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info@researchersjournal.org
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Practical Technical Design Solutions for Sound and Effect for Nigerian Live Theatre and Media Paradigms

Apeh Columba A., Lilian A. Okoro & Nancy Ernest Irek
Department of Theatre and Media Studies, University of Calabar-Nigeria
apehcolumba@gmail.com, apehcolumba@unical.edu.ng, apehcolumba1976@gmail.com,
Phone: +2348105722745; https://orcid.org/0000-0001-7107-3512
apaciafrica@yahoo.com, Phone: +234803624721; https://orcid.org/0000-0002-3434-8385
nancyirek@gmail.com/ nannest@yahoo.com,+2347033503179, https://orcid.org/0000-0001-5463-0276

Abstract

The main points of this research paper are to discover dependable practical technical design solutions for sound and sound effects for Nigerian live theatre and media productions. The research design is survey method in order to discover solution for sound and effects in theatre performances. The findings of this research are that considering the challenges facing technical theatre in Nigeria-poor funding, architectural limitations and technical installation mayhem, both technical directors and technical students need forums where they can share ideas and solutions to specific sound and sound effects production challenges. This paper concludes that this work will be of great value to the technical directors and technical students whenever a production problem arises concerning managing sound and effects. This work will be the first resource to which technical students should turn to for practical solutions.

Keywords: Technical Design, Practical Solution, Sound Effects, and Live Theatre

Introduction

Since man first inhabited this planet, it has been one long struggle for survival between him and nature. Man has had to live and to find his place in the universe. In the process man has left behind traces of his achievements at various levels of his development, and the cumulative knowledge of his various achievements in sound, music, dance and dramatic performance constitutes his total way of existence from primordial epoch to the in the 21st Century Theatre and Media practice. According to Apeh on this backdrop writes that,

> Sound is a normal part of human life, accompanying human beings everywhere they go and in everything they do. They themselves make sounds as they move and live. The objects they create make sounds and other animal and they elements make sounds. Sound is so much a part of life, in fact, that absolute silence, like loss of movement is associated with death (1).

In life, human beings to some extent create their personal psychological environment by hearing sound and music selectively. In theatre and Media productions, the artistic and technical directors have the same control not only in the selection of sounds to be heard, but control over a particular quality of sound or music and over the relationship of sound to sound and sound to action

Based on the communicative values of sound and sound effects, theatre and media productions rely almost entirely on sound and visual components for meaningful results. The audience comes to the theatre to see and hear the performance, which is the daring meaning and aesthetics from the stage where the events unfold in a drawing process comprising of all other theatrical elements facilitating. Sound and sound effects contribute to meaningful and aesthetic communication in theatre and media productions as asserted by Meyer in his *On Rehearsing Music,* where he distinguished three theories about significant in sound and music as being formal, kinetic syntactic and the referential. They aid to articulate the authentic aesthetic enrichment effected by the reproduction and rehearsal of any good musical number or sound and effects (Adam and Georges 106). In the theatre and media milieu to have or gain perfect audibility and vision of the performance from any legitimate seating and standing, (if applicable) positions in the house must be considered.

In a nutshell, Taylor (39) writes that it is the corporate effort of the production team that makes sound and sound effects a complete necessity in the arts of the theatre and the media. By this, where is creative effort and steady experimentation. When accomplished, the joined effort may achieve the following communicative values in the theatre and media production in view: (1) establish time and place; (2) provide an idea to facilitate the plot of the production; (3) suggest atmosphere or feelings, especially in romantic or crisis scenes. When emotions mount, a sound underlie can support hook the audience on; (4) reproduce physical happenings, spot cues, like clocks striking, babies crying, and car arriving, to mention a few; (5) establish the following conditions – weather, day or night and locale of the action; (6) link one scene to another; (7) reinforce the thematic concept of the performance; and (8) create theatrical aesthetics

Theoretical framework

This research adopts Systems Theory approach as propounded by Ludwig Von Bertantffy which deals with general system theory. He introduced the idea that systems in theatre production experience consist of interrelated parts working together, which is useful for analyzing sound systems in performance environments (57).

The second theory used in this research is the theory of Design Thinking by Herbert A. Simon which centres on the science of the artificial and change by design. The contribution of this is that it frames design as a problem -solving process, which supports developing practical sound and effects solutions. (93).

Designers must therefore be creative and bring out fresh ideas to create originality of concept instead of repetition of ideas in similar assigned task. For instance, when costumes are often used in a particular, theatre company, the audience would not expect any significant or spectacular creativity or innovation in the use of costume, because they had established mental stock image of costume anytime they are coming to watch a play in such theatre. Designers must break that barrier through innovations. Also, as an aesthetic factor, there is the need for oneness (unity) in the visual sch/eme. The design must unite with other visual elements to create a harmonized theatre experience for the audience. This means costume must portray the characters as expected when they are under the stage light and the background set. Characters should not be absorbed by the stage backdrop and light making it difficult for the audience to identify and separate them.

Methodology

This research adopts the experimental research methodology for this investigation. This methodology is most suitable because this a method whereby functional relations, which derived is strictly controlled and contrived conditions, are verified as pertinent and usable in human situations. The Avante Garde theatre has, in its varieties and variations been largely experimental. So have artistic and technical director's approach to creating meaning in performative arts from traditional based productions to modern as well as postmodern African performances of semiotics in theatre practice including acoustic and psycho acoustic development in theatre and media productions, According to Bracht and Glass, the paramount characteristic of experimental research methodology following its definitive context involves making a change in the value of one variable called the independent variable and observing the effect of that change on another variable called the dependent variable (437).

Practical technical sound and effects design solutions for Nigerian live theatre and media productions

Sound design is ongoing throughout the production process in the arts of the theatre and media milieus in preproduction planning, during the runs of production and in postproduction respectively. It is a continuous process that covers the spectrum from music on one end to dialogue at the other end with sound effects in between. The key to integrating it all is that the concept of sound design is to make sure these various elements work together and don't step on each other's toes (Audsley 328).

According to Askenfelt, there are three domains to work with in creating a sound design, speech, sound effects, and music. Paradoxically, silence and the ability of sound to evoke a picture in the mind's eye may be considered two other domains (96). All sound is made up of the same basic components; pitch loudness, timbre, tempo, rhythm, attack, duration, and decay. Determining a sound design involves consideration of how the audience is to think or feel about a particular story scene, character or action, from what point of view; and whether that is to be carried out mainly in the sound design also requires the awareness that doing so is often tantamount to defining a production's conceptual and emotional intent (Sundberg 76). In this research work, nine practical technical sound and effects design solutions are successfully achieved and tested for Nigerian live theatre and media productions through field work and with effective exploration of the embedded potentials in the World-Wide-Web. These practical technical sound and effects design solutions are: how to use your headset as a page mic; amplified-to-line signal converter; two head attachment methods for wireless microphones; a device to determine sound system polarity; marking a prop tape player cue able; a Bench built cable jester; in-expensive digital noise reduction; two-scene sound control; and recreating an acoustic space with discrete-6 recording.

Solution 1: How to use your headset as a page microphone (Mic)

This article describes a simple adapter box that allows an intercom headset with a dynamic mic to function additionally as a page mic. With the adaptation, a Stage Manager can make backstage and in-house pages without having to pick up a second microphone. The device acts somewhat like a direct box in an orchestra pit. The mic signal from the headset is split and then fed to both the intercom belt pack and (through a balancing transformer) to any external amplifier with microphone inputs.

Construction

Any small metal project box can be used as an adapter enclosure. On one end of the box, drill holes for and mount the two connectors that will serve as outputs to your page system. On the opposite end of the box, drill holes for and mount the Connectors and install strain relief for a short cable tail between the adapter and your belt pack. Install the two switches. Mount the transformer in the box. Solder a 12" piece of cable to the connector. Wire the connectors, switch, and transformer as shown in the schematic.

Use of Adapter

The headset plugs into the connector, and the tail with the connector plugs into the belt pack. Normal belt pack to base station connections must be made. Mic cables are then used to attach the two D3M connectors to your page system. The output can be connected to any low-impedance mic input on a page system. The toggle switch is used to select backstage or in-house paging. The momentary switch is used as a push-to-talk (page) button. With this device, all standard headset functions are normal. When you want to make a page, you select in-house or backstage and push the momentary switch.

Solution 2: Amplified-to-line signal converter

The audio equipment in theaters offers enough inputs and outputs to accommodate most production needs. Occasionally, however, it may be necessary to feed an amplified output into a line-level input. The simple solution described here is within the resources of most production organizations. The amplified-to-line signal circuit illustrated below uses a resistor and an audio transformer to convert the low-impedance, high-current output of the amplified signal to the low-impedance, low- current signal required by a line-level input. A switched, pass through ground is included to help eliminate ground loops. Connectors and their wiring are matched to the hardware and polarity of existing equipment.

The resistor, rated for 10 ohms at 1 watt, is connected in parallel to the amplifier's output terminals. If the output of the amplifier cannot be limited to 1 watt or less, a resistor with a larger power capacity must be used.

The audio transformer is also connected in parallel to the output terminals. The transformer must be capable of handling the full audio frequency range without distortion. Appropriate transformers cost ₦200,000 to ₦250,000 and are available from Andy systems in Calabar and similar suppliers.

The converter's operating principle is quite straightforward. The 10-ohm resistor loads the amplifier to keep it operating normally. The transformer performs electrical isolation and reduces the amplifier's output voltage to a level usable as a line-level input signal for semi-professional equipment. While not required frequently, this particular amplified-to-line converter provides an inexpensive solution to what might otherwise become a show-stopping problem.

Solution 3: Two Head Attachment Methods for Wireless Microphones

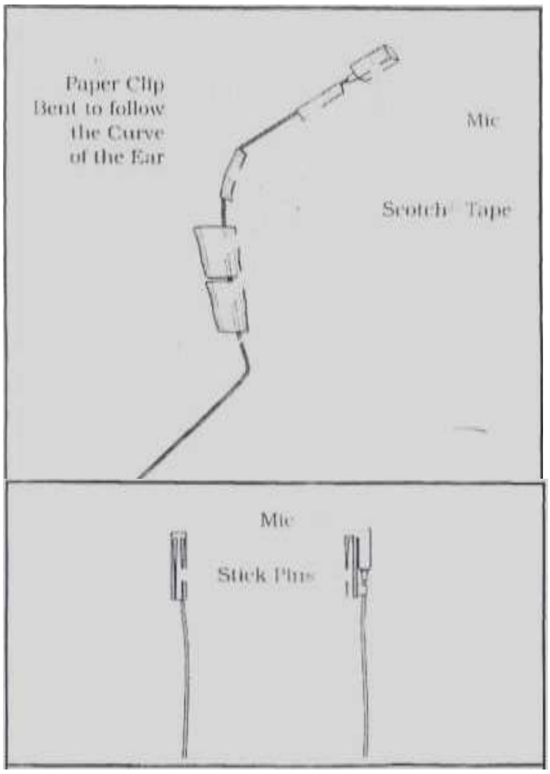


FIGURE 2: A MIC AND VIPER CLAW HEADBAND ATTACHMENT

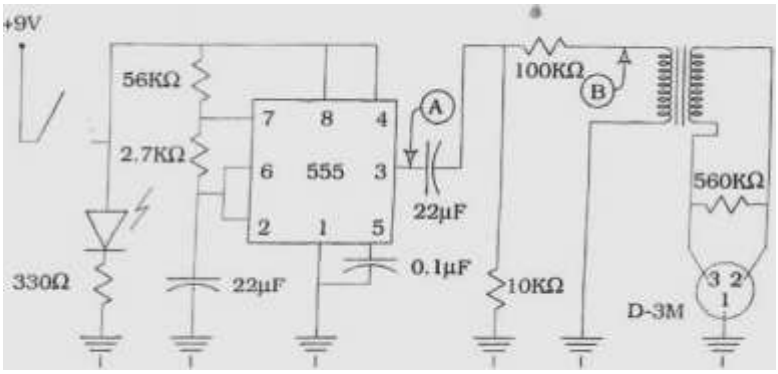
Source: Fieldwork 2020

Sound designers and engineers are often plagued by the problem of securing wireless body microphones to a performer's head, which is important for providing consistent sound pickup. You can hide most body mics fairly well in the hair of the performer. But simply attaching mics to the hair by using bobby pins often provides too little stability. The microphones frequently fall out of the hair or twist the wrong way, resulting in poor pickup for directional microphones.

One good solution to the problem involves providing a larger base for bobby pins to attach to. As Figure 1 illustrates, an ordinary paper clip partly unfolded and molded to fit around the performer's ear creates a stable base for bobby pins. Taping the mic wire securely to a bent paper clip in this manner also helps improve pickup by keeping the microphone from twisting.

A different approach is particularly useful with short or particularly fine hair. Most body microphones come with the kind of viper-claw accessory shown in Figure 2, a plastic microphone holder backed with two stick pins. A mic and its viper claw can be easily pinned to a headband made of a piece of the clear elastic (found in most costume shops), as long as you're careful not to tear the elastic. You can further secure the viper claw by taping it to the headband with Scotch tape or clear Band-Aids. The microphone will be well hidden if worn on the side of the head, just above the ear.

Solution 4: A Device to determine sound system polarity



A1.1 Isolation Transformer. 600-900x (Radio Shack #273-1374)

Balanced Output to Sound-System Input)

FIGURE 3: THE IMPULSE GENERATOR

Maintaining polarity among several speakers is absolutely critical to the successful performance of large-scale public address systems, musical reinforcement systems, page/monitor systems, and the like. The circuit described in Figure 3 provides a reference signal to the input of a sound system, and the circuit described' in Figure 2 decodes that signal as it comes from a speaker, indicating whether polarity through the system has been maintained or reversed. Commercial devices cost upwards of $350, but this device can be built for around $30.

The impulse generator's 555 timer chip generates positive and negative impulses of a known time relationship. Connected to the sound system input, this circuit produces recognizable excursions and incursions at a speaker's cone. The impulse decoder uses an electret mic positioned in front of the speaker to detect those excursions and incursions. The succession of positive and negative pressures is translated into digital pulses, revealing whether an excursion or an incursion happened first. If polarity is maintained, the green LED lights up; otherwise, the red LED lights up. Each circuit is powered by a 9-volt battery and a power-supply circuit. Both circuits and their power supplies can be mounted in small project boxes for portability.

Solution 5: Making a prop tape player cueable

The sound quality of an inexpensive cassette tape player was a key element in theatre production experience. Wanting to make a prop tape player sound as realistic as possible without sacrificing cueing or volume control, Sound Designer Kevin Hodgson came up with the idea of putting a wireless receiver in the tape player that the actors would use and transmitting a signal to it from the sound board. The wireless unit is primarily used as a remote sound pickup for video camcorders. The package contains two parts: a radio transmitter with a small lavalier mic, and a receiver with both line-level and headphone output jacks. I added some parts to make the transmitter take a balanced line input, and built an amplifier to boost the signal from the receiver and send it to the speaker. With advice from electronics specialist Alan Hendrickson, I designed the circuits described below.

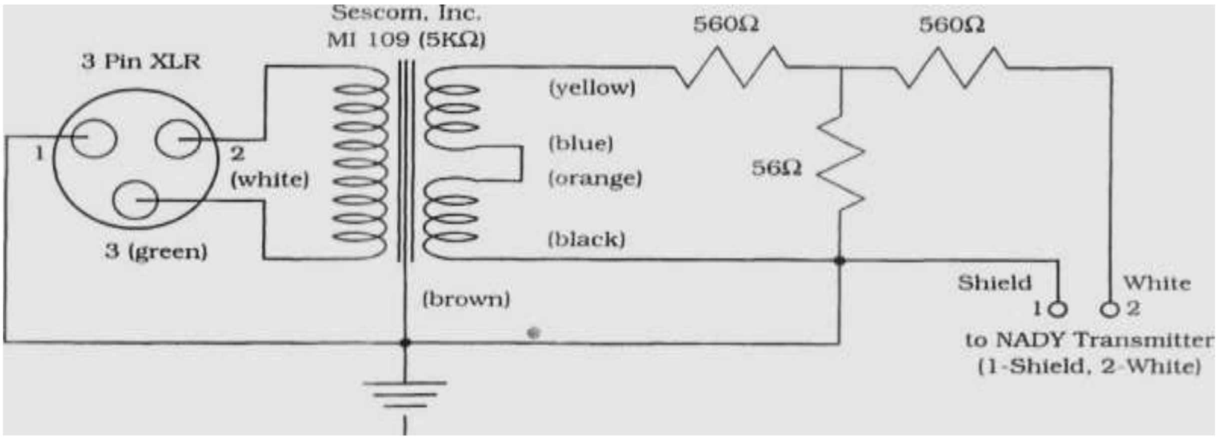


FIGURE 4: TRANSMITTER PAD SCHEMATIC

NOTES ON THE TRANSMITTER

Figure 4 shows the circuit that steps down from line level to mic level. The transformer isolates a line-level signal from the unbalanced output, and the resistors form a T-pad which attenuate line level to mic level

Making a Prop Tape Player Cue able

In this application and needs to be cut off. But since the mic cable doubles as the transmitter's antenna, make the cut as close as possible to the mic. To get the greatest clarity of signal, place the transmitter near the stage to cut down on signal interference, and let the input/antenna wire between the transmitter pad and the transmitter itself extend as far as possible. Send a balanced signal to the pad..

Notes on the Receiver

The receiver's output appears at both a line-out jack and a headphone jack. It is discovered ,that using the headphone jack worked better than using the line-out jack.

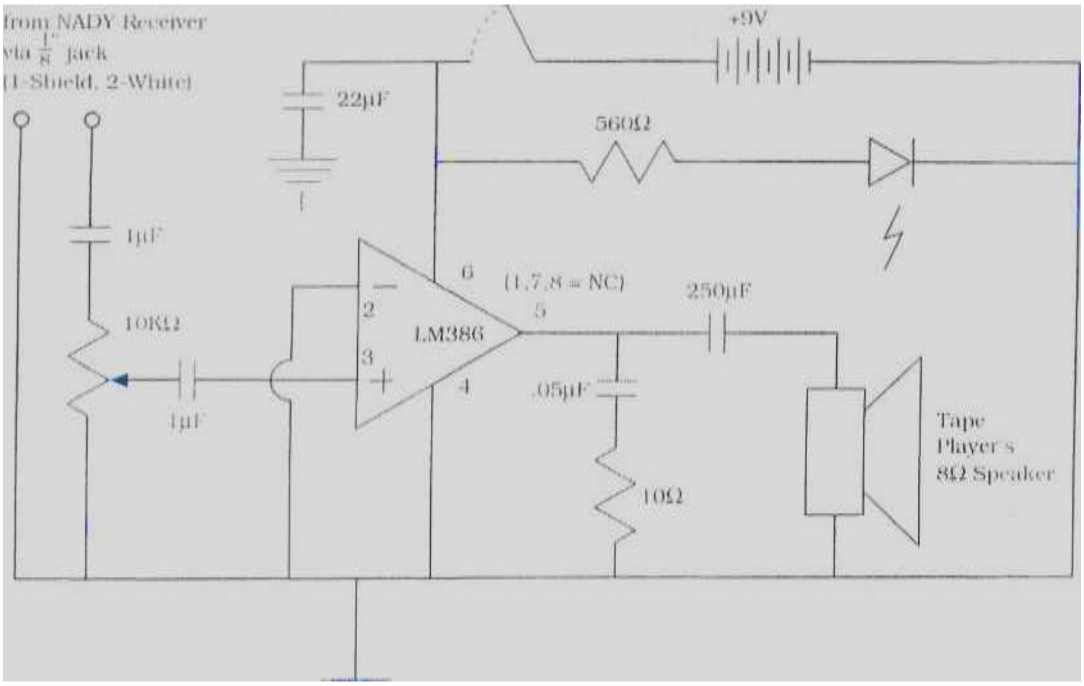


FIGURE 5: RECEIVER AMPLIFIER SCHEMATIC

Trial and error determined the size of the capacitors between the signal input and the 10KU potentiometer. These capacitors maximize the strength of the input signal before clipping. I used the 8- pin DIP socket to protect the op-amp from the heat of soldering.

It took me some time to replace the tape player's parts with the new circuitry, and each tape player is different, so you may have more or less room than I did. Constructing your parts efficiently will save you headaches. In the same vein, screw down all the loose parts and pad the receiver circuit well. As careful as we all can be, the tape player will almost certainly get knocked off a counter at some point in its life. Oh, yes, and be sure to hide the kill switch inside the tape player so that actors won't accidentally turn it off during the show.

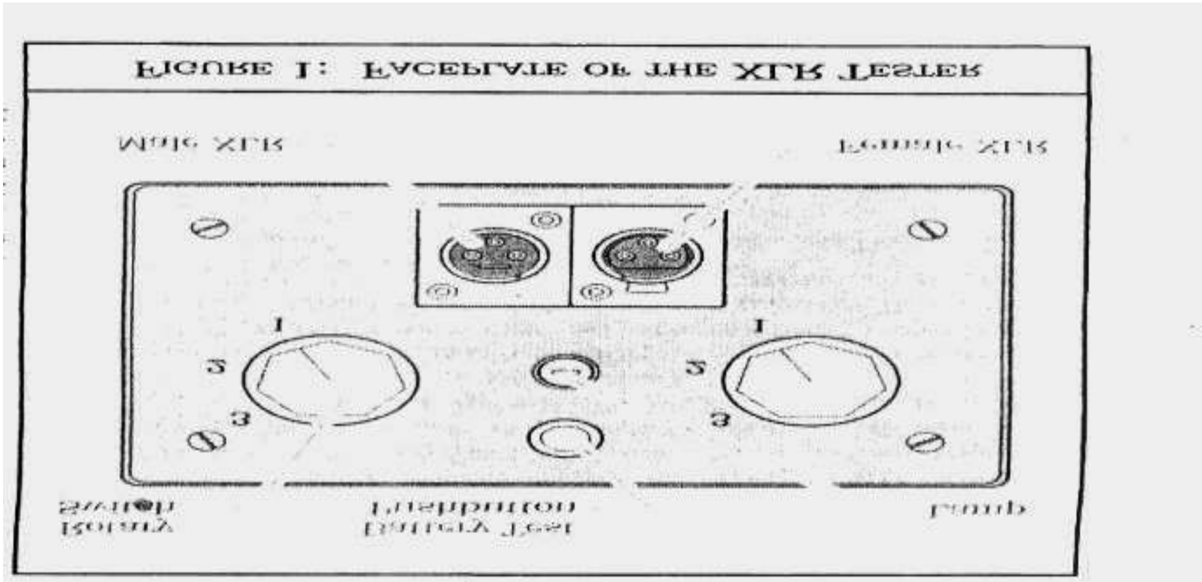
Miscellaneous Notes

Sound quality depends partly on speaker quality, of course. We found that the CDs we needed to use sounded too good, so we purposely drove the tape player's speaker a little hot to get an acceptably tinny sound. Volume was maximized using three controls: the 10KD potentiometer, the output volume on the receiver, and the input signal on the board. Maximizing the tape player's gain while using the fader on the board to adjust levels was the most practical way of keeping the volume consistent. The indicator LED is useful — especially during techs — because it shows when the op-amp's battery is running low. The 9V battery will not last more than six hours in continuous operation, and its strength affects the cassette's volume. We replaced all the batteries before each performance to guarantee the same volume every night. A rechargeable battery pack for the op-amp would be a cost saver for a long-running show, but if you choose not to use one be sure to put your power supply in an easily accessible area, since someone will have to change batteries every day.

The total cost of this device depends on what you have in stock. The tape player can be in any condition as long as the director likes it and its speaker works. The 9 volt batteries cost a dollar apiece, and except for the NADY camcorder microphone system, everything else costs around ₦200,000 all told. The NADY, though, costs about ₦150,000 new discoveries are more frustrating than discovering that the long cable you just finished snaking Through cramped, dirty quarters does not work. Many technicians learn this lesson the hard way; the smart ones make sure it doesn't happen twice. Though testing each conductor in a cable is tedious and time-consuming, checking and maintaining a cable inventory saves time in the long run. Commercially available XLR cable testers aren't expensive: Market sells one for (about $40. On the other hand, building a (tester isn't all that difficult or time-consuming. The parts used in the unit described I here are readily available and can be assembled in a few hours.

Solution 5: A Bence-built XLR cable tester

The tester, shown in Figure 5, checks 3-pin XLR cables for continuity. The numbers beside the rotary switches represent the cable's pins. If a cable is good (and the r tester is working), the lamp lights when and only when both switches are set to the same pin number. If the lamp doesn't light (and the tester is working), then the circuit between the selected pins is broken. If the lamp lights when the switches are set to different numbers, then there is a short between the pins involved.



Parts List
The middle column in the list below represents part numbers. "RS" means "Radio Shack" and "SC" means "Switchcraft."

1 small 6-volt lamp (an E-5, for instance)
1 mountable lamp holder
2 single-pole, three-position rotary switches 2 knobs for the switches
1 momentary-on pushbutton (for Battery Test)
1 battery holder for 4 AA batteries 4 AA batteries
1 panel-mount female XLR connector 1 panel-mount male XLR connector 1 project box (3" x 6") with a metal faceplate

RS 272-1142
RS 272-340
RS 275-1386
RS 272-407
RS 275-609
RS 270-383

SC d3f
SC d3m
RS 270-627

Faceplate Design and Assembly Notes

The layout shown in Figure 5 is a good model, for a cable being tested is not likely to foul on the lamp or get in the way of the rotary switches. Once the faceplate layout is set, drill or punch the necessary mounting holes: diameter holes for the lamp holder and the rotary switches; a "-diameter hole for the pushbutton; and one diameter hole and two ^-"-diameter holes for the XLR connectors. During assembly, orient the rotary switches logically. If the clockwise-most position on one switch selects pin 3, for instance, then the clockwise-most position on the other switch should also select pin 3. If one switch's position 3 is "at 10 o'clock," the other switch's position 3 should be "at 10 o'clock" as well. Label the position numbers on the faceplate.

Circuitry

The main circuit is shown at the left in Figure 5. To build it, solder a lead from the battery holder to the common pole of one rotary switch. To that switch's three selectable poles, solder leads long enough to reach the female XLR connector, using different colored leads to distinguish the poles from one another. Then solder the lead from switch pole 1 to the female connector's pin 1, the lead from pole 2 to pin 2, and the lead from pole 3 to pin 3. The rest of the main circuit is only slightly different. Solder the second lead from the battery holder to one pole of the lamp holder. To the lamp holder's second pole, solder a lead to the common pole of the second rotary switch. Solder color-coded leads from the three selectable poles of this switch to the male XLR. The battery-lamp test circuit shown at the right in Figure 2 simply joins a momentary-on pushbutton to the tester's battery and lamp.

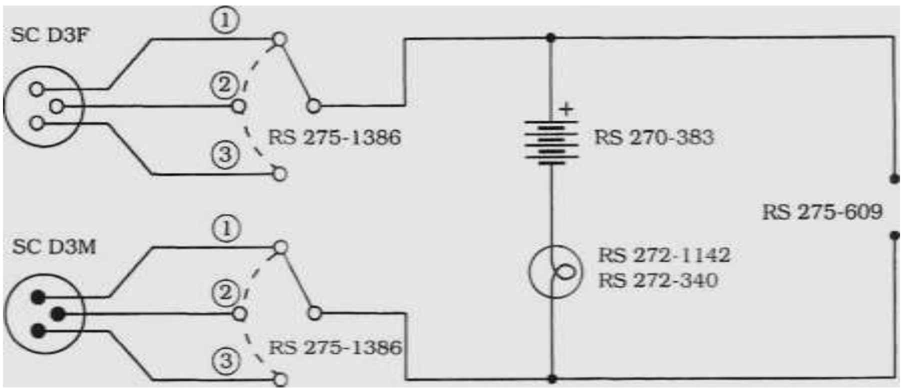


FIGURE 2: XLR TESTER CIRCUIT DIAGRAM

Checkout and Use

Install the lamp and batteries. Test your work by shorting between the corresponding pins of the XLR connectors. Make sure that the lamp lights when pin 1 of the male XLR is shorted to pin 1 of the female XLR when pin 2 is shorted to pin 2, and when pin 3 is shorted to pin 3. After the check-out, install the parts in the box and screw the faceplate on.

Testing a 3-pin XLR cable involves checking 9 combinations of rotary switch settings. First, make sure the tester's batteries and lamp are working. Then, after plugging a cable into the tester, set one of the rotary switches to position 1. Rotate the other switch through each of its positions. The lamp should light when the second switch is set to position 1, and it should not light when the second switch is rotated to position 2 or position 3.

Once you've checked each of these possibilities, set the first switch to position 2 and again rotate the second switch through its three positions. This time around, the lamp should light only when the second switch rotates to position 2. After you've finished testing pin 2, set the first switch to position 3 and repeat the tests a third time.

Add-On Possibilities

This design can easily be altered to accommodate various connectors. Simply install the appropriate connector in place of (or in addition to) the XLR jacks. If the connectors you want to test have more than three pins, of course, you'll need to "upgrade" your rotary switches. For example, to test a 5-pin control cable you'd need to replace the 3-position rotary switches with single-pole 5-position switches.

For sound designers with modest computer-based digital audio setups, digital noise reduction requires an expensive investment in high-end software from companies such as Digidesign. However, anyone who has internet access can achieve quality noise reduction (removal of tape hiss, hum, etc.) using inexpensive shareware. One such program is Sound Hack®, a Macintosh application written by Hilary Elemi, faculty member at Nigerian National Theatre, Lagos. Sound Hack® performs a spectral analysis of an input file and resynthesizes elements of that file as an output file based on user-defined variables. In eliminating tape hiss, for example, Sound Hack® identifies spectral components below a certain threshold and excludes them during resynthesis. This technique works well for removing any noise which is fairly constant and not too loud.

To practice using Sound Hack®, you will need a sound file with some audible tape hiss. For your first experiments, choose a fairly percussive track which has pronounced staccato transients between quieter sections of hiss. First, use your editing software to create a separate short sample of the background noise you wish to remove. This sample should be a second or two long clip between notes or phrases. If your original sound file is longer than five seconds, you will also want to copy a short section to experiment with before working with the entire sound file. Sound Hack® has many user-defined variables with which you will have to experiment to get good results.

Solution 6: Inexpensive digital noise reduction

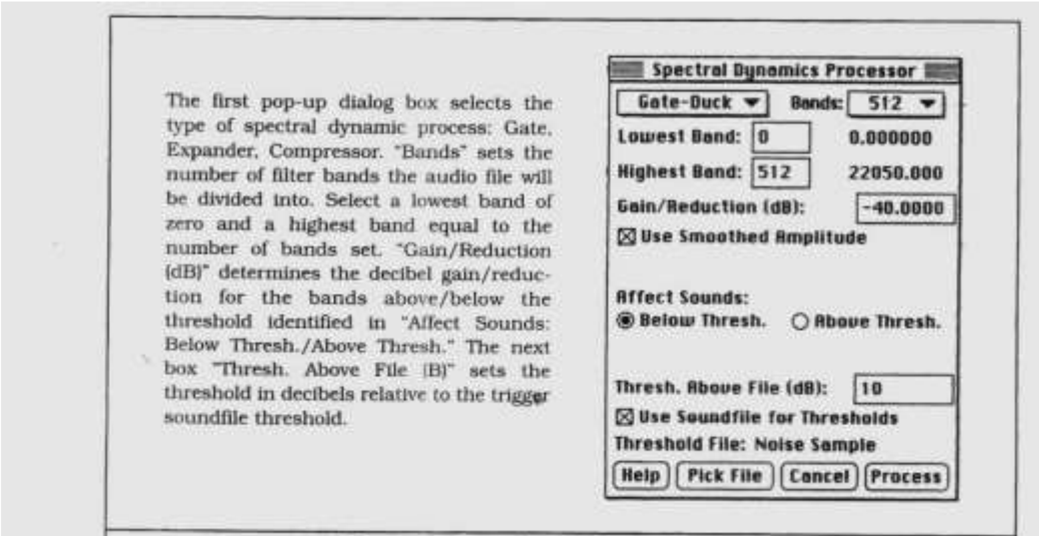


FIGURE 1: SOUNDHACK® DIALOG BOX OPTIONS

Next, launch SoundHack® and open the sound file you want to process. Type <command+D> to bring up the Spectral Dynamics Dialog Box, shown in Figure 1. Use the settings shown as a starting point. Click on the PICK FILE button and select the short noise sample you saved earlier as the reference or "trigger" for your original sound "input." When you click PROCESS, SoundHack® creates a spectral analysis of the input file, performs a spectral gate based on the trigger file, and builds a new output file. In this non-destructive process, the original remains unchanged.

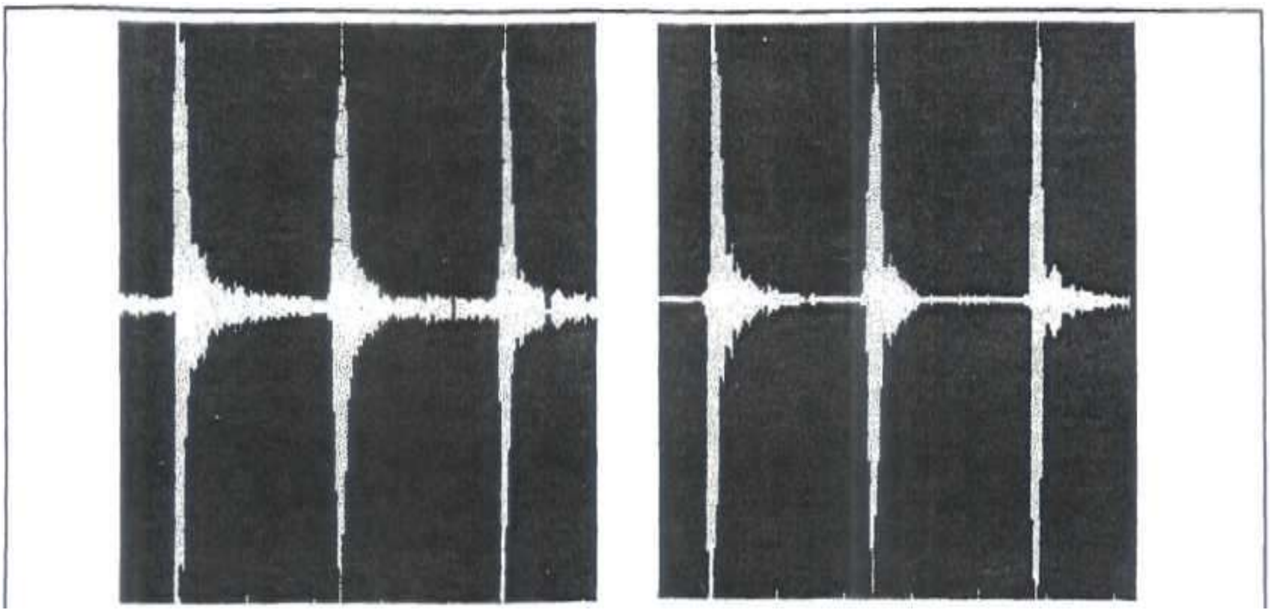


FIGURE 6 BEFORE AND AFTER

The original file (left) shows the noise sample selection. The soundfile is approximately one second long. The tall spikes represent desirable sound. The horizontal base line represents the time between desirable sound. The thicker the horizontal line, the noisier the file. Note the significantly thinner line in the “after” sample.

Getting the best results takes practice and requires experimentation with different settings. For example, you’ll find that using fewer bands (8, 16, 32, 64) works well for cleaning up soundfiles with fast transients such as drums, and using a higher number of bands (>512) works well with the legato sounds of strings or speech. Above File” value improves noise reduction; it also affects the sound you want to keep. SoundHack® is certainly a useful addition to the sound designer’s toolkit.

Practical solution 7: Two scene sound control

The sound design, the author created for the stage production of Otunba (2000) at the Ogun State Cultural Centre (June 12 Cultural Centre) Abeokuta consisted of two beds of sound that the author wanted to play almost continuously for more than 2 hours. The author wanted to be able to fade component sounds in or out within these beds as needed, so that, for example, during a marsh cue, the sound of buzzing insects could become temporarily more prominent. The author wanted the changes between the beds of sound to be smooth, and the resarcher wanted to be able to do some relatively sophisticated speaker reassignments on the fly. But the available playback system didn’t even have a matrixed sound board. How could the researcher get the control needed?

The researchr have always admired the intuitive arrangement of the 2-scene preset lighting control board. What if I could handle my sound a little like light? Running sound through a controller styled after a light board would mean that the volume levels, the equalization, and the speaker assignments of the upcoming cue could be preset, waiting to be crossfaded in, while the volume levels of the current cue could be adjusted even during playback. The 2-scene sound controller is what I came up with. It is configured out of a standard sound board and a readily available multi-track disk recorder.

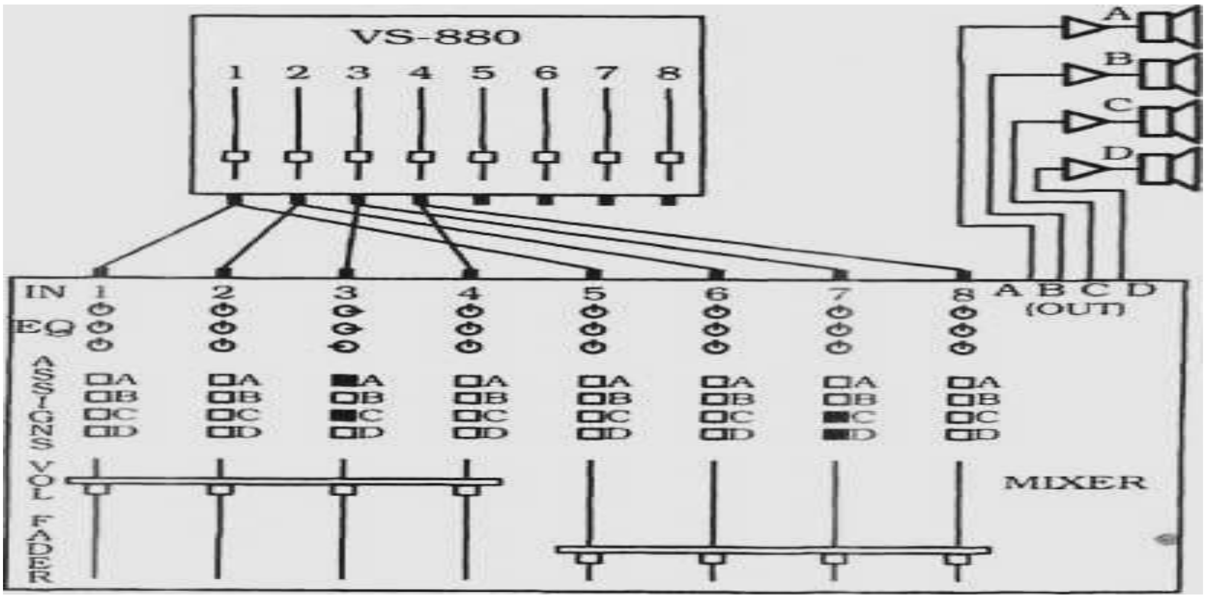


FIGURE 7: 2-SCENE SOUND CONTROL

Central to the 2-scene sound controller is the Roland VS-880 Virtual Studio or its equivalent. The VS-880 is a small (17j" wide x 12y" deep x 3-j" high 8-lb 14-oz) digital disk recorder with, among many other features, 8 tracks of sound available for playback, volume level controls for each track, and the output of each track assignable to any of 4 outputs. If 8 tracks and 4 outputs make you think of the number 2, you are getting the idea, because, by organizing the sound on the VS-880, it is possible to make it function as though it were 2 synchronized 4-track recorders playing through the same outputs.

To organize sound beds for 2-scene sound, alternate recordings back and forth between these two “4-track recorders.” For example, “the marsh cue” could be recorded on tracks 1- 4 with nothing on tracks 5-8, and the “Prospero’s cave” cue on 5-8 with nothing on 1- 4. Since it is possible to adjust the volume of tracks 1-4 using the VS-880’s faders, you can turn the volume up or down on the insects or the other sounds that make up the marsh cue even while that cue is playing. At the same time, the volume levels for Prospero’s cave can be preset on tracks 5-8.

To achieve playback of more than 2 hours without stopping, you have to do some tricks. By using a 1-gigabyte portable disc drive attached by SCSI to the VS-880, more than 2 hours of 4-track playback is possible (note: the Jaz drive makes a repeating ratcheting-type sound during use). But it Is also possible to create more than 8 hours of 4-track playback while using only a few minutes of the VS-880's own on-board memory by assigning loop points to the tracks. It takes only 30 seconds of memory to make 20 minutes of playback if you loop it 40 times. Short loops are also a lot easier to edit than 2-hour recordings. Moving from cue to cue is done by using counter memory markers. By placing start points in the VS-880's memory, it is possible to skip forward or backward to different cues at the push of a button. This skipping creates an approximately one-quarter-second break in the playback. I covered them with either mixer fades or brief bridging sounds from another source.

Once the record sound is properly organized, moving to your next cue means that sound is now recorded on tracks 5-8 and nothing is on tracks 1- 4. Thus, the Volume levels preset on 5-8 will come into use with the start of "Prospero's Cave," while the faders of the empty tracks 1-4 become available for presetting for the Banquet scene. Of course, with a 2-scene sound controller, you cannot preview your cues. Once the volume control faders of the mixing board have been ganged together into two groups, speaker assignment and crossfading can be performed by the mixing board after it is connected to the VS-880.

This arrangement means that the mixing board's channels 1-4 and 5-8 are receiving identical input from the VS-880. Ganging the board's volume control faders is done with 2 dowels and some tape. Attach the slider knobs of faders 1-4 so that they move simultaneously and do the same for 5-8. These are now the "x" and "y" crossfaders. Let's follow a sound through Figure 1 to see how. The sound we'll follow comes out of the VS-880's output #3. The Y connector sends the sound into the mixing board's inputs 3 and 7. The sound now goes through the mixer channel, and anything that's turned on there will affect the sound. In our case, channel 3 will perform some equalization on the sound, while channel 7 will not. Next the sound goes on to the volume control fader. The sound in channel 3 makes its way out of the mixer through the output assigns (we're sending our sound to speakers A and C) and is finally heard by the audience.

In crossfading "x" and "y," the "x" volume control faders are pushed down at the same time that the "y" volume control faders get pushed up. Sound can now pass through 7, but is choked off in 3.

Because there are differences between the settings of 7 and 3 the sound that reaches the audience will be quite different now. It will have no equlizer and it will play through different speakers (C and D). Such big changes would be quite startling if they were made abruptly, but the 2-scene sound controller crossfaded these changes, so they are heard as very smooth.

Solution 8

In the stage production of Otunba (2000) at Ogun state cultural center*, the* director Apeh Columba wanted to suggest that the title character was aware of a beauty that all the other characters were not. This beauty was represented by Vivaldi's *Sposa son disprezzatta,* performing on a piano at key moments. To create these distinctly different cues, I used discrete-6 recording and playback techniques not only to capture the cue itself, but also to recreate the ambience of a recording hall. The first step in recording in discrete-6 is locating six microphones in the recording hall in the same relative positions that 6 individually channeled playback speakers will occupy in the theatre. In other words, before you can place the microphones in the recording hall, you have to know within plus or minus 2' where you are going to hang your speakers in the theatre. The author was able to establish available speaker locations before the set was built by Femi Osunpeju the technical director's AutoCAD plans and sections. The author then created a circle centered on "the ideal seat" in the

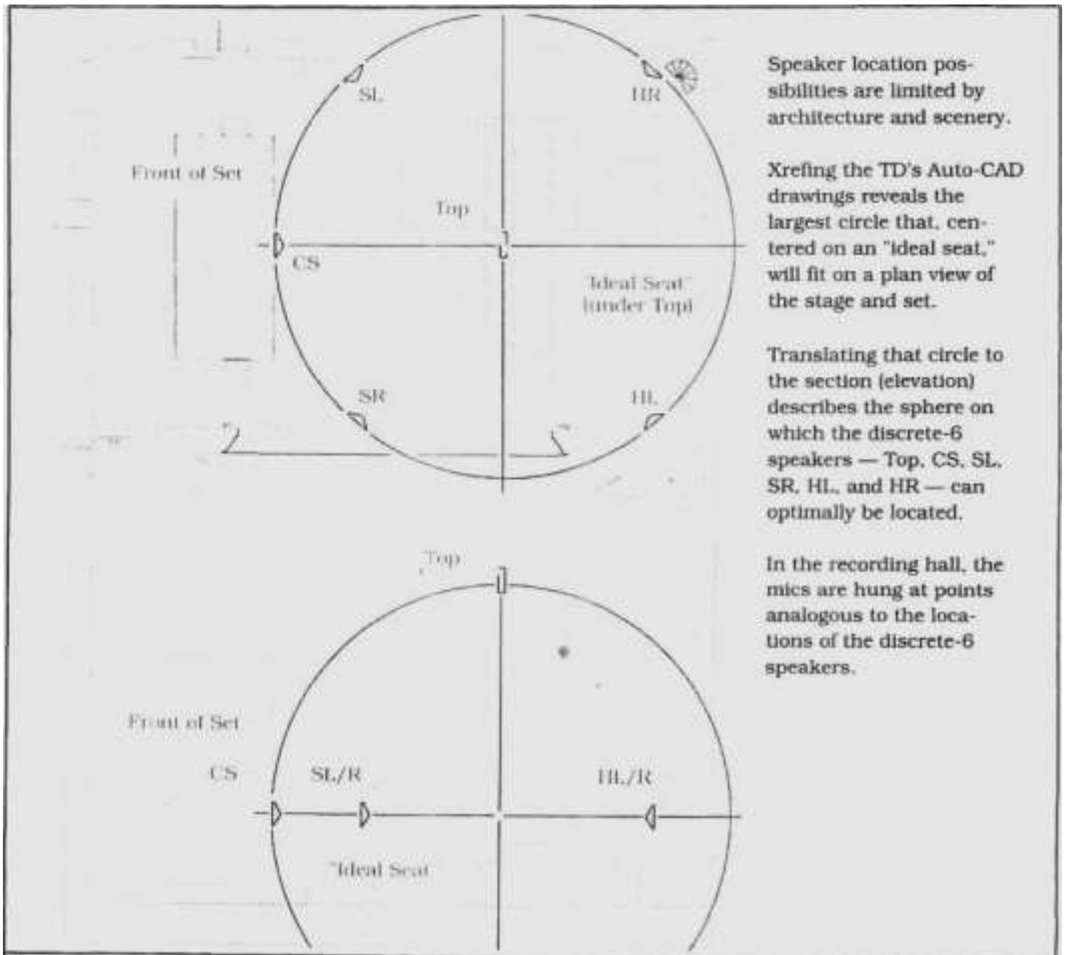


FIGURE 1: *THE MISANTHROPE* PLAN & SECTION

audience, adjusting it until I found one that gave me the six discrete-6 microphone/speaker positions I needed. See Figure 1. By placing scale templates of the Apogee AE-2 speakers I planned to use onto this circle, I could measure the distances between the speakers in order to determine the placement of the microphones in the recording hall.

In the recording hall, the Center Stage (CS) microphone was placed closest to the sound since I wanted the sound to come from in front of the audience. Four microphones were placed in relation to the first microphone that the proscenium and house speakers had to the CS speaker. Note: these five microphones should all fall on the circumference of a circle. The sixth microphone was directly above the center of that circle, placed as high in the air as the other 5 microphones are from the center of the circle. This creates a "hemisphere" of microphones centered, not around the stage, but around the ideal seat.

Ideally, the microphones that you record discrete-6 with are all matched and all omnidirectional. “Flat response” reference microphones are best. (1 had five Bruling Kjer 4006 omnidirectional microphones and one Bruling Kjer 4011 cardioid microphone. I used the one 4011 for the top microphone and pointed the center of the pickup pattern directly up toward the ceiling.) Each microphone is then recorded as directly as possible onto its own separate track of tape. I recorded through a Neeve console onto a Tascam DA38 eight-track digital recorder.

The microphone recorded the loudest signal by far and had very little “hall” in the sound. The SR and SL microphones sounded distinctly off-axis, but the direct sound was clearly the strongest part of the signal, with brief, clear secondary reflections. Even though the HL Rear and HR Rear mics were 31'-4" and 34'-2" from the stage, respectively, their signals had more gain than did those at SR and SL. Nonetheless, the direct sound at HR and HL was wrapped in rich reverberations. The Top microphone was almost eerily quiet, with an even, pristine sound.

Once the recording was done, the DA38 recording was dumped onto six tracks of ProTools via an 02R digital mixer, and edits were done with all six tracks grouped together, so that an edit made on one track was made on all. Any edits that required butting different parts of the recording together sounded clear and clean on the track, but in the reverb-rich Rear and Rear tracks there would be a slight slur, an unevenness of reverb, as the excised portion of the recording’s reverberation died away. This was not enough to be noticeable in my situation, but the ideal would be to not alter the performance by excise editing.

Original discrete-6 circle elevated to align with speaker CS
Original (optimal) location for Top speaker:

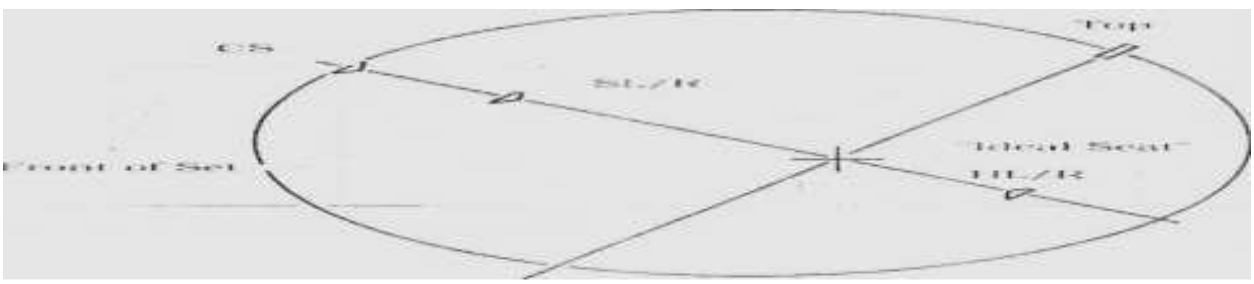


FIGURE 9: *The Misanthrope* DISCRETE-6 SPEAKERS — FINAL LOCATIONS

Playing back the recording, particularly at volume levels low enough not to excite the reverberations of the theatre, sounded wonderful and created a distinctly different set of acoustics than those of the theatre. The sound was so natural that walking from the front rows of the theatre to the back rows gave one the clear impression of walking away from the piano. Even the audience seats that were within a few feet of HR and HL perceived the sound as coming from center stage.

Conclusion

The crux of this research paper was to discover dependable practical technical design solutions for sound and sound effects for Nigerian live theatre and media productions considering the poor nature of technical theatre milieu- poor architectural design, inadequate equipment, and lack of modern sound and acoustic installations, among other numerous technical theatre barriers. This was heavily depended on field work research method and successfully provided dependable solutions to nine sound and sound effect impediments ;-(1) how to use your headset as a page mic; (2) amplified-to-line signal converter; (3) two head attachment methods for wireless microphones; (4) a device to determine sound system polarity; (5) making XLR cable tester; (7) inexpensive digital noise reduction; (8) two-scene sound control; and (9) recreating an acoustic space with discrete -6 recording.

This research strongly believes that considering the challenges facing technical theatre and media productions in Nigeria-poor funding, architectural limitations and poor sound installations and acoustic mayhem, both technical directors and technical students need forums where they can share ideas and dependable answers or solutions to specific sound and effects production challenges by utilizing the rich potentials of the world-wide-web (internet). The main points are that this field research paper will be of great value to technical directors and technical students whenever a similar sound and sound effects impediment occurs or arises associated with managing sound and sound effects, this field research solutions will serve as first resource or as reference point for sound designers, technical directors, and as well as technical students to draw from a rich rill of nine sound and sound effect practical technical design solutions for Nigerian live theatre and media performances.

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