs should utilize a linear fader (or a triple ganged linear fader for each left, center, right sub-mix and the main output). Where ganged faders are utilized, tracking should be maintained to within ±1 dB throughout the operating range.

All outputs have a level indicator having VU characteristics and a PPM calibrated peak flasher. Master and send outputs are equipped with output transformers. Group outputs rarely leave the console, and thus do not require transformers. The solo function on the effects send outputs allow operator monitoring of the buss.

EFFECTS RETURNS

Figure 3 shows possible implementations of monophonic and stereo effects returns. An alternate variation of the stereo return would allow switching a mono sum of the left and right returns to the main center channel and to the center channels of the sub-mixes. An alternate implementation of the monophonic return might allow the output of a left, center, right pan pot to access the main channels and the sub-mixes. In both alternates, the effects returns to the sends would remain unchanged.

EFFECTS PLAYBACK AND SUB-WOOFER MATRIX

Figure 4 shows an implementation of an effects playback matrix. The system is really two matrices, one a pre-selector routing matrix composed of interlocked switches and the other a proportional mixing matrix composed of potentiometers.

Effects playback sources enter the pre-selector matrix via XLR connectors at the top or from the left. Those entering from the top are assigned to a matrix mix buss by choosing the appropriate horizontally-interlocked selector switch. Those entering from the left are normalled through the matrix to the matrix mix buss when no switches on their horizontal row are selected. Each matrix mix buss may thus be used as a "pre-set" for a given sound effect, distributing it to any combination of matrix outputs.

The top inputs to the pre-selector matrix are intended for use with tape recorders as sources, where a series of sound effects will be recorded and distributed to a number of different combinations of loudspeakers during the course of a show. The left inputs to the pre-selector matrix are intended for applications where a console's mix buss or direct outputs of input modules might be patched directly to the matrix for complex feeds to the effects playback system.

The matrix output modules are essentially the same as the other output modules, but receive their signal from the matrix mix busses rather than the effects busses. Monitor function is shown pre-fader.

The top inputs to the pre-selector of the effects playback matrix may also be fed to the sub-woofer. The sub-woofer output module is similar to the output modules for other busses, but has more inputs. It is fed by the console left, center, and right outputs, the matrix outputs, and the sub-woofer effects buss (#9). The output module has a level indicator and can feed the solo buss.

MONITORING

Figure 5 shows the required function of the console's monitor system. A bank of non-interlocked selector switches chooses the inputs to each summing matrix so that any panned combination of the main and sub-mix busses may be monitored "in-place". The switches should be arranged in a 3 x 4 matrix, so that groups 1, 2, and 3 are at the top, 4, 5, and 6 below them, 7, 8, and 9 below them, and the main left, center, and right below them. Next below (or above) them should be an 4-button interlocked switch to select between the output of the 4x3 switch, the stereo effects send busses, the external monitor line inputs, or the hall monitor inputs.

At some non-confusing location, a positive action pushbutton should force the headphone and control room outputs to the monophonic sum of the selected L, C, and R inputs. When the control room output is forced mono, a circuit board jumper should allow the monophonic signal to be routed either to the center channel only or to the left and right speakers only.

The headphone output is shown having only one pair of output amplifiers and level control but with separate isolation resistors for two pairs of headphones. A more ideal arrangement would call for two pairs of headphone amplifiers, each with its own level control.

An input is provided for monitoring a production intercom system. It is assumed that the interface to the intercom system will be accomplished by driving the console from the headphone output of an intercom user station, so that the console manufacturer need not be concerned with the intercom system interconnection protocol, but simply provide an input transformer and level control for isolation.

The preamplifiers for hall monitor microphone inputs should be of high quality, but need only phantom power, screwdriver-adjustable high and low frequency equalization and gain trim, and a level detented level control. The patch points allow for more extensive equalization to account for acoustic anomalies that may exist in the room, or for distribution of the pre-amplified mics for recording or feeds to dressing rooms and lobbies.

TALKBACK

Figure 6 shows the talkback system. A single mic input receives phantom power, is preamplified, and provided with screwdriver adjustable equalization and gain trim. It will be utilized for talkback to the stage during rehearsals, as a signal source to effects sends during system setups, and announcements to the audience during performances. Input gains should be such that an input overload point of at least -10 dBu can be obtained.

ACKNOWLEDGMENTS

The author would like to acknowledge the contribution of Jim Gundlach, who articulated and helped develop the effects playback matrix. Rick Talaske, Larry Kirkegaard, and Jim Gundlach have worked with the author to implement the system concepts in real performance spaces. David Kimm, Chris Foreman, Steve Wooley, Arnold Toshner, Gary Snow, and Sam Spennacchio have all put up with the author's ramblings on the subject and provided input on the practicality (and lack thereof) of his ideas. Pete Tappan reviewed the manuscript. Thanks are due to them all.

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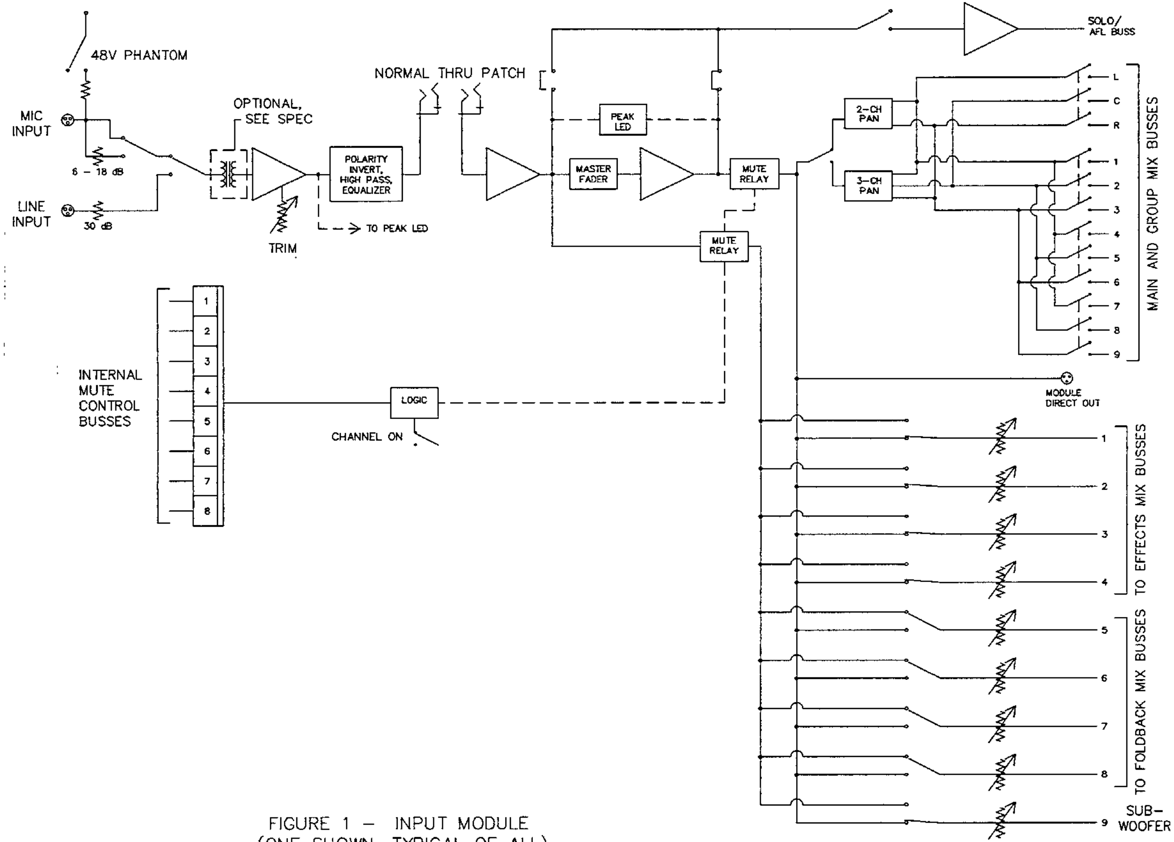


FIGURE 1 – INPUT MODULE
(ONE SHOWN, TYPICAL OF ALL)

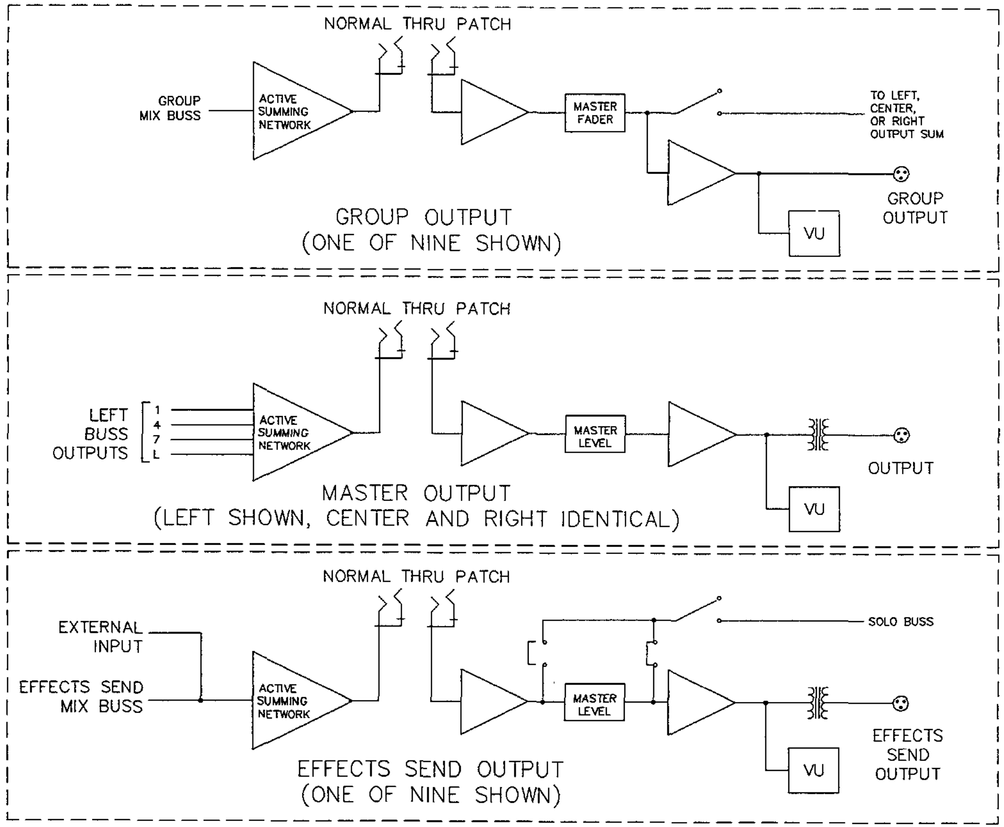


FIGURE 2 – OUTPUT MODULES

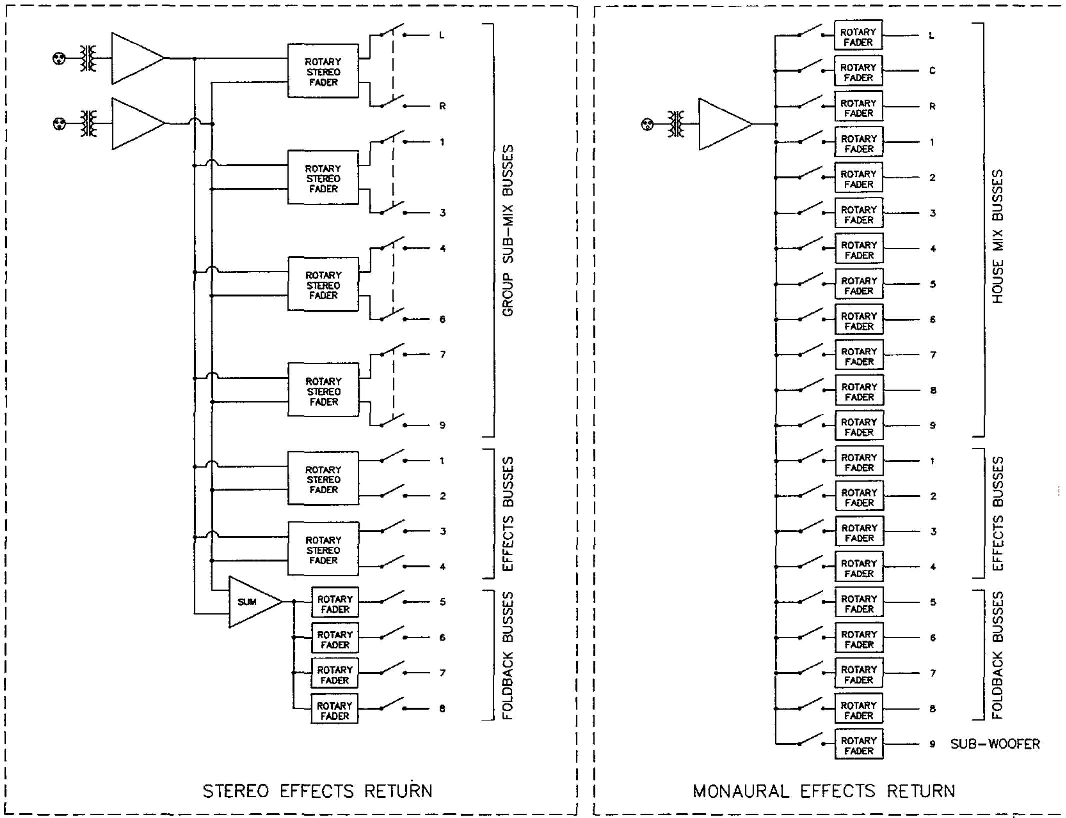


FIGURE 3 – EFFECTS RETURNS

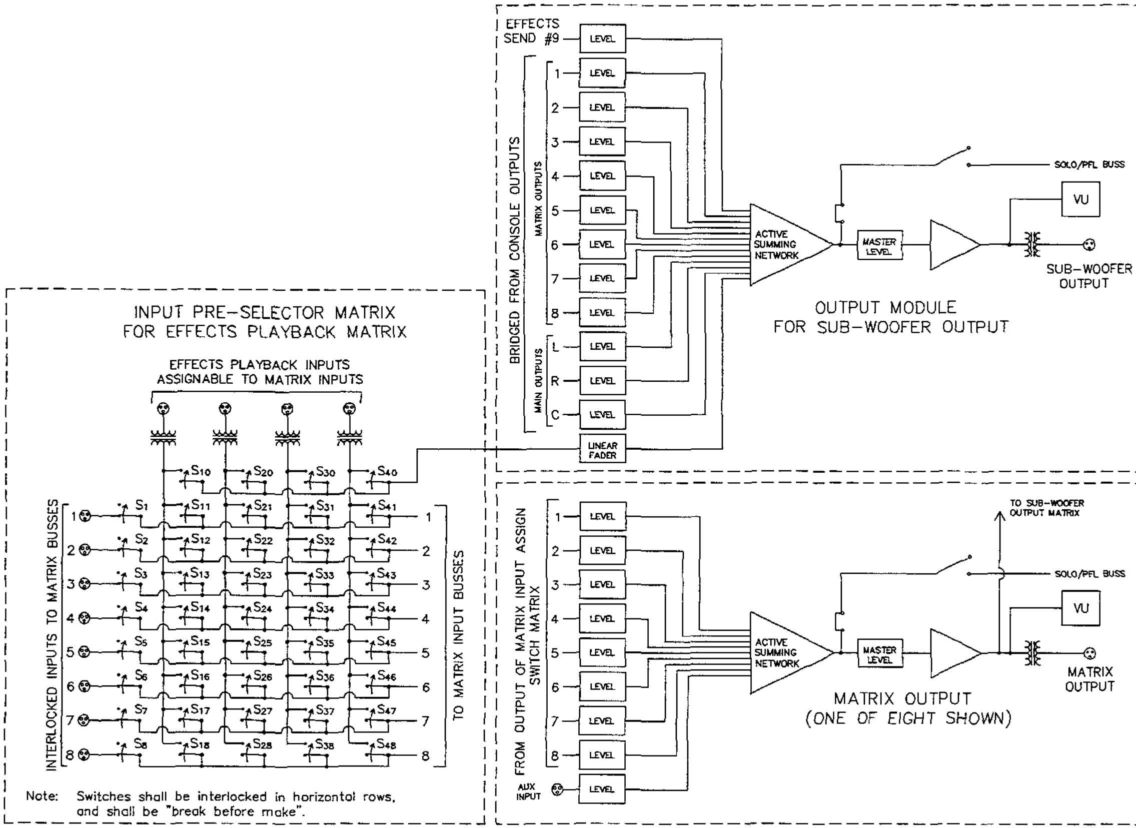
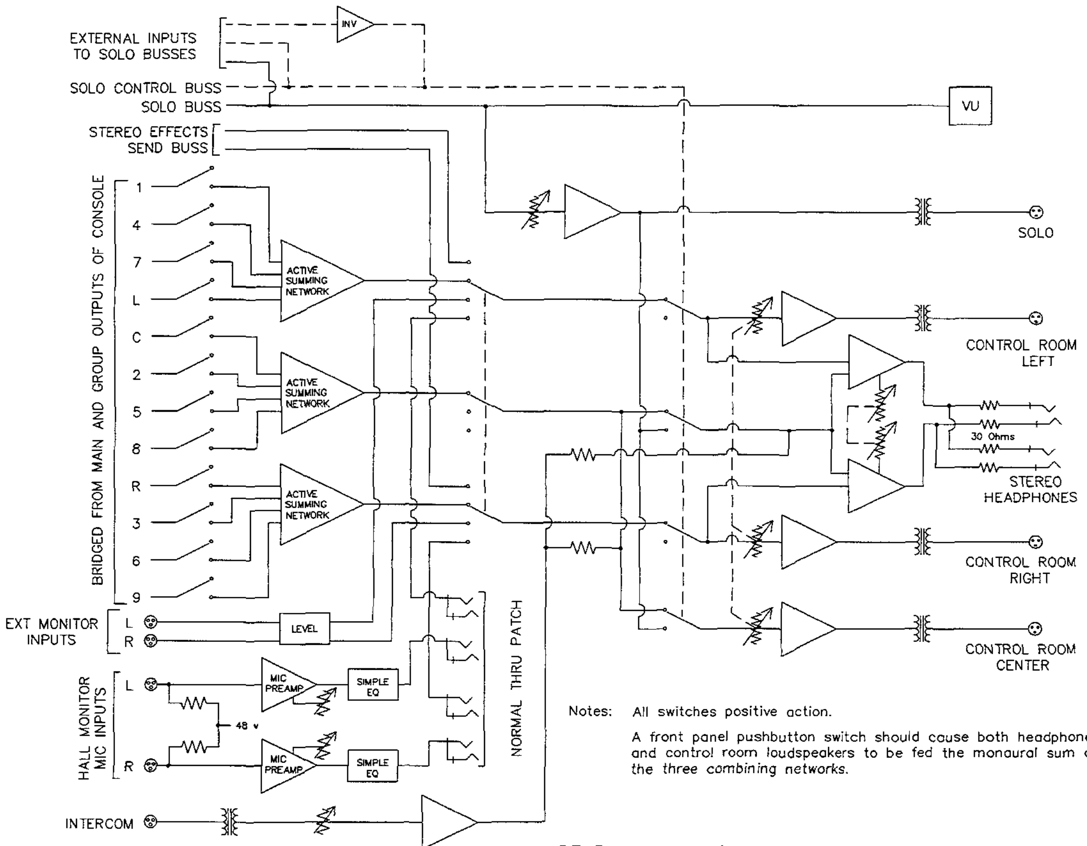


FIGURE 4 – EFFECTS AND SUB-WOOFER MATRIX SYSTEM



Notes: All switches positive action.

A front panel pushbutton switch should cause both headphone and control room loudspeakers to be fed the monaural sum of the three combining networks.

FIGURE 5 – MONITOR SYSTEM

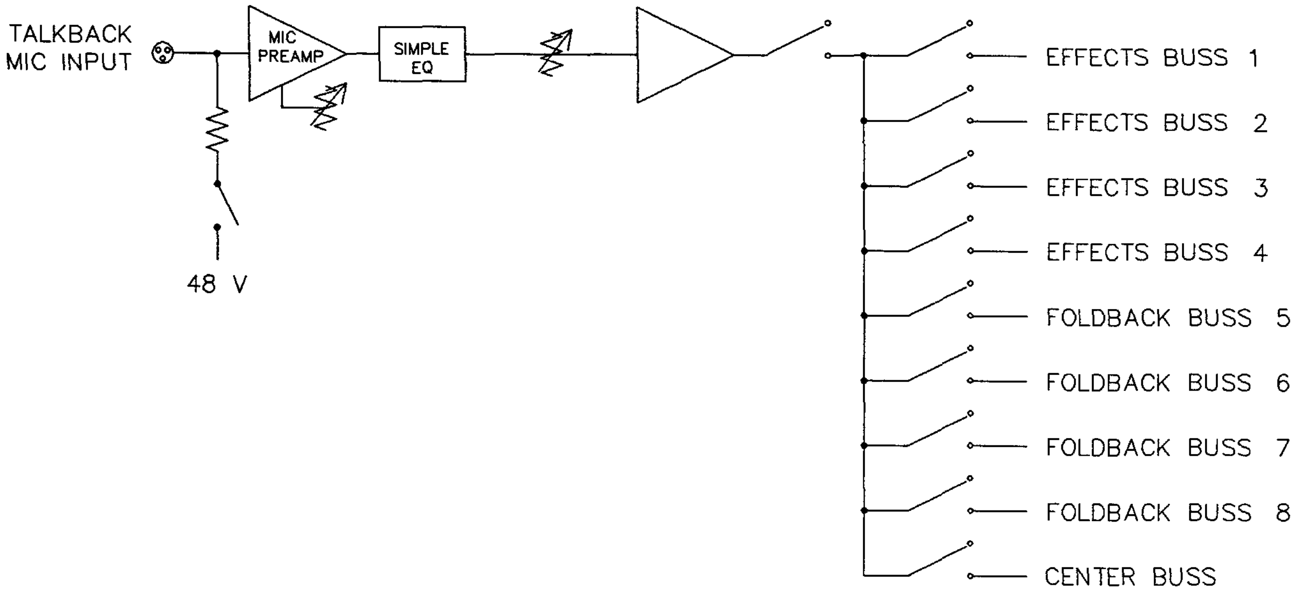


FIGURE 6
TALKBACK SYSTEM