THE EFFECTS OF MULTI-CHANNEL AUDIO PRODUCTION TECHNIQUES ON LISTENING PARAMETERS

By

Theresa L. Forbush

A THESIS

Submitted to
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for the degree of

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Department of Telecommunication

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ABSTRACT

THE EFFECTS OF MULTI-CHANNEL AUDIO PRODUCTION TECHNIQUES ON LISTENING PARAMETERS

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This thesis examined the listener's perception of the sonic dimensions of depth, width, and height through some of the possible combinations of storage mode, transmission medium, and playback environment. The aim of this research was to determine the listener's perception of these dimensions and their effect on the listener's liking of the production.

In order to accomplish this task, a musical selection was produced using current multi-channel audio production techniques. In addition, a simple music video was produced to accompany the song.

Analysis of the testing results show only the dimension of width to be significantly perceived by the listener. However, an amateur audience cannot distinguish between the width of stereo and bi-polar mono playback formats. Rather only between those and a purely monaural playback format.

The author was responsible for all aspects of the production phase as well as the testing. This manuscript details all of these processes.

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Accepted by the faculty of the Department of Telecommunication, College of Communication Arts and Sciences, Michigan State University, in partial fulfillment of the requirements for the Master of Arts degree.

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I would also like to thank my parents, Marshall and Peg Forbush and my brothers and sister: Mark, Todd, Michelle and Tate and my Grandmother, Arlene Yats for their continued encouragement.

Lastly, I would like to thank my friends for seeing me through this. I hope that I can be as supportive and understanding for them as they have been for me.

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The production which accompanies this manuscript may be viewed by contacting the author or Gary A. Reid, Department of Telecommunication, 409 Communication Arts and Sciences Building, Michigan State University, East Lansing. MI 48824-1212.

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CHAPTER I

INTRODUCTION

This thesis will investigate multi-channel audio recording techniques. The recording and mixing processes involved in determining stereo placement, width, depth and overall perspective will be analyzed as to their relative value. In addition, mono vs. stereo and low quality vs. high quality speaker variables will be considered. The purpose is to provide the audio producer with some research supported guidelines for multi-channel audio production. This research will help to define the consumer's ability to distinguish the qualities of stereo placement, width, depth and overall perspective, as well as the effect these variables have on a listener's enjoyment of the music. With the wide range of possible release formats available today, the audio producer is in need of research supported information to best utilize the medium at hand. The advent of the "music video" as a popular art form has added new parameters to a producers decision-making process. The impact of video release formats on the audio producer will also be examined.

Within the ranks of the audio recording industry, the methods used during the recording and mixing process are as individual as the producers themselves. Some multi-channel audio releases are created making full use of the available production possibilities while others

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only use pieces of the production spectrum. In particular, some of the sonic dimensions dealt with in this thesis will be stereo placement, width, depth, height and overall perspective of the final production.

Through the use of panaramic potentiometers, different instruments can be placed at specific locations across the width of the stereo spread.[1] Another parameter, that of depth, gains a portion of its reality from stereo placement. An additional component responsibile for sonic depth is artificial reverberation added to the recordings of individual instruments and voices. By establishing the delay time of each echo, the overall decay time of the reverberation, and the equalization of the artifically repeated sounds, each instrument and voice will appear to be located at different distances from the listener.[2] For example, to replicate a live performance, the producer would probably place the voice of the lead singer in the center of the stereo placement and fairly "close" to the listener in the layout of depth. Backup vocals, however, would probably appear to be further behind the lead vocalist and possibly moved to the right, to the left, or even both directions from center. In addition, the dimension of height can be added to the overall sonic picture by giving emphasis, in appropriate places, to those frequencies at the very top and very bottom of the audible spectrum.

When used together, these dimensions can create the illusion of sonic movement. Various sounds can be manipulated to appear as if soaring about the audio picture. Each instrument and voice can be assigned a specific localization and their placement, relative to each other, can then be manipulated. By employing these sonic dimensions

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during the mixdown process, a producer can bring to life a piece of music, by stimulating the visual imagery within the mind of the listener. When properly achieved, this imagery can enhance a video counterpart which will, in turn, reflect back positively on the audio recording.

Regardless of how good a recording may sound in the studio, it is ultimately subject to the limitations of the medium in which it is released to the public. Currently the most common of these are the 45 rpm record, full length record album, cassette tape and 8-track cartridge. Others, such as the open reel tape, compact disc and beta high-fi are less common among consumers, but are nonetheless potential release formats available to an audio producer. In addition, the audio accompanying a video can appear in three or more forms also. Each of these storage mediums offers a different set of limitations inherent to the medium itself. For example, songs placed close to the center of a vinyl disc have a higher distortion ratio due to the slowed groove velocity.[3] Sonic imperfections created in the mastering process as well as surface noise due to the movement of the needle along the vinyl are both much more noticable within songs placed in the last position on each side. Therefore, an audio producer would not choose to put a selection with a wide dynamic range toward the center of an album. While tapes and compact discs do not exhibit this problem, they have limitations of their own.

Not only do the various formats affect the sonic quality of the music, but the transmission medium also contributes its effect as well. Broadcast radio now offers AM monaural, AM stereo, FM monaural and FM stereo transmissions. Though the AM spectrum can be divided and trans

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transmitted in the left and right matrix to produce a true stereo broadcast, the fidelity of AM stereo is very low compared to that of FM stereo. This is due to the very limited bandwidth of AM radio transmission, as well as the form of modulation itself. Therefore, the sound quality loss for AM stereo is even greater than that which occurs in a monaural broadcast on the AM spectrum.

Another variable beyond the control of the producer is the listening environment surrounding the consumer. From portable AM monaural radios through varying qualities of car stereos to expensive home stereo components, each type of playback setup adds its own coloration to the sound it outputs. Indeed, each component can vary greatly in sonic reproduction according to the individual manufacturer.

Therefore, a producer must specialize his/her mode of production in relation to their marketing decisions. Each possible combination of these variables will result in a different sound available to the consumer. Therefore, producers often find it necessary to create several different mixes designed to maximize the quality of the output for various possible combinations of release format, transmission medium, and playback environment. While this approach has its advantages, it is very costly in studio time and it also disrupts consistency among mediums. That is, a consumer will hear a song not only on FM radio, but also on AM radio and television and in a variety of different playback environments. Ideally an audio producer would like to create one, or at most two different mixes which would be sonically similar across all combinations of release format, transmission medium, and playback environment. Traditionally these variables have been exercised within a purely audio realm. However, a substantial portion of multi-channel audio

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audio releases become incorporated in video productions. The popularity of the music videos has forced audio producers to contend with the transmission methods of broadcast and cable television as well as their respective playback environments.

The music video is emerging as a production hybred which couples a multi-channel audio work, originally intended for use within the radio/record industry, with a video creation. Due to the inherent quality limitations of the speakers generally installed in television sets, recording mixes created for records and radio are not effectively demonstrated during a television transmission. Most television speakers use one cone to reproduce the entire audible spectrum. However, to achieve clear replication of very high frequencies, a very small specialized cone is necessary in order to vibrate at the required frequency. Similarily, a large cone is necessary to reproduce the low frequency tones. Therefore, a speaker enclosure containing only one cone cannot reproduce the entire audible spectrum to the extent that enclosures using two or three cones can.[4] Since the majority of the radio/record industry, including most probably the producer who created the mix, use the higher quality speakers, the extremes of the frequency spectrum are included in the overall sound. When these productions are played on a single cone speaker, the upper and lower frequency extremes become lost and therefore, the sonic balance is disrupted.

In addition, the various sonic dimensions discussed earlier rely on a discrete left and right signal to be fully created. However, most television audio is currently transmitted in the monaural format. On March 29, 1984, the FCC approved a Second Report and Order authorizing the use of subcarriers on the broadcast television aural baseband for a

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variety of purposes, among them stereo audio for television.[5] The Broadcast Television Systems Committee (BTSC) and the Electronic Industries Association (EIA) have endorsed the Zenith proposal for the transmission of stereo television audio. While this system has not been authorized as the only multi-channel television sound format allowable, the FCC has granted protection from interference to the pilot subcarrier of 15.735 kHz [Sec. 73.682 (c)(3)].[6] This is, however, only an initial step toward the implementation of stereo television. The networks and ultimately the local television stations must equip their transmitters to broadcast in stereo. Currently only one network, NBC, has announced its intention to broadcast stereo programming. The remaining networks, ABC, CBS and PBS are engaged in internal testing of the technology, but have not yet released any intentions for using it. Likewise, the local affiliates must also accommodate for the retransmission of stereo television. Only six stations across the nation have announced that they will invest in the facilities necessary for multi-channel television sound.[7] Therefore, the majority of the broadcast television viewing audience will continue to receive a monaural audio signal for at least the near future.

Due to this factor, the various sonic dimensions which require distinct left and right channels are lost entirely to the television viewer. Therefore, recordings produced making use of the various sonic dimensions have ultimately used large expenditures of time and money to create a product which cannot be received by most television viewers.

In an effort to bypass this problem, Music Television (MTV), a primary programmer of music videos, has created a simulcast FM transmission to accompany its cable television signal. Customers can request an FM

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an FM radio hookup from their cable operator, and tune into this simulcast transmission of MTV on their home stereo. While the average consumer probably believes they are receiving stereo television, they are actually listening to a simulcast radio/television transmission. While this form of reception does bypass many of the limitations of television audio, it is a rather crude and cumbersome method. In order to receive MTV audio in this manner, the consumer must obtain a second cable feed and connect it to their FM stereo receive. Also, since the television and the FM receiver are both necessary to create the MTV signal, the consumer must tune in both units to the appropriate channel. With this setup, the audio and the video portion of the signal are now being controlled by two separate units. Should the viewer decide to switch channels, the video will change, however, the audio portion, received through the FM tuner, will remain on MTV.

Music videos are conceived and produced through some combination of the artist and their record company, usually after the completion of the song or album. Some record companies will handle the full range of video production decision, requiring the artist to simply show up on the appointed day for the video recording. In other instances, such as Michael Jackson's "Thriller" video, the artist maintains creative control throughout the entire production process. Regardless of who accepts responsibility for the video production, these music videos are created primarily as a promotional/marketing tool. It is hoped that these videos spark consumer interest in the album which the selection is taken from and therefore, generate record sales. Ultimately, it is the ability of the video to complement the audio, and visa versa, which will determine the success of this promotional form. The production of these videos usually begins after the recording of the corresponding album has been completed. Therefore, the sonic picture created for the album release, will ultimately be constrained within the television format for the promotional release. That is, generally a monaural signal, received through a single cone speaker enclosure after transmission through television broadcast. With the loss of distinct left and right channels, most of the sonic dimensions discussed earlier are lost. Specifically, width is entirely distroyed and depth is severely distorted due to the compilation of the instruments and voices into one center image. Because of the single cone speaker enclosure, very high and low frequencies are lost and/or distorted, resulting in disruption of the sonic balance as well as the dimension of height. Therefore, the musical portion of these promotional videos suffer a large decrease in audio quality from the original mix. This phenomenon appears contradictory to the intended purpose of the music videos, that is, to promote the corresponding audio release. Therefore, this research will measure the consumers ability to distinguish the sonic dimensions created for an audio release. Also, subjects will be divided and tested, so as to identify their ability to recognize the loss of those dimensions of stereo placement, width, depth, height and overall perspective. The information derived from the testing of this production will be compiled into a summary of consumer's perceptions of the sonic dimensions of stereo placement, height, width, depth, and overall perspective. This summary will offer an audio producer information defining the role these dimensions play in stimulating listener enjoyment of the piece. Without this information, audio producers cannot take into consideration the consumer'sability or inability to distinguish the various production parameters within a song. Therefore, producers are forced to direct their efforts toward the medium rather than the consumer.

While the creative effort in producing the audio and video components will be the major portion of this thesis, the research/testing component will be used to derive guidelines to define the consumer's perception and appreciation of the applicable production techniques. This, in turn, will aid the audio producer in decisions considering the expenditures of time and money when producing for various release formats.

NOTES--CHAPTER I

[1]Robert E. Runstein, Modern Recording Techniques, (Indianapolis: 1974)., p. 35.

[2]John M. Woram, The Recording Studio Handbook, (New York: 1976), p. 204.

[3]Robert E. Runstein, Modern Recording Techniques, (Indianapolis: 1974), p. 292.

[4]John M. Woram, The Recording Studio Handbook (New York: 1976), p. 183.

[5]"Multichannel Television Sound - Part I," NAB Special Bulletin, Science and Technology Department, (June 15, 1984), p. 1.

[6]Ibid., p. 2.

[7]Susan Spillman, "Stereo TV Sparks Little Enthusiasm," Advertising Age, (June 25, 1984).

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CHAPTER II

MULTI-CHANNEL AUDIO RECORDING

The first phase of this thesis dealt with the multi-channel audio production process, beginning with the recording of an analog signal onto magnetic tape. A local musical group, "The 80's," was recorded using the 24-track audio studio. This production facility offers the capability of recording, on a 2-inch magnetic tape, 24 distinct signals or "tracks" which will ultimately be combined to produce the final product. Each instrument or voice is assigned to a different track and can be manipulated independently of all other tracks during the recording and mixing processes. To assure the highest possible quality of the final recording, each track must be initially recorded as free of noise and distortion as possible. With 24 separate signals combined to create the final piece, slight imperfections in each track will be multiplied in the ending mix.

One of the limitations of magnetic recording which contributes to the problem of creating a low noise signal, is the noise floor or hiss generated as the tape moves through the tape recorder. In order to effectively mask this hiss, the level of the recording should be placed as far above the noise floor as possible without causing distortion. One piece of equipment which aids in this processing is the noise reduc

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reduction unit, in this case, the dbx system was used. The dbx Noise Reduction Unit compresses or squashes the dynamic range of the signal. That is, the variance between the maximum and minimum levels is decreased allowing the overall recording level to be higher than without the use of noise reduction. This increases the signal to noise ratio or the distance between the lowest level of the signal and level of the noise floor. Therefore making the noise floor less audible within the recorded signal. When this recording is played back, the noise reduction unit reverses the process by expanding the dynamic range. The benefit of the noise reduction unit is that it allows the signal to be influenced less by the noise floor present in tape storage.

Another place to consider the possible introduction of noise into each track is the environment in which the recording takes place. When recording The 80's for this production, it was necessary to have several microphones operational in the studio at the same time. Therefore, caution was exercised in the layout of the specific instruments in the studio. Ideally, total isolation of each sound would result in the most distinct individual tracks. However, due to the design of the studio and the necessity for the majority of the musicians to perform at the same time, total isolation is not possible, nor desirable. It is therefore important to control, as much as possible, the sound which each microphone will be exposed to. A portion of this control comes from the positioning of the musicians in the recording studio. The drum kit, requiring the largest amount of space, was placed in the center of the room. (See Figure 1)

The majority of the drum kit consists of Simmons electronic drums. The signal from the drums is routed through a synthesizer control box

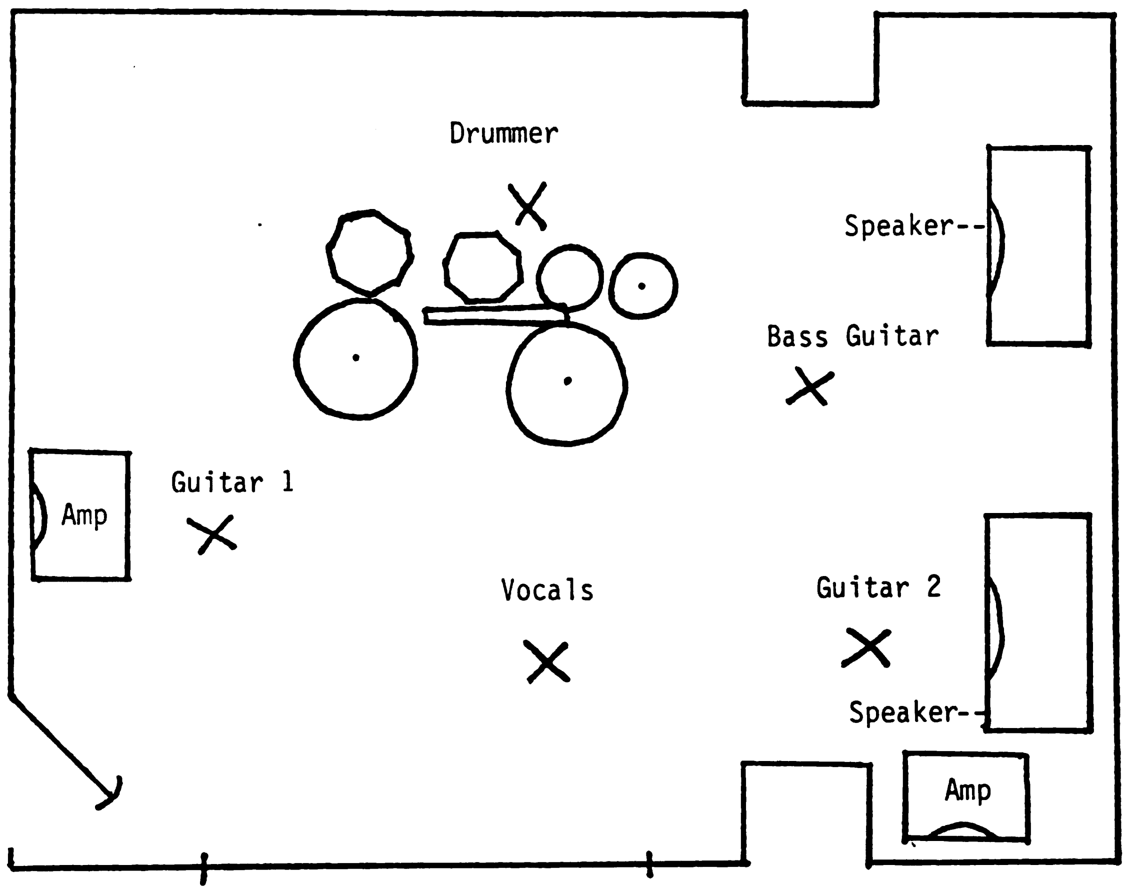

Figure 1. Audio Studio Layout

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which allows the drummer to electronically process the signal and therefore create a variety of sounds. This synthesizer allows the musician to control volume, tone, and musical envelope, or the portion of tone divided among the attack, release sustain and decay. (See Figure 2) Different configurations of these elements allow the drummer to achieve a variety of sounds using the same drum kit. The output of each drum, processed through this synthesizer control box, is then connected into a microphone line and direct injected into the mixing console. Therefore, these drums do not require the use of microphones, and only the tapping of the sticks, striking the pickup heads of the drums is audible in the studio. The snare drum, however, is acoustic, and requires a microphone to detect the sound created as the head of the drums is hit, which in turn causes the vibration of the snares. For the recording of the snare drum, a Shure SM57 dynamic moving coil microphone was used. This microphone is beneficial in capturing the snap of the snare drum and is also strong enough to withstand the transient peak levels which the snare exhibits. The head of the SM57 was placed approximately one inch from the drum head, slightly inward of one of the tuning lugs. (See Figure 3) This position for the microphone is optimal for two reasons. First, since the drummer is striking the snare drum in the central portion of the head, the microphone stands a greater chance of being hit the further inward on the drum head it is placed. The reason for aiming the SM57 at a tuning lug is that the vibration pattern of the head, activated by the striking of the sticks, is the purest at the locations where the drum head is anchored, those locations being the tuning lugs.

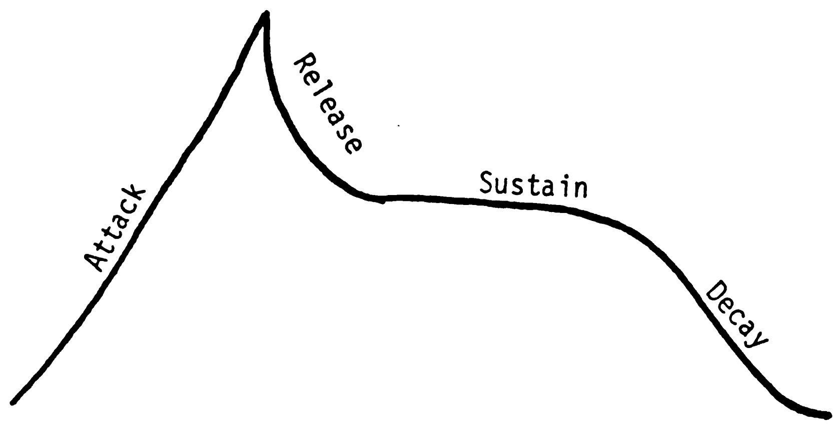

Figure 2. Musical Envelope

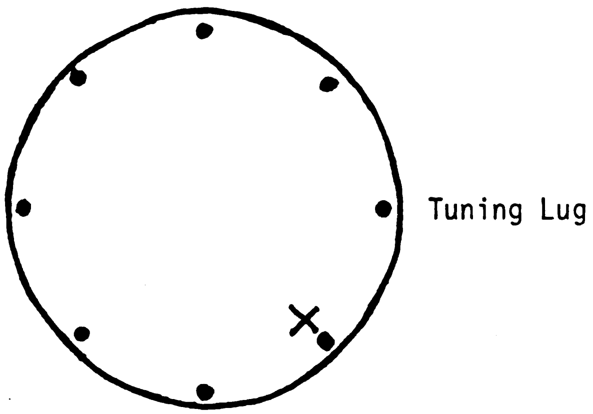

Figure 3. Snare Drum Microphone Placement

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In addition, a hi-hat, a ride and two crash cymbols are included in the drum kit. With the cymbols being very close together and very loud, it would be impossible to record each cymbol seperately without the sound of the other cymbols and drums also bleeding into that track. Furthermore, in a performance situation, the drummer determines a mix between the level of these cymbols according to how hard he strikes each one. Therefore, two microphones known as a stereo pair are suspended above the drum kit to record all of the cymbols. These mics will also pick up some of the sound of the acoustic snare drum. It is therefore necessary during the mixdown session to recheck the overhead stereo pair in combination with the snare drum microphone lines for possible cancellation between them. The possibility of cancellation arises due to the fact that the same instrument, for instance the snare drum, is being picked up on multiple microphones. Because the microphones are at different distances from the source, the signal enters them at different stages of the waveform. When combined together, the two replications of the signal may be out of phase with each other, and would therefore result in frequency cancellation.

A check for any frequency cancellation can be done very simply during the recording process. The mixing console contains a button on the monitoring section, which compiles a stereo signal into one monoaural output. This "mono-button" only acts upon the monitor system and does not affect the recording process. By listening to the entire drum kit with the mono-button depressed, any frequency cancellation which would occur during the mixing process will demonstrate itself at this point. In the case of this recording, no frequency cancellation occurred.

Also in the studio with the drum kit were two electric guitars and an electric bass guitar. The bass guitar is recorded with the use of a direct box. This direct box allows the signal created by the guitar to be directly injected into the microphone line without the use of an amplifier/speaker combination. Therefore, the signal created by the bass guitar is taken from the guitar directly into the mixing console. This serves two purposes, first it provides a clean, unaltered signal to the recorder and second it eliminates the need for a bass amplifier in the studio. The loud, low frequency signals coming from the bass amplifier can cause distructive interference by being picked up by the other microphones in the studio.

The two electric guitars sound significantly different when recorded direct than when played through an amplifier/speaker. This is due primarily to the amplifiers ability to alter the sound of the guitar according to the artist's adjustments of the amplifier controls. In addition, The 80's make use of several guitar peripherals such as chorusers and delay boxes which further alter the sonic make up of the guitar output. The signal from the guitar is routed through these peripherals and then to the amplifier/speaker. In order to record the guitar sound as the artist performs it, it is necessary to play the guitars through the peripherals and amplifier/speaker combination, and record the resultant signal with a microphone. Several steps must be taken to discourage the sound which eminates from these speakers from bleeding through other microphones, such as the overhead cymbol mics, and leaking onto other tracks. First the two amplifier/speakers are placed on opposite ends of the studio facing the acoustically tiled walls. (See Figure 1) The sound absorbing panels mounted on the walls help to trap the signals which leave the speakers and therefore cut down on the level of guitar sounds which ultimately enter the studio environment. In addition, the amplifier/speaker combination is placed on top of a chair to discourage sympathetic vibrations in the floor which may transfer up other microphone stands and enter into other tracks. To record these two guitars, a Shure SM57 and Shure SM58 are used. These are both dynamic moving coil microphones with cardioid pick-up patterns. The heads of these microphones are placed just inside the edge of the speaker cone approximately one inch from the speaker surface. As with the vibration pattern of the snare drum head, the waveform is truest at the edge of the speaker cone, where it is anchored. While there is no danger to the microphone if placed closer to the center of the speaker cone, another problem becomes more prominent here. The center of the speaker moves back and forth to set the cone in vibration. For certain frequencies, the distance of the movement is large enough to cause an air disturbance which would be picked up by the microphone. This would be recorded as a swishing sound along with the desired signal. Therefore, it is desirable to keep the microphone closer to the edge of the cone, and approximately one inch away from its surface.

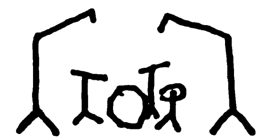
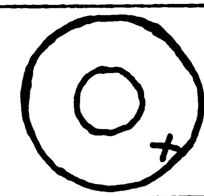
An additional microphone is set up in the studio for a scratch vocal track. This mic serves two purposes. First, it allows a rough vocal track to be monitored during the recording process so that the musicians can easily follow the flow of the song. This scratch vocal track can also be helpful in establishing timing during a ccapela portions of the piece and during vocal overdubs. Secondly, this additional microphone allows the musicians a means of communication with the engineer/producer in the control room. (See Table 1)

Once the instruments are set up in the studio, and the microphones and direct lines are in place, it is necessary to establish the track assignments and recording level for each signal. The optimal meter peaks for a recording at good levels would be in the range of -5 to +1 db (decibels) from 0 VU (volume units). In addition to the VU meters measuring levels, each track also has a peak light which corresponds to that particular signal. These peak lights flash when high level, transient signal peaks are detected which occur too rapidly to be displayed by the needle on that channels VU meter. When these transient peaks are present, it is necessary to activate the microphone trim control for that particular channel. This trim control limits these transient peaks within the signal which can cause distortion in the recording.

Through the microphone inputs in the studio, each signal has been routed to a certain input channel on the mixing console. These channels must then be assigned to one of the 24 available tracks on the multi-track tape recorder. This is done by simply depressing the button labeled with the appropriate track number on each input channel. For example, the bass guitar is routed to input channel #4 on the mixing console and will be recorded on track #12. Therefore, on channel #4 the button labeled for track #12 is depressed, and the bass guitar may now be recorded on track #12 on the 24-track tape recorder.

In turn each musician plays their instrument so that an appropriate level can be established. The trim control on each channel is adjusted to the highest possible setting which does not allow the peak light to flash. The fader, or potentiometer, is then used to establish the

Table 1. Microphone Choice and Placement

Instruments	Recording Method	Microphone Placement	Graphic Display of Mic Placement
Bass Drum	Direct Injection	From Synthesizer Control Box	
Snare Drum	Shure SM 57	Capsule of microphone approximately one inch from head of drum, next to tuning lug	See Figure 3
Tom Toms	Direct Injection	From Snythesizer Control Box	
Cymbals	AKG 452EB	Stereo pair, suspended from booms on both sides of the drum kit, over symbals	
Bass Guitar	Direct Injection	Through Direct Box	
Guitar 1	Shure SM57	Microphone capsule placed approximately one inch from surface of amplifier, just inside the rim of the cone.	
Guitar 2	Shure SM58	Same as Guitar 1	Same as Guitar 1
Vocals	Neumann U87	Placed in shock mount with wind screen approximately 18 inches from singer.	

level of signal sent to the recorder. This level is measured on the VU meter for each respective channel and should predominantly remain in the -5 to +1 db range. The fader on the monitoring section of the board is then used to determine the volume of each specific track in the control room monitor mix. The levels of the various instruments in the monitor mix has no effect on the recording taking place. While the recording levels are being established and the control room monitor mix is being created, a third mix must be derived for the musicians to hear in their headphones. Due to the fact that some instruments are recorded by direct injection, it is necessary to create a headphone feed containing all instruments and voices for the musicians in the studio. Without the headphones, none of the instruments recorded directly into the mixing console can be heard in the studio.

The musicians headphone mix can be created in two different ways. The first is to simply route the control room monitor mix into the musicians headphones. The second is to use the headphone cue feeds on each track to derive an overall mix which is satisfactory to the artists. The second method is generally the more desirable of the two means. Musicians often prefer to hear more of one specific instrument, like the bass drum, to serve as a cue for when a certain part is played, or to maintain timing in a difficult portion of a song. This method of deriving the headphone mix also allows the producer to work with the monitor section of the board to begin a rough mixdown during the recording process without interfering with the headphone feed.

At the head of each tape, prior to the recording of any music, calibration tones must be recorded. These calibration tones consist of 30 seconds at 1000 Hz, 30 seconds at 10,000 Hz and 30 seconds at

100 Hz. Once placed on the head of the tape, these tones will aid the engineer in checking the recording alignment.

For this recording project, the rhythm tracks and scratch vocals were layed down simultaneously. The rhythm tracks consist of drum tracks, bass guitar, and two rhythm guitars. The lead guitar, all vocal parts, and an additional snare drum part, however, were all added during overdub sessions.

The overdubbing process allows additional tracks to be added to a recording without altering the original tracks. This process may be useful for adding solos or pieces which are very difficult to play and may require several takes to perfect. It is also a primary means of adding acoustic tracks which would require additional microphones in the studio during the initial session. Often softer acoustic instruments, such as acoustic guitars, pick up significant leakage when recorded within a rhythm track session. Conversely very sharp acoustic instruments, such as trumpets, would leak into drum and guitar mics if recorded with rhythm tracks. Vocals are nearly always overdubbed. This is done for a couple of reasons. First, voices are not very loud in comparison to instruments. Therefore, it is necessary to raise the fader, thereby increasing the level of the voice, during the recording process. However, any other signal present in the recording studio, such as leakage from other instruments, also becomes increased in level. This leakage can be very disruptive when adding the voice level to the overall mix. In addition, should the vocalist sing the wrong words, the entire group does not have to re-record their parts, only the vocal tracks must be re-recorded.

When in the playback mode the pre-recorded tracks are monitored at the play head. This playback head is located several inches past the record head. Because of this head arrangement, parts overdubbed in time with pre-recorded tracks being monitored on the playback head, would actually appear several inches apart on the tape. If this were the case, these tracks would be out of synchronization with each other during the playback process, when all tracks are monitored at the playback head. To make the overdubbing process work, the tape machine is placed into the selective synchronization mode. In this mode, the record head on the tape recorder serves as a playback head for the tracks already recorded. Therefore, tracks being overdubbed will be synchronized with those previously recorded to allow mixing of all of the tracks at once. Track assignment and level established for overdubbed tracks are performed in the same manner as previously described for rhythm track sessions. The channels on the monitor section of the mixing board which correspond to the pre-recorded tracks are placed in the tape return mode. This mode allows the signals already recorded on the corresponding tracks to be included in the monitoring process during the overdub session. The headphone mix for the musicians doing the overdub is then derived from the headphone cue feeds on the monitoring section of the mixing console, and as before is independent of the control room monitor mix and the levels being sent to the tape recorder.

Due to the style of the drum part during the beginning and at various places throughout the song, the hi-hat rhythm and snare drum hit come so close together in timing that each sounds cleaner when played seperately. Therefore, the snare drum part was recorded in an overdub session. The drummer listened to the pre-recorded tracks through the headphones and played only the snare part. By this method, a solid recording of both the snare drum and the hi-hat part can be achieved for use in the final mix.

Another overdub session was conducted during which the lead guitar part was recorded. Instead of using the standard amplifier/speaker combination, however, a Rockman was used to amplify the guitar output. This Rockman is a very small guitar amplifier which can be attached to the musician's belt. The output of this amplifier is fed to a set of headphones which allows the musician to practice without creating the loud room volume necessary with a conventional amplifier/speaker combination. A specialized adapter was used to convert the headphone output on the Rockman to a microphone input. The signal was then sent through a microphone line and direct injected into the mixing console. There are several benefits to this method of overdubbing the lead guitar part. First, since the Rockman is the amplifier and is then direct injected into the board, there is no need for an amplifier/speaker combination nor an open mic in the studio. Therefore, the musician can monitor the pre-recorded tracks and the sound of his guitar through the studio speakers. This eliminates the need for the musician to wear headphones. In addition, while the system acts like a direct box recording, the sonic quality of the signal still resembles that of a guitar being played through an amplifier/speaker.

One of the disadvantages of using the Rockman to record from is its introduction of extra noise into the system. When not being played, the Rockman contributes a significant amount of noise to the recording. Since the lead guitar, being recorded through the Rockman was only present during portions of the song, it was not necessary to record on that track through the entire piece. Therefore, a process known as "punching in" was used during the lead guitar overdub. The theory of punching in an instrument revolves around the recording of a track only at specific parts of the song. With the tape recorder set up for an overdub, the pre-recorded tracks roll in the play mode. A few seconds prior to where the lead guitar part is to come in, the record button is depressed and the tape recorder begins recording on the overdubbed tracks. When that specific part is finished, the play button is depressed and the machine continues to play the pre-recorded tracks with the record circuitry disengaged. In this same manner the lead guitar part was punched into each specific place where it was to appear.

During another overdub session, the vocal parts were added to the recording. This song consisted of a lead vocal part with a back up vocal spaced throughout and four-part harmony in two separate places. Each of the four vocalists were recorded on separate tracks to allow processing of each voice independently. The lead vocalists' part was recorded first. Then, in turn, each of the remaining three vocal tracks were recorded on a separate overdub pass of the tape. This was done primarily to assure that each vocal track was free of stray signals, such as those that may have been picked up from an additional vocalist singing elsewhere in the studio. All four vocal tracks were recorded using the same microphone, a Neumann U87. This is a studio condenser microphone which tends to produce a flat frequency response. That is, there is relatively little variance in the reproduction level of all of the frequencies along the audible frequency spectrum.

Once all of the parts were recorded at good levels on the 24-track master tape, the recording was ready to move into the mixing phase. No additional instrument or vocal parts were added after this point.

CHAPTER III

MULTI-CHANNEL AUDIO MIXING

In its most crude definition, the mixing process revolves around determining the levels of each instrument and voice which will ultimately create the final sound. The art of mixing, however, goes much beyond this definition. The use of various pieces of signal processing equipment can enhance and transform the sounds initially recorded. Indeed, the creation of the sonic dimensions of height, width and depth revolve around maticulous uses of signal processing.

Ambient sounds present in a normal performance setting helps the listener to establish the size and makeup of the room, which ultimately influences the sound of the music. Large gymnasiums, for instance, offer a much larger amount of echo and reverberation then do chamber music halls. Also influencing the sound which the listener receives are the materials used to create and fill the performance environment. Flat brick or cement walls will have a large amount of sound reflection, especially of higher frequencies. Rooms filled with padded chairs will have less sound reflection with higher frequencies being absorbed by the furniture, therefore, leaving a more bassy sound.

The manner in which the multi-track tape of The 80's was recorded offers a very clean and stark sound. That is, the use of direct injection recording and close micing techniques do not detect the ambient sounds of the room. Rather, these techniques present each instrument as it would be heard at a very close distance, that of the microphone. Hearing an instrument at a matter of inches from it is a very unnatural listening distance. Normally the listener is present in the room where the musicians are performing. Therefore, the sound of the music takes on the ambient characteristic of the performance environment. An audience can judge the relative distance of each instrument by the length of its echo. With direct injection and close micing techniques these room sounds are lost. Therefore, without signal processing, the listener would be presented with a very stark and dimensionless sound. Because of this characteristic of multi-track recording, it is necessary for the audio producer to artifically create the ambient sounds which would normally be present in a performance environment. These ambient sounds will then establish for the listener, the dimensions of the music as it is heard. The re-creation of the performance of the band as a unit, including the room ambience of the performance setting, is the central focus of the mixing process.

To begin the mixing process, the channels on the mixing console which correspond to the pre-recorded tracks are placed in the line position and the mix mode. This mode enables the producer/engineer to hear and manipulate the sounds of each track independently and to compile the signals into a final output. Two separate output forms, stereo and mono, were created in individual mixing sessions for this thesis. First, the creation of the stereo mix will be addressed.

The rhythm tracks, that is drums and bass and rhythm guitars, are most consistantly played instruments throughout a piece. Since the presence of these instruments is fairly continuous they are a good place to begin the mixing process. In creating an initial balance among signals, the drum tracks were dealt with first. During the snare overdub session, a series of three drum hits were accidently left out of the overdub track. In any recording session, a wide range of variables are responsible for the sound recorded in that particular session. With the snare drum part specifically things such as the tenacity of the drum head and the particular tuning set that day can cause variance in the instrument itself. In addition, the exact position of the microphone recording the snare could vary slightly and change the sound detected. The tape machine may have built up dirt on the record head during the original session which would color the recording of those tracks, again causing a difference in sound. In addition, the changes in mood, manner and playing style of the musician would also induce differences between sessions. For these reasons, it would be nearly impossible to call back the drummer to play those three missing beats and have their sound be consistant with the remaining snare overdub track. Therefore, another method of replacing those three missing beats must be chosen.

The best possibility for maintaining sonic consistancy would be to choose three drum beats recorded elsewhere in the song during the same session. While variance may still exist within playing style as the song progresses, many of the other variables mentioned will remain quite consistant. The rhythm and timing of the three missing beats may appear in several locations throughout the song. However, the drum patterns flanking those specific beats will effect the way they are played, and ultimately the way they are recorded. Therefore, it was necessary to locate three beats containing not only the correct rhythm and timing, but actually surrounded by the same parts. Fortunately, the three beats which were missing were located in a portion of the song which was repeated three times throughout the tune.

The first step in the process of replacing the missing beats entailed making an audio copy of the snare drum part containing the piece needed. This is done by a process known as dubbing. During the dubbing process, the signal present on one tape recording is essentially copied onto another piece of tape on a different machine. The signal which needed to be dubbed was that of the snare drum overdub track. While the remaining pre-recorded tracks were necessary to locate the correct portion, they were not desirable on the copy. Therefore, it became necessary to separate the snare drum overdub track from the remaining tracks, while still being able to monitor the combined sound of all the instruments. Each channel has a control known as a panaramic potentiometer, or pan pot, which determines how much of that particular instrument will be distributed to the left and the right channels of the overall mix. Using these pan pots, the channel containing the snare overdub track was routed entirely to the left channel. The remaining tracks on the master tape were routed entirely to the right channel. Therefore, the level of the snare overdub track could now be controlled independently by manipulating the level of the left channel. With the setup described, a dub was made onto another tape machine of the portion of the song containing the snare drum beats.

The next step was to replace only the three missing snare drum beats from the left channel of the dub into the appropriate spot of the snare overdub track on the 24-track master tape. This was to be done using the same method of "punching in" as described in the overdubbing discussion in Chapter II. The two machines were to roll synchronously in the play mode, and the 24-track tape machine was to be punched into record only during the empty portion where the three drum beats were to be placed. A problem arose, however, due to the inconsistancy in speed of the 2-track machine containing the snare dub and the 24-track tape recorder. Several practice dubs were performed while slightly adjusting the speed of the 2-track machine, using the variable speed control. The results were that the 24-track tape machine was two percent faster then the 2-track recorder during the length of the three beats needed. This variance was not consistant throughout the entire song, however, because as the two machines continued to play, they would vary in and out of synchronization with each other. This slight speed variance can be caused by a variety of factors within the tape machines themselves.[1] While changing speeds is not enough to create an audible pitch change, it does make tasks, such as the one engaged in here, more difficult due to its effect on the timing of the song.

Once the three missing beats were replaced in the snare overdub track, the next task was to balance the sound of the drum kit as a whole. Each channel contains a solo button. This button allows that particular track to be heard on the monitor system without the signals from all of the remaining tracks. This function, however, is operational on a circuit independent of the recording process. Therefore, soloing one particular channel, or group of channels does not effect the remaining tracks, except that they are muted from the monitoring system.[2] This function is useful for working with the equalization of any one track, or the balance of certain tracks relative to each other, for example two guitar tracks.

Using the solo button, it was determined that the bass drum needed more edge and punch. To achieve this the equalizer was set to increase 16 kHz 3db, 3kHz 4db and to drop 1.2 kHz by 3 db. The snare drum track was also equalized to give the snare more snap. The EQ setting here was +3 db at 16 kHz, + 7 db at 5 kHz and +7 db at .7kHz. A balance was reached between the level of the snare and that of the bass drum, and the snare overdub track was also added in. Due to the fact that the snare overdub was recorded at a lower level, it had to be set at 10 db higher than the snare track in order to achieve a volume balance of the snare drum between the two tracks.

The two overhead microphones responsible for the sound of the cymbols were brought in and balanced in level with the other drum tracks. The right overhead track was equalized with +3 db at 16 kHz, + 3db at k kHz, -6 db at 1.2 kHz and +3 db at 70 Hz. This equalization setting helped to tone down the snare parts leaking onto this track, and also to give more ring to the crash cymbol appearing on this side of the drum kit. No equalization was done to the left overhead track.

Using panoramic potentiometers or pan pots, the signal from any individual track can be routed through the left and right channels in varying amounts. Therefore, the pan pot determines where across the stereo spread the sound of the instrument or voice will appear to be coming from. Using the pan pots, the overhead left and right tracks were panned to their respective sides. Due to the musical arrangement of this tune, the hi-hat, which shows up in the right overhead track, is played alot throughout the song. As a result, the right side of the overhead tracks feels heavier, or louder than the left. Therefore, the right side is brought back slightly closer to the center, and decreased in level 2 db lower than the left overhead.

Next the rack and floor toms were added to the mix of the drum kit. These tracks were panned to the left and right sides of the stereo spread approximately half the distance. (See Figure 4)

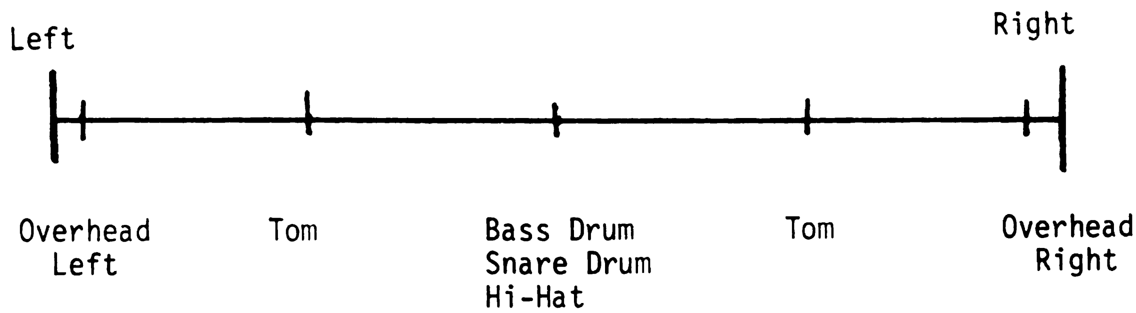


Figure 4. Stereo Drum Placement

The drum kit, as a whole, is then monitored for level balance among parts. When a satisfactory level balance had been achieved, artificial reverberation was added. For this project, a Lexicon Digital Reverberator was used, with a delay time set for 2.4 msec (milliseconds). That is, the reverberation, or repeated echo of any signal sent to the digital reverb unit, would repeat and delay for 2.4 msec before it has decreased below being audible, or "died out." The reverb was processed through two separate channels to allow the panning of the final return to the left and right. This capability of having independent control of the left and right reverb returns can be used to create a more distant stereo sound by allowing the reverberation of specific instruments to come more priminantly from one side or the other of the sonic picture. By adding artificial reverberation to an instrument or voice, it can give the illusion of depth. That is, some sounds will appear to be further away from the listener than others. This is due to the fact that sounds which have more of an echo to them are attributed to being further away. Conversely, these sounds which are heard with little or no associated reverberation are sensed as being closer. For the instruments which comprise the drum kit the largest amount of reverb was used on the overhead tracks. The reverb for the left overhead was brought up on the right side of the reverb returns. Likewise with the right overhead, the reverb was used on the left reverb return. This placement of reverberation can be best explained by visualizing the drum kit in a performance environment. The primary sound of the symbols on the left side of the drum kit will appear to the listener to be coming from their left. The sound wave from that symbol will also be reaching the listener in delayed echos after it has reflected back from the characteristics of the environment, such as the walls of the room. Due to the fact that the primary sound is much louder than these reflected signals or echoes, those refelections will be most obvious from directions other than that from which the primary sound is coming. Therefore, to replicate this phenomenon in the mixing process, the artificial reverberation should be placed most prominantly in the locations where those reflected sounds would appear. The snare track was also given reverberation. However, since the snare drum was placed in the center of the stereo spread, its reverberation was returned equally to both the left and right channels. The snare reverb was less than used on the overhead tracks. Reverb was not used on the snare overdub track, bass drum track or tom tracks. These tracks were left "dry," or without reverberation, to give them a stronger presence, that is, a closer feeling within the drum kit and ultimately the final mix.

The bass guitar, another rhythm instrument, was then brought into the mix. In order to establish a more punchy and solid sound, 3 db was added at 14 kHz and 6 db was added at 250 Hz. While the level is balanced with all other tracks in the mix, it should be carefully checked in balance with the bass drum. Since these two instruments have their primary tones in a very similar frequency range, it is important to see that they are distinct enough to differentiate between them in the final mix. This is known as separation and applies to any instruments with similar harmonic structures. The bass guitar is panned to the very center of the stereo spread. This is an optimal place for this instrument because of its very loud, low frequencies, it would cause the two channels to feel unbalanced if placed on either side of the stereo spread. In addition, the needle track on the record album contains the left and right signal information on the two sides of the groove. Should one side be radically different in level than the other, the groove would be cut very lopsided and cause the playback needle to mistrack, or skip.

Next the two rhythm guitars were added to the mix. The first rhythm guitar was panned to the center and was not processed in any manner. The second rhythm guitar was responsible for playing the "hook" of the song. This hook is one particular musical phrase which is repeated several times within the tune. The hook is hoped to be an identification tag for that particular piece, such that an audience would be able to hear the hook, and recognize the song. At several points throughout the tune, the hook is played with very sparce instrumentation. In order to help establish the hook as an identification tag it needed to receive special emphasis and body, to set it apart from the remainder of the song. To help fill out the sound of this guitar, a digital delay unit was used, in this case, the Lexicon Prime Time. This digital delay unit differs from a digital reverb unit in that it repeats each signal only once, while the reverb unit repeats each signal several times with a continual decay over time.

By processing the guitar playing the hook through the Prime Time Delay Unit, the illusion of two separate guitars, or one guitar coming from two separate directions can be created. A delay time of 11 msec was chosen for the repeat of the signal. The choice of 11 msec is based on two separate reasons. First, this short of a delay time is too close in time to the original signal to be perceived by the listener as an echo. Therefore, the two will not appear to be an original and a reflected sound, but rather two replications of the same part, played simultaneously. This allows these two signals to be placed opposite sides of the stereo spread to give a larger sound to the guitar. A potential problem arises, however, when these two signals are separated in this manner, that is the possibility of frequency cancellation. The second reason for the choice of the 11 msec delay time is this potential for frequency cancellation. Being a prime number, 11 is not evenly divisable by any number except one and itself. Therefore, the cancellation of equal but opposite frequencies in the harmonic structure of the guitar is kept to the lowest possible amount by using a prime delay time. By taking the original and the delayed guitar signals and placing them on the opposite sides of the stereo spread the guitar appeared to have a fuller and fatter guitar sound.

The remaining guitar part was that of the lead guitar which was recorded through the Rockman. The manner in which this guitar was recorded offers several additional considerations. First, the Rockman delivers a stereo output, that is, a separate left and right channel. Therefore, it was necessary to record the Rockman on two separate tracks which contain within them some left and right movement. These tracks are panned to the left and right side of the stereo spread to allow this recorded movement to be heard. Also, the Rockman induced into the recorded tracks a significant amount of hiss and unwanted noise. For this reason it became necessary to set up a mute on the Rockman tracks. This would allow those tracks to be turned off, with the push of one button, when the lead guitar was not playing. Therefore, this additional hiss and noise would not be present throughout the entire song. Most importantly, the mute was used for the a capella portions of the tune, where excess noise became extremely noticable. Artificial reverberation was added to the lead guitar tracks in the same manner as the overhead tracks. That is, reverb to the left on the right track and reverb to the right on the left track. While the reverb on the lead guitar did offer some distance to its sound, the guitar still appeared to be very upfront. This was due partially to a low amount of reverb, but also to the nature of the solos and the very high pitch cutting sound of the guitar itself.

The final tracks added to the music mix were the vocal tracks. There are four vocal parts present throughout the song. First the lead vocal part was placed in the mix at a level which caused this voice to be present above the entire music mix, or louder than the instrumentation. The lead vocal is placed in the center of the stereo spread. Since the relative volume of this track is higher than all others, this not only causes the lead singer to appear in the center of the sonic stage, it is also useful in maintaining equal levels to the left and right channels.

In the recording setting, the microphone acts like the listener's ear. That is, it transmits for recording what the listener will ultimately hear. While a vocalist generally sings into a microphone at a distance of only a few inches, this is a very unnatural distance for an audience to be listening at. Therefore, artificial reverberation is particularly important on the vocal tracks. A substantially larger amount of reverb can be added to a vocal track than to an instrumental track before the vocalist will appear to be far away from the listener. This reverb not only serves to back the vocalist off to a more natural distance, it also recreates the room ambience lost in the recording process. Therefore, a fairly large amount of reverb was added to the lead vocal tracks. This reverb was distributed evenly between the left and right channels.

The second vocal part, that of the high harmony, is present throughout the song in addition to the lead vocal part. This vocal part was processed similar to the guitar part in the hook of the song, through a digital delay unit, to provide a second duplication of the same part. The delay time for this voice was chosen to be 11 msec for the reasons stated earlier. The two duplications were then panned slightly to either side of center, and artificial reverberation was added to each. This method of using the harmony serves two purposes. The high harmony part often feels very thin in relation to the other vocal tracks due to the frequency, pitch and performance style of the part. By returning the second duplication opposite the original, the result is a fuller and thicker sound because the part has now been doubled. The final sound will be the lead singer with the high harmony part appearing to be coming from slightly behind him. Since the high harmony part is meant to supplement the lead, placement directly behind the lead singer will be less distracting and more enhancing. If, however, this high harmony part were placed to one side or the other of the stereo spread, it would draw the listeners attention as a second separate voice. This would be disrupting due to the sparce presence of the high harmony.

The remaining two vocal tracks are present only during the four-part harmony segments. Therefore, the two parts were panned to opposite sides of the stereo spread and artificial reverb was added. Therefore, during the portions of the song when these vocal parts are present they will appear to balance the sonic layout.

During the entire mixing process, the overall levels of the stereo signal being created must be monitored. While each individual track has been recorded at good levels, that is, -5 to +1 db from 0 VU, the summation of different amounts of these tracks must also be kept in the -5 to +1 db range. One major aspect of the mixing process is determining how much level of each track will be included in the final sound. Therefore, adjustments must be continually made to assure a relatively even distribution of the resultant signal to both the left and right channels.

Once the producer has established a mix between the various instruments and voices which he/she is satisified with, there are several conditions of the listening environment which must be taken into consideration. Both the volume of the playback and the quality of the speakers used to monitor the sound effect what the listener will hear.

For example, the monitor speakers in most recording studios are of very good quality. The listening public, however, may have a wide range of different quality speakers. Indeed, transistor radios and most clock radios have speakers of quite poor quality. Therefore, it is advisable to check the mix on several systems with different quality playback speakers. Speakers of lower quality may cause the overall sound of the song to appear "tinnie" and bass deficient.

According to the Fletcher-Munson Equal-Loudness Contours, at any constant playback level, different frequency tones will appear to be different levels of loudness.[3] For example, as volume decreases, low frequency tones drop in loudness much more rapidly than the higher frequency tones. Since it cannot be predetermined at what level the audience will listen to the music, the song should be mixed so that all parts can be heard at a wide range of playback volumes. While control rooms are often kept at a very high volume while mixing, the mix should be checked also at very low volumes. At these lower volumes special attention should be paid to the bass instruments because of their characteristic shifts on the equal-loudness contours.

The final dub onto the stereo master also included the recording of calibration tones. Thirty seconds of tone at 10,000 Hz, 30 seconds of tone at 1,000 Hz and also at 100 Hz were recorded onto the head of the tape. These calibration tones were to be used when transferring the audio onto the video tape. This provided a reference point for the equalization of the original audio recording. During the production process it became apparent that a 3 db loss at 10,000 Hz was suffered by the audio track each time it was dubbed or transferred onto video tape. This loss severely alters the sonic dimensions being studied in this project. Therefore, these calibration tones allowed the signal to be equalized in the playback process to accommodate for this frequency decrease. The signal to noise ratio of videotape is considerably less than that for audio, therefore, while the equalization of the playback restored the frequency makeup of the song, the noise level was much higher coming off of the videotape.

With stereo version complete, a monaural mix needed to be created for use during the testing phase of this research. This began by placing all of the pan pots to the center position, thereby placing all of the tracks in the very middle of the spread. Since their main purpose was to create width, the delayed tracks for the high harmony and guitar were dropped out entirely, leaving only the original tracks of each. With all of the instruments compiled into the center, the levels of many of the tracks had to be adjusted. For instance, the voices and guitars which had been placed to the right and left needed to be increased in level to maintain the mix once they were placed into the monaural format. The equalization of each instrument remained the same. However, the artificial reverberation had to be adjusted to accommodate for the fact that all of the parts were now placed in the center. As with the stereo version, the levels of the final mix should remain within the -5 to +1 db from 0 VU range. Also with mono, different quality speakers and different playback volumes were checked for sonic consistency.

NOTES--CHAPTER III

[1]John M. Woram, The Recording Studio Handbook, (New York: 1976), p. 290.

[2]Ibid., p. 258-259.

[3]Robert E. Runstein, Modern Recording Techniques, (Indianapolis: 1974), p. 31-32.

CHAPTER IV

VIDEO PRODUCTION

A key factor in determining a listener's perception of the sonic dimensions of depth, width and heigth is the transmission medium involved in the listening experience. The traditional mediums encountered by audio producers have recently expanded to include the video realm. Promotional music videos are taking the work of the audio producer and supplementing it with a video counterpart.

In order to incorporate this transmission medium in the study at hand, a simple music video was produced to corresponds with the musical selection used in audio production process. This music video was produced after the audio production was completed. The sequence of production steps used in this thesis were, therefore, reflective of those implemented in an industry setting.

The video was produced on 3/4" videotape and shooting was done at a ratio of approximately 20:1. The production involved in creating the music video used in this thesis can be divided into four separate divisions.

1. Preproduction/Planning
2. Videotaping of Performance Segments
3. Videotaping of Concept Segments
4. Postproduction/Editing

Preproduction/Planning

The first stage of the preproduction phase was to establish a concept and tone which would dominate the final product. The song, Restless Hearts, contains the following lyrical content.

> Restless Hearts Beat Strong
> Restless Dreams Die Hard
> You Can't Hide
> Your Broken Heart
> You Can't Hide
> Your Broken Dreams
> Restless Hearts Beat Strong
> Restless Hearts Beat Strong

The concept developed revolves around the lead singer Dan Keough's search for his girlfriend, Debbie, who is a ballerina. As the storyline progresses, it becomes apparent that she has died. The focus of the cutaways, which contained the concept, was to be his memories of her and the places she used to be. He also has in his possession a film of her dancing which he watches obsessively. The final scene of the video was to be Dan walking past a woman who he thinks is his girlfriend, with the final shot being a freeze frame, closeup of his confused expression.

A rough script outlining the progression and basic content of the cutaways was written. Throughout the video, Dan would have with him the pink ribbon from Debbie's ballerina slippers. The ribbon would provide a link between present action, where it is in his possession, and memory, where the ribbon is still connected to Debbie's slipper.

Four different locations were chosen to videotape the cutaways These were a dance studio, a cemetery, a parking ramp and a projection room in which Dan would watch his film of Debbie dancing. Nancie Bauer Dance Studio in Holt was chosen for videotaping of the dance scenes. The cemetery on Mt. Hope Road in Okemos provided the

the setting for the cemetery scenes and a projection room was created in the Communication Arts Building. The closing scenes of the video were shot outside a parking ramp in East Lansing.

To maintain continuity between the storyline and the production work, the performance portion of the video was scheduled to be done with limbo lighting. This lighting effect would create a feeling that the action was taking place in a void by highlighting the band in a very dark background.

The performance portion of the video was scheduled for shooting in video Studio E. The band would be arranged in the studio under limbo lighting and would panomine the music. The song was to be taped in its entirety, several times, and the best of these takes would be used as a whole unit upon which to assemble the final project. This method was chosen as the simplist and most precise way of maintaining the best possible timing between the audio and the video portions of the signal. Another method which could have been selected would have been to shoot the video in several small pieces, each containing their own audio. These pieces would have then required assembling during the editing phase. While certainly not impossible, the difficulty of performing music edits on a video editing machine is very great. Another possible option which was refused would have been to shoot the several segments, devoid of any audio track. Once the video track was complete, the audio could have been dubbed onto the videotape. The primary reason why this method was rejected is due to the variance in operating speed of an audio tape recorder. The variance is due to the fact that there is no controlling meter for the tape machine to lock onto which would force a consistent tape transport speed.

Videotaping of Performance Segments

The band was arranged for the performance portion of the video in a circle located in the middle of the studio with all band members facing toward the center. (See Figure 5). This set up provided several benefits to the project. First it allowed each of the band members to be placed in a tight circle of light. By having them set away from the background and curtains, they could each be lit with tightly spotted lights, so as to create the high contrast effect desired. This special lighting, in combination with their placement, allowed cameras to be located around the circumference of the group in the dark, or void. Therefore, shots of each band member could be constructed from several different angles. In addition, this configuration did not allow for any shot to contain more than two of the band members at one time. In order to establish the listener's ability to locate instruments within the sonic layout it was important that the video not suggest the location of particular instruments in relation to other instruments. For example, one question asked the subjects to state if the bass guitar was closer to them or further away from them than the lead singer. By never showing the lead singer and the bass guitar in the same picture, there are no video cues to the sonic placement. Therefore, the subjects who saw the video were forced to rely on the audio signal to respond to the questionnaire, as were the subjects who did not see the video.

During the videotaping, a pair of auxiliary speakers were placed in the studio from which the band would hear the pre-recorded tape. Since the snare drum and symbols are acoustic instruments, when being played for the videotaping session, these instruments would actually be

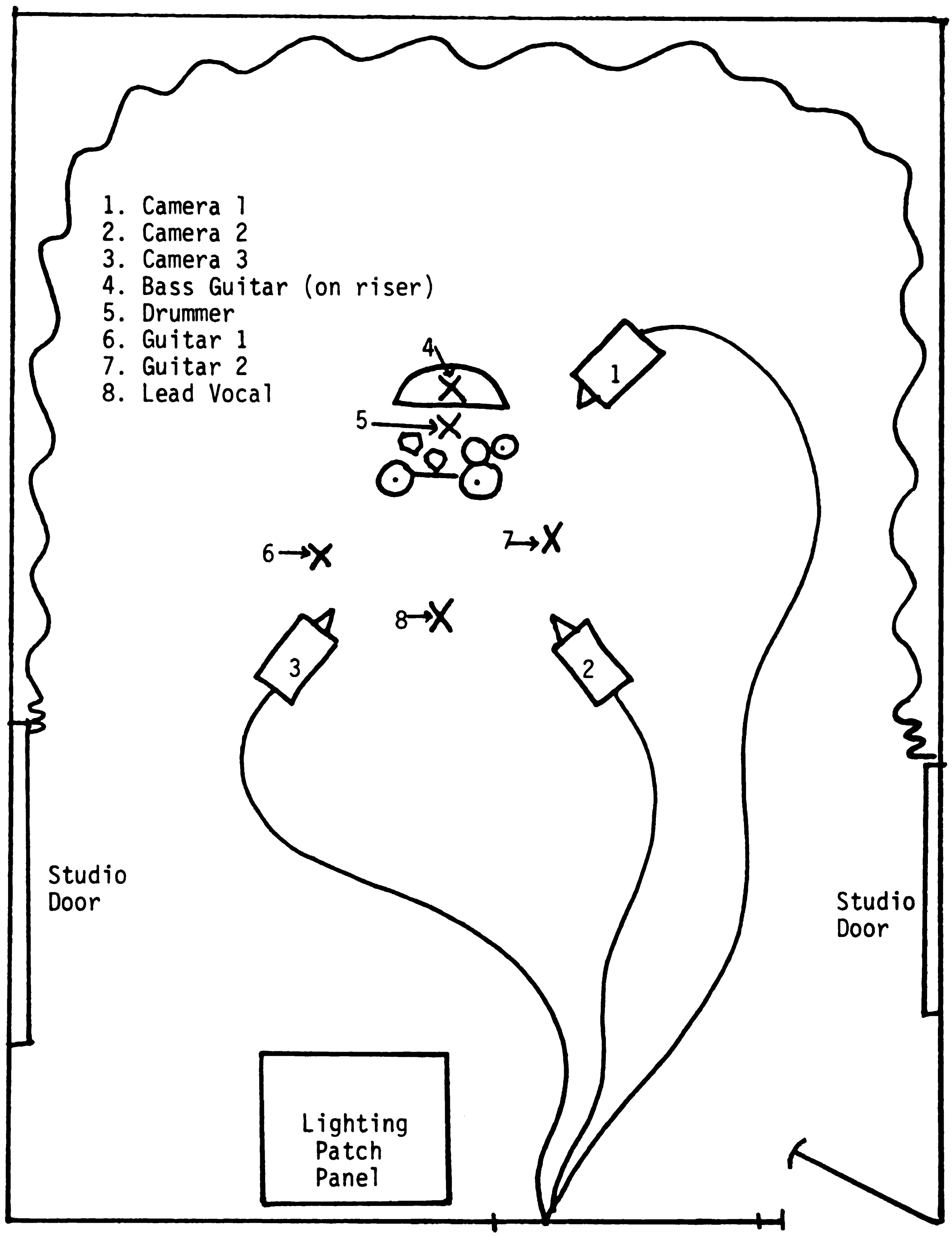


Figure 5. Video Studio Layout

heard in the studio. Therefore, it was necessary for the volume of the auxiliary speakers to be fairly loud in order for the instruments audible only on the tape to be heard over the snare drum and cymbols. In addition, the louder volume would create for the musicians, the feeling of a live performance. This is due to the fact that electronic instruments cannot be heard without an amplifier/speaker combination, and musicians become accustomed to hearing their instrument from speakers in a performance setting.

During several run throughs the band bacame more comfortable with the studio setting and the practice of panomining. These run throughs also helped solidify the best locations for each of the three cameras. With the necessity that the video be shot in its entirety each time, and the full production being only 4 minutes and 11 seconds long, it was too difficult to move the cameras between different physical locations. Therefore, the locations were established where each camera could shoot the widest variety of well composed shots.

Before the video recording could begin, color bars were recorded onto the head of the videotape in combination with the calibration tones from the pre-recorded audio tape, those of 10,000 Hz, 1,000 Hz, and 100 Hz. As discussed in Chapter III, these tones provide an equalization reference which can be used to calibrate the playback to the originally recorded equalization. As mentioned earlier, this equalization calibration is paramount in this study, due to the fact that the dimensions being tested suffer severe degradation with the loss of the high frequency signals.

Once the crew and musicians were fairly comfortable with the flow of the process, the recording began. The performance was recorded onto videotape five times, each marked with a character generated slate. Any of these recordings were of a quality which could have been used to create the final project. In addition, they provided several different combinations of shots from which to choose in creating the final piece. For example, the performance video which was selected to build the production upon did not contain any shots of the bass player by himself. Therefore, shots from one of the other four takes containing only the bass player were edited into the final video.

Also during this recording session two cutaways were recorded which needed to be in time with the audio. The first was that of a foot tapping in time with the beat, and the second, a closeup of the snare drum part during the introduction of the song. Both of these were to be edited into the beginning of the video and needed to be shot with the music running to assure consistant timing.

Videotaping of Concept Segments

The cutaways containing the concept portion of the video were taped on three different days. The first session, taping those segments in the dance studio, was done the day before the performance session. This session contained the taping of several specific ballet movements as well as improvisation by Debbie Miller, the ballerina. Also recorded at the dance studio were various shots of Dan walking along the barr and around the empty studio, as if to be looking for Debbie.

The second videotaping session of cutaways where the shots to comprise the ending segment outside the parking ramp. The projection room was created in a classroom in the Communication Arts Building. One movie light covered with a blue gel, gave the impression that the room light level was very low. The light from an overhead projector was then focused on Dan's face to illuminate his expressions without adding an excess amount of light to the overall room. Scenes shot in this setting were videotaped with Dan wearing several different shirts. This was done to give the impression that he watched the film of her dancing on several different occasions. No "film" was actually ever created of Debbie dancing. During the videotaping sessions in the projection room, Dan actually viewed a drivers education film. The image of Debbie dancing, as seen in the final project, was electronically added during postproduction. In order to accomplish this a video cassette machine and monitor were placed in the darkened television studio, and the monitor was wrapped in a black cloth with only the screen showing. A camera was then used to create a picture with the monitor screen in the center of the frame. This monitor played the videotape of Debbie dancing. In the control room, the videotape of Dan watching the movie screen was run through the switches, and a wipe was used which placed the monitor from inside the studio on the portion of the picture where the movie screen was on the projection room tape. With both tapes rolling, a dub was made onto a third videocassette. The result was the scene in the video where Dan appears to be watching a "film" of Debbie dancing.

The final videotaping session, in the last setting, was that of the cemetery. Shots of the cemetery were composed viewing the backs of the tombstones. This was done to avoid any possible complications with names which would appear on the front of the markers.

Postproduction/Editing

Once all of the videotaping was completed and the contents of the final project were assembled on four separate videocassettes, the postproduction process began. The primary components of the post-production phase are the logging of the videotapes and the actual editing itself. In order to determine where the cutaways should appear, a detailed log of the performance video to be used was created. The segments to be kept as performance segments in the final project were noted and the places where cutaways were to appear were timed. From detailed logs of the cutaway tapes, those shots to be used in the final production were chosen, and a final production script was derived.

The final videotape was done using insert edits. The video portion of the cutaways were inserted over the video signal of the performance tape at the appropriate places. Therefore, the audio track of the performance tape was uninterrupted through the entire video. This editing was carried out on the master tape of the performance recording. This was done in order to spare the signal degradation which would have occurred by dubbing the performance recording to another tape to start with.

The finished video was then dubbed off through the switcher where the freeze frame was implemented on the end. In addition, the character generator provided a title slate and countdown on the front, and credits were added after the ending fade to black.

CHAPTER V

METHODOLOGY

The audio recordings created with this project demonstrate the sonic dimension of width, depth, and height. Together, these dimensions create a size to the music. The research component of this thesis tested the listener's perception of these dimensions and their effect on how well the listener liked the song. Nine research questions comprise the core of this study.

1. Do the sonic dimensions of depth, width, and height contribute to a listener's liking of a song?
2. Do the playback formats of stereo, mono and bi-polar mono effect the perception of these dimensions?
3. Does the quality of speaker used in playback effect the perception of these dimensions?
4. Does the audio knowledge of the individual effect the perceptions of these sonic dimensions?
5. Does the addition of a video signal to the audio signal, as in a music video, effect the perceptions of these dimensions?
6. Does the playback format of stereo, mono, or bi-polar mono effect the listener's liking of the song?

7. Does the quality of the playback speakers effect the listener's liking of the song?
8. Does an increase in audio knowledge of the individual cause an increase in the individuals liking of the song?
9. Does an added video signal, such as is present in a music video effect the liking of the song?

These research questions focus around four separate independent and four separate dependent variables.

Independent Variables

1. Playback Format: This was tested using three conditions: stereo, mono, and bi-polar mono. Stereo was represented by a distinct left and right channel from two different speakers, mono was one signal/ one speaker, and bi-polar mono is a foil by which listeners hear two speakers sending the same signal.

2. Speaker Quality: This variable was operationalized by offering different subjects the same audio tracks through different quality speakers. Two quality distinctions were made: good quality, similar to those that would be purchased with a good stereo system, and poor quality, which would be similar to those available in factory installed car radios.

3. Audio or Video: Approximately half of the subjects heard the recording as if they were listening to a record or tape. The remainder of the respondants heard the production in connection with the music video.

4. Audio Knowledge Level: The subjects were classified according to their level of audio production knowledge. Amateur listeners were

considered to be representative of the general public. The next classification, audiophiles, express an intense interest in audio, but do not possess the knowledge and/or experience to be considered professionals in the field, and finally professional audio producers who are trained and active in the audio industry.

The dependent variables involved in this study were examined by creating indexes to describe each respondent's perception of the sonic dimensions of depth, width and height. The fourth dependent variable was an index measuring the degree to which the subject liked the song.

The following hypothesis were derived from the research questions and variable lists.

1. The dimensions of height, width and depth contribute to the overall size of the sound.

2. Listeners will be able to correctly identify more instruments when placed across a stereo spread than when all the instruments are panned to the center, in mono.

3. Listeners sonic dimension index scores will be higher with a stereo playback format than with that of bi-polar mono or mono.

4. Listeners will have a higher liking index score from stereo playback formats than for bi-polar mono of mono playback formats.

5. Good speakers will produce higher sonic dimension index scores than poor speakers.

6. Good speakers will produce higher liking index scores than poor speakers.

7. As the audio knowledge of the individual increases, the sonic dimension index scores will raise.

8. As the audio knowledge level of the individual raises, the liking index will be more positively correlated with the sonic dimension index scores.

9. The addition of a video signal to the audio signal will not effect the sonic dimension index scores.

10. The addition of a video signal to the audio signal will not effect the liking index score.

As an initial step toward devising a procedure to test these hypothesis, a sample questionnaire was drafted and pretested on a group of individuals with different levels of audio production knowledge, who have all taken graduate level research courses. Upon completion of the pretest, a focus group discussion took place during which critiques of the questionnaire and proposed study methods were given. A revision of the questionnaire was completed and the testing conditions were established.

Subjects were solicited from the basic audio production and advanced audio production courses at Michigan State University. In addition, a group of four industry professionals were enlisted to participate in this study. Those subjects enrolled in the basic audio production course were classified as amateurs and those in the advanced course as audiophiles. The total sample size tested was 74.

Subjects were tested in groups of four or less. They were seated in chairs in the center of a recording studio with both sets of speakers and the television monitor in front of them. Regardless of which condition the respondents were offered, the same equipment was always present. All possible combinations of the independent variable of playback format, speaker quality, and presence or absence of video were administered to the amateur group. Due to a limitation in group size, the audiophile group were tested under the conditions of good quality speakers, as stereo without video, stereo with video and bi-mono without video. Also, due to the size limitation, the professional group was tested only under the "good speaker, stereo audio, with video" condition. (Appendix C contains a complete list of testing conditions administered.)

Each group was read the following statement prior to the beginning of the test.

> "This project deals with the various dimensions which music can take on, such as width, depth and height--the overall size of the sound. This study will help to determine a listener's perception of these dimensions. I will play this song (video) for you and then come back in and give you a questionnaire. You will have a few minutes to look through the questionnaire, and then I will play the song (video) for you again, and you may work on the questionnaire while it is playing."

Subjects were allowed approximately three minutes to look through the questionnaire before the second administration of the testing. They were then given as much time as needed to complete the questionnaire.

CHAPTER VI

QUESTIONNAIRE DESIGN

The questionnaire to be administered during the testing phase of this thesis was designed to contain several different types of questions. For instance, each of the sonic dimension indexes is derived from the average of a pictoral question and a semantic differential. In addition, Likert scales and open-ended questions are also a part of the design.

This explanation of the questionnaire will deal with the instruments used to test the video groups. Items 19 and 20 were omitted on the questionnaire administered to the audio only groups. Aside from this exception, the two questionnaires were identical.

1. Sounds often appear as if they are coming from different locations, as if the band were standing in front of you. On the line provided, please place an "X" at the place where the bass drum sounds like it is located.

LEFT CENTER RIGHT

Regardless of what condition was given to the subjects, the bass drum is always panned to the center. Therefore, this questions serves to check the accuracy of the listeners bi-aural perception.

2. Please list the instruments that you can identify in this song.

It is hoped that listeners will be able to distinguish more instruments when they are spread out across a stereo width, than when compiled in the center of a mono, or bi-polar mono playback format.

3. How many guitars do you hear? __________

   Because of the delay methods used on the one guitar in the stereo version it may appear to the listener to be two different guitars. Therefore, there is a possibility that the listener would hear more guitars in the stereo version than in the mono or bi-polar mono versions.

4. Music creates a feeling of <u>depth</u>. Various instruments or voices seem further from you than others. With the lead singer as a point of reference, please mark the column signifying whether each instrument is closer than the lead signer or further from you than the lead singer.

	CLOSER	FURTHER
a. bass guitar	_____	_____
b. cymbols	_____	_____
c. drums	_____	_____

   This question will guage a listener's perception of depth when given a point of reference in the mix, that of the lead singer.

5. If the band were standing in front of you performing, place an "X" on the line provided at the location where you hear the guitar solo coming from.

LEFT CENTER RIGHT

   Due to the separate left and right channels of the Rockman used to record the lead guitar, it should appear to be in different locations on the stereo version than on the mono and bi-polar mono versions.

6. How many voices do you hear? ________

   Again due to the delay method used on the high harmony, the stereo version may appear to have more voices than the mono or bi-polar mono versions.

7. Place an "X" on the line provided where you hear each of these voices. You may use as many "X's" as necessary.

LEFT CENTER RIGHT

7. (continued)

   In the stereo version, the voices are spread across the entire width of the stereo spread, while they are all in the center of the mono, bi-polar mono versions.

8. Do any of the instruments appear as if they move from side to side?

   _____ Yes

   _____ No

   _____ Don't Know

   If you answered yes, which instrument(s)?

   ____

   In the stereo version, at one point, a guitar is panned from side to side. This does not occur in the other versions. If the listener says any instrument other than a guitar moved, their "yes" response to movement may be of questionable validity.

9. Music creates a feeling of <u>heigth</u>. Considering the entire song, place an "X" indicating how far toward the ceiling, or how tall the music feels to be.

   CEILING

   FLOOR

   This question gives the listener a pictoral description of their perception of height.

10. If the fullest <u>depth</u> the music could exibit were the distance between you and the far wall, place an "X" on the line provided which indicates how much depth this song has.

   WALL

   YOU

10. (continued)

    This item gives the subject a pictoral description of their perception of depth.

11. Music creates a feeling of <u>width</u>. Given the two side walls of the room as <u>boundaries</u>, please darken in the portion of the line which you feel represents the width of this song. (For example, a very narrow sound would have a small length of the line darkened in, probably toward the center, while a wide sound would have a larger portion of the line darkened).

    WALL ____ WALL

    Once again, item 11 gives the subject a pictoral description, this time for their perception of width.

12. For each of the following pairs, please place an "X" on the line which most appropriately expresses your response to the statement:

    The sound of the music seemed to be:

    a. ____ ____ ____ ____ ____ ____ ____
       TALL SHORT

    b. ____ ____ ____ ____ ____ ____ ____
       WIDE NARROW

    c. ____ ____ ____ ____ ____ ____ ____
       SHALLOW DEEP

    d. ____ ____ ____ ____ ____ ____ ____
       LARGE SMALL

    This item offers semantic differentials for depth width, height and overall size. The respective semantics are averaged with their appropriate pictoral descriptions to derive the depth, width, and height index score for each individual.

		STRONGLY AGREE						STRONGLY DISAGREE
13.	I liked the song I just heard.	1	2	3	4	5	6	7
14.	I liked the sound of the group playing this song.	1	2	3	4	5	6	7
15.	I would like to hear more songs by this group.	1	2	3	4	5	6	7
16.	Given that money was not a factor, I would buy a record containing this song.	1	2	3	4	5	6	7
17.	I would <u>NOT</u> like this song played on my favorite radio station.	1	2	3	4	5	6	7
18.	I felt the sound quality of the music could have been improved.	1	2	3	4	5	6	7

These questions measure on Likert scales various aspects of a person's liking of the song. Items 17 and 18 are worded to give the opposite response of Items 13-16 so as to discourage response set.

19.	I liked the video I just saw.	1	2	3	4	5	6	7
20.	The video story was <u>NOT</u> related to the words in the song.	1	2	3	4	5	6	7

These two questions measure liking of video.

21.	The volume of the music was too loud.	1	2	3	4	5	6	7

The volume of the music could certainly effect the listener's enjoyment level. Therefore, this question solicites their opinion of the testing volume.

22.	The sound of the music seemed to fill the room.	1	2	3	4	5	6	7

Whether or not the sound of the music seems to fill the room may be related to volume, although it may also be related to the sonic dimension indexes. Item 22 offers the opportunity to test this idea.

23. In general, what did you think of the song you just heard? What aspects of the recording did you like and what didn't you like?

    This question solicites general opinions which may or may not have been covered in previous questions.

24. The type of music I listen to most often is (Please choose one).

    _____ Pop/Rock
    _____ Heavy Metal Rock
    _____ Easy Listening
    _____ Country
    _____ Jazz
    _____ Soul
    _____ Classical
    _____ Folk
    _____ Other

    The type of music that a subject listens to most frequently will probably effect their liking of this song. The 80's music could be considered pop/rock and may be totally unappealing to a classical listener.

25. Approximately how many hours each day do you spend listening to music?

    _____ Hours

26. Please divide your music listening time among the following categories--use percentages.

    _____ Records
    _____ Tapes
    _____ Radio
    _____ TV
    _____ Live

    (Must total 100%)

27. Approximately how many albums and tapes do you have in your personal music collection?

    _____ Albums
    _____ Tapes

27. (continued)

    These items help to determine if those who are avid music listeners have a better perception of the sonic dimensions than those who are not.

28. What is your age? ______ Years

29. What is your sex? _____ Male _____ Female

    Age and sex may also effect liking of the song. Particularly in the audio and video conditions where the sex and approximate age of the band members can be estimated.

30. From participating in this study, do you feel that in the future you will listen to music in a different way? If so, how?

    This will allow a comparison of those who say they always listen discriminately with their sonic dimension index scores.

Keeping in mind the various possible dimensions which music can take on, the dimensions of depth, width and height were studied within this project.

Table 2. Sonic Dimensions Used in Study

Dimensions	Creation Method	Measurement Method
Depth	The distance of the instruments and voices from the listener. This is created by variance in amount and equalization of artificial reverberation and also variance in level.	Average of pictoral question response, item 10, and semantic differential response, item 12e.
Width	The placement of the sounds across the stereo spread as determined by the panaramic potentiometer setting during the mix-down process.	Average of pictoral question response, item 11, and semantic differential response, item 12b.
Height	Emphasis given to very high and very low frequency tones within the music to create an illusion of heigth.	Average of pictoral question response, item 9, and semantic differential response, item 12a.

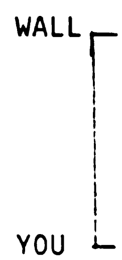
CHAPTER VII

ANALYSIS OF DATA

To begin analysis of the data received during the testing phase, the dependent variable scores had to be calculated. These variables describe the subjects' perception of depth, width and height, and also their level of liking of the song. For each respondent, three sonic dimension indexes were calculated to quantify the individual's perceptions. In addition to the depth, width and height indexes, a liking index was also compiled to describe the subjects' degree of liking of the song.

Each of the sonic dimension indexes is an average of two separate question responses from similar questions of different forms. For example, the depth index comes from the average of a semantic differential from "deep" to "shallow" and a pictoral question soliciting the respondent's perception of depth within the music. This pictoral question read:

> "If the fullest depth the music could exhibit were the distance between you and the far wall, place an "X" on the line provided which indicates how much depth this song has."



A key was created separating this distance line into segments in order to convert the pictoral response into numerical form. Likewise, the pictoral questions for height and width used the dimensions of the room as boundaries for the listener's perceptions. In the case of each of the sonic dimension indexes, the pictoral score was averaged with the semantic differential score which corresponded to that index. For their use in certain statistical programs available through the Statistical Package for the Social Sciences (SPSS), these index scores were rounded to the nearest whole number.

The liking index was derived from scores of six statements dealing with different aspects of an individual's liking of a song. The responses to these statements were measured on a seven-point Likert Scale anchored with "Strongly Agree" and "Strongly Disagree." The statements used to devise this index were:

> I liked the song I just heard.
>
> I liked the sound of the group playing the song.
>
> I would like to hear more songs by this group.
>
> Given that money was not a factor, I would buy a record containing this song.
>
> I would not like this song played on my favorite radio station.
>
> I felt the sound quality of the music could have been improved.

From the averaging of these responses, the liking index was calculated for each of the respondents and used to define their level of liking.

Each of the final indexes has a possible score of 1 to 7. In the case of the sonic dimension indexes, the higher scores correspond to a wider, deeper, or higher perception of the dimension. Likewise the higher the liking index, the greater the level of liking.

Each different testing condition was assigned a classification number. That number contained four digits which defined the combination of each of the four independent variables present in that condition. A Pearson Correlation between these classification numbers on the various conditions and the dependent variable indexes was calculated. Accepting significance levels of $P < .05$, one significant correlation emerged. The classification numbers were correlated with the width index, with a correlation coefficient of .2960 ($P = .005$).

With the classification numbers broken down into each of the independent variables of playback format, speaker quality and audio or video contingency tables were created with the various indexes (dependent variables). The width index again was significant in a Chi-square Test, this time with the playback format, that is, stereo, mono or bi-polar mono ($P < .05$).

A four-way analysis of variance between the independent and the dependent variables did not produce a significant amount of explained variance. A second four-way analysis of variance was calculated taking into consideration two possible covariates. These covariates were the type of music that the subjects usually listen to, and their feelings toward the playback volume of the music in the testing room. The amount of variance explained in this calculation was not significant either. However, these two tests pointed out possibly significant variance between the width index and the playback format and also between the liking index and the covariate of the type of music the subject prefers. A one-way analysis of variance was performed on each of these variable pairs.

The explained variance between the liking index and the type of music which the respondent usually listens to was significant at P=.033. Across the entire population, the subjects musical preference explained 22.3 percent of the variance in the liking index. Indeed, a Pearson Correlation between these two variables showed them to be positively correlated with a correlation coefficient of .3000 (P=.006). With 64.7 percent of the sample naming the pop/rock category as their preferred musical format, and the vairance in liking being significantly related to musical preference, it would appear that the choice of a pop/rock musical group for the testing conditions was appropriate for the sample studied.

When analyzing the variance between the width index and the playback formats of stereo, mono and bi-polar mono, these formats explain 21.7 percent of the variance in the width index. This amount of explained variance is significant at P = .001. Therefore, the different playback formats for the testing conditions accounted for 21.7 percent of the variance in the respondents' width index scores. This relates to the hypothesis which states:

(hypothesis) Listeners sonic dimension scores will be higher with a stereo playback format than with that of bi-polar mono or mono.

The scores for the width index were higher for the stereo conditions than for the remaining playback formats. A t-test for the difference of means on the width index scores between all respondents in the stereo and the mono conditions produced a t of 4.00 (P=.001). Also, the difference of means between bi-polar mono and mono was significant at P = .006 (T=2.97). However, the difference in the width scores between the stereo and the bi-polar mono conditions were not significant. This information, in concert with earlier stated statistics between these two variables, would seem to suggest that playback format does effect individual's perception of width. However, since the significant distinctions are between scores for either stereo or bi-polar mono and mono, the possibility exists that the number of speakers or their placement could be effecting the width index. Possibly the listener perceives width more in terms of the direction(s) from which the music is disseminated rather than from the sonic layout of the instrumentation. The remaining sonic dimension indexes were not significantly related to the playback format.

(hypothesis) Listeners will have a higher liking index for stereo playback formats than for bi-polar mono or mono playback formats.

Contingency tables were created for the liking index and the independent variable of playback format. The most frequency occurring liking index score for the stereo condition was 4 (35.1%); the mono condition was also 4 (37.5%) and the bi-polar mono conditions most frequently occurring response was 3 (42.9%). None of the statistical attempts to relate these variables, t-test, Pearson Correlation or Chi-Square, provided significant results.

(hypothesis) Listeners will be able to correctly identify more instruments when placed across a stereo spread than when all the instruments are panned to the center, as in mono.

Respondents who were given one of the stereo or bi-polar mono conditions most frequently could correctly identify three of the instruments in the mix. However, those receiving the mono conditions were evenly split between two and three correct identifications at 37.5 percent of the respondents choosing each. In a similar manner, when asked to indicate the number of voices they could hear, the majority of these in the stereo and bi-polar mono groups identified three (63.9% and 45% respectively). The mono group again was split. At 43.8 percent each between two and three voices present. In both instances, a strong concentration appears in the mono conditions around the mode, while the stereo and bi-polar mono conditions demonstrate a wider distribution acorss the entire range of options. If the bi-polar mono condition could be ignored, then suppport might be lent to the hypothesis that a stereo spread makes individual instruments more easy to identify. Since, however, there seems to be relatively little difference between the responses of those subjects in the stereo and bi-polar mono groups, a stronger possibility exsists that the number of speakers may contribute more directly to the correct identification of instruments. Respondent's identification of the number of guitars present does not assist in this analysis either. Sixty-six percent (66%) of the entire population said there were two guitars present and this figure was consistent across all three playback formats, with at least 64 percent in each identifying two. When asked to place the voices they could identify along a left to right spectrum, the subject's responses showed some difference across playback formats. The furthest left and right options were cited more by the stereo groups than by those in the mono or bi-polar mono conditions. This may suggest a comprehension of width, even though that dimension may not assist in the separation of individual instruments.

At one location in the stereo mix, a guitar effect is panned across the sonic layout from the right to the left. When asked if any instruments appeared to move from side to side, 57.6 percent of those subjects receiving the stereo condition responded yes. However, 41.7 percent of those receiving mono and 64.7 percent of those receiving bi-polar mono also responded yes. Furthermore, of those affirmative responses, only 65.9 percent could attribute that movement to a guitar sound. Other frequent responses included movement by drums or synthesizers. This could be attributed to several different occurrences. Possibly the respondent subject to the stereo condition mistook the sound of different instruments panned to opposite sides, such as different drums, for the same instrument and therefore detected movement that didn't actually occur. This possibility is quite likely considering that the mean across the entire population for correctly identified instruments was only 3.1. Another possibility is the confusion among respondents of perspective for movement. That is, the sonic placement of each instrument's sound within the sonic layout, could be confused by the average listener for movement due to the instruments relative placement.

Item 1 and 5 asked the subjects to place an "X" on a given line anchored with "left," "center" and "right," where the bass drum and the guitar solo appeared to be coming from. There seemed to be confusion among the stereo and bi-polar mono listeners as to the direction of the single speaker the signal was coming from. This might suggest confusion on the part of the respondents between the location of the instrument within the sonic layout and the location of the music as it is disseminated from the speaker.

(hypothesis) The dimensions of height, width and depth contribute to the overall size of the sound.

A size index was calculated for each respondent. A semantic differential from "large" to "small" was averaged with a seven-point Likert Scale score for the statement:

"The sound of the music seemed to fill the room."

This score was attributed to a measure of the subjects perception of the overall size of the music. Pearson Correlations between this size score and each of the sonic dimension indexes of depth, width and height were calculated. Each of these indexes correlated with the size score at a significance level of $P = .006$ or better. This would seem to indicate that the perceptions of width, depth, and height of a sound do relate to a listener's overall perception of the size of the music.

(hypothesis) Good speakers will produce higher sonic dimension index scores than poor speakers.

(hypothesis) Good speakers will produce higher liking index scores than poor speakers.

Crosstabs between the liking index and the independent varaible of speaker quality produced an insignificant Chi-Square. The most frequently occurring liking scores for both good and poor speaker quality conditions were 4 (30.6% and 44.0% respectively). Likewise, a statistically signficant relationship could not be established between the sonic dimension indexes and speaker quality.

(hypothesis) As the audio knowledge of the individual increases, the sonic dimension index scores will raise.

(hypothesis) As the audio knowledge of the individual raises, the liking index will be more positively correlated with the sonic dimensions.

No distinguishable trends occurred within the liking, width, depth and height indexes through a progression of audio knowlege. This could possibly be due to the radically different sample sizes between the three classifications. While 59 amateurs participated in the study only 11 audiophiles and 4 professionals provided their input. With the sample sizes being this varied, it is very difficult to make generalizations across categories. The highest mean scores of the depth and width indexes, however, were from the professional group, with means of 4.50 and 4.37 respectively.

(hypothesis) The addition of a video signal to an audio signal will not effect the sonic dimension index scores.

(hypothesis) The addition of a video signal to an audio signal will not effect the liking index score.

Crosstabs between the liking index of those who heard the audio while watching the video both had the highest frequency of responses at the level of 4. However, when broken down by sex, the liking index for these two categories shows an interesting trend. The most frequently occurring score among females who heard only the audio was 2 (30.0%), that for the males of the same group was 4 (38.9%). For those respondents who saw the video with the audio, the most frequent male score again was 4 (43.5%). However, the most common liking score of the females in this category was equally distributed between 3 and 4 at 27.3 percent each. Therefore, females who saw the video had a higher liking score than females who only heard the audio. Indeed, in comments among the entire population on the open ended question, those who heard only the audio cited several times a dislike toward the lead singer. Among those who saw the video, however, this response was not received at all.

While none of the statistical tests of these two hypothesis provided significant results, the trends cited here suggest a need for further research on the effect which a video in connection with the music may have on a musician's popularity.

CHAPTER VIII

CONCLUSIONS

The indexes for depth, width, and height were positively correlated at a significant level (P =.006 or better) with the size index. Therefore, there is a positive relationship between these dimension indexes and the overall size of the sound which the listener attributes to the music.

When dealing with these indexes to the independent variables, however, in relation the depth, height, and liking indexes produced were not significantly different between the playback formats of stereo, mono, and bi-polar mono. Nor were there significant differences in these indexes between the conditions of good and bad speakers, or the audio or audio/video conditions.

For the width index, however, significant differences exist between the playback formats. The width index scores of the individual respondents within each of the playback format conditions suggest that the average listener definitely views the stereo and bi-polar mono formats as wider than the mono format. As discussed earlier, however, this may be due to number and placement of the speakers rather than the sonic layout of the instrumentation. However, there was no significant distance between the liking scores of these three groups. This may suggest, that a respondent's perception of greater width does not necessarily change the degree of liking. In addition, the creation of a signal which places the various instruments across a stereo spread does not appear to make those instruments more easily identifiable. Also, speaker quality made no significant difference in the width index scores between the playback format conditions.

Furthermore, no trends emerged within the various indexes across the three levels of audio knowledge. Therefore, it cannot be inferred that an increase in audio knowledge would relate to a change in the index scores. This could, however, be related to the vastly different number of respondents in each category, as analyzed within this study.

It should therefore be concluded the sonic dimension of width is significantly effected by the playback format. The listener can identify the playback format differences, most prominently, between stereo or bi-polar mono and mono. Furthermore, there appears to be no preference, as measured by the liking index, for any one of the three playback formats. Therefore, this would suggest that the listener does not perceive more width in a stereo signal nor do they prefer stereo playback to that of bi-polar mono. If, indeed, this is the case, then decision makers dealing with new technologies, such as stereo television, should consider the marketing of bi-polar mono rather than the conversion to stereo audio.

Since the listener cannot significantly identify depth and height, possibly these dimensions should be given less emphasis by the audio producer than the dimension of width. Also, with listener distinction of width between stereo and bi-polar mono being insignificant, the audio producer may wish to reconsider the creation of a stereo mix at all. One mix, that of mono, may be the only playback format necessary to please the majority of the audience.

Further research should be conducted dealing with physical distance between playback speakers and also number of speakers used. This type of a study should take into consideration the format of quadraphonic sound in which four separate speakers are used in playback. An arrangement of mono from all four speakers, two sets of stereo and discrete quadraphonic sound could be used to further the conclusions reached within this study.

In addition, more discriminate divisions between levels of speaker quality may produce different results in relation to liking scores. Possibly a division between four or five different quality levels ranging from television speakers through home speakers to studio quality monitors would produce significant differences in the liking scores of the respondents.

Finally, further research involving larger samples from the audiophile and professional categories may produce significant results between persons. While the amateur can identify width, and sees it as a contributor to size, this does not significantly effect his/her liking of the music. However, it would be hoped, that as knowledge of the industry increases, appreciation for these created dimensions could increase the individual's liking of the song. Therefore, making the creation of the dimensions to some aesthetic gain.

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APPENDIX A

MULTI-CHANNEL AUDIO TRACK ASSIGNMENTS

TRACK #	
1	Blank
2	Kick Drum
3	Rack Tom
4	Floor Tom
5	Overhead Left
6	Overhead Right
7	Snare
8	Snare Overdub
9	Blank
10	Blank
11	Blank
12	Bass Guitar
13	Guitar #1 - Mark
14	Guitar #2 - Scott
15	Blank
16	Scratch Vocal
17	Rockman Left
18	Rockman Right
19	Danny Vocals
20	Marty Vocals
21	Scott Vocals
22	Mark Vocals
23	Blank
24	Blank

APPENDIX B

PERFORMANCE VIDEO SCRIPT

Cam	Video	Audio
(1)	Fade to CU Snare	- Hook
(2)	take CU Dan's Guitar	
(3)	take CU Scott's Guitar	
(2)	take CU Dan's Guitar	
(3)	take CU Scott's Guitar	
(2)	take CU Dan's Guitar	
(1)	take SC Dan	- First Verse
(3)	take WS Scott	
(1)	take CS Dan	
(2)	take WS Mark	
(1)	take CS Dan	
(3)	take WS Scott	
(1)	take CS Dan	
(1)	zoom out to WS Dan	
(2)	take WS Mark	- Break
(3)	take WS Paul	
(2)	take WS Mark	
(3)	2-shot Paul & Marty	
(1)	take WS Dan	
(2)	take WS Mark	
(3)	take KS Scott	- Solo

Cam	Video	Audio
(3)	zoom in to WS Scott	
(2)	take WS Mark	
(3)	take WS Scott	
(1)	take WS Dan	
(3)	take WS Scott	
(2)	take KS Mark	
(3)	take WS Marty	
(1)	WS Dan	- Hook
(2)	take KS Mark	
(1)	take WS Dan	
(3)	2-shot Marty & Paul	- Verse 2
(1)	take WS Dan	
(1)	zoom into CS Dan	
(3)	take WS Paul	
(2)	take WS Mark	
(1)	take SS Dan	
(2)	take 2-shot Dan & Scott	
(3)	take WS Scott	
(1)	take CS Dan	
(3)	take WS Scott	- Hook
(2)	take 2-shot Dan & Scott	
(3)	take WS Scott	
(2)	take 2-shot Dan & Scott	
(1)	take CS Dan	- a capella
(2)	take WS Mark	

Cam	Video	Audio
(3)	take WS Scott	
(1)	take Face Shot Dan	
	Fade to Black	

APPENDIX C

FINAL VIDEO SCRIPT

Video	Audio	Time (minutes; seconds; frames)
CU Snare Drum	- hook 1	0:00:00
CU Dan's Boot		0:03:23
CU Snare Drum		0:05:21
CU Ballerina's Slippers		0:07:07
CU Snare Drum		0:08:21
CU Dan's Guitar	- hook 2	0:10:20
CU Scott's Guitar		0:16:02
CU Dan's Guitar	- hook 3	0:18:04
FS Dan (projection room)		0:18:25
CU Dan's Guitar		0:19:17
FS Ballerina Spinning		0:20:28
CU Dan's Guitar		0:21:28
CU Scott's Guitar		0:24:08
LS Dan and Tombstone		0:25:08
CU Dan's Guitar	- hook 4	0:28:07
CS Dan	Verse 1	0:31:24
WS Scott		0:38:21
OSS Projection Room Screen		0:39:22
Ballerina Dancing in Studio		0:41:10
FS Dan in Projection Room		0:44:18
Zoom out to WS Dan		0:44:27
WS Mark		0:49:21
CS Dan		0:53:11
WS Scott		0:56:12
CU Ballerina's Feet		0:59:02
Ballerina Dancing in Studio		1:02:10
CS Dan in Projection Room		1:06:06
CS Dan		1:09:06

Final Video Script (continued)

Video	Audio	Time (minutes; seconds; frames)
Zoom Out to KS Dan		1:09:25
CS Dan in Projection Room		1:14:18
FS Ballerina		1:19:23
Ballerina Dancing in Studio		1:22:00
LS Dan in Projection Room		1:26:20
WS Paul		1:29:04
WS Mark		1:32:22
2-Shot Paul and Marty		1:37:16
KS Dan		1:41:01
CS Mark		1:44:21
WS Scott	- Solo	1:47:27
LS Dan and Tombstone		1:53:25
Zoom In and Roll Focus		1:54:09
WS Scott		1:59:09
Zoom in to CS Scott		2:01:00
WS Mark		2:04:00
CS Dan in Projection Room		2:06:05
Zoom in to CU Hand		2:10:03
WS Scott		2:18:16
WS Dan in Dance Studio		2:25:12
WS Marty		2:33:02
CS Dan by Door		2:36:16
LS Empty Dance Studio	- hook	2:38:06
CS Dan by Door		2:39:16
WS Mark		2:42:17
KS Dan in Studio	- hook	2:44:06
CS Dan		2:46:06
Rack Focus to Ballerina		2:46:13
CU Ballerina Slippers		2:49:22
LS Ballerina Tying Slippers	- Verse 2	2:51:10
OSS Projection Room of Screen		2:52:07
WS Dan		2:55:12

Final Video Script (continued)

Video	Audio	Time (minutes; seconds; frames)
CS Dan Smoking		3:05:04
WS Paul		3:10:10
WS Dan in Projection Room		3:14:02
Zoom In and Roll Focus		3:14:05
Ballerina Dancing in Studio		3:15:23
CU Dan's Eyes		3:20:22
Ballerina Dancing in Studio		3:21:05
CS Dan		3:23:17
2-Shot Dan and Scott		3:30:07
WS Scott		3:33:26
KS of Dan in Studio	- hook	3:35:19
WS Dan in Projection Room		3:41:28
Ribbon Blowing Off Tombstone		3:44:21
LS Dan in Projection Room	- hook	3:47:05
LS Dan Leaving Projection Room		3:49:15
CS Dan	- a ccapela	3:53:16
LS Dan by Parking Ramp		3:58:19
WS Woman Leaving Ramp		4:05:00
Zoom to FS Woman		4:06:08
KS Entering Ramp		4:08:00
LS Woman and Dan Passing Each Other		4:08:27
FS Dan		4:10:04
Freeze Frame		4:10:23
Fade to Black		4:13:04

APPENDIX D

LIST OF TESTING CONDITIONS AND NUMBER OF SUBJECTS

Amateurs

Stereo - Good Speakers - Audio Only	(8)
Stereo - Bad Speakers - Audio Only	(6)
Stereo - Good Speakers - Audio/Video	(8)
Stereo - Bad Speakers - Audio/Video	(4)
Mono - Good Speakers - Audio Only	(5)
Mono - Bad Speakers - Audio Only	(4)
Mono - Good Speakers - Audio/Video	(3)
Mono - Bad Speakers - Audio/Video	(4)
Bi-Polar Mono - Good Speakers - Audio Only	(6)
Bi-Polar Mono - Bad Speakers - Audio Only	(4)
Bi-Polar Mono - Good Speakers - Audio/Video	(4)
Bi-Polar Mono - Bad Speakers - Audio/Video	(3)

Audiophiles

Stereo - Good Speakers - Audio Only	(3)
Stereo - Good Speakers - Audio/Video	(4)
Bi-Polar Mono - Good Speakers - Audio Only	(4)

Professionals

Stereo - Good Speakers - Audio/Video	(4)

APPENDIX E

FINAL QUESTIONNAIRE

Appendix E contains a copy of the final questionnaire to the groups seeing the video along with the audio. The questionnaire administered to the groups which heard the audio without the video was identical to this one, except that questions 19 and 20 were omitted.

The study you are participating in deals with the various dimensions of music, and listener perceptions of those dimensions.

Your thoughtful response is appreciated.

Thank you for your time.

Classification Number __________

Respondent Number ______________

1. Sounds often appear as if they are coming from different locations, as if the band were standing in front of you. On the line provided, please place an "X" at the place where the bass drum sounds like it is located.

LEFT CENTER RIGHT

2. Please list the instruments that you can identify in this song.

3. How many guitars do you hear? ____________

4. Music creates a feeling of depth. Various instruments or voices seem further from you than others. With the lead singer as point of reference, please mark the column signifying whether each instrument is closer than the lead singer or further from you than the lead singer.

	CLOSER	FURTHER
a. bass guitar	_____	_____
b. cymbols	_____	_____
c. drums	_____	_____

5. If the band were standing in front of you performing, place an "X" on the line provided at the location where you hear the guitar solo coming from.

LEFT CENTER RIGHT

6. How many voices do you hear? _____

7. Place an "X" on the line provided where you hear each of these voices. You may use as many "X's" as necessary.

LEFT CENTER RIGHT

8. Do any of the instruments appear as if they move from side to side?

    _____ Yes

    _____ No

    _____ Don't Know

    If you answered yes, which instrument(s)?

    ____

9. Music creates a feeling of <u>height</u>. Considering the entire song, place an "X" indicating how far toward the ceiling, or how tall the music feels to be.

CEILING

FLOOR

10. If the fullest <u>depth</u> the music could exhibit were the distance between you and the far wall, place an "X" on the line provided which indicates how much depth this song has.

WALL

YOU

11. Music creates a feeling of <u>width</u>. Given the two side walls of the room as boundaries, please <u>darken</u> in the portion of the line which you feel represents the width of this song. (For example, a very narrow sound would have a small length of the line darkened in, probably toward the center, while a wide sound would have a larger portion of the line darkened.)

WALL ____ WALL

12. For each of the following pairs, please place an "X" on the line which most appropriately expresses your response to the statement:

The sound of the music seemed to be:

a. ____ ____ ____ ____ ____ ____ ____
TALL SHORT

b. ____ ____ ____ ____ ____ ____ ____
WIDE NARROW

c. ____ ____ ____ ____ ____ ____ ____
SHALLOW DEEP

d. ____ ____ ____ ____ ____ ____ ____
LARGE SMALL

PLEASE RESPOND TO THE FOLLOWING QUESTIONS BY CIRCLING THE NUMBER MOST APPROPRIATE FOR YOUR RESPONSE.

	STRONGLY AGREE						STRONGLY DISAGREE
13. I liked the song I just heard.	1	2	3	4	5	6	7
14. I liked the sound of the group playing the song.	1	2	3	4	5	6	7
15. I would like to hear more songs by this group.	1	2	3	4	5	6	7
16. Given that money was not a factor, I would buy a record containing this song.	1	2	3	4	5	6	7
17. I would <u>NOT</u> like this song played on my favorite radio station.	1	2	3	4	5	6	7
18. I felt the sound quality of the music could have been improved.	1	2	3	4	5	6	7
19. I liked the video I just saw.	1	2	3	4	5	6	7
20. The video story was <u>NOT</u> related to the words in the song.	1	2	3	4	5	6	7

	STRONGLY AGREE						STRONGLY DISAGREE
21. The volume of the music was too loud.	1	2	3	4	5	6	7
22. The sound of the music seemed to fill the room.	1	2	3	4	5	6	7

23. In general, what did you think of the song you just heard? What aspects of the recording did you like and what didn't you like?

24. The type of music I listen to most often is (Please choose one).

_____ Pop/Rock
_____ Heavy Metal Rock
_____ Easy Listening
_____ Country
_____ Jazz
_____ Soul
_____ Classical
_____ Folk
_____ Other

25. Approximately how many hours each day do you spend listening to music?

_____ Hours

26. Please divide your music listening time among this following categories--use percentages.

_____ Records
_____ Tapes
_____ Radio
_____ TV
_____ Live

(Must total 100%)

27. Approximately how many albums and tapes do you have in your personal music collection?

_____ Albums

_____ Tapes

28. What is your age? _________ Years

29. What is your sex? _____ Male

_____ Females

30. From participating in this study, do you feel that in the future you will listen to music in a different way? If so, how?

APPENDIX F

SELECTED STATISTICAL RESULTS

Key

LIKING	Liking Index Score
WDINDEX	Width Index Score
DPINDEX	Depth Index Score
HGINDEX	Height Index Score
FORMAT	Listener's Musical Preference
SMB	Playback Format--Stereo, Mono, or Bi-Polor Mono
SZINDEX	Size Index

* * * * * * * * * A N A L Y S I S O F V A R I A N C E * * * * * * * * *

LIKING
BY FORMAT

SOURCE OF VARIATION	SUM OF SQUARES	DF	MEAN SQUARE	F	SIGNIF OF F
MAIN EFFECTS	19.450	6	3.242	2.509	.033
FORMAT	19.450	6	3.242	2.509	.033
EXPLAINED	19.450	6	3.242	2.509	.033
RESIDUAL	69.782	54	1.292		
TOTAL	89.231	60	1.487		

* * * * * * * * * A N A L Y S I S O F V A R I A N C E * * * * * * * * *

WDINDEX
BY SMB

SOURCE OF VARIATION	SUM OF SQUARES	DF	MEAN SQUARE	F	SIGNIF OF F
MAIN EFFECTS	20.532	2	10.266	9.843	.001
SMB	20.532	2	10.266	9.843	.001
EXPLAINED	20.532	2	10.266	9.843	.001
RESIDUAL	74.053	71	1.043		
TOTAL	94.584	73	1.296		

P E A R S O N C O R R E L A T I O N C O E F F I C I E N T S

	FORMAT
LIKING	.3000 (68) P= .006

P E A R S O N C O R R E L A T I O N C O E F F I C I E N T S

	WDINDEX	HGINDEX	DPINDEX
SZINDEX	.2920 (74) P= .006	.2898 (74) P= .006	.3320 (74) P= .002

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GROUP 1 - Stereo
GROUP 2 - Mono

VARIABLE	NUMBER OF CASES	MEAN	STANDARD DEVIATION	STANDARD ERROR
WDINDEX				
GROUP 1	37	3.8108	.945	.155
GROUP 2	16	2.4688	1.190	.297

		POOLED VARIANCE ESTIMATE			SEPARATE VARIANCE ESTIMATE		
F VALUE	2-TAIL PROB.	T VALUE	DEGREES OF FREEDOM	2-TAIL PROB.	T VALUE	DEGREES OF FREEDOM	2-TAIL PROB.
1.58	.256	4.38	51	.000	4.00	23.58	.001

- - - - - - - - - - - - - - - - T - T E S T - - - - - - - - - - - - - - - -

GROUP 1 - Stereo
GROUP 2 - Bi-Polar Mono

VARIABLE WDINDEX	NUMBER OF CASES	MEAN	STANDARD DEVIATION	STANDARD ERROR
GROUP 1	37	3.8108	.945	.155
GROUP 2	21	3.5714	1.016	.222

		POOLED VARIANCE ESTIMATE			SEPARATE VARIANCE ESTIMATE		
F VALUE	2-TAIL PROB.	T VALUE	DEGREES OF FREEDOM	2-TAIL PROB.	T VALUE	DEGREES OF FREEDOM	2-TAIL PROB.
1.15	.688	.90	56	.371	.88	39.23	.382

- - - - - - - - - - - - - - - - T - T E S T - - - - - - - - - - - - - - - -

GROUP 1 - Bi-Polar Mono
GROUP 2 - Mono

VARIABLE WDINDEX	NUMBER OF CASES	MEAN	STANDARD DEVIATION	STANDARD ERROR
GROUP 1	21	3.5714	1.016	.222
GROUP 2	16	2.4688	1.190	.297

		POOLED VARIANCE ESTIMATE			SEPARATE VARIANCE ESTIMATE		
F VALUE	2-TAIL PROB.	T VALUE	DEGREES OF FREEDOM	2-TAIL PROB.	T VALUE	DEGREES OF FREEDOM	2-TAIL PROB.
1.37	.502	3.04	35	.004	2.97	29.47	.006
